User's Manual

AudioCodes Mediant™ Family of Media Gateways & Session Border Controllers

Mediant 500L

Enterprise Session Border Controller (E-SBC) & Media Gateway

Version 7.0

audiocodes

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Notice

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This document is subject to change without notice.

Date Published: September-12-2018

WEEE EU Directive

Pursuant to the WEEE EU Directive, electronic and electrical waste must not be disposed of with unsorted waste. Please contact your local recycling authority for disposal of this product.

Customer Support

Customer technical support and services are provided by AudioCodes or by an authorized AudioCodes Service Partner. For more information on how to buy technical support for AudioCodes products and for contact information, please visit our Web site at https://www.audiocodes.com/services-support/maintenance-and-support.

Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.

Throughout this manual, unless otherwise specified, the term *device* refers to Mediant 500L Gateway & E-SBC.

Related Documentation

Manual Name
SIP CPE Release Notes
Mediant 500L Gateway & E-SBC Hardware Installation Manual
Complementary Guides
CLI Reference Guide
CPE Configuration Guide for IP Voice Mail
SNMP User's Guide
SBC Design Guide
Recommended Security Guidelines Configuration Note
SIP Message Manipulations Quick Reference Guide
Utility Guides
INI Viewer & Editor Utility User's Guide

Manual Name
DConvert User's Guide
AcBootP Utility User's Guide
CLI Wizard User's Guide

Notes and Warnings

Note: The device is an indoor unit and therefore, must be installed only **INDOORS**. In addition, Ethernet port interface cabling must be routed only indoors and must not exit the building.

Note: The scope of this document does not fully cover security aspects for deploying the device in your environment. Security measures should be done in accordance with your organization’s security policies. For basic security guidelines, refer to AudioCodes *Recommended Security Guidelines* document.

Note: Throughout this manual, unless otherwise specified, the term *device* refers to your AudioCodes product.

Note: Before configuring the device, ensure that it is installed correctly as instructed in the *Hardware Installation Manual*.

Notes:

- This device includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit (http://www.openssl.org/).
- This device includes cryptographic software written by Eric Young (eay@cryptsoft.com).

Note: Some of the features listed in this document are available only if the relevant Software License Key has been purchased from AudioCodes and installed on the device. For a list of Software License Keys that can be purchased, please consult your AudioCodes sales representative.

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Document Revision Record

LTRT	Description
10520	Initial document release for Version 7.0.
10521	New sections: Enabling LDAP Searches for Numbers with Characters New parameters: LdapConfiguration_VerifyCertificate; GW Group Registered IP Address; GW Group Registered Status; PREFIX_ForkingGroup; CDRLocalMaxNomOfFiles; CDR Syslog Sequence Number; HAUnitIdName; HARemoteAddress; HARevertiveEnabled; HAPriority; PublicationIPGroupID; LDAPNumericAttributes; FeatureKeyURL G.727 removed; NFS removed; ISDN BRI North American variants (partial support); Web browser requirement for Web interface Parameters removed: SRD_SBCRegisteredUsersClassificationMethod; IpProfile_RemoteBaseUDPPort Updated parameter descriptions: TLSContexts_TLSVersion; DeviceTable_Tagging; BaseUDPPort; HTTPRemoteServices_Policy; HTTPRemoteServices_VerifyCertificate; HTTPInterface_VerifyCert; HTTPProxyHost_VerifyCert; CpMediaRealm_PortRangeEnd; CpMediaRealm_PortRangeStart; SIPInterface_UDPPort; IPGroup_Type; IPGroup_SIPGroupName; IPGroup_ClassifyByProxySet; IPGroup_InboundManSet; IPGroup_OutboundManSet; IPGroup_SBCPSAPMode; MessageManipulations_RowRole; CodersGroup0_CoderSpecific; CodersGroupX_CoderSpecific; Classification_SrcAddress; IP2IPRouting_GroupPolicy; BaseUDPport; RTCPXREscIP; MinOverlapDigitsForRouting; ISDNTxOverlap; ISDNRxOverlap; SBCUserRegistrationGraceTime; ProtocolType; Coder payload types Updated sections: Configuring Underlying Ethernet Devices; Configuring RTP Base UDP Port; Enabling the E9-1-1 Feature; Configuring IP Groups (matching methodology); Configuring SIP Message Manipulation; Configuring Source/Destination Number Manipulation Rules; SIP Calling Name Manipulations; Configuring Redirect Number IP to Tel; Configuring Tel-to-IP Routing Rules; Configuring IP-to-Trunk Group Routing Rules; Configuring Classification Rules; Configuring SBC IP-to-IP Routing; Network Topology Types and Rx/Tx Ethernet Port Group Settings; Initial HA Configuration; Configuring RTCP XR; Configuring CDR Reporting; Customizing CDRs for SBC Calls; Storing CDRs on the Device; Configuring RADIUS Accounting; Interworking ISDN Overlap Dialing with SIP According to RFC 3578
10524	Updated sections: Accessing the Web Interface (autocomplete); Advanced User Accounts Configuration (password); CLI-Based Management; Understanding Configuration Modes (enable command); Configuring TLS Certificate Contexts (TLS version); Configuring Underlying Ethernet Devices (max.); First Incoming Packet Mechanism (NAT by Signaling); No-Op Packets (note); Configuring the Device's LDAP Cache; Centralized Third-Party Routing Server or ARM (ARM added); Configuring the SEM Server (port removed); Configuring IP Groups (note); Configuring Tel-to-IP Routing Rules (note); Configuring IP-to-Trunk Group Routing Rules (notes); BRI Call Forwarding; Feature List; Registration Refreshes; Direct Media; Configuring Message Condition Rules; Configuring SBC IP-to-IP Routing; Obtaining IP Destination from Dial Plan File (note); Configuring SBC User Info Table in Loadable Text File; Software Upgrade Wizard (power source); Automatic Provisoining (CLI Script); MAC Address Placeholder in Configuration File Name; Configuring RTCP XR; Storing CDRs on the Device (FTP removed); Maintaining Same Syslog SID/BID over Multiple Devices (removed); Session Capacity per Configuration. New sections: Refreshing the LDAP Cache; Clearing the LDAP Cache; Utilizing Gateway Channel Resources for SBC; Configuring Dial Plans; Importing and

LTRT	Description
	Exporting Dial Plans; Creating Dial Plan Files; Using Dial Plan Tags for IP-to-IP Routing; Using Dial Plan Tags for Outbound Manipulation; File Template for Automatic Provisioning; Configuring PacketSmart for Network Monitoring; PacketSmart Parameters. Updated parameters: TLSContexts_TLSVersion; CallSetupRules_ActionType (enums); SIPInterface_SBCDirectMedia; IpProfile_DisconnectOnBrokenConnection; IPGroup_ProxySetName (note); IpProfile_MediaIPVersionPreference; IpProfile_SBCAllowedMediaTypes; IPProfile_SBCDirectMediaTag; ForwardOnBusyTrunkDest_ForwardDestination; IP2IPRouting_SrcUsernamePrefix; IP2IPRouting_DestUsernamePrefix; IP2IPRouting_Trigger; IPOutboundManipulation_SrcUsernamePrefix; SBCCDRFormat_Title; CLIPrivPass; NATMode; SendAcSessionIDHeader (removed); QOEPort (removed); SIPChallengeCachingMode; MaxGeneratedRegistersRate; RejectCancelAfterConnect; RTCPXRESCTransportType (removed); RTCPXREscIP (removed); RTCPXRReportMode; PublicationIPGroupID; DisconnectOnBrokenConnection; SBCUserRegistrationGraceTime; MSLDAPOCSNumAttributeName; GWRoutingServer. New parameters: IP2IPRouting_SrcTags; IP2IPRouting_DestTags; IPOutboundManipulation_SrcTags; IPOutboundManipulation_DestTags; WebLoginBlockAutoComplete; EnforcePasswordComplexity; AUPDCliScriptURL; TemplateUrl; AupdFilesList; PacketSmartAgentMode; PacketSmartIpAddress; PacketSmartIpAddressPort; PacketSmartMonitorInterface; PacketSmartNetworkInterface; GeneratedRegistersInterval; MaxCallDuration; UseFacilityInRequest; TrunkLifeLineType; DialPlans; DialPlanRule.
10527	▪ Updated sections: Configuring Proxy Sets; BRI Call Forwarding; Configuring Multi-Line Extensions and Supplementary Services; Pre-Configured IP Groups; Normal Mode; Emergency Mode; Auto Answer to Registrations; Auto Answer to Registrations; Software License Key; Viewing Device Information; Viewing Proxy Set Status; Configuring CDR Reporting; Configuring RADIUS Accounting; Technical Specifications ▪ New sections: Customizing the Web Interface; Local Handling of BRI Call Forwarding • Updated parameters: TLSContexts_ServerCipherString; TLSContexts_ClientCipherString; NATTranslation_SourceStartPort; NATTranslation_SourceEndPort; NATTranslation_TargetStartPort; NATTranslation_TargetEndPort ; IPOutboundManipulation_PrivacyRestrictionMode; DialPlans_Name; DialPlanRule_RuleIndex; DialPlanRule_Name; UserInfoFileURL; EnableCoreDump; ProxySet; PrackMode; SessionExpiresDisconnectTime; ISDNSuppServ; RADIUSRetransmission; RadiusTO ▪ New parameters: ProxySet_SuccessDetectionRetries; ProxySet_SuccessDetectionInterval; ProxySet_FailureDetectionRetransmissions and ProxySet_MinActiveServersLB; ISDNSuppServ_CFB2PhoneNumber; ISDNSuppServ_CFNR2PhoneNumber; ISDNSuppServ_CFU2PhoneNumber; ISDNSuppServ_NoReplyTime; WebFaviconFileUrl]; EnableNonCallCdr; BRICallForwardHandling ▪ Removed parameters: WelcomeMessage
10529	▪ Updates sections: CLI; Configuring Web User Accounts (TYPO); Configuring TLS Certificate Contexts; Assigning CSR-based Certificates to TLS Contexts (SHAI); Generating Private Keys for TLS Contexts (4096); Configuring IP Network Interfaces (NOTE); Configuring Firewall Settings; Assigning IDS Policies; Configuring LDAP Servers (max); Configuring Call Setup Rules; Fixed Mapping of SIP Response to ISDN Release Reason; Fixed Mapping of ISDN Release Reason to SIP Response; Alternative Routing Based on IP Connectivity; Alternative Routing Based on SIP Responses; Configuring SIP Response Codes for

LTRT	Description
	Alternative Routing Reasons; Configuring Dial Plans (priority); Creating Core Dump and Debug Files upon Device Crash (reset); Configuring DTMF Tones for Test Calls; Configuring Basic Test Call; Configuring SBC Test Call with External Proxy (removed) ▪ Updated parameters: TLSContexts_ServerCipherString; TLSContexts_ClientCipherString; AccessList_Start_Port; AccessList_End_Port; SIPInterface_InterfaceName; ProxySet_ProxyName; MessageManipulations_ManipulationName; MessagePolicy_Name; TelProfile_ProfileName; IpProfile_SBCUseSilenceSupp; [_ManipulationName; PREFIX_RouteName; PstnPrefix_RouteName; GWRoutingPolicy_Name; SBCAdmissionControl_AdmissionControlName; SBCAdmissionControl_Rate; Classification_ClassificationName; IP2IPRouting_RouteName; SBCRoutingPolicy_Name; IPInboundManipulation_ManipulationName; IPOutboundManipulation_ManipulationName; Test_Call_Play (tone type); EnableWebAccessFromAllInterfaces; ResetWebPassword; DisableSNMP; KeepAliveTrapPort (default); SBCtestID (removed); EnableCoreDump; SSHMaxLoginAttempts; IgnoreAlertAfterEarlyMedia; EnableBusyOut; ECNLPMode; ISDNInCallsBehavior; SecureCallsFromIP; AltRoutingTel2IPEnable; ProxySet_IsProxyHotSwap; IpProfile_SBCUseSilenceSupp (removed); EnablePChargingVector; ProtocolType (BRI IUA removed); ▪ New parameters: TLSContexts_DTLSVersion; TLSContexts_DHKeySize; CustomerSN; TLSContexts_ServerCipherString
10534	▪ Updated sections: Disabling Enabling SNMP; Configuring NAT Translation per IP Interface; Silence Suppression (removed); Comfort Noise Generation; Configuring Media (SRTP) Security; SIP-based Media Recording (France URL); Configuring SIP Recording Rules (timestamp); Enabling LDAP Searches for Numbers with Characters; Configuring Call Setup Rules; Configuring Media Realm Extensions; Configuring SIP Message Manipulation (max.); Interworking SIP Early Media; Configuring Call Preemption for SBC Emergency Calls (note); DHCP-based Provisioning (note); Automatic Update from Remote Servers; Viewing Active Alarms (note); Configuring CDR Reporting (note); Configuring RADIUS Accounting (figure) ▪ Updated parameters: MediaRealmExtension_IPv4IF; MediaRealmExtension_IPv6IF; SRD_SBCOperationMode; ProxySet_ProxyName; ProxySet_EnableProxyKeepAlive; IpProfile_SCE (removed); IpProfile_SBCPlayHeldTone; IpProfile_SBCSDPPtimeAnswer (Preferred Value); IpProfile_SBCPreferredPTime; PstnPrefix_SourceAddress; TelnetServerEnable; DisableSNMP; EnableLanWatchDog (removed); SyslogOptimization (default): IsCiscoSCEMode; EnableBusyOut; EnableSilenceCompression (removed); EnablePChargingVector ; FaxBypassPayloadType (range); removed – EnableSilenceDisconnect / FarEndDisconnectSilencePeriod / FarEndDisconnectSilenceMethod / FarEndDisconnectSilenceThreshold / BrokenConnectionDuringSilence; TrunkLifeLineType; BriTEIAssignTrigger; BriTEIRemoveTrigger; UseDisplayNameAsSourceNumber; SBCKeepContactUserinRegister ▪ New parameters: CallSetupRules_QueryTarget; IpProfile_SBCAdaptRFC2833BWToVoiceCoderBW; ActiveAlarmTableMaxSize; NoAlarmForDisabledPort; SBCRemoveSIPSFromNonSecuredTransport; TimeZoneFormat; SRTPTunnelingValidateRTPRxAuthentication; SRTPTunnelingValidateRTCPRxAuthentication; HookFlashFromMediaIP; SBCRemoveSIPSFromNonSecuredTransport; SIPRecTimeStamp; TimeZoneFormat

LTRT	Description
10537	▪ Updated sections: Gateway Application; Fax / Modem NSE Mode; Configuring Call Setup Rules; Configuring IP-to-Trunk Group Routing Rules; Interworking Media Security Protocols; Configuring Call Preemption for SBC Emergency Call; Event Representation in Syslog Messages. ▪ Updated parameters: Mode; Rules Set ID; Silence Suppression; Media Security Method; Request Type; Play; Telnet Server Idle Timeout; IsCiscoSCEMode

Documentation Feedback

AudioCodes continually strives to produce high quality documentation. If you have any comments (suggestions or errors) regarding this document, please fill out the Documentation Feedback form on our Web site at https://online.audiocodes.com/documentation-feedback.

1 Introduction

This User's Manual is intended for the professional person responsible for installing, configuring and managing the AudioCodes product (hereafter, referred to as *device*). The document provides the information you need to configure and manage the device.

1.1 Product Overview

The Mediant 500L Gateway and Enterprise Session Border Controller (hereafter referred to as device) is a member of AudioCodes family of enterprise session border controllers (E-SBC), enabling connectivity and security between small medium businesses (SMB) and service providers' VoIP networks. The device provides voice-over-IP (VoIP) SBC functionality. The device offers enhanced dialing plans and voice routing capabilities along with SIP-to-SIP mediation, allowing enterprises to implement SIP Trunking services and IP-based Unified Communications.

The device is designed as a secured VoIP platform. A fully featured enterprise-class SBC provides a secured voice network deployment based on a Back-to-Back User Agent (B2BUA) implementation. The SBC functionality provides perimeter defense for protecting the enterprise from malicious VoIP attacks; mediation for allowing the connection of any PBX and/or IP PBX to any service provider; and service assurance for service quality and manageability.

The device offers call survivability solutions, ensuring service continuity to enterprises served by a centralized SIP-based IP-Centrex server or branch offices of distributed enterprises. Call survivability enables internal office communication between SIP clients in the case of disconnection from the centralized SIP IP-Centrex server or IP-PBX.

The device supports the following interfaces:

- Four Fast Ethernet (10/100Base-T) LAN ports
- Up to four ISDN BRI port interfaces, supporting up to eight voice channels as well as PSTN fallback
- One USB port for optional, USB storage services
- Serial console port (RJ-45) for device management

The device supports local and remote management through various management platforms such as an HTTP/S-based Web server, a command-line interface (CLI), SNMP, and serial (RS-232).

Note: For maximum call capacity figures, see "SBC and DSP Channel Capacity" on page 1033.

1.2 Typographical Conventions

This document uses the following typographical conventions to convey information:

Table 1-1: Typographical Conventions

Convention	Description	Example
Boldface font	▪ Buttons in the Web interface. ▪ Optional parameter values in the Web interface. ▪ Navigational path in the Web interface. ▪ Toolbar buttons in the Web interface.	Click the **Add** button.
Text enclosed by double apostrophe (" ")	Text that you need to type.	Enter the value "10.10.1.1".
Courier font	CLI commands.	At the prompt, type the following: `# configure system`
Text enclosed by square brackets ([ ])	Ini file parameter.	Configure the [GWDebugLevel] parameter to 1.
Text enclosed by single apostrophe (' ')	Web parameters.	From the 'Debug Level' drop-down list, select **Basic**.
⚠	Notes highlight important or useful information.	-
⚡	Warnings alert you to potentially serious problems if a specific action is not taken.	-

1.3 Getting Familiar with Configuration Concepts and Terminology

Before using your device, it is recommended that you familiarize yourself with the basic configuration concepts and terminology. An understanding of the basic concepts and terminology will help you configure and manage your device more effectively and easily.

1.3.1 SBC Application

The objective of your configuration is to enable the device to forward calls between telephony endpoints in the SIP-based Voice-over-IP (VoIP) network. The endpoints (SIP entities) can be servers such as SIP proxy servers and IP PBXs, or end users such as IP phones. In the SIP world, the endpoints are referred to as SIP user agents (UA). The UA that initiates the call is referred to as the user agent client (UAC); the UA that accepts the call is referred to as the user-agent server (UAS).

The following table describes the main configuration concepts and terminology.

Table 1-2: Configuration Concepts and Terminology

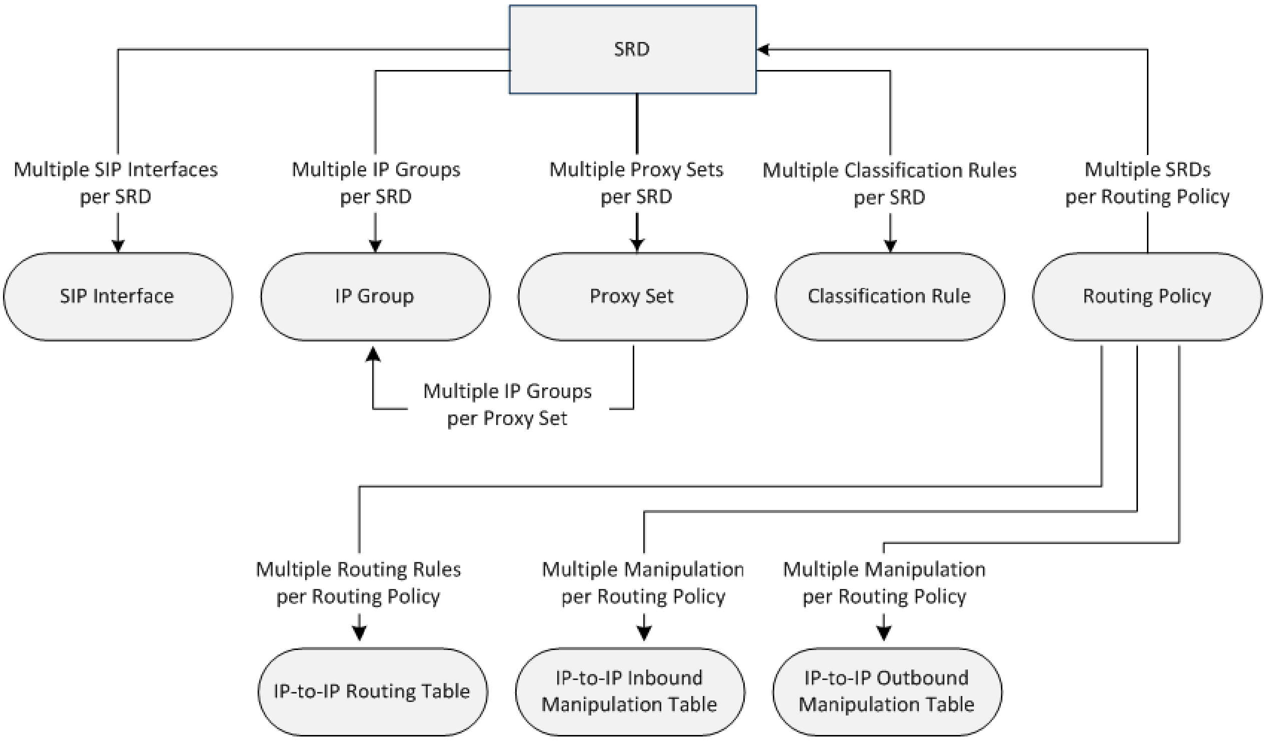
Configuration Terms	Description
IP Group	The IP Group is a logical representation of the SIP entity (UA) with which the device receives and sends calls. The SIP entity can be a server (e.g., IP PBX or SIP Trunk) or it can be a group of users (e.g., LAN IP phones). For servers, the IP Group is typically used to define the address of the entity (by its associated Proxy Set). IP Groups are used in IP-to-IP routing rules to denote the source and destination of the call.
Proxy Set	The Proxy Set defines the actual address (IP address or FQDN) of SIP entities that are servers (e.g., IP PBX). As the IP Group represents the SIP entity, to associate an address with the SIP entity, the Proxy Set is assigned to the IP Group. You can assign the same Proxy Set to multiple IP Groups (belonging to the same SRD).
SIP Interface	The SIP Interface represents a Layer-3 network. It defines a local listening port for SIP signaling traffic on a local, logical IP network interface. The term *local* implies that it's a logical port and network interface on the device. The SIP Interface is used to receive and send SIP messages with a specific SIP entity (IP Group). Therefore, you can create a SIP Interface for each SIP entity in the VoIP network with which your device needs to communicate. For example, if your VoIP network consists of three SIP entities -- a SIP Trunk, a LAN IP PBX, and remote WAN users -- a SIP Interface can be created for each of these Layer-3 networks. The SIP Interface is associated with the SIP entity, by assigning it to an SRD that is in turn, assigned to the IP Group of the SIP entity.
Media Realm	The Media Realm defines a local UDP port range for RTP (media) traffic on any one of the device's logical IP network interfaces. The Media Realm is used to receive and send media traffic with a specific SIP entity (IP Group). The Media Realm can be associated with the SIP entity, by assigning the Media Realm to the IP Group of the SIP entity, or by assigning it to the SIP Interface associated with the SIP entity.

Configuration Terms	Description
SRD	The SRD is a logical representation of your entire SIP-based VoIP network (Layer 5) containing groups of SIP users and servers. The SRD is in effect, the foundation of your configuration to which all other previously mentioned configuration entities are associated. For example, if your VoIP network consists of three SIP entities -- a SIP Trunk, a LAN IP PBX, and remote WAN users -- the three SIP Interfaces defining these Layer-3 networks would all assigned to the same SRD. Typically, only a single SRD is required and this is the recommended configuration topology. As the device provides a default SRD, in a single SRD topology, the device automatically assigns the SRD to newly created configuration entities. Thus, in such scenarios, there is no need to get involved with SRD configuration. Multiple SRDs are required only for multi-tenant deployments, where it "splits" the device into multiple logical devices. For multiple SRDs, the SRD can be configured with a Sharing Policy. The Sharing Policy simply means whether the SRD's resources (SIP Interfaces, IP Groups, and Proxy Sets) can be used by other SRDs. For example, if all tenants route calls with the same SIP Trunking service provider, the SRD of the SIP Trunk would be configured as a *Shared* Sharing Policy. SRDs whose resources are not shared, would be configured with an *Isolated* Sharing Policy.
IP Profile	The IP Profile is an optional configuration entity that defines a wide range of call settings for a specific SIP entity (IP Group). The IP Profile includes signaling and media related settings, for example, jitter buffer, voice coders, fax signaling method, SIP header support (local termination if not supported), and media security method. The IP Profile is in effect, the interoperability "machine" of the device, enabling communication between SIP endpoints that "speak" different call "languages". The IP Profile is associated with the SIP entity, by assigning the IP Profile to the IP Group of the SIP entity.
Classification	Classification is the process that identifies the incoming call (SIP dialog request) as belonging to a specific SIP entity (IP Group). There are three chronological classification stages, where each stage is done only if the previous stage fails. The device first attempts to classify the SIP dialog by checking if it belongs to a user that is already registered in the device's registration database. If this stage fails, the device checks if the source IP address is defined for a Proxy Set and if yes, it classifies it to the IP Group associated with the Proxy Set. If this fails, the device classifies the SIP dialog using the Classification table, which defines various characteristics of the incoming dialog that if matched, classifies the call to a specific IP Group. The main characteristics of the incoming call is the SIP Interface that is associated with the SRD for which the Classification rule is configured.
IP-to-IP Routing	IP-to-IP routing rules define the routes for routing calls between SIP entities. As the SIP entities are represented by IP Groups, the routing rules typically employ IP Groups to denote the source and destination of the call. For example, to route calls from the IP PBX to the SIP Trunk, the routing rule can be configured with the IP PBX as the source IP Group and the SIP Trunk as the destination IP Group. Instead of IP Groups, various other source and destination methods can be used. For example, the source can be a source host name while the destination can be an IP address or based on an LDAP query.

Configuration Terms	Description
IP-to-IP Inbound and Outbound Manipulation	IP-to-IP inbound and outbound manipulation lets you manipulate the user part of the SIP URI in the SIP message for a specific entity (IP Group). Inbound manipulation is done on messages received from the SIP entity; outbound manipulation is done on messages sent to the SIP entity. Inbound manipulation lets you manipulate the user part of the SIP URI for source (e.g., in the SIP From header) and destination (e.g., in the Request-URI line) in the incoming SIP dialog request. Outbound manipulation lets you manipulate the user part of the Request-URI for source (e.g., in the SIP From header) or destination (e.g., in the SIP To header) or calling name, in outbound SIP dialog requests. The IP-to-IP inbound and outbound manipulation are associated with the SIP entity, by configuring the rules with incoming characteristics such as source IP Group and destination host name. The manipulation rules are also assigned an SBC Routing Policy, which in turn, is assigned to IP-to-IP routing rules. As most deployments require only one SBC Routing Policy, the default Routing Policy is automatically assigned to the manipulation rules and to the routing rules.
SBC Routing Policy	SBC Routing Policy logically groups routing and manipulation (inbound and outbound) rules to a specific SRD. It also enables Least Cost Routing (LCR) for routing rules and associates an LDAP server for LDAP-based routing. However, as multiple Routing Policies are required only for multi-tenant deployments, for most deployments only a single Routing Policy is required. When only a single Routing Policy is required, handling of this configuration entity is not required as a default Routing Policy is provided, which is automatically associated with all relevant configuration entities.
Call Admission Control	Call Admission Control (CAC) lets you configure the maximum number of permitted concurrent calls (SIP dialogs) per IP Group, SIP Interface, SRD, or user.
Accounts	Accounts are used to register or authenticate a "served" SIP entity (e.g., IP PBX) with a "serving" SIP entity (e.g., a registrar or proxy server). The device does this on behalf of the "served" IP Group. Authentication (SIP 401) is typically relevant for INVITE messages forwarded by the device to a "serving" IP Group. Registration is for REGISTER messages, which are initiated by the device on behalf of the "serving" SIP entity.

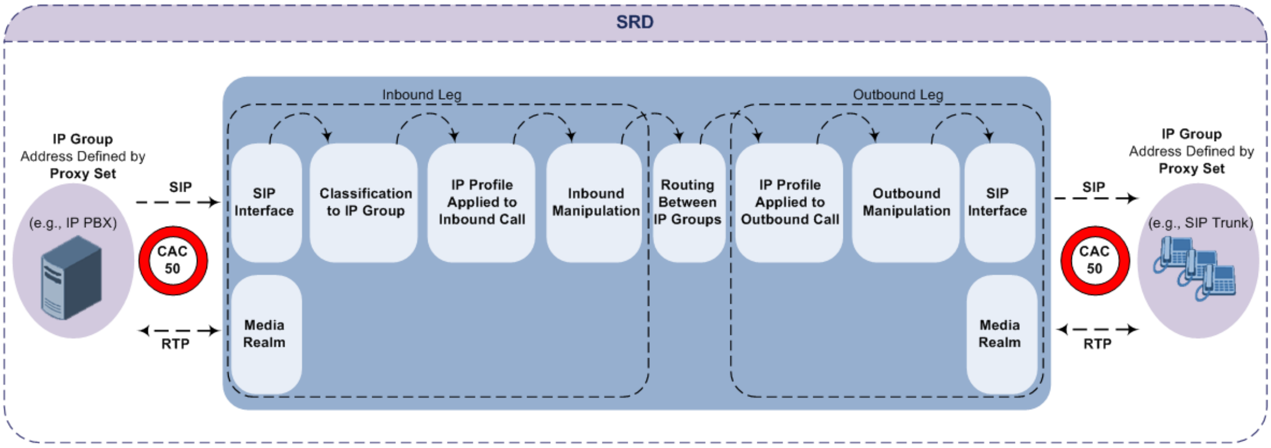
The associations between the configuration entities are summarized in the following figure:

Figure 1-1: Association of Configuration Entities


SRD
Multiple SIP Interfaces per SRD
Multiple IP Groups per SRD
Multiple Proxy Sets per SRD
Multiple Classification Rules per SRD
Multiple SRDs per Routing Policy
SIP Interface
IP Group
Proxy Set
Classification Rule
Routing Policy
Multiple IP Groups per Proxy Set
Multiple Routing Rules per Routing Policy
Multiple Manipulation per Routing Policy
Multiple Manipulation per Routing Policy
IP-to-IP Routing Table
IP-to-IP Inbound Manipulation Table
IP-to-IP Outbound Manipulation Table

The main configuration entities and their involvement in the call processing is summarized in following figure. The figure is used only as an example to provide basic understanding of the configuration terminology. Depending on configuration and network topology, the call process may include additional stages or a different order of stages.

Figure 1-2: SBC Configuration Terminology for Call Processing



1. The device determines the SIP Interface on which the incoming SIP dialog is received and thus, determines its associated SRD.
2. The device classifies the dialog to an IP Group (origin of dialog), using a specific Classification rule that is associated with the dialog's SRD and that matches the incoming characteristics of the incoming dialog defined for the rule.
3. IP Profile and inbound manipulation can be applied to incoming dialog.
4. The device routes the dialog to an IP Group (destination), using the IP-to-IP Routing table. The destination SRD (and thus, SIP Interface and Media Realm) is the one assigned to the IP Group. Outbound manipulation can be applied to the outgoing dialog.

1.3.2 Gateway Application

The objective of your configuration is to enable the device to forward calls between the IP-based endpoints and PSTN-based endpoints. The PSTN-based endpoints can be digital endpoints such as ISDN trunks. The IP-based endpoints (SIP entities) can be servers such as SIP proxy servers and IP PBXs, or end users such as LAN IP phones. In the SIP world, the endpoints are referred to as SIP user agents (UA). The UA that initiates the call is referred to as the user agent client (UAC); the UA that accepts the call is referred to as the user-agent server (UAS).

The following table describes the main configuration concepts and terminology.

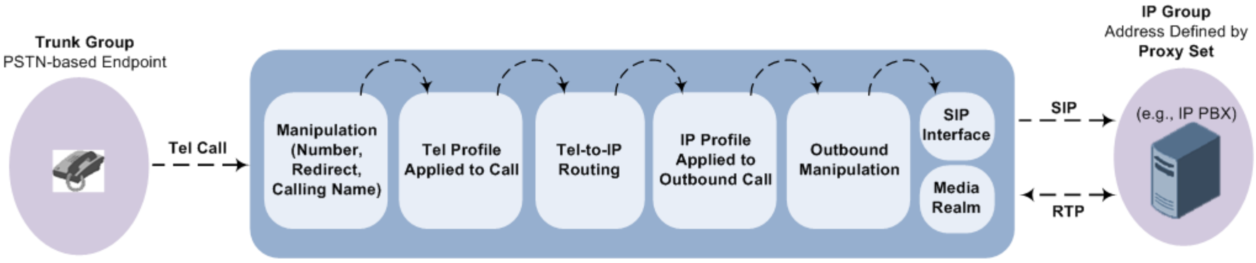
Table 1-3: Configuration Concepts and Terminology

Configuration Terms	Description
IP Groups	The IP Group is a logical representation of the SIP entity (UA) with which the device receives and sends calls. The SIP entity can be a server (e.g., IP PBX or SIP Trunk) or it can be a group of users (e.g., LAN IP phones). For servers, the IP Group is typically used to define the address of the entity (by its associated Proxy Set). IP Groups are typically used in Tel-to-IP routing rules to denote the destination of the call.
Proxy Sets	The Proxy Set defines the actual address (IP address or FQDN) of SIP entities that are servers (e.g., IP PBX). As the IP Group represents the SIP entity, to associate an address with the SIP entity, the Proxy Set is assigned to the IP Group.
SIP Interfaces	The SIP Interface represents a Layer-3 network for the IP-based SIP entity. It defines a local listening port for SIP signaling traffic on a local, logical IP network interface. The term *local* implies that it's a logical port and network interface on the device. The SIP Interface is used to receive and send SIP messages with a specific SIP entity (IP Group). Therefore, you can create a SIP Interface for each SIP entity in the VoIP network with which your device needs to communicate. The SIP Interface is associated with the SIP entity, by assigning the SIP Interface to an SRD that is in turn, assigned to the IP Group of the SIP entity.
Media Realms	The Media Realm defines a local UDP port range for RTP (media) traffic on any one of the device's logical IP network interfaces. The Media Realm is used to receive and send media traffic with a specific SIP entity (IP Group). The Media Realm can be associated with the SIP entity, by assigning the Media Realm to the IP Group of the SIP entity, or by assigning it to the SIP Interface associated with the SIP entity.
SRDs	The SRD is a logical representation of your entire VoIP network. The SRD is in effect, the foundation of your configuration to which all other previously mentioned configuration entities are associated. Typically, only a single SRD is required and this is the recommended configuration topology. As the device provides a default SRD, in a single SRD topology, the device automatically assigns the SRD to newly created configuration entities. Thus, in such scenarios, there is no need to get involved with SRD configuration. Multiple SRDs are required only for multi-tenant deployments.

Configuration Terms	Description
IP Profiles	The IP Profile is an optional configuration entity that defines a wide range of call settings for a specific SIP entity (IP Group). The IP Profile includes signaling and media related settings, for example, jitter buffer, voice coders, fax signaling method, SIP header support (local termination if not supported), and media security method. The IP Profile is in effect, the interoperability "machine" of the device, enabling communication with SIP endpoints supporting different call "languages". The IP Profile is associated with the SIP entity, by assigning the IP Profile to the IP Group of the SIP entity.
Tel Profiles	The Tel Profile is an optional configuration entity that defines a wide range of call settings for a specific PSTN-based endpoint. The IP Profile includes settings such as message waiting indication (MWI), input gain, voice volume and fax signaling method. The Tel Profile is associated with the PSTN-based endpoint, by assigning it to the Trunk Group belonging to the endpoint.
Tel-to-IP Routing Rules	Tel-to-IP routing rules are used to route calls from PSTN-based endpoints to an IP destination (SIP entity). The PSTN side can be denoted by a specific Trunk Group, or calling or called telephone number prefix and suffix. The SIP entity can be denoted by an IP Group or other IP destinations such as IP address, FQDN, E.164 Telephone Number Mapping (ENUM service), and Lightweight Directory Access Protocol (LDAP).
IP-to-Tel (Trunk Group) Routing Rules	IP-to-Tel routing rules are used to route incoming IP calls to Trunk Groups. The specific channel pertaining to the Trunk Group to which the call is routed can also be configured.
Accounts	Accounts are used to register or authenticate PSTN-based endpoints with a SIP entity (e.g., a registrar or proxy server). The device does this on behalf of the PSTN-based endpoint. Authentication (SIP 401) is typically relevant for INVITE messages forwarded by the device to a SIP entity. Registration is for REGISTER messages, which are initiated by the device on behalf of the PSTN-based endpoint.

The following figure shows the main configuration entities and their involvement in call processing. The figure is used only as an example to provide basic understanding of the configuration terminology. Depending on configuration and network topology, the call process may include additional stages or a different order of stages.

Figure 1-3: Gateway Configuration Terminology for Call Processing



Part I

Getting Started with Initial Connectivity

2 Introduction

This part describes how to initially access the device's management interface and change its default IP address to correspond with your networking scheme.

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3 Default OAMP IP Address

The device is shipped with a factory default IP address for operations, administration, maintenance, and provisioning (OAMP), through its VoIP LAN interface. You can use this address to initially access the device from any of its management tools (embedded Web server, EMS, or Telnet/SSH). You can also access the device through the console CLI, by connecting the device's serial (RS-232) port to a PC.

The table below lists the device's default IP address.

Table 3-1: Default VoIP LAN IP Address for OAMP

IP Address	Value
Application Type	OAMP + Media + Control
IP Address	192.168.0.2
Prefix Length	255.255.255.0 (24)
Default Gateway	192.168.0.1
Underlying Device	vlan 1
Interface Name	O+M+C

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4 Configuring VoIP LAN Interface for OAMP

You can change the IP address of the VoIP-LAN interface for OAMP, using any of the following methods:

- Embedded HTTP/S-based Web server - see "Web Interface" on page 39
- Embedded command line interface (CLI) - see "CLI" on page 41

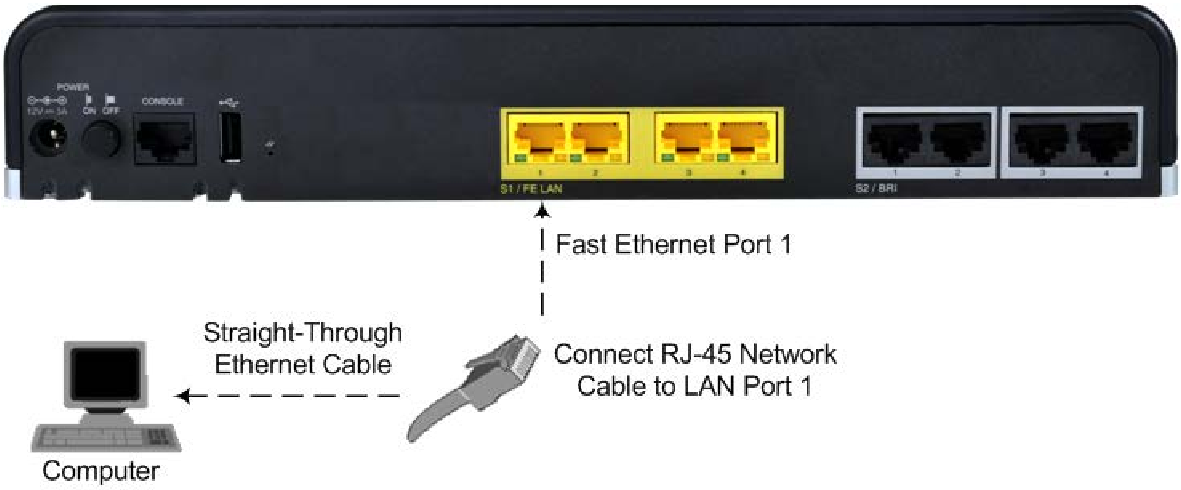
4.1 Web Interface

The following procedure describes how to change the IP address of the OAMP on the VoIP-LAN interface, using the Web-based management tool (Web interface). The default IP address is used to initially access the device.

➢ **To configure the VoIP-LAN IP Address for OAMP, using the Web interface:**

1. Connect Port 1 (left-most LAN port) located on the front panel directly to the network interface of your computer, using a straight-through Ethernet cable.

Figure 4-1: LAN Cabling to PC for Initial Connectivity

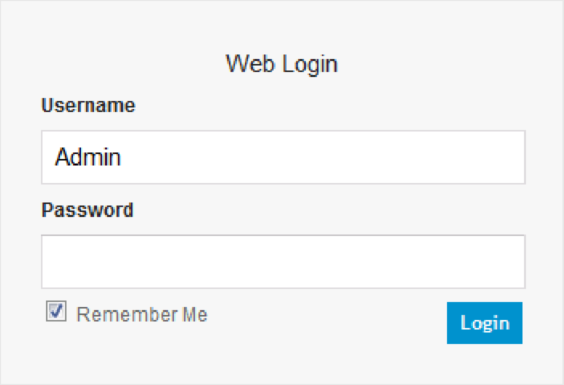

POWER
12V 3A
ON OFF
CONSOLE
S1 / FE LAN
S2 / BRI
Fast Ethernet Port 1
Straight-Through Ethernet Cable
Connect RJ-45 Network Cable to LAN Port 1
Computer

2. Change the IP address and subnet mask of your computer to correspond with the default OAMP IP address and subnet mask of the device.

3. Access the Web interface:

   a. On your computer, start a Web browser and in the URL address field, enter the default IP address of the device; the Web interface's Web Login screen appears:

Figure 4-2: Web Login Screen



Web Login

Username

Admin

Password

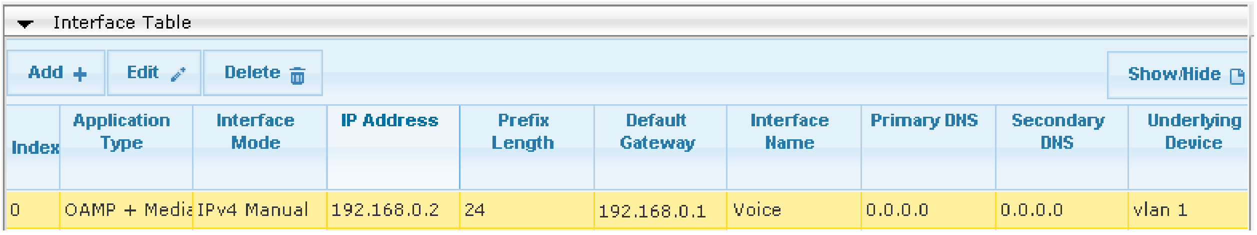
☑ Remember Me

   b. In the 'Username' and 'Password' fields, enter the case-sensitive, default login username ("Admin") and password ("Admin").

   c. Click **Login**.

4. Open the Physical Ports Settings page (Configuration tab > VoIP menu > Network > Physical Ports Table) and then configure the device's physical Ethernet port-pair (group) that you want to later assign to the OAMP interface. For more information, see Configuring Physical Ethernet Ports on page 127.

5. Open the Interface table (**Configuration** tab > **VoIP** menu > **Network** > **IP Interfaces Table**).



Interface Table

Index	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Interface Name	Primary DNS	Secondary DNS	Underlying Device
0	OAMP + Media	IPv4 Manual	192.168.0.2	24	192.168.0.1	Voice	0.0.0.0	0.0.0.0	vlan 1

6. Select the index row corresponding to the **OAMP + Media + Control** application type, and then click **Edit**.

7. Change the IP address to correspond with your network IP addressing scheme, for example:

   - IP Address: 10.8.6.86
   - Prefix Length: 24 (for 255.255.255.0)
   - Gateway: 10.8.6.85
   - Underlying Device: Select the Ethernet Device (VLAN and associated Ethernet port group) for OAMP

8. Click **Add**.

9. Save your settings by resetting the device with a flash burn (see "Resetting the Device" on page 641).
10. Disconnect the device from the PC and cable the device to your network. You can now access the management interface using the new OAMP IP address.

> **Note:** When you complete the above procedure, change your PC's IP address to correspond with your network requirements.

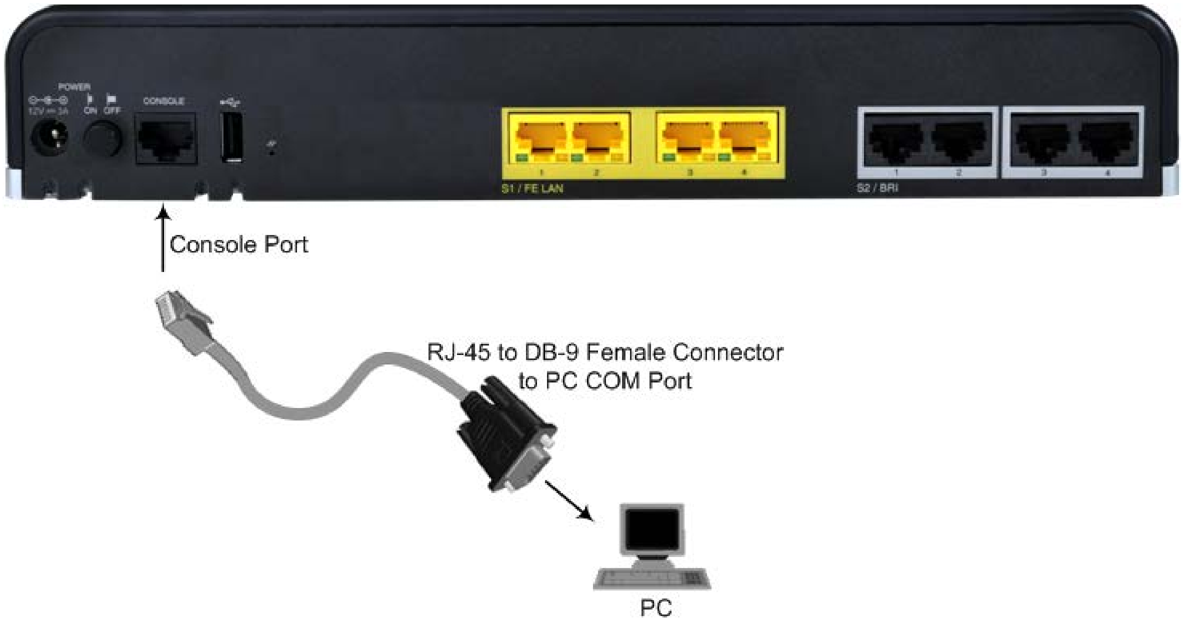
4.2 CLI

This procedure describes how to configure the VoIP-LAN IP address for OAMP using the device's CLI. The procedure uses the regular CLI commands. Alternatively, you can use the CLI Wizard utility to set up your device with the initial OAMP settings. The utility provides a fast-and-easy method for initial configuration of the device through CLI. For more information, refer to the *CLI Wizard User's Guide*.

➢ **To configure the OAMP IP address in the CLI:**

1. Connect the RS-232 port of the device to the serial communication port on your computer. For more information, refer to the Hardware Installation Manual.

Figure 4-3: Serial Cabling to PC



2. Establish serial communication with the device using a terminal emulator program such as HyperTerminal, with the following communication port settings:
   - Baud Rate: 115,200 bps
   - Data Bits: 8
   - Parity: None
   - Stop Bits: 1
   - Flow Control: None

3. At the CLI prompt, type the username (default is "Admin" - case sensitive):

   ```
   Username: Admin
   ```

4. At the prompt, type the password (default is "Admin" - case sensitive):

   ```
   Password: Admin
   ```

5. At the prompt, type the following:

   ```
   enable
   ```

6. At the prompt, type the password again:

   ```
   Password: Admin
   ```

7. Access the VoIP configuration mode:

   ```
   # configure voip
   ```

8. Access the Interface table:

   ```
   (config-voip)# interface network-if 0
   ```

9. Configure the IP address:

   ```
   (network-if-0)# ip-address <IP address>
   ```

10. Configure the prefix length:

    ```
    (network-if-0)# prefix-length <prefix length / subnet mask, e.g., 16>
    ```

11. Configure the Default Gateway address:

    ```
    (network-if-0)# gateway <IP address>
    ```

12. Apply your settings:

    ```
    (network-if-0)# activate
    ```

13. Cable the device to your network. You can now access the device's management interface using this new OAMP IP address.

Part II

Management Tools

5 Introduction

This part provides an overview of the various management tools that can be used to configure the device. It also provides step-by-step procedures on how to configure these management tools.

The device provides the following management tools:

- Embedded HTTP/S-based Web server - see "Web-based Management" on page 47
- Command Line Interface (CLI) - see "CLI-Based Management" on page 81
- Simple Network Management Protocol (SNMP) - see "SNMP-Based Management" on page 91
- Configuration *ini* file - see "INI File-Based Management" on page 99

Notes:

- Some configuration settings can only be done using a specific management tool. For example, some configuration can only be done using the Configuration *ini* file method.
- Throughout this manual, whenever a parameter is mentioned, its corresponding Web, CLI, and ini file parameter is mentioned. The *ini* file parameters are enclosed in square brackets [...].
- For a list and description of all the configuration parameters, see "Configuration Parameters Reference" on page 813.

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6 Web-Based Management

The device provides an embedded Web server (hereafter referred to as *Web interface*), supporting fault management, configuration, accounting, performance, and security (FCAPS), including the following:

- Full configuration
- Software and configuration upgrades
- Loading Auxiliary files, for example, the Call Progress Tones file
- Real-time, online monitoring of the device, including display of alarms and their severity
- Performance monitoring of voice calls and various traffic parameters

The Web interface provides a user-friendly, graphical user interface (GUI), which can be accessed using any standard Web browser (e.g., Microsoft™ Internet Explorer).

Access to the Web interface is controlled by various security mechanisms such as login user name and password, read-write privileges, and limiting access to specific IP addresses.

Notes:

- The Web interface allows you to configure most of the device's settings. However, additional configuration parameters may exist that are not available in the Web interface and which can only be configured using other management tools.
- Some Web interface pages and/or parameters are available only for certain hardware configurations or software features. The software features are determined by the installed Software License Key (see "Software License Key" on page 668).

6.1 Getting Acquainted with the Web Interface

This section provides a description of the Web interface.

6.1.1 Computer Requirements

The client computer requires the following to work with the Web interface of the device:

- A network connection to the device
- One of the following Web browsers:
  - Microsoft™ Internet Explorer™ (Version 11.0.13 and later)
  - Mozilla Firefox® (Versions 5 through 9.0)
- Recommended screen resolutions: 1024 x 768 pixels, or 1280 x 1024 pixels

Note: Your Web browser must be JavaScript-enabled to access the Web interface.

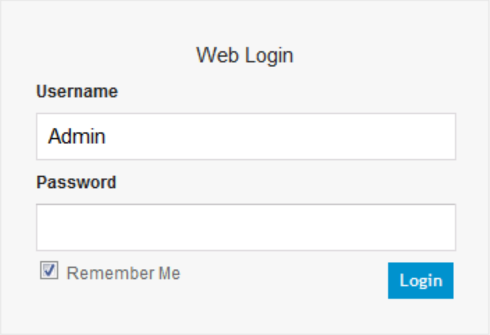
6.1.2 Accessing the Web Interface

The following procedure describes how to access the Web interface.

➢ **To access the Web interface:**

1. Open a standard Web browser (see "Computer Requirements" on page 47).
2. In the Web browser, specify the OAMP IP address of the device (e.g., http://10.1.10.10); the Web interface's Login window appears, as shown below**:**

Figure 6-1: Web Login Screen



Web Login

Username

Admin

Password

☑ Remember Me

3. In the 'Username' and 'Password' fields, enter the case-sensitive, user name and password respectively.
4. Click **Login**; the Web interface is accessed, displaying the Home page. For a detailed description of the Home page, see "Viewing the Home Page" on page 68.

Notes:

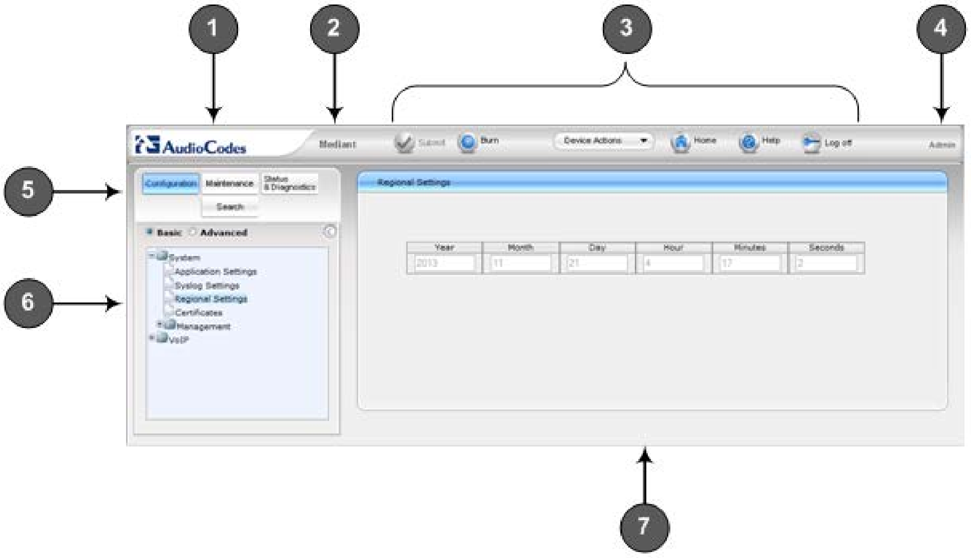
- By default, Web access is only through the IP address of the OAMP interface. However, you can allow access from all of the device's IP network interfaces, by setting the EnableWebAccessFromAllInterfaces parameter to 1.
- The default login username and password is "Admin". To change the login credentials, see "Configuring the Web User Accounts" on page 70.
- If you want the Web browser to remember your password, select the 'Remember Me' check box and then agree to the browser's prompt (depending on your browser) to save the password for future logins. On your next login attempt, simply press the Tab or Enter keys to auto-fill the 'Username' and 'Password' fields, and then click **Login**.
- By default, autocompletion of the login username is enabled whereby the 'Username' field offers previously entered usernames. To disable autocompletion, use the WebLoginBlockAutoComplete ini file parameter.
- Depending on your Web browser's settings, a security warning box may be displayed. The reason for this is that the device's certificate is not trusted by your PC. The browser may allow you to install the certificate, thus skipping the warning box the next time you connect to the device. If you are using Windows Internet Explorer, click **View Certificate**, and then **Install Certificate**. The browser also warns you if the host name used in the URL is not identical to the one listed in the certificate. To resolve this, add the IP address and host name (ACL_nnnnnn, where *nnnnnn* is the serial number of the device) to your hosts file, located at /etc/hosts on UNIX or C:\Windows\System32\Drivers\ETC\hosts on Windows; then use the host name in the URL (e.g., https://ACL_280152). Below is an example of a host file:

```
127.0.0.1    localhost
10.31.4.47   ACL_280152
```

6.1.3 Areas of the GUI

The areas of the Web interface's GUI are shown in the figure below and described in the subsequent table.

Figure 6-2: Main Areas of the Web Interface GUI



Year	Month	Day	Hour	Minutes	Seconds
2013	11	21	4	17	2

Table 6-1: Description of the Web GUI Areas

Item #	Description
1	AudioCodes company logo.
2	Product name.
3	Toolbar, providing frequently required command buttons. For more information, see "Toolbar Description" on page 50.
4	Displays the username of the Web user that is currently logged in.
5	Navigation bar, providing the following tabs for accessing various functionalities in the Navigation tree: ▪ **Configuration**, **Maintenance**, and **Status & Diagnostics** tabs: Access the configuration menus (see "Working with Configuration Pages" on page 53) ▪ **Search** tab: Enables a search engine for searching configuration parameters (see "Searching for Configuration Parameters" on page 60)
6	Navigation tree, displaying a tree-like structure of elements (configuration menus or search engine) pertaining to the selected tab on the Navigation bar. For more information, see "Navigation Tree" on page 50.
7	Work pane, displaying the configuration page of the selected menu in the Navigation tree. This is where configuration is done. For more information, see "Working with Configuration Pages" on page 53.

6.1.4 Toolbar Description

The toolbar provides frequently required command buttons, described in the table below:

Table 6-2: Description of Toolbar Buttons

Icon	Button Name	Description
✓	**Submit**	Applies parameter settings to the device (see "Saving Configuration" on page 643). **Note:** This icon is grayed out when not applicable to the currently opened page.
◉	**Burn**	Saves parameter settings to flash memory (see "Saving Configuration" on page 643).
Device Actions ▼	**Device Actions**	Opens a drop-down list with frequently needed commands: ▪ **Load Configuration File:** Opens the Configuration File page for loading an *ini* file to the device (see "Backing Up and Loading Configuration File" on page 677). ▪ **Save Configuration File:** Opens the Configuration File page for saving the *ini* file to a folder on your PC (see "Backing Up and Loading Configuration File" on page 677). ▪ **Reset:** Opens the Maintenance Actions page for performing various maintenance procedures such as resetting the device (see "Resetting the Device" on page 641). ▪ **Software Upgrade Wizard:** Starts the Software Upgrade Wizard for upgrading the device's software (see "Software Upgrade Wizard" on page 673).
⌂	**Home**	Opens the Home page (see "Viewing the Home Page" on page 68).
?	**Help**	Opens the Online Help topic of the currently opened configuration page (see "Getting Help" on page 63).
🔑	**Log off**	Logs off a session with the Web interface (see "Logging Off the Web Interface" on page 64).
-	**Reset**	If you modify a parameter on a page that takes effect only after a device reset, after you click the **Submit** button, the toolbar displays "Reset". This is a reminder that you need to later save your settings to flash memory and reset the device.

6.1.5 Navigation Tree

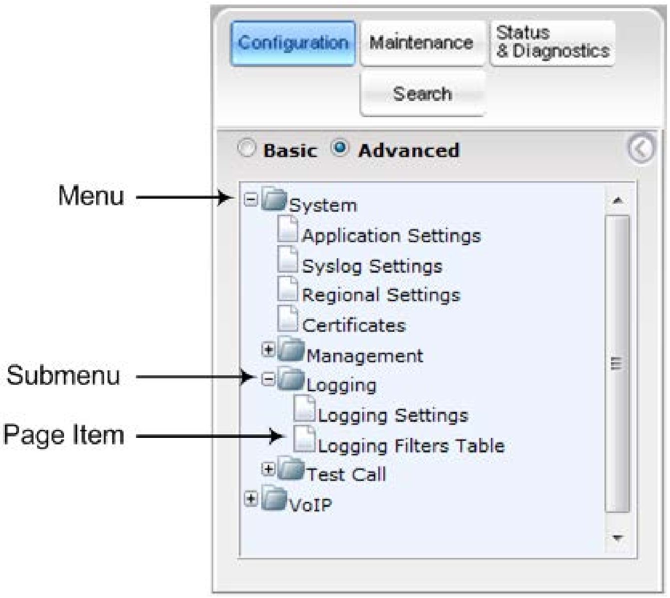
The Navigation tree is located in the Navigation pane and displays a tree-like structure of menus pertaining to the selected tab on the Navigation bar. You can drill-down to the required page item level to open its corresponding page in the Work pane.

The terminology used throughout this manual for referring to the hierarchical structure of the tree is as follows:

- *Menu*: first level (highest level)
- *Submenu*: second level - contained within a menu

- *Page item*: last level (lowest level in a menu) - contained within a menu or submenu

Figure 6-3: Navigating in Hierarchical Menu Tree (Example)



> **Note:** The figure above is used only as an example. The displayed menus depend on supported features based on the Software License Key installed on your device.

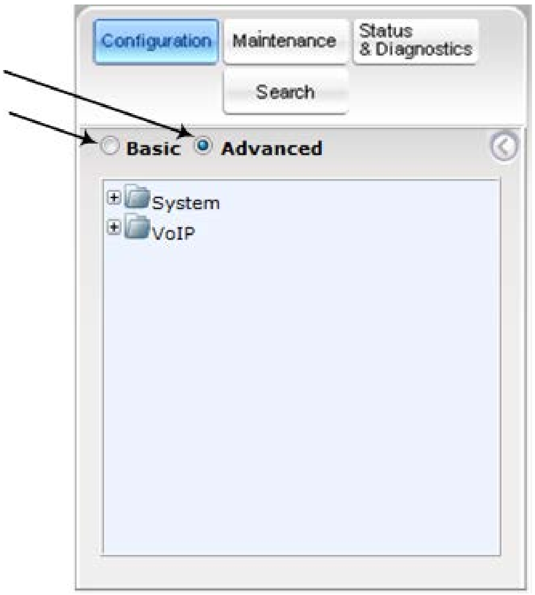
6.1.5.1 Displaying Navigation Tree in Basic and Full View

You can view an expanded or reduced display of the Navigation tree. This affects the number of displayed menus and submenus in the tree. The expanded view displays all the menus pertaining to the selected configuration tab; the reduced view displays only commonly used menus.

- To display a reduced menu tree, select the **Basic** option (default).

- To display all menus and submenus, select the **Advanced** option.

Figure 6-4: Basic and Full View Options



○ Basic ● Advanced

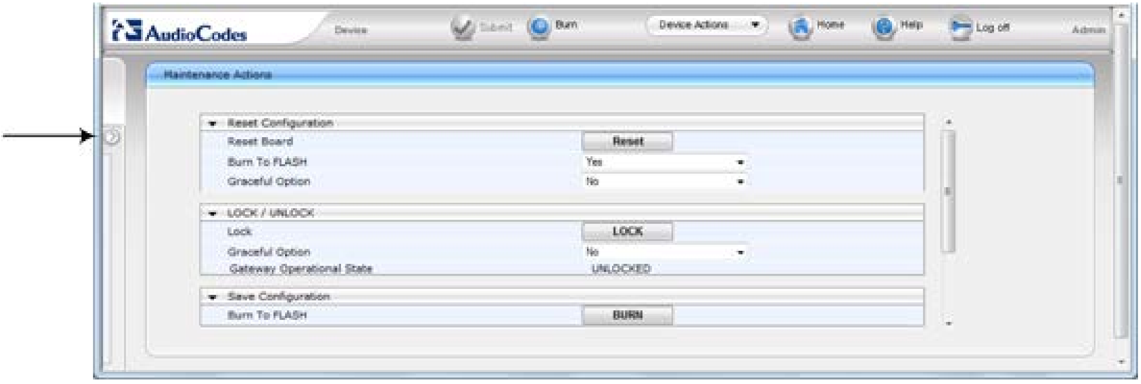
> **Note:** After you reset the device, the Web GUI is displayed in **Basic** view.

6.1.5.2 Showing / Hiding the Navigation Pane

You can hide the Navigation pane to provide more space for elements displayed in the Work pane. This is especially useful when the Work pane displays a wide table. The arrow button located below the Navigation bar is used to hide and show the pane.

- To hide the Navigation pane, click the left-pointing arrow ; the pane is hidden and the button is replaced by the right-pointing arrow button.
- To show the Navigation pane, click the right-pointing arrow ; the pane is displayed and the button is replaced by the left-pointing arrow button.

Figure 6-5: Show and Hide Button (Navigation Pane in Hide View)



Maintenance Actions

Reset Configuration

Reset Board	Reset
Burn To FLASH	Yes
Graceful Option	No

LOCK / UNLOCK

Lock	LOCK
Graceful Option	No
Gateway Operational State	UNLOCKED

Save Configuration

Burn To FLASH	BURN

6.1.6 Working with Configuration Pages

The configuration pages contain the parameters for configuring the device and are displayed in the Work pane.

6.1.6.1 Accessing Pages

The configuration pages are accessed by clicking the required page item in the Navigation tree.

➢ **To open a configuration page:**

1. On the Navigation bar, click the required tab (**Configuration**, **Maintenance**, or **Status & Diagnostics**); the menus pertaining to the selected tab appear in the Navigation tree.
2. Navigate to the required page item, by performing the following:
    - Drill-down using the **plus** ⊞ sign to expand the menu and submenus.
    - Drill-up using the **minus** ⊟ sign to collapse the menu and submenus.
3. Click the required page item; the page opens in the Work pane.

You can also access previously opened pages by clicking the Web browser's **Back** button until you have reached the required page. This is useful if you want to view pages in which you have performed configurations in the current Web session.

> **Note:** Depending on the access level of your Web user account, certain pages may not be accessible or may be read-only (see "Configuring Web User Accounts" on page 70). If a page is read-only, "Read-Only Mode" is displayed at the bottom of the page.

6.1.6.2 Viewing Parameters

Some pages allow you to view a reduced or expanded display of parameters. The Web interface provides two methods for displaying page parameters:

- Displaying "basic" and "advanced" parameters - see "Displaying Basic and Advanced Parameters" on page 53
- Displaying parameter groups - see "Showing / Hiding Parameter Groups" on page 54

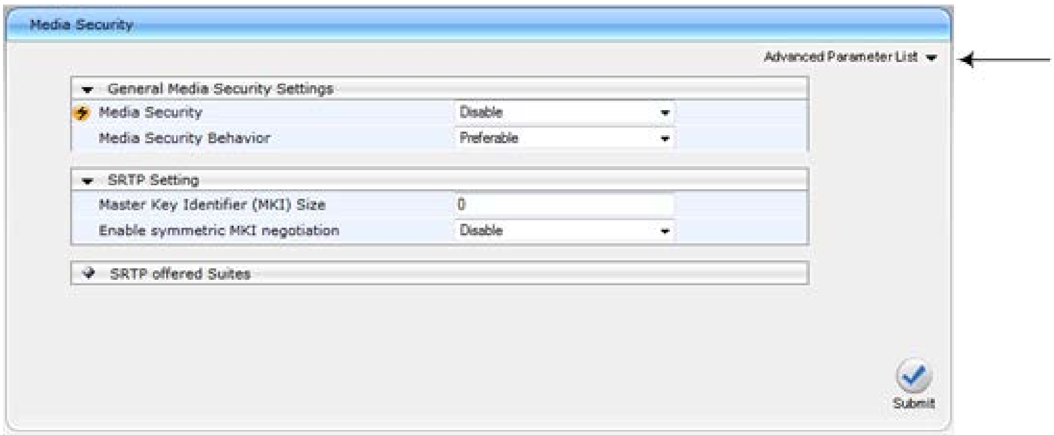
6.1.6.2.1 Displaying Basic and Advanced Parameters

Some pages provide a toggle button that allows you to show and hide parameters. This button is located on the top-right corner of the page and has two display states:

- **Advanced Parameter List** button with down-pointing arrow: click this button to display all parameters.
- **Basic Parameter List** button with up-pointing arrow: click this button to show only common (*basic*) parameters.

The figure below shows an example of a page displaying basic parameters only. If you click the **Advanced Parameter List** button (shown below), the page will also display the advanced parameters.

Figure 6-6: Toggling between Basic and Advanced View



Media Security

Advanced Parameter List

General Media Security Settings

Media Security	Disable
Media Security Behavior	Preferable

SRTP Setting

Master Key Identifier (MKI) Size	0
Enable symmetric MKI negotiation	Disable

SRTP offered Suites

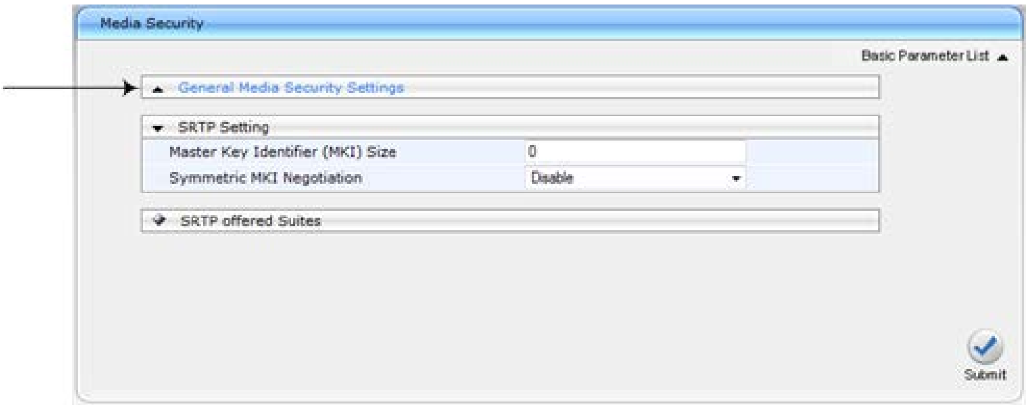
Notes:

- When the Navigation tree is in **Advanced** display mode (see "Navigation Tree" on page 50), configuration pages display all their parameters.
- If you reset the device, the Web pages display only the basic parameters.
- The basic parameters are displayed in a different background color to the advanced parameters.

6.1.6.2.2 Showing / Hiding Parameter Groups

Some pages group parameters under sections, which can be hidden or shown. To toggle between hiding and showing a group, simply click the group title name that appears above each group. The button appears with a down-pointing or up-pointing arrow, indicating that it can be collapsed or expanded when clicked, respectively.

Figure 6-7: Expanding and Collapsing Parameter Groups



Media Security

Basic Parameter List

General Media Security Settings

SRTP Setting

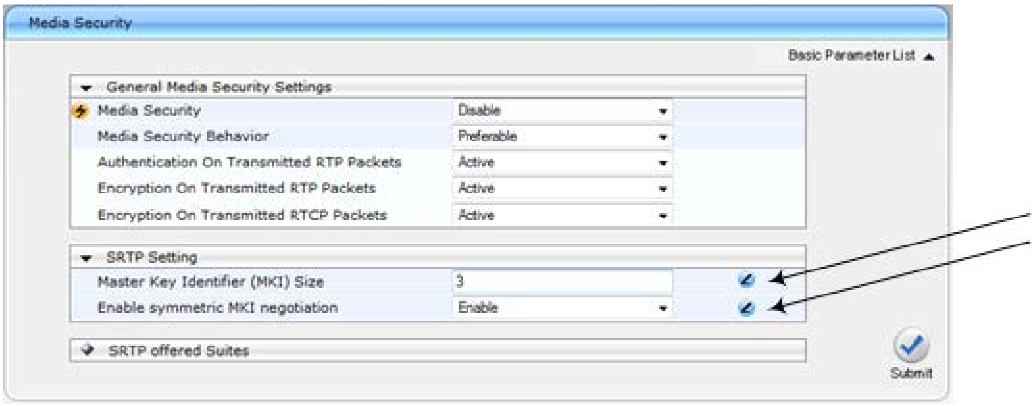
Master Key Identifier (MKI) Size	0
Symmetric MKI Negotiation	Disable

SRTP offered Suites

6.1.6.3 Modifying and Saving Parameters

When you modify a parameter value on a page, the **Edit** ✎ icon appears to the right of the parameter. This indicates that the parameter has been modified, but has yet to be applied (submitted). After you click **Submit** the ✎ icon disappears.

Figure 6-8: Edit Symbol after Modifying Parameter Value



Media Security

Basic Parameter List ▲

▾ General Media Security Settings

⚡ Media Security	Disable ▾	
Media Security Behavior	Preferable ▾	
Authentication On Transmitted RTP Packets	Active ▾	
Encryption On Transmitted RTP Packets	Active ▾	
Encryption On Transmitted RTCP Packets	Active ▾	

▾ SRTP Setting

Master Key Identifier (MKI) Size	3	✎
Enable symmetric MKI negotiation	Enable ▾	✎

▸ SRTP offered Suites

- **To save configuration changes on a page to the device's volatile memory (RAM):**

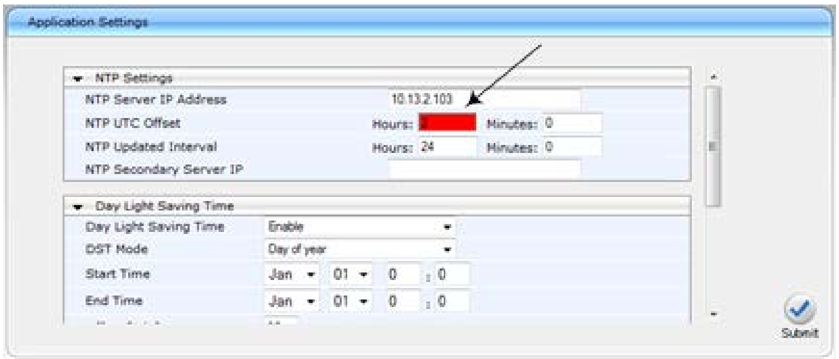
- On the toolbar, click the **Submit** ✓ button.
- At the bottom of the page, click the **Submit** ✓ button.

When you click **Submit**, modifications to parameters with on-the-fly capabilities are immediately applied to the device and take effect. Parameters displayed on the page with the lightning ⚡ icon take effect only after a device reset. For resetting the device, see "Resetting the Device" on page 641.

Note: Parameters saved to the volatile memory (by clicking **Submit**), revert to their previous settings after a hardware or software reset, or if the device is powered down. Thus, to ensure parameter changes (whether on-the-fly or not) are retained, save ('burn') them to the device's non-volatile memory, i.e., flash (see "Saving Configuration" on page 643).

If you enter an invalid value (e.g., not in the range of permitted values) and then click **Submit**, a message box appears notifying you of the invalid value. In addition, the parameter value reverts to its previous value and is highlighted in red, as shown in the figure below:

Figure 6-9: Value Reverts to Previous Valid Value



Application Settings

▾ NTP Settings

NTP Server IP Address	10.13.2.103			
NTP UTC Offset	Hours:	🟥	Minutes:	0
NTP Updated Interval	Hours:	24	Minutes:	0
NTP Secondary Server IP				

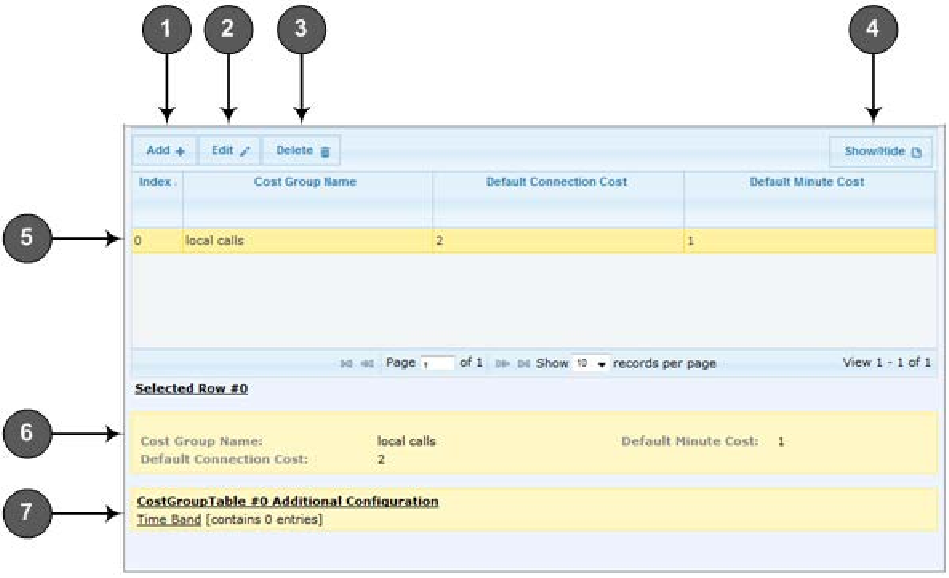
▾ Day Light Saving Time

Day Light Saving Time	Enable ▾			
DST Mode	Day of year ▾			
Start Time	Jan ▾	01 ▾	0	: 0
End Time	Jan ▾	01 ▾	0	: 0

6.1.7 Working with Tables

Many of the Web configuration pages provide tables for configuring various functionalities of the device. The figure below and subsequent table describe the areas of a typical configuration table:

Figure 6-10: Displayed Details Pane



Index	Cost Group Name	Default Connection Cost	Default Minute Cost
0	local calls	2	1

Selected Row #0

Cost Group Name: local calls
Default Connection Cost: 2
Default Minute Cost: 1

CostGroupTable #0 Additional Configuration
Time Band [contains 0 entries]

Table 6-3: Enhanced Table Design Description

Item #	Button	
1	**Add**	Adds a new index entry row to the table. When you click this button, a dialog box appears with parameters for configuring the new entry. When you have completed configuration, click the **Submit** button in the dialog box to add it to the table.
2	**Edit**	Edits the selected row.
3	**Delete**	Removes the selected row from the table. When you click this button, a confirmation box appears requesting you to confirm deletion. Click **Delete** to accept deletion.
4	**Show/Hide**	Toggles between displaying and hiding the full configuration of a selected row. This configuration is displayed below the table (see Item #6) and is useful for large tables that cannot display all its columns in the work pane.
5	**-**	Selected index row entry for editing, deleting and showing configuration.
6	**-**	Displays the full configuration of the selected row when you click the **Show/Hide** button.
7	**-**	Links to access additional configuration tables related to the current configuration.

6.1.7.1 Table Toolbar Description

The configuration tables provide a toolbar with various buttons, as described below.

Table 6-4: Table Toolbar Description

Button	Name	
	Add	Adds a new index entry row to the table. When you click this button, the Add Row dialog box appears with parameters for configuring the new entry. When you have completed configuration, click the **Add** button in the dialog box to add it to the table.
	Edit	Edits the selected row. When you click this button, the Edit Row dialog box appears to modify the row entry. When you have completed configuration, click the **Save** button in the dialog box.
	Delete	Removes the selected row from the table. When you click this button, the Delete Row confirmation box appears requesting you to confirm deletion. Click **Delete** to accept deletion.
	Show / Hide	Toggles between displaying and hiding the full configuration of a selected row. The configuration is displayed below the table and is useful for tables containing many parameters, which cannot all be displayed in the work pane.
	Action	Provides a drop-down list with commands (e.g., Register and Unregister) relevant to the specific table (e.g., Account table). **Note:** The button only appears in certain tables.
	Insert	Adds a new table row at a selected index. You can add the row at any existing (configured) index number. If you select a row and then click the button, after configuring the parameters, the row is automatically added to the index of the selected row and the row that previously occupied the index row and all rows below it are moved one index down in the table. For example, if you select Index 2 and then click the button, the new row is assigned Index 2 and the row previously occupying Index 2 is moved down to Index 3 and so on. **Notes:** ▪ The button is available only if the table is sorted according to 'Index' column; otherwise, the button is grayed out. For sorting tables, see "Sorting Tables by Column" on page 60. ▪ The button appears only in certain tables.
	Down	Moves a selected row one index down. The index number of the row changes according to its new position in the table. The row that previously occupied the index row and all rows below it are moved one index down in the table. **Notes:** ▪ The button is available only if the table contains more than one row and is sorted according to 'Index' column; otherwise, the button is grayed out. For sorting tables, see "Sorting Tables by Column" on page 60. ▪ The button appears only in certain tables.
	Up	Moves a selected row one index up. The index number of the row changes according to its new position in the table. The row that previously occupied the index row and all rows below it are moved one index down in the table. **Notes:**

Button	Name	
		▪ The button is available only if the table contains more than one row and is sorted according to 'Index' column; otherwise, the button is grayed out. For sorting tables, see "Sorting Tables by Column" on page 60. ▪ The button appears only in certain tables.
Clone	**Clone**	Adds a new row with identical settings as the selected row. **Note:** The button only appears in certain tables.

6.1.7.2 Toggling Display Mode of Table Dialog Boxes

Add and Edit dialog boxes that appear when you click the **Add** and **Edit** buttons respectively, by default display parameters in Tab view, whereby parameters are grouped under tabs according to functionality (e.g., Rule, Action, and Status). You can change the view mode to Classic view, whereby all parameters appear in one list and whereby parameters are separated according to functionality by a heading instead of a tab.

➢ **To toggle between Tab and Classic view:**

- Click the **Classic View** or **Tabs View** link located at the bottom of the dialog box.

6.1.7.3 Scrolling through Table Pages

You can define the maximum number of rows (indices) to display in the table. To view additional rows, you can scroll through the table pages. The figure below shows the table page navigation area, which is located below the table:

Figure 6-11: Viewing Table Rows per Page

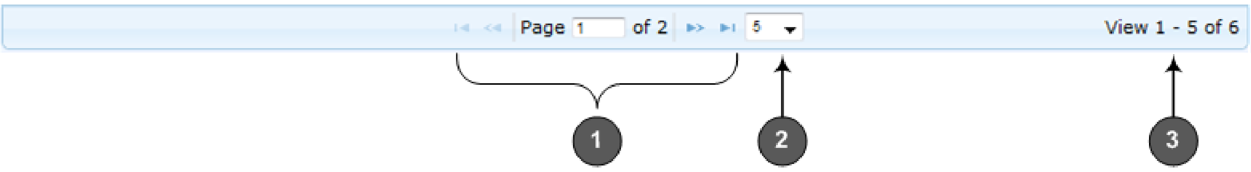


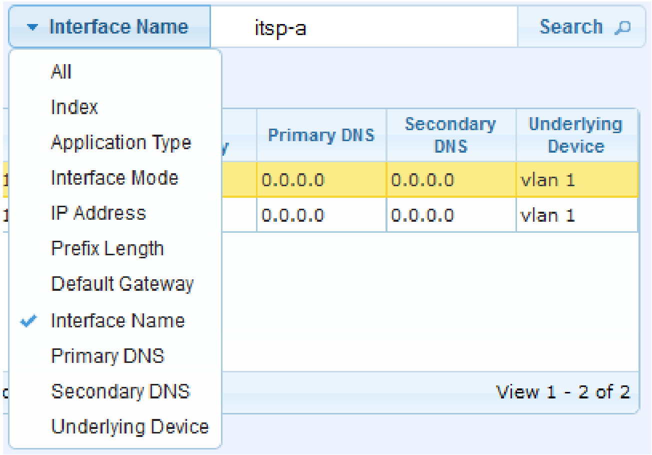
Table 6-5: Row Display and Page Navigation

Item #	Description
1	Defines the page that you want to view. Enter the required page number or use the following page navigation buttons: ▪ ▶ - Displays the next page ▪ ▶| - Displays the last page ▪ ◀ - Displays the previous page ▪ |◀ - Displays the first page
2	Defines the number of rows to display per page. You can select 5, 10 (default), or 20.
3	Displays the currently displayed number of rows out of the maximum configured.

6.1.7.4 Searching Table Entries

The configuration tables provide you with a search feature that lets you search any value (string or IP address) of a specified parameter (column) in the table. By default, searches are performed on all the table's parameters. You define the search using the search features, located on top of the table (on the table's toolbar), as shown in the example below:

Figure 6-12: Searching Table Entries



Interface Name: itsp-a

Primary DNS	Secondary DNS	Underlying Device
0.0.0.0	0.0.0.0	vlan 1
0.0.0.0	0.0.0.0	vlan 1

View 1 - 2 of 2

➢ **To search for a table entry:**

1. From the search drop-down list, select the table column name in which you want to search for the value.
2. In the search text box, enter the value for which you want to search.
3. Click **Search**.

If the device locates searched entries, the table displays only the rows in which the entries were found. If the search was unsuccessful, no rows are displayed and a message is displayed notifying you that no records were found.

To quit the search feature, click the **X** icon, displayed alongside the "Showing results for" message below the **Add** button.

Note: The search feature is supported only for certain tables.

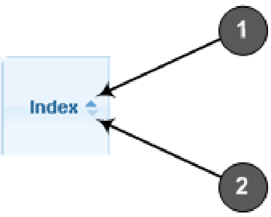
6.1.7.5 Sorting Tables by Column

You can sort table rows by any table column and in ascending (e.g., 1, 2 and 3, or a, b, and c) or descending (3, 2, and 1, or c, b, and a) order. For example, instead of the rows being sorted by the Index column in ascending order (e.g., 1, 2, and 3), you can sort the rows by Index column in descending order (e.g., 3, 2, and 1). By default, most tables are sorted by Index column in ascending order.

➢ **To sort table rows by column:**

1. Click the heading name of the column that you want to sort the table rows by; the up-down arrows appear alongside the heading name and the up button is bolded (see Item 1 in the figure below), indicating that the column is sorted in ascending order:

Figure 6-13: Sorting Table Rows by Column



2. To sort the column in descending order, click the heading name of the column again; only the down arrow appears bolded (see Item 2 in the figure above), indicating that the column is sorted in descending order.

6.1.8 Searching for Configuration Parameters

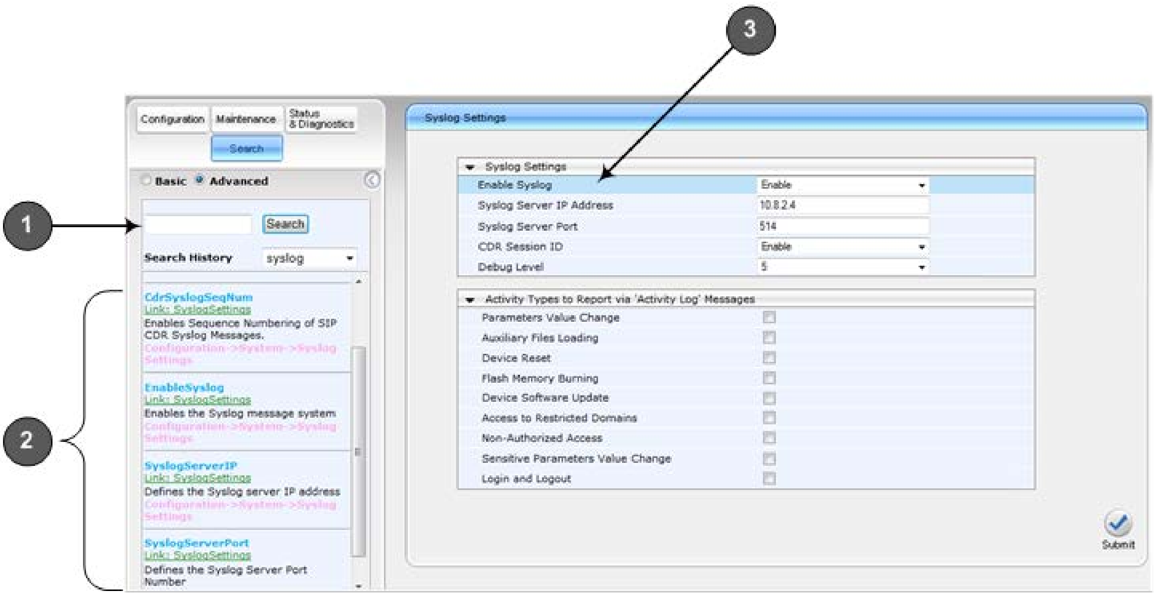
You can locate the exact Web page on which a specific parameter appears, by using the Search feature. To search for a Web parameter, you must use the *ini* file parameter name as the search key. The search key can include the full parameter name (e.g., "EnableSyslog") or a substring of it (e.g., "sys"). If you search for a substring, all parameters containing the specified substring in their names are listed in the search result.

➢ **To search for a parameter:**

1. On the Navigation bar, click the **Search** tab; the Search engine appears in the Navigation pane.
2. In the field alongside the **Search** button, enter the parameter name or a substring of the name for which you want to search. If you have done a previous search for such a parameter, instead of entering the required string, you can use the 'Search History' drop-down list to select the string saved from a previous search.
3. Click **Search**; a list of found parameters based on your search key appears in the Navigation pane. Each searched result displays the following:
   - *ini* file parameter name
   - Link (in green) to the Web page on which the parameter appears
   - Brief description of the parameter
   - Menu navigation path to the Web page on which the parameter appears
4. In the searched list, click the required parameter (green link) to open the page on which the parameter appears; the relevant page opens in the Work pane and the searched

parameter is highlighted in the page for easy identification, as shown in the figure below:

Figure 6-14: Searched Result Screen



○ Basic ● Advanced

Search History: syslog

CdrSyslogSeqNum
Link: SyslogSettings
Enables Sequence Numbering of SIP CDR Syslog Messages.
Configuration->System->Syslog Settings

EnableSyslog
Link: SyslogSettings
Enables the Syslog message system
Configuration->System->Syslog Settings

SyslogServerIP
Link: SyslogSettings
Defines the Syslog server IP address
Configuration->System->Syslog Settings

SyslogServerPort
Link: SyslogSettings
Defines the Syslog Server Port Number

Syslog Settings

Syslog Settings

Enable Syslog	Enable
Syslog Server IP Address	10.8.2.4
Syslog Server Port	514
CDR Session ID	Enable
Debug Level	5

Activity Types to Report via 'Activity Log' Messages

Parameters Value Change	□
Auxiliary Files Loading	□
Device Reset	□
Flash Memory Burning	□
Device Software Update	□
Access to Restricted Domains	□
Non-Authorized Access	□
Sensitive Parameters Value Change	□
Login and Logout	□

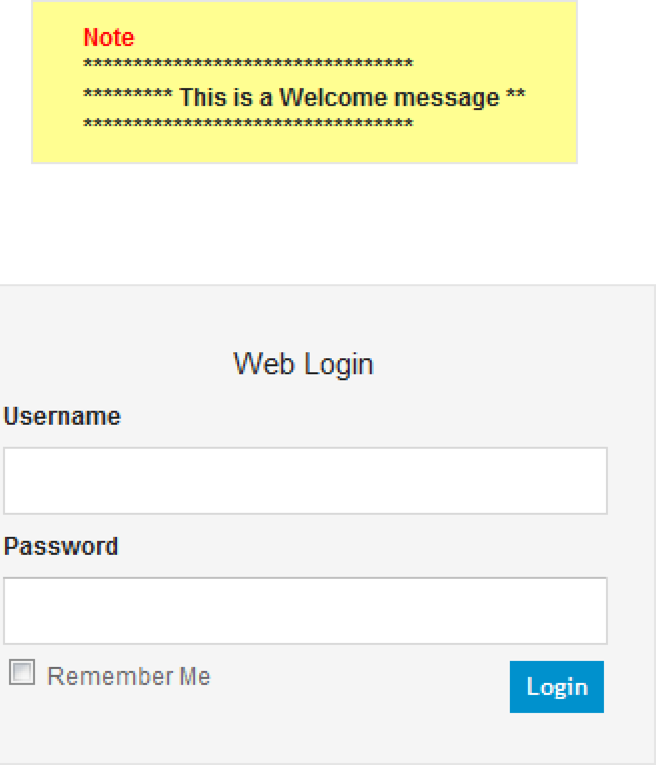
Table 6-6: Search Description

Item #	Description
1	Search field for entering search key and **Search** button for activating the search process.
2	Search results listed in Navigation pane.
3	Found parameter, highlighted on relevant Web page

6.1.9 Creating a Login Welcome Message

You can create a Welcome message box that is displayed on the Web Login page. The figure below displays an example of a Welcome message:

Figure 6-15: User-Defined Web Welcome Message after Login



Note
****
********* This is a Welcome message **
****

To enable and create a Welcome message, use the WelcomeMessage table ini file parameter, as described in the table below. If the parameter is not configured, no Welcome message is displayed.

Table 6-7: *ini* File Parameter for Welcome Login Message

Parameter	Description
[WelcomeMessage]	Enables and defines a Welcome message that appears on the Web Login page for logging in to the Web interface. The format of the ini file table parameter is: [WelcomeMessage] FORMAT WelcomeMessage_Index = WelcomeMessage_Text; [\WelcomeMessage] For Example: [WelcomeMessage] FORMAT WelcomeMessage_Index = WelcomeMessage_Text; WelcomeMessage 1 = "**********************************"; WelcomeMessage 2 = "********* This is a Welcome message **"; WelcomeMessage 3 = "**********************************"; [\WelcomeMessage] Each index row represents a line of text in the Welcome message box. Up to 20 lines (or rows) of text can be defined.

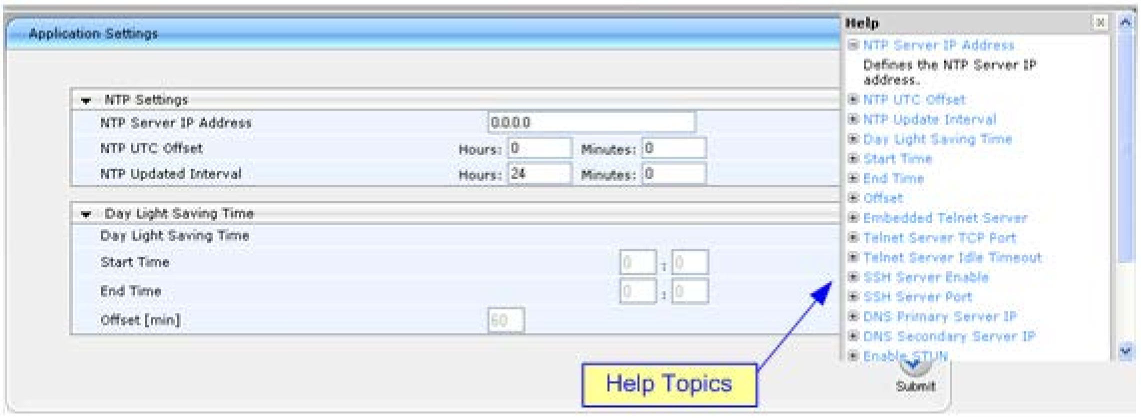
6.1.10 Getting Help

The Web interface provides you with context-sensitive Online Help. The Online Help provides brief descriptions of parameters pertaining to the currently opened page.

➢ **To view the Help topic of a currently opened page:**

1. On the toolbar, click the **Help** button; the Help topic pertaining to the opened page appears, as shown below:

Figure 6-16: Help Topic for Current Page



Application Settings

NTP Settings

NTP Server IP Address: 0.0.0.0

NTP UTC Offset: Hours: 0 Minutes: 0

NTP Updated Interval: Hours: 24 Minutes: 0

Day Light Saving Time

Day Light Saving Time

Start Time: 0 : 0

End Time: 0 : 0

Offset [min]: 60

Help

NTP Server IP Address

Defines the NTP Server IP address.

2. To view a description of a parameter, click the **plus** sign to expand the parameter. To collapse the description, click the **minus** sign.
3. To close the Help topic, click the **close** button located on the top-right corner of the Help topic window or simply click the **Help** button.

Note: Instead of clicking the **Help** button for each page you open, you can open it once for a page and then simply leave it open. Each time you open a different page, the Help topic pertaining to that page is automatically displayed.

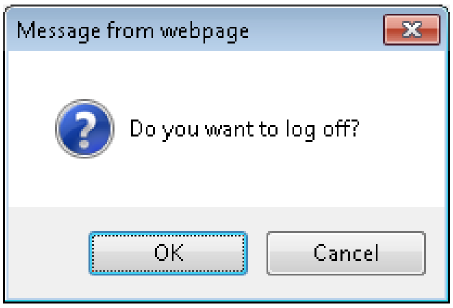
6.1.11 Logging Off the Web Interface

The following procedure describes how to log off the Web interface.

➢ **To log off the Web interface:**

1. On the toolbar, click the **Log Off** icon; the following confirmation message box appears:

Figure 6-17: Log Off Confirmation Box



Message from webpage

Do you want to log off?

2. Click **OK**; you are logged off the Web session and the Web Login dialog box appears enabling you to re-login, if required.

6.2 Customizing the Web Interface

You can customize the following elements of the device's Web interface (GUI):

- Corporate logo (see Replacing the Corporate Logo on page 64)
- Device's (product) name (see Customizing the Product Name on page 66)
- Favicon (see Customizing the Favicon on page 66)
- Login welcome message (see Creating a Login Welcome Message on page 62)

Note:

- The product name also affects other management interfaces.
- In addition to Web-interface customization, you can customize the following to reference your company instead of AudioCodes:
  - √ SNMP Interface: Product system OID (see the SNMPSysOid parameter) and trap Enterprise OID (see the SNMPTrapEnterpriseOid parameter).
  - √ SIP Messages: User-Agent header (see the UserAgentDisplayInfo parameter), SDP "o" line (see the SIPSDPSessionOwner parameter), and Subject header (see the SIPSubject parameter).

6.2.1 Replacing the Corporate Logo

You can replace the default corporate logo image (i.e., AudioCodes logo) that is displayed in the Web interface. The logo appears in the following Web areas:

- Web Login screen
- Menu bar

You can replace the logo with one of the following:

- A different image (see Replacing the Corporate Logo with an Image on page 65)
- Text (see Replacing the Corporate Logo with Text on page 65)

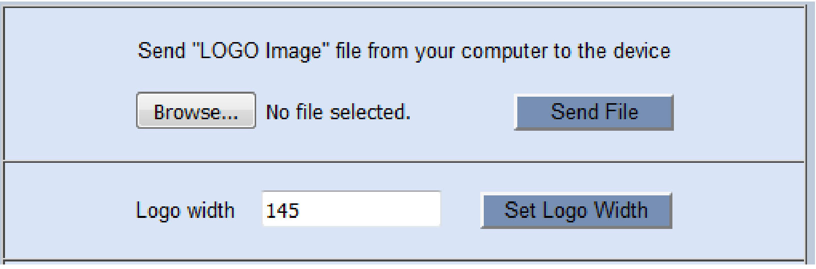
6.2.1.1 Replacing the Corporate Logo with an Image

You can replace the logo with a different image.

➢ **To customize the logo:**

1. Save your new logo image file in a folder on the same PC that you are using to access the device's Web interface.
2. In your browser's URL address field, append the case-sensitive suffix "/AdminPage" to the device's IP address (e.g., http://10.1.229.17/AdminPage).
3. Log in with your credentials; the Admin page appears.
4. On the left pane, click **Image Load to Device**; the right pane displays the following:

Figure 6-18: Customizing Web Logo



Send "LOGO Image" file from your computer to the device

No file selected.

Logo width 145

5. Use the **Browse** button to select your logo file, and then click **Send File**; the device loads the file.
6. If you want to modify the width of the image, in the 'Logo Width' field, enter the new width (in pixels) and then click the **Set Logo Width** button.
7. On the left pane, click **Back to Main** to exit the Admin page.
8. Reset the device with a save-to-flash for your settings to take effect.

Note:

- The logo image file type can be GIF, PNG, JPG, or JPEG.
- The logo image must have a fixed height of 24 pixels. The width can be up to 199 pixels (default is 145).
- The maximum size of the image file can be 64 Kbytes.
- Ignore the **ini Parameters** option, which is located on the left pane of the Admin page.

6.2.1.2 Replacing the Corporate Logo with Text

You can replace the logo with text.

➢ **To replace the logo with text:**

1. Create an ini file that includes the following parameter settings:

```
UseWebLogo = 1
WebLogoText = < your text >
```

2. Load the ini file using the Auxiliary Files page (see Loading Auxiliary Files on page 647).
3. Reset the device with a save-to-flash for your settings to take effect.

6.2.2 Customizing the Product Name

You can customize the device's product name. The name is displayed in various places in the management interfaces, as shown below using the customized name, "My Product Name":

- **Web Login screen**
- **Ini file "Board" field:**

```
Board: My Product Name
```

- **CLI prompt:**

```
My Product Name(config-system)#
```

To customize the device's product name:

1. Create an ini file that includes the following parameter settings:

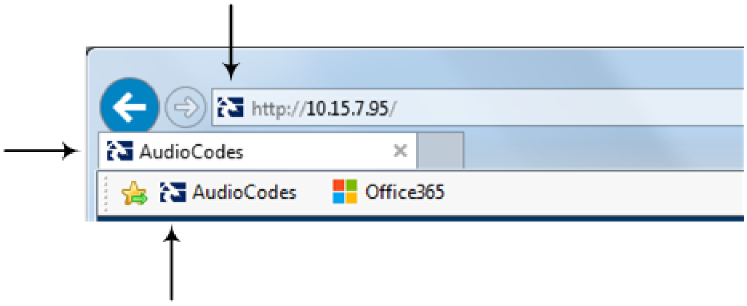
```
UseProductName = 1
UserProductName = < name >
```

2. Load the ini file using the Auxiliary Files page (see Loading Auxiliary Files on page 647).
3. Reset the device with a save-to-flash for your settings to take effect.

6.2.3 Customizing the Favicon

You can replace the default favicon (i.e., AudioCodes) with your own personalized favicon. Depending on the browser, the favicon is displayed in various areas of your browser, for example, in the URL address bar, on the page tab, and when bookmarked:

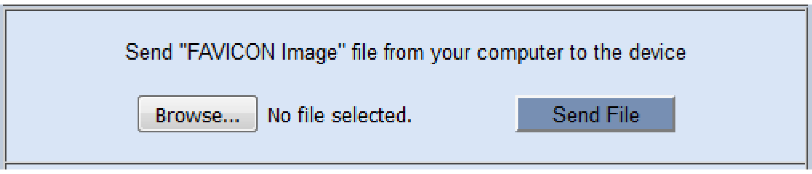
Figure 6-19: Favicon Display in Browser



To customize the favicon:

1. Save your new favicon file (.ico) in a folder on the same PC that you are using to access the device's Web interface.
2. In your browser's URL address field, append the case-sensitive suffix "/AdminPage" to the device's IP address (e.g., http://10.1.229.17/AdminPage).
3. Log in with your credentials; the Admin page appears.
4. On the left pane, click **Image Load to Device**; the right pane displays the following:

Figure 6-20: Customizing Favicon



Send "FAVICON Image" file from your computer to the device

5. Use the **Browse** button to select your favicon file, and then click **Send File**; the device loads the image file.
6. On the left pane, click **Back to Main** to exit the Admin page.

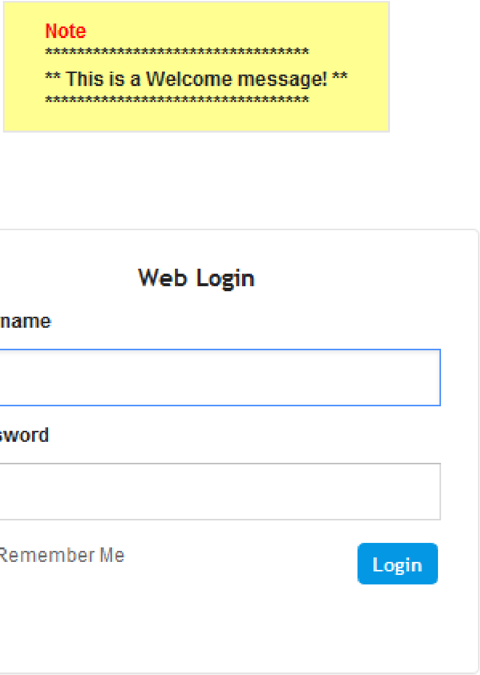
7. Reset the device with a save-to-flash for your settings to take effect.

> **Note:**
>
> - The logo image file type can be ICO, GIF, or PNG.
> - The maximum size of the image file can be 16 Kbytes.
> - Ignore the **ini Parameters** option, which is located on the left pane of the Admin page.

6.2.4 Creating a Login Welcome Message

You can create a personalized welcome message that is displayed on the Web Login screen. The message always begins with the title "Note" and has a color background, as shown in the example below:

Figure 6-21: Creating Login Welcome Message



Note
****
** This is a Welcome message! **
****

To create a login welcome message:

1. Create an ini file that includes the WelcomeMessage table parameter. Use the parameter to configure your message, where each index row is a line in your message, for example:

```
[WelcomeMessage ]
FORMAT WelcomeMessage_Index = WelcomeMessage_Text;
WelcomeMessage 1 = "*********************************";
WelcomeMessage 2 = "** This is a Welcome message! **";
WelcomeMessage 3 = "*********************************";
[\WelcomeMessage]
```

2. Load the ini file using the Auxiliary Files page (see Loading Auxiliary Files on page 647).
3. Reset the device with a save-to-flash for your settings to take effect.

➢ **To remove the welcome message:**

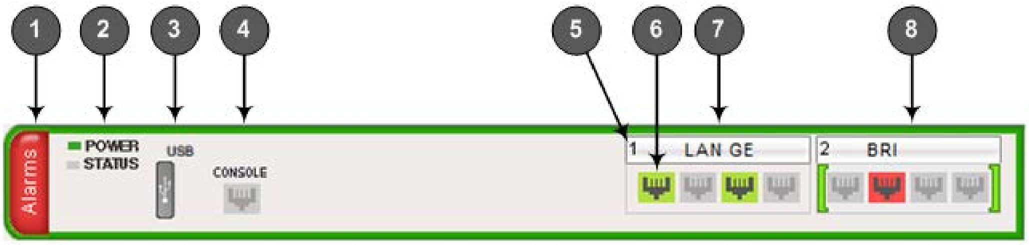
1. Load an empty ini file, using the Auxiliary Files page.
2. Reset the device with a save-to-flash for your settings to take effect.

6.3 Viewing the Home Page

The Home page is displayed when you access the device's Web interface. The Home page provides you with a graphical display of the device's front panel, showing color-coded status icons for various operations device.

➢ **To access the Home page:**

- On the toolbar, click the **Home** 🏠 icon.



In addition to the color-coded status information depicted on the graphical display of the device, the Home page displays various read-only information in the General Information pane:

- **IP Address:** IP address of the device
- **Subnet Mask:** Subnet mask address of the device
- **Default Gateway Address:** Default gateway used by the device
- **BRI Port Number:** Number of BRI ports (depending on ordered hardware configuration)
- **Firmware Version:** Software version running on the device
- **Protocol Type:** Signaling protocol currently used by the device (i.e. SIP)

- **Gateway Operational State:**
  - "LOCKED": device is locked (i.e. no new calls are accepted)
  - "UNLOCKED": device is not locked
  - "SHUTTING DOWN": device is currently shutting down

  To perform these operations, see "Basic Maintenance" on page 641.

The table below describes the areas of the Home page.

Table 6-8: Home Page Description

<table>
<tr><th>Item #</th><th colspan="2">Description</th></tr>
<tr><td>1</td><td colspan="2">Displays the highest severity of an active alarm raised (if any) by the device:
▪ Green = No alarms
▪ Red = Critical alarm
▪ Orange = Major alarm
▪ Yellow = Minor alarm
To view active alarms, click the Alarms area to open the Active Alarms page (see Viewing Active Alarms on page 709).</td></tr>
<tr><td>2</td><td colspan="2">STATUS LED displaying the operating status.</td></tr>
<tr><td>3</td><td colspan="2">USB port for USB storage services.</td></tr>
<tr><td>4</td><td colspan="2">RS-232 interface port (RJ-45).</td></tr>
<tr><td>5</td><td colspan="2">Module number of LAN or telephony interfaces</td></tr>
<tr><td>6</td><td colspan="2">Ethernet LAN module with port status icons:
▪ 🟩 (green): Link is working
▪ ⬜ (gray): Link is not configured
▪ 🟥 (red): Link error
To view detailed port information, click the port icon (see Viewing Ethernet Port Information on page 706).</td></tr>
<tr><td rowspan="8">7</td><td colspan="2">Port (trunk or channel) status icon.</td></tr>
<tr><td>Icon</td><td>Trunk Description
(Digital Module)</td></tr>
<tr><td>⬜
(gray)</td><td>Disable: Trunk not configured (not in use)</td></tr>
<tr><td>🟩
(green)</td><td>Active - OK: Trunk synchronized</td></tr>
<tr><td>🟨
(yellow)</td><td>RAI Alarm: Remote Alarm Indication (RAI), also known as the Yellow Alarm</td></tr>
<tr><td>🟥
(red)</td><td>LOS/LOF Alarm: Loss due to LOS (Loss of Signal) or LOF (Loss of Frame)</td></tr>
<tr><td>🟦
(blue)</td><td>AIS Alarm: Alarm Indication Signal (AIS), also known as the Blue Alarm</td></tr>
<tr><td>🟧
(orange)</td><td>D-Channel Alarm: D-channel alarm</td></tr>
</table>

Item #	Description	
	🟧 (dark orange)	NFAS Alarm
	If you click a port, a shortcut menu appears with commands allowing you to do the following: ▪ Port Settings: Displays trunk status (see Viewing Trunk and Channel Status on page 721) ▪ Update Port Info: Assigns a name to the port (see Assigning a Port Name)	

6.4 Configuring Web User Accounts

Web user accounts define users for the Web interface and CLI. User accounts permit login access to these interfaces as well as different levels of read and write privileges. Thus, user accounts prevent unauthorized access to these interfaces, permitting access only to users with correct credentials (i.e., username and password).

Each user account is based on the following:

- **Username and password:** Credentials that enable authorized login access to the Web interface.
- **User level (user type):** Access privileges specifying what the user can view in the Web interface and its read/write privileges. The table below describes the different types of Web user account access levels:

Table 6-9: Web User Access Levels and Privileges

User Level	Numeric Representation in RADIUS	Privileges
Security Administrator	200	Read / write privileges for all pages. It can create all user types and is the only one that can create the first Master user. **Note:** At least one Security Administrator user must exist.
Master	220	Read / write privileges for all pages. Can create all user types, including additional Master users and Security Administrators. It can delete all users except the last Security Administrator.
Administrator	100	Read / write privileges for all pages, except security-related pages (read-only).
Monitor	50	No access to security-related and file-loading pages; read-only access to all other pages.
No Access	0	No access to any page. **Note:** This access level is not applicable when using advanced Web user account configuration in the Web Users table.

By default, the device is pre-configured with the following two Web user accounts:

Table 6-10: Pre-configured Web User Accounts

User Access Level	Username (Case-Sensitive)	Password (Case-Sensitive)
Security Administrator	Admin	Admin
Monitor	User	User

After you log in to the Web interface, the username is displayed on the toolbar.

Notes:

- For security, it's recommended that you change the default username and password of the pre-configured users (i.e., Security Administrator and Monitor users).
- The Security Administrator user can change all attributes of all Web user accounts. Web users with access levels other than Security Administrator can change only their username and password.
- To restore the two Web user accounts to default settings (usernames and passwords), set the *ini* file parameter ResetWebPassword to 1.
- To log in to the Web interface with a different Web user, click the **Log off** button and then login with with a different username and password.
- You can set the entire Web interface to read-only (regardless of Web user access levels) using the *ini* file parameter DisableWebConfig (see "Web and Telnet Parameters" on page 813).
- You can define additional Web user accounts using a RADIUS server (see "RADIUS Authentication" on page 227).

6.4.1 Basic User Accounts Configuration

This section describes basic Web user account configuration. This is relevant only if the two default, pre-configured Web user accounts--Security Administrator ("Admin") and Monitor ("User")--are sufficient for your management scheme.

The Web user account parameters that can be modified depends on the access level of the currently logged-in Web user:

Table 6-11: Allowed Modifications per Web User Level

Logged-in User	Web User Level	Allowed Modifications
Security Administrator	(Default) Security Administrator	Username and password
	Monitor	Username, password, and access level
Monitor	(Default) Security Administrator	None
	Monitor	Username and password

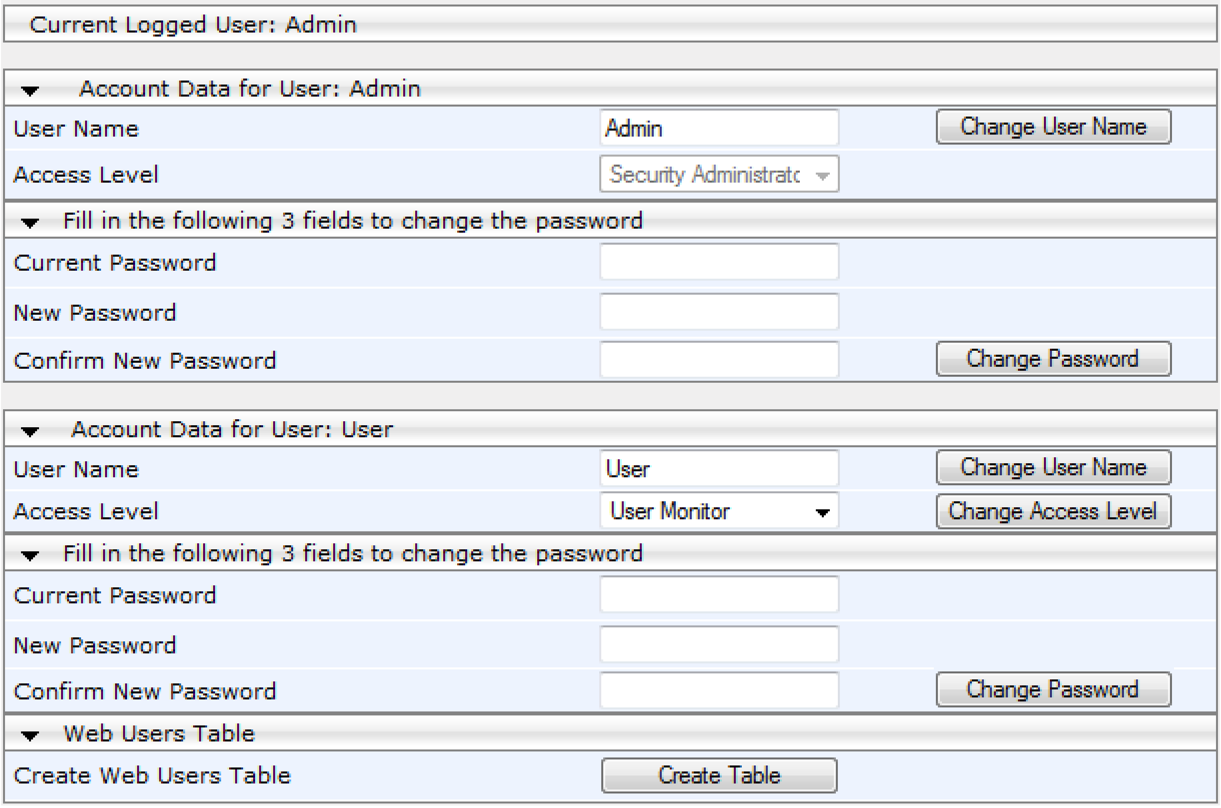
Notes:

- The username and password can be a string of up to 19 characters and are case-sensitive.
- When only the basic user accounts are being used, up to two users can be concurrently logged in to the Web interface, and they can be the same user.

➢ To configure the two pre-configured Web user accounts:

1. Open the Web User Accounts page (**Configuration** tab > **System** menu > **Web User Accounts**). If you are logged in as Security Administrator, both Web user accounts are displayed (as shown below). If you are logged in with the second user account, only the details of this user account are displayed.

Figure 6-22: WEB User Accounts Page (for Users with 'Security Administrator' Privileges)



Current Logged User: Admin

▼ Account Data for User: Admin

User Name	Admin	Change User Name
Access Level	Security Administratc ▼	

▼ Fill in the following 3 fields to change the password

Current Password		
New Password		
Confirm New Password		Change Password

▼ Account Data for User: User

User Name	User	Change User Name
Access Level	User Monitor ▼	Change Access Level

▼ Fill in the following 3 fields to change the password

Current Password		
New Password		
Confirm New Password		Change Password

▼ Web Users Table

Create Web Users Table	Create Table

2. To change the username of an account:
   a. In the 'User Name' field, enter the new user name.
   b. Click **Change User Name**; if you are currently logged in to the Web interface with this account, the 'Web Login' dialog box appears.
   c. Log in with your new user name.
3. To change the password of an account:
   a. In the 'Current Password' field, enter the current password.
   b. In the 'New Password' and 'Confirm New Password' fields, enter the new password.
   c. Click **Change Password**; if you are currently logged in to the Web interface with this account, the 'Web Login' dialog box appears.
   d. Log in with your new password.

4. To change the access level of the optional, second account:
   a. Under the **Account Data for User: User** group, from the 'Access Level' drop-down list, select a new access level user.
   b. Click **Change Access Level**; the new access level is applied immediately.

6.4.2 Advanced User Accounts Configuration

The Web Users table lets you configure advanced Web user accounts. This configuration is relevant only if you need the following management schemes:

- Enhanced security settings per Web user (e.g., limit session duration)
- More than two Web user accounts (up to 10 Web user accounts)
- Master users

> **Notes:**
>
> - Only the Security Administrator user can **initially** access the Web Users table. Admin users have read-only privileges in the Web Users table. Monitor users have no access to this table.
> - Only Security Administrator and Master users can add, edit, or delete users.
> - For advanced user accounts, up to five users can be concurrently logged in to the Web interface, and they can be the same user.
> - If you delete a user who is currently in an active Web session, the user is immediately logged off by the device.
> - All user types can change their own passwords. This is done in the Web Security Settings page (see "Configuring Web Security Settings" on page 77).
> - To remove the Web Users table and revert to the Web User Accounts page with the pre-configured, default Web user accounts, set the ResetWebPassword *ini* file parameter to 1. This also deletes all other Web users.
> - Once the Web Users table is accessed, Monitor users and Admin users can change only their passwords in the Web Security Settings page (see "Configuring Web Security Settings" on page 77). The new password must have at least four different characters than the previous password. (The Security Administrator users and Master users can change their passwords in the Web Users table and in the Web Security Settings page.)

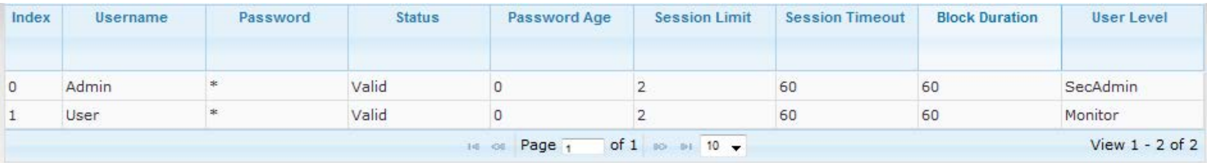
The following procedure describes how to configure Web users through the Web interface. You can also configure it through CLI (configure system > create-users-table).

➢ **To add Web user accounts with advanced settings:**

1. Open the Web Users table:
   - Upon initial access:
     a. Open the Web User Accounts page (**Configuration** tab > **System** menu > **Web User Accounts**).
     b. Under the **Web Users Table** group, click the **Create Table** button.
   - Subsequent access: **Configuration** tab > **System** menu > **Web User Accounts**.

The Web Users table appears, listing the two default, pre-configured Web use accounts - Security Administrator ("Admin") and Monitor ("User"):

Figure 6-23: Web Users Table Page

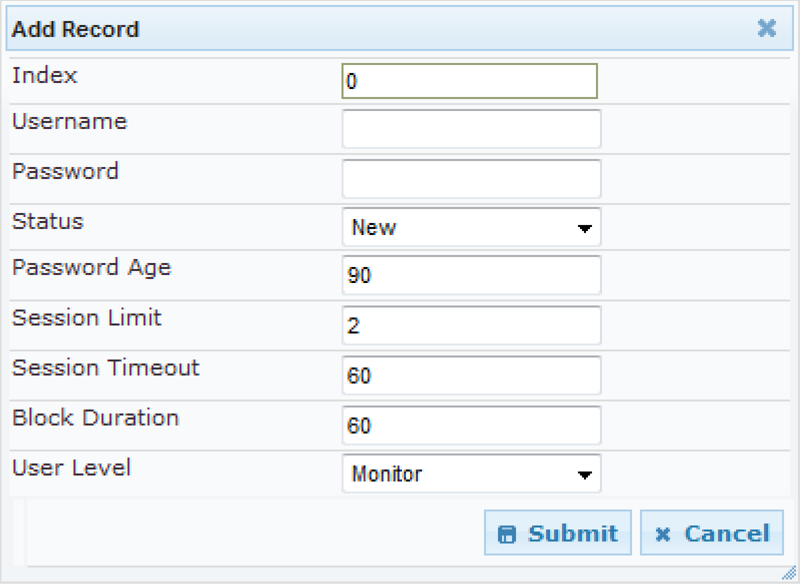


Index	Username	Password	Status	Password Age	Session Limit	Session Timeout	Block Duration	User Level
0	Admin	*	Valid	0	2	60	60	SecAdmin
1	User	*	Valid	0	2	60	60	Monitor

Page 1 of 1 10 View 1 - 2 of 2

2. Click **Add**; the following dialog box is displayed:

Figure 6-24: Web Users Table - Add Record Dialog Box



Add Record

Index	0
Username	
Password	
Status	New
Password Age	90
Session Limit	2
Session Timeout	60
Block Duration	60
User Level	Monitor

3. Configure a Web user according to the parameters described in the table below.
4. Click **Add**, and then save ("burn") your settings to flash memory.

Table 6-12: Web User Table Parameter Descriptions

Parameter	Description
Index	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Username `user`	Defines the Web user's username. The valid value is a string of up to 40 alphanumeric characters, including the period ".", underscore "_", and hyphen "-" signs.
Password `password`	Defines the Web user's password. The valid value is a string of 8 to 40 ASCII characters. To ensure strong passwords, adhere to the following password complexity requirements: ▪ Contain at least eight characters. ▪ Contain at least two letters that are upper case (e.g., A). ▪ Contain at least two letters that are lower case (e.g., a). ▪ Contain at least two numbers (e.g., 4). ▪ Contain at least two symbols (non-alphanumeric characters) (e.g., $, #, %). ▪ No spaces.

Parameter	Description
	- Contain at least four new characters that were not used in the previous password. **Note:** To enforce the password complexity requirements mentioned above, configure the EnforcePasswordComplexity to 1.
Status `status`	Defines the status of the Web user. - New = (Default) User is required to change its password on the next login. When the user logs in to the Web interface, the user is immediately prompted to change the current password. - Valid = User can log in to the Web interface as normal. - Failed Login = This state is automatically set for users that exceed a user-defined number of failed login attempts, set by the 'Deny Access on Fail Count' parameter (see "Configuring Web Session and Access Settings" on page 78). These users can log in only after a user-defined timeout configured by the 'Block Duration' parameter (see below) or if their status is changed (to New or Valid) by a System Administrator or Master. - Inactivity = This state is automatically set for users that have not accessed the Web interface for a user-defined number of days, set by the 'User Inactivity Timer' (see "Configuring Web Session and Access Settings" on page 78). These users can only log in to the Web interface if their status is changed (to New or Valid) by a System Administrator or Master. **Notes:** - The Inactivity status is applicable only to Admin and Monitor users; System Administrator and Master users can be inactive indefinitely. - For security, it is recommended to set the status of a newly added user to New in order to enforce password change.
Password Age `password-age`	Defines the duration (in days) of the validity of the password. When this duration elapses, the user is prompted to change the password; otherwise, access to the Web interface is blocked. The valid value is 0 to 10000, where 0 means that the password is always valid. The default is 90.
Session Limit `session-limit`	Defines the maximum number of concurrent Web interface sessions allowed for the specific user. For example, if configured to 2, the same user account can be logged into the device’s Web interface (i.e., same username-password combination) from two different management stations (i.e., IP addresses) at any one time. Once the user logs in, the session is active until the user logs off (by clicking the **Log off** icon on the toolbar) or until the session expires if the user is inactive for a user-defined duration (see the ‘Session Timeout’ parameter below). The valid value is 0 to 5. The default is 2. **Note:** Up to 5 users can be concurrently logged in to the Web interface.
Session Timeout `session-timeout`	Defines the duration (in minutes) of inactivity of a logged-in user in the Web interface, after which the user is automatically logged off the Web session. In other words, the session expires when the user has not performed any operations (activities) in the Web interface for the configured timeout duration. The valid value is 0 to 100000. A value of 0 means no timeout. The default value is according to the settings of the WebSessionTimeout global parameter (see "Configuring Web Session and Access Settings" on page 78).

Parameter	Description
Block Duration `block-duration`	Defines the duration (in seconds) for which the user is blocked when the user exceeds a user-defined number of failed login attempts. This is configured by the 'Deny Access On Fail Count' parameter (see "Configuring Web Session and Access Settings" on page 78). The valid value is 0 to 100000, where 0 means that the user can do as many login failures without getting blocked. The default is according to the settings of the 'Deny Authentication Timer' parameter (see "Configuring Web Session and Access Settings" on page 78). **Note:** The 'Deny Authentication Timer' parameter relates to failed Web logins from specific IP addresses.
User Level `privilege`	Defines the user's access level. ▪ Monitor = (Default) Read-only user. This user can only view Web pages and access to security-related pages is denied. ▪ Administrator = Read/write privileges for all pages, except security-related pages including the Web Users table where this user has only read-only privileges. ▪ Security Administrator = Read/write privileges for all pages. This user is the Security Administrator. ▪ Master = Read/write privileges for all pages. This user also functions as a security administrator. **Notes:** ▪ At least one Security Administrator must exist. The last remaining Security Administrator cannot be deleted. ▪ The first Master user can be added only by a Security Administrator user. ▪ Additional Master users can be added, edited and deleted only by Master users. ▪ If only one Master user exists, it can be deleted only by itself. ▪ Master users can add, edit, and delete Security Administrators (but cannot delete the last Security Administrator). ▪ Only Security Administrator and Master users can add, edit, and delete Administrator and Monitor users.

6.5 Displaying Login Information upon Login

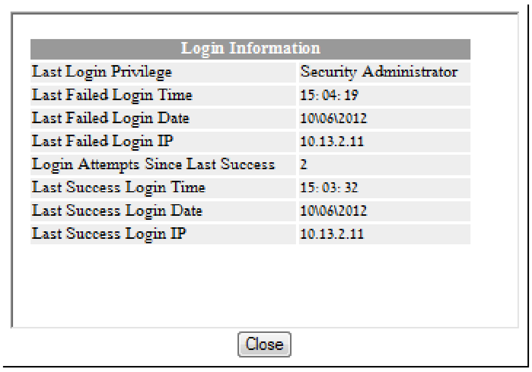
The device can display login information immediately upon Web login.

➢ **To enable display of user login information upon a successful login:**

1. Open the Web Security Settings page (**Configuration** tab > **System** menu > **Management** > **Web Security Settings**).
2. From the 'Display Login Information' drop-down list, select **Yes**.
3. Click **Submit**.

Once enabled, the Login Information window is displayed upon a successful login, as shown in the example below:

Figure 6-25: Login Information Window



Login Information	
Last Login Privilege	Security Administrator
Last Failed Login Time	15: 04: 19
Last Failed Login Date	10\06\2012
Last Failed Login IP	10.13.2.11
Login Attempts Since Last Success	2
Last Success Login Time	15: 03: 32
Last Success Login Date	10\06\2012
Last Success Login IP	10.13.2.11

6.6 Configuring Web Security Settings

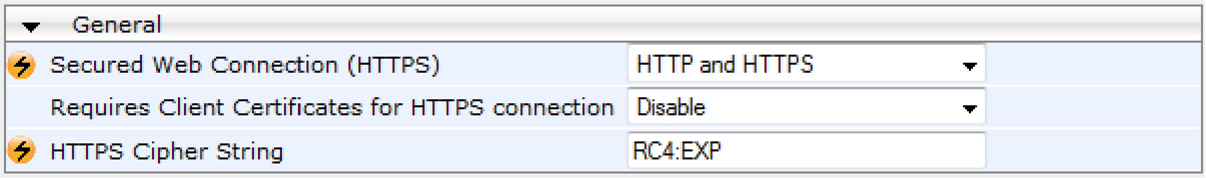
This section describes how to secure Web-based management.

6.6.1 Configuring Secured (HTTPS) Web

By default, the device allows remote management (client) through HTTP and HTTPS. However, you can enforce secure Web access communication by configuring the device to accept only HTTPS.

➢ **To configure secure Web access:**

1. Open the Web Security Settings page (**Configuration** tab > **System** menu > **Management** > **Web Security Settings**).



General

Secured Web Connection (HTTPS)	HTTP and HTTPS
Requires Client Certificates for HTTPS connection	Disable
HTTPS Cipher String	RC4:EXP

2. From the 'Secured Web Connection (HTTPS)' drop-down list, select **HTTPS Only**.
3. To enable two-way authentication whereby both management client and server are authenticated using X.509 certificates, from the 'Requires Client Certificates for HTTPS connection' drop-down list, select **Enable**.
4. In the 'HTTPS Cipher String' field, enter the cipher string for HTTPS (in OpenSSL cipher list format).
5. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

For more information on secured Web-based management including TLS certificates, see "TLS for Remote Device Management" on page 118.

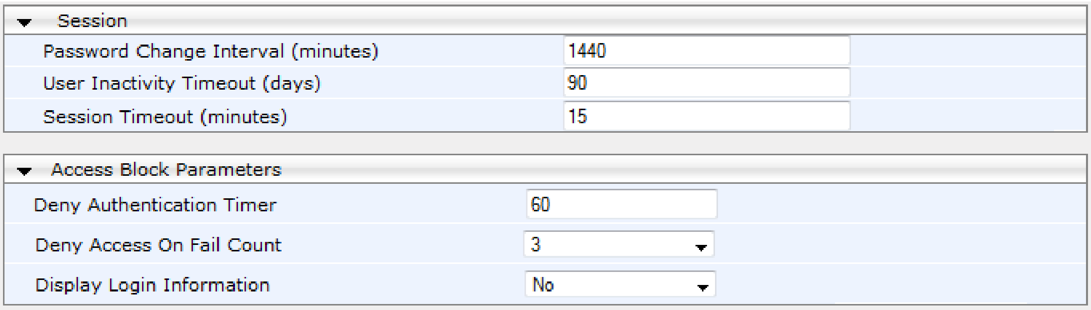
6.6.2 Configuring Web Session and Access Settings

You can configure security features related to Web user sessions and access.

➢ **To configure Web user sessions and access security:**

1. Open the Web Security Settings page (**Configuration** tab > **System** menu > **Management** > **Web Security Settings**).

Figure 6-26: Configuring Security Related to Web User Sessions and Access



Session

Password Change Interval (minutes)	1440
User Inactivity Timeout (days)	90
Session Timeout (minutes)	15

Access Block Parameters

Deny Authentication Timer	60
Deny Access On Fail Count	3
Display Login Information	No

2. Web user sessions:
   - 'Password Change Interval': Duration of the validity of Web login passwords. When the duration expires, the Web user must change the password in order to log in again.
   - 'User Inactivity Timeout': If the user has not logged into the Web interface within this defined duration, the status of the user becomes inactive and the user can no longer access the Web interface. The user can only log in to the Web interface if its status is changed (to "New" or "Valid") by an Administrator or a Master user.
   - 'Session Timeout': Duration of Web inactivity (i.e., no actions are performed in the Web interface) of a logged-in user, after which the Web session expires and the user is automatically logged off the Web interface and needs to log in again to continue the session. You can also configure the functionality per user in the Web Users table (see Advanced User Accounts Configuration on page 73), which overrides this global setting.
3. Web user access:
   - 'Deny Authentication Timer': Interval (in seconds) that the user needs to wait before the user can attempt to log in from the same IP address after reaching the maximum number of failed login attempts (see next step).
   - 'Deny Access On Fail Count': Number of failed login attempts after which the user is prevented access to the device for a user-defined duration (previous step).
4. Click **Submit**.

For a detailed description of the above parameters, see "Web Parameters" on page 814.

6.7 Web Login Authentication using Smart Cards

You can enable Web login authentication using certificates from a third-party, common access card (CAC) with user identification. When a user attempts to access the device through the Web browser (HTTPS), the device retrieves the Web user's login username (and other information, if required) from the CAC. The user attempting to access the device is only required to provide the login password. Typically, a TLS connection is established between the CAC and the device's Web interface, and a RADIUS server is implemented to authenticate the password with the username. Therefore, this feature implements a two-factor authentication - what the user has (i.e., the physical card) and what the user knows (i.e., the login password).

This feature is enabled using the EnableMgmtTwoFactorAuthentication parameter.

> **Note:** For specific integration requirements for implementing a third-party smart card for Web login authentication, contact your AudioCodes representative.

➢ To log in to the Web interface using CAC:

1. Insert the Common Access Card into the card reader.
2. Access the device using the following URL: https://<host name or IP address>; the device prompts for a username and password.
3. Enter the password only. As some browsers require that the username be provided, it's recommended to enter the username with an arbitrary value.

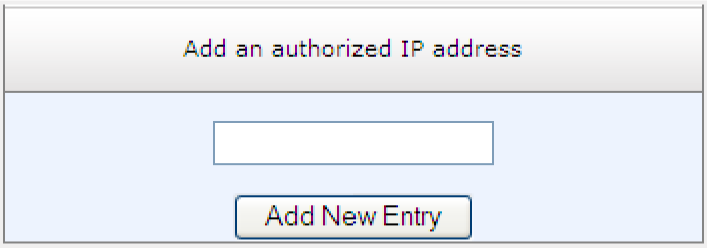
6.8 Configuring Web and Telnet Access List

The Web & Telnet Access List page is used to define IP addresses (up to ten) that are permitted to access the device's Web, Telnet, and SSH interfaces. Access from an undefined IP address is denied. If no IP addresses are defined, this security feature is inactive and the device can be accessed from any IP address. The Web and Telnet Access List can also be defined using the *ini* file parameter WebAccessList_x (see "Web Parameters" on page 814).

➢ To add authorized IP addresses for Web, Telnet, and SSH interfaces access:

1. Open the Web & Telnet Access List page (**Configuration** tab > **System** menu > **Management** > **Web & Telnet Access List**).

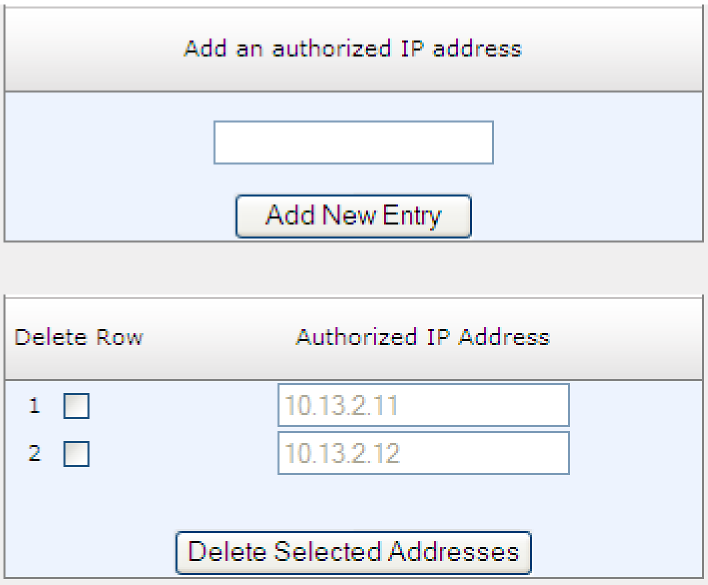
Figure 6-27: Web & Telnet Access List Page - Add New Entry



2. To add an authorized IP address, in the 'Add an authorized IP address' field, enter the required IP address, and then click **Add New Entry**; the IP address you entered is

added as a new entry to the Web & Telnet Access List table.

Figure 6-28: Web & Telnet Access List Table



Add an authorized IP address

Delete Row		Authorized IP Address
1	□	10.13.2.11
2	□	10.13.2.12

3. To delete authorized IP addresses, select the Delete Row check boxes corresponding to the IP addresses that you want to delete, and then click **Delete Selected Addresses**; the IP addresses are removed from the table and these IP addresses can no longer access the Web and Telnet interfaces.
4. To save the changes to flash memory, see "Saving Configuration" on page 643.

Notes:

- The first authorized IP address in the list must be your PC's (terminal) IP address; otherwise, access from your PC is denied.
- Delete your PC's IP address last from the 'Web & Telnet Access List page. If it is deleted before the last, subsequent access to the device from your PC is denied.

7 CLI-Based Management

This chapter provides an overview of the CLI-based management and provides configuration relating to CLI management.

Notes:

- For security, CLI is disabled by default.
- The CLI can only be accessed by management users with the following user levels:
  - √ Administrator
  - √ Security Administrator
  - √ Master
- For a description of the CLI commands, refer to the CLI Reference Guide.

7.1 Getting Familiar with CLI

This section describes the basic structure of the device's CLI, which you may need to know before configuring the device through CLI.

7.1.1 Understanding Configuration Modes

Before you begin your CLI session, you should familiarize yourself with the CLI command modes. Each command mode provides different levels of access to commands, as described below:

- **Basic command mode**: This is the initial mode that is accessed upon a successful CLI login authentication. Any user level can access this mode and thus, the commands supported by this command tier are limited, as is interaction with the device itself. This mode allows you to view various information (using the show commands) and activate various debugging capabilities.

  ```
  Welcome to AudioCodes CLI
  Username: Admin
  Password:
  >
  ```

  The Basic mode prompt is ">".

- **Enable command mode:** This mode is the high-level tier in the command hierarchy, one step up from the Basic Mode. A password ("Admin", by default) is required to access this mode **after** you have accessed the Basic mode. This mode allows you to configure all the device's settings. The Enable mode is accessed by typing the following commands:

  ```
  > enable
  Password: <Enable mode password>
  #
  ```

  The Enable mode prompt is "#".

Note: The default password for accessing the Enable mode is "Admin" (case-sensitive). To change the password, use the CLIPrivPass ini file parameter.

The Enable mode groups the configuration commands under the following command sets:

- **config-system:** Provides the general and system related configuration commands, for example, Syslog configuration. This set is accessed by typing the following command:

```
# configure system
(config-system)#
```

- **config-voip:** Provides the VoIP-related configuration commands, for example, SIP and media parameters, and VoIP network interface configuration. This set is accessed by typing the following command:

```
# configure voip
(config-voip)#
```

7.1.2 Using CLI Shortcuts

The CLI provides several editing shortcut keys to help you configure your device more easily, as listed in the table below.

Table 7-1: CLI Editing Shortcut keys

<table>
<tr><th>Shortcut Key</th><th>Description</th></tr>
<tr><td><b>Up</b> arrow key</td><td>Retypes the previously entered command. Continuing to press the <b>Up</b> arrow key cycles through all commands entered, starting with the most recent command.</td></tr>
<tr><td><b>&lt;Tab&gt;</b> key</td><td>Pressing the <b>&lt;Tab&gt;</b> key after entering a partial (but unique) command automatically completes the command, displays it on the command prompt line, and waits for further input.
Pressing the <b>&lt;Tab&gt;</b> key after entering a partial and not unique command displays all completing options.</td></tr>
<tr><td><b>?</b> (question mark)</td><td>
<ul><li>Displays a list of all subcommands in the current mode, for example:</li></ul>
<pre>
(config-voip)# voip-network ?
  dns   Enter voip-network dns
  ip-group IP Group table
  nat-translation   NATTranslationtable
...
</pre>
<ul><li>Displays a list of available commands beginning with certain letter(s), for example:</li></ul>
<pre>
(config)# voip-network d?
  dns   Enter voip-network dns
</pre>
<ul><li>Displays syntax help for a specific command by entering the command, a space, and then a question mark (?). This includes the range of valid values and a brief description of the next parameter expected for that particular command. For example:</li></ul>
<pre>
(config)# voip-network dns srv2ip ?
  [0-9]   index
</pre>
If a command can be invoked (i.e., all its arguments have been entered), the question mark at its end displays "<cr>" to indicate that a carriage return (Enter) can now be entered to run the command, for example:
<pre>
(config)# logging host 10.1.1.1 ?
  <cr>
</pre>
</td></tr>
<tr><td>&lt;<b>Ctrl + A</b>&gt;</td><td>Moves the cursor to the beginning of the command line.</td></tr>
<tr><td>&lt;<b>Ctrl + E</b>&gt;</td><td>Moves the cursor to the end of the command line.</td></tr>
</table>

Shortcut Key	Description
<**Ctrl + U**>	Deletes all the characters on the command line.
auto finish	You need only enter enough letters to identify a command as unique. For example, entering "int G 0/0" at the configuration prompt provides you access to the configuration parameters for the specified Gigabit-Ethernet interface. Entering "interface GigabitEthernet 0/0" would work as well, but is not necessary.
Space Bar at the --More--prompt	Displays the next screen of output. You can configure the size of the displayed output, as described in "Configuring Displayed Output Lines in CLI Terminal Window" on page 89.

7.1.3 Common CLI Commands

The following table contains descriptions of common CLI commands.

Table 7-2: Common CLI Commands

Command	Description

do	Provides a way to execute commands in other command sets without taking the time to exit the current command set. The following example shows the **do** command, used to view the GigabitEthernet interface configuration while in the virtual-LAN interface command set:
	`(config)# interface vlan 1`
	`(conf-if-VLAN 1)# do show interfaces GigabitEthernet 0/0`

no	Undoes an issued command or disables a feature. Enter **no** before the command: `# no debug log`
activate	Activates a command. When you enter a configuration command in the CLI, the command is not applied until you enter the **activate** and **exit** commands. **Note:** Offline configuration changes require a reset of the device. A reset can be performed at the end of the configuration changes. A required reset is indicated by an asterisk (*) before the command prompt.

exit	Leaves the current command-set and returns one level up. If issued on the top level, the session ends.
	For online parameters, if the configuration was changed and no **activate** command was entered, the **exit** command applies the **activate** command automatically. If issued on the top level, the session will end:
	`(config)# exit`
	`# exit`
	`(session closed)`

display	Displays the configuration of current configuration set.
help	Displays a short help how-to string.
history	Displays a list of previously run commands.
list	Displays the available command list of the current command-set.
| <filter>	Applied to a command output. The filter should be typed after the command with a pipe mark (|). Supported filters: ▪ **include <word>** – filter (print) lines which contain <word> ▪ **exclude <word>** – filter lines which does not contain <word>

Command	Description

	▪ **grep <options>** - filter lines according to *grep* common Unix utility options
	▪ **egrep <options>** - filter lines according to e*grep* common Unix utility options
	▪ **begin <word>** – filter (print) lines which begins with <word>
	▪ **between <word1> <word2>** – filter (print) lines which are placed between <word1> and <word2>
	▪ **count** – show the output's line count
	Example:
	`# show system version | grep Number`
	`;Serial Number: 2239835;Slot Number: 1`

7.1.4 Configuring Tables through CLI

Throughout the CLI, many configuration elements are in table format where each table row is represented by an index number. When you add a new row to a table, the device automatically assigns it the next consecutive, available index number. However, you can specify a different index number.

Table rows are added using the **new** command:

```
# <table name> new
```

When you add a new table row, the device accesses the row's configuration mode. For example, if three rows are configured in the Account table (account-0, account-1, and account-2) and you then add a new row, account-3 is automatically created and its' configuration mode is accessed:

```
(config-voip)# sip-definition account new
(account-3)#
```

You can also add a new table row to any specific index number, even if a row has already been configured for that index number. The row that was previously assigned that index number is incremented to the next consecutive index number, as well as all the index rows listed below it in the table. To add a new table row to a specific index number, use the **insert** command:

```
# <table name> <index> insert
```

For example, if three rows are configured in the Account table (account-0, account-1, and account-2) and you then add a new row with index 1, the previous account-1 becomes account-2 and the previous account-2 becomes account-3, and so on. The following command is run for this example:

```
(config-voip)# sip-definition account 1 insert
```

Note: The insert table row feature is applicable only to tables that do not have "child" tables (sub-tables).

You can also change the position (index) of a configured row by moving it one row up or one row down in the table, using the following command:

```
# <table> <index to move> move-up|move-down
```

For example, to move the row at Index 1 down to Index 2 in the IP-to-IP Routing table:

```
<config-voip># sbc routing ip2ip-routing 1 move-down
```

In this example, the previous row at Index 2 is moved up to Index 1.

Note: Changing of row position is applicable only to certain tables.

7.1.5 Understanding CLI Error Messages

The CLI provides feedback on commands by displaying informative messages:

- Failure reason of a run command. The failure message is identical to the notification failure message sent by Syslog. For example, an invalid Syslog server IP address is displayed in the CLI as follows:

```
(logging)# syslog-ip 1111.1.1.1
Parameter 'SyslogServerIP' does NOT accept the IP-Address:
1111.1.1.1, illegal IPAddress.
Configuration failed
Command Failed!
```

- "Invalid command" message: The command may not be valid in the current command mode, or you may not have entered sufficient characters for the command to be recognized. Use "?" to determine your error.
- "Incomplete command" message: You may not have entered all of the pertinent information required to make the command valid. Use "?" to determine your error.

7.2 Enabling CLI

By default, access to the device's CLI through Telnet and SSH is disabled. This section describes how to enable these protocols.

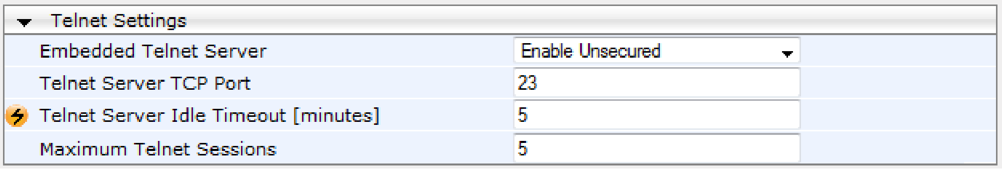
7.2.1 Enabling Telnet for CLI

The following procedure describes how to enable Telnet. You can enable a secured Telnet that uses Secure Socket Layer (SSL) where information is not transmitted in the clear. If SSL is used, a special Telnet client is required on your PC to connect to the Telnet interface over a secured connection; examples include C-Kermit for UNIX and Kermit-95 for Windows.

For security, some organizations require the display of a proprietary notice upon starting a Telnet session. You can use the configuration ini file parameter, WelcomeMessage to configure such a message (see "Creating a Login Welcome Message" on page 62).

➢ **To enable Telnet:**

1. Open the Telnet/SSH Settings page (**Configuration** tab > **System** menu > **Management** > **Telnet/SSH Settings**).



Telnet Settings

Embedded Telnet Server	Enable Unsecured
Telnet Server TCP Port	23
Telnet Server Idle Timeout [minutes]	5
Maximum Telnet Sessions	5

2. From the 'Embedded Telnet Server' drop-down list, select **Enable Unsecured** or **Enable Secured** (i.e, SSL).
3. In the 'Telnet Server TCP Port' field, enter the port number for the embedded Telnet server.
4. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

For a detailed description of the Telnet parameters, see "Telnet Parameters" on page 818.

7.2.2 Enabling SSH with RSA Public Key for CLI

Unless configured for TLS, Telnet is not secure as it requires passwords to be transmitted in clear text. To overcome this, Secure SHell (SSH) is used, which is the de-facto standard for secure CLI. SSH 2.0 is a protocol built above TCP, providing methods for key exchange, authentication, encryption, and authorization.

SSH requires appropriate client software for the management PC. Most Linux distributions have OpenSSH pre-installed; Windows-based PCs require an SSH client software such as PuTTY, which can be downloaded from http://www.chiark.greenend.org.uk/~sgtatham/putty/.

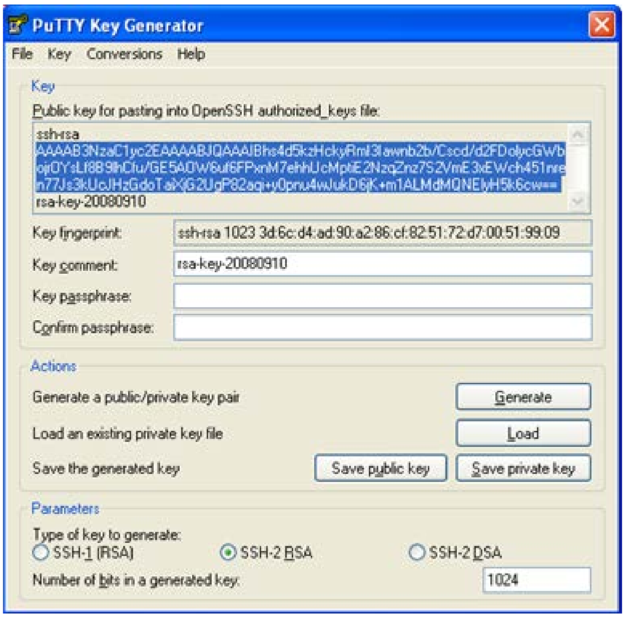
By default, SSH uses the same username and password as the Telnet and Web server. SSH supports 1024/2048-bit RSA public keys, providing carrier-grade security. Follow the instructions below to configure the device with an administrator RSA key as a means of strong authentication.

➢ **To enable SSH and configure RSA public keys for Windows (using PuTTY SSH software):**

1. Start the PuTTY Key Generator program, and then do the following:
   a. Under the 'Parameters' group, do the following:
      - Select the **SSH-2 RSA** option.
      - In the 'Number of bits in a generated key' field, enter "1024" bits.

   b. Under the 'Actions' group, click **Generate** and then follow the on-screen instructions.
   c. Under the 'Actions' group, click **Save private key** to save the new private key to a file (*.ppk) on your PC.
   d. Under the 'Key' group, select the displayed encoded text between "ssh-rsa" and "rsa-key-….", as shown in the example below:

Figure 7-1: Selecting Public RSA Key in PuTTY



Public key for pasting into OpenSSH authorized_keys file:

```
ssh-rsa
AAAAB3NzaC1yc2EAAAABJQAAAIBhs4d5kzHckyRml3lawnb2b/Csod/d2FDolycGWb
ojOYsLf8B9hCfu/GE5A0W6uf6FPxnM7ehhUcMptE2NzqZnz7S2VmE3xEWch451nre
n77Js3kUcJHzGdoTaXjG2UgP82aqi+y0pnu4wJukD6jK+m1ALMdMQNElyH5k6cw==
rsa-key-20080910
```

Key fingerprint: ssh-rsa 1023 3d:6c:d4:ad:90:a2:86:cf:82:51:72:d7:00:51:99:09

Key comment: rsa-key-20080910

Key passphrase:

Confirm passphrase:

Generate a public/private key pair

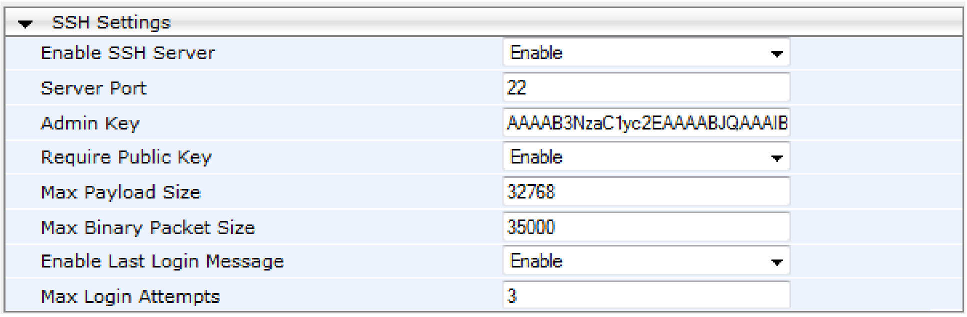
Load an existing private key file

Save the generated key

Type of key to generate:
○ SSH-1 (RSA) ● SSH-2 RSA ○ SSH-2 DSA

Number of bits in a generated key: 1024

2. Open the Telnet/SSH Settings page (**Configuration** tab > **System** menu > **Management** > **Telnet/SSH Settings**), and then do the following:
   a. Set the 'Enable SSH Server' parameter to **Enable**.
   b. Paste the public key that you copied in Step 1.d into the 'Admin Key' field, as shown below:



SSH Settings

Enable SSH Server	Enable
Server Port	22
Admin Key	AAAAB3NzaC1yc2EAAAABJQAAAIB
Require Public Key	Enable
Max Payload Size	32768
Max Binary Packet Size	35000
Enable Last Login Message	Enable
Max Login Attempts	3

c. For additional security, you can set the 'Require Public Key' to **Enable**. This ensures that SSH access is only possible by using the RSA key and not by using user name and password.

d.

e. Configure the other SSH parameters as required. For a description of these parameters, see "SSH Parameters" on page 857.

f. Click **Submit**.

3. Start the PuTTY Configuration program, and then do the following:

   a. In the 'Category' tree, drill down to **Connection**, then **SSH**, and then **Auth**; the 'Options controlling SSH authentication' pane appears.

   b. Under the 'Authentication parameters' group, click **Browse** and then locate the private key file that you created and saved in Step 4.

4. Connect to the device with SSH using the username "Admin"; RSA key negotiation occurs automatically and no password is required.

➢ To configure RSA public keys for Linux (using OpenSSH 4.3):

1. Run the following command to create a new key in the admin.key file and to save the public portion to the admin.key.pub file:

```
ssh-keygen -f admin.key -N "" -b 1024
```

2. Open the admin.key.pub file, and then copy the encoded string from "ssh-rsa" to the white space.

3. Open the Telnet/SSH Settings page (**Configuration** tab > **System** menu > **Management** > **Telnet/SSH Settings**), and then paste the value copied in Step 2 into the 'Admin Key' field.

4. Click **Submit**.

5. Connect to the device with SSH, using the following command:

```
ssh -i admin.key xx.xx.xx.xx
```

   where *xx.xx.xx.xx* is the device's IP address. RSA-key negotiation occurs automatically and no password is required.

7.3 Configuring Maximum Telnet/SSH Sessions

You can configure the maximum number of concurrent Telnet/SSH sessions (up to five) permitted on the device.

> **Note:** Before changing the setting, make sure that not more than this number of sessions are currently active; otherwise, the new setting will not take effect.

➢ To configure the maximum number of concurrent Telnet/SSH sessions:

1. Open the Telnet/SSH Settings page (**Configuration** tab > **System** menu > **Management** > **Telnet/SSH Settings**).
2. In the 'Maximum Telnet Sessions' field, enter the maximum number of concurrent sessions.
3. Click **Submit**.

7.4 Establishing a CLI Session

The device's CLI can be accessed using any of the following methods:

- RS-232: The device can be accessed through its RS-232 serial port, by connecting a VT100 terminal to it or using a terminal emulation program (e.g., HyperTerminal) with a PC. For connecting to the CLI through RS-232, see CLI on page 41.
- **Secure SHell (SSH):** The device can be accessed through its Ethernet interface by the SSH protocol using SSH client software. A popular and freeware SSH client software is Putty, which can be downloaded from http://www.chiark.greenend.org.uk/~sgtatham/putty/download.html
- **Telnet:** The device can be accessed through its Ethernet interface by the Telnet protocol using Telnet client software.

The following procedure describes how to access the CLI through Telnet/SSH.

> **Note:** The CLI login credentials are the same as all the device's other management interfaces (such as Web interface). The default username and password is "Admin" and "Admin" (case-sensitive), respectively. For configuring login credentials, see "Configuring Web User Accounts" on page 70.

➢ To establish a CLI session with the device:

1. Connect the device to the network.
2. Establish a Telnet or SSH session using the device's OAMP IP address.
3. Log in to the session using the username and password assigned to the Admin user of the Web interface:
   a. At the Username prompt, type the username, and then press Enter:
   ```
   Username: Admin
   ```
   b. At the Password prompt, type the password, and then press Enter:
   ```
   Password: Admin
   ```

c. At the prompt, type the following, and then press Enter:

```
> enable
```

d. At the prompt, type the password again, and then press Enter:

```
Password: Admin
```

7.5 Viewing Current CLI Sessions

You can view users that are currently logged in to the device's CLI. This applies to users logged in to the CLI through RS-232 (console), Telnet, or SSH. For each logged-in user, the following is displayed: the type of interface (console, Telnet, or SSH), user's username, remote IP address from where the user logged in, and the duration (days and time) of the session. Each user is displayed with a unique index (session ID).

➢ **To view currently logged-in CLI users:**

```
# show users
[0]  console      Admin       local            0d00h03m15s
[1]  telnet       John        10.4.2.1         0d01h03m47s
[2]* ssh          Alex        192.168.121.234  12d00h02m34s
```

The current session from which this show command was run is displayed with an asterisk (*).

Note: The device can display management sessions of up to 24 hours. After this time, the duration counter is reset.

7.6 Terminating a User's CLI Session

You can terminate users that are currently logged in to the device's CLI. This applies to users logged in to the CLI through RS-232 (console), Telnet, or SSH.

➢ **To terminate the CLI session of a specific CLI user:**

```
# clear user <session ID>
```

The *session ID* is a unique identification of each currently logged in user. You can view the session ID by running the **show users** command (see "Viewing Current CLI Sessions" on page 89).

Note: The session from which the command is run cannot be terminated.

7.7 Configuring Displayed Output Lines in CLI Terminal Window

You can configure the maximum number of lines (height) displayed in the terminal window for the output of CLI commands (Telnet and SSH). The number of displayed lines can be specified from 0 to 65,535, or determined by re-sizing the terminal window by mouse-dragging the window's border.

➢ To configure a specific number of output lines:

```
(config-system)# cli-terminal
<cli-terminal># window-height [0-65535]
```

If window-height is set to 0, the entire command output is displayed. In other words, even if the output extends beyond the visible terminal window length, the --MORE-- prompt is not displayed.

➢ To configure the number of lines according to dragged terminal window:

```
(config-system)# cli-terminal
<cli-terminal># window-height automatic
```

When this mode is configured, each time you change the height of the terminal window using your mouse (i.e., dragging one of the window's borders or corners), the number of displayed output command lines is changed accordingly.

8 SNMP-Based Management

The device provides an embedded SNMP Agent that allows it to be managed by AudioCodes Element Management System (EMS) or a third-party SNMP Manager (e.g., element management system). The SNMP Agent supports standard Management Information Base (MIBs) and proprietary MIBs, enabling a deeper probe into the interworking of the device. The SNMP Agent can also send unsolicited events (SNMP traps) towards the SNMP Manager. All supported MIB files are supplied to customers as part of the release.

AudioCodes EMS is an advanced solution for standards-based management that covers all areas vital for the efficient operation, administration, management and provisioning (OAM&P) of the device. The standards-compliant EMS uses distributed SNMP-based management software, optimized to support day-to-day Network Operation Center (NOC) activities, offering a feature-rich management framework. It supports fault management, configuration and security.

This section provides configuration relating to SNMP management.

Notes:

- SNMP-based management is enabled by default.
- For more information on the device's SNMP support (e.g., SNMP traps), refer to the *SNMP User's Guide*.
- EMS support is available only if the device is installed with a Software License Key that includes this feature. For installing a Software License Key, see "Software License Key" on page 668.
- For more information on using the EMS tool, refer to the *EMS User's Manual* and *EMS Server IOM Manual*.

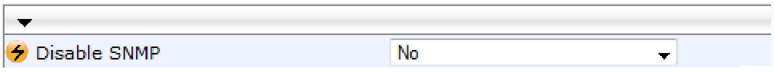
8.1 Disabling SNMP

By default, SNMP is enabled. You can change the setting, as described in the following procedure.

➢ **To disable SNMP:**

1. Open the SNMP Community String page (**Configuration** tab > **System** menu > **Management** > **SNMP** > **SNMP Community Settings**).

Figure 8-1: Disabling SNMP



Disable SNMP No

2. From the 'Disable SNMP' drop-down list (DisableSNMP parameter), select **Yes**.
3. Click **Submit**, and then reset the device with a save-to-flash for your settings to take effect.

8.2 Configuring SNMP Community Strings

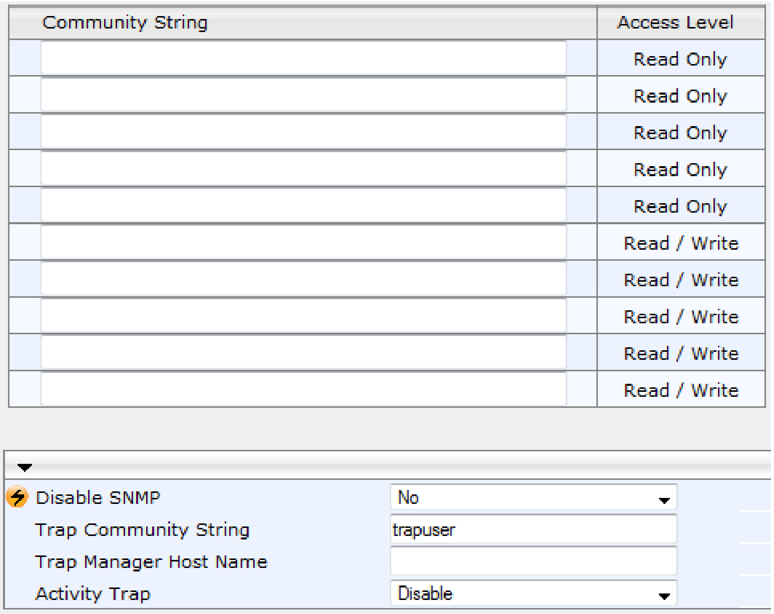
The SNMP Community String page lets you configure up to five read-only and up to five read-write SNMP community strings and to configure the community string that is used for sending traps. The SNMP community string determines the access privileges (read-only or read-write) of SNMP clients to the device's SNMP.

> **Note:** SNMP community strings are used only for SNMPv1 and SNMPv2c; SNMPv3 uses username-password authentication along with an encryption key (see "Configuring SNMP V3 Users" on page 95).

For detailed descriptions of the SNMP parameters, see "SNMP Parameters" on page 819.

➢ To configure SNMP community strings:

1. Open the SNMP Community String page (**Configuration** tab > **System** menu > **Management** > **SNMP** > **SNMP Community String**).



Community String	Access Level
	Read Only
	Read Only
	Read Only
	Read Only
	Read Only
	Read / Write
	Read / Write
	Read / Write
	Read / Write
	Read / Write

Disable SNMP	No
Trap Community String	trapuser
Trap Manager Host Name	
Activity Trap	Disable

2. Configure SNMP community strings according to the table below.
3. Click **Submit**, and then save ("burn") your settings to flash memory.

To delete a community string, select the **Delete** check box corresponding to the community string that you want to delete, and then click **Submit**.

Table 8-1: SNMP Community String Parameter Descriptions

Parameter	Description
Community String - Read Only `configure system > snmp > ro-community-string` [SNMPReadOnlyCommunityString_x]	Defines a read-only SNMP community string. Up to five read-only community strings can be configured. The valid value is a string of up to 19 characters that can include only the following: ▪ Upper- and lower-case letters (a to z, and A to Z) ▪ Numbers (0 to 9) ▪ Hyphen (-) ▪ Underline (_) For example, "Public-comm_string1".

Parameter	Description
	The default is "public".
Community String - Read / Write `configure system > snmp > rw-community-string` [SNMPReadWriteCommunityString_x]	Defines a read-write SNMP community string. Up to five read-write community strings can be configured. The valid value is a string of up to 19 characters that can include only the following: ▪ Upper- and lower-case letters (a to z, and A to Z) ▪ Numbers (0 to 9) ▪ Hyphen (-) ▪ Underline (_) For example, "Private-comm_string1". The default is "private".
Trap Community String `configure system > snmp trap > community-string` [SNMPTrapCommunityString]	Defines the community string for SNMP traps. The valid value is a string of up to 19 characters that can include only the following: ▪ Upper- and lower-case letters (a to z, and A to Z) ▪ Numbers (0 to 9) ▪ Hyphen (-) ▪ Underline (_) For example, "Trap-comm_string1". The default is "trapuser".

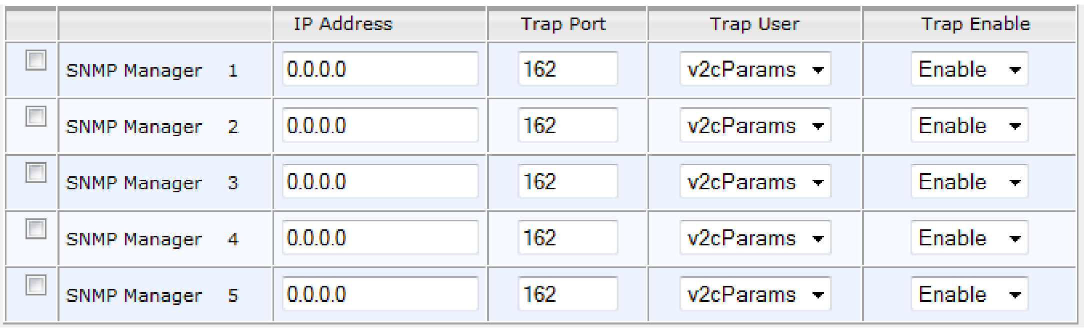
8.3 Configuring SNMP Trap Destinations

The SNMP Trap Destinations table lets you to configure up to five SNMP trap managers. You can associate a trap destination with SNMPv2 users and specific SNMPv3 users. Associating a trap destination with SNMPv3 users sends encrypted and authenticated traps to the SNMPv3 destination. By default, traps are sent unencrypted using SNMPv2.

➢ **To configure SNMP trap destinations:**

1. Open the SNMP Trap Destinations table (**Configuration** tab > **System** menu > **Management** > **SNMP** > **SNMP Trap Destinations**).

Figure 8-2: SNMP Trap Destinations Table



		IP Address	Trap Port	Trap User	Trap Enable
□	SNMP Manager 1	0.0.0.0	162	v2cParams	Enable
□	SNMP Manager 2	0.0.0.0	162	v2cParams	Enable
□	SNMP Manager 3	0.0.0.0	162	v2cParams	Enable
□	SNMP Manager 4	0.0.0.0	162	v2cParams	Enable
□	SNMP Manager 5	0.0.0.0	162	v2cParams	Enable

2. Configure the SNMP trap manager parameters according to the table below.
3. Select the check box corresponding to the SNMP Manager that you wish to enable.
4. Click **Submit**.

Notes:

- Only row entries whose corresponding check boxes are selected are applied when clicking **Submit**; otherwise, settings revert to their defaults.
- To enable the sending of the trap event acPerformanceMonitoringThresholdCrossing, which is sent every time a threshold (high or low) of a performance monitored SNMP object is crossed, configure the ini file parameter PM_EnableThresholdAlarms to 1.

Table 8-2: SNMP Trap Destinations Table Parameters Description

Parameter	Description
SNMP Manager [SNMPManagerIsUsed_x]	Enables the SNMP Manager to receive traps and checks the validity of the configured destination (IP address and port number). ▪ **[0]** (check box cleared) = (Default) Disables SNMP Manager ▪ **[1]** (check box selected) = Enables SNMP Manager
IP Address [SNMPManagerTableIP_x]	Defines the IP address (in dotted-decimal notation, e.g., 108.10.1.255) of the remote host used as the SNMP Manager. The device sends SNMP traps to this IP address.
Trap Port [SNMPManagerTrapPort_x]	Defines the port number of the remote SNMP Manager. The device sends SNMP traps to this port. The valid value range is 100 to 4000. The default is 162.
Trap User [SNMPManagerTrapUser]	Associates a trap user with the trap destination. This determines the trap format, authentication level, and encryption level. ▪ v2cParams (default) = SNMPv2 user community string ▪ SNMPv3 user configured in "Configuring SNMP V3 Users" on page 95
Trap Enable [SNMPManagerTrapSendingEnable_x]	Activates the sending of traps to the SNMP Manager. ▪ **[0]** Disable ▪ **[1]** Enable (Default)

8.4 Configuring SNMP Trusted Managers

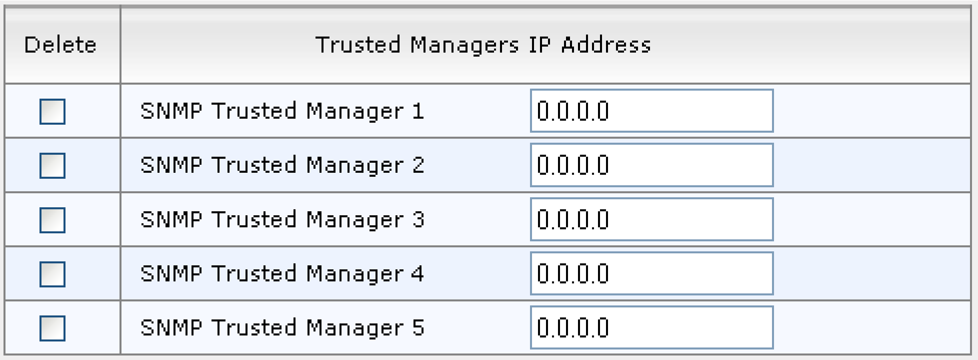
The SNMP Trusted Managers table lets you configure up to five SNMP Trusted Managers based on IP addresses. By default, the SNMP agent accepts SNMP Get and Set requests from any IP address as long as the correct community string is used in the request. Security can be enhanced by using Trusted Managers, which is an IP address from which the SNMP agent accepts and processes SNMP requests.

The following procedure describes how to configure SNMP trusted managers through the Web interface. You can also configure it through ini file (SNMPTrustedMgr_x) or CLI (configure system > snmp > trusted-managers).

➢ **To configure SNMP Trusted Managers:**

1. Open the SNMP Trusted Managers table (**Configuration** tab > **System** menu > **Management** > **SNMP** > **SNMP Trusted Managers**).

Figure 8-3: SNMP Trusted Managers Table



Delete	Trusted Managers IP Address	
□	SNMP Trusted Manager 1	0.0.0.0
□	SNMP Trusted Manager 2	0.0.0.0
□	SNMP Trusted Manager 3	0.0.0.0
□	SNMP Trusted Manager 4	0.0.0.0
□	SNMP Trusted Manager 5	0.0.0.0

2. Select the check box corresponding to the SNMP Trusted Manager that you want to enable and for whom you want to define an IP address.
3. Define an IP address in dotted-decimal notation.
4. Click **Submit**, and then save ("burn") your settings to flash memory.

8.5 Configuring SNMP V3 Users

The SNMPv3 Users table lets you configure up to 10 SNMP v3 users for authentication and privacy.

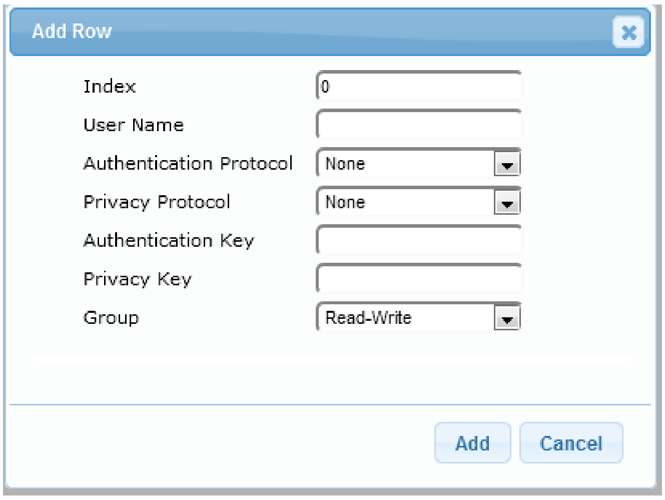
The following procedure describes how to configure SNMP v3 users through the Web interface. You can also configure it through ini file (SNMPUsers) or CLI (configure system > snmp v3-users).

➢ **To configure an SNMP v3 user:**

1. Open the SNMPv3 Users table (**Configuration** tab > **System** menu > **Management** > **SNMP** > **SNMP V3 Users**).

2. Click **Add**; the following dialog box appears:

Figure 8-4: SNMPv3 Users Table - Add Row Dialog Box



3. Configure the SNMP V3 parameters according to the table below.
4. Click **Add**, and then save ("burn") your settings to flash memory.

Note: If you delete a user that is associated with a trap destination (see "Configuring SNMP Trap Destinations" on page 93), the configured trap destination becomes disabled and the trap user reverts to default (i.e., SNMPv2).

Table 8-3: SNMPv3 Users Table Parameters Description

Parameter	Description
Index [SNMPUsers_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
User Name `username` [SNMPUsers_Username]	Name of the SNMP v3 user. This name must be unique.
Authentication Protocol `auth-protocol` [SNMPUsers_AuthProtocol]	Authentication protocol of the SNMP v3 user. ▪ **[0]** None (default) ▪ **[1]** MD5 ▪ **[2]** SHA-1
Privacy Protocol `priv-protocol` [SNMPUsers_PrivProtocol]	Privacy protocol of the SNMP v3 user. ▪ **[0]** None (default) ▪ **[1]** DES ▪ **[2]** 3DES ▪ **[3]** AES-128 ▪ **[4]** AES-192 ▪ **[5]** AES-256
Authentication Key `auth-key` [SNMPUsers_AuthKey]	Authentication key. Keys can be entered in the form of a text password or long hex string. Keys are always persisted as long hex strings and keys are localized.

Parameter	Description
Privacy Key `priv-key` [SNMPUsers_PrivKey]	Privacy key. Keys can be entered in the form of a text password or long hex string. Keys are always persisted as long hex strings and keys are localized.
Group `group` [SNMPUsers_Group]	The group with which the SNMP v3 user is associated. - **[0]** Read-Only (default) - **[1]** Read-Write - **[2]** Trap **Note:** All groups can be used to send traps.

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9 INI File-Based Management

The device can be configured through an ini file, which is a text-based file with an *ini* file extension name that can be created using any standard text-based editor such as Notepad. Each configuration element of the device has a corresponding ini file parameter that you can use in the ini file for configuring the device. When you have created the ini file with your ini file parameter settings, you apply these settings to the device by installing (loading) the ini file to the device.

Notes:

- For a list and description of the *ini* file parameters, see "Configuration Parameters Reference" on page 813.
- To restore the device to default settings through the *ini* file, see "Restoring Factory Defaults" on page 697.

9.1 INI File Format

The *ini* file can be configured with any number of parameters. These *ini* file parameters can be one of the following types:

- Individual parameters - see "Configuring Individual ini File Parameters" on page 99
- Table parameters - see "Configuring Table ini File Parameters" on page 99

9.1.1 Configuring Individual ini File Parameters

The syntax for configuring individual *ini* file parameters in the ini file is as follows:

- An optional, subsection name (or group name) enclosed in square brackets "[...]". This is used to conveniently group similar parameters by their functionality.
- Parameter name, followed by an equal "=" sign and then its value.
- Comments must be preceded by a semicolon ";".

```
[subsection name]
parameter name = value
parameter name = value
; this is a comment line
; for example:
[System Parameters]
SyslogServerIP = 10.13.2.69
EnableSyslog = 1
```

For general *ini* file formatting rules, see "General ini File Formatting Rules" on page 101.

9.1.2 Configuring Table ini File Parameters

The table ini file parameters allow you to configure tables, which include multiple parameters (*columns*) and row entries (*indices*). When loading an *ini* file to the device, it's recommended to include only tables that belong to applications that are to be configured (dynamic tables of other applications are empty, but static tables are not).

The table ini file parameter is composed of the following elements:

- **Title of the table:** The name of the table in square brackets, e.g., [MY_TABLE_NAME].
- **Format line:** Specifies the columns of the table (by their string names) that are to be configured.

   - The first word of the Format line must be "FORMAT", followed by the Index field name and then an equal "=" sign. After the equal sign, the names of the columns are listed.
   - Columns must be separated by a comma ",".
   - The Format line must only include columns that can be modified (i.e., parameters that are not specified as read-only). An exception is Index fields, which are mandatory.
   - The Format line must end with a semicolon ";".

- **Data line(s):** Contain the actual values of the columns (parameters). The values are interpreted according to the Format line.
   - The first word of the Data line must be the table's string name followed by the Index field.
   - Columns must be separated by a comma ",".
   - A Data line must end with a semicolon ";".
- **End-of-Table Mark:** Indicates the end of the table. The same string used for the table's title, preceded by a backslash "\", e.g., [\MY_TABLE_NAME].

The following displays an example of the structure of a table ini file parameter.

```
[Table_Title]
; This is the title of the table.
FORMAT Index = Column_Name1, Column_Name2, Column_Name3;
; This is the Format line.
Index 0 = value1, value2, value3;
Index 1 = value1, $$, value3;
; These are the Data lines.
[\Table_Title]
; This is the end-of-the-table-mark.
```

The table ini file parameter formatting rules are listed below:

- Indices (in both the Format and the Data lines) must appear in the same order. The Index field must never be omitted.
- The Format line can include a subset of the configurable fields in a table. In this case, all other fields are assigned with the pre-defined default values for each configured line.
- The order of the fields in the Format line isn't significant (as opposed to the Index fields). The fields in the Data lines are interpreted according to the order specified in the Format line.
- The double dollar sign ($$) in a Data line indicates the default value for the parameter.
- The order of the Data lines is insignificant.
- Data lines must match the Format line, i.e., it must contain exactly the same number of Indices and Data fields and must be in exactly the same order.
- A row in a table is identified by its table name and Index field. Each such row may appear only once in the *ini* file.
- Table dependencies: Certain tables may depend on other tables. For example, one table may include a field that specifies an entry in another table. This method is used to specify additional attributes of an entity, or to specify that a given entity is part of a larger entity. The tables must appear in the order of their dependency (i.e., if Table X is referred to by Table Y, Table X must appear in the *ini* file before Table Y).

For general *ini* file formatting rules, see "General ini File Formatting Rules" on page 101.

The table below displays an example of a table ini file parameter:

```
[ CodersGroup0 ]
FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce,
CodersGroup0_CoderSpecific;
```

```
CodersGroup0 0 = g711Alaw64k, 20, 0, 255, 0, 0;
CodersGroup0 1 = eg711Ulaw, 10, 0, 71, 0, 0;
[ \CodersGroup0 ]
```

> **Note:** Do not include read-only parameters in the table ini file parameter as this can cause an error when attempting to load the file to the device.

9.1.3 General ini File Formatting Rules

The *ini* file must adhere to the following formatting rules:

- The *ini* file name must not include hyphens "-" or spaces; if necessary, use an underscore "_" instead.
- Lines beginning with a semi-colon ";" are ignored. These can be used for adding remarks in the *ini* file.
- A carriage return (i.e., Enter) must be done at the end of each line.
- The number of spaces before and after the equals sign "=" is irrelevant.
- Subsection names for grouping parameters are optional.
- If there is a syntax error in the parameter name, the value is ignored.
- Syntax errors in the parameter's value can cause unexpected errors (parameters may be set to the incorrect values).
- Parameter string values that denote file names (e.g., CallProgressTonesFileName) must be enclosed with inverted commas, e.g., CallProgressTonesFileName = 'cpt_usa.dat'.
- The parameter name is not case-sensitive.
- The parameter value is not case-sensitive, except for coder names.
- The *ini* file must end with at least one carriage return.

9.2 Configuring an ini File

There are different methods that you can use for configuring the ini file before you load it to the device.

- Modifying the device's current ini file. This method is recommended if you mainly need to change the settings of parameters that you have previously configured.
  1. Save the device's current configuration as an *ini* file on your computer, using the Web interface (see "Saving Configuration" on page 643).
  2. Open the file using a text file editor, and then modify the *ini* file as required.
  3. Save and close the file.
  4. Load the file to the device.
- Creating a new ini file that includes only updated configuration:
  1. Open a text file editor such as Notepad.
  2. Add only the required parameters and their settings.
  3. Save the file with the ini file extension name (e.g., myconfiguration.ini).
  4. Load the file to the device.

For loading the ini file to the device, see "Loading an ini File to the Device" on page 102.

Notes:

- If you save an ini file from the device and a table row is configured with invalid values, the ini file displays the row prefixed with an exclamation mark (!), for example:

  ```
  !CpMediaRealm 1 = "ITSP", "Voice", "", 60210, 2,
  6030, 0, "", "";
  ```

- To restore the device to default settings through the *ini* file, see "Restoring Factory Defaults" on page 697.

9.3 Loading an ini File to the Device

You can load an *ini* file to the device using the following methods:

- CLI:
  - Voice Configuration: # copy voice-configuration from <URL>
- Web interface:
  - Load Auxiliary Files page (see "Loading Auxiliary Files" on page 647): The device updates its configuration according to the loaded ini file, while preserving the remaining current configuration.
  - Configuration File page (see "Backing Up and Loading Configuration File" on page 677): The device updates its configuration according to the loaded ini file, and applies default values to parameters that were not included in the loaded ini file. Thus, all previous configuration is overridden.

When you load an ini file to the device, its configuration settings are saved to the device's non-volatile memory.

Note: Before you load an *ini* file to the device, make sure that the file extension name is *.ini*.

9.4 Secured Encoded ini File

The *ini* file contains sensitive information that is required for the functioning of the device. The file may be loaded to the device using HTTP. These protocols are not secure and are vulnerable to potential hackers. To overcome this security threat, the AudioCodes DConvert utility allows you to binary-encode (encrypt) the *ini* file before loading it to the device. For more information, refer to the *DConvert Utility User's Guide*.

Note: If you save an ini file from the device to a folder on your PC, an *ini* file that was loaded to the device encoded is saved as a regular *ini* file (i.e., unencoded).

9.5 Configuring Password Display in ini File

Passwords can be displayed in the ini file in one of the following formats, configured by the INIPasswordsDisplayType ini file parameter:

- Obscured: The password characters are concealed and displayed as encoded. The password is displayed using the syntax, *$1$<obscured password>*, for example, $1$S3p+fno=.
- Hidden: the password is replaced with an asterisk (*).

When you save an ini file from the device to a PC, the passwords are displayed according to the enabled format. When you load an ini file to the device, obscured passwords are parsed and applied to the device; hidden passwords are ignored.

By default, the enabled format is obscured passwords, thus enabling their full recovery in case of configuration restore or copy to another device.

When obscured password mode is enabled, you can enter a password in the ini file using any of the following formats:

- $1$<obscured password>: Password in obscured format as generated by the device; useful for restoring device configuration and copying configuration from one device to another.
- $0$<plain text>: Password can be entered in plain text; useful for configuring a new password. When the ini file is loaded to the device and then later saved from the device to a PC, the password is displayed obscured (i.e., $1$<obscured password>).

9.6 INI Viewer and Editor Utility

AudioCodes INI Viewer & Editor utility provides a user-friendly graphical user interface (GUI) that lets you easily view and modify the device's ini file. This utility is available from AudioCodes Web site at www.AudioCodes.com/downloads, and can be installed on any Windows-based PC.

For more information, refer to the *INI Viewer & Editor User's Guide*.

Part III

General System Settings

10 Configuring SSL/TLS Certificates

The TLS Contexts page lets you configure X.509 certificates, which are used for secure management of the device, secure SIP transactions, and other security applications.

Notes:

- The device is shipped with an active, default TLS setup. Thus, configure certificates only if required.
- Since X.509 certificates have an expiration date and time, you must configure the device to use Network Time Protocol (NTP) to obtain the current date and time from an NTP server. Without the correct date and time, client certificates cannot work. For configuring NTP, see "Configuring Automatic Date and Time using SNTP" on page 121.
- Only **Base64 (PEM)** encoded X.509 certificates can be loaded to the device.

10.1 Configuring TLS Certificate Contexts

The TLS Contexts table lets you configure up to 12 TLS certificates, referred to as *TLS Contexts*. The Transport Layer Security (TLS), also known as Secure Socket Layer (SSL), is used to secure the device's SIP signaling connections, Web interface, and Telnet server. The TLS/SSL protocol provides confidentiality, integrity, and authenticity between two communicating applications over TCP/IP.

The device is shipped with a default TLS Context (ID 0 and string name "default"), which includes a self-generated random private key and a self-signed server certificate. The subject name for the default certificate is "ACL_nnnnnnn", where *nnnnnnn* denotes the serial number of the device. The default TLS Context can be used for SIP over TLS (SIPS) or any other supported application such as Web (HTTPS), Telnet, and SSH.The default TLS Context cannot be deleted.

The user-defined TLS Contexts are used **only** for SIP over TLS (SIPS). This enables you to use different TLS certificates for your IP Groups (SIP entities). This is done by assigning a specific TLS Context to the Proxy Set and/or SIP Interface associated with the IP Group. TLS Contexts are applicable to Gateway and SBC calls.

Each TLS Context can be configured with the following:

- Context ID and name
- TLS version (SSL 3.0, TLS 1.0, TLS 1.1, TLS 1.2)
- Encryption ciphers for server and client - DES, RC4 compatible, Advanced Encryption Standard (AES)
- Online Certificate Status Protocol (OCSP). Some Public-Key Infrastructures (PKI) can revoke a certificate after it has been issued. You can configure the device to check whether a peer's certificate has been revoked, using the OCSP. When OCSP is enabled, the device queries the OCSP server for revocation information whenever a peer certificate is received (TLS client mode, or TLS server mode with mutual authentication).
- Private key - externally created and then uploaded to device
- X.509 certificates - self-signed certificates or signed as a result of a certificate signing request (CSR)
- Trusted root certificate authority (CA) store (for validating certificates)

When the device establishes a TLS connection (handshake) with a SIP user agent (UA), the TLS Context is determined as follows:

- **Incoming calls:**
  1. Proxy Set: If the incoming call is successfully classified to an IP Group based on Proxy Set (i.e., IP address of calling party) and the Proxy Set is configured for TLS ('Transport Type' parameter is set to **TLS**), the TLS Context assigned to the Proxy Set is used. For configuring Proxy Sets, see "Configuring Proxy Sets" on page 352.
  2. SIP Interface: If the Proxy Set is either not configured for TLS (i.e., the 'Transport Type' parameter is set to **UDP**) or not assigned a TLS Context, and/or classification to a Proxy Set fails, the device uses the TLS Context assigned to the SIP Interface used for the call. For configuring SIP Interfaces, see "Configuring SIP Interfaces" on page 333.
  3. Default TLS Context (ID 0): If the SIP Interface is not assigned a TLS Context or no SIP Interface is used for the call, the device uses the default TLS Context.
- **Outgoing calls:**
  1. Proxy Set: If the outgoing call is sent to an IP Group associated with a Proxy Set that is assigned a TLS Context and the Proxy Set is configured for TLS (i.e., 'Transport Type' parameter is set to **TLS**), the TLS Context is used. If the 'Transport Type' parameter is set to **UDP**, the device uses UDP to communicate with the proxy and no TLS Context is used.
  2. SIP Interface: If the Proxy Set is not assigned a TLS Context, the device uses the TLS Context assigned to the SIP Interface used for the call.
  3. Default TLS Context (ID 0): If the SIP Interface is not assigned a TLS Context or no SIP Interface is used for the call, the device uses the default TLS Context.

Notes:

- If the TLS Context used for an existing TLS connection is changed during the call by the user agent, the device ends the connection.
- The device does not query OCSP for its own certificate.
- Some PKIs do not support OCSP, but generate Certificate Revocation Lists (CRLs). For such scenarios, set up an OCSP server such as OCSPD.

TLS Context certification also enables employing different levels of security strength (key size) per certificate. This feature also enables the display of the list of all trusted certificates currently installed on the device. For each certificate, detailed information such as issuer and expiration date is shown. Certificates can be deleted or added from/to the Trusted Root Certificate Store.

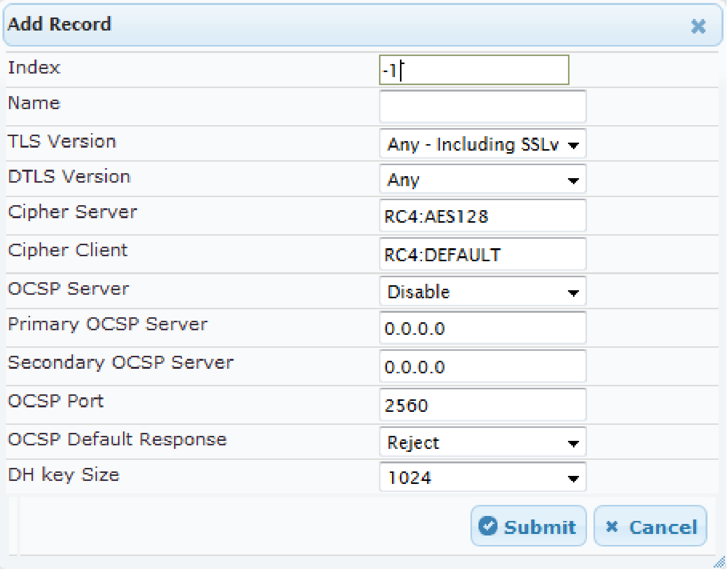
You can also configure TLS certificate expiry check, whereby the device periodically checks the validation date of the installed TLS server certificates and sends an SNMP trap event if a certificate is nearing expiry. This feature is configured globally for all TLS Contexts. For configuring TLS certificate expiry check, see "Configuring TLS Server Certificate Expiry Check" on page 119.

The following procedure describes how to configure a TLS Context through the Web interface. You can also configure it through ini file (TLSContexts) or CLI (configure system > tls <ID>).

➢ To configure a TLS Context:

1. Open the TLS Contexts page (**Configuration** tab > **System** menu > **TLS Contexts**).
2. Click **Add**; the following dialog box appears:

Figure 10-1: TLS Contexts Table - Add Record Dialog Box



Add Record

Index	-1
Name	
TLS Version	Any - Including SSLv
DTLS Version	Any
Cipher Server	RC4:AES128
Cipher Client	RC4:DEFAULT
OCSP Server	Disable
Primary OCSP Server	0.0.0.0
Secondary OCSP Server	0.0.0.0
OCSP Port	2560
OCSP Default Response	Reject
DH key Size	1024

3. Configure the TLS Context according to the parameters described in the table below.
4. Click **Add**, and then save ("burn") your settings to flash memory.

Table 10-1: TLS Context Parameter Descriptions

Parameter	Description
Index `tls <ID>` [TLSContexts_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `name` [TLSContexts_Name]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 31 characters.
TLS Version `tls-version` [TLSContexts_TLSVersion]	Defines the supported SSL/TLS protocol version. Clients attempting to communicate with the device using a TLS version that is not configured are rejected. ▪ [0] Any - Including SSLv3 = (Default) SSL 3.0 and all TLS versions are supported. ▪ [1] TLSv1.0 = Only TLS 1.0. ▪ [2] TLSv1.1 = Only TLS 1.1. ▪ [3] TLSv1.0 and TLSv1.1 = Only TLS 1.0 and TLS 1.1. ▪ [4] TLSv1.2 = Only TLS 1.2. ▪ [5] TLSv1.0 and TLSv1.2 = Only TLS 1.0 and TLS 1.2. ▪ [6] TLSv1.1 and TLSv1.2 = Only TLS 1.1 and TLS 1.2. ▪ [7] TLSv1.0 TLSv1.1 and TLSv1.2 = Only TLS 1.0, TLS 1.1 and TLS 1.2 (excludes SSL 3.0).

Parameter	Description
DTLS Version [TLSContexts_DTLSVersion]	Defines the Datagram Transport Layer Security (DTLS) version, which is used to negotiate keys for WebRTC calls. ▪ [0] Any (default) ▪ [1] DTLSv1.0 ▪ [2] DTLSv1.2 **Note:** The parameter is applicable only to the SBC application.
Cipher Server `ciphers-server` [TLSContexts_ServerCipherString]	Defines the supported cipher suite for the TLS server (in OpenSSL cipher list format). The default is AES:RC4. For valid values, visit the OpenSSL website at https://www.openssl.org/docs/man1.0.2/apps/ciphers.html.
Cipher Client `ciphers-client` [TLSContexts_ClientCipherString]	Defines the supported cipher suite for TLS clients. The default is DEFAULT. For possible values and additional details, visit the OpenSSL website at https://www.openssl.org/docs/man1.0.2/apps/ciphers.html.
OCSP Server `ocsp-server` [TLSContexts_OcspEnable]	Enables or disables certificate checking using OCSP. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Primary OCSP Server `ocsp-server-primary` [TLSContexts_OcspServerPrimary]	Defines the IP address (in dotted-decimal notation) of the primary OCSP server. The default IP address is 0.0.0.0.
Secondary OCSP Server `ocsp-server-secondary` [TLSContexts_OcspServerSecondary]	Defines the IP address (in dotted-decimal notation) of the secondary OCSP server (optional). The default IP address is 0.0.0.0.
OCSP Port `ocsp-port` [TLSContexts_OcspServerPort]	Defines the OCSP server's TCP port number. The default port number is 2560.
OCSP Default Response `ocsp-default-response` [TLSContexts_OcspDefaultResponse]	Determines whether the device allows or rejects peer certificates if it cannot connect to the OCSP server. ▪ **[0]** Reject (default) ▪ **[1]** Allow
DH Key Size [TLSContexts_DHKeySize]	Defines the Diffie-Hellman (DH) key size (in bits). DH is an algorithm used chiefly for exchanging cryptography keys used in symmetric encryption algorithms such as AES. ▪ [1024] 1024 (default) ▪ [2048] 2048

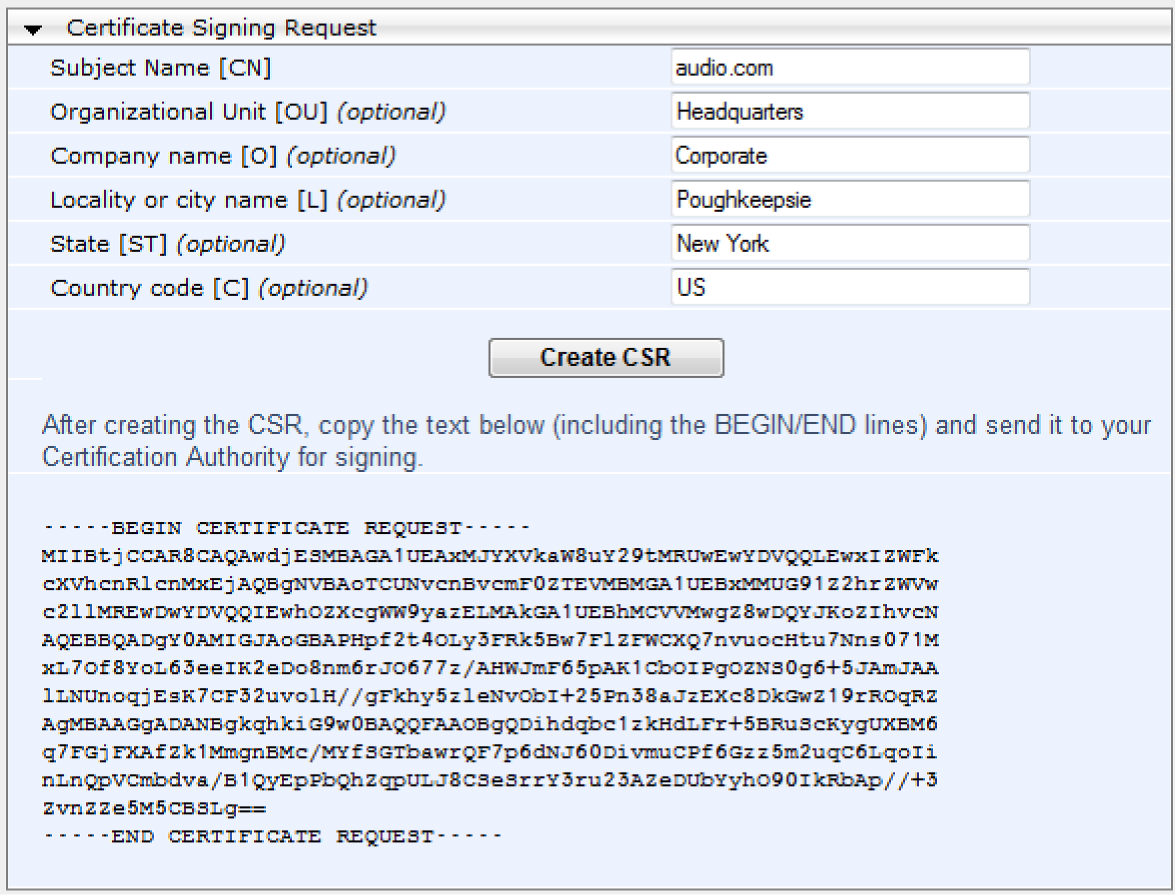
10.2 Assigning CSR-based Certificates to TLS Contexts

The following procedure describes how to request a digitally signed certificate from a Certification Authority (CA) for a TLS Context. This process is referred to as a certificate signing request (CSR) and is required if your organization employs a Public Key Infrastructure (PKI) system. The CSR contains information identifying the device (such as a distinguished name in the case of an X.509 certificate).

➢ To assign a CSR-based certificate to a TLS Context:

1. Your network administrator should allocate a unique DNS name for the device (e.g., dns_name.corp.customer.com). This DNS name is used to access the device and therefore, must be listed in the server certificate.
2. Open the TLS Contexts page (**Configuration** tab > **System** menu > **TLS Contexts**).
3. In the table, select the required TLS Context index row, and then click the **TLS Context Certificate** ➡ button, located below the table; the Context Certificates page appears.
4. Under the **Certificate Signing Request** group, do the following:
   a. In the 'Subject Name [CN]' field, enter the DNS name.
   b. From the 'Signature Algorithm' drop-down list, select the hash function algorithm (SHA-1, SHA-256, or SHA-512) with which to sign the certificate.
   c. Fill in the rest of the request fields according to your security provider's instructions.
   d. Click the **Create CSR** button; a textual certificate signing request is displayed in the area below the button:

Figure 10-2: Certificate Signing Request Group



▼ Certificate Signing Request

Subject Name [CN]	audio.com
Organizational Unit [OU] *(optional)*	Headquarters
Company name [O] *(optional)*	Corporate
Locality or city name [L] *(optional)*	Poughkeepsie
State [ST] *(optional)*	New York
Country code [C] *(optional)*	US

After creating the CSR, copy the text below (including the BEGIN/END lines) and send it to your Certification Authority for signing.

```
-----BEGIN CERTIFICATE REQUEST-----
MIIBtjCCAR8CAQAwdjESMBAGA1UEAxMJYXVkaW8uY29tMRUwEwYDVQQLEwxIZWFk
cXVhcnRlcnMxEjAQBgNVBAoTCUNvcnBvcmF0ZTEVMBMGA1UEBxMMUG91z2hrZWVw
c2llMREwDwYDVQQIEwhOZXcgWW9yazELMAkGA1UEBhMCVVMwgZ8wDQYJKoZIhvcN
AQEBBQADgY0AMIGJAoGBAPHpf2t4OLy3FRk5Bw7FlZFWCXQ7nvuocHtu7Nns071M
xL7Of8YoL63eeIK2eDo8nm6rJO677z/AHWJmF65pAK1CbOIPgOZNS0g6+5JAmJAA
lLNUnoqjEsK7CF32uvolH//gFkhy5zleNvObI+25Pn38aJzEXc8DkGwz19rROqRZ
AgMBAAGgADANBgkqhkiG9w0BAQQFAAOBgQDihdgbc1zkHdLFr+5BRuScKygUXBM6
q7FGjFXAfZk1MmgnBMc/MYfSGTbawrQF7p6dNJ60DivmuCPf6Gzz5m2uqC6LqoIi
nLnQpVCmbdva/B1QyEpPbQhZqpULJ8CSeSrrY3ru23AZeDUbYyhO90IkRbAp//+3
ZvnZZe5M5CBSLg==
-----END CERTIFICATE REQUEST-----
```

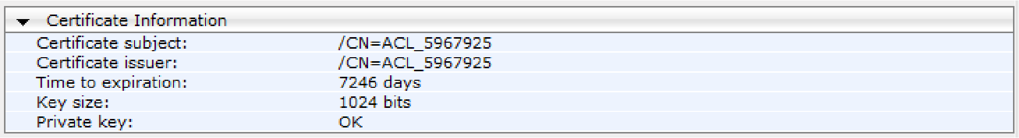
5. Copy the text and send it to your security provider (CA) to sign this request.
6. When the CA sends you a server certificate, save the certificate to a file (e.g., cert.txt). Ensure that the file is a plain-text file containing the"'BEGIN CERTIFICATE" header, as shown in the example of a Base64-Encoded X.509 Certificate below:

```
-----BEGIN CERTIFICATE-----
MIIDkzCCAnugAwIBAgIEAgAAADANBgkqhkiG9w0BAQQFADA/MQswCQYDVQQGEw
JGUjETMBEGA1UEChMKQ2VydGlwb3N0ZTEbMBkGA1UEAxMSQ2VydGlwb3N0ZSBT
ZXJ2ZXVyMB4XDTk4MDYyNDA4MDAwMFoXDTE4MDYyNDA4MDAwMFowPzELMAkGA1
UEBhMCRlIxEzARBgNVBAoTCkNlcnRpcG9zdGUxGzAZBgNVBAMTEkNlcnRpcG9z
```

```
dGUgU2VydmV1cjCCASEwDQYJKoZIhvcNAQEBBQADggEOADCCAQkCggEAPqd4Mz
iR4spWldGRx8bQrhZkonWnNm`+Yhb7+4Q67ecf1janH7GcN/SXsfx7jJpreWUL
f7v7Cvpr4R7qIJcmdHIntmf7JPM5n6cDBv17uSW63er7NkVnMFHwK1QaGFLMyb
FkzaeGrvFm4k3lRefiXDmuOe+FhJgHYezYHf44LvPRPwhSrzi9+Aq3o8pWDguJ
uZDIUP1F1jMa+LPwvREXfFcUW+w==
-----END CERTIFICATE-----
```

7. Scroll down to the **Upload certificates files from your computer** group, click the **Browse** button corresponding to the 'Send Device Certificate...' field, navigate to the cert.txt file, and then click **Send File**.
8. After the certificate successfully loads to the device, save the configuration with a device reset.
9. Open the TLS Contexts page again, select the TLS Context index row, and then verify that under the **Certificate Information** group, the 'Private key' field displays "OK"; otherwise, consult your security administrator:

Figure 10-3: Private key "OK" in Certificate Information Group



▼ Certificate Information

Certificate subject:	/CN=ACL_5967925
Certificate issuer:	/CN=ACL_5967925
Time to expiration:	7246 days
Key size:	1024 bits
Private key:	OK

Notes:

- The certificate replacement process can be repeated when necessary (e.g., the new certificate expires).
- It is possible to use the IP address of the device (e.g., 10.3.3.1) instead of a qualified DNS name in the Subject Name. This is not recommended since the IP address is subject to change and may not uniquely identify the device.
- The device certificate can also be loaded via the Automatic Update Facility by using the HTTPSCertFileName *ini* file parameter.

10.3 Assigning Externally Created Private Keys to TLS Contexts

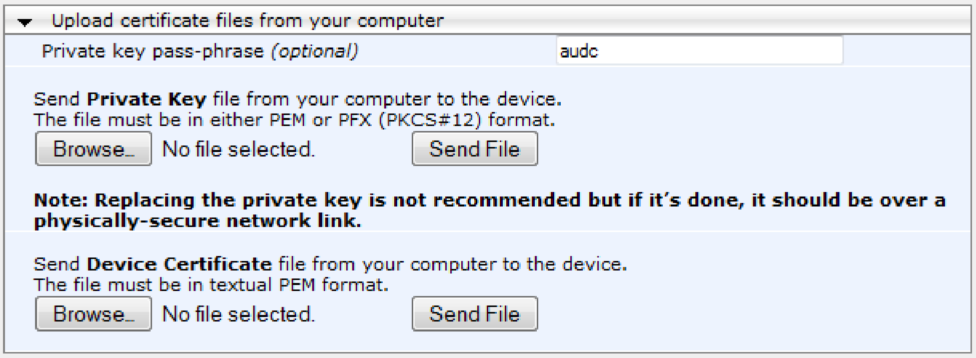
The following procedure describes how to assign an externally created private key to a TLS Context.

➢ To assign an externally created private key to a TLS Context:

1. Obtain a private key in either textual PEM (PKCS #7) or PFX (PKCS #12) format (typically provided by your security administrator). The file may be encrypted with a short pass-phrase.
2. Open the TLS Contexts page (**Configuration** tab > **System** menu > **TLS Contexts**).
3. In the table, select the required TLS Context index row, and then click the **TLS Context Certificate** ➡ button, located below the table; the Context Certificates page appears.

4. Scroll down to the **Upload certificate files from your computer** group.

Figure 10-4: Upload Certificate Files from your Computer Group



Upload certificate files from your computer

Private key pass-phrase *(optional)*: audc

Send **Private Key** file from your computer to the device.
The file must be in either PEM or PFX (PKCS#12) format.

Note: Replacing the private key is not recommended but if it's done, it should be over a physically-secure network link.

Send **Device Certificate** file from your computer to the device.
The file must be in textual PEM format.

5. Fill in the 'Private key pass-phrase' field, if required.

6. Click the **Browse** button corresponding to the 'Send Private Key' field, navigate to the private key file (Step 1), and then click **Send File**.

7. If the security administrator has provided you with a device certificate file, load it using the 'Send Device Certificate' field.
8. After the files successfully load to the device, save the configuration with a device reset.
9. Open the TLS Contexts page again, select the TLS Context index row, and then verify that under the **Certificate Information** group, the 'Private key' field displays "OK"; otherwise, consult your security administrator.

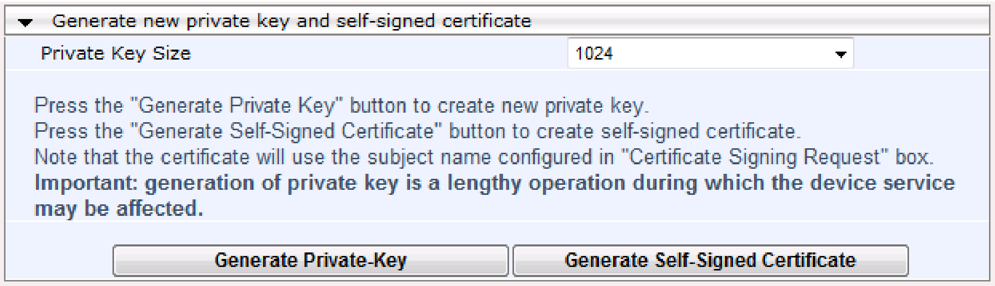
10.4 Generating Private Keys for TLS Contexts

The device can generate the private key for a TLS Context, as described in the following procedure. The private key can be generated for CSR or self-signed certificates.

➢ **To generate a new private key for a TLS Context:**

1. Open the TLS Contexts page (**Configuration** tab > **System** menu > **TLS Contexts**).
2. In the table, select the required TLS Context index row, and then click the **Context Certificates** ➡ button, located below the table; the Context Certificates page appears.
3. Scroll down to the **Generate new private key and self-signed certificate** group:

Figure 10-5: Generate new private key and self-signed certificate Group



▾ Generate new private key and self-signed certificate

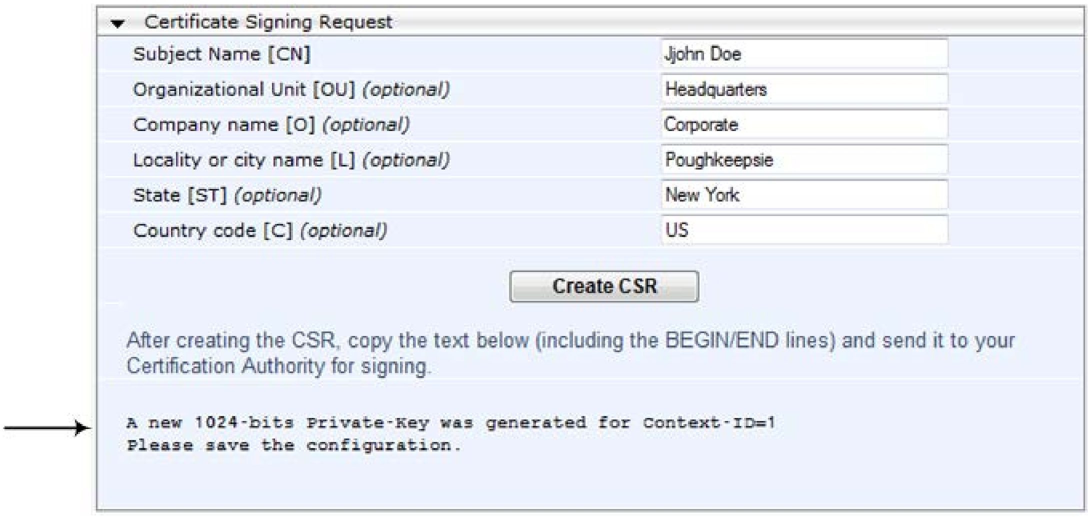
Private Key Size: 1024

Press the "Generate Private Key" button to create new private key.
Press the "Generate Self-Signed Certificate" button to create self-signed certificate.
Note that the certificate will use the subject name configured in "Certificate Signing Request" box.
Important: generation of private key is a lengthy operation during which the device service may be affected.

4. From the 'Private Key Size' drop-down list, select the desired private key size (in bits) for RSA public-key encryption for newly self-signed generated keys:
   - 512
   - 1024 (default)
   - 2048
   - 4096
5. Click **Generate Private Key**; a message appears requesting you to confirm key generation.
6. Click **OK** to confirm key generation; the device generates a new private key, indicated

by a message in the **Certificate Signing Request** group.

Figure 10-6: Indication of Newly Generated Private Key



Certificate Signing Request

Subject Name [CN]	Jjohn Doe
Organizational Unit [OU] *(optional)*	Headquarters
Company name [O] *(optional)*	Corporate
Locality or city name [L] *(optional)*	Poughkeepsie
State [ST] *(optional)*	New York
Country code [C] *(optional)*	US

After creating the CSR, copy the text below (including the BEGIN/END lines) and send it to your Certification Authority for signing.

```
A new 1024-bits Private-Key was generated for Context-ID=1
Please save the configuration.
```

7. Continue with the certificate configuration, by either creating a CSR or generating a new self-signed certificate.
8. Save the configuration with a device reset for the new certificate to take effect.

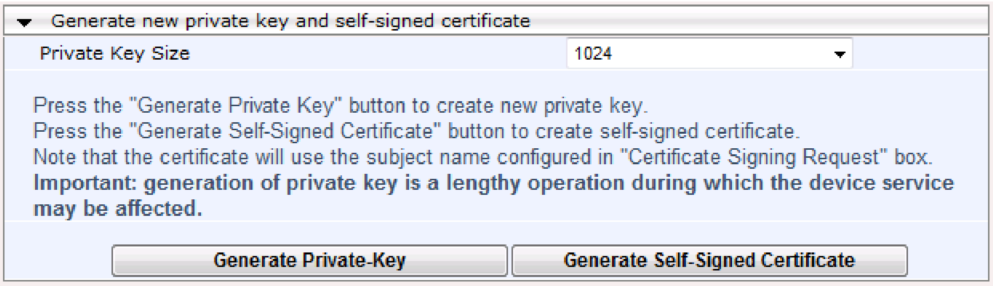
10.5 Creating Self-Signed Certificates for TLS Contexts

The following procedure describes how to assign a certificate that is digitally signed by the device itself to a TLS Context. In other words, the device acts as a CA.

➢ To assign a self-signed certificate to a TLS Context:

1. Before you begin, make sure that:
   - You have a unique DNS name for the device (e.g., dns_name.corp.customer.com). This name is used to access the device and therefore, must be listed in the server certificate.
   - No traffic is running on the device. The certificate generation process is disruptive to traffic and should be done during maintenance time.
2. Open the TLS Contexts page (**Configuration** tab > **System** menu > **TLS Contexts**).
3. In the table, select the required TLS Context index row, and then click the **TLS Context Certificate** ➡ button, located below the table; the Context Certificates page appears.
4. Under the **Certificate Signing Request** group, in the 'Subject Name [CN]' field, enter the fully-qualified DNS name (FQDN) as the certificate subject.
5. Scroll down the page to the **Generate new private key and self-signed certificate** group:

Figure 10-7: Generate new private key and self-signed certificate Group



Generate new private key and self-signed certificate

Private Key Size: 1024

Press the "Generate Private Key" button to create new private key.
Press the "Generate Self-Signed Certificate" button to create self-signed certificate.
Note that the certificate will use the subject name configured in "Certificate Signing Request" box.
Important: generation of private key is a lengthy operation during which the device service may be affected.

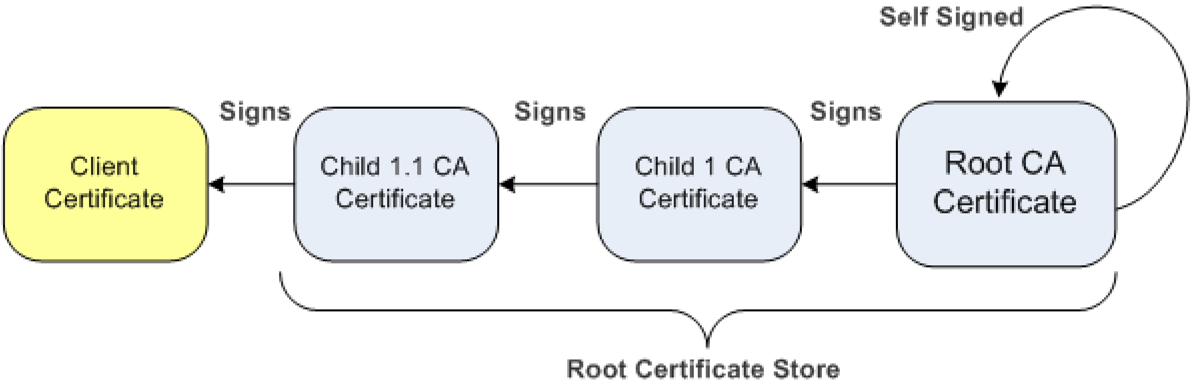
6. Click **Generate Self-Signed Certificate**; a message appears (after a few seconds) displaying the new subject name.
7. Save the configuration with a device reset for the new certificate to take effect.

10.6 Importing Certificates and Certificate Chain into Trusted Certificate Store

The device provides its own Trusted Root Certificate Store. This lets you manage certificate trust. You can add up to 20 certificates to the store per TLS Context (but this may be less depending on certificate file size).

The trusted store can also be used for certificate chains. A certificate chain is a sequence of certificates where each certificate in the chain is signed by the subsequent certificate. The last certificate in the list of certificates is the Root CA certificate, which is self-signed. The purpose of a certificate chain is to establish a chain of trust from a child certificate to the trusted root CA certificate. The CA vouches for the identity of the child certificate by signing it. A client certificate is considered trusted if one of the CA certificates up the certificate chain is found in the server certificate directory.

Figure 10-8: Certificate Chain Hierarchy



For the device to trust a whole chain of certificates per TLS Context, you need to add them to the device's Trusted Certificates Store, as described below.

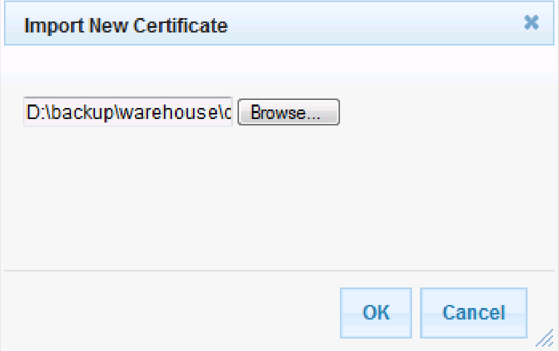
> **Note:** Only Base64 (PEM) encoded X.509 certificates can be loaded to the device.

➢ **To import certificates into device's Trusted Root Certificate Store:**

1. Open the TLS Contexts page (**Configuration** tab > **System** menu > **TLS Contexts**).
2. In the table, select the required TLS Context index row, and then click the **TLS Context Trusted Root Certificates** ➡ button, located below the table; the Trusted Certificates page appears.

3. Click the **Import** button, and then select the certificate file to load.

Figure 10-9: Importing Certificate into Trusted Certificates Store



4. Click **OK**; the certificate is loaded to the device and listed in the Trusted Certificates store.

You can also do the following with certificates that are in the Trusted Certificates store:

- Delete certificates: Select the required certificate, click **Remove**, and then in the Remove Certificate dialog box, click **Remove**.
- Save certificates to a file on your PC: Select the required certificate, click **Export**, and then in the Export Certificate dialog box, browse to the folder on your PC where you want to save the file and click **Export**.

10.7 Configuring Mutual TLS Authentication

This section describes how to configure mutual (two-way) TLS authentication.

10.7.1 TLS for SIP Clients

When Secure SIP (SIPS) is implemented using TLS, it is sometimes required to use two-way (mutual) authentication between the device and a SIP user agent (client). When the device acts as the TLS server in a specific connection, the device demands the authentication of the SIP client’s certificate. Both the device and the client use certificates from a CA to authenticate each other, sending their X.509 certificates to one another during the TLS handshake. Once the sender is verified, the receiver sends its' certificate to the sender for verification. SIP signaling starts when authentication of both sides completes successfully.

TLS mutual authentication can be configured for specific calls by enabling mutual authentication on the SIP Interface used by the call. The TLS Context associated with the SIP Interface or Proxy Set belonging to these calls are used.

Note: SIP mutual authentication can also be configured globally for all calls, using the 'TLS Mutual Authentication' parameter (SIPSRequireClientCertificate) in the General Security Settings page (**Configuration** tab > **VoIP** menu > **Security** > **General Security Settings**).

➢ **To configure mutual TLS authentication for SIP messaging:**

1. Enable two-way authentication on the specific SIP Interface:
   a. In the SIP Interface table (see "Configuring SIP Interfaces" on page 333), configure the 'TLS Mutual Authentication' parameter to **Enable** for the specific SIP Interface.
   b. Reset the device with a burn-to-flash for your settings to take effect.

2. Configure a TLS Context with the following certificates:
   - Import the certificate of the CA that signed the certificate of the SIP client into the Trusted Root Store so that the device can authenticate the client (see "Importing Certificates and Certificate Chain into Trusted Certificate Store" on page 116).
   - Make sure that the TLS certificate is signed by a CA that the SIP client trusts so that the client can authenticate the device.

10.7.2 TLS for Remote Device Management

By default, servers using TLS provide one-way authentication. The client is certain that the identity of the server is authentic. When an organizational PKI is used, two-way authentication may be desired - both client and server should be authenticated using X.509 certificates. This is achieved by installing a client certificate on the management PC and loading the root CA's certificate to the device's Trusted Root Certificate Store. The Trusted Root Certificate file may contain more than one CA certificate combined, using a text editor.

➢ **To enable mutual TLS authentication for HTTPS:**

1. On the Web Security Settings page (see "Configuring Web Security Settings" on page 77), configure the 'Secured Web Connection (HTTPS)' field to **HTTPS Only**. The setting ensures that you have a method for accessing the device in case the client certificate doesn't work. Restore the previous setting after testing the configuration.
2. In the TLS Contexts table (see "Configuring TLS Certificate Contexts" on page 107), select the required TLS Context row, and then click the **TLS Context Trusted Root Certificates** button, located below the table; the Trusted Certificates page appears.
3. Click the **Import** button, and then select the certificate file.
4. Wait until the import operation finishes successfully.
5. On the Web Security Settings page, configure the 'Requires Client Certificates for HTTPS connection' field to **Enable**.
6. Reset the device with a burn-to-flash for your settings to take effect.

When a user connects to the secured Web interface of the device:

- If the user has a client certificate from a CA that is listed in the Trusted Root Certificate file, the connection is accepted and the user is prompted for the system password.
- If both the CA certificate and the client certificate appear in the Trusted Root Certificate file, the user is not prompted for a password (thus, providing a single-sign-on experience - the authentication is performed using the X.509 digital signature).
- If the user does not have a client certificate from a listed CA or does not have a client certificate, the connection is rejected.

Notes:

- The process of installing a client certificate on your PC is beyond the scope of this document. For more information, refer to your operating system documentation, and/or consult your security administrator.
- The root certificate can also be loaded via the Automatic Update facility, using the HTTPSRootFileName *ini* file parameter.
- You can enable the device to check whether a peer's certificate has been revoked by an OCSP server per TLS Context (see "Configuring TLS Certificate Contexts" on page 107).

10.8 Configuring TLS Server Certificate Expiry Check

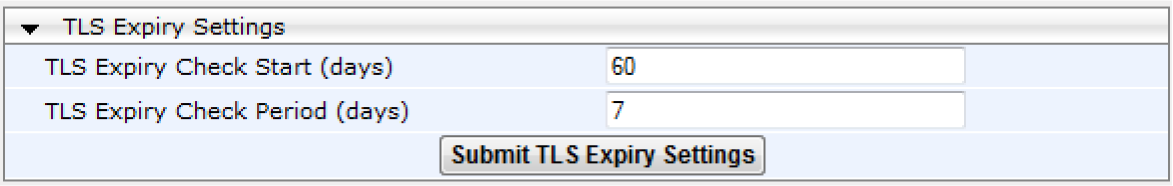
You can also configure the TLS Server Certificate Expiry Check feature, whereby the device periodically checks the validation date of the installed TLS server certificates. You can also configure the device to send a notification SNMP trap event (acCertificateExpiryNotification) at a user-defined number of days before the installed TLS server certificate is to expire. The trap indicates the TLS Context to which the certificate belongs.

> **Note:** TLS certificate expiry check is configured globally for all TLS Contexts.

➢ To configure TLS certificate expiry checks and notification:

1. Open the TLS Contexts page (**Configuration** tab > **System** menu > **TLS Contexts**).
2. Scroll down the page to the **TLS Expiry Settings** group:

Figure 10-10: TLS Expiry Settings Group



TLS Expiry Settings

TLS Expiry Check Start (days)	60
TLS Expiry Check Period (days)	7

3. In the 'TLS Expiry Check Start' field, enter the number of days before the installed TLS server certificate is to expire at which time the device sends an SNMP trap event to notify of this.
4. In the 'TLS Expiry Check Period' field, enter the periodical interval (in days) for checking the TLS server certificate expiry date. By default, the device checks the certificate every 7 days.
5. Click the **Submit TLS Expiry Settings** button.

This page is intentionally left blank.

11 Date and Time

The date and time of the device can be configured manually or it can be obtained automatically from a Simple Network Time Protocol (SNTP) server.

11.1 Configuring Automatic Date and Time using SNTP

The device's Simple Network Time Protocol (SNTP) client functionality generates requests and reacts to the resulting responses using the NTP Version 3 protocol definitions (according to RFC 1305). Through these requests and responses, the device, as an NTP client, synchronizes the system time to a time source within the network, thereby eliminating any potential issues should the local system clock 'drift' during operation. The NTP client follows a simple process in managing system time: the NTP client requests an NTP update, receives an NTP response, and then updates the local system clock based on an NTP server within the network. The client requests a time update from the user-defined NTP server (IP address or FQDN) at a user-defined update interval. Typically, this update interval is every 24 hours based on when the system was restarted.

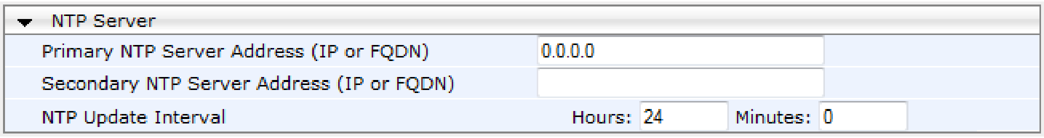
You can also configure the device to authenticate and validate the NTP messages received from the NTP server. Authentication is done using an authentication key with the MD5 cryptographic hash algorithm. When this feature is enabled, the device ignores NTP messages received without authentication.

The following procedure describes how to configure SNTP. For detailed descriptions of the configuration parameters, see "NTP and Daylight Saving Time Parameters" on page 838.

➢ **To configure SNTP using the Web interface:**

1. Open the Time and Date page (**Configuration** tab > **System** menu > **Time And Date**), and then scroll down to the 'NTP Sever' group:

Figure 11-1: NTP Parameters on Time and Date Page



NTP Server

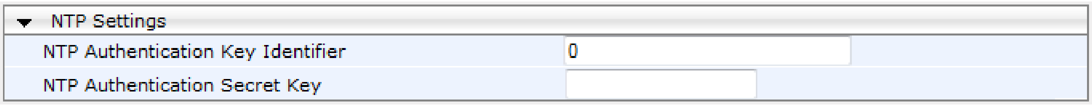
Primary NTP Server Address (IP or FQDN) 0.0.0.0

Secondary NTP Server Address (IP or FQDN)

NTP Update Interval Hours: 24 Minutes: 0

2. Configure the NTP server address:
   - In the 'Primary NTP Server Address' (NTPServerIP) field, configure the primary NTP server's address (IP or FQDN).
   - In the 'Secondary NTP Server Address' (NTPSecondaryServerIP) field, configure the secondary NTP server.
3. In the 'NTP Updated Interval' (NTPUpdateInterval) field, configure the period after which the date and time of the device is updated.
4. Open the Application Settings page (**Configuration** tab > **System** menu > **Application Settings**), and then scroll down to the 'NTP Settings' group:

Figure 11-2: NTP Authentication Parameters on Application Settings Page



NTP Settings

NTP Authentication Key Identifier 0

NTP Authentication Secret Key

5. Configure NTP message authentication:
   - In the 'NTP Authentication Key Identifier' field, configure the NTP authentication key identifier.
   - In the 'NTP Authentication Secret Key' field, configure the secret authentication key shared between the device and the NTP server.
6. Verify that the device has received the correct date and time from the NTP server. The

date and time is displayed in the 'UTC Time' read-only field on the Time and Date page.

Note: If the device receives no response from the NTP server, it polls the NTP server for 10 minutes. If there is still no response after this duration, the device declares the NTP server as unavailable, by sending an SNMP alarm (acNTPServerStatusAlarm). The failed response could be due to incorrect configuration.

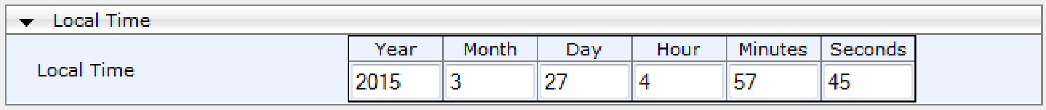
11.2 Configuring Date and Time Manually

You can manually configure the date and time of the device instead of using an NTP server (as described in "Configuring Automatic Date and Time using SNTP" on page 121).

➢ **To manually configure the device's date and time, using the Web interface:**

1. Open the Time and Date page (**Configuration** tab > **System** menu > **Time And Date**), and then scroll down to the 'Local Time' group:

Figure 11-3: Manually Configured Date and Time on Time and Date Page



▼ Local Time

	Year	Month	Day	Hour	Minutes	Seconds
Local Time	2015	3	27	4	57	45

2. In the 'Local Time' fields, enter the current date and time of the geographical location in which the device is installed:
   - Date:
     - 'Year' in yyyy format (e.g., "2015")
     - 'Month' in mm format (e.g., "3" for March)
     - 'Day' in dd format (e.g., "27")
   - Time:
     - 'Hour' in 24-hour format (e.g., "4" for 4 am)
     - 'Minutes' in mm format (e.g., "57")
     - 'Seconds' in ss format (e.g., "45")
3. Click **Submit**; the date and time is displayed in the 'UTC Time' read-only field.

Notes:

- If the device is configured to obtain the date and time from an SNTP server, the fields on this page are read-only, displaying the date and time received from the NTP server.
- After performing a hardware reset, the date and time are returned to default values and thus, you should subsequently update the date and time.

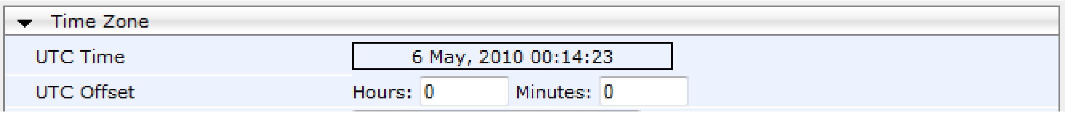
11.3 Configuring the Time Zone

You can configure the time zone in which the device is deployed. This is referred to as the Coordinated Universal Time (UTC) time offset and defines how many hours the device is from Greenwich Mean Time (GMT). For example, Germany Berlin is one hour ahead of GMT (UTC/GMT is +1 hour) and therefore, you would configure the offset to "1". USA New York is five hours behind GMT (UTC/GMT offset is -5 hours) and therefore, the offset is a minus value and configured as "-5".

➢ **To configure the time zone (UTC offset):**

1. Open the Time and Date page (**Configuration** tab > **System** menu > **Time And Date**), and then scroll down to the 'Time Zone' group:

Figure 11-4: UTC Offset on Time and Date Page



▾ Time Zone

UTC Time: 6 May, 2010 00:14:23

UTC Offset: Hours: 0 Minutes: 0

2. In the 'UTC Offset' fields (NTPServerUTCOffset), configure the time offset in relation to the UTC. For example, if your region is GMT +1 (an hour ahead), enter "1" in the 'Hours' field.
3. Click **Submit**; the updated time is displayed in the 'UTC Time' read-only field and the 'Local Time' fields.

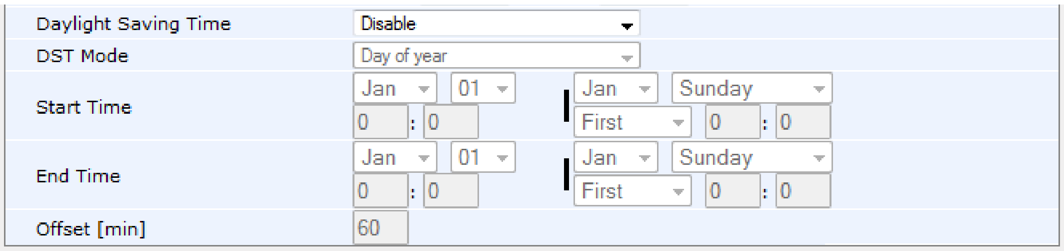
11.4 Configuring Daylight Saving Time

You can apply daylight saving time (DST) to the date and time of the device. DST defines a date range in the year (summer) where the time is brought forward so that people can experience more daylight. DST applies an offset of up to 60 minutes (default) to the local time. For example, Germany Berlin has DST from 30 March to 26 October, where the time is brought forward by an hour (e.g., 02:00 to 03:00 on 30 March). Therefore, you would configure the DST offset to 60 minutes (one hour).

➢ To configure DST through the Web interface:

1. Open the Time and Date page (**Configuration** tab > **System** menu > **Time And Date**), and then scroll down to the 'Time Zone' group:

Figure 11-5: Configuring DST



Daylight Saving Time	Disable			
DST Mode	Day of year			
Start Time	Jan 01 0 : 0		Jan Sunday First 0 : 0	
End Time	Jan 01 0 : 0		Jan Sunday First 0 : 0	
Offset [min]	60			

2. From the 'Daylight Saving Time' (DayLightSavingTimeEnable) drop-down list, select **Enable**.
3. From the 'DST Mode' drop-down list, select the range type for configuring the start and end dates for DST:
   - **Day of year:** The range is configured by exact date (day number of month), for example, from March 30 to October 30. If 'DST Mode' is set to **Day of year**, in the 'Start Time' (DayLightSavingTimeStart) and 'End Time' (DayLightSavingTimeEnd) drop-down lists, configure the period for which DST is relevant.
   - **Day of month:** The range is configured by month and day type, for example, from the last Sunday of March to the last Sunday of October. If 'DST Mode' is set to **Day of month**, in the 'Day of Month Start' and 'Day of Month End' drop-down lists, configure the period for which DST is relevant.
4. In the 'Offset' (DayLightSavingTimeOffset) field, configure the DST offset in minutes.
5. If the current date falls within the DST period, verify that it has been successful applied to the device's current date and time. You can view the device's date and time in the 'UTC Time' read-only field.

Part IV

General VoIP Configuration

12 Network

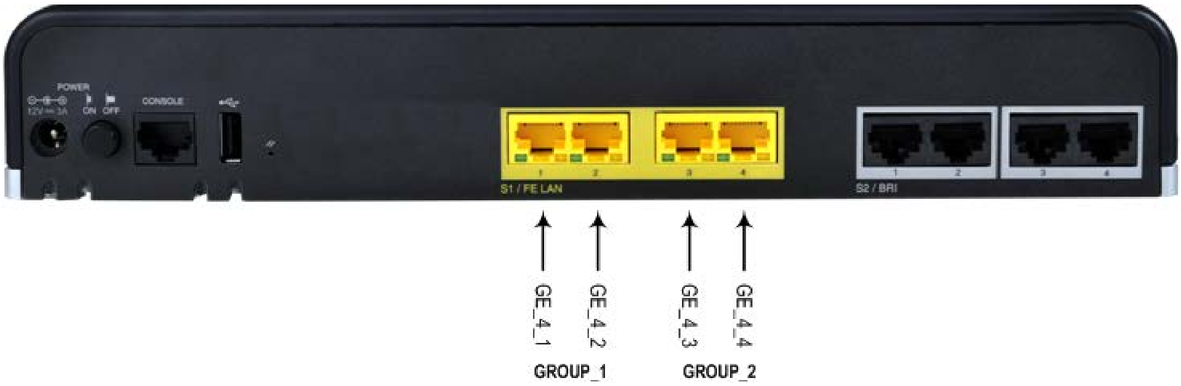
This section describes the network-related configuration.

12.1 Configuring Physical Ethernet Ports

The Physical Ports Settings table lets you configure the device's Ethernet ports. This includes port speed and duplex mode (half or full), and a brief description of the port.

The table also displays the status of the port (e.g., active) as well as the port group (*Ethernet Group*) to which the port belongs. You can assign up to two ports to an Ethernet Group. Ethernet Groups with two ports are used for 1+1 Ethernet port redundancy. For more information on Ethernet Groups and for assigning ports to Ethernet Groups, see "Configuring Ethernet Port Groups" on page 129.

The device's management tools (e.g., Web interface) use hard-coded strings to represent the physical ports, as shown below:



To view the mapping of the physical ports to these logical ports (strings) as well as view port status, use the CLI command, show voip ports. This displays the MAC address and port status (up or down) of the physical port and its corresponding logical port. Below shows an example of the mapping results from running this command:

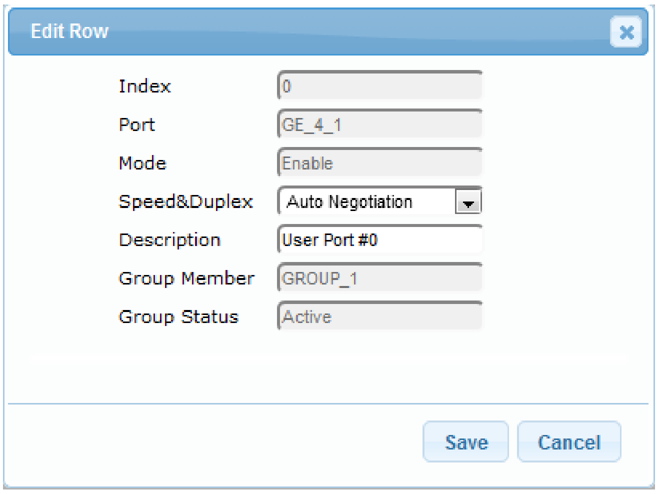
```
# show voip ports
Port Num    Port Name    MAC Address       Speed       Duplexity  Link Status Native VLAN
--------    ---------    -----------       -----       ---------  ----------- ------------
 1          GE4_1       00:1e:67:11:7c:28   100Mbps     FULL       UP          1
 2          GE4_2       00:1e:67:11:7c:28   100Mbps     FULL       DOWN
```

Note: All the LAN ports have the same MAC address. This is the MAC address of the device itself.

The following procedure describes how to configure the Ethernet ports through the Web interface. You can also configure it through ini file (PhysicalPortsTable) or CLI (configure voip > physical-port).

➢ To configure the physical Ethernet ports:

1. Open the Physical Ports Settings page (**Configuration** tab > **VoIP** menu > **Network** > **Physical Ports Table**).
2. Select a port that you want to configure by clicking its table row, and then clicking **Edit**; the following dialog box appears:



Index: 0
Port: GE_4_1
Mode: Enable
Speed&Duplex: Auto Negotiation
Description: User Port #0
Group Member: GROUP_1
Group Status: Active

3. Configure the port according to the parameters described in the table below.
4. Click **Submit**, and then save ("burn") your settings to flash memory.

Table 12-1: Physical Port Settings Table Parameter Descriptions

Parameter	Description
Port `port` [PhysicalPortsTable_Port]	(Read-only) Displays the Ethernet port number. The figure in the beginning of this section shows the mapping of this GUI port number to the physical port on the chassis.
Mode `mode` [PhysicalPortsTable_Mode]	(Read-only) Displays the mode of the port: ▪ **[0]** Disable ▪ **[1]** Enable (default)
Speed & Duplex `speed-duplex` [PhysicalPortsTable_SpeedDuplex]	Defines the speed and duplex mode of the port. ▪ **[0]** 10BaseT Half Duplex ▪ **[1]** 10BaseT Full Duplex ▪ **[2]** 100BaseT Half Duplex ▪ **[3]** 100BaseT Full Duplex ▪ **[4]** Auto Negotiation (default)
Description `port-description` [PhysicalPortsTable_PortDescription]	Defines an arbitrary description of the port. By default, the value is "User Port #<row index>".
Group Member `group-member` [PhysicalPortsTable_GroupMember]	(Read-only) Displays the Ethernet Group to which the port belongs. To assign the port to a different Ethernet Group, see "Configuring Ethernet Port Groups" on page 129.
Group Status `group-status` [PhysicalPortsTable_GroupStatus]	(Read-only) Displays the status of the port: ▪ "Active": Active port. When the Ethernet Group includes two ports and their transmit/receive mode is configured to 2RX 1TX or 2RX 2TX, both ports show "Active" ▪ "Redundant": Standby (redundant) port.

12.2 Configuring Ethernet Port Groups

The Ethernet Group Settings table lets you configure Ethernet Groups. An Ethernet Group represents a physical Ethernet port(s) on the device. You can assign an Ethernet Group with one, two, or no ports (*members*). When two ports are assigned to an Ethernet Group, 1+1 Ethernet port redundancy can be implemented in your network. In such a configuration, one port can be active while the other in standby mode or both ports can be active, depending on the ports' transmit (Tx) and receive (Rx) settings. This provides port redundancy within the Ethernet Group, whereby if an active port is disconnected, the device switches over to the other port in the Ethernet Group. If you configure an Ethernet Group with only one port, the Ethernet Group operates as a single port, without redundancy. You can also configure a combination of Ethernet Group types, where some contain one port and others two ports.

The Ethernet Group Settings table also lets you configure the transmit (Tx) and receive (Rx) settings for the Ethernet ports per Ethernet Group. The Tx/Rx setting applies only to Ethernet Groups that contain two ports. This setting determines whether either both ports or only one of the ports can receive and/or transmit traffic.

The maximum number of Ethernet Groups that can be configured is the same as the number of Ethernet ports provided by the device. Thus, the device supports up to four Ethernet Groups, each containing one port, or up to two Ethernet Groups, each containing two ports. By default, each Ethernet Group is assigned two ports; the other Ethernet Groups are empty.

You can assign Ethernet ports to IP network interfaces. This is done by first configuring an Ethernet Device with the required Ethernet Group containing the port or ports (see "Configuring Underlying Ethernet Devices" on page 131). Then by assigning the Ethernet Device to the IP network interface in the Interface table (see "Configuring IP Network Interfaces" on page 133). This enables physical separation of network interfaces, providing a higher level of segregation of sub-networks. Equipment connected to different physical ports is not accessible to one another; the only connection between them can be established by cross connecting them with media streams (VoIP calls).

The port names (strings) displayed in the Ethernet Group Settings table represent the physical ports on the device. For the mapping of these strings to the physical ports, see Configuring Physical Ethernet Ports on page 127.

The following procedure describes how to configure Ethernet Groups through the Web interface. You can also configure it through ini file (EtherGroupTable) or CLI (configure voip > ether-group).

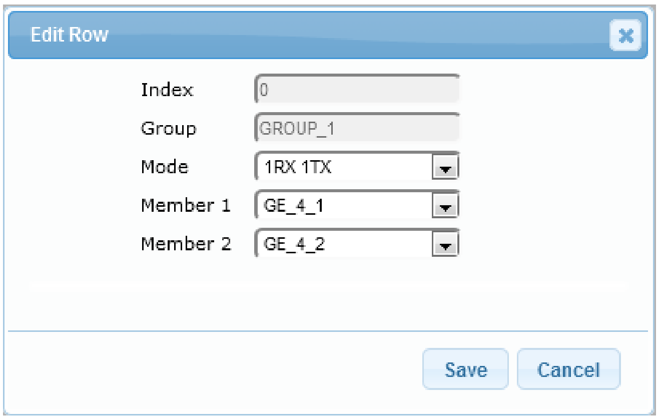
Notes:

- Before you can re-assign a port to a different Ethernet Group, you must first remove the port from its current Ethernet Group. To remove the port, either set the 'Member' field to **None** or to a different port.
- As all the ports have the same MAC address, you must connect each port to a different Layer-2 switch.
- When implementing 1+1 Ethernet port redundancy, each port in the Ethernet Group (port pair) must be connected to a different switch, but in the same subnet.

➢ **To configure Ethernet Groups:**

1. Open the Ethernet Group Settings table (**Configuration** tab > **VoIP** menu > **Network** > **Ethernet Groups Table**).

2. If the port that you want to assign to a specific Ethernet Group is already associated with another Ethernet Group, you must first **remove** the port from the currently associated Ethernet Group before you can associate it with the desired Ethernet Group:
   a. Select the Ethernet Group to which the port is currently associated, and then click **Edit**; the following dialog box appears:



Edit Row

Index: 0

Group: GROUP_1

Mode: 1RX 1TX

Member 1: GE_4_1

Member 2: GE_4_2

   b. Set the 'Member 1' or 'Member 2' field (depending on where the port appears) to **None** (or assign it a different port).
   c. Click **Submit**; the port is removed from the Ethernet Group.
3. Select the Ethernet Group that you want to configure and associate a port(s), and then click **Edit**.
4. Configure the Ethernet Group according to the parameters described in the table below.
5. Click **Submit**, and then save ("burn") your settings to flash memory.

Table 12-2: Ethernet Group Settings Parameter Descriptions

Parameter	Description
Group `group` [EtherGroupTable_Group]	(Read-only) Displays the Ethernet Group number.
Mode `mode` [EtherGroupTable_Mode]	Defines the mode of operation of the ports in the Ethernet Group. This applies only to Ethernet Groups containing two ports. ▪ **[2]** 1RX/1TX = (Default) At any given time, only a single port in the Ethernet Group can transmit and receive packets. If a link exists on both ports, then the active one is either the first to have a link up or the lower-numbered port if both have the same link up from start. ▪ **[3]** 2RX/1TX = Both ports in the Ethernet Group can receive packets, but only one port can transmit. The transmitting port is determined arbitrarily by the device. If the selected port fails at a later stage, a switchover to the redundant port is done, which begins to transmit as well as receive. ▪ **[4]** 2RX/2TX = Both ports in the Ethernet Group can receive and transmit packets. ▪ **[5]** Single = If the Ethernet Group contains only one port, use this option. ▪ **[6]** None = If no port is assigned to the Ethernet Group, use this option.
Member 1 `member1`	Assigns the first port to the Ethernet Group. To assign no port, set this field to **None**.

Parameter	Description
[EtherGroupTable_Member1]	**Note:** Before you can re-assign a port to a different Ethernet Group, you must first remove the port from its current Ethernet Group. To remove the port, either set this field to **None** or to a different port.
Member 2 `member2` [EtherGroupTable_Member2]	Assigns the second port to the Ethernet Group. To assign no port, set this field to **None**. **Note:** Before you can re-assign a port to a different Ethernet Group, you must first remove the port from its current Ethernet Group. To remove the port, either set this field to **None** or to a different port.

12.3 Configuring Underlying Ethernet Devices

The Ethernet Device table lets you configure up to 16 *Ethernet Devices*. An Ethernet Device represents a Layer-2 bridging device and is assigned a VLAN ID and an Ethernet Port Group. Multiple Ethernet Devices can be associated with the same Ethernet Group.

Once configured, you need to assign the Ethernet Device to an IP network interface in the Interface table ('Underlying Device' field) and/or with a static route in the Static Route table ('Device Name' field). You can assign the same Ethernet Device to multiple IP network interfaces, thereby implementing multi-homing (multiple addresses on the same interface/VLAN).

Each Ethernet Device (VLAN) can be configured with a VLAN tagging policy, which determines whether the Ethernet Device accepts tagged or untagged packets received on the Ethernet port associated with the Ethernet Device.

By default, the device provides a pre-configured Ethernet Device at Index 0 with the following settings:

- Name: "vlan 1"
- VLAN ID: 1
- Ethernet Group: GROUP 1
- Tagging Policy: Untagged

The pre-configured Ethernet Device is associated with the default IP network interface (OAMP) in the Interface table. The Untagged policy of the pre-configured Ethernet Device enables you to connect to the device using the default OAMP interface.

You can view configured Ethernet Devices that have been successfully applied to the device (saved to flash), in the Ethernet Device Status table. This page is accessed by clicking the **Ethernet Device Status Table** button, located at the bottom of the Ethernet Device table. The Ethernet Device Status table can also be accessed from the **Status & Diagnostics** tab > **VoIP Status** menu > **Ethernet Device Status Table** (see "Viewing Ethernet Device Status" on page 722).

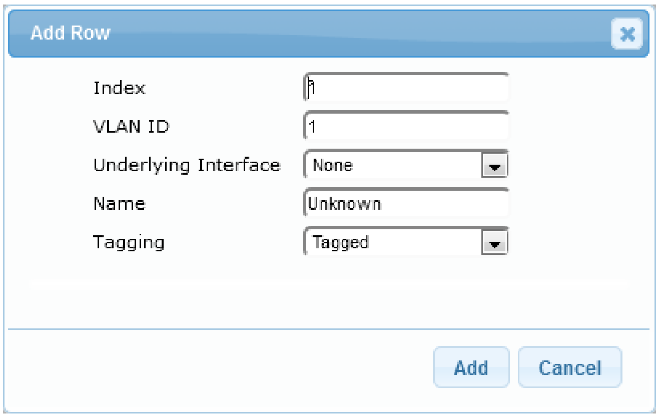
Note: You cannot delete an Ethernet Device that is associated with an IP network interface (in the Interface table). You can only delete it once you have disassociated it from the IP network interface.

The following procedure describes how to configure Ethernet devices through the Web interface. You can also configure it through ini file (DeviceTable) or CLI (config-voip > interface network-dev).

➢ **To configure an Ethernet Device:**

1. Open the Ethernet Device table (**Configuration** tab > **VoIP** menu > **Network** > **Ethernet Device Table**).

2. Click **Add**; the following dialog box appears:



Add Row

Index: 1
VLAN ID: 1
Underlying Interface: None
Name: Unknown
Tagging: Tagged

3. Configure an Ethernet Device according to the parameters described in the table below.
4. Click **Add**.

Table 12-3: Ethernet Device Table Parameter Descriptions

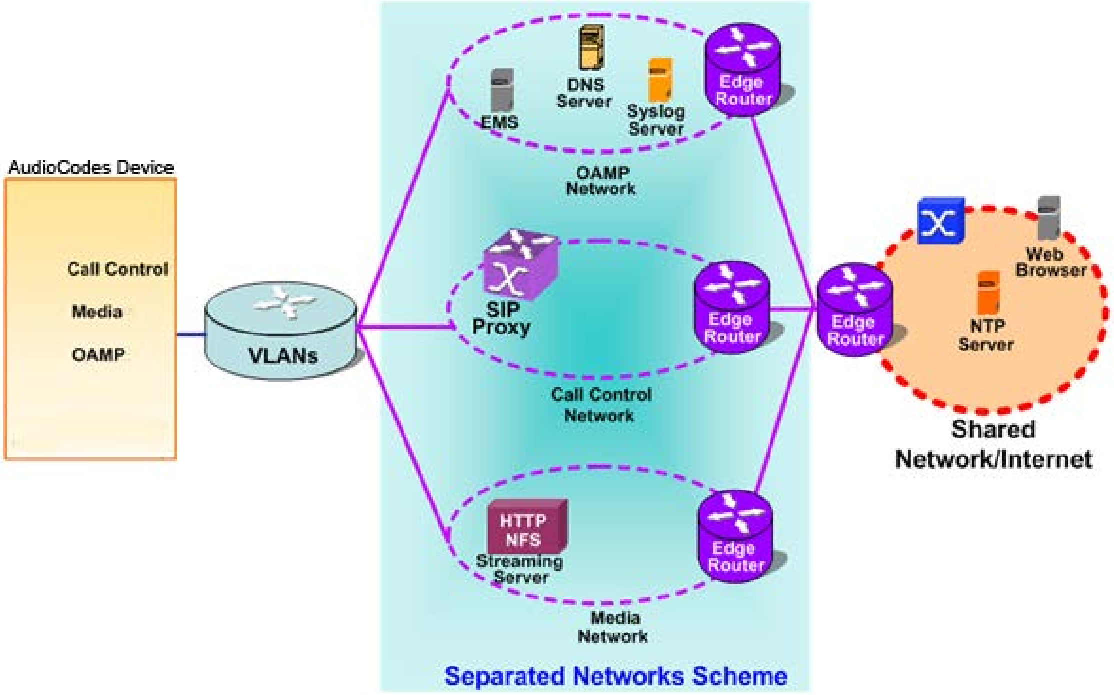
Parameter	Description
Index [DeviceTable_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
VLAN ID `vlan-id` [DeviceTable_VlanID]	Defines a VLAN ID for the Ethernet Device. The valid value is 1 to 3999. The default value is 1. **Note:** Each Ethernet Port Group must have a unique VLAN ID.
Underlying Interface `underlying-if` [DeviceTable_UnderlyingInterface]	Assigns an Ethernet Port Group to the Ethernet Device. For configuring Ethernet Port Groups, see Configuring Ethernet Port Groups on page 129. **Note:** The parameter is mandatory.
Name `name` [DeviceTable_DeviceName]	Defines a name for the Ethernet Device. This name is used to associate the Ethernet Device with an IP network interface in the Interface table ('Underlying Device' field - see "Configuring IP Network Interfaces" on page 133) and/or with a static route in the Static Route table ('Device Name' field - see "Configuring Static IP Routing" on page 141).

Parameter	Description
Tagging `tagging` [DeviceTable_Tagging]	Defines VLAN tagging per Ethernet Device. ▪ [0] Untagged = (Default of pre-configured Ethernet Device) The Ethernet Device accepts untagged packets as well as packets with the same VLAN ID as configured for the Ethernet Device. Incoming untagged packets are assigned the VLAN ID of the Ethernet Device. The Ethernet Device sends these VLAN packets untagged (i.e., removes the VLAN ID). ▪ [1] Tagged = ((Default for new Ethernet Devices) The Ethernet Device accepts packets that have the same VLAN ID as configured for the Ethernet Device and sends packets with this VLAN ID. For all Ethernet Devices that are associated with the same Ethernet Group ('Underlying Interface') and set to **Tagged**, incoming untagged packets received on this Ethernet Group are discarded. **Note:** Only one Ethernet Device can be configured as Untagged per associated Ethernet Group (port group). In other words, if multiple Ethernet Devices are associated with the same Ethernet Group, only one of these Ethernet Devices can be set to **Untagged**; all the others must be set to **Tagged**.

12.4 Configuring IP Network Interfaces

You can configure a single VoIP network interface for all applications, including OAMP (management traffic), call control (SIP signaling messages), and media (RTP traffic), or you can configure multiple logical, IP network interfaces for these applications. You may need to logically separated network segments for these applications for administration and security. This can be achieved by employing Layer-2 VLANs and Layer-3 subnets. The figure below illustrates a typical network architecture where the device is configured with three network interfaces, each representing the OAMP, call control, and media applications. The device is connected to a VLAN-aware switch for directing traffic from and to the device to the three separated Layer-3 broadcast domains according to VLAN tags (middle pane).

Figure 12-1: Multiple Network Interfaces



The device is shipped with a default OAMP interface. For more information, see "Default OAMP IP Address" on page 37. The Interface table lets you change this OAMP interface and configure additional network interfaces for control and media, if necessary. You can configure up to 12 interfaces, consisting of up to 11 Control and Media interfaces, and 1 OAMP interface. Each IP interface is configured with the following:

- Application type allowed on the interface:
  - Control: call control signaling traffic (i.e., SIP)
  - Media: RTP traffic
  - Operations, Administration, Maintenance and Provisioning (OAMP): management (i.e., Web, CLI, and SNMP based management)
- IP address (IPv4 and IPv6) and subnet mask (prefix length)
- For configuring Quality of Service (QoS), see "Configuring the QoS Settings" on page 144.
- Default Gateway: Traffic from this interface destined to a subnet that does not meet any of the routing rules (local or static) are forwarded to this gateway
- Primary and secondary domain name server (DNS) addresses (optional)

- Underlying Ethernet Device: Layer-2 bridging device and assigned a VLAN ID. As the Ethernet Device is associated with an Ethernet Port Group, this is useful for setting trusted and un-trusted networks on different physical Ethernet ports. Multiple entries in the Interface table may be associated with the same Ethernet Device, providing multi-homing IP configuration (i.e., multiple IP addresses on the same interface/VLAN).

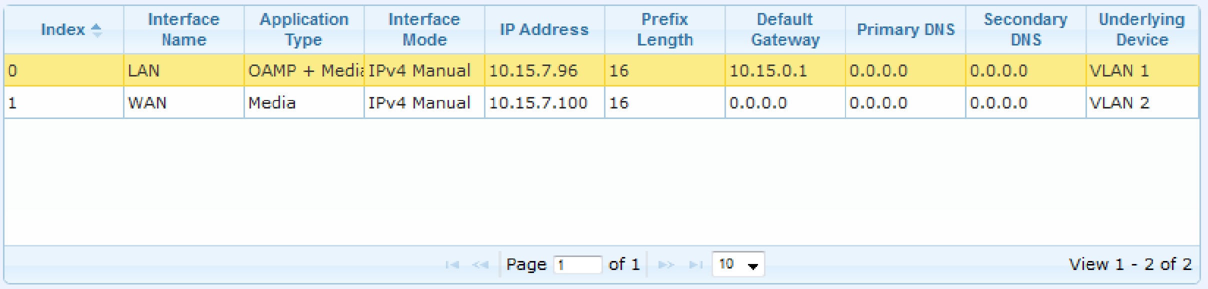
Complementing the Interface table is the Static Route table, which lets you configure static routing rules for non-local hosts/subnets. For more information, see "Configuring Static IP Routing" on page 141.

Note: Before configuring IP interfaces, it is recommended that you read the IP interface configuration guidelines in "Interface Table Configuration Guidelines" on page 137.

The following procedure describes how to configure the IP network interfaces through the Web interface. You can also configure it through ini file (InterfaceTable) or CLI (configure voip/interface network-if).

➢ To configure IP network interfaces:

1. Open the Interface Table (**Configuration** tab > **VoIP** menu > **Network** > **IP Interfaces Table**).



Index	Interface Name	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Primary DNS	Secondary DNS	Underlying Device
0	LAN	OAMP + Medi	IPv4 Manual	10.15.7.96	16	10.15.0.1	0.0.0.0	0.0.0.0	VLAN 1
1	WAN	Media	IPv4 Manual	10.15.7.100	16	0.0.0.0	0.0.0.0	0.0.0.0	VLAN 2

2. Click **Add**; a dialog box appears.
3. Configure the IP network interface according to the parameters described in the table below.
4. Click **Add**.

Notes:

- If you modify the OAMP interface's address, after clicking **Add** in the dialog box you will lose connectivity with the device and need to access the device with the new address.
- If you edit or delete an IP interface, current calls using the interface are immediately terminated.
- If you delete an IP interface, row indices of other tables (e.g., Media Realm table) that are associated with the deleted IP interface, lose their association with the interface ('Interface Name' field displays "None") and the row indices become invalid.

To view configured network interfaces that are currently active, click the **IP Interface Status Table** button. For more information, see "Viewing Active IP Interfaces" on page 722.

Table 12-4: Interface Table Parameters Description

Parameter	Description
Table parameters	
Index `network-if` [InterfaceTable_Index]	Table index row of the interface. The range is 0 to 11.
Application Type `application-type` [InterfaceTable_ApplicationTypes]	Defines the applications allowed on the interface. ▪ **[0]** OAMP = Operations, Administration, Maintenance and Provisioning (OAMP) applications (e.g., Web, Telnet, SSH, and SNMP). ▪ **[1]** Media = Media (i.e., RTP streams of voice). ▪ **[2]** Control = Call Control applications (e.g., SIP). ▪ **[3]** OAMP + Media = OAMP and Media applications. ▪ **[4]** OAMP + Control = OAMP and Call Control applications. ▪ **[5]** Media + Control = Media and Call Control applications. ▪ **[6]** OAMP + Media + Control = All application types are allowed on the interface.
Interface Mode [InterfaceTable_InterfaceMode]	Defines the method that the interface uses to acquire its IP address. ▪ [3] IPv6 Manual Prefix = IPv6 manual prefix IP address assignment. The IPv6 prefix (higher 64 bits) is set manually while the interface ID (the lower 64 bits) is derived from the device's MAC address. ▪ [4] IPv6 Manual = IPv6 manual IP address (128 bits) assignment. ▪ [10] IPv4 Manual = IPv4 manual IP address (32 bits) assignment.
IP Address `ip-address` [InterfaceTable_IPAddress]	Defines the IPv4/IPv6 address, in dotted-decimal notation.
Prefix Length `prefix-length` [InterfaceTable_PrefixLength]	Defines the prefix length of the related IP address. This is a Classless Inter-Domain Routing (CIDR)-style representation of a dotted-decimal subnet notation. The CIDR-style representation uses a suffix indicating the number of bits which are set in the dotted-decimal format. For example, 192.168.0.0/16 is synonymous with 192.168.0.0 and subnet 255.255.0.0. This CIDR lists the number of ‘1’ bits in the subnet mask (i.e., replaces the standard dotted-decimal representation of the subnet mask for IPv4 interfaces). For example, a subnet mask of 255.0.0.0 is represented by a prefix length of 8 (i.e., 11111111 00000000 00000000 00000000) and a subnet mask of 255.255.255.252 is represented by a prefix length of 30 (i.e., 11111111 11111111 11111111 11111100). The prefix length is a Classless Inter-Domain Routing (CIDR) style presentation of a dotted-decimal subnet notation. The CIDR-style presentation is the latest method for interpretation of IP addresses. Specifically, instead of using eight-bit address blocks, it uses the variable-length subnet masking technique to allow allocation on arbitrary-length prefixes. The prefix length for IPv4 must be set to a value from 0 to 30. The prefix length for IPv6 must be set to a value from 0 to 64.

Parameter	Description
Default Gateway `gateway` [InterfaceTable_Gateway]	Defines the IP address of the default gateway for the interface. When traffic is sent from this interface to an unknown destination (i.e., not in the same subnet and not defined for any static routing rule), it is forwarded to this default gateway.
Interface Name `name` [InterfaceTable_InterfaceName]	Defines a name for the interface. This name is used in various configuration tables to associate the network interface with other configuration entities such as Media Realms. It is also displayed in management interfaces (Web, CLI, and SNMP) for clarity where it has no functional use. The valid value is a string of up to 16 characters.
Primary DNS `primary-dns` [InterfaceTable_PrimaryDNSServerIPAddress]	(Optional) Defines the primary DNS server's IP address (in dotted-decimal notation), which is used for translating domain names into IP addresses for the interface. By default, no IP address is defined.
Secondary DNS `secondary-dns` [InterfaceTable_SecondaryDNSServerIPAddress]	(Optional) Defines the secondary DNS server's IP address (in dotted-decimal notation), which is used for translating domain names into IP addresses for the interface. By default, no IP address is defined.
Underlying Device `underlying-dev` [InterfaceTable_UnderlyingDevice]	Assigns an Ethernet Device to the IP interface. An Ethernet Device is a VLAN ID associated with a physical Ethernet port (Ethernet Group). To configure Ethernet Devices, see Configuring Underlying Ethernet Devices on page 131.

12.4.1 Assigning NTP Services to Application Types

You can associate the Network Time Protocol (NTP) application with the OAMP or Control application type. This is done using the EnableNTPasOAM ini file parameter.

12.4.2 Multiple Interface Table Configuration Summary and Guidelines

The Interface table configuration must adhere to the following rules:

- Multiple Control and Media interfaces can be configured with overlapping IP addresses and subnets.
- The prefix length replaces the dotted-decimal subnet mask presentation and **must** have a value of 0-30 for IPv4 addresses and a value of 0-64 for IPv6 addresses.
- **One** OAMP interface must be configured and this **must** be an IPv4 address. This OAMP interface can be combined with Media and Control.
- At least one Control interface **must** be configured.
- At least one Media interface **must** be configured.
- Multiple Media and/or Control interfaces can be configured with an IPv6 address.
- The network interface types can be combined:
    - Example 1:
        - One combined OAMP-Media-Control interface with an IPv4 address
    - Example 2:
        - One OAMP interface with an IPv4 address

    - ♦ One or more Control interfaces with IPv4 addresses
    - ♦ One or more Media interfaces with IPv4 interfaces
  - Example 3:
    - ♦ One OAMP with an IPv4 address
    - ♦ One combined Media-Control interface with IPv4 address
    - ♦ One combined Media-Control interface with IPv6 address
- Each network interface can be configured with a Default Gateway. The address of the Default Gateway **must** be in the same subnet as the associated interface. Additional static routing rules can be configured in the Static Route table.
- The interface name **must** be configured (mandatory) and must be unique for each interface.
- For IPv4 addresses, the 'Interface Mode' column must be set to IPv4 Manual. For IPv6 addresses, this column must be set to IPv6 Manual or IPv6 Manual Prefix.

> **Note:** Upon device start up, the Interface table is parsed and passes comprehensive validation tests. If any errors occur during this validation phase, the device sends an error message to the Syslog server and falls back to a "safe mode", using a single interface without VLANs. Ensure that you view the Syslog messages that the device sends in system startup to see if any errors occurred.

12.4.3 Networking Configuration Examples

This section provides configuration examples of networking interfaces.

12.4.3.1 One VoIP Interface for All Applications

This example describes the configuration of a single VoIP interface for all applications:

1. **Interface table:** Configured with a single interface for OAMP, Media and Control:

Table 12-5: Example of Single VoIP Interface in Interface Table

Index	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Underlying Device	Interface Name
0	OAMP, Media & Control	IPv4	192.168.0.2	16	192.168.0.1	1	myInterface

2. **Static Route table:** Two routes are configured for directing traffic for subnet 201.201.0.0/16 to 192.168.11.10, and all traffic for subnet 202.202.0.0/16 to 192.168.11.1:

Table 12-6: Example of Static Route Table

Destination	Prefix Length	Gateway
201.201.0.0	16	192.168.11.10
202.202.0.0	16	192.168.11.1

3. The **NTP** applications remain with their default application types.

12.4.3.2 VoIP Interface per Application Type

This example describes the configuration of three VoIP interfaces; one for each application type:

1. **Interface table:** Configured with three interfaces, each for a different application type, i.e., one for OAMP, one for Call Control, and one for RTP Media, and each with a different VLAN ID and default gateway:

Table 12-7: Example of VoIP Interfaces per Application Type in Interface Table

Index	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Underlying Device	Interface Name
0	OAMP	IPv4 Manual	192.168.0.2	16	192.168.0.1	1	ManagementIF
1	Control	IPv4 Manual	200.200.85.14	24	200.200.85.1	200	myControlIF
2	Media	IPv4 Manual	211.211.85.14	24	211.211.85.1	211	myMediaIF

2. **Static Route table:** A routing rule is required to allow remote management from a host in 176.85.49.0 / 24:

Table 12-8: Example Static Route Table

Destination	Prefix Length	Gateway
176.85.49.0	24	192.168.11.1

3. All other parameters are set to their respective default values. The NTP application remains with its default application types.

12.4.3.3 VoIP Interfaces for Combined Application Types

This example describes the configuration of multiple interfaces for the following applications:

- One interface for the OAMP application.
- Interfaces for Call Control and Media applications, where two of them are IPv4 interfaces and one is an IPv6 interface.

1. **Interface table:**

Table 12-9: Example of VoIP Interfaces of Combined Application Types in Interface Table

Index	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Underlying Device	Interface Name
0	OAMP	IPv4 Manual	192.168.0.2	16	192.168.0.1	1	Mgmt
1	Media & Control	IPv4 Manual	200.200.85.14	24	200.200.85.1	201	MediaCntrl1
2	Media & Control	IPv4 Manual	200.200.86.14	24	200.200.86.1	202	MediaCntrl2
3	Media & Control	IPv6 Manual	2000::1:200:200:86:14	64	::	202	V6CntrlMedia2

2. **Static Route table:** A routing rule is required to allow remote management from a host in 176.85.49.0/24:

Table 12-10: Example of Static Route Table

Destination	Prefix Length	Gateway
176.85.49.0	24	192.168.0.10

3. The NTP application is configured (through the ini file) to serve as OAMP applications:

```
EnableNTPasOAM = 1
```

4. DiffServ table:
   - Layer-2 QoS values are assigned:
     - For packets sent with DiffServ value of 46, set VLAN priority to 6
     - For packets sent with DiffServ value of 40, set VLAN priority to 6
     - For packets sent with DiffServ value of 26, set VLAN priority to 4
     - For packets sent with DiffServ value of 10, set VLAN priority to 2
   - Layer-3 QoS values are assigned:
     - For Media Service class, the default DiffServ value is set to 46
     - For Control Service class, the default DiffServ value is set to 40
     - For Gold Service class, the default DiffServ value is set to 26
     - For Bronze Service class, the default DiffServ value is set to 10

12.4.3.4 VoIP Interfaces with Multiple Default Gateways

Below is a configuration example using default gateways per IP network interface. In this example, the default gateway for OAMP is 192.168.0.1 and for Media and Control it is 200.200.85.1.

Table 12-11: Configured Default Gateway Example

Index	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Underlying Device	Interface Name
0	OAMP	IPv4 Manual	192.168.0.2	16	192.168.0.1	100	Mgmt
1	Media & Control	IPv4 Manual	200.200.85.14	24	200.200.85.1	200	CntrlMedia

A separate Static Route table lets you configure static routing rules. Configuring the following static routing rules enables OAMP applications to access peers on subnet 17.17.0.0 through the gateway 192.168.10.1 (which is not the default gateway of the interface), and Media & Control applications to access peers on subnet 171.79.39.0 through the gateway 200.200.85.10 (which is not the default gateway of the interface).

Table 12-12: Separate Static Route Table Example

Destination	Prefix Length	Gateway	Underlying Device
17.17.0.0	16	192.168.10.1	100
171.79.39.0	24	200.200.85.10	200

12.5 Configuring Static IP Routes

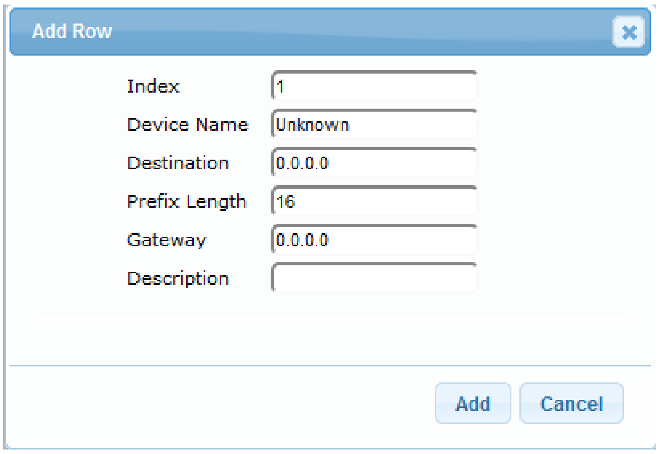
The Static Route table lets you configure up to 30 static IP routing rules. Using static routes lets you communicate with LAN networks that are not located behind the Default Gateway specified for the IP network interface, configured in the Interface table, from which the packets are sent. Before sending an IP packet, the device searches the Static Route table for an entry that matches the requested destination host/network. If an entry is found, the device sends the packet to the gateway that is configured for the static route. If no explicit entry is found, the packet is sent to the Default Gateway configured for the IP network interface.

You can view the status of the configured static routes in the IP Routing Status table. This page can be accessed by clicking the **Static Route Status Table** button, located at the bottom of the Static Route table page, or it can be accessed from the Navigation tree under the **Status & Diagnostics** tab (see "Viewing Static Routes Status" on page 723).

The following procedure describes how to configure static routes through the Web interface. You can also configure it through ini file (StaticRouteTable) or CLI (configure voip > routing static).

➢ **To configure a static IP route:**

1. Open the Static Route table (**Configuration** tab > **VoIP** menu > **Network** > **Static Route Table**).
2. Click **Add**; the following dialog box appears:



Index 1
Device Name Unknown
Destination 0.0.0.0
Prefix Length 16
Gateway 0.0.0.0
Description

3. Configure a static route according to the parameters described in the table below.
4. Click **Add**, and then reset the device with a burn-to-flash for your settings to take effect.

Note: You can delete only static routing rules that are inactive.

Table 12-13: Static Route Table Parameter Descriptions

Parameter	Description
Index [StaticRouteTable_Index]	Defines an index number for the new table row. The valid value is 0 to 29. **Note:** Each row must be configured with a unique index.

Parameter	Description
Device Name `device-name` [StaticRouteTable_DeviceName]	Assigns an IP network interface through which the static route's Gateway is reached. The Device Name (or underlying device) represents the IP network interface, including VLAN ID and associated physical port(s). The value must be identical to the value in the 'Underlying Device' parameter of the required IP network interface in the Interface table (see Configuring IP Network Interfaces on page 133). For configuring Ethernet Devices, see Configuring Underlying Ethernet Devices on page 131.
Destination `destination` [StaticRouteTable_Destination]	Defines the IP address of the destination host/network. The destination can be a single host or a whole subnet, depending on the prefix length configured for this routing rule.
Prefix Length `prefix-length` [StaticRouteTable_PrefixLength]	Defines the Classless Inter-Domain Routing (CIDR)-style representation of a dotted-decimal subnet notation of the destination host/network. The CIDR-style representation uses a suffix indicating the number of bits that are set in the dotted-decimal format. For example, the value 16 represents subnet 255.255.0.0. The value must be 0 to 31 for IPv4 interfaces and a value of 0 to 64 for IPv6 interfaces.
The address of the host/network you want to reach is determined by an AND operation that is applied to the fields 'Destination' and 'Prefix Length'. For example, to reach the network 10.8.x.x, enter 10.8.0.0 in the 'Destination' field and 16 in the 'Prefix Length'. As a result of the AND operation, the value of the last two octets in the 'Destination' field is ignored. To reach a specific host, enter its IP address in the 'Destination' field and 32 in the 'Prefix Length' field.	
Gateway `gateway` [StaticRouteTable_Gateway]	Defines the IP address of the Gateway (next hop) used for traffic destined to the subnet/host defined in the 'Destination' / 'Prefix Length' field. **Notes:** ▪ The Gateway's address must be in the same subnet as the IP address of the network interface that is associated with the static route (using the 'Device Name' parameter - see above). ▪ The IP network interface associated with the static route must be of the same IP address family (IPv4 or IPv6).
Description `description` [StaticRouteTable_Description]	Defines an arbitrary name to easily identify the static route rule. The valid value is a string of up to 20 characters.

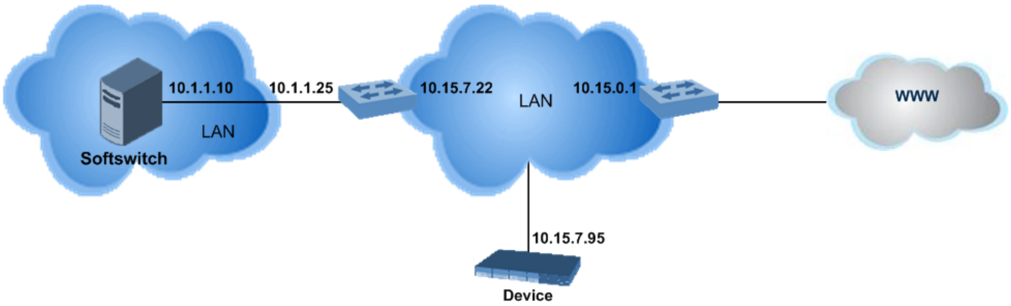
12.5.1 Configuration Example of Static IP Routes

An example of the use for static routes is shown in the figure below. In the example scenario, the device needs to communicate with a softswitch at IP address 10.1.1.10. However, the IP network interface from which packets destined for 10.1.1.10 is sent, is configured to send the packets to a Default Gateway at 10.15.0.1. Therefore, the packets do not reach the softswitch. To resolve this problem, a static route is configured to specify the correct gateway (10.15.7.22) in order to reach the softswitch.

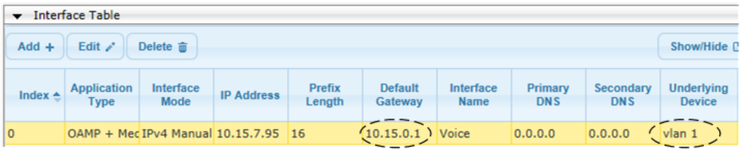
Note the following configuration:

- The static route is configured with a subnet mask of 24 (255.255.255.0), enabling the device to use the static route to send all packets destined for 10.1.1.**x** to this gateway and therefore, to the network in which the softswitch resides.
- The static route in the Static Route table is associated with the IP network interface in the Interface table, using the 'Device Name' and 'Underlying Device' fields, respectively.
- The static route's Gateway address in the Static Route table is in the same subnet as the IP address of the IP network interface in the Interface table.

Figure 12-2: Example of using a Static Route

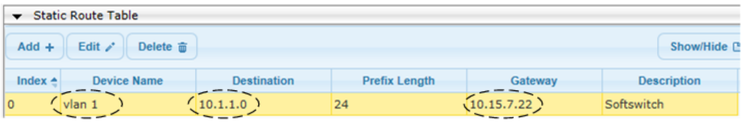


No Static Route:
The device sends packets to 10.15.0.1, which is the Default Gateway defined for this IP network interface in the Interface table. Therefore, the device will not succeed in reaching the softswitch.



Index	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Interface Name	Primary DNS	Secondary DNS	Underlying Device
0	OAMP + Mec	IPv4 Manual	10.15.7.95	16	10.15.0.1	Voice	0.0.0.0	0.0.0.0	vlan 1

Static Route Configured:
A static route with the correct gateway is needed for routing to the softswitch. The device communicates with the softswitch (10.1.1.0/24) using the gateway 10.15.7.22.
Note: The device first searches for a matching route in the Static Route table. If not found, it uses the default gateway defined in the Interface table.



Index	Device Name	Destination	Prefix Length	Gateway	Description
0	vlan 1	10.1.1.0	24	10.15.7.22	Softswitch

12.5.2 Troubleshooting the Routing Table

When adding a new static route to the Static Route table, the added rule passes a validation test. If errors are found, the static route is rejected and not added to the table. Failed static route validations may result in limited connectivity (or no connectivity) to the destinations specified in the incorrect static route. For any error found in the Static Route table or failure to configure a static route, the device sends a notification message to the Syslog server reporting the problem.

Common static routing configuration errors may include the following:

- The IP address specified in the 'Gateway' field is unreachable from the IP network interface associated with the static route.
- The same destination is configured in two different static routes.
- More than 30 static routes have been configured.

Note: If a static route is required to access OAMP applications (for remote management, for example) and the route is not configured correctly, the route is not added and the device is not accessible remotely. To restore connectivity, the device must be accessed locally from the OAMP subnet and the required routes be configured.

12.6 Configuring Quality of Service

The QoS Settings page lets you configure Layer-2 and Layer-3 Quality of Service (QoS). Differentiated Services (DiffServ) is an architecture providing different types or levels of service for IP traffic. DiffServ (according to RFC 2474), prioritizes certain traffic types based on priority, accomplishing a higher-level QoS at the expense of other traffic types. By prioritizing packets, DiffServ routers can minimize transmission delays for time-sensitive packets such as VoIP packets.

You can assign DiffServ to the following class of services (CoS) and assign VLAN priorities (IEEE 802.1p) to various values of DiffServ:

-
- Media Premium – RTP packets sent to the LAN
- Control Premium – control protocol (SIP) packets sent to the LAN
- Gold – HTTP streaming packets sent to the LAN
- Bronze – OAMP packets sent to the LAN

The Layer-3 QoS parameters define the values of the DiffServ field in the IP header of the frames related to a specific service class. The Layer-2 QoS parameters define the values for the 3 priority bits in the VLAN tag according to the value of the DiffServ field in the packet IP header (according to the IEEE 802.1p standard). The DiffServ table lets you configure up to 64 DiffServ-to-VLAN Priority mapping (Layer 2 class of service). For each packet sent to the LAN, the VLAN Priority of the packet is set according to the DiffServ value in the IP header of the packet.

The mapping of an application to its CoS and traffic type is shown in the table below:

Table 12-14: Traffic/Network Types and Priority

Application	Traffic / Network Types	Class-of-Service (Priority)
Debugging interface	Management	Bronze
Telnet	Management	Bronze
DHCP	Management	Network
Web server (HTTP)	Management	Bronze
SNMP GET/SET	Management	Bronze
Web server (HTTPS)	Management	Bronze
RTP traffic	Media	Premium media
RTCP traffic	Media	Premium media
T.38 traffic	Media	Premium media
SIP	Control	Premium control
SIP over TLS (SIPS)	Control	Premium control
Syslog	Management	Bronze
SNMP Traps	Management	Bronze

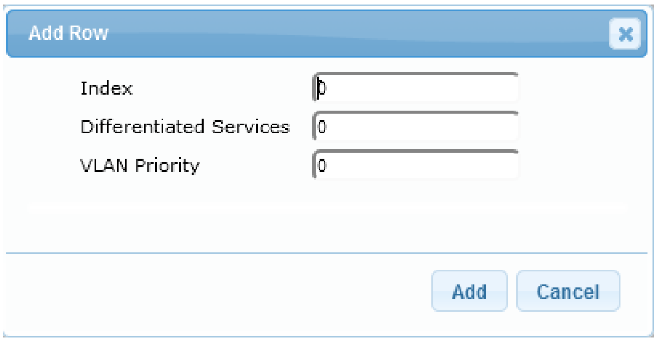
Application	Traffic / Network Types	Class-of-Service (Priority)
DNS client	Varies according to DNS settings: ▪ OAMP ▪ Control	Depends on traffic type: ▪ Control: Premium Control ▪ Management: Bronze
NTP	Varies according to the interface type associated with NTP (see "Assigning NTP Services to Application Types" on page 137): ▪ OAMP ▪ Control	Depends on traffic type: ▪ Control: Premium control ▪ Management: Bronze

The following procedure describes how to configure DiffServ-to-VLAN priority mapping through the Web interface. You can also configure it through ini file (DiffServToVlanPriority) or CLI (configure voip > qos vlan-mapping).

➢ **To configure QoS:**

1. Open the Diff Serv table (**Configuration** tab > **VoIP** menu > **Network** > **QoS Settings**).
2. Configure DiffServ-to-VLAN priority mapping (Layer-2 QoS):
   a. Click Add; the following dialog box appears:

Figure 12-3: DiffServ Table Page - Add Row Dialog Box



Add Row

Index: 0
Differentiated Services: 0
VLAN Priority: 0

   b. Configure a DiffServ-to-VLAN priority mapping (Layer-2 QoS) according to the parameters described in the table below.
   c. Click Add, and then save ("burn") your settings to flash memory.

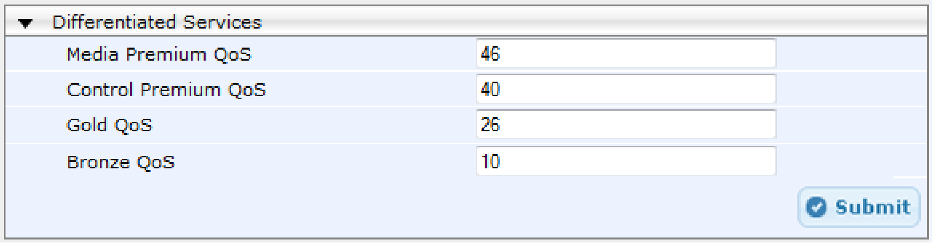
Table 12-15: DiffServ Table Parameter Descriptions

Parameter	Description
Index	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Differentiated Services `diff-serv` [DiffServToVlanPriority_DiffServ]	Defines a DiffServ value. The valid value is 0 to 63.
VLAN Priority `vlan-priority` [DiffServToVlanPriority_VlanPriority]	Defines the VLAN priority level. The valid value is 0 to 7.

3. Under the Differentiated Services group, configure DiffServ (Layer-3 QoS) values per

CoS.

Figure 12-4: QoS Settings Page - Differentiated Services



Differentiated Services

Media Premium QoS	46
Control Premium QoS	40
Gold QoS	26
Bronze QoS	10

12.7 Configuring ICMP Messages

Internet Control Message Protocol (ICMP) is one of the core protocols of the Internet Protocol suite. It is used by network devices such as routers to send error messages indicating, for example, that a requested service is unavailable.

You can configure the device to handle ICMP messages as follows:

- Send and receive ICMP Redirect messages.
- Send ICMP Destination Unreachable messages. The device sends this message in response to a packet that cannot be delivered to its destination for reasons other than congestion. The device sends a Destination Unreachable message upon any of the following:
  - Address unreachable
  - Port unreachable

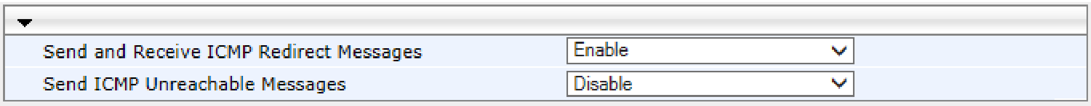
  This feature is applicable to IPv4 and IPv6 addressing schemes.

The following procedure describes how to configure ICMP messaging through the Web interface. You can also configure it through ini file - DisableICMPUnreachable (ICMP Unreachable messages) and DisableICMPRedirects (ICMP Redirect messages).

➢ **To configure handling of ICMP messages:**

1. Open the Network Settings page (**Configuration** tab > **VoIP** menu > **Network** > **Network Settings**).

Figure 12-5: Configuring ICMP Messaging in Network Settings Page



Send and Receive ICMP Redirect Messages	Enable
Send ICMP Unreachable Messages	Disable

2. To enable or disable sending and receipt of ICMP Redirect messages, use the 'Send and Received ICMP Redirect Messages' parameter.
3. To enable or disable the sending of ICMP Destination Unreachable messages, use the 'Send ICMP Unreachable Messages' parameter.
4. Click **Submit**.

12.8 DNS

You can use the device's embedded domain name server (DNS) or an external, third-party DNS to translate domain names into IP addresses. This is useful if domain names are used as the destination in call routing. The device supports the configuration of the following DNS types:

- Internal DNS table - see "Configuring the Internal DNS Table" on page 147
- Internal SRV table - see "Configuring the Internal SRV Table" on page 148

12.8.1 Configuring the Internal DNS Table

The Internal DNS table, similar to a DNS resolution, translates up to 20 host (domain) names into IP addresses. This functionality can be used when a domain name (FQDN) is configured as an IP destination in a routing rule. Up to three different IP addresses can be assigned to the same host name. This is typically used for alternative Tel-to-IP call routing.

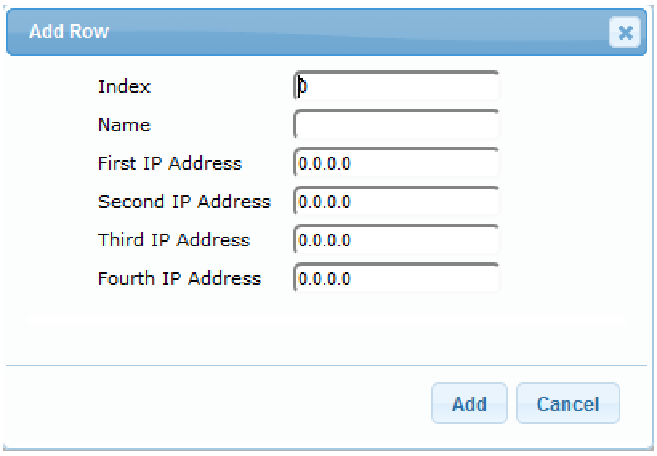
Note: The device initially attempts to resolve a domain name using the Internal DNS table. If the domain name is not configured in the table, the device performs a DNS resolution using an external DNS server for the related IP network interface (see "Configuring IP Network Interfaces" on page 133).

The following procedure describes how to configure the DNS table through the Web interface. You can also configure it through ini file (DNS2IP) or CLI (configure voip > voip-network dns dns-to-ip).

➢ **To configure the internal DNS table:**

1. Open the Internal DNS table (**Configuration** tab > **VoIP** menu > **Network** > **DNS** > **Internal DNS Table**).
2. Click **Add**; the following dialog box appears:

Figure 12-6: Internal DNS Table - Add Row Dialog Box



Add Row

Index	0
Name	
First IP Address	0.0.0.0
Second IP Address	0.0.0.0
Third IP Address	0.0.0.0
Fourth IP Address	0.0.0.0

3. Configure the DNS rule, as required. For a description of the parameters, see the table below.
4. Click **Add**; the DNS rule is added to the table.

Table 12-16: Internal DNS Table Parameter Description

Parameter	Description
Domain Name `domain-name` [Dns2Ip_DomainName]	Defines the host name to be translated. The valid value is a string of up to 31 characters.
First IP Address `first-ip-address` [Dns2Ip_FirstIpAddress]	Defines the first IP address (in dotted-decimal format notation) to which the host name is translated. The IP address can be configured as an IPv4 and/or IPv6 address.
Second IP Address `second-ip-address` [Dns2Ip_SecondIpAddress]	Defines the second IP address (in dotted-decimal format notation) to which the host name is translated.
Third IP Address `third-ip-address` [Dns2Ip_ThirdIpAddress]	Defines the third IP address (in dotted-decimal format notation) to which the host name is translated.
Fourth IP Address `fourth-ip-address` [Dns2Ip_FourthIpAddress]	Defines the fourth IP address (in dotted-decimal format notation) to which the host name is translated. **Note:** Currently, this parameter is not supported.

12.8.2 Configuring the Internal SRV Table

The Internal SRV table resolves host names to DNS A-Records. Three different A-Records can be assigned to each host name, where each A-Record contains the host name, priority, weight, and port.

Note: If you configure the Internal SRV table, the device initially attempts to resolve a domain name using this table. If the domain is not configured in the table, the device performs a Service Record (SRV) resolution using an external DNS server, configured in the Interface table (see "Configuring IP Network Interfaces" on page 133).

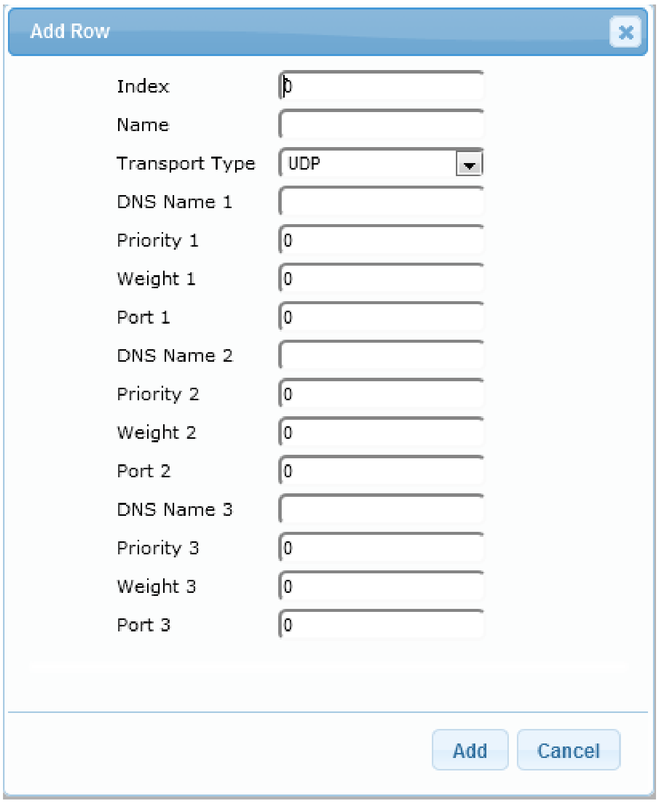
The following procedure describes how to configure the Internal SRV table through the Web interface. You can also configure it through ini file (SRV2IP) or CLI (configure voip > voip-network dns srv2ip).

➢ **To configure an SRV rule:**

1. Open the Internal SRV table (**Configuration** tab > **VoIP** menu > **Network** > **DNS** > **Internal SRV Table**).

2. Click **Add**; the following dialog box appears:

Figure 12-7: Internal SRV Table - Add Row Dialog Box



Add Row

Index	0
Name	
Transport Type	UDP
DNS Name 1	
Priority 1	0
Weight 1	0
Port 1	0
DNS Name 2	
Priority 2	0
Weight 2	0
Port 2	0
DNS Name 3	
Priority 3	0
Weight 3	0
Port 3	0

3. Configure an SRV rule according to the parameters described in the table below.
4. Click **Add**, and then save ("burn") your settings to flash memory.

Table 12-17: Internal SRV Table Parameter Descriptions

Parameter	Description
Domain Name `domain-name` [Srv2Ip_InternalDomain]	Defines the host name to be translated. The valid value is a string of up to 31 characters. By default, no value is defined.
Transport Type `transport-type` [Srv2Ip_TransportType]	Defines the transport type. ▪ **[0]** UDP (default) ▪ **[1]** TCP ▪ **[2]** TLS
DNS Name (1-3) `dns-name-1|2|3` [Srv2Ip_Dns1/2/3]	Defines the first, second or third DNS A-Record to which the host name is translated. By default, no value is defined.
Priority (1-3) `priority-1|2|3` [Srv2Ip_Priority1/2/3]	Defines the priority of the target host. A lower value means that it is more preferred. By default, no value is defined.
Weight (1-3) `weight-1|2|3` [Srv2Ip_Weight1/2/3]	Defines a relative weight for records with the same priority. By default, no value is defined.

Parameter	Description
Port (1-3) `port-1|2|3` [Srv2Ip_Port1/2/3]	Defines the TCP or UDP port on which the service is to be found. By default, no value is defined.

12.9 Network Address Translation Support

Network Address Translation (NAT) is a mechanism that maps internal IP addresses (and ports) used within a private network to global IP addresses and vice versa, providing transparent routing to end hosts. The primary advantages of NAT include (1) reduction in the number of global IP addresses required in a private network (global IP addresses are only used to connect to the Internet) and (2) better network security by hiding the internal architecture.

The design of SIP creates a problem for VoIP traffic to pass through NAT. SIP uses IP addresses and port numbers in its message body. However, the NAT server is unable to modify the SIP messages and thus, can't change local addresses to global addresses.

This section discusses the device's solutions for overcoming NAT traversal issues.

12.9.1 Device Located behind NAT

Two different streams traverse through NAT - signaling and media. A device located behind a NAT that initiates a signaling path has problems receiving incoming signaling responses as they are blocked by the NAT server. Therefore, the initiating device must inform the receiving device where to send the media. To resolve this NAT problem, the following solutions are provided by the device, listed in priority of the selected method used by the device:

a. If configured, uses the single Static NAT IP address for all interfaces - see "Configuring a Static NAT IP Address for All Interfaces" on page 151.

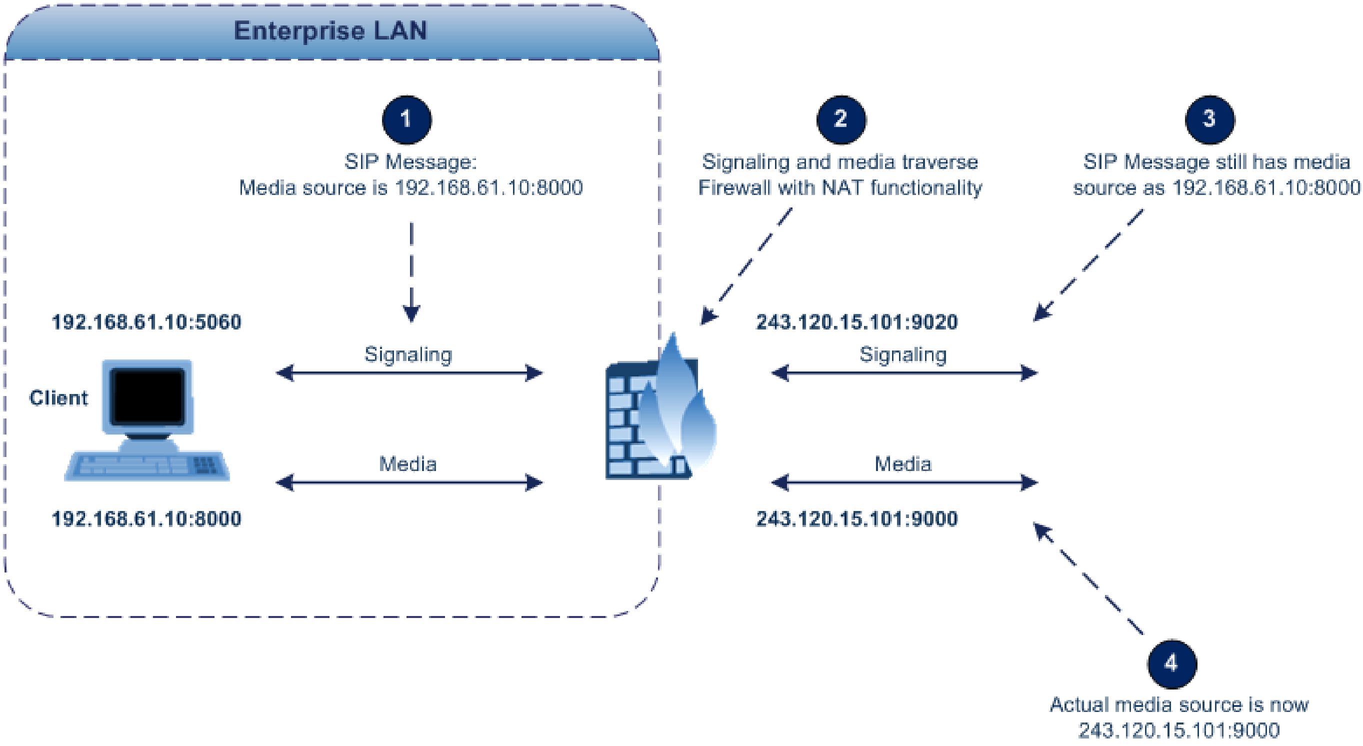
b. If configured, uses the NAT Translation table which configures NAT per interface - see Configuring NAT Translation per IP Interface on page 151.

If NAT is not configured by any of the above-mentioned methods, the device sends the packet according to its IP address configured in the Interface table.

Note: The priority list above is applicable only to the Gateway calls.

The figure below illustrates the NAT problem faced by the SIP networks where the device is located behind a NAT:

Figure 12-8: Device behind NAT and NAT Issues


Enterprise LAN
1
SIP Message:
Media source is 192.168.61.10:8000
2
Signaling and media traverse
Firewall with NAT functionality
3
SIP Message still has media
source as 192.168.61.10:8000
192.168.61.10:5060
Client
192.168.61.10:8000
Signaling
Media
243.120.15.101:9020
Signaling
Media
243.120.15.101:9000
4
Actual media source is now
243.120.15.101:9000

12.9.1.1 Configuring a Static NAT IP Address for All Interfaces

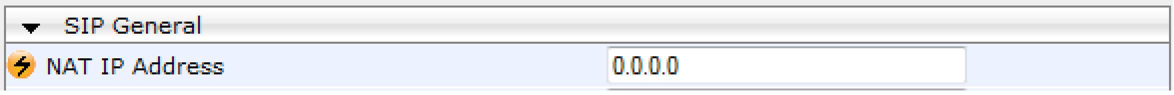
You can configure a global (public) IP address of the router to enable static NAT between the device and the Internet for all network interfaces. Thus, the device replaces the source IP address for media of all outgoing SIP messages sent on any of its network interfaces to this public IP address.

The following procedure describes how to configure a static NAT address through the Web interface. You can also configure it through ini file (StaticNATIP) or CLI (configure voip > sip-definition general-settings > nat-ip-addr).

➢ **To configure a single static NAT IP address:**

1. Open the SIP General Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**).

Figure 12-9: Configuring Static NAT IP Address in SIP General Parameters Page


SIP General

NAT IP Address: 0.0.0.0

2. In the 'NAT IP Address' field, enter the NAT IP address in dotted-decimal notation.
3. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

12.9.1.2 Configuring NAT Translation per IP Interface

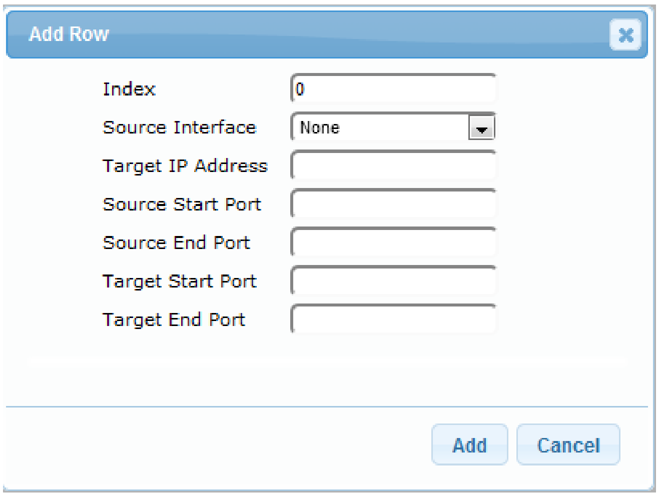
The NAT Translation table lets you configure up to 32 network address translation (NAT) rules for translating source IP addresses into NAT IP addresses (*global* - *public*) when the device is located behind NAT. The device's NAT traversal mechanism replaces the source IP address of SIP messages sent from a specific VoIP interface (Control and/or Media) in the IP Interfaces table to a public IP address. This allows, for example, the separation of VoIP traffic between different ITSP's, and topology hiding of internal IP addresses from the "public" network. Each IP network interface, configured in the Interface table, can be associated with a NAT rule, translating the source IP address and port of the outgoing packet into the NAT address (IP address and port range).

The following procedure describes how to configure NAT translation rules through the Web interface. You can also configure it through ini file (NATTranslation) or CLI (voip-network nattranslation).

➢ **To configure NAT translation rules:**

1. Open the NAT Translation table (**Configuration** tab > **VoIP** menu > **VoIP Network** > **NAT Translation Table**).
2. Click **Add**; the following dialog box appears:

Figure 12-10: NAT Translation Table - Add Row Dialog Box



Index: 0
Source Interface: None
Target IP Address:
Source Start Port:
Source End Port:
Target Start Port:
Target End Port:

3. Configure a NAT translation rule according to the parameters described in the table below.
4. Click **Add**, and then save ("burn") your settings to flash memory.

Table 12-18: NAT Translation Table Parameter Descriptions

Parameter	Description
Index `index` [NATTranslation_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Source Interface `src-interface-name` [NATTranslation_SrcIPInterfaceName]	Assigns an IP network interface to the rule. Outgoing packets sent from the specified network interface are NAT'ed. By default, no value is defined (**None**). For configuring IP network interfaces, see "Configuring IP Network Interfaces" on page 133.
Target IP Address `target-ip-address` [NATTranslation_TargetIPAddress]	Defines the global (public) IP address. The device adds the address to the SIP Via header, Contact header, 'o=' SDP field, and 'c=' SDP field, in the outgoing packet.

Parameter	Description
Source Start Port `src-start-port` [NATTranslation_SourceStartPort]	Defines the optional starting port range (0-65535) of the IP interface, used as matching criteria for the NAT rule. If not configured, the match is done on the entire port range. Only IP addresses and ports of matched source ports will be replaced.
Source End Port `src-end-port` [NATTranslation_SourceEndPort]	Defines the optional ending port range (0-65535) of the IP interface, used as matching criteria for the NAT rule. If not configured, the match is done on the entire port range. Only IP addresses and ports of matched source ports will be replaced.
Target Start Port `target-start-port` [NATTranslation_TargetStartPort]	Defines the optional starting port range (0-65535) of the global address. If not configured, the ports are not replaced. Matching source ports are replaced with the target ports. This address is set in the SIP Via and Contact headers, as well as in the o= and c= SDP fields.
Target End Port `target-end-port` [NATTranslation_TargetEndPort]	Defines the optional ending port range (0-65535) of the global address. If not configured, the ports are not replaced. Matching source ports are replaced with the target ports. This address is set in the SIP Via and Contact headers, as well as in the o= and c= SDP fields.

12.9.2 Remote UA behind NAT

12.9.2.1 SIP Signaling Messages

By default, the device resolves NAT issues for SIP signaling, using its NAT Detection mechanism. The NAT Detection mechanism checks whether the endpoint is located behind NAT, by comparing the incoming packet's source IP address with the SIP Contact header's IP address. If the packet's source IP address is a public address and the Contact header's IP address is a local address, the device considers the endpoint as located behind NAT. In this case, the device sends the SIP messages to the endpoint, using the packet's source IP address. Otherwise (or if you have disabled the NAT Detection mechanism), the device sends the SIP messages according to the SIP standard RFC 3261, where requests within the SIP dialog are sent using the IP address in the Contact header, and responses to INVITEs are sent using the IP address in the Via header. To enable or disable the device's NAT Detection mechanism, use the 'SIP NAT Detection' parameter.

If necessary, you can also configure the device to always consider incoming SIP INVITE messages as sent from endpoints that are located behind NAT. When this is enabled, the device sends responses to the INVITE (to the endpoint), using the the source IP address of the packet (INVITE) initially received from the endpoint. This is especially useful in scenarios where the endpoint is located behind a NAT firewall and the device (for whatever reason) is unable to identify NAT using its regular NAT Detection mechanism. This feature is enabled per specific calls using IP Groups. To configure this feature, use the 'Always Use Source Address' parameter in the IP Group table (see "Configuring IP Groups" on page 340). If this feature is disabled, the device's NAT detection is according to the settings of the global parameter, 'SIP NAT Detection' parameter.

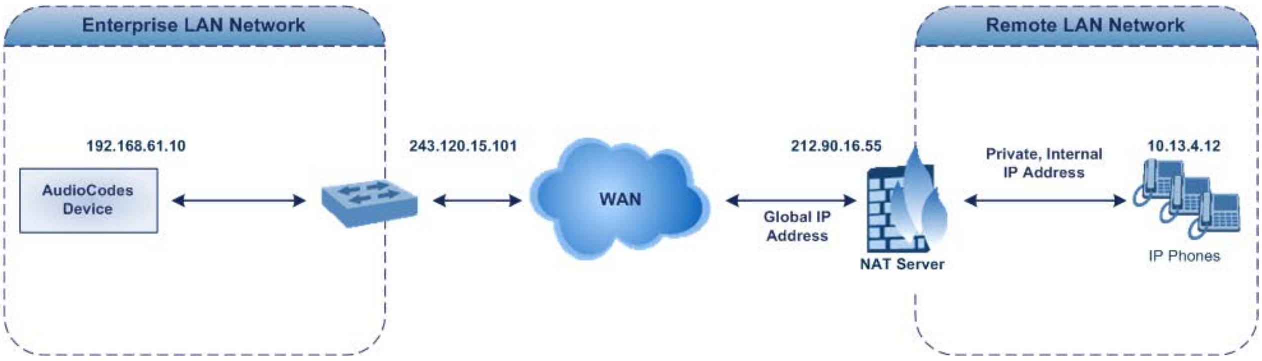
12.9.2.2 Media (RTP/RTCP/T.38)

When a remote UA initiates a call and is not located behind a NAT server, the device sends the RTP (or RTCP, T.38) packets to the remote UA using the IP address:port (UDP) indicated in the SDP body of the SIP message received from the UA. However, if the UA is located behind NAT, the device sends the RTP with the IP address of the UA (i.e., private IP address) as the destination, instead of that of the NAT server. Thus, the RTP will not reach the UA. To resolve this NAT traversal problem, the device offers the following features:

- First Incoming Packet Mechanism - see "First Incoming Packet Mechanism" on page 154
- RTP No-Op packets according to the avt-rtp-noop draft - see "No-Op Packets" on page 155

The figure below illustrates a typical network architecture where the remote UA is located behind NAT:

Figure 12-11: Remote UA behind NAT



12.9.2.2.1First Incoming Packet Mechanism

In scenarios where the remote user agent (UA) resides behind a NAT server, it's possible that the device, if not configured for NAT traversal, will send the media (RTP, RTCP and T.38) streams to an invalid IP address / UDP port (i.e., private IP address:port of UA and not the public address). When the UA is located behind a NAT, although the UA sends its private IP address:port in the original SIP message (INVITE), the device receives the subsequent media packets with a source address of a public IP address:port (i.e., allocated by the NAT server). Therefore, to ensure that the media reaches the UA, the device must send it to the public address.

The device identifies whether the UA is located behind NAT, by comparing the source IP address of the first received media packet, with the IP address and UDP port of the first received SIP message (INVITE) when the SIP session was started. This is done for each media type--RTP, RTCP and T.38--and therefore, they can have different destination IP addresses and UDP ports than one another.

You can configure the device's NAT feature to operate in one of the following modes:

- [0] Enable NAT Only if Necessary: NAT traversal is performed only if the UA is located behind NAT:
    - UA behind NAT: The device sends the media packets to the IP address:port obtained from the source address of the first media packet received from the UA.
    - UA not behind NAT: The device sends the packets to the IP address:port specified in the SDP 'c=' line (Connection) of the first received SIP message.

  Note: If the SIP session is established (ACK) and the device (not the UA) sends the first packet, it sends it to the address obtained from the SIP message and only after the device receives the first packet from the UA does it determine whether the UA is behind NAT.
- [1] Disable NAT: (Default) The device considers the UA as not located behind NAT and sends media packets to the UA using the IP address:port specified in the SDP 'c=' line (Connection) of the first received SIP message.
- [2] Force NAT: The device always considers the UA as behind NAT and sends the media packets to the IP address:port obtained from the source address of the first media packet received from the UA. The device only sends packets to the UA after it

receives the first packet from the UA (to obtain the IP address).

- [3] NAT by Signaling = The device identifies whether or not the UA is located behind NAT based on the SIP signaling. The device assumes that if signaling is behind NAT that the media is also behind NAT, and vice versa. If located behind NAT, the device sends media as described in option [2] Force NAT; if not behind NAT, the device sends media as described in option [1] Disable NAT. This option is applicable only to SBC calls. If the parameter is configured to this option, Gateway calls use option [0] Enable NAT Option, by default.

➢ **To enable NAT resolution using the First Incoming Packet mechanism:**

1. Open the General Settings page (**Configuration** tab > **VoIP** menu > **Media** > **General Media Settings**).
2. Set the 'NAT Mode' parameter (NATMode).
3. Click **Submit**.

12.9.2.2.2No-Op Packets

The device's No-Op packet support can be used to verify Real-Time Transport Protocol (RTP) and T.38 connectivity, and to keep NAT bindings and Firewall pinholes open. The No-Op packets are available for sending in RTP and T.38 formats.

You can control the activation of No-Op packets by using the *ini* file parameter NoOpEnable. If No-Op packet transmission is activated, you can control the time interval in which No-Op packets are sent in the case of silence (i.e., no RTP or T.38 traffic). This is done using the *ini* file parameter NoOpInterval.

- **RTP No-Op:** The RTP No-Op support complies with IETF Internet-Draft draft-wing-avt-rtp-noop-03 ("A No-Op Payload Format for RTP"). This IETF document defines a No-Op payload format for RTP. The draft defines the RTP payload type as dynamic. You can control the payload type with which the No-Op packets are sent. This is performed using the RTPNoOpPayloadType *ini* parameter. The default payload type is 120.
- **T.38 No-Op:** T.38 No-Op packets are sent only while a T.38 session is activated. Sent packets are a duplication of the previously sent frame (including duplication of the sequence number).

Note:

- The No-OP Packet feature requires DSP resources.
- Receipt of No-Op packets is always supported.

12.9.2.2.3Fax Transmission behind NAT

The device supports transmission from fax machines (connected to the device) located inside (behind) a Network Address Translation (NAT). Generally, the firewall blocks T.38 (and other) packets received from the WAN, unless the device behind the NAT sends at least one IP packet from the LAN to the WAN through the firewall. If the firewall blocks T.38 packets sent from the termination IP fax, the fax fails.

To overcome this, the device sends No-Op (“no-signal”) packets to open a pinhole in the NAT for the answering fax machine. The originating fax does not wait for an answer, but immediately starts sending T.38 packets to the terminating fax machine upon receipt of a re-INVITE with T.38 only in the SDP, or T.38 and audio media in the SDP. This feature is configured using the T38FaxSessionImmediateStart parameter. The No-Op packets are enabled using the NoOpEnable and NoOpInterval parameters.

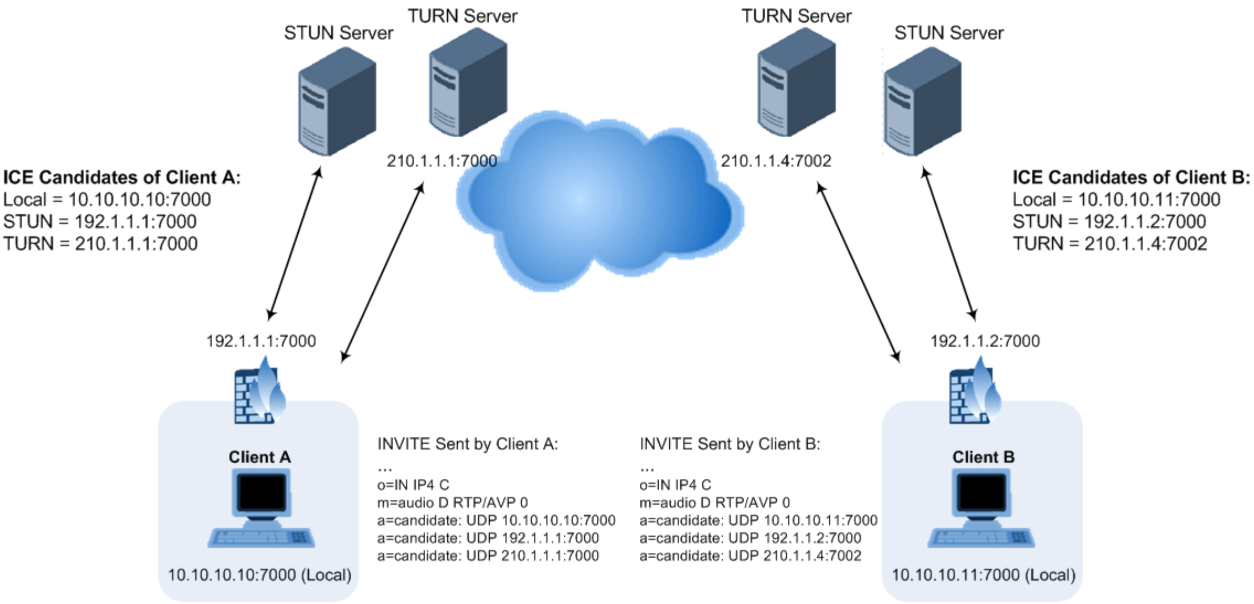
12.9.2.2.4ICE Lite

The device supports Interactive Connectivity Establishment (ICE) Lite for SBC calls. ICE is a methodology for NAT traversal, enabling VoIP interoperability across networks to work better across NATs and firewalls. It employs Session Traversal Utilities for NAT (STUN) and Traversal Using Relays around NAT (TURN) protocols to provide a peer with a public IP address and port that can be used to connect to a remote peer.

In order for clients behind NATs and/or firewalls to send media (RTP) between one another, they need to discover each others P address and port as seen by the "outside" world. If both peers are located in different private networks behind a NAT, the peers must coordinate to determine the best communication path between them.

ICE first tries to make a connection using the client's private local address. If that fails (which it will for clients behind NAT), ICE obtains an external (public) address using a STUN server. If that fails, traffic is routed through a TURN relay server (which has a public address).

These addresses:ports (local, STUN, TURN and any other network address) of the client are termed "candidates". Each client sends its' candidates to the other in the SDP body of the INVITE message. Peers then perform connectivity checks per candidate of the other peer, using STUN binding requests sent on the RTP and RTCP ports. ICE tries each candidate and selects the one that works (i.e., media can flow between the clients). The following figure shows a simple illustration of ICE:



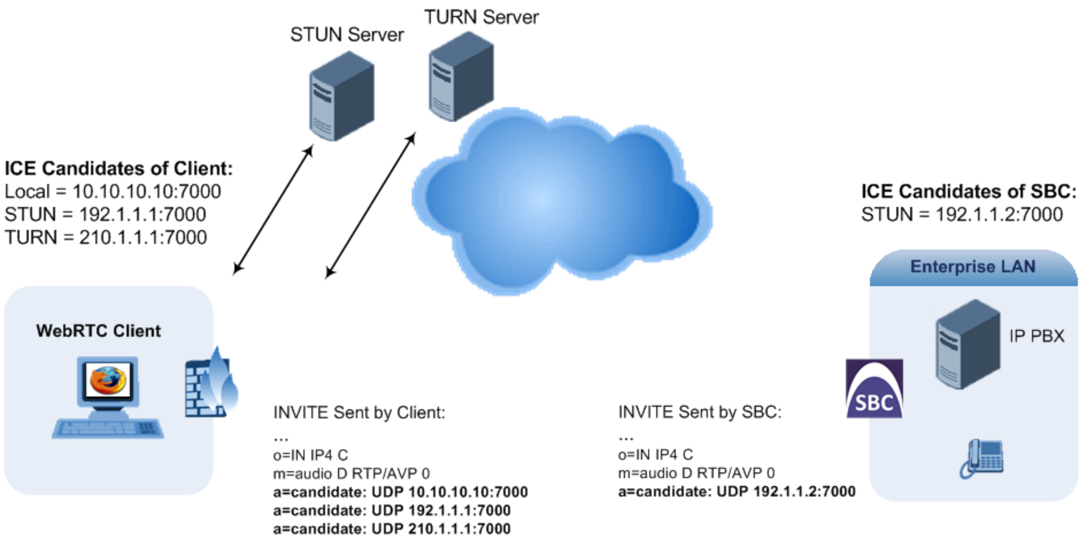
INVITE Sent by Client A:

```
...
o=IN IP4 C
m=audio D RTP/AVP 0
a=candidate: UDP 10.10.10.10:7000
a=candidate: UDP 192.1.1.1:7000
a=candidate: UDP 210.1.1.1:7000
```

INVITE Sent by Client B:

```
...
o=IN IP4 C
m=audio D RTP/AVP 0
a=candidate: UDP 10.10.10.11:7000
a=candidate: UDP 192.1.1.2:7000
a=candidate: UDP 210.1.1.4:7002
```

The device's support for ICE-Lite means that it does not initiate the ICE process. Instead, it supports remote endpoints that initiate ICE to discover their workable public IP address with the device. Therefore, the device supports the receipt of STUN binding requests for connectivity checks of ICE candidates and responds to them with STUN responses. Note that in the response to the INVITE message received from the remote endpoint, the device sends only a single candidate for its' own IP address. This is the IP address of the device that the client uses. To support ICE, the SBC leg interfacing with the ICE-enabled client (SIP entity) must be enabled for ICE. This is done using the IP Profile parameter, IPProfile_SBCIceMode (see "Configuring IP Profiles" on page 387).



ICE Candidates of Client:
Local = 10.10.10.10:7000
STUN = 192.1.1.1:7000
TURN = 210.1.1.1:7000

ICE Candidates of SBC:
STUN = 192.1.1.2:7000

INVITE Sent by Client:

```
...
o=IN IP4 C
m=audio D RTP/AVP 0
a=candidate: UDP 10.10.10.10:7000
a=candidate: UDP 192.1.1.1:7000
a=candidate: UDP 210.1.1.1:7000
```

INVITE Sent by SBC:

```
...
o=IN IP4 C
m=audio D RTP/AVP 0
a=candidate: UDP 192.1.1.2:7000
```

12.10 Robust Receipt of Media Streams by Media Latching

The Robust Media mechanism (or media latching) is an AudioCodes proprietary mechanism to filter out unwanted media (RTP, RTCP, SRTP, SRTCP, and T.38) streams that are sent to the same port number of the device. Media ports may receive additional multiple unwanted media streams (from multiple sources of traffic) as result of traces of previous calls, call control errors, or deliberate malicious attacks (e.g., Denial of Service). When the device receives more than one media stream on the same port, the Robust Media mechanism detects the valid media stream and ignores the rest. Thus, this can prevent an established call been stolen by a malicious attacker on the media stream.

For the involved voice channel, the device latches onto the first stream of the first received packet. All packets (of any media type) received from the same IP address and SSRC are accepted (for T.38 packets, the device considers only the IP address). If the channel receives subsequent packets from a non-latched source, the device can either ignore this new stream and remain latched to the first original stream (IP address:port), or it can latch onto this new stream. The media latch mode is configured using the InboundMediaLatchMode parameter. If this mode is configured to latch onto new streams, you also need to configure the following:

- Minimum number of continuous media packets that need to be received from a different source(s) before the channel can latch onto this new incoming stream.
- Period (msec) during which if no packets are received from the current stream, the channel latches onto the next packet received from any other stream.

Depending on media latch mode, if the device has latched onto a new stream and a packet from the original (first latched onto) IP address:port is received at any time, the device latches onto this original stream.

Latching onto a new T.38 stream is reported in CDR using the CDR fields, LatchedT38Ip (new IP address) and LatchedT38Port (new port). In addition, the SIP PUBLISH message updates the latched RTP SSRC, for example:

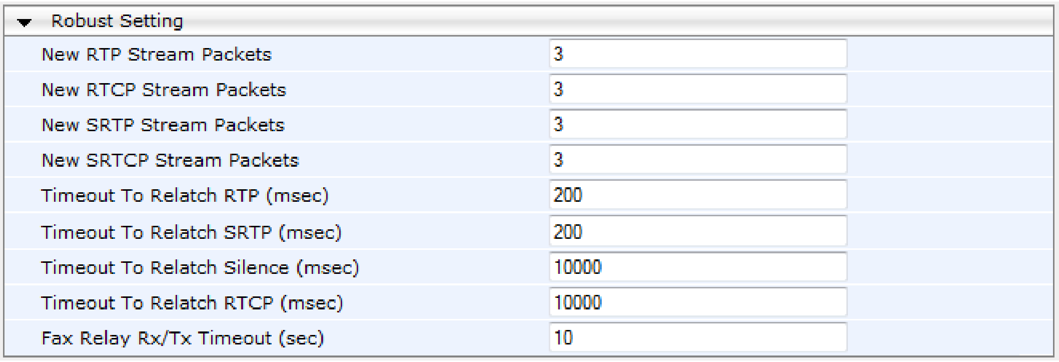
```
RemoteAddr: IP=10.33.2.55 Port=4000 SSRC=0x66d510ec
```

➢ To configure media latching:

1. Define the Robust Media method, using the InboundMediaLatchMode ini file parameter.
2. Open the General Settings page (Configuration tab > VoIP menu > Media > General

Media Settings).

Figure 12-12: General Settings Page - Robust Setting



▼ Robust Setting

New RTP Stream Packets	3
New RTCP Stream Packets	3
New SRTP Stream Packets	3
New SRTCP Stream Packets	3
Timeout To Relatch RTP (msec)	200
Timeout To Relatch SRTP (msec)	200
Timeout To Relatch Silence (msec)	10000
Timeout To Relatch RTCP (msec)	10000
Fax Relay Rx/Tx Timeout (sec)	10

3. If you have set the InboundMediaLatchMode parameter to 1 or 2, scroll down to the Robust Settings group and do the following:
   - Define the minimum number of continuous media (RTP, RTCP, SRTP, and SRTCP) packets that need to be received by the channel before it can latch onto this new incoming stream:
     - 'New RTP Stream Packets'
     - 'New RTCP Stream Packets'
     - 'New SRTP Stream Packets'
     - 'New SRTCP Stream Packets'
   - Define a period (msec) during which if no packets are received from the current media session, the channel can re-latch onto another stream:
     - 'Timeout To Relatch RTP'
     - 'Timeout To Relatch SRTP'
     - 'Timeout To Relatch Silence'
     - 'Timeout To Relatch RTCP'
     - 'Fax Relay Rx/Tx Timeout'
4. Click Submit, and then save ("burn") your settings to flash memory.

For a detailed description of the robust media parameters, see "General Security Parameters" on page 849.

12.11 Multiple Routers Support

Multiple routers support is designed to assist the device when it operates in a multiple routers network. The device learns the network topology by responding to Internet Control Message Protocol (ICMP) redirections and caches them as routing rules (with expiration time).

When a set of routers operating within the same subnet serve as devices to that network and intercommunicate using a dynamic routing protocol, the routers can determine the shortest path to a certain destination and signal the remote host the existence of the better route. Using multiple router support, the device can utilize these router messages to change its next hop and establish the best path.

Note: Multiple Routers support is an integral feature that doesn't require configuration.

13 Security

This section describes the VoIP security-related configuration.

13.1 Configuring Firewall Settings

The Firewall Settings table lets you configure the device's Firewall, which defines network traffic filtering rules (*access list*) for incoming traffic. You can add up to 50 firewall rules. The access list offers the following firewall possibilities:

- Block traffic from known malicious sources
- Allow traffic only from known "friendly" sources, and block all other traffic
- Mix allowed and blocked network sources
- Limit traffic to a user-defined rate (blocking the excess)
- Limit traffic to specific protocols, and specific port ranges on the device

For each packet received on the network interface, the table is scanned from top to bottom until the first matching rule is found. This rule can either permit (*allow*) or deny (*block*) the packet. Once a rule in the table is located, subsequent rules further down the table are ignored. If the end of the table is reached without a match, the packet is accepted.

Notes:

- This firewall applies to a very low-level network layer and overrides all your other security-related configuration. Thus, if you have configured higher-level security features (e.g., on the Application level), you must also configure firewall rules to permit this necessary traffic. For example, if you have configured IP addresses to access the Web and Telnet interfaces in the Web Access List (see ''Configuring Web and Telnet Access List'' on page 79), you must configure a firewall rule that permits traffic from these IP addresses.
- Only users with Security Administrator or Master access levels can configure firewall rules.
- Setting the 'Prefix Length' field to **0** means that the rule applies to **all** packets, regardless of the defined IP address in the 'Source IP' field. Thus, it is highly recommended to set the parameter to a value other than 0.
- It is recommended to add a rule at the end of your table that blocks all traffic and to add firewall rules above it that allow required traffic (with bandwidth limitations). To block all traffic, use the following firewall rule:
  - √ Source IP: 0.0.0.0
  - √ Prefix Length: 0 (i.e., rule matches all IP addresses)
  - √ Start Port - End Port: 0-65535
  - √ Protocol: **Any**
  - √ Action Upon Match: **Block**

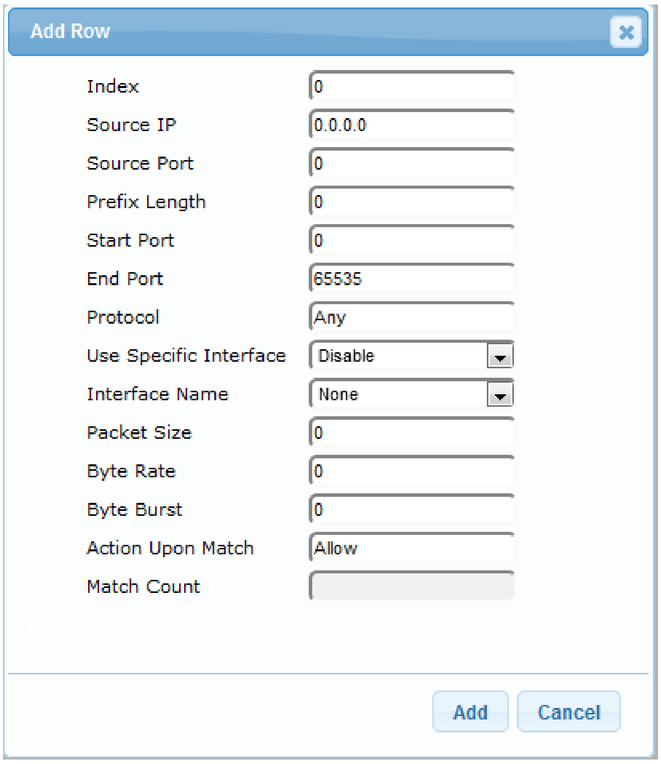
The following procedure describes how to configure Firewall rules through the Web interface. You can also configure it through ini file (AccessList) or CLI (configure voip > access-list).

➢ To configure a Firewall rule:

1. Open the Firewall Settings page (**Configuration** tab > **VoIP** menu > **Security** > **Firewall Settings**).

2. Click **Add**; the following dialog box appears:

Figure 13-1: Firewall Settings Table - Add Row Dialog Box



Add Row

Index	0
Source IP	0.0.0.0
Source Port	0
Prefix Length	0
Start Port	0
End Port	65535
Protocol	Any
Use Specific Interface	Disable
Interface Name	None
Packet Size	0
Byte Rate	0
Byte Burst	0
Action Upon Match	Allow
Match Count	

3. Configure a Firewall rule according to the parameters described in the table below.
4. Click **Add**, and then reset the device with a burn-to-flash for your settings to take effect.

Table 13-1: Firewall Settings Table Parameter Descriptions

Parameter	Description
Index	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Source IP `source-ip` [AccessList_Source_IP]	Defines the IP address (or DNS name) or a specific host name of the source network from where the device receives the incoming packet. The default is 0.0.0.0.
Source Port `src-port` [AccessList_Source_Port]	Defines the source UDP/TCP ports of the remote host from where the device receives the incoming packet. The valid range is 0 to 65535. The default is 0. **Note:** When set to 0, this field is ignored and any source port matches the rule.

Parameter	Description
Prefix Length `prefixLen` [AccessList_PrefixLen]	(**Mandatory**) Defines the IP network mask - 32 for a single host or the appropriate value for the source IP addresses. ▪ A value of 8 corresponds to IPv4 subnet class A (network mask of 255.0.0.0). ▪ A value of 16 corresponds to IPv4 subnet class B (network mask of 255.255.0.0). ▪ A value of 24 corresponds to IPv4 subnet class C (network mask of 255.255.255.0). The IP address of the sender of the incoming packet is trimmed in accordance with the prefix length (in bits) and then compared to the parameter 'Source IP'. The default is 0 (i.e., applies to all packets). You **must** change this value to any of the above options. **Note:** A value of 0 applies to **all** packets, regardless of the defined IP address. Therefore, you must set the parameter to a value other than 0.
Start Port `start-port` [AccessList_Start_Port]	Defines the first UDP/TCP port in the range of ports on the device on which the incoming packet is received. From the perspective of the remote IP entity, this is the destination port. To configure the last port in the range, see the 'End Port' parameter (below). The valid range is 0 to 65535. **Note:** When the protocol type isn't TCP or UDP, the entire range must be provided.
End Port `end-port` [AccessList_End_Port]	Defines the last UDP/TCP port in the range of ports on the device on which the incoming packet is received. From the perspective of the remote IP entity, this is the destination port. To configure the first port in the range, see the 'Start Port' parameter (above). The valid range is 0 to 65535 (default). **Note:** When the protocol type isn't TCP or UDP, the entire range must be provided.
Protocol `protocol` [AccessList_Protocol]	Defines the protocol type (e.g., **UDP**, **TCP**, **ICMP**, **ESP** or **Any**) or the IANA protocol number in the range of 0 (Any) to 255. The default is **Any**. **Note:** The parameter also accepts the abbreviated strings "SIP" and "HTTP". Specifying these strings implies selection of the TCP or UDP protocols and the appropriate port numbers as defined on the device.
Use Specific Interface `use-specific-interface` [AccessList_Use_Specific_Interface]	Determines whether you want to apply the rule to a specific network interface defined in the Interface table (i.e., packets received from that defined in the Source IP field and received on this network interface): ▪ **[0]** Disable (default) ▪ **[1]** Enable **Notes:** ▪ If enabled, then in the 'Interface Name' field (described below), select the interface to which the rule is applied. ▪ If disabled, then the rule applies to all interfaces.

Parameter	Description
Interface Name `network-interface-name` [AccessList_Interface_x]	Defines the network interface to which you want to apply the rule. This is applicable if you enabled the 'Use Specific Interface' field. The list displays interface names as defined in the Interface table in "Configuring IP Network Interfaces" on page 133.
Packet Size `packet-size` [AccessList_Packet_Size]	Defines the maximum allowed packet size. The valid range is 0 to 65535. **Note:** When filtering fragmented IP packets, this field relates to the overall (re-assembled) packet size, and not to the size of each fragment.
Byte Rate `byte-rate` [AccessList_Byte_Rate]	Defines the expected traffic rate (bytes per second), i.e., the allowed bandwidth for the specified protocol. In addition to this field, the 'Burst Bytes' field provides additional allowance such that momentary bursts of data may utilize more than the defined byte rate, without being interrupted. For example, if 'Byte Rate' is set to 40000 and 'Burst Bytes' to 50000, then this implies the following: the allowed bandwidth is 40000 bytes/sec with extra allowance of 50000 bytes; if, for example, the actual traffic rate is 45000 bytes/sec, then this allowance would be consumed within 10 seconds, after which all traffic exceeding the allocated 40000 bytes/sec is dropped. If the actual traffic rate then slowed to 30000 bytes/sec, then the allowance would be replenished within 5 seconds.
Burst Bytes `byte-burst` [AccessList_Byte_Burst]	Defines the tolerance of traffic rate limit (number of bytes). The default is 0.
Action Upon Match `allow-type` [AccessList_Allow_Type]	Defines the firewall action to be performed upon rule match. ▪ "Allow" = (Default) Permits these packets ▪ "Block" = Rejects these packets
Match Count [AccessList_MatchCount]	(Read-only) Displays the number of packets accepted or rejected by the rule.

The table below provides an example of configured firewall rules:

Table 13-2: Configuration Example of Firewall Rules

Parameter	Firewall Rule				
	1	2	3	4	5
Source IP	12.194.231.76	12.194.230.7	0.0.0.0	192.0.0.0	0.0.0.0
Prefix Length	16	16	0	8	0
Start Port and End Port	0-65535	0-65535	0-65535	0-65535	0-65535
Protocol	Any	Any	icmp	Any	Any
Use Specific Interface	Enable	Enable	Disable	Enable	Disable
Interface Name	WAN	WAN	None	Voice-Lan	None

Parameter	Firewall Rule				
	1	2	3	4	5
Byte Rate	0	0	40000	40000	0
Burst Bytes	0	0	50000	50000	0
Action Upon Match	Allow	Allow	Allow	Allow	Block

The firewall rules in the above configuration example do the following:

- **Rules 1 and 2:** Typical firewall rules that allow packets ONLY from specified IP addresses (e.g., proxy servers). Note that the prefix length is configured.
- **Rule 3:** A more "advanced” firewall rule - bandwidth rule for ICMP, which allows a maximum bandwidth of 40,000 bytes/sec with an additional allowance of 50,000 bytes. If, for example, the actual traffic rate is 45,000 bytes/sec, then this allowance would be consumed within 10 seconds, after which all traffic exceeding the allocated 40,000 bytes/sec is dropped. If the actual traffic rate then slowed to 30,000 bytes/sec, the allowance would be replenished within 5 seconds.
- **Rule 4:** Allows traffic from the LAN voice interface and limits bandwidth.
- **Rule 5:** Blocks all other traffic.

13.2 Configuring General Security Settings

The device uses TLS over TCP to encrypt and optionally, authenticate SIP messages. This is referred to as Secure SIP (SIPS). SIPS uses the X.509 certificate exchange process, as described in "Configuring SSL/TLS Certificates" on page 107, where you need to configure certificates (TLS Context).

Note: When a TLS connection with the device is initiated by a SIP client, the device also responds using TLS, regardless of whether or not TLS was configured.

➢ **To configure SIPS:**

1. Configure a TLS Context as required.
2. Assign the TLS Context to a Proxy Set or SIP Interface (see "Configuring Proxy Sets" on page 352 and "Configuring SIP Interfaces" on page 333, respectively).
3. Configure a SIP Interface with a TLS port number.
4. Configure various SIPS parameters in the General Security Settings page (**Configuration** tab > **VoIP** menu > **Security** > **General Security Settings**).

   For a description of the TLS parameters, see "TLS Parameters" on page 855.
5. By default, the device initiates a TLS connection only for the next network hop. To enable TLS all the way to the destination (over multiple hops), set the 'Enable SIPS' (EnableSIPS) parameter to **Enable** in the SIP General Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**).

13.3 Intrusion Detection System

The device's Intrusion Detection System (IDS) feature detects malicious attacks on the device and reacts accordingly. A remote host is considered malicious if it has reached or exceeded a user-defined threshold (counter) of specified malicious attacks.

If malicious activity is detected, the device can do the following:

- Block (blacklist) remote hosts (IP addresses / ports) considered by the device as malicious. The device automatically blacklists the malicious source for a user-defined period after which it is removed from the blacklist.
- Send SNMP traps to notify of malicious activity and/or whether an attacker has been added to or removed from the blacklist. For more information, see "Viewing IDS Alarms" on page 170.

The Intrusion Detection System (IDS) is an important feature for Enterprises to ensure legitimate calls are not being adversely affected by attacks and to prevent Theft of Service and unauthorized access.

There are many types of malicious attacks, the most common being:

- **Denial of service:** This can be Denial of Service (DoS) where an attacker wishing to prevent a server from functioning correctly directs a large amount of requests – sometimes meaningless and sometimes legitimate, or it can be Distributed Denial of Service (DDoS) where the attacker controls a large group of systems to coordinate a large scale DoS attack against a system:
    - Message payload tampering: Attacker may inject harmful content into a message, e.g., by entering meaningless or wrong information, with the goal of exploiting a buffer overflow at the target. Such messages can be used to probe for vulnerabilities at the target.
    - Message flow tampering: This is a special case of DoS attacks. These attacks disturb the ongoing communication between users. An attacker can then target the connection by injecting fake signaling messages into the communication channel (such as CANCEL messages).
    - Message Flooding: The most common DoS attack is where an attacker sends a huge amount of messages (e.g., INVITEs) to a target. The goal is to overwhelm the target's processing capabilities, thereby rendering the target inoperable.
- **SPAM over Internet Telephony (SPIT):** VoIP spam is unwanted, automatically dialed, pre-recorded phone calls using VoIP. It is similar to e-mail spam.
- **Theft of Service (ToS):** Service theft can be exemplified by phreaking, which is a type of hacking that steals service (i.e., free calls) from a service provider, or uses a service while passing the cost to another person.

The IDS configuration is based on IDS Policies, where each policy can be configured with a set of IDS rules. Each rule defines a type of malicious attack to detect and the number of attacks during an interval (threshold) before an SNMP trap is sent. Each policy is then applied to a target under attack (SIP interface) and/or source of attack (Proxy Set and/or subnet address).

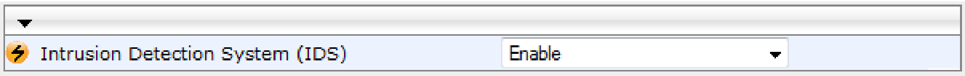
13.3.1 Enabling IDS

The following procedure describes how to enable IDS.

➢ **To enable IDS:**

1. Open the IDS Global Parameters page (**Configuration** tab > **VoIP** menu > **Security** > **Intrusion Detection and Prevention** > **Global Parameters**).

Figure 13-2: Enabling IDS on IDS Global Parameters Page



Intrusion Detection System (IDS) Enable

2. From the 'Intrusion Detection System' drop-down list, select **Enable**.
3. Click **Submit**, and then reset the device with a burn-to-flash for the setting to take effect.

13.3.2 Configuring IDS Policies

Configuring IDS Policies is a two-stage process that includes the following tables:

1. **IDS Policy (parent table):** Defines a name and description for the IDS Policy. You can configure up to 20 IDS Policies.
2. **IDS Rules table (child table):** Defines the actual rules for the IDS Policy. Each IDS Policy can be configured with up to 20 rules.

Note: A maximum of 100 IDS rules can be configured (regardless of how many rules are assigned to each policy).

The device provides the following pre-configured IDS Policies that can be used in your deployment (if they meet your requirements):

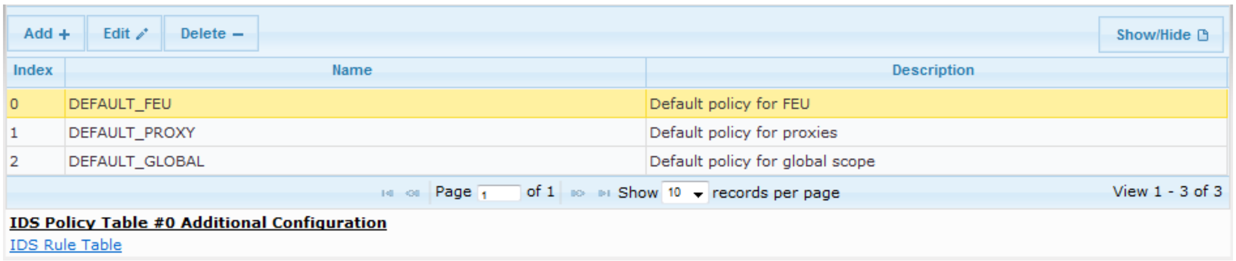
- "DEFAULT_FEU": IDS Policy for far-end users in the WAN
- "DEFAULT_PROXY": IDS Policy for proxy server
- "DEFAULT_GLOBAL": IDS Policy with global thresholds

These default IDS Policies are read-only and cannot be modified.

➢ To configure an IDS Policy:

1. Open the IDS Policy table (**Configuration** tab > **VoIP** menu > **Security** > **Intrusion Detection and Prevention** > **Policy Table**); the table shows the pre-configured IDS policies:

Figure 13-3: IDS Policy Table with Default Rules

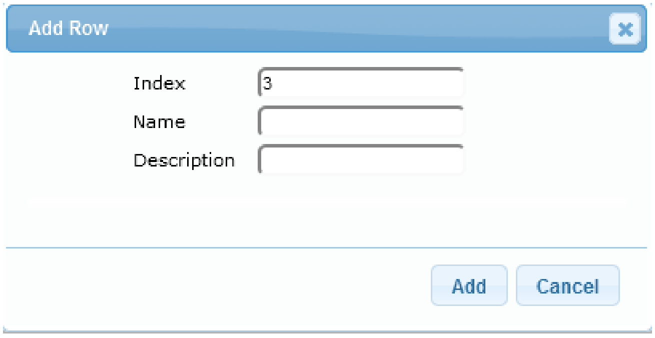


Index	Name	Description
0	DEFAULT_FEU	Default policy for FEU
1	DEFAULT_PROXY	Default policy for proxies
2	DEFAULT_GLOBAL	Default policy for global scope

IDS Policy Table #0 Additional Configuration
IDS Rule Table

2. Click **Add**; the following dialog box appears:

Figure 13-4: IDS Policy Table - Add Row Dialog Box



Index 3
Name
Description

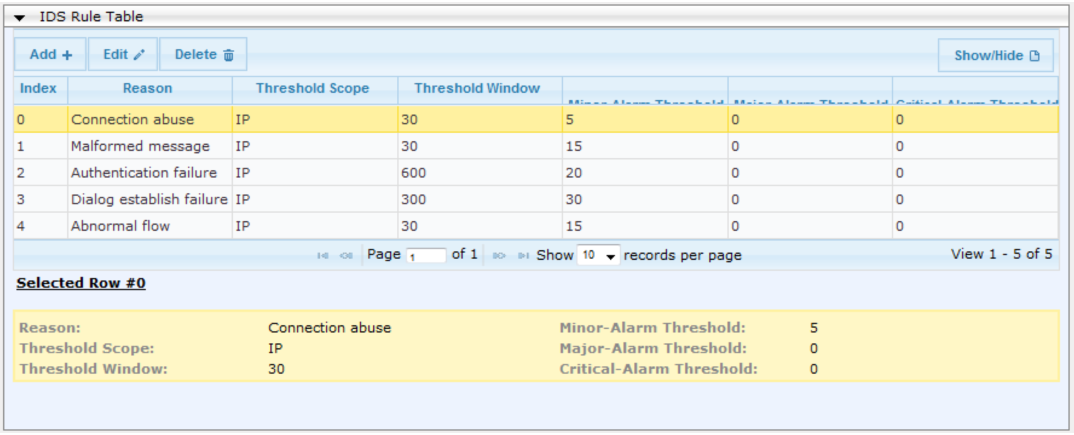
3. Configure an IDS Policy name according to the parameters described in the table below.
4. Click **Add**.

Table 13-3: IDS Policy Table Parameter Descriptions

Parameter	Description
Index `policy` [IDSPolicy_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `rule` [IDSPolicy_Name]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters.
Description [IDSPolicy_Description]	Defines a brief description for the IDS Policy. The valid value is a string of up to 100 characters.

5. In the IDS Policy table, select the required IDS Policy row, and then click the **IDS Rule Table** link located below the table; the IDS Rule table opens:

Figure 13-5: IDS Rule Table of Selected IDS Policy



IDS Rule Table

Index	Reason	Threshold Scope	Threshold Window	Minor-Alarm Threshold	Major-Alarm Threshold	Critical-Alarm Threshold
0	Connection abuse	IP	30	5	0	0
1	Malformed message	IP	30	15	0	0
2	Authentication failure	IP	600	20	0	0
3	Dialog establish failure	IP	300	30	0	0
4	Abnormal flow	IP	30	15	0	0

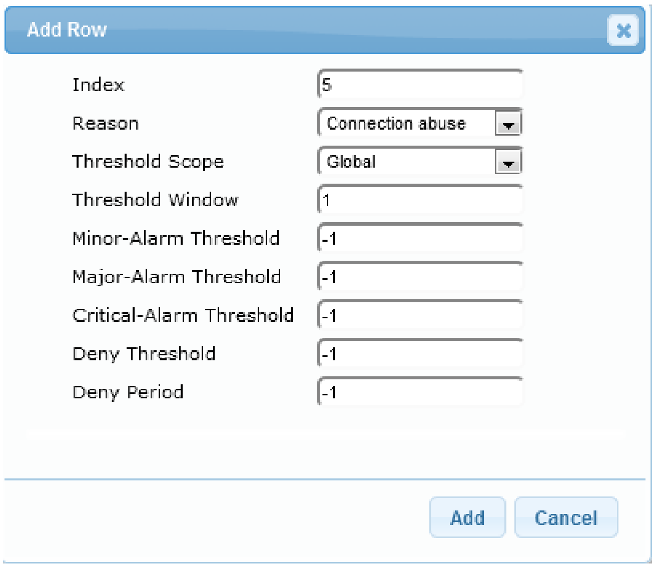
Page 1 of 1 Show 10 records per page View 1 - 5 of 5

Selected Row #0

Reason:	Connection abuse	Minor-Alarm Threshold:	5
Threshold Scope:	IP	Major-Alarm Threshold:	0
Threshold Window:	30	Critical-Alarm Threshold:	0

6. Click **Add**; the following dialog box appears:

Figure 13-6: IDS Rule Table - Add Record



Add Row

Index	5
Reason	Connection abuse
Threshold Scope	Global
Threshold Window	1
Minor-Alarm Threshold	-1
Major-Alarm Threshold	-1
Critical-Alarm Threshold	-1
Deny Threshold	-1
Deny Period	-1

The figure above shows a configuration example. If 15 malformed SIP messages are received within a period of 30 seconds, a minor alarm is sent. Every 30 seconds, the rule's counters are cleared. In addition, if more than 25 malformed SIP messages are received within this period, the device blacklists the remote IP host from where the messages were received for 60 seconds.

7. Configure an IDS Rule according to the parameters described in the table below.
8. Click **Add**, and then save ("burn") your settings to flash memory.

Table 13-4: IDS Rule Table Parameter Descriptions

Parameter	Description
Index `rule-id` [IDSRule_RuleID]	Defines an index number for the new table record.
Reason `reason` [IDSRule_Reason]	Defines the type of intrusion attack (malicious event). ▪ **[0]** Any = All events listed below are considered as attacks and are counted together. ▪ **[1]** Connection abuse = (Default) TLS authentication failure. ▪ **[2]** Malformed message = ✓ Message exceeds a user-defined maximum message length (50K) ✓ Any SIP parser error ✓ Message Policy match (see "Configuring SIP Message Policy Rules") ✓ Basic headers not present ✓ Content length header not present (for TCP) ✓ Header overflow ▪ **[3]** Authentication failure = ✓ Local authentication ("Bad digest" errors) ✓ Remote authentication (SIP 401/407 is sent if original message includes authentication) ▪ **[4]** Dialog establish failure = ✓ Classification failure (see "Configuring Classification Rules" on page 569) ✓ Routing failure ✓ Other local rejects (prior to SIP 180 response) ✓ Remote rejects (prior to SIP 180 response) ▪ **[5]** Abnormal flow = ✓ Requests and responses without a matching transaction user (except ACK requests) ✓ Requests and responses without a matching transaction (except ACK requests)
Threshold Scope `threshold-scope` [IDSRule_ThresholdScope]	Defines the source of the attacker to consider in the device's detection count. ▪ **[0]** Global = All attacks regardless of source are counted together during the threshold window. ▪ **[2]** IP = Attacks from each specific IP address are counted separately during the threshold window. ▪ **[3]** IP+Port = Attacks from each specific IP address:port are counted separately during the threshold window. This option is useful for NAT servers, where numerous remote machines use the same IP address but different ports. However, it is not recommended to use this option as it may degrade detection capabilities.

Parameter	Description
Threshold Window `threshold-window` [IDSRule_ThresholdWindow]	Defines the threshold interval (in seconds) during which the device counts the attacks to check if a threshold is crossed. The counter is automatically reset at the end of the interval. The valid range is 1 to 1,000,000. The default is 1.
Minor-Alarm Threshold `minor-alrm-thr` [IDSRule_MinorAlarmThreshold]	Defines the threshold that if crossed a minor severity alarm is sent. The valid range is 1 to 1,000,000. A value of 0 or -1 means not defined.
Major-Alarm Threshold `major-alrm-thr` [IDSRule_MajorAlarmThreshold]	Defines the threshold that if crossed a major severity alarm is sent. The valid range is 1 to 1,000,000. A value of 0 or -1 means not defined.
Critical-Alarm Threshold `critical-alrm-thr` [IDSRule_CriticalAlarmThreshold]	Defines the threshold that if crossed a critical severity alarm is sent. The valid range is 1 to 1,000,000. A value of 0 or -1 means not defined.
Deny Threshold [IDSRule_DenyThreshold]	Defines the threshold that if crossed, the device blocks (blacklists) the remote host (attacker). The default is -1 (i.e., not configured). **Note:** The parameter is applicable only if the 'Threshold Scope' parameter is set to **IP** or **IP+Port**.
Deny Period [IDSRule_DenyPeriod]	Defines the duration (in sec) to keep the attacker on the blacklist. The valid range is 0 to 1,000,000. The default is -1 (i.e., not configured).

13.3.3 Assigning IDS Policies

The IDS Match table lets you implement your configured IDS Policies. You do this by assigning IDS Policies to any, or a combination of, the following configuration entities:

- **SIP Interface:** For detection of malicious attacks on specific SIP Interface(s). For configuring SIP Interfaces, see "Configuring SIP Interfaces" on page 333.
- **Proxy Sets:** For detection of malicious attacks from specified Proxy Set(s). For configuring Proxy Sets, see "Configuring Proxy Sets" on page 352.
- **Subnet addresses:** For detection of malicious attacks from specified subnet addresses.

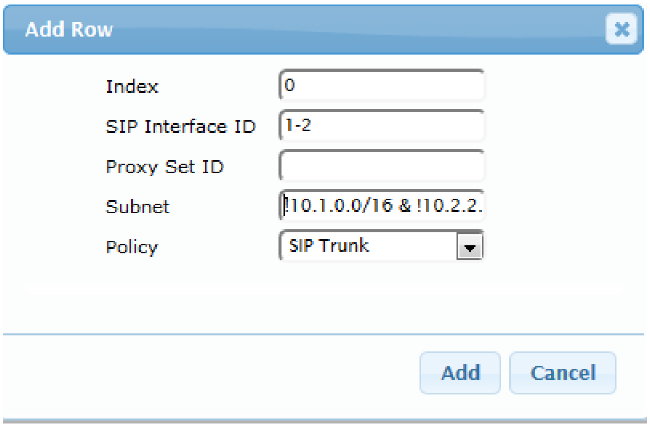
You can configure up to 20 IDS Policy-Matching rules.

➢ **To configure an IDS Policy-Matching rule:**

1. Open the IDS Match table (**Configuration** tab > **VoIP** menu > **Security** > **Intrusion Detection and Prevention** > **Match Table**).

2. Click **Add**; the following dialog box appears:

Figure 13-7: IDS Match Table - Add Row Dialog Box



Index: 0
SIP Interface ID: 1-2
Proxy Set ID:
Subnet: !10.1.0.0/16 & !10.2.2.
Policy: SIP Trunk

The figure above shows a configuration example where the IDS Policy "SIP Trunk" is applied to SIP Interfaces 1 and 2, and all source IP addresses outside of subnet 10.1.0.0/16 and IP address 10.2.2.2.

3. Configure a rule according to the parameters described in the table below.
4. Click **Add**, and then save ("burn") your settings to flash memory.

Table 13-5: IDS Match Table Parameter Descriptions

Parameter	Description
Index [IDSMatch_Index]	Defines an index number for the new table record.
SIP Interface ID `sip-interface` [IDSMatch_SIPInterface]	Defines the SIP Interface(s) to which you want to assign the IDS Policy. This indicates the SIP Interfaces that are being attacked. The valid value is the ID of the SIP Interface. The following syntax is supported: ▪ A comma-separated list of SIP Interface IDs (e.g., 1,3,4) ▪ A hyphen "-" indicates a range of SIP Interfaces (e.g., 3,4-7 means IDs 3, and 4 through 7) ▪ A prefix of an exclamation mark "!" means negation of the set (e.g., !3,4-7 means all indexes excluding 3, and excluding 4 through 7)
Proxy Set ID `proxy-set` [IDSMatch_ProxySet]	Defines the Proxy Set(s) to which the IDS Policy is assigned. This indicates the Proxy Sets from where the attacks are coming from. The following syntax is supported: ▪ A comma-separated list of Proxy Set IDs (e.g., 1,3,4) ▪ A hyphen "-" indicates a range of Proxy Sets (e.g., 3,4-7 means IDs 3, and 4 through 7) ▪ A prefix of an exclamation mark "!" means negation of the set (e.g., !3,4-7 means all indexes excluding 3, and excluding 4 through 7) **Notes:** ▪ Only the IP address of the Proxy Set is considered (not port). ▪ If a Proxy Set has multiple IP addresses, the device considers the Proxy Set as one entity and includes all its IP addresses in the same IDS count.

Parameter	Description
Subnet `subnet` [IDSMatch_Subnet]	Defines the subnet to which the IDS Policy is assigned. This indicates the subnets from where the attacks are coming from. The following syntax can be used: ▪ Basic syntax is a subnet in CIDR notation (e.g., 10.1.0.0/16 means all sources with IP address in the range 10.1.0.0–10.1.255.255) ▪ An IP address can be specified without the prefix length to refer to the specific IP address. ▪ Each subnet can be negated by prefixing it with "!", which means all IP addresses outside that subnet. ▪ Multiple subnets can be specified by separating them with "&" (and) or "|" (or) operations. For example: ✓ 10.1.0.0/16 | 10.2.2.2: includes subnet 10.1.0.0/16 and IP address 10.2.2.2. ✓ !10.1.0.0/16 & !10.2.2.2: includes all addresses except those of subnet 10.1.0.0/16 and IP address 10.2.2.2. Note that the exclamation mark "!" appears before each subnet. ✓ 10.1.0.0/16 & !10.1.1.1: includes subnet 10.1.0.0/16, except IP address 10.1.1.1.
Policy `policy` [IDSMatch_Policy]	Assigns an IDS Policy (configured in "Configuring IDS Policies" on page 164).

13.3.4 Viewing IDS Alarms

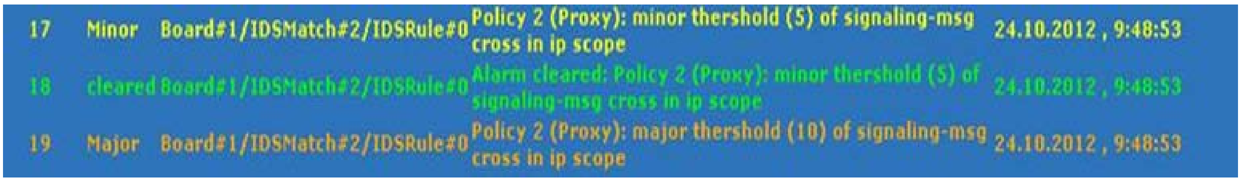
For the IDS feature, the device sends the following SNMP traps:

- Traps that notify the detection of malicious attacks:
  - **acIDSPolicyAlarm:** The device sends this alarm whenever a threshold of a specific IDS Policy rule is crossed. The trap displays the crossed severity threshold (Minor or Major), IDS Policy and IDS Rule, and the IDS Policy-Match index.
  - **acIDSThresholdCrossNotification:** The device sends this event for each scope (IP address) that crosses the threshold. In addition to the crossed severity threshold (Minor or Major) of the IDS Policy-Match index, this event shows the IP address (or IP address:port) of the malicious attacker.

    If the severity level is raised, the alarm of the former severity is cleared and the device sends a new alarm with the new severity. The alarm is cleared after a user-defined period (configured by the ini file parameter, IDSAlarmClearPeriod) during which no thresholds have been crossed. However, this "quiet" period must be at least twice the 'Threshold Window' value (configured in "Configuring IDS Policies" on page 164). For example, if you set IDSAlarmClearPeriod to 20 sec and 'Threshold Window' to 15 sec, the IDSAlarmClearPeriod parameter is ignored and the alarm is cleared only after 30 seconds (2 x 15 sec).

The figure below displays an example of IDS alarms in the Active Alarms table ("Viewing Active Alarms" on page 709). In this example, a Minor threshold alarm is cleared and replaced by a Major threshold alarm:

Figure 13-8: IDS Alarms in Active Alarms Table



17	Minor	Board#1/IDSMatch#2/IDSRule#0	Policy 2 (Proxy): minor thershold (5) of signaling-msg cross in ip scope	24.10.2012 , 9:48:53
18	cleared	Board#1/IDSMatch#2/IDSRule#0	Alarm cleared: Policy 2 (Proxy): minor thershold (5) of signaling-msg cross in ip scope	24.10.2012 , 9:48:53
19	Major	Board#1/IDSMatch#2/IDSRule#0	Policy 2 (Proxy): major thershold (10) of signaling-msg cross in ip scope	24.10.2012 , 9:48:53

- acIDSBlacklistNotification event: The device sends this event whenever an attacker (remote host at IP address and/or port) is added to or removed from the blacklist.

You can also view IDS alarms in the CLI, using the following commands:

- To view all active IDS alarms:

```
# show voip security ids active-alarm all
```

- To view all IP addresses that have crossed the threshold for an active IDS alarm:

```
# show voip security ids active-alarm match <IDS Match Policy ID> rule <IDS Rule ID>
```

  The IP address is displayed only if the 'Threshold Scope' parameter is set to IP or IP+Port; otherwise, only the alarm is displayed.

- To view the blacklist:

```
# show voip security ids blacklist active
```

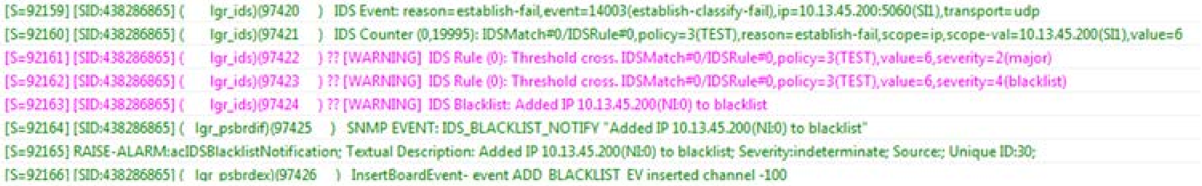
  For example:

```
Active blacklist entries:
  10.33.5.110(NI:0) remaining 00h:00m:10s in blacklist
```

  Where SI is the SIP Interface and NI is the network interface.

The device also sends IDS notifications and alarms in Syslog messages to a Syslog server. This occurs only if you have configured Syslog (see "Enabling Syslog" on page 783). An example of a Syslog message with IDS alarms and notifications is shown below:

Figure 13-9: Syslog Message Example with IDS Alarms and Notifications



```
[S=92159] [SID:438286865] (     lgr_ids)(97420    )  IDS Event: reason=establish-fail,event=14003(establish-classify-fail),ip=10.13.45.200:5060(SI1),transport=udp
[S=92160] [SID:438286865] (     lgr_ids)(97421    )  IDS Counter (0,19995): IDSMatch#0/IDSRule#0,policy=3(TEST),reason=establish-fail,scope=ip,scope-val=10.13.45.200(SI1),value=6
[S=92161] [SID:438286865] (     lgr_ids)(97422    ) ?? [WARNING]  IDS Rule (0): Threshold cross. IDSMatch#0/IDSRule#0,policy=3(TEST),value=6,severity=2(major)
[S=92162] [SID:438286865] (     lgr_ids)(97423    ) ?? [WARNING]  IDS Rule (0): Threshold cross. IDSMatch#0/IDSRule#0,policy=3(TEST),value=6,severity=4(blacklist)
[S=92163] [SID:438286865] (     lgr_ids)(97424    ) ?? [WARNING]  IDS Blacklist: Added IP 10.13.45.200(NI:0) to blacklist
[S=92164] [SID:438286865] (  lgr_psbrdif)(97425    )  SNMP EVENT: IDS_BLACKLIST_NOTIFY "Added IP 10.13.45.200(NI:0) to blacklist"
[S=92165] RAISE-ALARM:acIDSBlacklistNotification; Textual Description: Added IP 10.13.45.200(NI:0) to blacklist; Severity:indeterminate; Source:; Unique ID:30;
[S=92166] [SID:438286865] (  lgr psbrdex)(97426    )  InsertBoardEvent- event ADD BLACKLIST  EV inserted channel -100
```

The table below lists the Syslog text messages per malicious event:

Table 13-6: Types of Malicious Events and Syslog Text String

Type	Description	Syslog String
Connection Abuse	TLS authentication failure	abuse-tls-auth-fail
Malformed Messages	▪ Message exceeds a user-defined maximum message length (50K) ▪ Any SIP parser error ▪ Message policy match ▪ Basic headers not present ▪ Content length header not present (for TCP) ▪ Header overflow	▪ malformed-invalid-msg-len ▪ malformed-parse-error ▪ malformed-message-policy ▪ malformed-miss-header

Type	Description	Syslog String
		▪ malformed-miss-content-len ▪ malformed-header-overflow
Authentication Failure	▪ Local authentication ("Bad digest" errors) ▪ Remote authentication (SIP 401/407 is sent if original message includes authentication)	▪ auth-establish-fail ▪ auth-reject-response
Dialog Establishment Failure	▪ Classification failure ▪ Routing failure ▪ Other local rejects (prior to SIP 180 response) ▪ Remote rejects (prior to SIP 180 response)	▪ establish-classify-fail ▪ establish-route-fail ▪ establish-local-reject ▪ establish-remote-reject
Abnormal Flow	▪ Requests and responses without a matching transaction user (except ACK requests) ▪ Requests and responses without a matching transaction (except ACK requests)	▪ flow-no-match-tu ▪ flow-no-match-transaction

14 Media

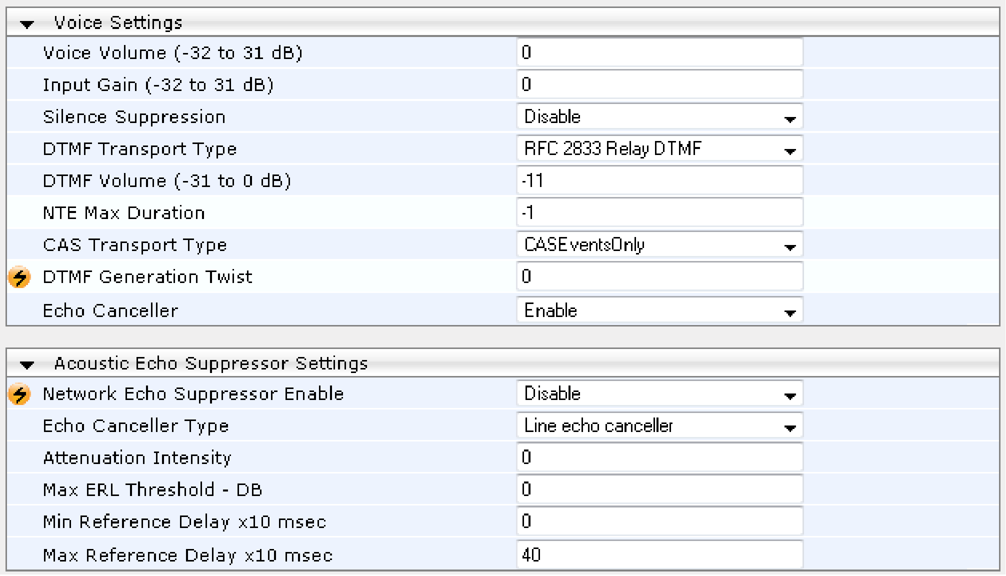
This section describes the media-related configuration.

14.1 Configuring Voice Settings

The Voice Settings page configures various voice parameters such as voice volume and DTMF transport type. For a detailed description of these parameters, see "Configuration Parameters Reference" on page 813.

➢ **To configure the voice parameters:**

1. Open the Voice Settings page (**Configuration** tab > **VoIP** menu > **Media** > **Voice Settings**).



▼ Voice Settings

Voice Volume (-32 to 31 dB)	0
Input Gain (-32 to 31 dB)	0
Silence Suppression	Disable
DTMF Transport Type	RFC 2833 Relay DTMF
DTMF Volume (-31 to 0 dB)	-11
NTE Max Duration	-1
CAS Transport Type	CASEventsOnly
⚡ DTMF Generation Twist	0
Echo Canceller	Enable

▼ Acoustic Echo Suppressor Settings

⚡ Network Echo Suppressor Enable	Disable
Echo Canceller Type	Line echo canceller
Attenuation Intensity	0
Max ERL Threshold - DB	0
Min Reference Delay x10 msec	0
Max Reference Delay x10 msec	40

2. Configure the Voice parameters as required.
3. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

14.1.1 Configuring Voice Gain (Volume) Control

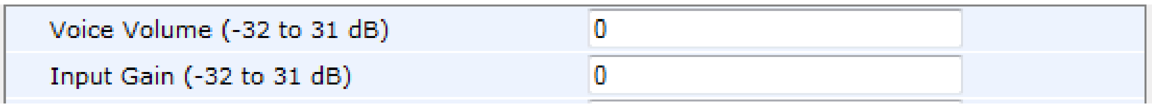
The device allows you to configure the level of the received (input gain) Tel-to-IP signal and the level of the transmitted (output gain) IP-to-Tel signal. The gain can be set between -32 and 31 decibels (dB).

The following procedure describes how to configure gain control using the Web interface.

➢ **To configure gain control using the Web interface:**

1. Open the Voice Settings page (**Configuration** tab > **VoIP** menu > **Media** > **Voice Settings**).

Figure 14-1: Voice Volume Parameters in Voice Settings Page



Voice Volume (-32 to 31 dB)	0
Input Gain (-32 to 31 dB)	0

2. Configure the following parameters:
   - 'Voice Volume' (*VoiceVolume*) - Defines the voice gain control (in decibels) of the transmitted signal.

- 'Input Gain' (*InputGain*) - Defines the PCM input gain control (in decibels) of the received signal.

3. Click **Submit**.

14.1.2 Configuring Echo Cancellation

The device supports adaptive linear (line) echo cancellation according to G.168-2002. Echo cancellation is a mechanism that removes echo from the voice channel. Echoes are reflections of the transmitted signal.

In this line echo, echoes are generated when two-wire telephone circuits (carrying both transmitted and received signals on the same wire pair) are converted to a four-wire circuit. Echoes are reflections of the transmitted signal, which result from impedance mismatch in the hybrid (bi-directional 2-wire to 4-wire converting device).

An estimated echo signal is built by feeding the decoder output signal to an RLS-like adaptive filter, which adapts itself to the characteristics of the echo path. The ‘estimated echo signal’ (the output of this filter) is then subtracted from the input signal (which is the sum of the desired input signal and the undesired echo) to provide a clean signal. To suppress the remaining residual echo, a Non Linear Processor (NLP) is used, as well as a double-talk (two people speak at the same time) detector that prevents false adaptation during near-end speech.

The following procedure describes how to configure echo cancellation using the Web interface:

➢ **To configure echo cancellation using the Web interface:**

1. Configure line echo cancellation:
   a. Open the Voice Settings page (**Configuration** tab > **VoIP** menu > **Media** > **Voice Settings**).
   b. Set the 'Echo Canceller' field (*EnableEchoCanceller*) to **Enable**.

Note: The following additional echo cancellation parameters are configurable only through the *ini* file:

- *ECHybridLoss* **-** defines the four-wire to two-wire worst-case Hybrid loss
- *ECNLPMode* **-** defines the echo cancellation Non-Linear Processing (NLP) mode
- *EchoCancellerAggressiveNLP* **-** enables Aggressive NLP at the first 0.5 second of the call

14.2 Fax and Modem Capabilities

This section describes the device's fax and modem capabilities and corresponding configuration. The fax and modem configuration is done in the Fax/Modem/CID Settings page.

Notes:

- Unless otherwise specified, the configuration parameters mentioned in this section are available on this page.
- Some SIP parameters override these fax and modem parameters. For example, the IsFaxUsed parameter and V.152 parameters in Section "V.152 Support" on page 187.
- For a detailed description of the parameters appearing on this page, see "Configuration Parameters Reference" on page 813.

➢ **To access the fax and modem parameters:**

1. Open the Fax/Modem/CID Settings page (**Configuration** tab > **VoIP** menu > **Media** > **Fax/Modem/CID Settings**).
2. Configure the parameters, as required.
3. Click **Submit**.

14.2.1 Fax/Modem Operating Modes

The device supports two modes of operation:

- Fax/modem negotiation that is not performed during the establishment of the call.
- Voice-band data (VBD) mode for V.152 implementation (see "V.152 Support" on page 187): fax/modem capabilities are negotiated between the device and the remote endpoint at the establishment of the call. During a call, when a fax/modem signal is detected, transition from voice to VBD (or T.38) is automatically performed and no additional SIP signaling is required. If negotiation fails (i.e., no match is achieved for any of the transport capabilities), fallback to existing logic occurs (according to the parameter IsFaxUsed).

14.2.2 Fax/Modem Transport Modes

The device supports the following transport modes for fax per modem type (V.22/V.23/Bell/V.32/V.34):

- T.38 fax relay (see "T.38 Fax Relay Mode" on page 175)
- G.711 Transport: switching to G.711 when fax/modem is detected (see "G.711 Fax / Modem Transport Mode" on page 178)
- Fax fallback to G.711 if T.38 is not supported (see "Fax Fallback" on page 178)
- Fax and modem bypass: a proprietary method that uses a high bit rate coder (see "Fax/Modem Bypass Mode" on page 179)
- NSE Cisco’s Pass-through bypass mode for fax and modem (see "Fax / Modem NSE Mode" on page 180)
- Transparent with events: passing the fax / modem signal in the current voice coder with adaptations (see "Fax / Modem Transparent with Events Mode" on page 181)
- Transparent: passing the fax / modem signal in the current voice coder (see "Fax / Modem Transparent Mode" on page 181)
- RFC 2833 ANS Report upon Fax/Modem Detection (see "RFC 2833 ANS Report upon Fax/Modem Detection" on page 182)

‘Adaptations’ refer to automatic reconfiguration of certain DSP features for handling fax/modem streams differently than voice.

14.2.2.1 T.38 Fax Relay Mode

In Fax Relay mode, fax signals are transferred using the T.38 protocol. T.38 is the ITU standard for sending fax across IP networks in real-time mode. The device currently supports only the T.38 UDP syntax.

T.38 can be configured in the following ways:

- Switching to T.38 mode using SIP Re-INVITE messages (see "Switching to T.38 Mode using SIP Re-INVITE" on page 176)
- Automatically switching to T.38 mode without using SIP Re-INVITE messages (see "Automatically Switching to T.38 Mode without SIP Re-INVITE" on page 176)

When fax transmission ends, the reverse switching from fax relay to voice is automatically performed at both the local and remote endpoints.

You can change the fax rate declared in the SDP, using the 'Fax Relay Max Rate' parameter (FaxRelayMaxRate). The parameter does not affect the actual transmission rate. You can also enable or disable Error Correction Mode (ECM) fax mode using the 'Fax Relay ECM Enable' parameter (FaxRelayECMEnable).

When using T.38 mode, you can define a redundancy feature to improve fax transmission over congested IP networks. This feature is activated using the 'Fax Relay Redundancy Depth' parameter (FaxRelayRedundancyDepth) and the 'Fax Relay Enhanced Redundancy Depth' parameter (FaxRelayEnhancedRedundancyDepth). Although this is a proprietary redundancy scheme, it should not create problems when working with other T.38 decoders.

14.2.2.1.1Switching to T.38 Mode using SIP Re-INVITE

In the Switching to T.38 Mode using SIP Re-INVITE mode, upon detection of a fax signal the terminating device negotiates T.38 capabilities using a Re-INVITE message. If the far-end device doesn't support T.38, the fax fails. In this mode, the 'Fax Transport Mode' parameter (FaxTransportMode) is ignored.

➢ **To configure T.38 mode using SIP Re-INVITE messages:**

1. In the SIP General Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**), set the 'Fax Signaling Method' parameter to **T.38 Relay** (IsFaxUsed = 1).
2. In the Fax/Modem/CID Settings page, configure the following optional parameters:
   - 'Fax Relay Redundancy Depth' (FaxRelayRedundancyDepth)
   - 'Fax Relay Enhanced Redundancy Depth' (FaxRelayEnhancedRedundancyDepth)
   - 'Fax Relay ECM Enable' (FaxRelayECMEnable)
   - 'Fax Relay Max Rate' (FaxRelayMaxRate)

Note: The terminating gateway sends T.38 packets immediately after the T.38 capabilities are negotiated in SIP. However, the originating device by default, sends T.38 (assuming the T.38 capabilities are negotiated in SIP) only after it receives T.38 packets from the remote device. This default behavior cannot be used when the originating device is located behind a firewall that blocks incoming T.38 packets on ports that have not yet received T.38 packets from the internal network. To resolve this problem, the device should be configured to send CNG packets in T.38 upon CNG signal detection (CNGDetectorMode = 1).

14.2.2.1.2Automatically Switching to T.38 Mode without SIP Re-INVITE

In the Automatically Switching to T.38 Mode without SIP Re-INVITE mode, when a fax signal is detected, the channel automatically switches from the current voice coder to answer tone mode and then to T.38-compliant fax relay mode.

➢ **To configure automatic T.38 mode:**

1. In the SIP General Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**), set the 'Fax Signaling Method' parameter to **No Fax** (IsFaxUsed = 0).
2. In the Fax/Modem/CID Settings page, set the 'Fax Transport Mode' parameter to **T.38 Relay** (FaxTransportMode = 1).
3. Configure the following optional parameters:
   - 'Fax Relay Redundancy Depth' (FaxRelayRedundancyDepth)

- 'Fax Relay Enhanced Redundancy Depth' (FaxRelayEnhancedRedundancyDepth)
- 'Fax Relay ECM Enable' (FaxRelayECMEnable)
- 'Fax Relay Max Rate' (FaxRelayMaxRate)

14.2.2.1.3Fax over IP using T.38 Transmission over RTP

The device supports Fax-over-IP (FoIP) transmission using T.38 over RTP, whereby the T.38 payload is encapsulated in the RTP packet, instead of being sent in dedicated T.38 packets (out-of-band). To configure this support, set the coder type to T.38 Over RTP.

To indicate T.38 over RTP, the SDP body uses "udptl" (Facsimile UDP Transport Layer) in the 'a=ftmp' line. The device supports T.38 over RTP according to this standard as well as according to AudioCodes proprietary method:

- **Call Parties belong to AudioCodes Devices:** AudioCodes proprietary T.38-over-RTP method is used, whereby the device encapsulates the entire T.38 packet (payload with all its headers) in the sent RTP. For T.38 over RTP, AudioCodes devices use the proprietary identifier "AcUdptl" in the 'a=ftmp' line of the SDP. For example:

```
v=0
o=AudiocodesGW 1357424688 1357424660 IN IP4 10.8.6.68
s=Phone-Call
c=IN IP4 10.8.6.68
t=0 0
m=audio 6080 RTP/AVP 18 100 96
a=ptime:20
a=sendrecv
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:100 t38/8000
a=fmtp:100 T38FaxVersion=0
a=fmtp:100 T38MaxBitRate=0
a=fmtp:100 T38FaxMaxBuffer=3000
a=fmtp:100 T38FaxMaxDatagram=122
a=fmtp:100 T38FaxRateManagement=transferredTCF
a=fmtp:100 T38FaxUdpEC=t38UDPRedundancy
a=fmtp:100 AcUdptl
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
```

- **AudioCodes Call Party with non-AudioCodes Party:** The device uses the standard T.38-over-RTP method, which encapsulates the T.38 payload only, without its headers (i.e., includes only fax data) in the sent RTP packet (RFC 4612).

The T.38-over-RTP method also depends on call initiator:

- **Device initiates a call:** The device always sends the SDP offer with the proprietary token "AcUdpTl" in the 'fmtp' attribute. If the SDP answer includes the same token, the device employs AudioCodes proprietary T.38-over-RTP mode; otherwise, the standard mode is used.
- **Device answers a call:** If the SDP offer from the remote party contains the 'fmtp' attribute with "AcUdpTl", the device answers with the same attribute and employs AudioCodes proprietary T.38-over-RTP mode; otherwise, the standard mode is used.

Note: If both T.38 (regular) and T.38 Over RTP coders are negotiated between the call parties, the device uses T.38 Over RTP.

14.2.2.2 G.711 Fax / Modem Transport Mode

In this mode, when the terminating device detects fax or modem signals (CED or AnsAM), it sends a Re-INVITE message to the originating device, requesting it to re-open the channel in G.711 VBD with the following adaptations:

- Echo Canceller = off
- Silence Compression = off
- Echo Canceller Non-Linear Processor Mode = off
- Dynamic Jitter Buffer Minimum Delay = 40
- Dynamic Jitter Buffer Optimization Factor = 13

After a few seconds upon detection of fax V.21 preamble or super G3 fax signals, the device sends a second Re-INVITE enabling the echo canceller (the echo canceller is disabled only on modem transmission).

A ‘gpmd’ attribute is added to the SDP according to the following format:

- **For G.711 A-law:**

  ```
  a=gpmd:0 vbd=yes;ecan=on (or off for modems)
  ```

- **For G.711 µ-law:**

  ```
  a=gpmd:8 vbd=yes;ecan=on (or off for modems)
  ```

The following parameters are ignored and automatically set to **Events Only**:

- 'Fax Transport Mode' (FaxTransportMode)
- 'Vxx ModemTransportType' (VxxModemTransportType)

➢ **To configure fax / modem transparent mode:**

- In the SIP General Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**), set the 'Fax Signaling Method' parameter to **G.711 Transport** (IsFaxUsed = 2).

14.2.2.3 Fax Fallback

In this mode, when the terminating device detects a fax signal, it sends a Re-INVITE message to the originating device with T.38. If the remote device doesn’t support T.38 (replies with SIP response 415 "Media Not Supported"), the device sends a new Re-INVITE with G.711 VBD with the following adaptations:

- Echo Canceller = on
- Silence Compression = off
- Echo Canceller Non-Linear Processor Mode = off
- Dynamic Jitter Buffer Minimum Delay = 40
- Dynamic Jitter Buffer Optimization Factor = 13

When the device initiates a fax session using G.711, a ‘gpmd’ attribute is added to the SDP according to the following format:

- **For G.711A-law:**

  ```
  a=gpmd:0 vbd=yes;ecan=on
  ```

- **For G.711 µ-law:**

  ```
  a=gpmd:8 vbd=yes;ecan=on
  ```

In this mode, the 'Fax Transport Mode' (FaxTransportMode) parameter is ignored and automatically set to **Disable** (transparent mode).

➢ **To configure fax fallback mode:**

- In the SIP General Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**), set the 'Fax Signaling Method' parameter to **Fax Fallback** (IsFaxUsed = 3).

14.2.2.4 Fax/Modem Bypass Mode

In this proprietary mode, when fax or modem signals are detected, the channel automatically switches from the current voice coder to a high bit-rate coder, according to the 'Fax/Modem Bypass Coder Type' parameter (FaxModemBypassCoderType). The channel is also automatically reconfigured with the following fax / modem adaptations:

- Disables silence suppression
- Enables echo cancellation for fax
- Disables echo cancellation for modem
- Performs certain jitter buffering optimizations

The network packets generated and received during the bypass period are regular voice RTP packets (per the selected bypass coder), but with a different RTP payload type according to the following parameters:

- 'Fax Bypass Payload Type' (FaxBypassPayloadType)
- ModemBypassPayloadType (ini file)

During the bypass period, the coder uses the packing factor, configured by the 'Fax/Modem Bypass Packing Factor' parameter (FaxModemBypassM). The packing factor determines the number of coder payloads (each the size of FaxModemBypassBasicRTPPacketInterval) that are used to generate a single fax/modem bypass packet. When fax/modem transmission ends, the reverse switching, from bypass coder to regular voice coder is performed.

➢ **To configure fax / modem bypass mode:**

1. In the SIP General Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**), set the 'Fax Signaling Method' parameter to **No Fax** (IsFaxUsed = 0).
2. In the Fax/Modem/CID Settings page, do the following:
   a. Set the 'Fax Transport Mode' parameter to **Bypass** (FaxTransportMode = 2).
   b. Set the 'V.21 Modem Transport Type' parameter to **Enable Bypass** (V21ModemTransportType = 2).
   c. Set the 'V.22 Modem Transport Type' parameter to **Enable Bypass** (V22ModemTransportType = 2).
   d. Set the 'V.23 Modem Transport Type' parameter to **Enable Bypass** (V23ModemTransportType = 2).
   e. Set the 'V.32 Modem Transport Type' parameter to **Enable Bypass** (V32ModemTransportType = 2).
   f. Set the 'V.34 Modem Transport Type' parameter to **Enable Bypass** (V34ModemTransportType = 2).
3. Set the ini file parameter, BellModemTransportType to 2 (Bypass).
4. Configure the following optional parameters:
   - 'Fax/Modem Bypass Coder Type' (FaxModemBypassCoderType).
   - 'Fax Bypass Payload Type' (FaxBypassPayloadType) - in the RTP/RTCP Settings page (**Configuration** tab > **VoIP** menu > **Media** > **RTP/RTCP Settings**).
   - ModemBypassPayloadType (ini file).
   - FaxModemBypassBasicRTPPacketInterval (ini file).
   - FaxModemBypasDJBufMinDelay (ini file).

Note: When the device is configured for modem bypass and T.38 fax, V.21 low-speed modems are not supported and fail as a result.

Tip: When the remote (non-AudioCodes) gateway uses the G.711 coder for voice and doesn't change the coder payload type for fax or modem transmission, it is recommended to use the Bypass mode with the following configuration:

- EnableFaxModemInbandNetworkDetection = 1.
- 'Fax/Modem Bypass Coder Type' = same coder used for voice.
- 'Fax/Modem Bypass Packing Factor'(FaxModemBypassM) = same interval as voice.
- ModemBypassPayloadType = 8 if voice coder is A-Law or 0 if voice coder is Mu-Law.

14.2.2.5 Fax / Modem NSE Mode

In this mode, fax and modem signals are transferred using Cisco-compatible Pass-through bypass mode. Upon detection of fax or modem answering tone signal, the terminating device sends three to six special NSE RTP packets (configured by the NSEpayloadType parameter; usually to 100). These packets signal the remote device to switch to G.711 coder, according to the 'Fax/Modem Bypass Packing Factor' parameter. After a few NSE packets are exchanged between the devices, both devices start using G.711 packets with standard payload type (8 for G.711 A-Law and 0 for G.711 Mu-Law). In this mode, no Re-INVITE messages are sent. The voice channel is optimized for fax/modem transmission (same as for usual bypass mode).

Note: This feature is applicable only to the Gateway application.

The parameters defining payload type for AudioCodes proprietary Bypass mode -- 'Fax Bypass Payload Type' (RTP/RTCP Settings page) and ModemBypassPayloadType (ini file) -- are not used with NSE Bypass.

When configured for NSE mode, the device includes in its SDP the following line:

```
a=rtpmap:100 X-NSE/8000
```

Where *100* is the NSE payload type.

The Cisco gateway must include the following definition:

```
modem passthrough nse payload-type 100 codec g711alaw
```

➢ **To configure NSE mode:**

1. In the SIP General Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**), set the 'Fax Signaling Method' parameter to **No Fax** (IsFaxUsed = 0).
2. In the Fax/Modem/CID Settings page, do the following:
   a. Set the 'Fax Transport Mode' parameter to **Bypass** (FaxTransportMode = 2).
   b. Set the 'V.21 Modem Transport Type' parameter to **Enable Bypass** (V21ModemTransportType = 2).
   c. Set the 'V.22 Modem Transport Type' parameter to **Enable Bypass** (V22ModemTransportType = 2).

d. Set the 'V.23 Modem Transport Type' parameter to **Enable Bypass** (V23ModemTransportType = 2).

e. Set the 'V.32 Modem Transport Type' parameter to **Enable Bypass** (V32ModemTransportType = 2).

f. Set the 'V.34 Modem Transport Type' parameter to **Enable Bypass** (V34ModemTransportType = 2).

3. Set the ini file parameter, BellModemTransportType to 2 (Bypass).
4. Set the ini file parameter, NSEMode parameter to 1 (enables NSE).
5. Set the ini file parameter, NSEPayloadType parameter to 100.

14.2.2.6 Fax / Modem Transparent with Events Mode

In this mode, fax and modem signals are transferred using the current voice coder with the following automatic adaptations:

- Echo Canceller = on (or off for modems)
- Echo Canceller Non-Linear Processor Mode = off
- Jitter buffering optimizations

➢ **To configure fax / modem transparent with events mode:**

1. In the SIP General Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**), set the 'Fax Signaling Method' parameter to **No Fax** (IsFaxUsed = 0).
2. In the Fax/Modem/CID Settings page, do the following:
   a. Set the 'Fax Transport Mode' parameter to **Events Only** (FaxTransportMode = 3).
   b. Set the 'V.21 Modem Transport Type' parameter to **Events Only** (V21ModemTransportType = 3).
   c. Set the 'V.22 Modem Transport Type' parameter to **Events Only** (V22ModemTransportType = 3).
   d. Set the 'V.23 Modem Transport Type' parameter to **Events Only** (V23ModemTransportType = 3).
   e. Set the 'V.32 Modem Transport Type' parameter to **Events Only** (V32ModemTransportType = 3).
   f. Set the 'V.34 Modem Transport Type' parameter to **Events Only** (V34ModemTransportType = 3).
3. Set the ini file parameter, BellModemTransportType to 3 (transparent with events).

14.2.2.7 Fax / Modem Transparent Mode

In this mode, fax and modem signals are transferred using the current voice coder without notifications to the user and without automatic adaptations. It's possible to use Profiles (see "Coders and Profiles" on page 379) to apply certain adaptations to the channel used for fax / modem. For example, to use the coder G.711, to set the jitter buffer optimization factor to 13, and to enable echo cancellation for fax and disable it for modem.

➢ **To configure fax / modem transparent mode:**

1. In the SIP General Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**), set the 'Fax Signaling Method' parameter to **No Fax** (IsFaxUsed = 0).
2. In the Fax/Modem/CID Settings page, do the following:
   a. Set the 'Fax Transport Mode' parameter to **Disable** (FaxTransportMode = 0).

   b. Set the 'V.21 Modem Transport Type' parameter to **Disable** (V21ModemTransportType = 0).
   c. Set the 'V.22 Modem Transport Type' parameter to **Disable** (V22ModemTransportType = 0).
   d. Set the 'V.23 Modem Transport Type' parameter to **Disable** (V23ModemTransportType = 0).
   e. Set the 'V.32 Modem Transport Type' parameter to **Disable** (V32ModemTransportType = 0).
   f. Set the 'V.34 Modem Transport Type' parameter to **Disable** (V34ModemTransportType = 0).
3. Set the ini file parameter, BellModemTransportType to 0 (transparent mode).
4. Configure the following optional parameters:
   a. Coders table - (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **Coders**).
   b. 'Dynamic Jitter Buffer Optimization Factor' (DJBufOptFactor) - RTP/RTCP Settings page (**Configuration** tab > **VoIP** menu > **Media** > **RTP/RTCP Settings**).
   c. 'Echo Canceller' (EnableEchoCanceller) - Voice Settings page.

> **Note:** This mode can be used for fax, but is not recommended for modem transmission. Instead, use the Bypass (see "Fax/Modem Bypass Mode" on page 179) or Transparent with Events modes (see "Fax / Modem Transparent with Events Mode" on page 181) for modem.

14.2.2.8 RFC 2833 ANS Report upon Fax/Modem Detection

The device (terminator gateway) sends RFC 2833 ANS/ANSam events upon detection of fax and/or modem answer tones (i.e., CED tone). This causes the originator to switch to fax/modem. The parameter is applicable only when the fax or modem transport type is set to bypass, Transparent-with-Events, V.152 VBD, or G.711 transport. When the device is located on the originator side, it ignores these RFC 2833 events

➢ **To configure RFC 2833 ANS Report upon fax/modem detection:**

1. In the SIP General Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**), set the 'Fax Signaling Method' parameter to **No Fax** or **Fax Fallback** (IsFaxUsed = 0 or 3).
2. In the Fax/Modem/CID Settings page, do the following:
   a. Set the 'Fax Transport Mode' parameter to **Bypass** (FaxTransportMode = 2).
   b. Set the 'V.xx Modem Transport Type' parameters to **Enable Bypass** (VxxModemTransportType = 2).
3. Set the ini file parameter, FaxModemNTEMode to 1 (enables this feature).

14.2.3 V.34 Fax Support

V.34 fax machines can transmit data over IP to the remote side using various methods. The device supports the following modes for transporting V.34 fax data over IP:

- T38 Version 3 - V.34 fax relay mode
- Bypass mechanism for V.34 fax transmission (see "Bypass Mechanism for V.34 Fax Transmission" on page 183)
- T38 Version 0 relay mode, i.e., fallback to T.38 (see "Relay Mode for T.30 and V.34 Faxes" on page 183)

To configure whether to pass V.34 over T38 fax relay, or use Bypass over the High Bit Rate coder (e.g. PCM A-Law), use the 'V.34 Fax Transport Type' parameter (V34FaxTransportType).

You can use the 'SIP T.38 Version' parameter (SIPT38Version) in the Advanced Parameters page (Configuration tab > VoIP menu > SIP Definitions > Advanced Parameters) to configure one of the following:

- Pass V.34 over T.38 fax relay using bit rates of up to 33,600 bps ('SIP T.38 Version' is set to Version 3).
- Use Fax-over-T.38 fallback to T.30, using up to 14,400 bps ('SIP T.38 Version' is set to Version 0).

Note: The CNG detector is disabled in all the subsequent examples. To disable the CNG detector, set the 'CNG Detector Mode' parameter (CNGDetectorMode) to **Disable**.

14.2.3.1 Bypass Mechanism for V.34 Fax Transmission

In this proprietary scenario, the device uses bypass (or NSE) mode to transmit V.34 faxes, enabling the full utilization of its speed.

➢ **To use bypass mode for T.30 and V.34 faxes:**

1. In the Fax/Modem/CID Settings page, do the following:
   a. Set the 'Fax Transport Mode' parameter to **Bypass** (FaxTransportMode = 2).
   b. Set the 'V.22 Modem Transport Type' parameter to **Enable Bypass** (V22ModemTransportType = 2).
   c. Set the 'V.23 Modem Transport Type' parameter to **Enable Bypass** (V23ModemTransportType = 2).
   d. Set the 'V.32 Modem Transport Type' parameter to **Enable Bypass** (V32ModemTransportType = 2).
   e. Set the 'V.34 Modem Transport Type' parameter to **Enable Bypass** (V34ModemTransportType = 2).
2. Set the ini file parameter, V34FaxTransportType to 2 (Bypass).

➢ **To use bypass mode for V.34 faxes, and T.38 for T.30 faxes:**

1. In the Fax/Modem/CID Settings page, do the following:
   a. Set the 'Fax Transport Mode' parameter to **T.38 Relay** (FaxTransportMode = 1).
   b. Set the 'V.22 Modem Transport Type' parameter to **Enable Bypass** (V22ModemTransportType = 2).
   c. Set the 'V.23 Modem Transport Type' parameter to **Enable Bypass** (V23ModemTransportType = 2).
   d. Set the 'V.32 Modem Transport Type' parameter to **Enable Bypass** (V32ModemTransportType = 2).
   e. Set the 'V.34 Modem Transport Type' parameter to **Enable Bypass** (V34ModemTransportType = 2).
2. Set the ini file parameter, V34FaxTransportType to 2 (Bypass).

14.2.3.2 Relay Mode for T.30 and V.34 Faxes

In this scenario, V.34 fax machines are forced to use their backward compatibility with T.30 faxes and operate in the slower T.30 mode.

➢ **To use T.38 mode for V.34 and T.30 faxes:**

1. In the Fax/Modem/CID Settings page, do the following:
   a. Set the 'Fax Transport Mode' parameter to **T.38 Relay** (FaxTransportMode = 1).
   b. Set the 'V.22 Modem Transport Type' parameter to **Disable** (V22ModemTransportType = 0).
   c. Set the 'V.23 Modem Transport Type' parameter to **Disable** (V23ModemTransportType = 0).
   d. Set the 'V.32 Modem Transport Type' parameter to **Disable** (V32ModemTransportType = 0).
   e. Set the 'V.34 Modem Transport Type' parameter to **Disable** (V34ModemTransportType = 0).
2. Set the ini file parameter, V34FaxTransportType to 1 (Relay).

➢ **To allow V.34 fax relay over T.38:**

- In the Advanced Parameters page (Configuration tab > VoIP menu > SIP Definitions > Advanced Parameters), set the 'SIP T.38 Version' parameter to Version 3 (SIPT38Version = 3).

➢ **To force V.34 fax machines to use their backward compatibility with T.30 faxes and operate in the slower T.30 mode:**

- Set the 'SIP T.38 Version' parameter to Version 0 (SIPT38Version = 0).

14.2.3.3 V.34 Fax Relay for SG3 Fax Machines

Super Group 3 (SG3) is a standard for fax machines that support speeds of up to 33.6 kbps through V.34 half duplex (HD) modulation. The following procedure describes how to configure V.34 (SG3) fax relay support based on ITU Specification T.38 version 3.

➢ **To enable support for V.34 fax relay (T.38) at SG3 speed:**

1. In the IP Profile table (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **IP Profile Settings**), configure an IP Profile with the 'Fax Signaling Method' parameter (IpProfile_IsFaxUsed) set to **T.38 Relay**.
2. In the Coders Table (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **Coders**), set the coder used by the device to G.729 (or any other supported codec).
3. On the Fax/Modem/CID Settings page, do the following settings:
   a. 'SIP T.38 Version' to **Version 3** (SIPT38Version = 3).
   b. 'Fax Relay Max Rate' (RelayMaxRate) to **33,600bps** (default).
   c. 'CNG Detector Mode' (CNGDetectorMode) to **Disable** (default).
   d. 'V.21 Modem Transport Type' to **Disable** (V21ModemTransportType = 0).
   e. 'V.22 Modem Transport Type' to **Disable** (V22ModemTransportType = 0).
   f. 'V.23 Modem Transport Type' to **Disable** (V23ModemTransportType = 0).
   g. 'V.32 Modem Transport Type' to **Disable** (V32ModemTransportType = 0).
   h. 'V.34 Modem Transport Type' to **Disable** (V34ModemTransportType = 0).
   i. 'CED Transfer Mode' to Fax Relay or VBD (CEDTransferMode = 0). (Applicable only to the Gateway application.)
4. Set the ini file parameter, V34FaxTransportType to 1 (i.e., relay).
5. Set the ini file parameter, T38MaxDatagramSize to 560 (default).

Notes:

- The T.38 negotiation should be completed at call start according to V.152 procedure (as shown in the INVITE example below).
- T.38 mid-call Re-INVITEs are supported.
- If the remote party supports only T.38 Version 0, the device "downgrades" the T.38 Version 3 to T.38 Version 0.

For example, the device sends or receives the following INVITE message, negotiating both audio and image media:

```
INVITE sip:2001@10.8.211.250;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.8.6.55;branch=z9hG4bKac1938966220
Max-Forwards: 70
From: <sip:318@10.8.6.55>;tag=1c1938956155
To: <sip:2001@10.8.211.250;user=phone>
Call-ID: 193895529241200022331@10.8.6.55
CSeq: 1 INVITE
Contact: <sip:318@10.8.6.55:5060>
Supported: em,100rel,timer,replaces,path,resource-priority,sdp-
anat
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUB
SCRIBE,UPDATE
Remote-Party-ID:
<sip:318@10.8.211.250>;party=calling;privacy=off;screen=no;screen-
ind=0;npi=1;ton=0
Remote-Party-ID: <sip:2001@10.8.211.250>;party=called;npi=1;ton=0
User-Agent: Audiocodes-Sip-Gateway-/v.6.80A.227.005
Content-Type: application/sdp
Content-Length: 433
v=0
o=AudiocodesGW 1938931006 1938930708 IN IP4 10.8.6.55
s=Phone-Call
c=IN IP4 10.8.6.55
t=0 0
m=audio 6010 RTP/AVP 18 97
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:97 telephone-event/8000
a=fmtp:97 0-15
a=ptime:20
a=sendrecv
m=image 6012 udptl t38
a=T38FaxVersion:3
a=T38MaxBitRate:33600
a=T38FaxMaxBuffer:1024
a=T38FaxMaxDatagram:122
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPRedundancy
```

14.2.4 V.150.1 Modem Relay

The device can be configured to transfer modem calls using a subset of the ITU-T V.150.1 Modem Relay protocol. The device also supports V.150.1 modem relay coder negotiation in the initial SIP INVITE and 200 OK, using the SDP body according to the USA Department of Defense (DoD) UCR-2008, Change 2 specification. This eliminates the need for sending a re-INVITE to negotiate V.150.1. The device sends an INVITE’s SDP offer in a format to negotiate V.150 modem relay using the same port as RTP, as shown in the example below:

```
a=cdsc:1 audio udpsprt 114\r\n
a=cpar:a=sprtmap:114 v150mr/8000\r\n
a=cpar:a=fmtp:114
mr=1;mg=0;CDSCselect=1;mrmods=1,3;jmdelay=no;versn=1.1\r\n\
```

You can configure the payload type for the outgoing SDP offer, using the NoAudioPayloadType parameter. You can set the parameter to "NoAudio", whereby RTP is not sent and the device adds an audio media only for the Modem Relay purpose. This is also in accordance to DOD UCR 2008 specification: "The AS-SIP signaling appliance MUST advertise the “NoAudio” payload type to interoperate with a “Modem Relay-Preferred” endpoint that immediately transitions to the Modem Relay state without first transmitting voice information in the Audio state."

Notes:

- The V.150.1 Modem Relay feature is available only if the device is installed with a Software License Key that includes this feature. For installing a Software License Key, see "Software License Key" on page 668.
- V.150.1 modem relay feature support is a subset of the full V.150.1 protocol and is designed according to the US DoD requirement document. It therefore, cannot be used for general purposes.
- V.150.1 modem relay is applicable only to the Gateway application.
- The V.150.1 feature has been tested with certain IP phones. For more details, please contact your AudioCodes sales representative.
- The V.150.1 SSE Tx payload type is according to the offered SDP of the remote side.
- The V.150.1 SPRT Rx payload type is according to the 'Payload Type' field in the Coders table.
- The V.150.1 SPRT Tx payload type is according to the remote side offered SDP.

➢ **To configure V.150.1 Modem relay:**

1. In the Coders Table (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **Coders**), set the coder to **V.150**.
2. On the Fax/Modem/CID Settings page, configure the V.150.1 parameters appearing under the 'V.150.1 Modem Relay Settings' group:
   a. Set the 'SSE Payload Type Rx' parameter to the V.150.1 SSE payload type that the device uses when it offers the SDP.
   b. Set the 'SSE Redundancy Depth' parameter to the number of sent SSE redundant packets. The parameter is important in case of network impairments.
   c. For additional V.150.1 related parameters, see "Fax and Modem Parameters" on page 915.

14.2.5 Simultaneous Negotiation of Fax (T.38) and Modem (V.150.1) Relay

The device can negotiate fax relay (T.38) and modem relay (V.150.1) sessions in the same, already established call channel. Fax relay sessions require bypass answering tone (CED) while modem relay requires RFC 2833 answering tone. As the device is not always aware at the start of the session whether the answering tone is fax or modem, it uses both methods for CED tone transfer and sends both answering tone types. Only when the answering tone is detected, does the device send the fax or modem.

To support this functionality, you need to configure a Coders Group (in the Coders Group table - see "Configuring Coder Groups" on page 382) that includes the T.38, V.150, and G.711/VBD coders.

14.2.6 V.152 Support

The device supports the ITU-T recommendation V.152 (Procedures for Supporting Voice-Band Data over IP Networks). Voice-band data (VBD) is the transport of modem, facsimile, and text telephony signals over a voice channel of a packet network with a codec appropriate for such signals.

For V.152 capability, the device supports T.38 as well as VBD codecs (i.e., G.711 A-law and G.711 µ-law). The selection of capabilities is performed using the coders table (see "Configuring Default Coders" on page 379).

When in VBD mode for V.152 implementation, support is negotiated between the device and the remote endpoint at the establishment of the call. During this time, initial exchange of call capabilities is exchanged in the outgoing SDP. These capabilities include whether VBD is supported and associated RTP payload types ('gpmd' SDP attribute), supported codecs, and packetization periods for all codec payload types ('ptime' SDP attribute). After this initial negotiation, no Re-INVITE messages are necessary as both endpoints are synchronized in terms of the other side's capabilities. If negotiation fails (i.e., no match was achieved for any of the transport capabilities), fallback to existing logic occurs (according to the parameter IsFaxUsed).

Below is an example of media descriptions of an SDP indicating support for V.152. In the example, V.152 implementation is supported (using the dynamic payload type 96 and G.711 u-law as the VBD codec) as well as the voice codecs G.711 µ-law and G.729.

```
v=0
o=-  0 0 IN IPV4 <IPAdressA>
s=-
t=0 0
p=+1
c=IN IP4  <IPAddressA
m=audio <udpPort A> RTP/AVP 18 0
a=ptime:10
a=rtpmap:96 PCMU/8000
a=gpmd: 96 vbd=yes
```

Instead of using VBD transport mode, the V.152 implementation can use alternative relay fax transport methods (e.g., fax relay over IP using T.38). The preferred V.152 transport method is indicated by the SDP ‘pmft’ attribute. Omission of this attribute in the SDP content means that VBD mode is the preferred transport mechanism for voice-band data. To configure T.38 mode, use the CodersGroup parameter.

Note: You can also configure the device to handle G.711 coders received in INVITE SDP offers as VBD coders, using the HandleG711asVBD parameter. For example, if the device is configured with G.729 and G.711 VBD coders and it receives an INVITE with an SDP offer containing G.729 and “regular” G.711 coders, it sends an SDP answer containing G.729 and G.711 VBD coders, allowing subsequent bypass (passthrough) sessions if fax / modem signals are detected during the call.

14.3 Configuring RTP/RTCP Settings

This section describes configuration relating to Real-Time Transport Protocol (RTP) and RTP Control Protocol (RTCP).

14.3.1 Configuring the Dynamic Jitter Buffer

Voice frames are transmitted at a fixed rate. If the frames arrive at the other end at the same rate, voice quality is perceived as good. However, some frames may arrive slightly faster or slower than the other frames. This is called jitter (delay variation) and degrades the perceived voice quality. To minimize this problem, the device uses a jitter buffer. The jitter buffer collects voice packets, stores them and sends them to the voice processor in evenly spaced intervals.

The device uses a dynamic jitter buffer that can be configured with the following:

- **Minimum delay:** Defines the starting jitter capacity of the buffer. For example, at 0 msec, there is no buffering at the start. At the default level of 10 msec, the device always buffers incoming packets by at least 10 msec worth of voice frames.
- **Optimization Factor:** Defines how the jitter buffer tracks to changing network conditions. When set at its maximum value of 12, the dynamic buffer aggressively tracks changes in delay (based on packet loss statistics) to increase the size of the buffer and doesn’t decay back down. This results in the best packet error performance, but at the cost of extra delay. At the minimum value of 0, the buffer tracks delays only to compensate for clock drift and quickly decays back to the minimum level. This optimizes the delay performance but at the expense of a higher error rate.

The default settings of 10 msec Minimum delay and 10 Optimization Factor should provide a good compromise between delay and error rate. The jitter buffer ‘holds’ incoming packets for 10 msec before making them available for decoding into voice. The coder polls frames from the buffer at regular intervals in order to produce continuous speech. As long as delays in the network do not change (jitter) by more than 10 msec from one packet to the next, there is always a sample in the buffer for the coder to use. If there is more than 10 msec of delay at any time during the call, the packet arrives too late. The coder tries to access a frame and is not able to find one. The coder must produce a voice sample even if a frame is not available. It therefore compensates for the missing packet by adding a Bad-Frame-Interpolation (BFI) packet. This loss is then flagged as the buffer being too small. The dynamic algorithm then causes the size of the buffer to increase for the next voice session. The size of the buffer may decrease again if the device notices that the buffer is not filling up as much as expected. At no time does the buffer decrease to less than the minimum size configured by the Minimum delay parameter.

In certain scenarios, the **Optimization Factor is set to 13**: One of the purposes of the Jitter Buffer mechanism is to compensate for clock drift. If the two sides of the VoIP call are not synchronized to the same clock source, one RTP source generates packets at a lower rate, causing under-runs at the remote Jitter Buffer. In normal operation (optimization factor 0 to 12), the Jitter Buffer mechanism detects and compensates for the clock drift by occasionally dropping a voice packet or by adding a BFI packet.

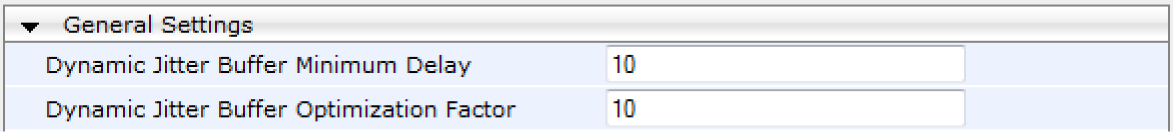
Fax and modem devices are sensitive to small packet losses or to added BFI packets. Therefore, to achieve better performance during modem and fax calls, the Optimization Factor should be set to 13. In this special mode the clock drift correction is performed less frequently - only when the Jitter Buffer is completely empty or completely full. When such condition occurs, the correction is performed by dropping several voice packets simultaneously or by adding several BFI packets simultaneously, so that the Jitter Buffer returns to its normal condition.

The following procedure describes how to configure the jitter buffer using the Web interface.

➢ To configure jitter buffer using the Web interface:

1. Open the RTP/RTCP Settings page (**Configuration** tab > **VoIP** menu > **Media** > **RTP/RTCP Settings**). The relevant parameters are listed under the 'General Settings' group, as shown below:

Figure 14-2: Jitter Buffer Parameters in the RTP/RTCP Settings Page



General Settings

Dynamic Jitter Buffer Minimum Delay	10
Dynamic Jitter Buffer Optimization Factor	10

2. Set the 'Dynamic Jitter Buffer Minimum Delay' parameter (DJBufMinDelay) to the minimum delay (in msec) for the Dynamic Jitter Buffer.

3. Set the 'Dynamic Jitter Buffer Optimization Factor' parameter (DJBufOptFactor) to the Dynamic Jitter Buffer frame error/delay optimization factor.
4. Click **Submit**.

14.3.2 Comfort Noise Generation

The device can generate artificial background noise, called *comfort* noise, in the voice channel during periods of silence (i.e. when no call party is speaking) for Gateway calls. This is useful in that it reassures the call parties that the call is still connected. The device detects silence using its Voice Activity Detection (VAD) mechanism. When Comfort Noise Generation is enabled and silence is detected, the device transmits Silence Identifier Descriptors (SIDs) parameters to reproduce the local background noise at the remote (receiving) side.

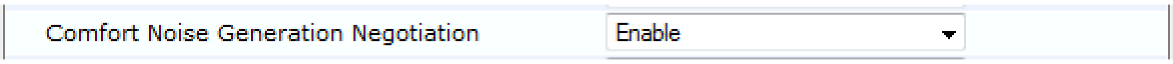
The Comfort Noise Generation support also depends on the silence suppression (SCE) setting for the coder used in the voice channel. For more information, see the description of the Comfort Noise Generation related parameters.

The following procedure describes how to configure Comfort Noise Generation using the Web interface.

➢ **To configure Comfort Noise Generation using the Web interface:**

1. Open the RTP/RTCP Settings page (**Configuration** tab > **VoIP** menu > **Media** > **RTP/RTCP Settings**). The relevant parameters are listed under the 'General Settings' group, as shown below:

Figure 14-3: Comfort Noise Parameter in RTP/RTCP Settings Page



Comfort Noise Generation Negotiation	Enable

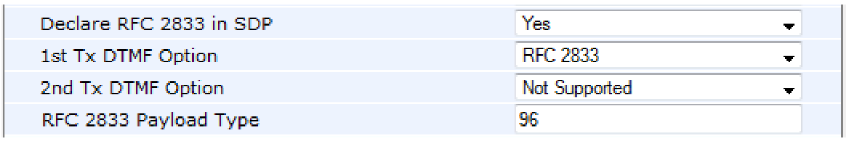
2. Set the 'Comfort Noise Generation Negotiation' parameter (ComfortNoiseNegotiation) to **Enable**.
3. Click **Submit**.

Note: This feature is applicable only to the Gateway application.

14.3.3 Configuring DTMF Transport Types

The device supports various methods for transporting DTMF digits over the IP network to the remote endpoint. The methods and their configuration can be configured in the DTMF & Dialing page (**Configuration** tab > **VoIP** menu > **Gateway** > **DTMF and Supplementary** > **DTMF & Dialing**):

Figure 14-4: DTMF Transport Parameters in Web Interface



Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option	RFC 2833
2nd Tx DTMF Option	Not Supported
RFC 2833 Payload Type	96

- **Using INFO message according to Nortel IETF draft:** DTMF digits are sent to the remote side in INFO messages. To enable the mode:
  a. Set the 'Declare RFC 2833 in SDP' parameter to **No** (RxDTMFOption = 0).
  b. Set the 'First Tx DTMF Option' parameter to **INFO Nortel** (FirstTxDTMFOption = 1).

Note: DTMF digits are removed from the audio stream (and the 'DTMF Transport Type' parameter is automatically set to **Mute DTMF**).

- **Using INFO message according to Cisco's mode:** DTMF digits are sent to the remote side in INFO messages. To enable the mode:

  a. Set the 'Declare RFC 2833 in SDP' parameter to **No** (RxDTMFOption = 0).

  b. Set the 'First Tx DTMF Option' parameter to **INFO Cisco** (FirstTxDTMFOption = 3).

  Note: DTMF digits are removed from the audio stream (and the 'DTMF Transport Type' parameter is automatically set to **Mute DTMF**).

- **Using NOTIFY messages according to IETF Internet-Draft draft-mahy-sipping-signaled-digits-01**: DTMF digits are sent to the remote side using NOTIFY messages. To enable the mode:

  a. Set the 'Declare RFC 2833 in SDP' parameter to **No** (RxDTMFOption = 0).

  b. Set the 'First Tx DTMF Option' parameter to **NOTIFY** (FirstTxDTMFOption = 2).

  Note: DTMF digits are removed from the audio stream (and the 'DTMF Transport Type' parameter is automatically set to **Mute DTMF**).

- **Using RFC 2833 relay with Payload type negotiation:** DTMF digits are sent to the remote side as part of the RTP stream according to RFC 2833. To enable the mode:

  a. Set the 'Declare RFC 2833 in SDP' parameter to **Yes** (RxDTMFOption = 3).

  b. Set the 'First Tx DTMF Option' parameter to **RFC 2833** (FirstTxDTMFOption = 4).

  Note: To set the RFC 2833 payload type with a value other than its default, use the RFC2833PayloadType parameter. The device negotiates the RFC 2833 payload type using local and remote SDP and sends packets using the payload type from the received SDP. The device expects to receive RFC 2833 packets with the same payload type as configured by the parameter. If the remote side doesn't include 'telephony-event' in its SDP, the device sends DTMF digits in transparent mode (as part of the voice stream).

- **Sending DTMF digits (in RTP packets) as part of the audio stream (DTMF Relay is disabled):** This method is typically used with G.711 coders. With other low-bit rate (LBR) coders, the quality of the DTMF digits is reduced. To enable the mode:

  a. Set the 'Declare RFC 2833 in SDP' parameter to **No** (RxDTMFOption = 0).

  b. Set the 'First Tx DTMF Option' parameter to **Not Supported** (FirstTxDTMFOption = 0).

  c. Set the ini file parameter, DTMFTransportType to 2 (i.e., transparent).

- **Using INFO message according to Korea mode:** DTMF digits are sent to the remote side in INFO messages. To enable this mode:

  a. Set the 'Declare RFC 2833 in SDP' parameter to **No** (RxDTMFOption = 0).

  b. Set the 'First Tx DTMF Option' parameter to **INFO Cisco** (FirstTxDTMFOption = 3).

  Note: DTMF digits are removed from the audio stream (and the 'DTMF Transport Type' parameter is automatically set to **Mute DTMF**).

Notes:

- The device is always ready to receive DTMF packets over IP in all possible transport modes: INFO messages, NOTIFY, and RFC 2833 (in proper payload type) or as part of the audio stream.
- To exclude RFC 2833 Telephony event parameter from the device's SDP, set the 'Declare RFC 2833 in SDP' parameter to **No**.
- You can use the following parameters to configure DTMF digit handling:
  - √ FirstTxDTMFOption, SecondTxDTMFOption, RxDTMFOption, RFC2833TxPayloadType, and RFC2833RxPayloadType
  - √ MGCPDTMFDetectionPoint, DTMFVolume, DTMFTransportType, DTMFDigitLength, and DTMFInterDigitInterval

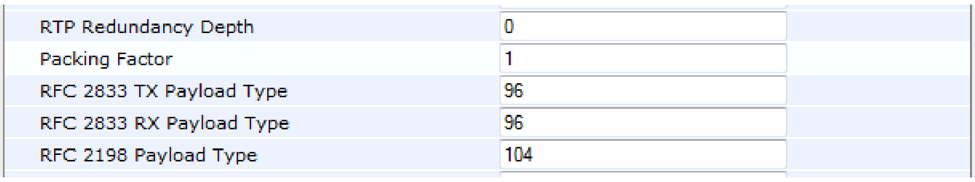
14.3.4 Configuring RFC 2833 Payload

The following procedure describes how to configure the RFC 2833 payload using the Web interface:

➢ **To configure RFC 2833 payload using the Web interface:**

1. Open the RTP/RTCP Settings page (**Configuration** tab > **VoIP** menu > **Media** > **RTP/RTCP Settings**). The relevant parameters are listed under the 'General Settings' group, as shown below:

Figure 14-5: RFC 2833 Payload Parameters on RTP/RTCP Settings Page



RTP Redundancy Depth	0
Packing Factor	1
RFC 2833 TX Payload Type	96
RFC 2833 RX Payload Type	96
RFC 2198 Payload Type	104

2. Configure the following parameters:
   - 'RTP Redundancy Depth' (RTPRedundancyDepth) - enables the device to generate RFC 2198 redundant packets.
   - For Gateway application only: 'Enable RTP Redundancy Negotiation' (EnableRTPRedundancyNegotiation) - enables the device to include the RTP redundancy dynamic payload type in the SDP, according to RFC 2198.
   - 'RFC 2833 TX Payload Type' (RFC2833TxPayloadType) - defines the Tx RFC 2833 DTMF relay dynamic payload type.
   - 'RFC 2833 RX Payload Type' (RFC2833RxPayloadType) - defines the Rx RFC 2833 DTMF relay dynamic payload type.
   - 'RFC 2198 Payload Type' (RFC2198PayloadType) - defines the RTP redundancy packet payload type according to RFC 2198.
3. Click **Submit**.

14.3.5 Configuring RTP Base UDP Port

You can configure the range of local UDP ports for RTP, RTCP, and T.38 media streams. The range of possible UDP ports that can be used, depending on configuration, is 6,000 through to 65,535. The device assigns ports **randomly** to the traffic within the configured port range.

For RTCP and T.38 traffic, the port offset from the RTP port used for the voice session is one and two, respectively. For example, if the voice session uses RTP port 6000, the RTCP port and T.38 port for the session is 6001 and 6002, respectively. However, you can configure the device to use the same port for RTP and T.38 packets, by setting the T38UseRTPPort parameter to 1.

Within the port range, the device allocates the UDP ports in "jumps" (spacing) of 10 (default). For example, if the port range starts at 6000 and the UDP port spacing is 10, the available ports include 6000, 6010, 6020, 6030, and so on.

The port range is calculated using the following equation:

```
BaseUDPPort to 65,535
```

Where, *BaseUDPPort* is a parameter for configuring the lower boundary of the port range (default is 6000) and *number of channels* is the maximum number of channels purchased from AudioCodes (included in the installed Software License Key).

For example, if the base UDP port is set to 6000, the port range is 6000 to 65,535.

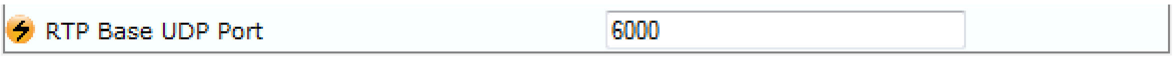
You can also configure specific port ranges for specific SIP entities, using Media Realms (see Configuring Media Realms on page 317). You can configure each Media Realm with a different UDP port range and then associate the Media Realm with a specific IP Group, for example. However, the port range of the Media Realm must be within the range configured by the BaseUDPPort parameter.

The following procedure describes how to configure the RTP base UDP port through the Web interface.

➢ To configure the RTP base UDP port:

1. Open the RTP/RTCP Settings page (**Configuration** tab > **VoIP** menu > **Media** > **RTP/RTCP Settings**). The relevant parameter is listed under the 'General Settings' group, as shown below:

Figure 14-6: RTP Based UDP Port in RTP/RTCP Settings Page



RTP Base UDP Port 6000

2. Set the 'RTP Base UDP Port' parameter to the required value.
3. Click **Submit**.
4. Reset the device for the settings to take effect.

Note:

- The RTP port must be different from ports configured for SIP signaling traffic (i.e., ports configured for SIP Interfaces). For example, if the RTP port range is 6000 to 6999, the SIP port can either be less than 6000 or greater than 6999.
- The base UDP port number (BaseUDPPort parameter) must be greater than the highest UDP port configured for a SIP Interface (see Configuring SIP Interfaces on page 333). For example, if your highest configured UDP port for a SIP Interface is 6060, you must configure the BaseUDPPort parameter to any value greater than 6060.

14.4 Event Detection and Notification using X-Detect Header

The device can detect certain events in the media stream (for SBC and Gateway calls) and notify of their detection to a remote application server, using the SIP X-Detect header. The request for event notification is done by the application server when establishing a SIP dialog (i.e., INVITE message) or during an already established call (SBC, IP-to-Tel, or Tel-to IP) using a re-INVITE message.

The device can detect the following event types:

- Answering Machine Detection (AMD): Detects events that are related to the AMD feature. AMD detects whether an answering machine or live voice has answered the call. It can also be used to detect silence, or the beep sound played by an answering machine to indicate the end of the greeting message after which a voice message can be left. For more information on AMD, see "Answering Machine Detection (AMD)" on page 198.
- Call Progress Tone (CPT): Detects whether a specific tone, defined in the installed CPT file is received from the call. It can be used to detect the beep sound played by an answering machine (as mentioned above), Special Information Tones (SIT) which indicate call failure with a recorded announcement describing the call failure, and the busy, reorder and ring tones.
- Fax/modem machine: Detects whether a fax has answered the call (preamble, CED, CNG, and modem).
- PTT: Detects the start and end of voice.

Notes:

- Currently, PTT is supported only for Gateway calls.
- Fax and SIT event detection is applicable only to Gateway calls.
- Event detection on SBC calls is supported only for calls using the G.711 coder.

The X-Detect header is used for event detection as follows:

- X-Detect header in the INVITE message received from the application server requesting a specific event detection:

  ```
  X-Detect: Request=[event type to detect]
  ```

- X-Detect header in the SIP response message -- SIP 183 (for early dialogs) or 200 OK (for confirmed dialogs) -- sent by the device to the application server specifying which of the requested events it can detect (absence of the X-Detect header indicates that the device cannot detect any of the events):

  ```
  X-Detect: Response=[supported event types]
  ```

- Each time the device detects the supported event, it sends an INFO message to the remote party with the following message body:

  ```
  Content-Type: Application/X-Detect
  Type = [event type]
  Subtype = [subtype of each event type]
  ```

The table below lists the event types and subtypes that the device can detect. The text shown in the table are the actual strings that are used in the X-Detect header. The table also provides a summary of the required configuration. For SBC calls, event detection is enabled using the IPProfile_SBCHandleXDetect parameter in the IP Profile table (see Configuring IP Profiles on page 387.

Table 14-1: Supported X-Detect Event Types

Event Type	Subtype	Description and Required Configuration
AMD	Voice (live voice) Automata (answering machine) Silence (no voice) Unknown Beep (greeting message of answering machine)	Event detection using the AMD feature. For more information, see Answering Machine Detection (AMD) on page 198.
CPT	SIT-NC SIT-IC SIT-VC SIT-RO Busy Reorder Ringtone Beep (greeting message of answering message)	Event detection of tones using the CPT file. 1 Create a CPT file with the required tone types of the events that you want to detect. 2 Install the CPT file on the device. 3 For SIT detection: a. Set the SITDetectorEnable parameter to 1. b. Set the UserDefinedToneDetectorEnable parameter to 1. **Notes:** ▪ For more information on SIT detection, see SIT Event Detection on page 194. ▪ For configuring beep detection, see Detecting Answering Machine Beep on page 195.
FAX	CED	▪ Set the IsFaxUsed parameter to any value other than 0. - or - ▪ Set the IsFaxUsed parameter to 0 and the FaxTransportMode parameter to any value other than 0.
	modem	Set the VxxModemTransportType parameter to 3.
PTT	voice-start voice-end	Set the EnableDSPIPMDetectors parameter to 1. **Note:** PTT is currently not supported for SBC calls.

14.4.1 SIT Event Detection

The device can detect and report the following Special Information Tones (SIT) types from the PSTN:

- SIT-NC (No Circuit found)
- SIT-IC (Operator Intercept)
- SIT-VC (Vacant Circuit - non-registered number)
- SIT-RO (Reorder - System Busy)

There are additional three SIT tones that are detected as one of the above SIT tones:

- The NC* SIT tone is detected as NC
- The RO* SIT tone is detected as RO
- The IO* SIT tone is detected as VC

The device can map these SIT tones to a Q.850 cause and then map them to SIP 5xx/4xx responses, using the parameters SITQ850Cause, SITQ850CauseForNC, SITQ850CauseForIC, SITQ850CauseForVC, and SITQ850CauseForRO.

Table 14-2: Special Information Tones (SITs) Reported by the device

Special Information Tones (SITs) Name	Description	First Tone Frequency Duration		Second Tone Frequency Duration		Third Tone Frequency Duration	
		(Hz)	(ms)	(Hz)	(ms)	(Hz)	(ms)
NC1	No circuit found	985.2	380	1428.5	380	1776.7	380
IC	Operator intercept	913.8	274	1370.6	274	1776.7	380
VC	Vacant circuit (non registered number)	985.2	380	1370.6	274	1776.7	380
RO1	Reorder (system busy)	913.8	274	1428.5	380	1776.7	380
NC*	-	913.8	380	1370.6	380	1776.7	380
RO*	-	985.2	274	1370.6	380	1776.7	380
IO*	-	913.8	380	1428.5	274	1776.7	380

The following example shows a SIP INFO message sent by the device to a remote application server notifying it that SIT detection has been detected:

```
INFO sip:5001@10.33.2.36 SIP/2.0
Via: SIP/2.0/UDP 10.33.45.65;branch=z9hG4bKac2042168670
Max-Forwards: 70
From: <sip:5000@10.33.45.65;user=phone>;tag=1c1915542705
To: <sip:5001@10.33.2.36;user=phone>;tag=WQJNIDDPCOKAPIDSCOTG
Call-ID: AIFHPETLLMVVFWPDXUHD@10.33.2.36
CSeq: 1 INFO
Contact: <sip:2206@10.33.45.65>
Supported: em,timer,replaces,path,resource-priority
Content-Type: application/x-detect
Content-Length: 28
Type= CPT
SubType= SIT-IC
```

14.4.2 Detecting Answering Machine Beeps

The device supports the detection of the beep sound played by an answering machine to indicate the end of the answering machine's greeting message. This is useful in that the device can then notify, for example, a third-party, application server that it can now leave a voice message on the answering machine. The device supports the following methods for detecting and reporting beeps:

- **AMD-based Detection:** The device uses its beep detector that is integrated in the AMD feature. You can configure the beep detection timeout and beep detection sensitivity level (for more information, see "Configuring AMD" on page 200). To enable the AMD beep detection, the received INVITE message must contain an X-Detect header with the value "Request=AMD",

  ```
  X-Detect: Request=AMD
  ```

  and the AMDBeepDetectionMode parameter must be set to 1 or 2. If set to 1, the beep is detected only after the answering machine is detected. If set to 2, the beep is detected even if the answering machine was not detected.

- **Tone-based Detection (Call Progress Tone):** The device detects the beep according to a call progress tone (CPT). This is enabled if the device receives a specific beep tone (Tone Type #46) that is also defined in the installed CPT file, and the received INVITE message contains an X-Detect header with the value "Request=CPT":

```
X-Detect: Request=CPT
```

For more information on the CPT file, see "Call Progress Tones File" on page 650.

The device reports beep detections to application servers, by sending a SIP INFO message that contains a body with one of the following values, depending on the method used for detecting the beep:

- AMD-detected Beep:

  ```
  Type= AMD
  SubType= Beep
  ```

- CPT-detected Beep:

  ```
  Type= CPT
  SubType=Beep
  ```

14.4.3 SIP Call Flow Examples of Event Detection and Notification

Two SIP call flow examples are provided below of event detection and notification:

- The following example shows a SIP call flow of the device's AMD and event detection feature, whereby the device detects an answering machine and the subsequent start and end of the greeting message, enabling the third-party application server to know when to play a recorded voice message to an answering machine:

  1. Upon detection of the answering machine, the device sends the following SIP INFO message to the application server:

     ```
     INFO sip:sipp@172.22.2.9:5060 SIP/2.0
     Via: SIP/2.0/UDP 172.22.168.249;branch=z9hG4bKac1566945480
     Max-Forwards: 70
     From: sut <sip:3000@172.22.168.249:5060>;tag=1c1505895240
     To: sipp <sip:sipp@172.22.2.9:5060>;tag=1
     Call-ID: 1-29758@172.22.2.9
     CSeq: 1 INFO
     Contact: <sip:56700@172.22.168.249>
     Supported: em,timer,replaces,path,resource-priority
     Allow:
     REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,I
     NFO,SUBSCRIBE,UPDATE
     User-Agent: Audiocodes-Sip-Gateway/v.6.80A.227.005
     Content-Type: application/x-detect
     Content-Length: 30
     Type= AMD
     SubType= AUTOMATA
     ```

  2. Upon detection of the start of voice (i.e., the greeting message of the answering machine), the device sends the following INFO message to the application server:

     ```
     INFO sip:sipp@172.22.2.9:5060 SIP/2.0
     Via: SIP/2.0/UDP 172.22.168.249;branch=z9hG4bKac482466515
     Max-Forwards: 70
     From: sut <sip:3000@172.22.168.249:5060>;tag=1c419779142
     To: sipp <sip:sipp@172.22.2.9:5060>;tag=1
     Call-ID: 1-29753@172.22.2.9
     CSeq: 1 INFO
     Contact: <sip:56700@172.22.168.249>
     Supported: em,timer,replaces,path,resource-priority
     Allow:
     REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,I
     NFO,SUBSCRIBE,UPDATE
     User-Agent: Audiocodes-Sip-Gateway/v.6.80A.227.005
     Content-Type: application/x-detect
     ```

```
Content-Length: 34
Type= PTT
SubType= SPEECH-START
```

3. Upon detection of the end of voice (i.e., end of the greeting message of the answering machine), the device sends the following INFO message to the application server:

```
INFO sip:sipp@172.22.2.9:5060 SIP/2.0
Via: SIP/2.0/UDP 172.22.168.249;branch=z9hG4bKac482466515
Max-Forwards: 70
From: sut <sip:3000@172.22.168.249:5060>;tag=1c419779142
To: sipp <sip:sipp@172.22.2.9:5060>;tag=1
Call-ID: 1-29753@172.22.2.9
CSeq: 1 INFO
Contact: <sip:56700@172.22.168.249>
Supported: em,timer,replaces,path,resource-priority
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,I
NFO,SUBSCRIBE,UPDATE
User-Agent: Audiocodes-Sip-Gateway/v.6.80A.227.005
Content-Type: application/x-detect
Content-Length: 34
Type= PTT
SubType= SPEECH-END
```

4. The application server sends its message to leave on the answering message.

- The following example shows a SIP call flow for event detection and notification of the beep of an answering machine:

1. The device receives a SIP message containing the X-Detect header from the remote application requesting beep detection:

```
INVITE sip:101@10.33.2.53;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.33.2.53;branch=z9hG4bKac5906
Max-Forwards: 70
From: "anonymous"
<sip:anonymous@anonymous.invalid>;tag=1c25298
To: <sip:101@10.33.2.53;user=phone>
Call-ID: 11923@10.33.2.53
CSeq: 1 INVITE
Contact: <sip:100@10.33.2.53>
X-Detect: Request=AMD,CPT
```

2. The device sends a SIP response message to the remote party, listing the events in the X-Detect header that it can detect:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.33.2.53;branch=z9hG4bKac5906
From: "anonymous"
<sip:anonymous@anonymous.invalid>;tag=1c25298
To: <sip:101@10.33.2.53;user=phone>;tag=1c19282
Call-ID: 11923@10.33.2.53
CSeq: 1 INVITE
Contact: <sip:101@10.33.2.53>
X-Detect: Response=AMD,CPT
```

3. The device detects the beep of an answering machine and sends an INFO message to the remote party:

```
INFO sip:101@10.33.2.53;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.33.2.53;branch=z9hG4bKac5906
Max-Forwards: 70
From: "anonymous"
<sip:anonymous@anonymous.invalid>;tag=1c25298
To: <sip:101@10.33.2.53;user=phone>
```

```
Call-ID: 11923@10.33.2.53
CSeq: 1 INVITE
Contact: <sip:100@10.33.2.53>
X- Detect: Response=AMD,CPT
Content-Type: Application/X-Detect
Content-Length: xxx
Type = CPT
Subtype = Beep
```

14.5 Answering Machine Detection (AMD)

The device's Answering Machine Detection (AMD) feature can detect whether an outbound call (SBC or Gateway call) has been answered by a human (including fax) or an answering machine. The device analyzes the sound (speech) patterns received in the first few seconds of the call to determine whether a human (live person) or machine has answered the call. Typically, when a human answers the call, there is a short "hello ..." followed by silence to wait for the other party to respond. In contrast, when an answering machine answers the call, there is constant speech (answering message) followed by a beep to leave a voice-mail message.

When the device detects what answered the call (human or machine), it can notify this detection type to, for example, a third-party application server used for automatic dialing applications. The X-Detect SIP header is used for requesting event detection and notification. For more information, see "Event Detection and Notification using X-Detect Header" on page 193. The device can also detect beeps played by an answering machine at the end of its greeting message. For more information, see "Detecting Answering Machine Beeps" on page 195. You can also configure the device to disconnect IP-to-Tel calls upon detection of an answering machine on the Tel side. For more information, see Enabling IP-to-Tel Call Disconnection upon Detection of Answering Machine on page 201.

The device's default AMD feature is based on voice detection for North American English (see note below). It uses AudioCodes' sophisticated speech detection algorithms which are based on hundreds of real-life recordings of answered calls by live voice and answering machines in English. The algorithms are used to detect whether it's human or machine based on voice and silence duration as well as speech patterns. The algorithms of the language-based recordings are compiled into a file called AMD Sensitivity. This file is provided by default, pre-installed on the device.

Note: As the main factor (algorithm) for detecting human and machine is the voice pattern and silence duration, the language on which the detection algorithm is based, is in most cases not important as these factors are similar across most languages. Therefore, the default, pre-installed AMD Sensitivity file, which is based on North American English, may suffice your deployment even if the device is located in a region where a language other than English is used.

However, if (despite the information stated in the note above) you wish to implement AMD in a different language or region or if you wish to fine-tune the default AMD algorithms to suit your specific deployment, please contact your AudioCodes sales representative for more information on this service. You will be typically required to provide AudioCodes with a database of recorded voices (calls) in the language on which the device's AMD feature can base its voice detector algorithms. The data needed for an accurate calibration should be recorded under the following guidelines:

- Statistical accuracy: The number of recorded calls should be as high as possible (at least 100) and varied. The calls must be made to different people. The calls must be made in the specific location in which the device's AMD feature is to operate.
- Real-life recording: The recordings should simulate real-life answering of a called person picking up the phone, and without the caller speaking.

- Normal environment interferences: The environment in which the recordings are done should simulate real-life scenarios, in other words, not sterile but not too noisy either. Interferences, for example, could include background noises of other people talking, spikes, and car noises.

Once you have provided AudioCodes with your database of recordings, AudioCodes compiles it into a loadable file. For a brief description of the file format and for installing the file on the device, see "AMD Sensitivity File" on page 667.

The device supports up to eight AMD algorithm suites called *Parameter Suites*, where each suite defines a range of detection sensitivity levels. Sensitivity levels refer to how accurately, based on AudioCodes' voice detection algorithms, the device can detect whether a human or machine has answered the call. Each level supports a different detection sensitivity to human and machine. For example, a specific sensitivity level may be more sensitive to detecting human than machine. In deployments where the likelihood of a call answered by an answering machine is low, it would be advisable to configure the device to use a sensitivity level that is more sensitive to human than machine. In addition, this allows you to tweak your sensitivity to meet local regulatory rules designed to protect consumers from automatic dialers (where, for example, the consumer picks up the phone and hears silence). Each suite can support up to 16 sensitivity levels (0 to 15), except for Parameter Suite 0, which supports up to 8 levels (0 to 7). The default, pre-installed AMD Sensitivity file, based on North American English, provides the following Parameter Suites:

- Parameter Suite 0 (normal sensitivity) - contains 8 sensitivity detection levels
- Parameter Suite 1 (high sensitivity) - contains 16 sensitivity detection levels

As Parameter Suite 1 provides a greater range of detection sensitivity levels (i.e., higher detection resolution), this may be the preferable suite to use in your deployment. The detected AMD type (human or machine) and success of detecting it correctly are sent in CDR and Syslog messages. For more information, see "Syslog Fields for Answering Machine Detection (AMD)" on page 778.

The Parameter Suite and sensitivity level can be applied globally for all calls, or for specific calls using IP Profiles. For enabling AMD and selecting the Parameter Suite and sensitivity level, see "Configuring AMD" on page 200.

The tables below show the success rates of the default, pre-installed AMD Sensitivity file (based on North American English) for correctly detecting "live" human voice and answering machine:

Table 14-3: Approximate AMD Normal Detection Sensitivity - Parameter Suite 0 (Based on North American English)

AMD Detection Sensitivity	Performance	
	Success Rate for Live Calls	Success Rate for Answering Machine
0 (Best for Answering Machine)	-	-
1	82.56%	97.10%
2	85.87%	96.43%
3	88.57%	94.76%
4	88.94%	94.31%
5	90.42%	91.64%
6	90.66%	91.30%
7 (Best for Live Calls)	94.72%	76.14%

Table 14-4: Approximate AMD High Detection Sensitivity - Parameter Suite 1 (Based on North American English)

AMD Detection Sensitivity	Performance	
	Success Rate for Live Calls	Success Rate for Answering Machine
0 (Best for Answering Machine)	72%	97%
1	77%	96%
2	79%	95%
3	80%	95%
4	84%	94%
5	86%	93%
6	87%	92%
7	88%	91%
8	90%	89%
9	90%	88%
10	91%	87%
11	94%	78%
12	94%	73%
13	95%	65%
14	96%	62%
15 (Best for Live Calls)	97%	46%

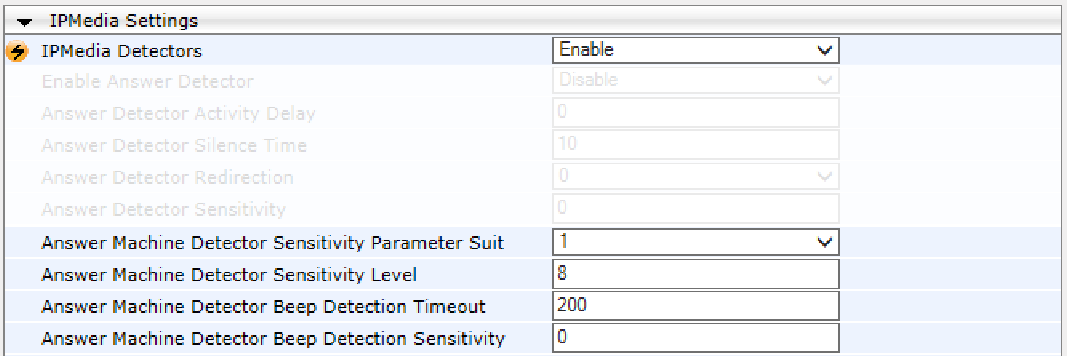
14.5.1 Configuring AMD

You can configure AMD for all calls using the global AMD parameters, or for specific calls using IP Profiles. The procedure below describes how to configure AMD for all calls. For configuring AMD for specific calls, use the AMD parameters in the IP Profile table (see "Configuring IP Profiles" on page 387). For Gateway calls, AMD can be configured per call based on the called number or Trunk Group. This is achieved by configuring AMD for a specific IP Profile and then assigning the IP Profile to a Trunk Group in the Inbound IP Routing table (see Configuring IP-to-Trunk Group Routing Rules on page 476).

➢ To enable and configure AMD for all calls:

1. Open the IPMedia Settings page (**Configuration** tab > **VoIP** > **Media** > **IPMedia Settings**):

Figure 14-7: Configuring AMD Parameters in the IPMedia Settings Page



IPMedia Settings

IPMedia Detectors	Enable
Enable Answer Detector	Disable
Answer Detector Activity Delay	0
Answer Detector Silence Time	10
Answer Detector Redirection	0
Answer Detector Sensitivity	0
Answer Machine Detector Sensitivity Parameter Suit	1
Answer Machine Detector Sensitivity Level	8
Answer Machine Detector Beep Detection Timeout	200
Answer Machine Detector Beep Detection Sensitivity	0

2. From the 'IPMedia Detectors' drop-down list (EnableDSPIPMDetectors), select **Enable** to enable AMD.
3. Select the AMD algorithm suite:
   a. In the 'Answer Machine Detector Sensitivity Parameter Suit' field, select the required Parameter Suite included in the installed AMD Sensitivity file.
   b. In the 'Answer Machine Detector Sensitivity' field, enter the required detection sensitivity level of the selected Parameter Suite.
4. Configure the answering machine beep detection:
   a. In the 'Answer Machine Detector Beep Detection Timeout' field (AMDBeepDetectionTimeout), enter the duration that the beep detector operates from when detection is initiated.
   b. In the 'Answer Machine Detector Beep Detection Sensitivity' field (AMDBeepDetectionSensitivity), enter the AMD beep detection sensitivity level.
5. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

For a complete list of AMD-related parameters, see "IP Media Parameters" on page 1019.

14.5.2 Enabling IP-to-Tel Call Disconnection upon Detection of Answering Machine

The device can disconnect an IP-to-Tel call upon detection of an answering machine on the Tel side. Once detected, the device disconnects the call after the receipt of an ISDN Connect from the Tel side and then sends a SIP BYE message to the IP side to disconnect the call. You can enable this feature for all calls (globally) using the AMDmode parameter, or for specific calls using IP Profiles where the IP Profile parameter 'AMD Mode' (IpProfile_AmdMode) is set to [1] **Disconnect on AMD** (see "Configuring IP Profiles" on page 387).

14.6 Configuring Various Codec Attributes

The following codec attribute settings can be configured in the General Media Settings page:

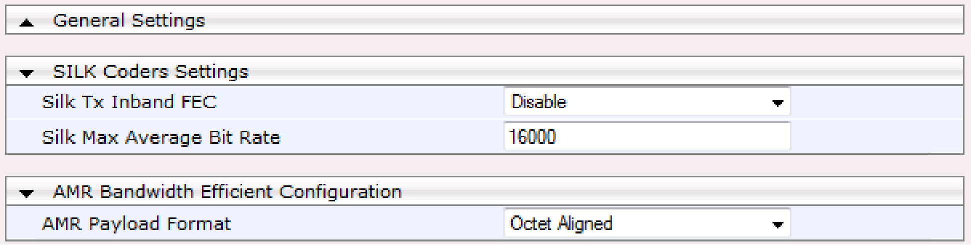
- AMR coder:
  - 'Payload Format': Defines the AMR payload format type.
- SILK coder (Skype's default audio codec):
  - 'Silk Tx Inband FEC': Enables forward error correction (FEC) for the SILK coder.
  - 'Silk Max Average Bit Rate': Defines the maximum average bit rate for the SILK coder.

For a detailed description of these parameters and for additional codec parameters, see "Coder Parameters" on page 908.

➢ **To configure codec attributes:**

1. Open the General Media Settings page (**Configuration** tab > **VoIP** menu > **Media** > **General Media Settings**).

Figure 14-8: Codec Settings in General Media Settings Page



Silk Tx Inband FEC: Disable
Silk Max Average Bit Rate: 16000
AMR Payload Format: Octet Aligned

2. Configure the parameters as required, and then click click **Submit**.
3. To save the changes to flash memory, see "Saving Configuration" on page 643.

14.7 Configuring Media (SRTP) Security

The device supports Secured RTP (SRTP) according to RFC 3711. SRTP is used to encrypt RTP and RTCP transport for protecting VoIP traffic. SRTP requires a cryptographic key exchange mechanism to negotiate the keys. To negotiate the keys, the device supports the Session Description Protocol Security Descriptions (SDES) protocol (according to RFC 4568) or Datagram Transport Layer Security (DTLS) protocol for SBC calls. For more information on DTLS, see SRTP using DTLS Protocol on page 204. The key exchange is done by adding the 'a=crypto' attribute to the SDP. This attribute is used (by both sides) to declare the various supported cipher suites and to attach the encryption key. If negotiation of the encryption data is successful, the call is established.

SRTP supports the following cipher suites (all other suites are ignored):

- AES_CM_128_HMAC_SHA1_32
- AES_CM_128_HMAC_SHA1_80
- ARIA_CM_128_HMAC_SHA1_80
- ARIA_CM_192_HMAC_SHA1_80

When the device is the offering side (SDP offer), it can generate a Master Key Identifier (MKI). You can configure the MKI size globally (using the SRTPTxPacketMKISize parameter) or per SIP entity (using the IP Profile parameter, IpProfile_MKISize). The length of the MKI is limited to four bytes. If the remote side sends a longer MKI, the key is ignored.

Note:

- Gateway application: The device only initiates the MKI size.
- SBC application: The device can forward MKI size transparently for SRTP-to-SRTP media flows or override the MKI size during negotiation (inbound or outbound leg).

The key lifetime field is not supported. However, if it is included in the key it is ignored and the call does not fail. For SBC calls belonging to a specific SIP entity, you can configure the device to remove the lifetime field in the 'a=crypto' attribute (using the IP Profile parameter, IpProfile_SBCRemoveCryptoLifetimeInSDP).

For SDES, the keys are sent in the SDP body ('a=crypto') of the SIP message and are typically secured using SIP over TLS (SIPS). The encryption of the keys is in plain text in the SDP. The device supports the following session parameters:

- UNENCRYPTED_SRTP
- UNENCRYPTED_SRTCP
- UNAUTHENTICATED_SRTP

Session parameters should be the same for the local and remote sides. When the device is the offering side, the session parameters are configured by the following parameter - 'Authentication On Transmitted RTP Packets', 'Encryption On Transmitted RTP Packets, and 'Encryption On Transmitted RTCP Packets'. When the device is the answering side, the device adjusts these parameters according to the remote offering. Unsupported session parameters are ignored, and do not cause a call failure.

Below is an example of crypto attributes usage:

```
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:PsKoMpHlCg+b5X0YLuSvNrImEh/dAe
a=crypto:2 AES_CM_128_HMAC_SHA1_32
inline:IsPtLoGkBf9a+c6XVzRuMqHlDnEiAd
```

The device also supports symmetric MKI negotiation, whereby it can forward the MKI size received in the SDP offer 'a=crypto' line in the SDP answer. You can enable symmetric MKI globally (using the EnableSymmetricMKI parameter) or per SIP entity (using the IP Profile parameter, IpProfile_EnableSymmetricMKI and for SBC calls, IpProfile_SBCEnforceMKISize). For more information on symmetric MKI, see "Configuring IP Profiles" on page 387.

You can configure the enforcement policy of SRTP, using the EnableMediaSecurity parameter for Gateway calls and IpProfile_SBCMediaSecurityBehaviour parameter for SBC calls. For example, if negotiation of the cipher suite fails or if incoming calls exclude encryption information, the device can be configured to reject the calls.

You can also enable the device to validate the authentication of packets for SRTP tunneling for RTP and RTCP. This applies only to SRTP-to-SRTP SBC calls and where the endpoints use the same key. This is configured using the 'SRTP Tunneling Authentication for RTP' and 'SRTP Tunneling Authentication for RTCP' parameters.

Notes:

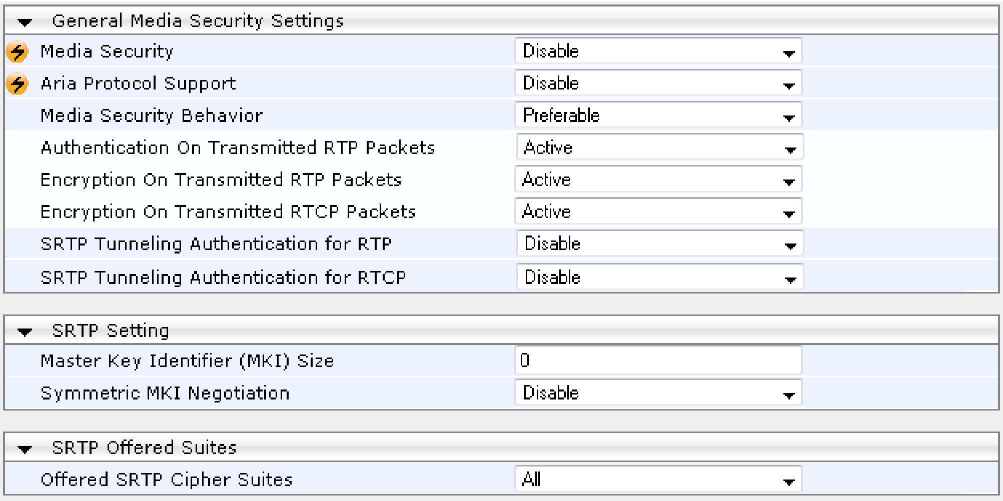
- For a detailed description of the SRTP parameters, see "Configuring IP Profiles" on page 387 and "SRTP Parameters" on page 852.
- When SRTP is used, the channel capacity may be reduced.

The procedure below describes how to configure SRTP through the Web interface.

➢ **To enable and configure SRTP:**

1. Open the Media Security page (**Configuration** tab > **VoIP** menu > **Media** > **Media Security**).

Figure 14-9: Media Security Page



▼ General Media Security Settings

Media Security	Disable
Aria Protocol Support	Disable
Media Security Behavior	Preferable
Authentication On Transmitted RTP Packets	Active
Encryption On Transmitted RTP Packets	Active
Encryption On Transmitted RTCP Packets	Active
SRTP Tunneling Authentication for RTP	Disable
SRTP Tunneling Authentication for RTCP	Disable

▼ SRTP Setting

Master Key Identifier (MKI) Size	0
Symmetric MKI Negotiation	Disable

▼ SRTP Offered Suites

Offered SRTP Cipher Suites	All

2. From the 'Media Security' drop-down list (EnableMediaSecurity), select **Enable** to enable SRTP.
3. Configure the other SRTP parameters as required.
4. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

14.7.1 SRTP using DTLS Protocol

For SBC calls, you can configure the device to use the Datagram Transport Layer Security (DTLS) protocol to secure UDP-based media streams (according to RFC 5763 and 5764) for specific SIP entities, using IP Profiles. DTLS allows datagram-based applications to communicate in a way that is designed to prevent eavesdropping, tampering or message forgery. The DTLS protocol is based on the stream-oriented TLS protocol, providing similar security. The device can therefore, interwork in mixed environments where one network may require DTLS and the other may require Session Description Protocol Security Descriptions (SDES) or even non-secure RTP. The device supports DTLS negotiation for RTP-to-SRTP and SRTP-to-SRTP calls.

In contrast to SDES, DTLS key encryption is done over the media channel (UDP), not signaling. Thus, DTLS-SRTP is generally known as "secured key exchange over media". DTLS is similar to TLS, but runs over UDP, whereas TLS is over TCP. Before the DTLS handshake, the peers exchange DTLS parameters (fingerprint and setup) and algorithm types in the SDP body of the SIP messages exchanged for establishing the call (INVITE request and response). The peers participate in a DTLS handshake during which they exchange certificates. These certificates are used to derive a symmetric key, which is used to encrypt data (SRTP) flow between the peers. A hash value calculated over the certificate is transported in the SDP using the 'a=fingerprint' attribute. At the end of the handshake, each side verifies that the certificate it received from the other side fits the fingerprint from the SDP. To indicate DTLS support, the SDP offer/answer of the SIP message uses the 'a=setup' attribute. The 'a=setup:actpass' attribute value is used in the SDP offer by the device. This indicates that the device is willing to be either a client ('act') or a server ('pass') in the handshake. The 'a=setup:active' attribute value is used in the SDP answer by the device. This means that the device wishes to be the client ('active') in the handshake.

```
a=setup:actpass
a=fingerprint: SHA-1
\4A:AD:B9:B1:3F:82:18:3B:54:02:12:DF:3E:5D:49:6B:19:E5:7C:AB
```

DTLS cipher suite reuses the TLS cipher suite. The DTLS handshake is done for every new call configured for DTLS. In other words, unlike TLS where the connection remains "open" for future calls, a new DTLS connection is required for every new call. Note that the entire authentication and key exchange for securing the media traffic is handled in the media path through DTLS. The signaling path is used only to verify the peers' certificate fingerprints. DTLS messages are multiplexed onto the same ports that are used for the media.

➢ To configure DTLS:

1. In the TLS Context table (see "Configuring TLS Certificate Contexts" on page 107), configure a TLS Context for DTLS.
2. Open the IP Group table (see "Configuring IP Groups" on page 340) and for the IP Group associated with the SIP entity, assign it the TLS Context for DTLS, using the 'DTLS Context' parameter (IPGroup_DTLSContext).
3. Open the IP Profile table (see "Configuring IP Profiles" on page 387) and for the IP Profile associated with the SIP entity, configure the following:
   - Configure the 'SBC Media Security Mode' parameter (IPProfile_SBCMediaSecurityBehavior) to **SRTP** or **Both**.
   - Configure the 'Media Security Method' parameter (IPProfile_SBCMediaSecurityMethod) to **DTLS**.
   - Configure the 'RTCP Mux' parameter (IpProfile_SBCRTCPMux) to **Supported**. Multiplexing is required as the DTLS handshake is done for the port used for RTP and thus, RTCP and RTP must be multiplexed onto the same port.
   - Configure the ini file parameter, SbcDtlsMtu (or CLI command configure voip > sbc general-setting > sbc-dtls-mtu) to define the maximum transmission unit (MTU) size for the DTLS handshake.

Notes:

- The 'Cipher Server' parameter must be configured to "ALL".
- The device does not support forwarding of DTLS transparently between endpoints.

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15 Services

This section describes configuration for various supported services.

15.1 DHCP Server Functionality

The device can serve as a Dynamic Host Configuration Protocol (DHCP) server that assigns and manages IP addresses from a user-defined address pool for DHCP clients. The DHCP server can also be configured to supply additional information to the requesting client such as the IP address of the TFTP server, DNS server, NTP server, and default router (gateway). The DHCP server functionality complies with IETF RFC 2131 and RFC 2132.

The DHCP server can service up to 800 DHCP clients. The DHCP clients are typically IP phones that are connected to the device's LAN port.

The DHCP server is activated when you configure a valid entry in the DHCP Servers table (see "Configuring the DHCP Server" on page 207) and associate it with an active IP network interface (listed in the Interface table). When an IP phone on the LAN requests an IP address, the DHCP server allocates one from the address pool. In scenarios of duplicated IP addresses on the LAN (i.e., an unauthorized network device using one of the IP addresses of the DHCP address pool), the DHCP server detects this condition using an Address Resolution Protocol (ARP) request and temporarily blacklists the duplicated address.

You can also configure the DHCP server to respond **only** to DHCPDiscover requests from DHCP clients that contain a specific value for Option 60 (Vendor Class Identification). For more information, see "Configuring the Vendor Class Identifier" on page 212.

15.1.1 Configuring the DHCP Server

The DHCP Servers table lets you configure the device's DHCP server. The DHCP Server table configures the DHCP server implementation. This includes configuring the DHCP IP address pool from where IP addresses are allocated to requesting DHCP clients, as well as configuring other information such as IP addresses of the DNS server, NTP server, default router (gateway), and SIP proxy server. The DHCP server sends the information in DHCP Options. The table below lists the DHCP Options that the DHCP server sends to the DHCP client and which are configurable in the DHCP Servers table.

Table 15-1: Configurable DHCP Options in DHCP Servers Table

DHCP Option Code	DHCP Option Name
Option 53	DHCP Message Type
Option 54	DHCP Server Identifier
Option 51	IP Address Lease Time
Option 1	Subnet Mask
Option 3	Router
Option 6	Domain Name Server
Option 44	NetBIOS Name Server
Option 46	NetBIOS Node Type
Option 42	Network Time Protocol Server
Option 2	Time Offset
Option 66	TFTP Server Name
Option 67	Boot file Name

DHCP Option Code	DHCP Option Name
Option 120	SIP Server

Once you have configured the DHCP server, you can configure the following:

- DHCP Vendor Class Identifier names (DHCP Option 60) - see "Configuring the Vendor Class Identifier" on page 212
- Additional DHCP Options - see "Configuring Additional DHCP Options" on page 213
- Static IP addresses for DHCP clients - see "Configuring Static IP Addresses for DHCP Clients" on page 215

> **Note:** If you configure additional DHCP Options in the DHCP Option table, they override the default ones, which are configured in the DHCP Servers table. For example, if you configure Option 67 in the DHCP Option table, the device uses the value configured in the DHCP Option table instead of the value configured in the DHCP Servers table.

To view and delete currently serviced DHCP clients, see "Viewing and Deleting DHCP Clients" on page 216.

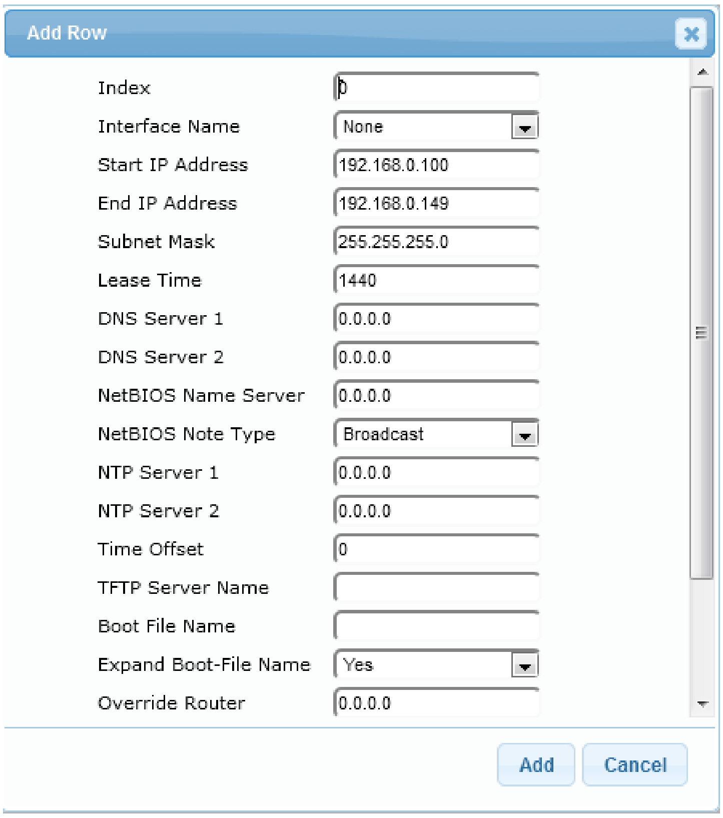
The following procedure describes how to configure the DHCP server through the Web interface. You can also configure it through ini file (DhcpServer) or CLI (configure voip > dhcp server <index>).

➢ **To configure the device's DHCP server:**

1. Open the DHCP Servers page (**Configuration** tab > **VoIP** menu > **Services** > **DHCP Severs**).

2. Click **Add**; the following dialog box appears:

Figure 15-1: DHCP Servers Table - Add Row Dialog Box



Index	0
Interface Name	None
Start IP Address	192.168.0.100
End IP Address	192.168.0.149
Subnet Mask	255.255.255.0
Lease Time	1440
DNS Server 1	0.0.0.0
DNS Server 2	0.0.0.0
NetBIOS Name Server	0.0.0.0
NetBIOS Note Type	Broadcast
NTP Server 1	0.0.0.0
NTP Server 2	0.0.0.0
Time Offset	0
TFTP Server Name	
Boot File Name	
Expand Boot-File Name	Yes
Override Router	0.0.0.0

3. Configure a DHCP server according to the parameters described in the table below.
4. Click **Add**.

Table 15-2: DHCP Servers Table Parameter Descriptions

Parameter	Description
Index `dhcp server <index>`	Defines an index number for the new table row. **Notes:** ▪ Each row must be configured with a unique index. ▪ Currently, only one index row can be configured.
Interface Name `network-if` [DhcpServer_InterfaceName]	Associates an IP interface on which the DHCP server operates. The IP interfaces are configured in the Interface table (see "Configuring IP Network Interfaces" on page 133). By default, no value is defined.
Start IP Address `start-address` [DhcpServer_StartIPAddress]	Defines the starting IP address (IPv4 address in dotted-decimal format) of the IP address pool range used by the DHCP server to allocate addresses. The default value is 192.168.0.100. **Note:** The IP address must belong to the same subnet as the associated interface's IP address.

Parameter	Description
End IP Address `end-address` [DhcpServer_EndIPAddress]	Defines the ending IP address (IPv4 address in dotted-decimal format) of the IP address pool range used by the DHCP server to allocate addresses. The default value is 192.168.0.149. **Note:** The IP address must belong to the same subnet as the associated interface's IP address and must be "greater or equal" to the starting IP address defined in 'Start IP Address'.
Subnet Mask `subnet-mask` [DhcpServer_SubnetMask]	Defines the subnet mask (for IPv4 addresses) for the DHCP client. The value is sent in DHCP Option 1 (Subnet Mask). The default value is 0.0.0.0. **Note:** The value must be "narrower" or equal to the subnet mask of the associated interface's IP address. If set to "0.0.0.0", the subnet mask of the associated interface is used.
Lease Time `lease-time` [DhcpServer_LeaseTime]	Defines the duration (in minutes) of the lease time to a DHCP client for using an assigned IP address. The client needs to request a new address before this time expires. The value is sent in DHCP Option 51 (IP Address Lease Time). The valid value range is 0 to 214,7483,647. The default is 1440. When set to 0, the lease time is infinite.
DNS Server 1 `dns-server-1` [DhcpServer_DNSServer1]	Defines the IP address (IPv4) of the primary DNS server that the DHCP server assigns to the DHCP client. The value is sent in DHCP Option 6 (Domain Name Server). The default value is 0.0.0.0.
DNS Server 2 `dns-server-2` [DhcpServer_DNSServer2]	Defines the IP address (IPv4) of the secondary DNS server that the DHCP server assigns to the DHCP client. The value is sent in DHCP Option 6 (Domain Name Server).
NetBIOS Name Server `netbios-server` [DhcpServer_NetbiosNameServer]	Defines the IP address (IPv4) of the NetBIOS WINS server that is available to a Microsoft DHCP client. The value is sent in DHCP Option 44 (NetBIOS Name Server). The default value is 0.0.0.0.
NetBIOS Node Type `netbios-node-type` [DhcpServer_NetbiosNodeType]	Defines the node type of the NetBIOS WINS server for a Microsoft DHCP client. The value is sent in DHCP Option 46 (NetBIOS Node Type). ▪ **[0]** Broadcast (default) ▪ **[1]** peer-to-peer ▪ **[4]** Mixed ▪ **[8]** Hybrid
NTP Server 1 `ntp-server-1` [DhcpServer_NTPServer1]	Defines the IP address (IPv4) of the primary NTP server that the DHCP server assigns to the DHCP client. The value is sent in DHCP Option 42 (Network Time Protocol Server). The default value is 0.0.0.0.
NTP Server 2 `ntp-server-2` [DhcpServer_NTPServer2]	Defines the IP address (IPv4) of the secondary NTP server that the DHCP server assigns to the DHCP client. The value is sent in DHCP Option 42 (Network Time Protocol Server). The default value is 0.0.0.0.

Parameter	Description
Time Offset `time-offset` [DhcpServer_TimeOffset]	Defines the Greenwich Mean Time (GMT) offset (in seconds) that the DHCP server assigns to the DHCP client. The value is sent in DHCP Option 2 (Time Offset). The valid range is -43200 to 43200. The default is 0.
TFTP Server `tftp-server-name` [DhcpServer_TftpServer]	Defines the IP address or name of the TFTP server that the DHCP server assigns to the DHCP client. The TFTP server typically stores the boot file image, defined in the 'Boot file name' parameter (see below). The value is sent in DHCP Option 66 (TFTP Server Name). The valid value is a string of up to 80 characters. By default, no value is defined.
Boot file name `boot-file-name` [DhcpServer_BootFileName]	Defines the name of the boot file image for the DHCP client. The boot file stores the boot image for the client. The boot image is typically the operating system the client uses to load (downloaded from a boot server). The value is sent in DHCP Option 67 (Bootfile Name). To define the server storing the file, use the 'TFTP Server' parameter (see above). The valid value is a string of up to 256 characters. By default, no value is defined. The name can also include the following case-sensitive placeholder strings that are replaced with actual values if the 'Expand Boot-file Name' parameter is set to **Yes**: ▪ <MAC>: Replaced by the MAC address of the client (e.g., *boot_<MAC>.ini*). The MAC address is obtained in the client's DHCP request. ▪ <IP>: Replaced by the IP address assigned by the DHCP server to the client.
Expand Boot-file Name `expand-boot-file-name` [DhcpServer_ExpandBootfileName]	Enables the use of the placeholders in the boot file name, defined in the 'Boot file name' parameter. ▪ **[0]** No ▪ **[1]** Yes (default)
Override Router `override-router-address` [DhcpServer_OverrideRouter]	Defines the IP address (IPv4 in dotted-decimal notation) of the default router that the DHCP server assigns the DHCP client. The value is sent in DHCP Option 3 (Router). The default value is 0.0.0.0. If not specified (empty or “0.0.0.0”), the IP address of the default gateway configured in the Interface table for the IP network interface that you associated with the DHCP server (see the 'Interface Name' parameter above) is used.
SIP Server `sip-server` [DhcpServer_SipServer]	Defines the IP address or DNS name of the SIP server that the DHCP server assigns the DHCP client. The client uses this SIP server for its outbound SIP requests. The value is sent in DHCP Option 120 (SIP Server). After defining the parameter, use the 'SIP server type' parameter (see below) to define the type of address (FQDN or IP address). The valid value is a string of up to 256 characters. The default is 0.0.0.0.

Parameter	Description
SIP server type `sip-server-type` [DhcpServer_SipServerType]	Defines the type of SIP server address. The actual address is defined in the 'SIP server' parameter (see above). Encoding is done per SIP Server Type, as defined in RFC 3361. ▪ **[0]** DNS names = (Default) The 'SIP server' parameter is configured with an FQDN of the SIP server. ▪ **[1]** IP address = The 'SIP server' parameter configured with an IP address of the SIP server.

15.1.2 Configuring the Vendor Class Identifier

The DHCP Vendor Class table lets you configure up to 10 Vendor Class Identifier (VCI) names (DHCP Option 60). When the table is configured, the device's DHCP server responds only to DHCPDiscover requests that contain Option 60 and that match one of the DHCP VCIs configured in the table. If you have not configured any entries in the table, the DHCP server responds to all DHCPDiscover requests, regardless of the VCI.

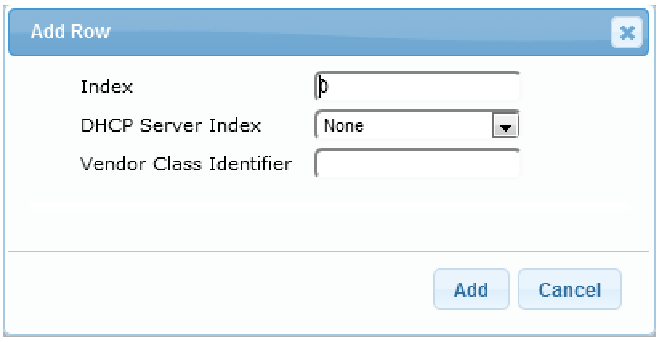
The VCI is a string that identifies the vendor and functionality of a DHCP client to the DHCP server. For example, Option 60 can show the unique type of hardware (e.g., "AudioCodes 440HD IP Phone") or firmware of the DHCP client. The DHCP server can then differentiate between DHCP clients and process their requests accordingly.

The following procedure describes how to configure the DHCP VCIs through the Web interface. You can also configure it through ini file (DhcpVendorClass) or CLI (configure voip > dhcp vendor-class).

➢ **To configure DHCP Vendor Class Identifiers:**

1. Open the DHCP Servers table (**Configuration** tab > **VoIP** menu > **Services** > **DHCP Severs**).
2. In the table, select the row of the desired DHCP server for which you want to configure VCIs, and then click the **DHCP Vendor Class Table** link, located below the table; the DHCP Vendor Class table opens.
3. Click **Add**; the following dialog box appears:

Figure 15-2: DHCP Vendor Class Table - Add Row Dialog Box



Index: 0
DHCP Server Index: None
Vendor Class Identifier:

4. Configure a VCI for the DHCP server according to the parameters described in the table below.
5. Click **Add**.

Table 15-3: DHCP Vendor Class Table Parameter Descriptions

Parameter	Description
Index `dhcp vendor-class <index>` [DhcpVendorClass_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
DHCP Server Index `dhcp-server-number` [DhcpVendorClass_DhcpServerIndex]	Associates the VCI table entry with a DHCP server that you configured in "Configuring the DHCP Server" on page 207. **Note:** Currently, only one DHCP server (Index 0) can be configured and therefore, the parameter is always set at 0.
Vendor Class Identifier `vendor-class` [DhcpVendorClass_VendorClassId]	Defines the value of the VCI DHCP Option 60. The valid value is a string of up to 80 characters. By default, no value is defined.

15.1.3 Configuring Additional DHCP Options

The DHCP Option table lets you configure up to 10 additional DHCP Options that the DHCP server can use to service the DHCP client. These DHCP Options are included in the DHCPOffer response sent by the DHCP server.

The following procedure describes how to configure DHCP Options through the Web interface. You can also configure it through ini file (DhcpOption) or CLI (configure voip > dhcp option).

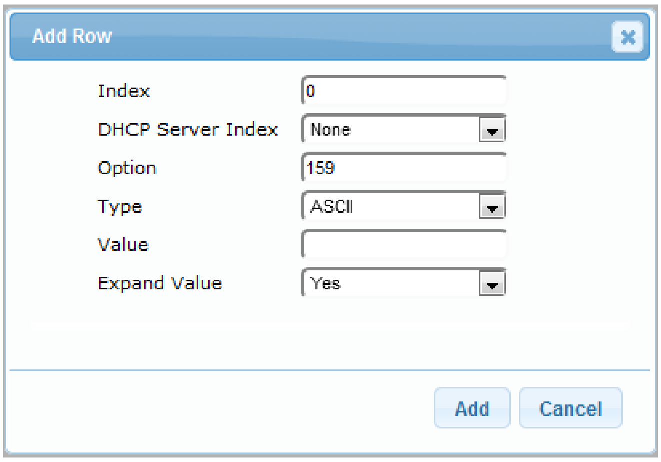
> **Note:** The additional DHCP Options configured in the DHCP Option table override the default ones, which are configured in the DHCP Servers table. In other words, if you configure Option 67 in the DHCP Option table, the device uses the value configured in the DHCP Option table instead of the value configured in the DHCP Servers table.

➢ **To configure DHCP Options:**

1. Open the DHCP Servers table (**Configuration** tab > **VoIP** menu > **Services** > **DHCP Severs**).
2. In the table, select the row of the desired DHCP server for which you want to configure additional DHCP Options, and then click the **DHCP Option Table** link, located below the table; the DHCP Option table opens.

3. Click **Add**; the following dialog box appears:

Figure 15-3: DHCP Option Table - Add Row Dialog Box



Add Row

Index: 0

DHCP Server Index: None

Option: 159

Type: ASCII

Value:

Expand Value: Yes

4. Configure additional DHCP Options for the DHCP server according to the parameters described in the table below.
5. Click **Submit**.

Table 15-4: DHCP Option Table Parameter Descriptions

Parameter	Description
Index `dhcp option` [DhcpOption_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
DHCP Server Index `dhcp-server-number` [DhcpOption_DhcpServerIndex]	Associates the DHCP Option table entry with a DHCP server that you configured in "Configuring the DHCP Server" on page 207. **Note:** Currently, only one DHCP server (Index 0) can be configured and therefore, the parameter is always set at 0.
Option `option` [DhcpOption_Option]	Defines the code of the DHCP Option. The valid value is 1 to 254. The default is 159. For example, for DHCP Option 150 (Cisco proprietary for defining multiple TFTP server IP addresses), enter the value 150.
Type `type` [DhcpOption_Type]	Defines the format (type) of the DHCP Option value that is configured in the 'Value' parameter (see below). ▪ **[0]** ASCII = (Default) Plain-text string (e.g., when the value is a domain name). ▪ **[1]** IP address = IPv4 address. ▪ **[2]** Hexadecimal = Hexadecimal-encoded string. For example, if you set the 'Value' parameter to "company.com", you need to set the 'Type' parameter to **ASCII**.

Parameter	Description
Value `value` [DhcpOption_Value]	Defines the value of the DHCP Option. For example, if you are using Option 66, the parameter is used for specifying the TFTP provisioning server (e.g., http://192.168.3.155:5000/provisioning/). The valid value is a string of up to 256 characters. By default, no value is defined. For IP addresses, the value can be one or more IPv4 addresses, each separated by a comma (e.g., 192.168.10.5,192.168.10.20). For hexadecimal values, the value is a hexadecimal string (e.g., c0a80a05). You can also configure the parameter with case-sensitive placeholder strings that are replaced with actual values if the 'Expand Value' parameter (see below) is set to **Yes**: ▪ <MAC>: Replaced by the MAC address of the client. The MAC address is obtained from the client's DHCP request. For example, the parameter can be set to: http://192.168.3.155:5000/provisioning/cfg_<MAC>.txt ▪ <IP>: Replaced by the IP address assigned by the DHCP server to the client. For example, the parameter can be set to: http://192.168.3.155:5000/provisioning/cfg_<IP>.txt
Expand Value `expand-value` [DhcpOption_ExpandValue]	Enables the use of the special placeholder strings, "<MAC>" and "<IP>" for configuring the 'Value' parameter (see above). ▪ **[0]** No ▪ **[1]** Yes (default) **Note:** The parameter is applicable only to values of type ASCII (see the 'Type' parameter above.

15.1.4 Configuring Static IP Addresses for DHCP Clients

The DHCP Static IP table lets you configure up to 100 DHCP clients with static IP addresses. The static IP address is a "reserved" IP address for a specified DHCP client defined by MAC address. In other words, instead of assigning the DHCP client with a different IP address upon each IP address lease renewal request, the DHCP server assigns the client the same IP address. For DHCP clients that are not listed in the table, the DHCP server assigns a random IP address from its address pool, as in normal operation.

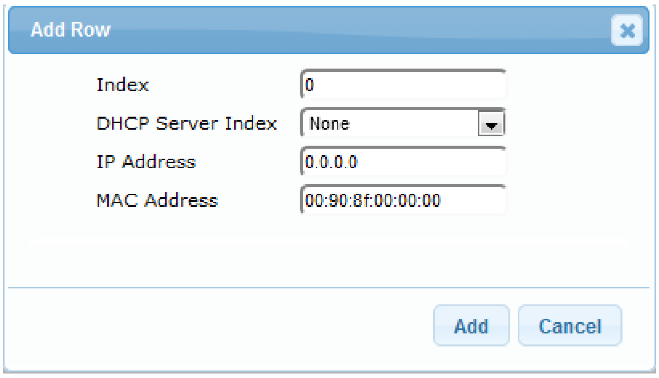
The following procedure describes how to configure static IP addresses for DHCP clients through the Web interface. You can also configure it through ini file (DhcpStaticIP) or CLI (configure voip > dhcp static-ip <index>).

➢ **To configure static IP addresses for DHCP clients:**

1. Open the DHCP Servers table (**Configuration** tab > **VoIP** menu > **Services** > **DHCP Severs**).
2. In the table, select the row of the desired DHCP server for which you want to configure static IP addresses for DHCP clients, and then click the **DHCP Static IP Table** link, located below the table; the DHCP Static IP table opens.

3. Click **Add**; the following dialog box appears:

Figure 15-4: DHCP Static IP Table - Add Row Dialog Box



Index: 0

DHCP Server Index: None

IP Address: 0.0.0.0

MAC Address: 00:90:8f:00:00:00

4. Configure a static IP address for a specific DHCP client according to the parameters described in the table below.
5. Click **Add**.

Table 15-5: DHCP Static IP Table Parameter Descriptions

Parameter	Description
Index `dhcp static-ip <index>` [DhcpStaticIP_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
DHCP Server Index `dhcp-server-number` [DhcpStaticIP_DhcpServerIndex]	Associates the DHCP Static IP table entry with a DHCP server that you configured in "Configuring the DHCP Server" on page 207. **Note:** Currently, only one DHCP server (Index 0) can be configured and therefore, the parameter is always set at 0.
IP Address `ip-address` [DhcpStaticIP_IPAddress]	Defines the "reserved", static IP address (IPv4) to assign the DHCP client. The default is 0.0.0.0.
MAC Address `mac-address` [DhcpStaticIP_MACAddress]	Defines the DHCP client by MAC address (in hexadecimal format). The valid value is a string of up to 20 characters. The format includes six groups of two hexadecimal digits, each separated by a colon. The default MAC address is 00:90:8f:00:00:00.

15.1.5 Viewing and Deleting DHCP Clients

The DHCP Clients table lets you view all currently serviced DHCP clients by the DHCP server. The table also lets you delete DHCP clients. If you delete a client, the DHCP server ends the lease of the IP address to the client and the IP address becomes available for allocation by the DHCP server to another client.

The following procedure describes how to view DHCP clients through the Web interface. You can also view this through CLI:

- To view DHCP clients:

```
# show voip dhcp clients
```

- To view DHCP clients according to IP address:

```
# show voip dhcp ip
```

- To view DHCP clients according to MAC address:

```
# show voip dhcp mac
```

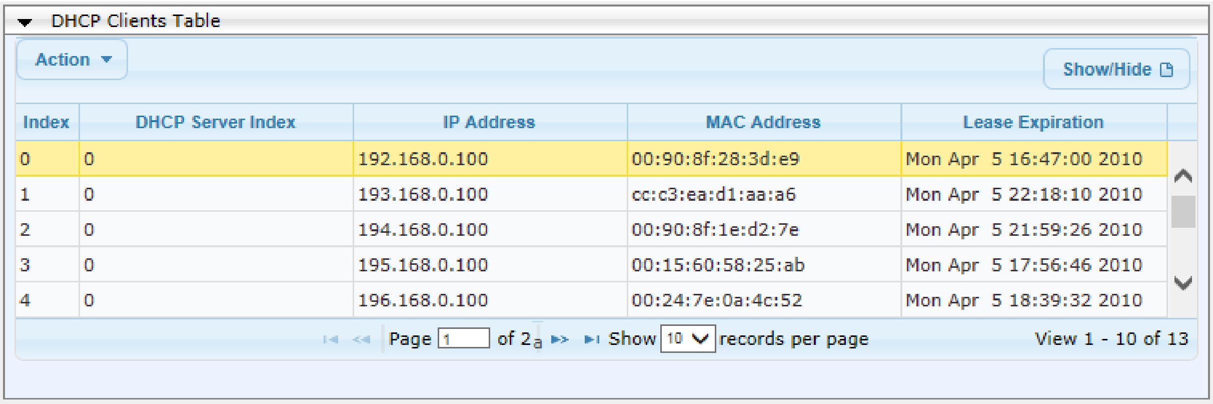
- To view DHCP clients that have been blacklisted from DHCP implementation (due to duplicated IP addresses in the network, where another device is using the same IP address as the one assigned to the client):

```
# show voip dhcp black-list
```

➢ **To view or delete DHCP clients:**

1. Open the DHCP Servers table (**Configuration** tab > **VoIP** menu > **Services** > **DHCP Severs**).
2. In the table, select the row of the desired DHCP server for which you want to view DHCP clients, and then click the **DHCP Clients Table** link, located below the table; the DHCP Clients table opens:

Figure 15-5: DHCP Clients Table



Index	DHCP Server Index	IP Address	MAC Address	Lease Expiration
0	0	192.168.0.100	00:90:8f:28:3d:e9	Mon Apr 5 16:47:00 2010
1	0	193.168.0.100	cc:c3:ea:d1:aa:a6	Mon Apr 5 22:18:10 2010
2	0	194.168.0.100	00:90:8f:1e:d2:7e	Mon Apr 5 21:59:26 2010
3	0	195.168.0.100	00:15:60:58:25:ab	Mon Apr 5 17:56:46 2010
4	0	196.168.0.100	00:24:7e:0a:4c:52	Mon Apr 5 18:39:32 2010

Page 1 of 2 Show 10 records per page View 1 - 10 of 13

The table displays the following per client:

- **Index:** Table index number.
- **DHCP Server Index:** The index number of the configured DHCP server scope in the DHCP Server table (see "Configuring the DHCP Server" on page 207) with which the client is associated.
- **IP Address:** IP address assigned to the DHCP client by the DHCP server.
- **MAC Address:** MAC address of the DHCP client.
- **Lease Expiration:** Date on which the lease of the DHCP client's IP address obtained from the DHCP server expires.

3. To delete a client:
   a. Select the table row index of the DHCP client that you want to delete.
   b. Click the **Action** button, and then from the drop-down menu, choose **Delete**; a confirmation message appears.
   c. Click **OK** to confirm deletion.

15.2 SIP-based Media Recording

The device can record SIP-based media call sessions traversing it for Gateway and SBC calls. The media recording support is in accordance with the Session Recording Protocol (siprec), which describes architectures for deploying session recording solutions and specifies requirements for extensions to SIP that will manage delivery of RTP media to a recording device. The siprec protocol is based on RFC 6341 (Use Cases and Requirements for SIP-Based Media Recording), Session Recording Protocol (draft-ietf-siprec-protocol-02), and Architecture o(draft-ietf-siprec-architecture-03).

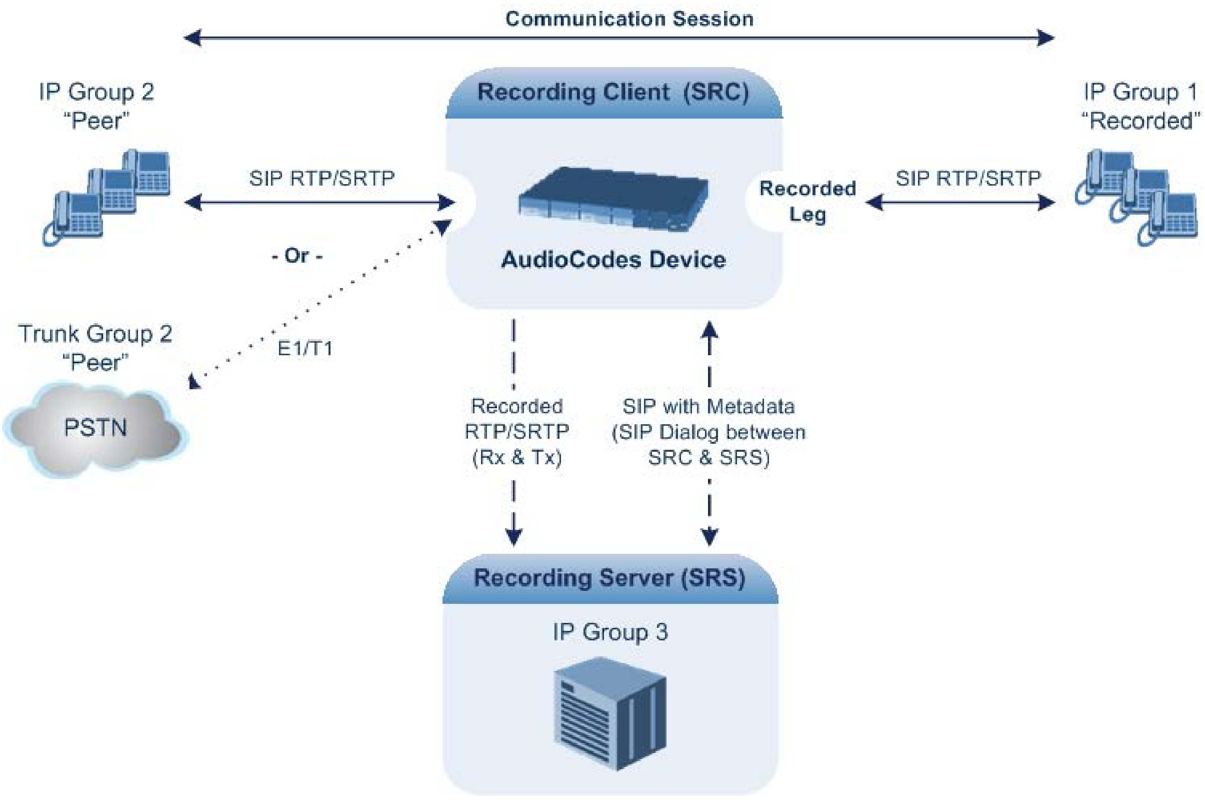
Warning for Deployments in France: The device supports SIP-based Media Recording (SIPREC) according to RFC 6341. As such, you must adhere to the Commission Nationale Informatique et Liberté's (CNIL) directive (https://www.cnil.fr/en/rights-and-obligations) and be aware that article R226-15 applies penalties to the malicious interception, diversion, use or disclosure of correspondence sent, transmitted or received by means of telecommunication, or the setting up of a device designed to produce such interceptions.

Notes:

- The SIP-based Media Recording feature is available only if the device is installed with a Software License Key that includes this feature. For installing a Software License Key, see "Software License Key" on page 668. The Software License Key also specifies the maximum number of supported SIP recording sessions.
- For the maximum number of concurrent sessions that the device can record, contact your AudioCodes sales representative.

Session recording is a critical requirement in many business communications environments such as call centers and financial trading floors. In some of these environments, all calls must be recorded for regulatory and compliance reasons. In others, calls may be recorded for quality control or business analytics. Recording is typically performed by sending a copy of the session media to the recording devices.

The siprec protocol specifies the use of SIP, SDP, and RTP to establish a Recording Session (RS) from the Session Recording Client (SRC), which is on the path of the Communication Session (CS), to a Session Recording Server (SRS) at the recording equipment. The device functions as the SRC, sending recording sessions to a third-party SRS, as shown in the figure below.



The device can record calls between two IP Groups, or between an IP Group and a Trunk Group for Gateway calls. The type of calls to record can be specified by source and/or destination prefix number or SIP Request-URI, as well as by call initiator. The side ("leg") on which the recording is done must be specified. Specifying the leg is important as it determines the various call media attributes of the recorded RTP (or SRTP) such as coder type.

The device can also record SRTP calls and send it to the SRS in SRTP. In such scenarios, the SRTP is used on the IP leg for Gateway calls, or on one of the IP legs for SBC calls. For an SBC RTP-SRTP session, the recorded IP Group in the SIP Recording table must be set to the RTP leg if recording is required to be RTP, or set to the SRTP leg if recording is required to be SRTP.

For SBC calls, the device can also be located between an SRS and an SRC and act as an RTP-SRTP translator. In such a setup, the device receives SIP recording sessions (as a server) from the SRC and translates SRTP media to RTP, or vice versa, and then forwards the recording to the SRS in the translated media format.

The device initiates a recording session by sending an INVITE message to the SRS when the recorded call is connected. The SIP From header contains the identity of the SRC and the To header contains the identity of the SRS. The SDP in the INVITE contains:

- Two 'm=' lines that represent the two RTP/SRTP streams (Rx and Tx).
- Two 'a=label:' lines that identify the streams.
- XML body (also referred to as metadata) that provides information on the participants of the call session:
    - <group id>: Logging Session ID (displayed as [SID:nnnnn] in Syslog), converted from decimal to hex. This number remains the same even if the call is forwarded or transferred. This is important for recorded calls.
    - <session id>: Originally recorded Call-ID, converted from decimal to hex.
    - <group-ref>: same as <group id>.
    - <participant id>: SIP From / To user.
    - <nameID aor>: From/To user@host.
    - <send> and <recv>: ID's for the RTP/SRTP streams in hex - bits 0-31 are the same as group, bits 32-47 are the RTP port.
    - <stream id>: Same as <send> for each participant.
    - <label>: 1 and 2 (same as in the SDP's 'a=label:' line).

The SRS can respond with 'a=recvonly' for immediate recording or 'a=inactive' if recording is not yet needed, and send re-INVITE at any later time with the desired RTP/SRTP mode change. If a re-INVITE is received in the original call (e.g. when a call is on hold), the device sends another re-INVITE with two 'm=' lines to the SRS with the updated RTP/SRTP data. If the recorded leg uses SRTP, the device can send the media streams to the SRS as SRTP; otherwise, the media streams are sent as RTP to the SRS.

Below is an example of an INVITE sent by the device to an SRS:

```
INVITE sip:VSRP@1.9.64.253 SIP/2.0
Via: SIP/2.0/UDP 192.168.241.44:5060;branch=z9hG4bKac505782914
Max-Forwards: 10
From: <sip:192.168.241.44>;tag=1c505764207
To: <sip:VSRP@1.9.64.253>
Call-ID: 505763097241201011157@192.168.241.44
CSeq: 1 INVITE
Contact: <sip:192.168.241.44:5060>;src
Supported: replaces,resource-priority
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUB
SCRIBE,UPDATE
Require: siprec
User-Agent: Mediant /v.6.80A.227.005
Content-Type: multipart/mixed;boundary=boundary_ac1fffff85b
Content-Length: 1832
```

```
--boundary_ac1fffff85b
Content-Type: application/sdp
v=0
o=AudiocodesGW 921244928 921244893 IN IP4 10.33.8.70
s=SBC-Call
c=IN IP4 10.33.8.70
t=0 0
m=audio 6020 RTP/AVP 8 96
c=IN IP4 10.33.8.70
a=ptime:20
a=sendonly
a=label:1
a=rtpmap:8 PCMA/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
m=audio 6030 RTP/AVP 8 96
c=IN IP4 10.33.8.70
a=ptime:20
a=sendonly
a=label:2
a=rtpmap:8 PCMA/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
--boundary_ac1fffff85b
Content-Type: application/rs-metadata
Content-Disposition: recording-session
<?xml version="1.0" encoding="UTF-8"?>
<recording xmlns='urn:ietf:params:xml:ns:recording'>
  <datamode>complete</datamode>
  <group id="00000000-0000-0000-0000-00003a36c4e3">
    <associate-time>2010-01-24T01:11:57Z</associate-time>
  </group>
  <session id="0000-0000-0000-0000-00000000d0d71a52">
    <group-ref>00000000-0000-0000-0000-00003a36c4e3</group-ref>
    <start-time>2010-01-24T01:11:57Z</start-time>
    <ac:AvayaUCID
xmlns="urn:ietf:params:xml:ns:Avaya">FA080030C4E34B5B9E59</ac:Avay
aUCID>
  </session>
  <participant id="1056" session="0000-0000-0000-0000-
00000000d0d71a52">
    <nameID aor="1056@192.168.241.20"></nameID>
    <associate-time>2010-01-24T01:11:57Z</associate-time>
    <send>00000000-0000-0000-0000-1CF23A36C4E3</send>
    <recv>00000000-0000-0000-0000-BF583A36C4E3</recv>
  </participant>
    <participant id="182052092" session="0000-0000-0000-0000-
00000000d0d71a52">
    <nameID aor="182052092@voicelab.local"></nameID>
    <associate-time>2010-01-24T01:11:57Z</associate-time>
    <recv>00000000-0000-0000-0000-1CF23A36C4E3</recv>
    <send>00000000-0000-0000-0000-BF583A36C4E3</send>
  </participant>
  <stream id="00000000-0000-0000-0000-1CF23A36C4E3" session="0000-
0000-0000-0000-00000000d0d71a52">
    <label>1</label>
  </stream>
  <stream id="00000000-0000-0000-0000-BF583A36C4E3" session="0000-
0000-0000-0000-00000000d0d71a52">
    <label>2</label>
  </stream>
</recording>
--boundary_ac1fffff85b—
```

15.2.1 Enabling SIP-based Media Recording

The following procedure describes how to enable the SIP-based media Recording feature. Once you have enabled this feature, your SIP Recording Routing rules (configured in "Configuring SIP Recording Rules" on page 221) become active.

➢ **To enable SIP-based media recording:**

1. Open the SIP Recording page (**Configuration** tab > **VoIP** menu > **Services** > **SIP Recording**).
2. From the 'SIP Recording Application' drop-down list, select **Enable**.
3. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

15.2.2 Configuring SIP Recording Rules

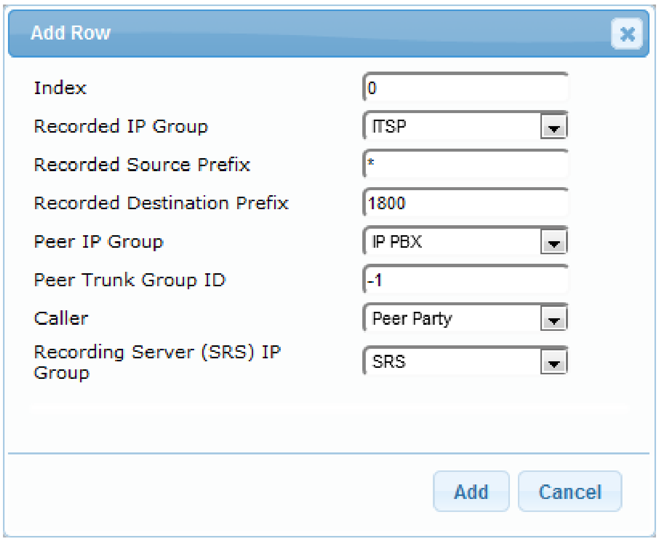
The SIP Recording table lets you configure up to 30 SIP-based media recording rules. A SIP Recording rule defines calls that you want to record. For an overview of this feature, see "SIP-based Media Recording" on page 217.

Note: To configure the device's timestamp format (local or UTC) in SIP messages sent to the SRS, see the SIPRecTimeStamp parameter.

The following procedure describes how to configure SIP Recording Routing rules through the Web interface. You can also configure it through ini file (SIPRecRouting) or CLI (configure voip > services sip-recording sip-rec-routing).

➢ **To configure a SIP Recording Routing rule:**

1. Open the SIP Recording page (**Configuration** tab > **VoIP** menu > **Services** > **SIP Recording**).
2. Click **Add**; the following dialog box appears:



Add Row

Index	0
Recorded IP Group	ITSP
Recorded Source Prefix	*
Recorded Destination Prefix	1800
Peer IP Group	IP PBX
Peer Trunk Group ID	-1
Caller	Peer Party
Recording Server (SRS) IP Group	SRS

The figure above shows a configuration example where the device records calls made by IP Group "ITSP" to IP Group "IP PBX" that have the destination number prefix, "1800". The device records the calls from the leg interfacing with IP Group "IP PBX", sending the recorded media to IP Group "SRS".

3. Configure a SIP recording route according to the parameters described in the table below.
4. Click **Add**, and then save ("burn") your settings to flash memory.

Table 15-6: SIP Recording Table Parameter Descriptions

Parameter	Description
Index [SIPRecRouting_Index]	Defines an index number for the new table record.
Recorded IP Group `recorded-ip-group-name` [SIPRecRouting_RecordedIPGroupName]	Defines the IP Group participating in the call and the recording is done on the leg interfacing with this IP Group. For configuring IP Groups, see "Configuring IP Groups" on page 340. By default, all IP Groups are defined (**Any**). **Note:** For an SBC RTP-SRTP session, the recorded IP Group must be set to the RTP leg if recording is required to be RTP, or set to the SRTP leg if recording is required to be SRTP.
Recorded Source Prefix `recorded-src-prefix` [SIPRecRouting_RecordedSourcePrefix]	Defines calls to record based on source number or URI. By default, all source numbers or URIs are defined (*****).
Recorded Destination Prefix `recorded-dst-prefix` [SIPRecRouting_RecordedDestinationPrefix]	Defines calls to record based on destination number or URI. By default, all destination numbers or URIs are defined (*****).
Peer IP Group `peer-ip-group-name` [SIPRecRouting_PeerIPGroupName]	Defines the peer IP Group that is participating in the call. By default, all IP Groups are defined (**Any**).
Peer Trunk Group ID `peer-trunk-group-id` [SIPRecRouting_PeerTrunkGroupID]	Defines the peer Trunk Group that is participating in the call (applicable only to Gateway calls). For configuring Trunk Groups, see Configuring Trunk Groups on page 433.
Caller `caller` [SIPRecRouting_Caller]	Defines which calls to record according to which party is the caller. ▪ **[0]** Both = (Default) Caller can be peer or recorded side ▪ **[1]** Recorded Party (in Gateway, IP-to-Tel call) ▪ **[2]** Peer Party (in Gateway, Tel-to-IP call)

Parameter	Description
Recording Server (SRS) IP Group `srs-ip-group-name` [SIPRecRouting_SRSIPGroupName]	Defines the IP Group of the recording server (SRS). By default, no value is defined. (**None**). **Note:** The SIP Interface used for communicating with the SRS is according to the SRD assigned to the SRS IP Group (in the IP Group table). If two SIP Interfaces are associated with the SRD - one for "SBC" and one for "GW" – the device uses the "SBC" SIP Interface. If no SBC SIP Interface type is configured, the device uses the “GW” interface.

15.2.3 Configuring SIP User Part for SRS

You can configure the SIP user part of the Request-URI for the recording server (SRS). The device inserts this user part in the SIP To header of the INVITE message sent to the SRS.

➢ **To configure the SIP user part for SRS:**

1. Open the SIP Recording page (**Configuration** tab > **VoIP** menu > **Services** > **SIP Recording**).
2. In the 'Recording Server (SRS) Destination Username' field, enter a user part value (string of up to 50 characters).
3. Click **Submit**, and then save ("burn") your settings to flash memory.

15.2.4 Interworking SIP-based Media Recording with Third-Party Vendors

The device can interwork the SIP-based Media Recording feature with third-party vendors, as described in the following subsections.

15.2.4.1 Genesys

The device's SIP-based media recording can interwork with Genesys' equipment. Genesys sends its proprietary X-Genesys-CallUUID header (which identifies the session) in the first SIP message, typically in the INVITE and the first 18x response. If the device receives a SIP message with Genesys SIP header, it adds the header's information to AudioCodes' proprietary tag in the XML metadata of the SIP INVITE that it sends to the recording server, as shown below:

```
<ac:GenesysUUID
xmlns="urn:ietf:params:xml:ns:Genesys">4BOKLLA3VH66JF112M1CC9VHKS1
4F0KP</ac:GenesysUUID>
```

No configuration is required for this support.

15.2.4.2 Avaya UCID

The device's SIP-based media recording can interwork with Avaya equipment. The Universal Call Identifier (UCID) is Avaya's proprietary call identifier used to correlate call records between different systems and identifies sessions. Avaya generates this in outgoing calls. If the device receives a SIP INVITE from Avaya, it adds the UCID value, received in the User-to-User SIP header to AudioCodes' proprietary tag in the XML metadata of the SIP INVITE that it sends to the recording server. For example, if the received SIP header is:

```
User-to-User: 00FA080019001038F725B3;encoding=hex
```

the device includes the following in the XML metadata:

```
xml metadata:
<ac:AvayaUCID xmlns="urn:ietf:params:xml:ns:Avaya">
FA080019001038F725B3</ac:AvayaUCID>
```

Note: For calls sent from the device to Avaya equipment, the device can generate the Avaya UCID, if required. To configure this support, use the following parameters:

- 'UUI Format' in the IP Group table - enables Avaya support.
- 'Network Node ID' - defines the Network Node Identifier of the device for Avaya UCID.

15.3 RADIUS-based Services

The device supports Remote Authentication Dial In User Service (RADIUS), by acting as a RADIUS client. You can use RADIUS for the following:

- Authentication and authorization of management users (login username and password) to gain access to the device's management interface.
- Accounting where the device sends accounting data of SIP calls as call detail records (CDR) to a RADIUS Accounting server (for third-party billing purposes).

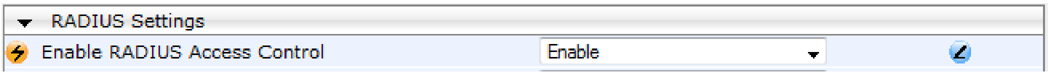
15.3.1 Enabling RADIUS Services

Before you can implement any RADIUS services, you must enable the RADIUS feature, as described in the procedure below.

➢ **To enable RADIUS:**

1. Open the Authentication Settings page (**Configuration** tab > **System** menu > **Management** > **Authentication Settings**).

Figure 15-6: Enabling RADIUS



RADIUS Settings

Enable RADIUS Access Control: Enable

2. From the 'Enable RADIUS Access Control' drop-down list, select **Enable.**
3. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

15.3.2 Configuring RADIUS Servers

The RADIUS Servers table lets you configure up to three RADIUS servers. The RADIUS servers can be used for RADIUS-based management-user login authentication and/or RADIUS-based accounting (sending of SIP CDRs to the RADIUS server).

When multiple RADIUS servers are configured, RADIUS server redundancy can be implemented. When the primary RADIUS server is down, the device sends a RADIUS request twice (one retransmission) and if both fail (i.e., no response), the device considers the server as down and attempts to send requests to the next server. The device continues sending RADIUS requests to the redundant RADIUS server even if the primary server returns to service later on. However, if a device reset occurs, the device sends RADIUS requests to the primary RADIUS server. By default, the device waits for up to two seconds (i.e., timeout) for a response from the RADIUS server for RADIUS requests and retransmission before it considers the server as down.

For each RADIUS server, the IP address, port, and shared secret can be configured. Each RADIUS server can be defined for RADIUS-based login authentication and/or RADIUS-based accounting. By setting the relevant port (authentication or accounting) to "0" disables the corresponding functionality. If both ports are configured, the RADIUS server is used for authentication and accounting. All servers configured with non-zero Authorization ports form an Authorization redundancy group and the device sends authorization requests to one of them, depending on their availability. All servers configured with non-zero Accounting ports form an Accounting redundancy group and the device sends accounting CDRs to one of them, depending on their availability. Below are example configurations:

- Only one RADIUS server is configured and used for authorization and accounting purposes (no redundancy). Therefore, both the Authorization and Accounting ports are defined.
- Three RADIUS servers are configured:
  - Two servers are used for authorization purposes only, providing redundancy. Therefore, only the Authorization ports are defined, while the Accounting ports are set to 0.
  - One server is used for accounting purposes only (i.e., no redundancy). Therefore, only the Accounting port is defined, while the Authorization port is set to 0.
- Two RADIUS servers are configured and used for authorization and accounting purposes, providing redundancy. Therefore, both the Authorization and Accounting ports are defined.

The status of the RADIUS severs can be viewed using the following CLI command:

```
# show system radius servers status
```

The example below shows the status of two RADIUS servers in redundancy mode for authorization and accounting:

```
servers 0
 ip-address 10.4.4.203
 auth-port 1812
 auth-ha-state "ACTIVE"
 acc-port 1813
 acc-ha-state "ACTIVE"
servers 1
 ip-address 10.4.4.202
 auth-port 1812
 auth-ha-state "STANDBY"
 acc-port 1813
 acc-ha-state "STANDBY"
```

Where *auth-ha-state* and *acc-ha-state* display the authentication and accounting redundancy status respectively. "ACTIVE" means that the server was used for the last sent authentication or accounting request; "STANDBY" means that the server was not used in the last sent request.

The following procedure describes how to configure a RADIUS server through the Web interface. You can also configure it through ini file (RadiusServers) or CLI configure system > radius > servers).

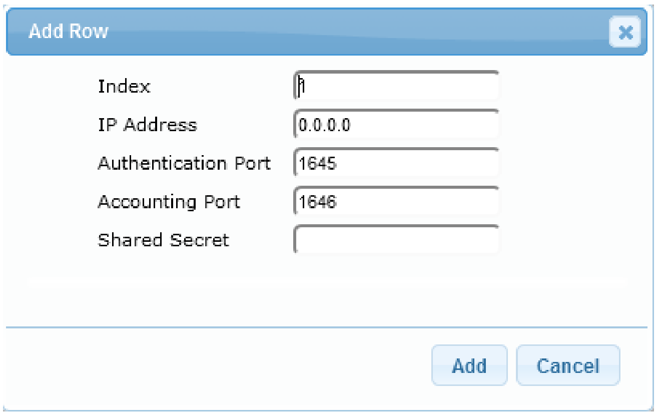
Note: To enable and configure RADIUS-based accounting, see "Configuring RADIUS Accounting" on page 760.

➢ **To configure a RADIUS server:**

1. Open the RADIUS Servers table (**Configuration** tab > **System** menu > **Management** > **RADIUS Servers**).

2. Click **Add**; the following dialog box appears:

Figure 15-7: RADIUS Servers Table - Add Row Dialog Box



Index: 1
IP Address: 0.0.0.0
Authentication Port: 1645
Accounting Port: 1646
Shared Secret:

3. Configure a RADIUS server according to the parameters described in the table below.
4. Click **Add**.

Table 15-7: RADIUS Servers Table Parameter Descriptions

Parameter	Description
Index [RadiusServers_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
IP Address `ip-address` [RadiusServers_IPAddress]	Defines the IP address of the RADIUS server (in dotted-decimal notation).
Authentication Port `auth-port` [RadiusServers_AuthenticationPort]	Defines the port of the RADIUS Authentication server for authenticating the device with the RADIUS server. When set to any value other than 0, the RADIUS server is used by the device for RADIUS-based management-user login authentication. When set to 0, RADIUS-based login authentication is not implemented. The valid value is 0 to any integer. The default is 1645.
Accounting Port `acc-port` [RadiusServers_AccountingPort]	Defines the port of the RADIUS Accounting server to where the device sends accounting data of SIP calls as call detail records (CDR). When set to any value other than 0, the RADIUS server is used by the device for RADIUS-based accounting (CDR). When set to 0, RADIUS-based accounting is not implemented. The valid value is 0 to any integer. The default is 1646.
Shared Secret `shared-secret` [RadiusServers_SharedSecret]	Defines the shared secret (password) for authenticating the device with the RADIUS server. This should be a cryptically strong password. The shared secret is also used by the RADIUS server to verify the authentication of the RADIUS messages sent by the device (i.e., message integrity). The valid value is up to 48 characters. By default, no value is defined.

15.3.3 Configuring Interface for RADIUS Communication

The device can communicate with the RADIUS server through its' OAMP (default) or SIP Control network interface. To change the interface used for RADIUS traffic, use the RadiusTrafficType parameter.

> **Note:** If set to Control, only one Control interface must be configured in the Interface table (see "Configuring IP Network Interfaces" on page 133); otherwise, RADIUS communication will fail.

15.3.4 Configuring General RADIUS Parameters

The procedure below describes the configuration of RADIUS parameters that are common between RADIUS-based user authentication and RADIUS-based accounting.

➢ **To configure general RADIUS parameters:**

1. Open the Authentication Settings page (**Configuration** tab > **System** menu > **Management** > **Authentication Settings**).
2. Scroll down the page to the RADIUS Settings group.
3. In the 'RADIUS VSA Vendor ID' field, enter the **same** vendor ID number as set on the third-party RADIUS server. The vendor-specific attribute (VSA) identifies the device to the RADIUS server using the Vendor ID. For an example of using the Vendor ID, see "Setting Up a Third-Party RADIUS Server" on page 228.
4. Configure RADIUS packet retransmission when no response is received from the RADIUS server:
   a. In the 'RADIUS Packets Retransmission' field (RADIUSRetransmission), enter the maximum number of RADIUS retransmissions that the device performs if no response is received from the RADIUS server.
   b. In the 'RADIUS Response Time Out' field (RadiusTO), enter the interval (in seconds) that the device waits for a response before sending a RADIUS retransmission.
5. Click **Submit**.

15.3.5 RADIUS-based Management User Authentication

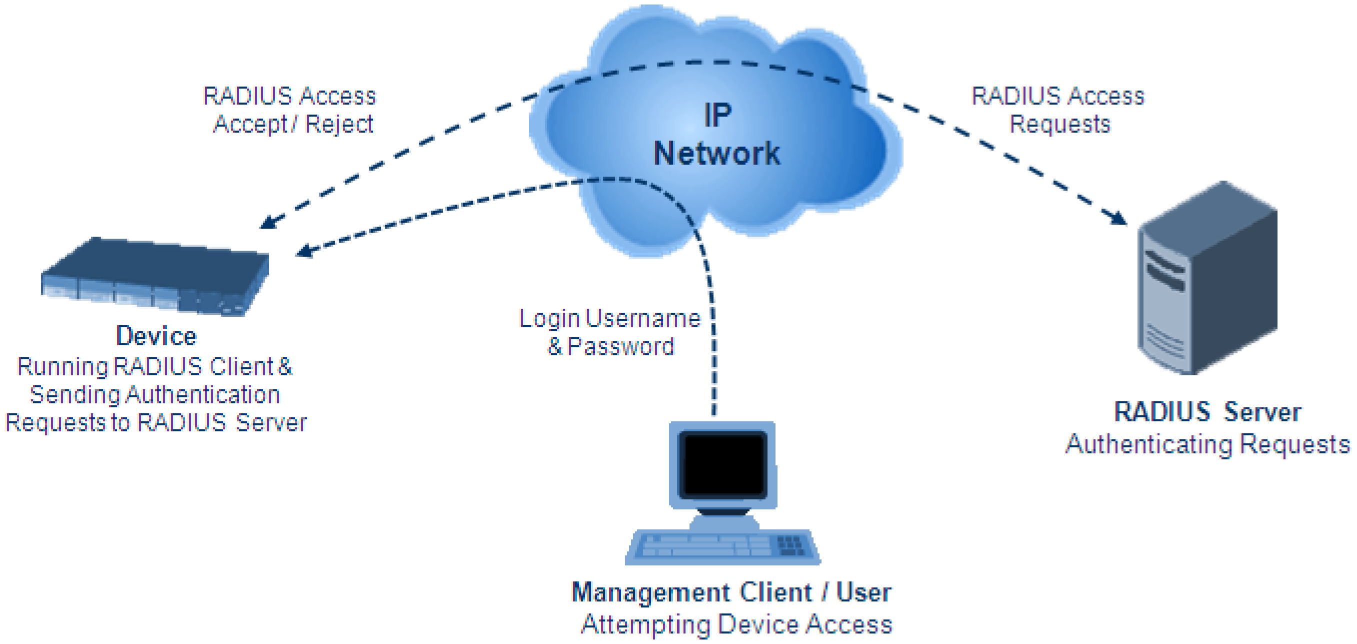
You can enhance security for your device by implementing Remote Authentication Dial-In User Service (RADIUS - RFC 2865) for authenticating multiple management user accounts of the device’s embedded Web and Telnet (CLI) servers. Thus, RADIUS also prevents unauthorized access to your device.

When RADIUS authentication is not used, the user's login username and password are locally authenticated by the device in its Web Users table (database). However, the Web Users table can be used as a fallback mechanism in case the RADIUS server does not respond. For configuring local user accounts, see "Configuring Web User Accounts" on page 70.

When RADIUS authentication is used, the RADIUS server stores the user accounts - usernames, passwords, and access levels (authorization). When a management user (client) tries to access the device, the device sends the RADIUS server the user's username and password for authentication. The RADIUS server replies with an acceptance or a rejection notification. During the RADIUS authentication process, the device’s Web interface is blocked until an acceptance response is received from the RADIUS server.

Note that communication between the device and the RADIUS server is done by using a shared secret, which is not transmitted over the network.

Figure 15-8: RADIUS Login Authentication for Management



For using RADIUS, you need to do the following:

- Set up a RADIUS server (third-party) to communicate with the device - see "Setting Up a Third-Party RADIUS Server" on page 228
- Configure the device as a RADIUS client for communication with the RADIUS server - see "Configuring RADIUS Authentication" on page 229

15.3.5.1 Setting Up a Third-Party RADIUS Server

The following procedure provides an example for setting up a third-party RADIUS sever, *FreeRADIUS*, which can be downloaded from www.freeradius.org. Follow the instructions on this Web site for installing and configuring the server. If you use a RADIUS server from a different vendor, refer to its appropriate documentation.

➢ **To set up a third-party RADIUS server (e.g., *FreeRADIUS*):**

1. Define the device as an authorized client of the RADIUS server, with the following:
   - Predefined *shared secret* (password used to secure communication between the device and the RADIUS server)
   - Vendor ID

   Below is an example of the *clients.conf* file (FreeRADIUS client configuration):

```
#
# clients.conf - client configuration directives
#
client 10.31.4.47 {
        secret          = FutureRADIUS
        shortname       = audc_device
}
```

2. If access levels are required, set up a Vendor-Specific Attributes (VSA) dictionary for the RADIUS server and select an attribute ID that represents each user's access level. The example below shows a dictionary file for FreeRADIUS that defines the attribute

"ACL-Auth-Level" with "ID=35". For the device's user access levels and their corresponding numeric representation in RADIUS servers, see "Configuring Web User Accounts" on page 70.

```
#
# AudioCodes VSA dictionary
#
VENDOR AudioCodes 5003
ATTRIBUTE ACL-Auth-Level 35 integer AudioCodes
VALUE ACL-Auth-Level ACL-Auth-UserLevel 50
VALUE ACL-Auth-Level ACL-Auth-AdminLevel 100
VALUE ACL-Auth-Level ACL-Auth-SecurityAdminLevel 200
```

3. Define the list of users authorized to use the device, using one of the password authentication methods supported by the server implementation. The example below shows a user configuration file for FreeRADIUS using a plain-text password:

```
# users - local user configuration database

john    Auth-Type := Local, User-Password == "qwerty"
        Service-Type = Login-User,
        ACL-Auth-Level = ACL-Auth-SecurityAdminLevel

sue   Auth-Type := Local, User-Password == "123456"
        Service-Type = Login-User,
        ACL-Auth-Level = ACL-Auth-UserLevel
```

4. Record and retain the IP address, port number, shared secret code, vendor ID, and VSA access level identifier (if access levels are implemented) used by the RADIUS server.

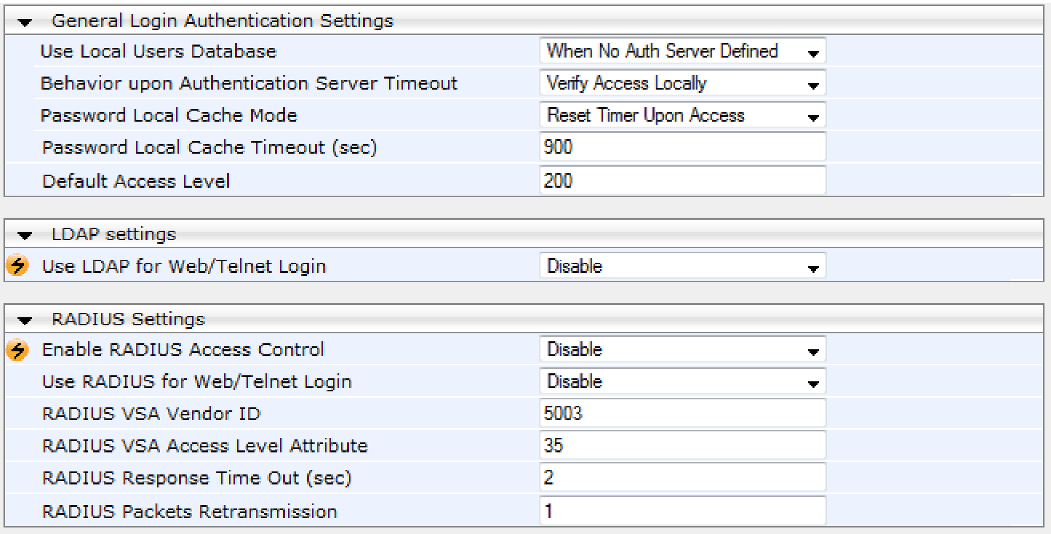
15.3.5.2 Configuring RADIUS-based User Authentication

The following procedure describes how to configure the RADIUS parameters specific to login authentication. For a detailed description of the RADIUS parameters, see "RADIUS Parameters" on page 1024.

➢ **To configure RADIUS parameters for login authentication:**

1. Open the Authentication Settings page (**Configuration** tab > **System** menu > **Management** > **Authentication Settings**).

Figure 15-9: Authentication Settings Page - RADIUS Configuration



General Login Authentication Settings

Use Local Users Database	When No Auth Server Defined
Behavior upon Authentication Server Timeout	Verify Access Locally
Password Local Cache Mode	Reset Timer Upon Access
Password Local Cache Timeout (sec)	900
Default Access Level	200

LDAP settings

Use LDAP for Web/Telnet Login	Disable

RADIUS Settings

Enable RADIUS Access Control	Disable
Use RADIUS for Web/Telnet Login	Disable
RADIUS VSA Vendor ID	5003
RADIUS VSA Access Level Attribute	35
RADIUS Response Time Out (sec)	2
RADIUS Packets Retransmission	1

2. From the 'Use RADIUS for Web/Telnet Login' drop-down list, select **Enable** to enable RADIUS authentication for Web and Telnet login.
3. When implementing Web user access levels, do one of the following:
   - **If the RADIUS server response includes the access level attribute:** In the 'RADIUS VSA Access Level Attribute' field, enter the code that indicates the access level attribute in the VSA section of the received RADIUS packet. For defining the RADIUS server with access levels, see "Setting Up a Third-Party RADIUS Server" on page 228.
   - **If the RADIUS server response does not include the access level attribute:** In the 'Default Access Level' field, enter the default access level that is applied to all users authenticated by the RADIUS server.
4. Configure RADIUS timeout handling:
   a. From the 'Behavior upon Authentication Server Timeout' drop-down list, select the option if the RADIUS server does not respond within five seconds:
      - **Deny Access**: device denies user login access.
      - **Verify Access Locally**: device checks the username and password configured locally for the user (in the Web User Accounts page or Web Users table), and if correct, allows access.

   b. In the 'Password Local Cache Timeout' field, enter a time limit (in seconds) after which the username and password verified by the RADIUS server becomes invalid and a username and password needs to be re-validated with the RADIUS server.
   c. From the 'Password Local Cache Mode' drop-down list, select the option for the local RADIUS password cache timer:
      - **Reset Timer Upon Access**: upon each access to a Web page, the timer resets (reverts to the initial value configured in the previous step).
      - **Absolute Expiry Timer**: when you access a Web page, the timer doesn’t reset, but continues its count down.
5. Configure when the Web Users table must be used to authenticate login users. From the 'Use Local Users Database' drop-down list, select one of the following:
   - **When No Auth Server Defined (default):** When no RADIUS server is configured or if a server is configured but connectivity with the server is down (if the server is up, the device authenticates the user with the server).
   - **Always:** First attempts to authenticate the user using the Web Users table, but if not found, it authenticates the user with the RADIUS server.
6. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

15.3.5.3 Securing RADIUS Communication

RADIUS authentication requires HTTP basic authentication (according to RFC 2617). However, this is insecure as the usernames and passwords are transmitted in clear text over plain HTTP. Thus, as digest authentication is not supported with RADIUS, it is recommended that you use HTTPS with RADIUS so that the usernames and passwords are encrypted.

To configure the device to use HTTPS, set the 'Secured Web Connection (HTTPS)' parameter to **HTTPS Only**, in the Web Security Settings page (**Configuration** tab > **System** menu > **Management** > **Web Security Settings**).

15.3.5.4 RADIUS-based User Authentication in URL

RADIUS authentication of the management user is typically done after the user accesses the Web interface by entering only the device's IP address in the Web browser's URL field (for example, http://10.13.4.12/) and then entering the username and password credentials in the Web interface's login screen. However, authentication with the RADIUS server can also be done immediately after the user enters the URL, if the URL also contains the login credentials. For example: http://10.4.4.112/Forms/RadiusAuthentication?WSBackUserName=John&WSBackPassword=1234

Note: This feature allows up to five simultaneous users only.

15.3.6 RADIUS-based CDR Accounting

Once you have configured a RADIUS server(s) for accounting in "Configuring RADIUS Servers" on page 224, you need to enable and configure RADIUS-based CDR accounting (see "Configuring RADIUS Accounting" on page 760).

15.4 LDAP-based Management and SIP Services

The device supports the Lightweight Directory Access Protocol (LDAP) application protocol and can operate with third-party, LDAP-compliant servers such as Microsoft Active Directory (AD).

You can use LDAP for the following LDAP services:

- **SIP-related (Control) LDAP Queries:** This can be used for routing or manipulation (e.g., calling name and destination address). The device connects and binds to the remote LDAP server (IP address or DNS/FQDN) during the service's initialization (at device start-up) or whenever you change the LDAP server's IP address and port. Binding to the LDAP server is based on username and password (Bind DN and Password). Service makes 10 attempts to connect and bind to the remote LDAP server, with a timeout of 20 seconds between attempts. If connection fails, the service remains in disconnected state until the LDAP server's IP address or port is changed. If connection to the LDAP server later fails, the service attempts to reconnect.

  For the device to run a search, the path to the directory's subtree, known as the distinguished name (DN), where the search is to be done must be configured (see "Configuring LDAP DNs (Base Paths) per LDAP Server" on page 241). The search key (filter), which defines the exact DN to search, and one or more attributes whose values must be returned to the device must also be configured. For more information on configuring these attributes and search filters, see "Active Directory-based Routing for Microsoft Lync" on page 254.

  The device can store recent LDAP queries and responses in its local cache. The cache is used for subsequent queries and/or in case of LDAP server failure. For more information, see "Configuring the Device's LDAP Cache" on page 244.

  If connection with the LDAP server disconnects (broken), the device sends the SNMP alarm, acLDAPLostConnection. Upon successful reconnection, the alarm clears. If connection with the LDAP server is disrupted during the search, all search requests are dropped and an alarm indicating a failed status is sent to client applications.

- **Management-related LDAP Queries:** This is used for authenticating and authorizing management users (Web and CLI) and is based on the user's login username and password (credentials) when attempting login to one of the device's management platforms. When configuring the login username (LDAP Bind DN) and password (LDAP Password) to send to the LDAP server, you can use templates based on the dollar ($) sign, which the device replaces with the actual username and password entered by the user during the login attempt. You can also configure the device to send the username and password in clear-text format or encrypted using TLS (SSL).

  The device connects to the LDAP server (i.e., an LDAP session is created) only when a login attempt occurs. The LDAP Bind operation establishes the authentication of the user based on the username-password combination. The server typically checks the password against the userPassword attribute in the named entry. A successful Bind operation indicates that the username-password combination is correct; a failed Bind operation indicates that the username-password combination is incorrect.

  Once the user is successfully authenticated, the established LDAP session may be used for further LDAP queries to determine the user's management access level and privileges (Operator, Admin, or Security Admin). This is known as the user authorization stage. To determine the access level, the device searches the LDAP directory for groups of which the user is a member, for example:

  ```
  CN=\# Support Dept,OU=R&D
  Groups,OU=Groups,OU=APC,OU=Japan,OU=ABC,DC=corp,DC=abc,DC=com
  CN=\#AllCellular,OU=Groups,OU=APC,OU=Japan,OU=ABC,DC=corp,DC=a
  bc,DC=com
  ```

  The device then assigns the user the access level configured for that group (in "Configuring Access Level per Management Groups Attributes" on page 243). The location in the directory where you want to search for the user's member group(s) is configured using the following:

- Search base object (distinguished name or DN, e.g., "ou=ABC,dc=corp,dc=abc,dc=com"), which defines the location in the directory from where the LDAP search begins, and is configured in "Configuring LDAP DNs (Base Paths) per LDAP Server" on page 241.
- Search filter, for example, (&(objectClass=person)(sAMAccountName=JohnD)), which filters the search in the subtree to include only the specific username. The search filter can be configured with the dollar ($) sign to represent the username, for example, (sAMAccountName=$). For configuring the search filter, see "Configuring the LDAP Search Filter Attribute" on page 242.
- Management attribute (e.g., memberOf), from where objects that match the search filter criteria are returned. This shows the user's member groups. The attribute is configured in the LDAP Configuration table (see "Configuring LDAP Servers" on page 236).

If the device finds a group, it assigns the user the corresponding access level and permits login; otherwise, login is denied. Once the LDAP response has been received (success or failure), the device ends the LDAP session.

For both of the previously discussed LDAP services, the following additional LDAP functionality is supported:

- Search method for searching DN object records between LDAP servers and within each LDAP server (see Configuring LDAP Search Methods).
- Default access level that is assigned to the user if the queried response does not contain an access level.
- Local users database (Web Users table) for authenticating users instead of the LDAP server (for example, when a communication problem occurs with the server). For more information, see "Configuring Local Database for Management User Authentication" on page 248.

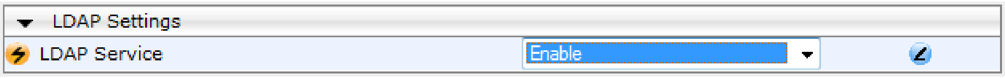
15.4.1 Enabling the LDAP Service

Before you can configure LDAP support, you need to enable the LDAP service.

➢ **To enable LDAP:**

1. Open the LDAP Settings page (**Configuration** tab > **VoIP** menu > **Services** > **LDAP** > **LDAP Settings**).

Figure 15-10: Enabling LDAP on the LDAP Settings Page



LDAP Settings

LDAP Service: Enable

2. Under LDAP Settings, from the 'LDAP Service' drop-down list, select **Enable**.
3. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

15.4.2 Enabling LDAP-based Web/CLI User Login Authentication and Authorization

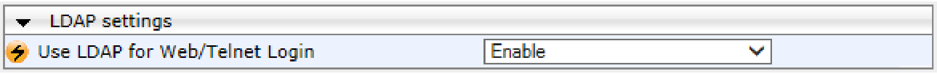
The LDAP service can be used for authenticating and authorizing device management users (Web and CLI), based on the user's login username and password (credentials). At the same, it can also be used to determine users' management access levels (privileges). Before you can configure LDAP-based login authentication, you must enable this type of LDAP service, as described in the following procedure.

➢ **To enable LDAP-based login authentication:**

1. Open the Authentication Settings page (**Configuration** tab > **System** menu >

Management > **Authentication Settings**).

Figure 15-11: Authentication Settings Page - Enabling LDAP-based Login



LDAP settings

Use LDAP for Web/Telnet Login: Enable

2. Under LDAP Settings, from the 'Use LDAP for Web/Telnet Login' drop-down list, select **Enable**.
3. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

15.4.3 Configuring LDAP Server Groups

The LDAP Server Groups table lets you configure up to 41 LDAP Server Groups. An LDAP Server Group is a logical configuration entity that contains up to two LDAP servers. LDAP servers are assigned to LDAP Server Groups in the LDAP Configuration table (see "Configuring LDAP Servers" on page 236). To use a configured LDAP server, you must assign it to an LDAP Server Group.

To use an LDAP server for call routing, you must configure its' LDAP Server Group as "Control" type, and then assign the LDAP Server Group to a Routing Policy. The Routing Policy in turn, needs to be assigned to the relevant routing rule(s). A Routing Policy can be assigned only one LDAP Server Group. Therefore, for multi-tenant deployments where multiple Routing Policies are employed, each tenant can be assigned a specific LDAP Server Group through its unique Routing Policy.

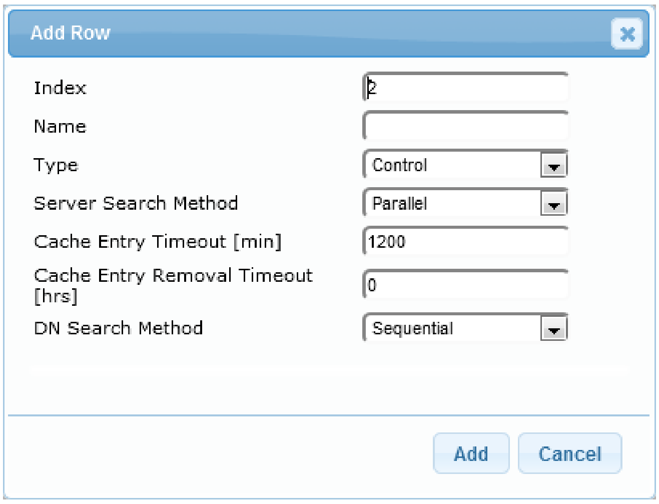
To use an LDAP server for management user login authentication and authorization, you must configure its' LDAP Server Group as "Management" type. Additional LDAP-based management parameters need to be configured, as described in "Enabling LDAP-based Web/CLI User Login Authentication and Authorization" on page 233 and "Configuring LDAP Servers" on page 236.

The following procedure describes how to configure an LDAP Server Group through the Web interface. You can also configure it through ini file (LDAPServersGroup) or CLI (configure voip/ldap/ldap-servers-group).

➢ **To configure an LDAP Server Group:**

1. Open the LDAP Server Groups table (**Configuration** tab > **VoIP** menu > **Services** > **LDAP** > **LDAP Server Groups**).
2. Click **Add**; the following dialog box appears:

Figure 15-12: LDAP Server Groups Table - Add Row Dialog Box



Index: 2

Name:

Type: Control

Server Search Method: Parallel

Cache Entry Timeout [min]: 1200

Cache Entry Removal Timeout [hrs]: 0

DN Search Method: Sequential

3. Configure an LDAP Server Group according to the parameters described in the table below.

4. Click **Add**.

Table 15-8: LDAP Server Groups Table Parameter Descriptions

Parameter	Description
Index [LdapServersGroup_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `name` [LdapServersGroup_Name]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 20 characters. **Note:** Each row must be configured with a unique name.
Type `server-type` [LdapServersGroup_ServerType]	Defines whether the servers in the group are used for SIP-related LDAP queries (Control) or management login authentication-related LDAP queries (Management). ▪ [0] Control (Default) ▪ [1] Management **Note:** Only one LDAP Server Group can be defined for management.
Server Search Method `server-search-method` [LdapServersGroup_SearchMethod]	Defines the method for querying between the two LDAP servers in the group. ▪ [0] Parallel = (Default) The device queries the LDAP servers at the same time. ▪ [1] Sequential = The device first queries one of the LDAP servers and if the DN object is not found or the search fails, it queries the second LDAP server.
Cache Entry Timeout `cache-entry-timeout` [LdapServersGroup_CacheEntryTimeout]	Defines the duration (in minutes) that an entry in the device's LDAP cache is valid. If the timeout expires, the cached entry is used only if there is no connectivity with the LDAP server. The valid range is 0 to 35791. The default is 1200. If set to 0, the LDAP entry is always valid.
Cache Entry Removal Timeout `cache-entry-removal-timeout` [LdapServersGroup_CacheEntryRemovalTimeout]	Defines the duration (in hours) after which the LDAP entry is deleted from the device's LDAP cache. The valid range is 0 to 596. The default is 0 (i.e., the entry is never deleted).
DN Search Method `search-dn-method` [LdapServersGroup_SearchDnsMethod]	Defines the method for querying the Distinguished Name (DN) objects within each LDAP server. ▪ [0] Sequential = (Default) The query is done in each DN object, one by one, until a result is returned. For example, a search for the DN object record "JohnD" is first run in DN object "Marketing" and if a result is not found, it searches in "Sales", and if not found, it searches in "Administration", and so on. ▪ [1] Parallel = The query is done in all DN objects at the same time. For example, a search for the DN object record "JohnD" is done at the same time in the "Marketing", "Sales" and "Administration" DN objects.

15.4.4 Configuring LDAP Servers

The LDAP Configuration table lets you configure up to 82 LDAP servers. This table defines the address and connectivity settings of the LDAP server. The LDAP server can be configured for SIP-related queries (e.g., routing and manipulation) or LDAP-based management user login authentication and authorization (username-password).

The following procedure describes how to configure an LDAP server through the Web interface. You can also configure it through ini file (LdapConfiguration) or CLI (configure voip > ldap > ldap-configuration).

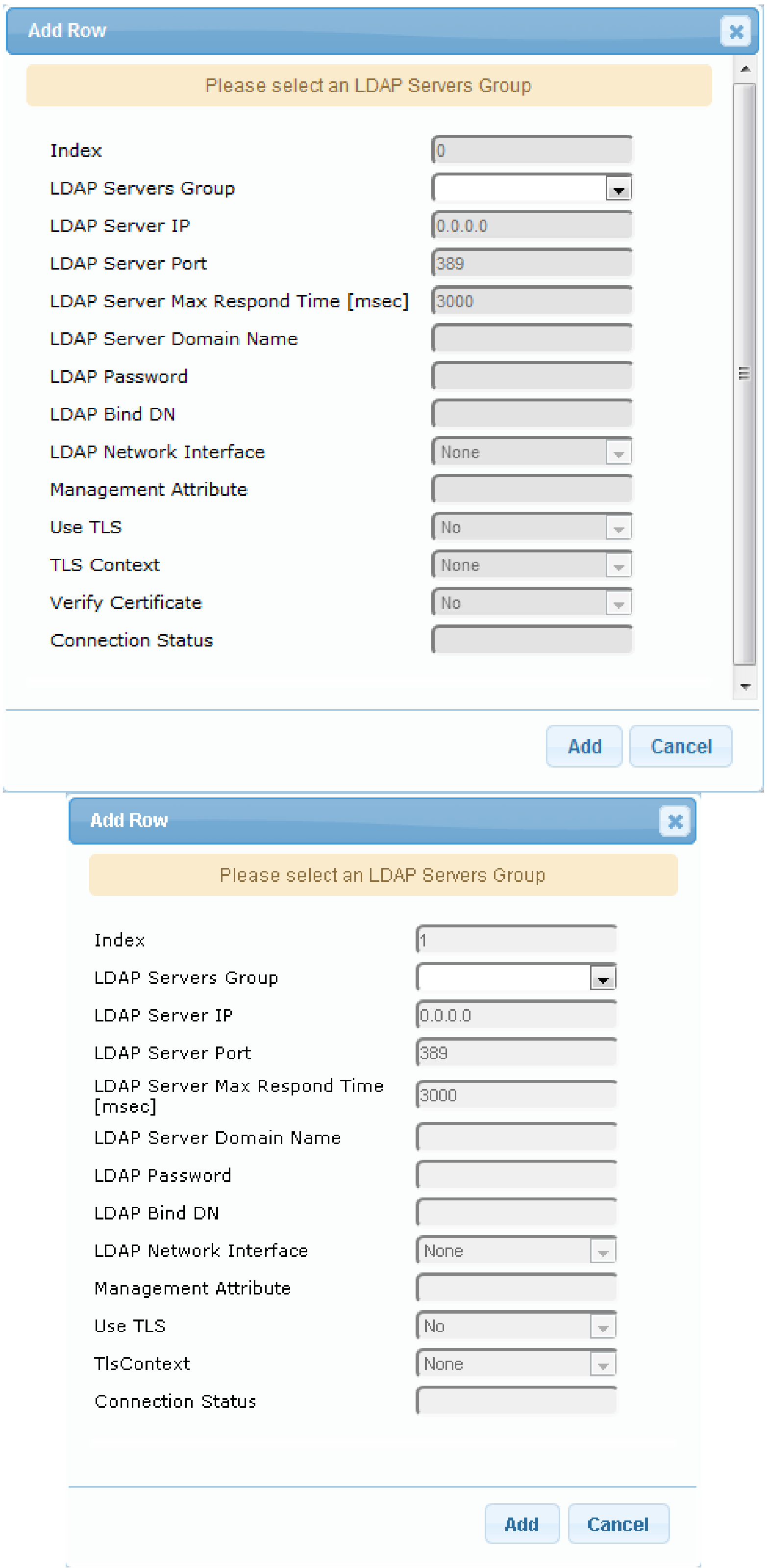
Note: When you configure an LDAP server, you need to assign it to an LDAP Server Group. Therefore, before you can configure an LDAP server in the table, you must first configure at least one LDAP Server Group in the LDAP Server Groups table (see "Configuring LDAP Server Groups" on page 234).

➢ **To configure an LDAP server:**

1. Open the LDAP Configuration Table (**Configuration** tab > **VoIP** menu > **Services** > **LDAP** > **LDAP Configuration Table**).

2. Click **Add**; the following dialog box appears:

Figure 15-13: LDAP Configuration Table - Add Row Dialog Box



Add Row

Please select an LDAP Servers Group

Index	0
LDAP Servers Group	
LDAP Server IP	0.0.0.0
LDAP Server Port	389
LDAP Server Max Respond Time [msec]	3000
LDAP Server Domain Name	
LDAP Password	
LDAP Bind DN	
LDAP Network Interface	None
Management Attribute	
Use TLS	No
TLS Context	None
Verify Certificate	No
Connection Status	

Add Row

Please select an LDAP Servers Group

Index	1
LDAP Servers Group	
LDAP Server IP	0.0.0.0
LDAP Server Port	389
LDAP Server Max Respond Time [msec]	3000
LDAP Server Domain Name	
LDAP Password	
LDAP Bind DN	
LDAP Network Interface	None
Management Attribute	
Use TLS	No
TlsContext	None
Connection Status	

3. Configure an LDAP server according to the parameters described in the table below.
4. Click **Add**.

Table 15-9: LDAP Configuration Table Parameter Descriptions

Parameter	Description
Index [LdapConfiguration_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
LDAP Servers Group `server-group` [LdapConfiguration_Group]	Assigns the LDAP server to an LDAP Server Group, configured in the LDAP Server Groups table (see "Configuring LDAP Server Groups" on page 234). **Notes:** ▪ The parameter is mandatory and must be set before configuring the other parameters in the table. ▪ Up to two LDAP servers can be assigned to the same LDAP Server Group.
LDAP Server IP `server-ip` [LdapConfiguration_LdapConfServerIp]	Defines the IP address of the LDAP server (in dotted-decimal notation, e.g., 192.10.1.255). By default, no IP address is defined. **Notes:** ▪ The parameter is mandatory. ▪ If you want to use an FQDN for the LDAP server, leave the parameter undefined and configure the FQDN in the 'LDAP Server Domain Name' parameter (see below).
LDAP Server Port `server-port` [LdapConfiguration_LdapConfServerPort]	Defines the port number of the LDAP server. The valid value range is 0 to 65535. The default port number is 389.
LDAP Server Max Respond Time `max-respond-time` [LdapConfiguration_LdapConfServerMaxRespondTime]	Defines the duration (in msec) that the device waits for LDAP server responses. The valid value range is 0 to 86400. The default is 3000. **Note:** If the response time expires, you can configure the device to use its local database (Web Users table) for authenticating the user. For more information, see "Configuring Local Database for Management User Authentication" on page 248.
LDAP Server Domain Name `domain-name` [LdapConfiguration_LdapConfServerDomainName]	Defines the domain name (FQDN) of the LDAP server. The device tries to connect to the LDAP server according to the IP address listed in the received DNS query. If there is no connection to the LDAP server or the connection to the LDAP server fails, the device tries to connect to the LDAP server with the next IP address in the DNS query list. **Note:** If the 'LDAP Server IP' parameter is configured, the 'LDAP Server Domain Name' parameter is ignored. Thus, if you want to use an FQDN, leave the 'LDAP Server IP' parameter undefined.

Parameter	Description
LDAP Password `password` [LdapConfiguration_LdapConfPassword]	Defines the user password for accessing the LDAP server during connection and binding operations. ▪ LDAP-based SIP queries: The parameter is the password used by the device to authenticate itself, as a client, to obtain LDAP service from the LDAP server. ▪ LDAP-based user login authentication: The parameter represents the login password entered by the user during a login attempt. You can use the $ (dollar) sign in this value to enable the device to automatically replace the $ sign with the user's login password in the search filter, which it sends to the LDAP server for authenticating the user's username-password combination. For example, $. **Notes:** ▪ The parameter is mandatory. ▪ By default, the device sends the password in clear-text format. You can enable the device to encrypt the password using TLS (see the 'Use SSL' parameter below).
LDAP Bind DN `bind-dn` [LdapConfiguration_LdapConfBindDn]	Defines the LDAP server's bind Distinguished Name (DN) or username. ▪ LDAP-based SIP queries: The DN is used as the username during connection and binding to the LDAP server. The DN is used to uniquely name an AD object. Below are example parameter settings: ✓ cn=administrator,cn=Users,dc=domain,dc=com ✓ administrator@domain.com ✓ domain\administrator ▪ LDAP-based user login authentication: The parameter represents the login username entered by the user during a login attempt. You can use the $ (dollar) sign in this value to enable the device to automatically replace the $ sign with the user's login username in the search filter, which it sends to the LDAP server for authenticating the user's username-password combination. An example configuration for the parameter is $@sales.local, where the device replaces the $ with the entered username, for example, JohnD@sales.local. The username can also be configured with the domain name of the LDAP server. **Note:** By default, the device sends the username in clear-text format. You can enable the device to encrypt the username using TLS (see the 'Use SSL' parameter below).
LDAP Network Interface `interface-type` [LdapConfiguration_Interface]	Assigns one of the device's IP network interfaces through which communication with the LDAP server is done. By default, no value is defined (**None**) and the device uses the OAMP network interface, configured in the Interface table. For configuring IP network interfaces, see "Configuring IP Network Interfaces" on page 133. **Note:** The parameter is mandatory.

Parameter	Description
Management Attribute `mgmt-attr` [LdapConfiguration_MngmAuthAtt]	Defines the LDAP attribute name to query, which contains a list of groups to which the user is a member. For Active Directory, this attribute is typically "memberOf". The attribute's values (groups) are used to determine the user's management access level; the group's corresponding access level is configured in "Configuring Access Level per Management Groups Attributes" on page 243. **Notes:** ▪ The parameter is applicable only to LDAP-based login authentication and authorization (i.e., the 'Type' parameter is set to **Management**). ▪ If this functionality is not used, the device assigns the user the configured default access level. For more information, see "Configuring Access Level per Management Groups Attributes" on page 243.
Use TLS `use-tls` [LdapConfiguration_useTLS]	Enables the device to encrypt the username and password (for Control and Management related queries) using TLS when sending them to the LDAP server. ▪ **[0]** No = (Default) Username and password are sent in clear-text format. ▪ **[1]** Yes
TLS Context [LdapConfiguration_ContextName]	Assigns a TLS Context for the connection with the LDAP server. By default, no value is defined (**None**) and the device uses the default TLS Context (ID 0). For configuring TLS Contexts, see "Configuring TLS Certificate Contexts" on page 107. **Note:** The parameter is applicable only if the 'Use TLS' parameter is configured to **Yes**.
Verify Certificate `verify-certificate` [LdapConfiguration_VerifyCertificate]	Enables certificate verification when the connection with the LDAP server uses TLS. ▪ [0] No = (Default) No certificate verification is done. ▪ [1] Yes = The device verifies the authentication of the certificate received from the LDAP server. The device authenticates the certificate against the trusted root certificate store associated with the associated TLS Context (see 'TLS Context' parameter above) and if ok, allows communication with the LDAP server. If authentication fails, the device denies communication (i.e., handshake fails). The device can also authenticate the certificate by querying with an Online Certificate Status Protocol (OCSP) server whether the certificate has been revoked. This is also configured for the associated TLS Context. **Note:** The parameter is applicable only if the 'Use TLS parameter is configured to **Yes**.

Parameter	Description
Connection Status `connection-status` [LdapConfiguration_ConnectionStatus]	(Read-only) Displays the connection status with the LDAP server. ▪ "Not Applicable" ▪ "LDAP Connection Broken" ▪ "Connecting" ▪ "Connected" **Note:** For more information about a disconnected LDAP connection, see your Syslog messages generated by the device.

15.4.5 Configuring LDAP DNs (Base Paths) per LDAP Server

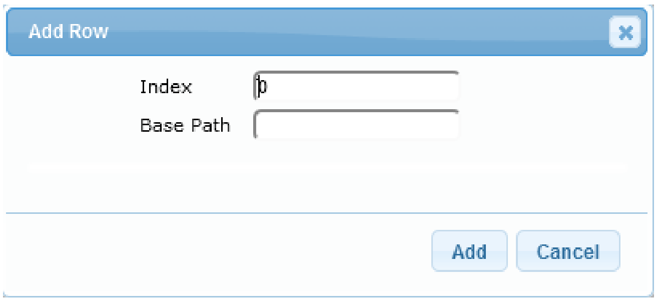
The LDAP Search DN table lets you configure LDAP base paths. The table is a "child" of the LDAP Configuration table (see "Configuring LDAP Servers" on page 236) and configuration is done per LDAP server. For the device to run a search using the LDAP service, the base path to the directory's subtree, referred to as the distinguished name object (or DN), where the search is to be done must be configured. For each LDAP server, you can configure up to three base paths.

The following procedure describes how to configure DNs per LDAP server through the Web interface. You can also configure it through ini file (LdapServersSearchDNs) or CLI (configure voip/ldap/ldap-servers-search-dns).

➢ To configure an LDAP base path per LDAP server:

1. Open the LDAP Configuration table (**Configuration** tab > **VoIP** menu > **Services** > **LDAP** > **LDAP Configuration Table**).
2. In the table, select the row of the LDAP server for which you want to configure DN base paths, and then click the **LDAP Servers Search DNs** link, located below the table; the LDAP Server Search Base DN table opens.
3. Click **Add**; the following dialog box appears:

Figure 15-14: LDAP Search Base DN Table - Add Row Dialog Box



Index: 0
Base Path:

4. Configure an LDAP DN base path according to the parameters described in the table below.
5. Click **Add**, and then save ("burn") your settings to flash memory.

Table 15-10: LDAP Server Search Base DN Table Parameter Descriptions

Parameter	Description
Index `set internal-index` [LdapServersSearchDNs_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.

Parameter	Description
Base Path `set base-path` [LdapServersSearchDNs_Base_Path]	Defines the full path (DN) to the objects in the AD where the query is done. The valid value is a string of up to 256 characters. For example: OU=NY,DC=OCSR2,DC=local. In this example, the DN path is defined by the LDAP names, OU (organizational unit) and DC (domain component).

15.4.6 Configuring the LDAP Search Filter Attribute

When the LDAP-based login username-password authentication succeeds, the device searches the LDAP server for all groups of which the user is a member. The LDAP query is based on the following LDAP data structure:

- **Search base object (distinguished name or DN, e.g., "ou=ABC,dc=corp,dc=abc,dc=com"):** The DN defines the location in the directory from which the LDAP search begins and is configured in "Configuring LDAP DNs (Base Paths) per LDAP Server" on page 241.
- **Filter (e.g., "(&(objectClass=person)(sAMAccountName=johnd))"):** This filters the search in the subtree to include only the login username (and excludes others). This is configured by the 'LDAP Authentication Filter' parameter, as described in the following procedure. You can use the dollar ($) sign to represent the username. For example, the filter can be configured as "(sAMAccountName=$)", where if the user attempts to log in with the username "SueM", the LDAP search is done only for the attribute sAMAccountName that equals "SueM".
- **Attribute (e.g., "memberOf") to return from objects that match the filter criteria:** The attribute is configured by the 'Management Attribute' parameter in the LDAP Configuration table (see "Configuring LDAP Servers" on page 236).

Therefore, the LDAP response includes only the groups of which the specific user is a member.

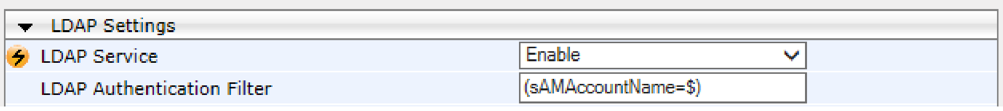
Notes:

- The search filter is applicable only to LDAP-based login authentication and authorization queries.
- The search filter is a global setting that applies to all LDAP-based login authentication and authorization queries, across all configured LDAP servers.

➢ To configure the LDAP search filter for management users:

1. Open the LDAP Settings page (**Configuration** tab > **VoIP** menu > **Services** > **LDAP** > **LDAP Settings**).

Figure 15-15: LDAP Settings Page - LDAP Search Filter



▼ LDAP Settings

LDAP Service	Enable
LDAP Authentication Filter	(sAMAccountName=$)

2. Make sure that the 'LDAP Service' parameter is configured to **Enable**.
3. In the 'LDAP Authentication Filter' parameter, enter the LDAP search filter attribute for searching the login username for user authentication.
4. Click **Submit**.

15.4.7 Configuring Access Level per Management Groups Attributes

The Management LDAP Groups table lets you configure LDAP group objects and their corresponding management user access level. The table is a "child" of the LDAP Configuration table (see "Configuring LDAP Servers" on page 236) and configuration is done per LDAP server. For each LDAP server, you can configure up to three table row entries of LDAP group(s) and their corresponding access level.

Notes:

- The Management LDAP Groups table is applicable only to LDAP-based login authentication and authorization queries.
- If the LDAP response received by the device includes multiple groups of which the user is a member and you have configured different access levels for some of these groups, the device assigns the user the highest access level. For example, if the user is a member of two groups where one has access level "Monitor" and the other "Administrator", the device assigns the user the "Administrator" access level.
- When the access level is unknown, the device assigns the default access level to the user, configured by the 'Default Access Level' parameter in the Authentication Settings page (**Configuration** tab > **System** menu > **Management** > **Authentication Settings**). This can occur in the following scenarios:
    - √ The user is not a member of any group.
    - √ The group of which the user is a member is not configured on the device (as described in this section).
    - √ The device is not configured to query the LDAP server for a management attribute (see "Configuring LDAP Servers" on page 236).

Group objects represent groups in the LDAP server of which the user is a member. The access level represents the user account's permissions and rights in the device's management interface (e.g., Web and CLI). The access level can either be Monitor, Administrator, or Security Administrator. For an explanation on the privileges of each level, see "Configuring Web User Accounts" on page 70.

When the username-password authentication with the LDAP server succeeds, the device searches the LDAP server for all groups of which the user is a member. The LDAP query is based on the following LDAP data structure:

- Search base object (distinguished name or DN, e.g., "ou=ABC,dc=corp,dc=abc,dc=com"), which defines the location in the directory from which the LDAP search begins. This is configured in "Configuring LDAP DNs (Base Paths) per LDAP Server" on page 241.
- Filter (e.g., "(&(objectClass=person)(sAMAccountName=johnd))"), which filters the search in the subtree to include only the login username (and excludes others). This is configured by the 'LDAP Authentication Filter' parameter.
- Attribute (e.g., "memberOf") to return from objects that match the filter criteria. This attribute is configured by the 'Management Attribute' parameter in the LDAP Configuration table.

The LDAP response includes all the groups of which the specific user is a member, for example:

```
CN=\# Support Dept,OU=R&D
Groups,OU=Groups,OU=APC,OU=Japan,OU=ABC,DC=corp,DC=abc,DC=com
CN=\#AllCellular,OU=Groups,OU=APC,OU=Japan,OU=ABC,DC=corp,DC=abc,D
C=com
```

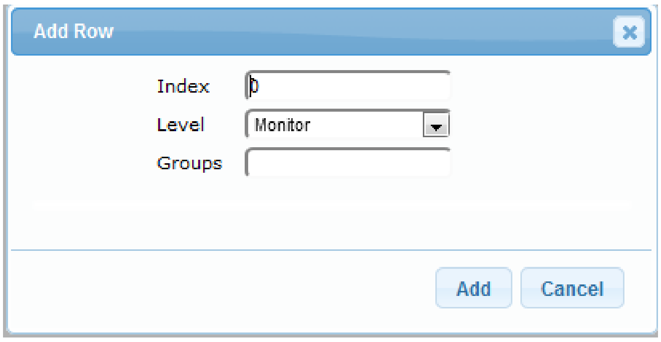
The device searches this LDAP response for the group names that you configured in the Management LDAP Groups table in order to determine the user's access level. If the device finds a group name, the user is assigned the corresponding access level and login is permitted; otherwise, login is denied. Once the LDAP response has been received (success or failure), the LDAP session terminates.

The following procedure describes how to configure an access level per management groups through the Web interface. You can also configure it through ini file (MgmntLDAPGroups) or CLI (configure voip > ldap > mgmt-ldap-groups).

- **To configure management groups and corresponding access level:**

1. Open the LDAP Configuration table (**Configuration** tab > **VoIP** menu > **Services** > **LDAP** > **LDAP Configuration Table**).
2. In the table, select the row of the LDAP server for which you want to configure management groups with a corresponding access level, and then click the **Management LDAP Groups** link, located below the table; the Management LDAP Groups table opens.
3. Click **Add**; the following dialog box appears:

Figure 15-16: Management LDAP Groups Table - Add Row Dialog Box



Index: 0
Level: Monitor
Groups:

4. Configure a group name(s) with a corresponding access level according to the parameters described in the table below.
5. Click **Add**, and then save ("burn") your settings to flash memory.

Table 15-11: Management LDAP Groups Table Parameter Descriptions

Parameter	Description
Index [MgmntLDAPGroups_GroupIndex]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Level `level` [MgmntLDAPGroups_Level]	Defines the access level of the group(s). ▪ **[0]** Monitor (Default) ▪ **[1]** Admin ▪ **[2]** Security Admin
Groups `groups` [MgmntLDAPGroups_Group]	Defines the attribute names of the groups in the LDAP server. The valid value is a string of up to 256 characters. To define multiple groups, separate each group name with a semicolon (;).

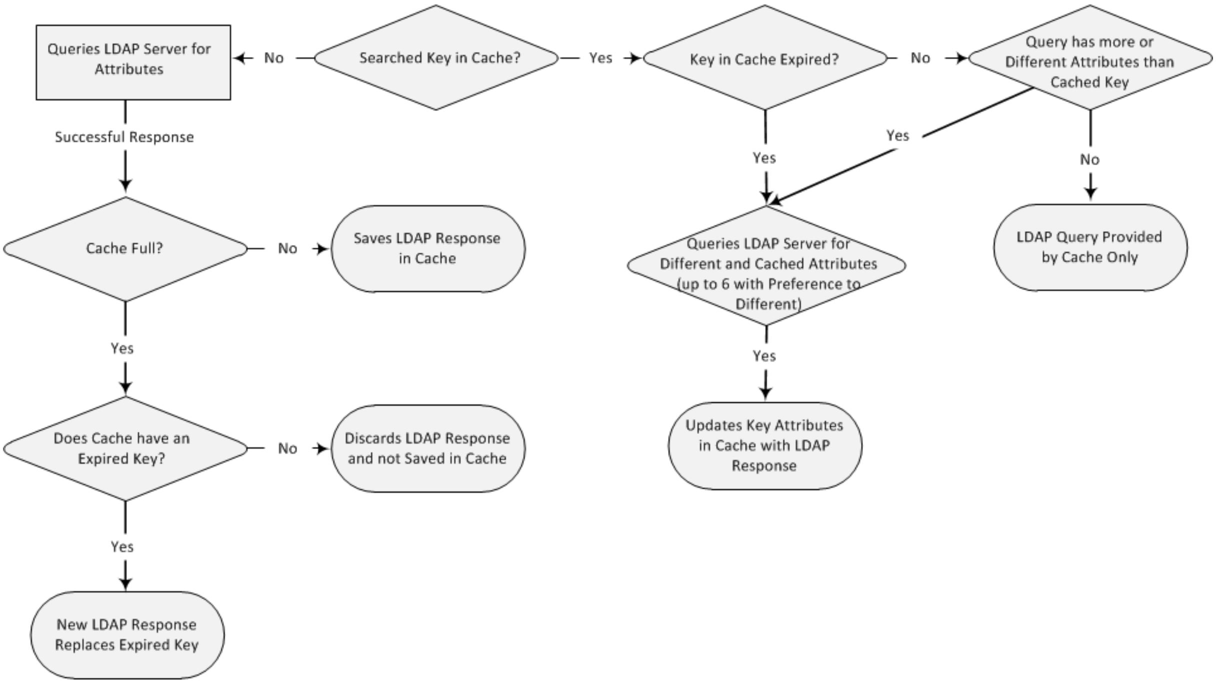
15.4.8 Configuring the Device's LDAP Cache

The device can optionally store LDAP queries of LDAP Attributes for a searched key with an LDAP server and the responses (results) in its local cache. The cache is used for subsequent queries and/or in case of LDAP server failure. The benefits of this feature include the following:

- Improves routing decision performance by using local cache for subsequent LDAP queries
- Reduces number of queries performed on an LDAP server and corresponding bandwidth consumption
- Provides partial survivability in case of intermittent LDAP server failure (or network isolation)

The handling of LDAP queries using the device's LDAP cache is shown in the flowchart below:

Figure 15-17: LDAP Query Process with Local LDAP Cache


Queries LDAP Server for Attributes
No
Searched Key in Cache?
Yes
Key in Cache Expired?
No
Query has more or Different Attributes than Cached Key
Successful Response
Yes
Yes
No
Cache Full?
No
Saves LDAP Response in Cache
Queries LDAP Server for Different and Cached Attributes (up to 6 with Preference to Different)
LDAP Query Provided by Cache Only
Yes
Yes
Does Cache have an Expired Key?
No
Discards LDAP Response and not Saved in Cache
Updates Key Attributes in Cache with LDAP Response
Yes
New LDAP Response Replaces Expired Key

If an LDAP query is required for an Attribute of a key that is already cached with that same Attribute, instead of sending a query to the LDAP server, the device uses the cache. However, if an LDAP query is required for an Attribute that does not appear for the cached key, the device queries the LDAP server and then saves the new Attribute (and response) in the cache for that key. When the device queries new Attributes for a cached key, the device also includes already cached Attributes of the key, while adhering to the maximum number of allowed saved Attributes (see note below), with preference to the new Attributes. In other words, if the cached key already contains the maximum Attributes and an LDAP query is required for a new Attribute, the device sends an LDAP query to the server for the new Attribute and for the five most recent Attributes already cached with the key. Upon the LDAP response, the new Attribute replaces the oldest cached Attribute while the values of the other Attributes are refreshed with the new response. The following table shows an example of different scenarios of LDAP queries of a cached key whose cached Attributes include a, b , c, and d, where a is the oldest and d the most recent Attribute:

Table 15-12: Example of LDAP Query for Cached Attributes

Attributes Requested in New LDAP Query for Cached Key	Attributes Sent in LDAP Query to LDAP Server	Attributes Saved in Cache after LDAP Response
e	**e**, a, b, c, d	**e**, a, b, c, d
e, **f**	**e**, **f**, a, b, c, d	**e**, **f**, a, b, c, d
e, **f**, **g**, **h**, **i**	**e**, **f**, **g**, **h**,**i**, a	**e**, **f**, **g**, **h**,**i**, a
e, **f**, **g**, **h**, **i**, **j**	**e**, **f**, **g**, **h**, **i**, **j**	**e**, **f**, **g**, **h**, **i**, **j**

Note:

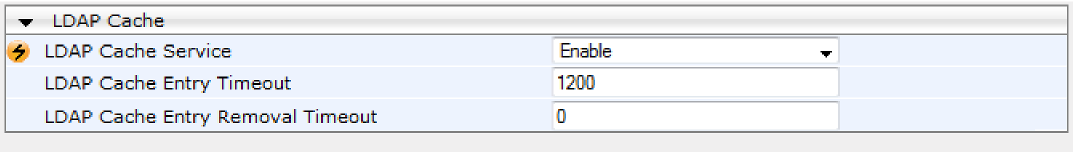
- The LDAP Cache feature is applicable only to LDAP-based SIP queries (Control).
- The maximum LDAP cache size is 10,000 entries.
- The device can save up to six LDAP Attributes in the cache per user (search LDAP key).
- The device also saves in the cache queried Attributes that do not have any values in the LDAP server.

The following procedure describes how to configure the device's LDAP cache through the Web interface. For a full description of the cache parameters, see 'LDAP Parameters' on page 1026.

➢ **To enable and configure the LDAP cache:**

1. Open the LDAP Settings page (**Configuration** tab > **VoIP** menu > **Services** > **LDAP** > **LDAP Settings**).

Figure 15-18: LDAP Settings Page - Cache Parameters



LDAP Cache Service	Enable
LDAP Cache Entry Timeout	1200
LDAP Cache Entry Removal Timeout	0

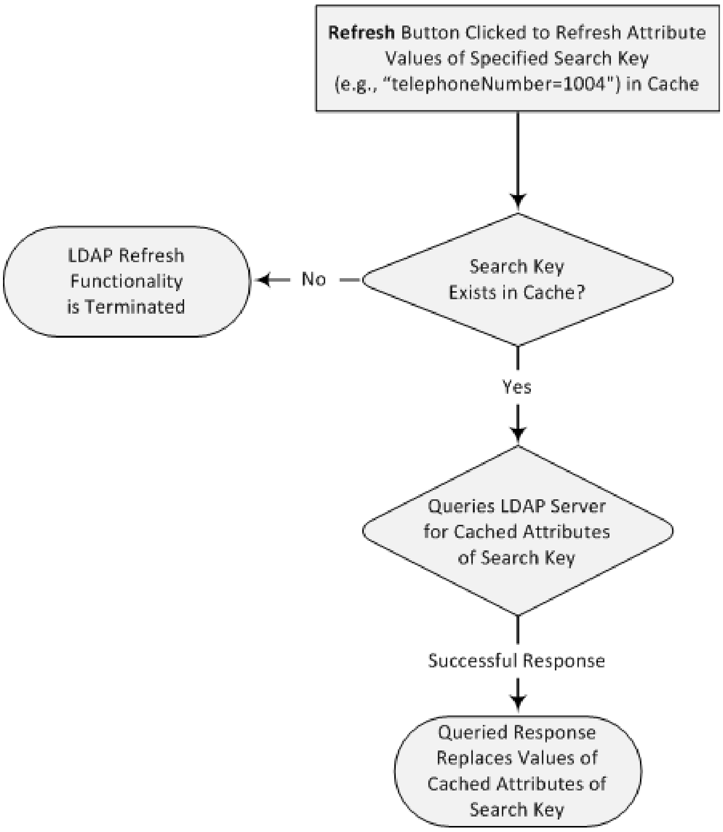
2. Under the Cache group, do the following:
   a. From the 'LDAP Cache Service' drop-down list, select **Enable** to enable LDAP cache.
   b. In the 'LDAP Cache Entry Timeout' field, enter the duration (in minutes) for which an entry in the LDAP cache is valid.
   c. In the 'LDAP Cache Entry Removal Timeout' field, enter the duration (in hours) after which the device removes the LDAP entry from the cache.
3. Click **Apply**, and then reset the device with a save-to-flash for your settings to take effect.

15.4.8.1 Refreshing the LDAP Cache

You can refresh values of LDAP Attributes associated with a specified LDAP search key that are stored in the device's LDAP cache. The device sends an LDAP query to the LDAP server for the cached Attributes of the specified search key and replaces the old values in the cache with the new values received in the LDAP response.

For example, assume the cache contains a previously queried LDAP Attribute "telephoneNumber=1004" whose associated Attributes include "displayName", "mobile" and "ipPhone". If you perform a cache refresh based on the search key "telephoneNumber=1004", the device sends an LDAP query to the server requesting values for the "displayName", "mobile" and "ipPhone" Attributes of this search key. When the device receives the LDAP response, it replaces the old values in the cache with the new values received in the LDAP response.

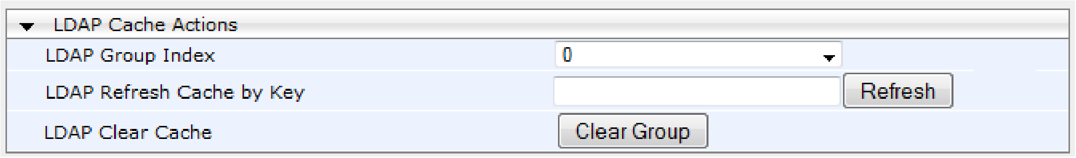
Figure 15-19: LDAP Cache Refresh Flowchart


Refresh Button Clicked to Refresh Attribute Values of Specified Search Key (e.g., "telephoneNumber=1004") in Cache
Search Key Exists in Cache?
No
LDAP Refresh Functionality is Terminated
Yes
Queries LDAP Server for Cached Attributes of Search Key
Successful Response
Queried Response Replaces Values of Cached Attributes of Search Key

➢ **To refresh the LDAP cache per LDAP Server Group:**

1. Open the LDAP Settings page (**Configuration** tab > **VoIP** menu > **Services** > **LDAP** > **LDAP Settings**).

Figure 15-20: Refreshing LDAP Cache



▾ LDAP Cache Actions

LDAP Group Index	0 ▾
LDAP Refresh Cache by Key	[ ] Refresh
LDAP Clear Cache	Clear Group

2. Under the Cache Actions group, do the following:
   a. From the 'LDAP Group Index' drop-down list, select the required LDAP Server Group (see 'Configuring LDAP Server Groups' on page 234).
   b. In the 'LDAP Refresh Cache by Key' field, enter the LDAP search key that you want to refresh (e.g., telephoneNumber=1004).
   c. Click **Refresh**; if a request with the specified key exists in the cache, a request is sent to the LDAP server for the Attributes associated in the cache with the search key.

15.4.8.2 Clearing the LDAP Cache

You can remove (clear) all LDAP entries in the device's LDAP cache for a specific LDAP Server Group, as described in the following procedure.

➢ To clear the LDAP cache:

1. Open the LDAP Settings page (**Configuration** tab > **VoIP** menu > **Services** > **LDAP** > **LDAP Settings**).
2. Under the Cache Actions group, do the following:
   a. From the 'LDAP Group Index' drop-down list, select the required LDAP Server Group (see 'Configuring LDAP Server Groups' on page 234).
   b. Click **Clear Group**.

15.4.9 Configuring Local Database for Management User Authentication

You can configure the device to use its local database (Web Users table) to authenticate management users based on the username-password combination. You can configure the device to use the Web Users table upon the following scenarios:

- LDAP or RADIUS server is not configured (or broken connection), or always use the Web Users table and only if the user is not found, to use the server.
- Connection with the LDAP or RADIUS server fails due to a timeout. In such a scenario, the device can deny access or verify the user's credentials (username-password) locally in the Web Users table.

If user authentication using the Web Users table succeeds, the device grants management access to the user; otherwise access is denied. The access level assigned to the user is also determined by the Web Users table. To configure local Web/CLI users in the Web Users table, see "Configuring Web User Accounts" on page 70.

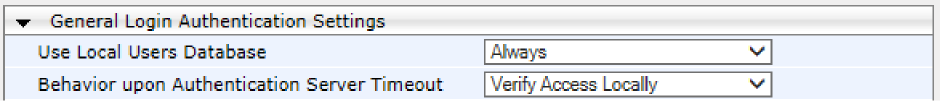
Notes:

- This feature is applicable to LDAP and RADIUS servers.
- This feature is applicable only to user management authentication.

➢ To use the Web Users table for authenticating management users:

1. Open the Authentication Settings page (**Configuration** tab > **System** menu > **Management** > **Authentication Settings**).

Figure 15-21: Authentication Settings Page - Local Database for Login Authentication



General Login Authentication Settings

Use Local Users Database	Always
Behavior upon Authentication Server Timeout	Verify Access Locally

2. Under General Login Authentication Settings:
   - Configure when the Web Users table must be used to authenticate login users. From the 'Use Local Users Database' drop-down list, select one of the following:
     - **When No Auth Server Defined (default):** When no LDAP/RADIUS server is configured or if a server is configured but connectivity with the server is down (if the server is up, the device authenticates the user with the server).
     - **Always:** First attempts to authenticate the user using the Web Users table, but if not found, it authenticates the user with the LDAP/RADIUS server.

- Configure whether the Web Users table must be used to authenticate login users upon connection timeout with the server. From the 'Behavior upon Authentication Server Timeout' drop-down list, select one of the following:
  - ♦ **Deny Access:** User is denied access to the management platform.
  - ♦ **Verify Access Locally (default):** The device verifies the user's credentials in the Web Users table.

3. Click **Submit**.

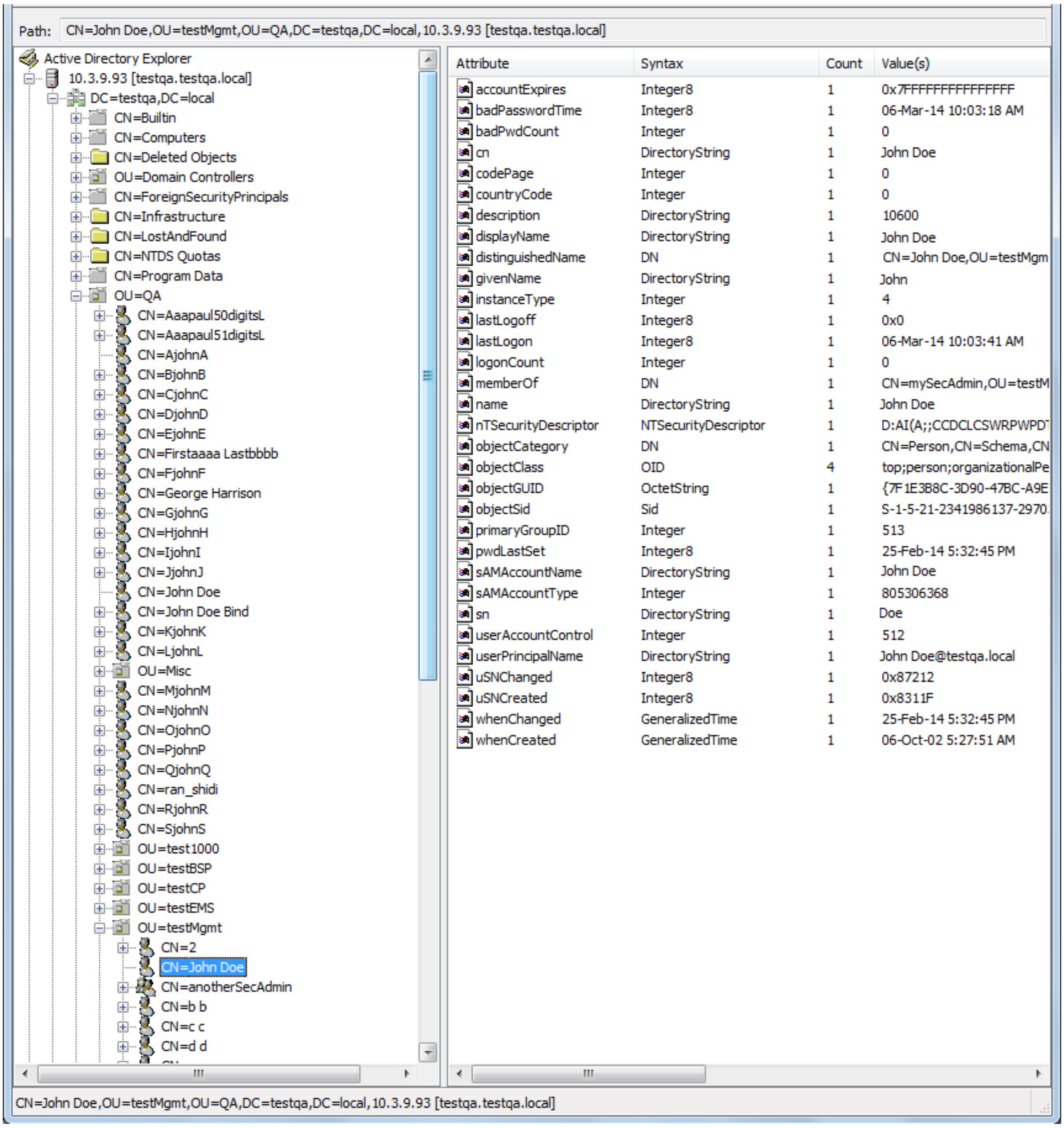
15.4.10 LDAP-based Login Authentication Example

To facilitate your understanding on LDAP entry data structure and how to configure the device to use and obtain information from this LDAP directory, a brief configuration example is described in this section. The example applies to LDAP-based user login authentication and authorization (access level), and assumes that you are familiar with other aspects of LDAP configuration (e.g., LDAP server's address).

The LDAP server's entry data structure schema in the example is as follows:

- **DN (base path):** OU=testMgmt,OU=QA,DC=testqa,DC=local. The DN path to search for the username in the directory is shown below:

Figure 15-22: Base Path (DN) in LDAP Server



Path: CN=John Doe,OU=testMgmt,OU=QA,DC=testqa,DC=local,10.3.9.93 [testqa.testqa.local]

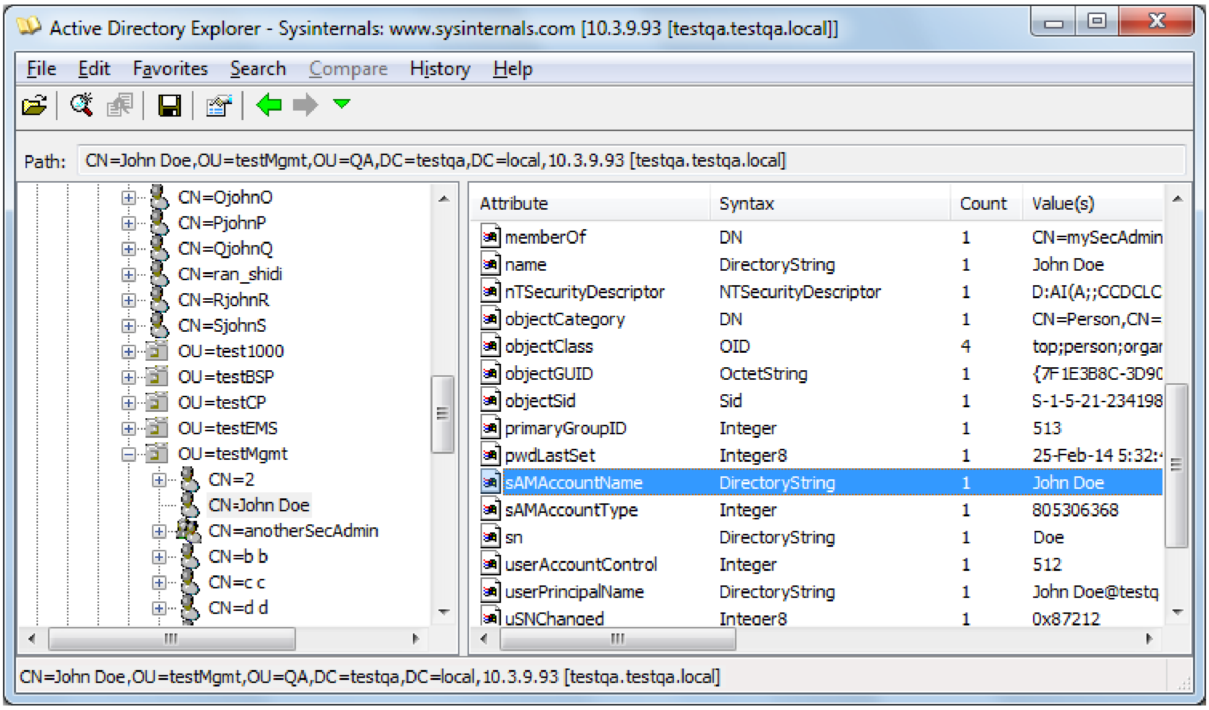
```
Active Directory Explorer
  10.3.9.93 [testqa.testqa.local]
    DC=testqa,DC=local
      CN=Builtin
      CN=Computers
      CN=Deleted Objects
      OU=Domain Controllers
      CN=ForeignSecurityPrincipals
      CN=Infrastructure
      CN=LostAndFound
      CN=NTDS Quotas
      CN=Program Data
      OU=QA
        CN=Aaapaul50digitsL
        CN=Aaapaul51digitsL
        CN=AjohnA
        CN=BjohnB
        CN=CjohnC
        CN=DjohnD
        CN=EjohnE
        CN=Firstaaaa Lastbbbb
        CN=FjohnF
        CN=George Harrison
        CN=GjohnG
        CN=HjohnH
        CN=IjohnI
        CN=JjohnJ
        CN=John Doe
        CN=John Doe Bind
        CN=KjohnK
        CN=LjohnL
        OU=Misc
        CN=MjohnM
        CN=NjohnN
        CN=OjohnO
        CN=PjohnP
        CN=QjohnQ
        CN=ran_shidi
        CN=RjohnR
        CN=SjohnS
        OU=test1000
        OU=testBSP
        OU=testCP
        OU=testEMS
        OU=testMgmt
          CN=2
          CN=John Doe
          CN=anotherSecAdmin
          CN=b b
          CN=c c
          CN=d d
```

Attribute	Syntax	Count	Value(s)
accountExpires	Integer8	1	0x7FFFFFFFFFFFFFFF
badPasswordTime	Integer8	1	06-Mar-14 10:03:18 AM
badPwdCount	Integer	1	0
cn	DirectoryString	1	John Doe
codePage	Integer	1	0
countryCode	Integer	1	0
description	DirectoryString	1	10600
displayName	DirectoryString	1	John Doe
distinguishedName	DN	1	CN=John Doe,OU=testMgm
givenName	DirectoryString	1	John
instanceType	Integer	1	4
lastLogoff	Integer8	1	0x0
lastLogon	Integer8	1	06-Mar-14 10:03:41 AM
logonCount	Integer	1	0
memberOf	DN	1	CN=mySecAdmin,OU=testM
name	DirectoryString	1	John Doe
nTSecurityDescriptor	NTSecurityDescriptor	1	D:AI(A;;CCDCLCSWRPWPDT
objectCategory	DN	1	CN=Person,CN=Schema,CN
objectClass	OID	4	top;person;organizationalPe
objectGUID	OctetString	1	{7F1E3B8C-3D90-47BC-A9E
objectSid	Sid	1	S-1-5-21-2341986137-2970
primaryGroupID	Integer	1	513
pwdLastSet	Integer8	1	25-Feb-14 5:32:45 PM
sAMAccountName	DirectoryString	1	John Doe
sAMAccountType	Integer	1	805306368
sn	DirectoryString	1	Doe
userAccountControl	Integer	1	512
userPrincipalName	DirectoryString	1	John Doe@testqa.local
uSNChanged	Integer8	1	0x87212
uSNCreated	Integer8	1	0x8311F
whenChanged	GeneralizedTime	1	25-Feb-14 5:32:45 PM
whenCreated	GeneralizedTime	1	06-Oct-02 5:27:51 AM

CN=John Doe,OU=testMgmt,OU=QA,DC=testqa,DC=local,10.3.9.93 [testqa.testqa.local]

- **Search Attribute Filter:** (sAMAccountName=$). The login username is found based on this attribute (where the attribute's value equals the username):

Figure 15-23: Username Found using sAMAccount Attribute Search Filter


Active Directory Explorer - Sysinternals: www.sysinternals.com [10.3.9.93 [testqa.testqa.local]]
File
Edit
Favorites
Search
Compare
History
Help

Path: CN=John Doe,OU=testMgmt,OU=QA,DC=testqa,DC=local,10.3.9.93 [testqa.testqa.local]

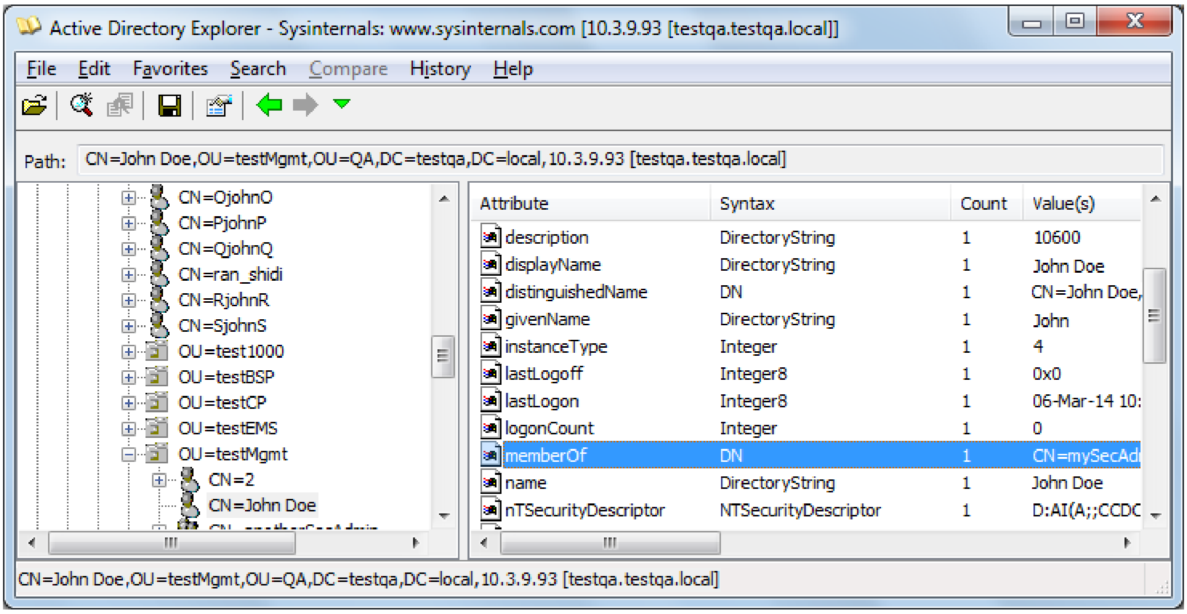
- CN=OjohnO
- CN=PjohnP
- CN=QjohnQ
- CN=ran_shidi
- CN=RjohnR
- CN=SjohnS
- OU=test1000
- OU=testBSP
- OU=testCP
- OU=testEMS
- OU=testMgmt
  - CN=2
  - CN=John Doe
  - CN=anotherSecAdmin
  - CN=b b
  - CN=c c
  - CN=d d

Attribute	Syntax	Count	Value(s)
memberOf	DN	1	CN=mySecAdmin
name	DirectoryString	1	John Doe
nTSecurityDescriptor	NTSecurityDescriptor	1	D:AI(A;;CCDCLC
objectCategory	DN	1	CN=Person,CN=
objectClass	OID	4	top;person;orgar
objectGUID	OctetString	1	{7F1E3B8C-3D90
objectSid	Sid	1	S-1-5-21-234198
primaryGroupID	Integer	1	513
pwdLastSet	Integer8	1	25-Feb-14 5:32:
sAMAccountName	DirectoryString	1	John Doe
sAMAccountType	Integer	1	805306368
sn	DirectoryString	1	Doe
userAccountControl	Integer	1	512
userPrincipalName	DirectoryString	1	John Doe@testq
uSNChanged	Integer8	1	0x87212

CN=John Doe,OU=testMgmt,OU=QA,DC=testqa,DC=local,10.3.9.93 [testqa.testqa.local]

- **Management Attribute:** memberOf. The attribute contains the member groups of the user:

Figure 15-24: User's memberOf Attribute


Active Directory Explorer - Sysinternals: www.sysinternals.com [10.3.9.93 [testqa.testqa.local]]
File
Edit
Favorites
Search
Compare
History
Help

Path: CN=John Doe,OU=testMgmt,OU=QA,DC=testqa,DC=local,10.3.9.93 [testqa.testqa.local]

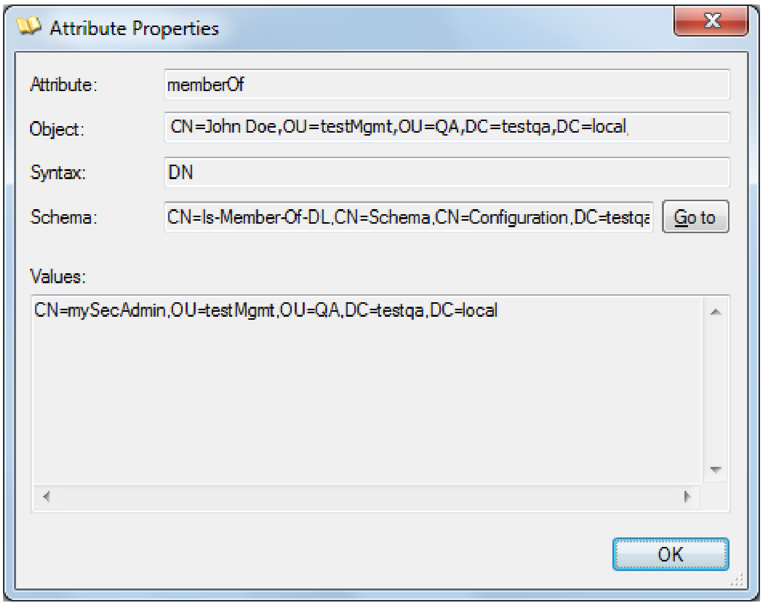
- CN=OjohnO
- CN=PjohnP
- CN=QjohnQ
- CN=ran_shidi
- CN=RjohnR
- CN=SjohnS
- OU=test1000
- OU=testBSP
- OU=testCP
- OU=testEMS
- OU=testMgmt
  - CN=2
  - CN=John Doe

Attribute	Syntax	Count	Value(s)
description	DirectoryString	1	10600
displayName	DirectoryString	1	John Doe
distinguishedName	DN	1	CN=John Doe,
givenName	DirectoryString	1	John
instanceType	Integer	1	4
lastLogoff	Integer8	1	0x0
lastLogon	Integer8	1	06-Mar-14 10:
logonCount	Integer	1	0
memberOf	DN	1	CN=mySecAd
name	DirectoryString	1	John Doe
nTSecurityDescriptor	NTSecurityDescriptor	1	D:AI(A;;CCDC

CN=John Doe,OU=testMgmt,OU=QA,DC=testqa,DC=local,10.3.9.93 [testqa.testqa.local]

- **Management Group:** mySecAdmin. The group to which the user belongs, as listed under the memberOf attribute:

Figure 15-25: User's mySecAdmin Group in memberOf Management Attribute



Attribute Properties

Attribute: memberOf

Object: CN=John Doe,OU=testMgmt,OU=QA,DC=testqa,DC=local

Syntax: DN

Schema: CN=Is-Member-Of-DL,CN=Schema,CN=Configuration,DC=testqa

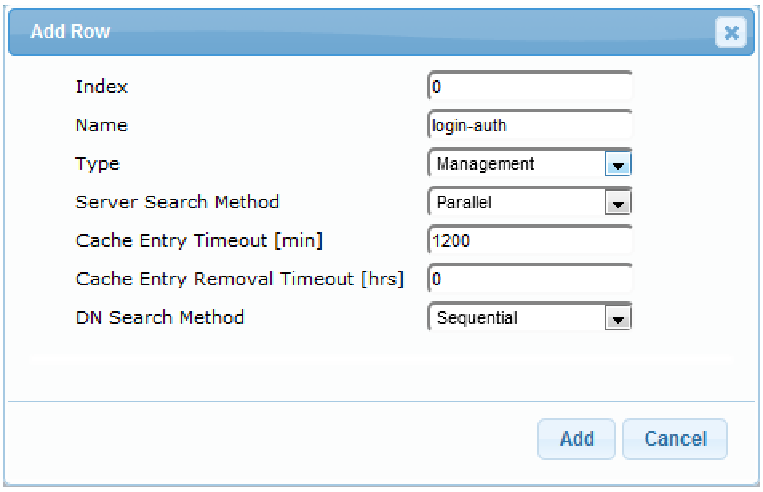
Values:

CN=mySecAdmin,OU=testMgmt,OU=QA,DC=testqa,DC=local

The configuration to match the above LDAP data structure schema is as follows:

- LDAP-based login authentication (management) is enabled in the LDAP Server Groups table (see "Configuring LDAP Server Groups" on page 234):

Figure 15-26: Configuring LDAP Server Group for Management

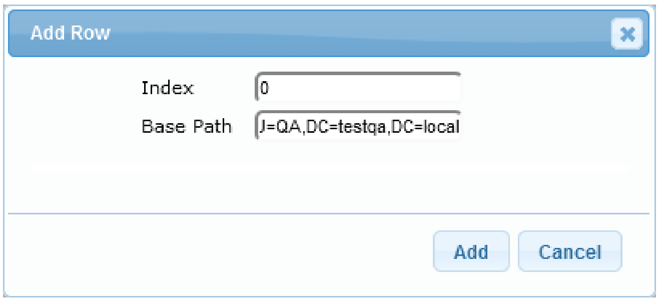


Add Row

Index	0
Name	login-auth
Type	Management
Server Search Method	Parallel
Cache Entry Timeout [min]	1200
Cache Entry Removal Timeout [hrs]	0
DN Search Method	Sequential

- The DN is configured in the LDAP Server Search Base DN table (see "Configuring LDAP DNs (Base Paths) per LDAP Server" on page 241):

Figure 15-27: Configuring DN

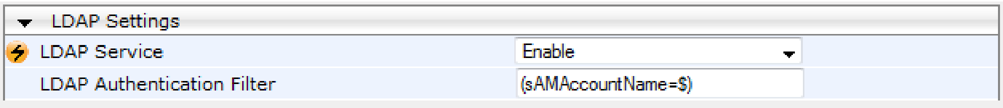


Add Row

Index	0
Base Path	J=QA,DC=testqa,DC=local

- The search attribute filter based on username is configured by the 'LDAP Authentication Filter' parameter in the LDAP Settings page (see "Configuring the LDAP Search Filter Attribute" on page 242):

Figure 15-28: Configuring Search Attribute Filter

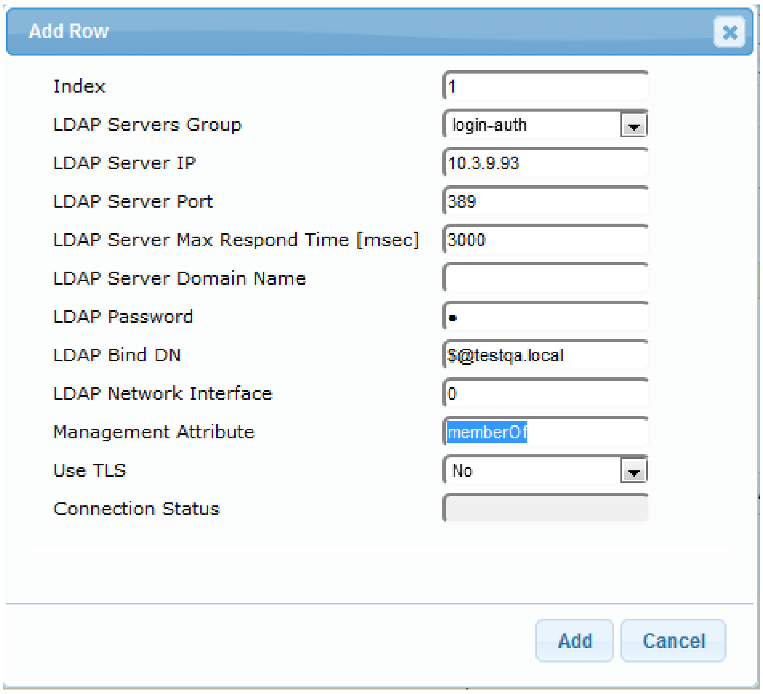


LDAP Settings

LDAP Service	Enable
LDAP Authentication Filter	(sAMAccountName=$)

- The group management attribute is configured by the 'Management Attribute' parameter in the LDAP Configuration table:

Figure 15-29: Configuring Management Attribute

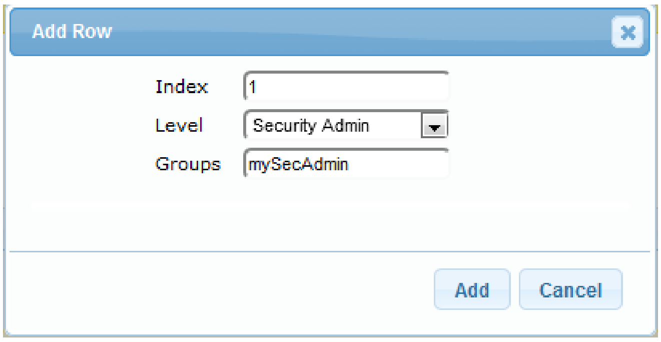


Add Row

Index	1
LDAP Servers Group	login-auth
LDAP Server IP	10.3.9.93
LDAP Server Port	389
LDAP Server Max Respond Time [msec]	3000
LDAP Server Domain Name	
LDAP Password	•
LDAP Bind DN	$@testqa.local
LDAP Network Interface	0
Management Attribute	memberOf
Use TLS	No
Connection Status	

- The management group and its corresponding access level is configured in the Management LDAP Groups table (see "Configuring Access Level per Management Groups Attributes" on page 243):

Figure 15-30: Configuring Management Group Attributes for Determining Access Level



Add Row

Index	1
Level	Security Admin
Groups	mySecAdmin

15.4.11 Enabling LDAP Searches for Numbers with Characters

Typically, the device performs LDAP searches in the AD for complete numbers where the digits are adjacent to one another (e.g., 5038234567). However, if the number is defined in the AD with characters (such as spaces, hyphens and periods) separating the digits (e.g., 503-823 4567), the LDAP query returns a failed result.

To enable the device to search the AD for numbers that may contain characters between its digits, you need to specify the Attribute (up to five) for which you want to apply this functionality, using the LDAPNumericAttributes parameter. For example, the telephoneNumber Attribute could be defined in AD with the telephone number "503-823-4567" (i.e., hyphens), "503.823.4567" (i.e., periods) or "503 823 4567" (i.e., spaces). If the device performs an LDAP search on this Attribute for the number 5038234567, the LDAP query will return results only if you configure the LDAPNumericAttributes parameter with the telephoneNumber Attribute (e.g., LDAPNumericAttributes=telephoneNumber). To search for the number with characters, the device inserts the asterisk (*) wildcard between all digits in the LDAP query (e.g., telephoneNumber = 5*0*3*8*2*3*4*5*6*7). As the AD server recognizes the * wildcard as representing any character, it returns all possible results to the device. Note that the wildcard represents only a character; a query result containing a digit in place of a wildcard is discarded and the device performs another query for the same Attribute. For example, it may return the numbers 533-823-4567 (second digit "3" and hyphens) and 503-823-4567. As the device discards query results where the wildcard results in a digit, it selects 503-823-4567 as the result. The correct query result is cached by the device for subsequent queries and/or in case of LDAP server failure.

15.4.12 Active Directory-based Routing for Microsoft Lync

Typically, enterprises wishing to deploy the Microsoft® Lync™ Server are faced with a complex, call routing dial plan when migrating users from their existing PBX or IP PBX to the Lync Server platform. As more and more end-users migrate to the new voice system, dialing plan management and PBX link capacity can be adversely impacted. To resolve this issue, enterprises can employ Microsoft's Active Directory (AD), which provides a central database to manage and maintain information regarding user's availability, presence, and location.

The device supports outbound IP call routing decisions based on information stored on the AD. Based on queries sent to the AD, the device can route the call to one of the following IP domains:

- Lync client - users connected to Lync Server through the Mediation Server
- PBX or IP PBX - users not yet migrated to Lync Server
- Mobile - mobile number
- Private - private telephone line for Lync users (in addition to the primary telephone line)

15.4.12.1 Querying the AD and Routing Priority

The device queries the AD using the initial destination number (i.e., called number). The query can return up to four user phone numbers, each pertaining to one of the IP domains (i.e., private number, Lync number, PBX / IP PBX number, and mobile number). The configuration parameters listed in the table below are used to configure the query attribute keys that defines the AD attribute that you wish to query in the AD:

Table 15-13: Parameters for Configuring Query Attribute Key

Parameter	Queried User Domain (Attribute) in AD	Query or Query Result Example
MSLDAPPBXNumAttributeName	PBX or IP PBX number (e.g., "telephoneNumber" - default)	telephoneNumber= +3233554447
MSLDAPOCSNumAttributeName	Mediation Server / Lync client number (e.g., "msRTCSIP-line")	msRTCSIP-line=john.smith@company.com
MSLDAPMobileNumAttributeName	Mobile number (e.g., "mobile")	mobile=+3247647156

Parameter	Queried User Domain (Attribute) in AD	Query or Query Result Example
MSLDAPPrivateNumAttributeName	Any attribute (e.g., "msRTCSIP-PrivateLine") **Note:** Used only if set to same value as Primary or Secondary key.	msRTCSIP-PrivateLine= +3233554480
MSLDAPPrimaryKey	Primary Key query search instead of PBX key - can be any AD attribute	msRTCSIP-PrivateLine= +3233554480
MSLDAPSecondaryKey	Secondary Key query key search if Primary Key fails - can be any attribute	-

The process for querying the AD and subsequent routing based on the query results is as follows:

1. If the Primary Key is configured, it uses the defined string as a primary key instead of the one defined in MSLDAPPBXNumAttributeName. It requests the attributes which are described below.
2. If the primary query is not found in the AD and the Secondary Key is configured, it does a second query for the destination number using a second AD attribute key name, configured by the MSLDAPSecondaryKey parameter.
3. If none of the queries are successful, it routes the call to the original dialed destination number according to the routing rule matching the "LDAP_ERR" destination prefix number value, or rejects the call with a SIP 404 "Not Found" response.
4. For each query (primary or secondary), it queries the following attributes (if configured):
   - MSLDAPPBXNumAttributeName
   - MSLDAPOCSNumAttributeName
   - MSLDAPMobileNumAttributeName

   In addition, it queries the special attribute defined in MSLDAPPrivateNumAttributeName, only if the query key (primary or secondary) is equal to its value.
5. If the query is found: The AD returns up to four attributes - Lync, PBX / IP PBX, private (only if it equals Primary or Secondary key), and mobile.
6. The device adds unique prefix keywords to the query results in order to identify the query type (i.e., IP domain). These prefixes are used as the prefix destination number value in the Tel-to-IP Routing table to denote the IP domains:
   - "PRIVATE" (PRIVATE:<private_number>): used to match a routing rule based on query results of the private number (MSLDAPPrivateNumAttributeName)
   - "OCS" (OCS:<Lync_number>): used to match a routing rule based on query results of the Lync client number (MSLDAPOCSNumAttributeName)
   - "PBX" (PBX:<PBX_number>): used to match a routing rule based on query results of the PBX / IP PBX number (MSLDAPPBXNumAttributeName)
   - "MOBILE" (MOBILE:<mobile_number>): used to match a routing rule based on query results of the mobile number (MSLDAPMobileNumAttributeName)
   - "LDAP_ERR": used to match a routing rule based on a failed query result when no attribute is found in the AD

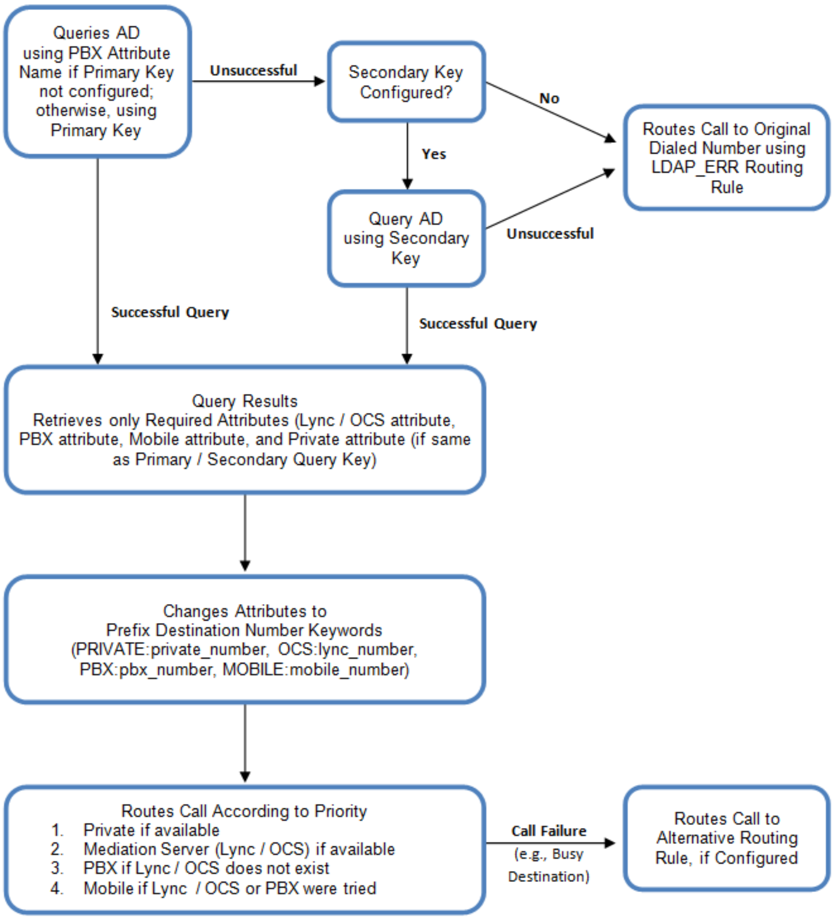
Note: These prefixes are involved only in the routing and manipulation processes; they are not used as the final destination number.

7. The device uses the Tel-to-IP Routing table to route the call based on the LDAP query result. The device routes the call according to the following priority:
   1. **Private line:** If the query is done for the private attribute and it's found, the device routes the call according to this attribute.
   2. **Mediation Server SIP address (Lync):** If the private attribute does not exist or is not queried, the device routes the call to the Mediation Server (which then routes the call to the Lync client).
   3. **PBX / IP PBX:** If the Lync client is not found in the AD, it routes the call to the PBX / IP PBX.
   4. **Mobile number:** If the Lync client (or Mediation Server) is unavailable (e.g., SIP response 404 "Not Found" upon INVITE sent to Lync client), and the PBX / IP PBX is also unavailable, the device routes the call to the user's mobile number (if exists in the AD).
   5. **Alternative route:** If the call routing to all the above fails (e.g., due to unavailable destination - call busy), the device can route the call to an alternative destination if an alternative routing rule is configured.
   6. **"Redundant" route:** If the query failed (i.e., no attribute found in the AD), the device uses the routing rule matching the "LDAP_ERR" prefix destination number value.

Note: For Enterprises implementing a PBX / IP PBX system, but yet to migrate to Lync Server, if the PBX / IP PBX system is unavailable or has failed, the device uses the AD query result for the user’s mobile phone number, routing the call through the PSTN to the mobile destination.

The flowchart below summarizes the device's process for querying the AD and routing the call based on the query results:

Figure 15-31: LDAP Query Flowchart


Queries AD using PBX Attribute Name if Primary Key not configured; otherwise, using Primary Key
Unsuccessful
Secondary Key Configured?
No
Routes Call to Original Dialed Number using LDAP_ERR Routing Rule
Yes
Query AD using Secondary Key
Unsuccessful
Successful Query
Successful Query
Query Results Retrieves only Required Attributes (Lync / OCS attribute, PBX attribute, Mobile attribute, and Private attribute (if same as Primary / Secondary Query Key)
Changes Attributes to Prefix Destination Number Keywords (PRIVATE:private_number, OCS:lync_number, PBX:pbx_number, MOBILE:mobile_number)
Routes Call According to Priority
1. Private if available
2. Mediation Server (Lync / OCS) if available
3. PBX if Lync / OCS does not exist
4. Mobile if Lync / OCS or PBX were tried
Call Failure (e.g., Busy Destination)
Routes Call to Alternative Routing Rule, if Configured

Note: If you are using the device's local LDAP cache, see "Configuring the Device's LDAP Cache" on page 244 for the LDAP query process.

15.4.12.2 Configuring AD-Based Routing Rules

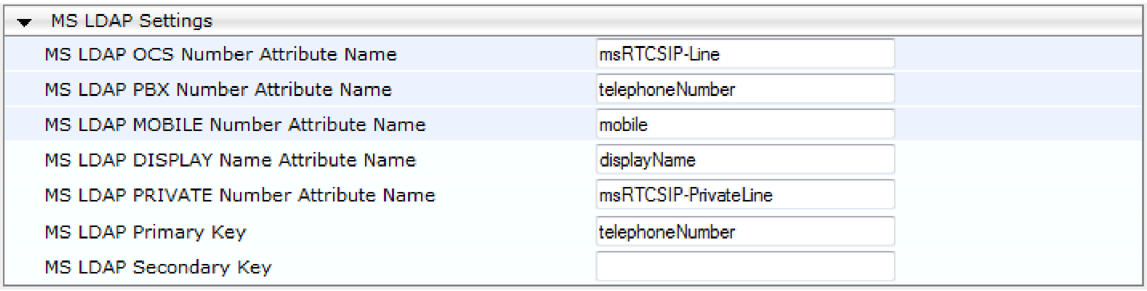
The following procedure describes how to configure outbound IP routing based on LDAP queries.

➢ **To configure LDAP-based IP routing for Lync Server:**

1. Configure the LDAP server parameters, as described in "Configuring LDAP Servers" on page 236.
2. Configure the AD attribute names used in the LDAP query:

a. Open the Advanced Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Advanced Parameters**).

Figure 15-32: LDAP Parameters for Microsoft Lync Server 2010



MS LDAP Settings

MS LDAP OCS Number Attribute Name	msRTCSIP-Line
MS LDAP PBX Number Attribute Name	telephoneNumber
MS LDAP MOBILE Number Attribute Name	mobile
MS LDAP DISPLAY Name Attribute Name	displayName
MS LDAP PRIVATE Number Attribute Name	msRTCSIP-PrivateLine
MS LDAP Primary Key	telephoneNumber
MS LDAP Secondary Key	

b. Configure the LDAP attribute names as desired.

3. Gateway application: Configure AD-based Tel-to-IP routing rules:

   a. Open the Tel-to-IP Routing table (Configuration tab > VoIP menu > Gateway > Routing > Tel to IP Routing). For more information, see Configuring Tel-to-IP Routing Rules on page 467.

   b. Configure query-result routing rules for each IP domain (private, PBX / IP PBX, Lync clients, and mobile), using the LDAP keywords (case-sensitive) for the prefix destination number:

      - PRIVATE: Private number
      - OCS: Lync client number
      - PBX: PBX / IP PBX number
      - MOBILE: Mobile number
      - LDAP_ERR: LDAP query failure

   c. Configure a routing rule for routing the initial Tel call to the LDAP server, using the value "LDAP" for denoting the IP address of the LDAP server.

   d. For alternative routing, enable the alternative routing mechanism and configure corresponding SIP reasons for alternative routing. For this feature, alternative routing starts from the table row located under the LDAP query row.

4. SBC application: Configure AD-based IP-to-IP routing rules:

   a. Open the IP-to-IP Routing table (Configuration tab > VoIP menu > SBC > Routing SBC > IP-to-IP Routing Table). For more information, see Configuring SBC IP-to-IP Routing Rules on page 578.

   b. Configure query-result routing rules for each IP domain (private, PBX / IP PBX, Lync clients, and mobile), using the LDAP keywords (case-sensitive) in the Destination Username Prefix field:

      - PRIVATE: Private number
      - OCS: Lync client number
      - PBX: PBX / IP PBX number
      - MOBILE: Mobile number
      - LDAP_ERR: LDAP query failure

   c. Configure a routing rule for routing the initial call (LDAP query) to the LDAP server, by setting the 'Destination Type' field to LDAP for denoting the IP address of the LDAP server.

   d. For alternative routing, enable the alternative routing mechanism and configure corresponding SIP reasons for alternative routing. For this feature, alternative routing starts from the table row located under the LDAP query row.

The table below shows an example for configuring AD-based Tel-to-IP routing rules in the Tel-to-IP Routing table:

Table 15-14: AD-Based Tel-to-IP Routing Rule Configuration Examples

Index	Destination Phone Prefix	Destination IP Address
1	PRIVATE:	10.33.45.60
2	PBX:	10.33.45.65
3	OCS:	10.33.45.68
4	MOBILE:	10.33.45.100
5	LDAP_ERR	10.33.45.80
6	*	LDAP
7	*	10.33.45.72

The table below shows an example for configuring AD-based SBC routing rules in the IP-to-IP Routing Table:

Table 15-15: AD-Based SBC IP-to-IP Routing Rule Configuration Examples

Index	Destination Username Prefix	Destination Type	Destination Address
1	PRIVATE:	Dest Address	10.33.45.60
2	PBX:	Dest Address	10.33.45.65
3	OCS:	Dest Address	10.33.45.68
4	MOBILE:	Dest Address	10.33.45.100
5	LDAP_ERR	Dest Address	10.33.45.80
6	*	LDAP	
7	*	Dest Address	10.33.45.72

The configured routing rule example is explained below:

- **Rule 1:** Sends call to private telephone line (at 10.33.45.60) upon successful AD query result for the private attribute.
- **Rule 2:** Sends call to IP PBX (at 10.33.45.65) upon successful AD query result for the PBX attribute.
- **Rule 3:** Sends call to Lync client (i.e., Mediation Server at 10.33.45.68) upon successful AD query result for the Lync attribute.
- **Rule 4:** Sends call to user's mobile phone number (to PSTN through the device's IP address at 10.33.45.100) upon successful AD query result for the Mobile attribute.
- **Rule 5:** Sends call to IP address of device (10.33.45.80) if AD query failure (e.g., no response from LDAP server or attribute not found).
- **Rule 6:** Sends query for original destination number of received call to the LDAP server.
- **Rule 7:** Alternative routing rule that sends the call of original dialed number to IP destination 10.33.45.72. This rule is applied in any of the following cases
    - LDAP functionality is disabled.
    - LDAP query is successful but call fails (due to, for example, busy line) to all the relevant attribute destinations (private, Lync, PBX, and mobile), and a relevant Tel-to-IP Release Reason (see Alternative Routing for Tel-to-IP Calls on page 485) or SBC Alternative Routing Reason (see Configuring SIP Response Codes for Alternative Routing Reasons on page 588) has been configured.

Once the device receives the original incoming call, the first rule that it uses is Rule 6, which queries the AD server. When the AD replies, the device searches the table, from the first rule down, for the matching destination phone prefix (i.e., "PRIVATE:, "PBX:", "OCS:", "MOBILE:", and "LDAP_ERR:"), and then sends the call to the appropriate destination.

15.4.12.3 Querying the AD for Calling Name

The device can retrieve the calling name (display name) from an LDAP-compliant server (for example, Microsoft Active Directory / AD) for Tel-to-IP calls that are received without a calling name.

The device uses the calling number (PBX or mobile number) for the LDAP query. Upon an incoming INVITE, the device queries the AD based on the Calling Number search key (tries to match the calling number with the appropriate "telephoneNumber" or "mobile" number AD attribute entry). It then searches for the corresponding calling name attribute, configured by the MSLDAPDisplayNameAttributeName parameter (e.g., “displayName”). The device uses the resultant calling name as the display name parameter in the SIP From header of the outgoing INVITE message.

To configure this feature, the following keywords are used in the Calling Name Manipulation Table for Tel-to-IP Calls table for the 'Prefix/Suffix to Add' fields, which can be combined with other characters:

- "$LDAP-PBX": LDAP query using the MSLDAPPBXAttrName parameter as the search key
- "$LDAP-MOBILE": LDAP query using MSLDAPMobileAttrName parameter as the search key

If the source (calling) number of the Tel-to-IP call matches the PBX / MOBILE (e.g., "telephoneNumber" and "mobile") number in the AD server, the device uses the resultant Display Name instead of the keyword(s).

For example, assume the following configuration in the Calling Name Manipulation Table for Tel-to-IP Calls:

- 'Source Prefix' field is set to "4".
- 'Prefix to Add' field is set to "$LDAP-PBX Office".

If the calling number is 4046 and the resultant LDAP query display name is "John Doe", the device sends the INVITE message with the following From header:

```
From: John Doe <sip:4064@company.com>
```

Notes:

- The Calling Name Manipulation Table for Tel-to-IP Calls table uses the numbers before manipulation, as inputs.
- The LDAP query uses the calling number after source number manipulation, as the search key value.

15.5 Least Cost Routing

This section provides a description of the device's least cost routing (LCR) feature and how to configure it.

15.5.1 Overview

The LCR feature enables the device to choose the outbound IP destination routing rule based on lowest call cost. This is useful in that it enables service providers to optimize routing costs for customers. For example, you may wish to define different call costs for local and international calls or different call costs for weekends and weekdays (specifying even the time of call). The device sends the calculated cost of the call to a Syslog server (as Information messages), thereby enabling billing by third-party vendors.

LCR is implemented by defining Cost Groups and assigning them to routing rules in the Tel-to-IP Routing table (Gateway calls) or IP-to-IP Routing table (SBC calls). The device searches the routing table for matching routing rules and then selects the rule with the lowest call cost. If two routing rules have identical costs, the rule appearing higher up in the table is used (i.e., first-matched rule). If the selected route is unavailable, the device selects the next least-cost routing rule.

Even if a matched routing rule is not assigned a Cost Group, the device can select it as the preferred route over other matched rules that are assigned Cost Groups. This is determined according to the settings of the 'Default Call Cost' parameter configured for the Routing Policy (associated with the routing rule for SBC calls). For configuring the Routing Policy, see Configuring a Gateway Routing Policy Rule on page 482 (for Gateway) and Configuring SBC Routing Policy Rules on page 590 (for SBC).

The Cost Group defines a fixed connection cost (*connection cost*) and a charge per minute (*minute cost*). Cost Groups can also be configured with time segments (*time bands*), which define connection cost and minute cost based on specific days of the week and time of day (e.g., from Saturday through Sunday, between 6:00 and 18:00). If multiple time bands are configured per Cost Group and a call spans multiple time bands, the call cost is calculated using only the time band in which the call was initially established.

In addition to Cost Groups, the device can calculate the call cost using an optional, user-defined average call duration value. The logic in using this option is that a Cost Group may be cheap if the call duration is short, but due to its high minute cost, may prove very expensive if the duration is lengthy. Thus, together with Cost Groups, the device can use this option to determine least cost routing. The device calculates the Cost Group call cost as follows:

*Total Call Cost = Connection Cost + (Minute Cost * Average Call Duration)*

The below table shows an example of call cost when taking into consideration call duration. This example shows four defined Cost Groups and the total call cost if the average call duration is 10 minutes:

Table 15-16: Call Cost Comparison between Cost Groups for different Call Durations

Cost Group	Connection Cost	Minute Cost	Total Call Cost per Duration	
			1 Minute	10 Minutes
A	1	6	7	61
B	0	10	10	100
C	0.3	8	8.3	80.3
D	6	1	7	**16**

If four matching routing rules are located in the routing table and each one is assigned a different Cost Group as listed in the table above, then the rule assigned Cost Group "D" is selected. Note that for one minute, Cost Groups "A" and "D" are identical, but due to the average call duration, Cost Group "D" is cheaper. Therefore, average call duration is an important factor in determining the cheapest routing role.

Below are a few examples of how you can implement LCR:

- **Example 1:** This example uses two different Cost Groups for routing local calls and international calls:

  Two Cost Groups are configured as shown below:

Cost Group	Connection Cost	Minute Cost
1. "Local Calls"	2	1
2. "International Calls"	6	3

The Cost Groups are assigned to routing rules for local and international calls:

Routing Index	Dest Phone Prefix	Destination IP	Cost Group ID
1	2000	x.x.x.x	1 "Local Calls"
2	00	x.x.x.x	2 "International Calls"

- **Example 2:** This example shows how the device determines the cheapest routing rule in the Tel-to-IP Routing table:

  The 'Default Call Cost' parameter in the Routing Policy rule is configured to **Lowest Cost**, meaning that if the device locates other matching routing rules (with Cost Groups assigned), the routing rule without a Cost Group is considered the lowest cost route.

  - The following Cost Groups are configured:

Cost Group	Connection Cost	Minute Cost
1. "A"	2	1
2. "B"	6	3

- The Cost Groups are assigned to routing rules:

Routing Index	Dest Phone Prefix	Destination IP	Cost Group
1	201	x.x.x.x	"A'
2	201	x.x.x.x	"B"
3	201	x.x.x.x	0
4	201	x.x.x.x	"B"

The device calculates the optimal route in the following index order: 3, 1, 2, and then 4, due to the following logic:

- Index 1 - Cost Group "A" has the lowest connection cost and minute cost
- Index 2 - Cost Group "B" takes precedence over Index 4 entry based on the first-matched method rule
- Index 3 - no Cost Group is assigned, but as the 'Default Call Cost' parameter is configured to **Lowest Cost**, it is selected as the cheapest route
- Index 4 - Cost Group "B" is only second-matched rule (Index 1 is the first)

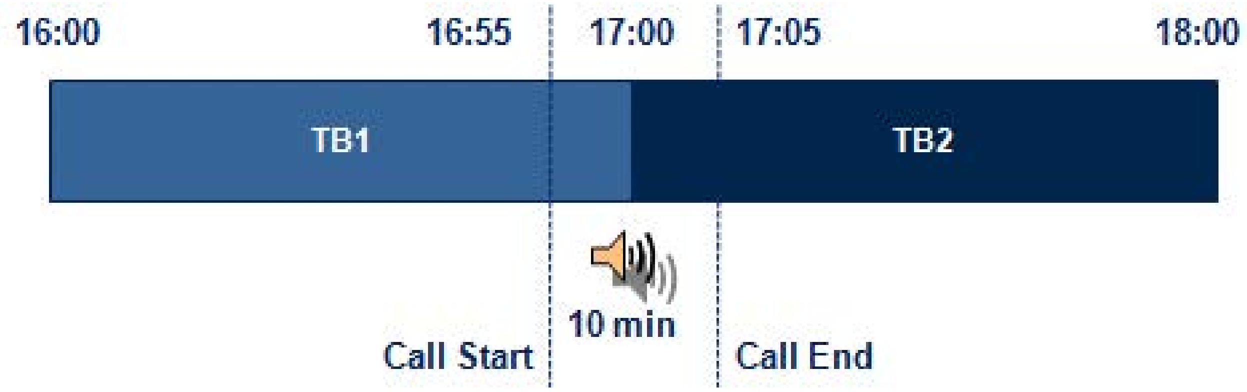
- **Example 3:** This example shows how the cost of a call is calculated if the call spans over multiple time bands:

  Assume a Cost Group, "CG Local" is configured with two time bands, as shown below:

Cost Group	Time Band	Start Time	End Time	Connection Cost	Minute Cost
CG Local	TB1	16:00	17:00	2	1
	TB2	17:00	18:00	7	2

Assume that the call duration is 10 minutes, occurring between 16:55 and 17:05. In other words, the first 5 minutes occurs in time band "TB1" and the next 5 minutes occurs in "TB2", as shown below:

Figure 15-33: LCR using Multiple Time Bands (Example)



The device calculates the call using the time band in which the call was initially established, regardless of whether the call spans over additional time bands:

Total call cost = "TB1" Connection Cost + ("TB1" Minute Cost x call duration) = 2 + 1 x 10 min = 12

15.5.2 Configuring LCR

To configure LCR, perform the following main steps:

1. Enable LCR - see Configuring a Gateway Routing Policy Rule on page 482 (for Gateway) and Configuring SBC Routing Policy Rules on page 590 (for SBC).
2. Configure Cost Groups - see "Configuring Cost Groups" on page 263.
3. Configure Time Bands for a Cost Group - see "Configuring Time Bands for Cost Groups" on page 264.
4. Assign Cost Groups to outbound IP routing rules - see "Assigning Cost Groups to Routing Rules" on page 266.

15.5.2.1 Configuring Cost Groups

The Cost Group table lets you configure Cost Groups. A Cost Group defines a fixed call connection cost and a call rate (charge per minute). Once configured, you can configure Time Bands per Cost Group. Up to 10 Cost Groups can be configured.

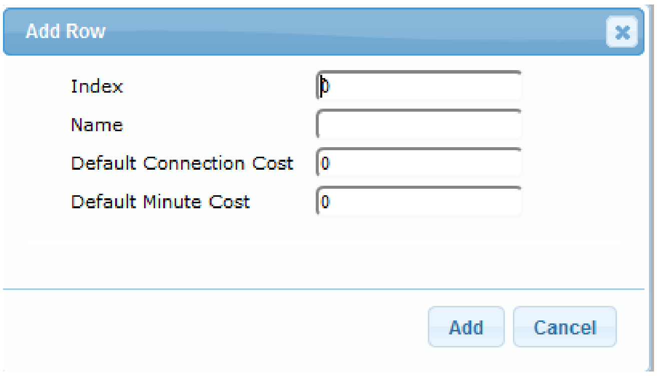
The following procedure describes how to configure Cost Groups through the Web interface. You can also configure it through ini file (CostGroupTable) or CLI (configure voip > services least-cost-routing cost-group).

➢ **To configure a Cost Group:**

1. Open the Cost Group table (**Configuration** tab > **VoIP** menu > **Services** > **Least Cost**

Routing > **Cost Group Table**).

2. Click **Add**; the following dialog box appears:



Add Row

Index: 0
Name:
Default Connection Cost: 0
Default Minute Cost: 0

3. Configure a Cost Group according to the parameters described in the table below.
4. Click **Add**, and then save ("burn") your settings to flash memory.

Table 15-17: Cost Group Table Parameter Descriptions

Parameter	Description
Index [CostGroupTable_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `cost-group-name` [CostGroupTable_CostGroupName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. **Note:** Each Cost Group must have a unique name.
Default Connection Cost `default-connection-cost` [CostGroupTable_DefaultConnectionCost]	Defines the call connection cost (added as a fixed charge to the call) for a call outside the time bands. The valid value range is 0-65533. The default is 0. **Note:** When calculating the cost of a call, if the current time of the call is not within a time band configured for the Cost Group, then this default connection cost is used.
Default Minute Cost `default-minute-cost` [CostGroupTable_DefaultMinuteCost]	Defines the call charge per minute for a call outside the time bands. The valid value range is 0-65533. The default is 0. **Note:** When calculating the cost of a call, if the current time of the call is not within a time band configured for the Cost Group, then this default charge per minute is used.

15.5.2.2 Configuring Time Bands for Cost Groups

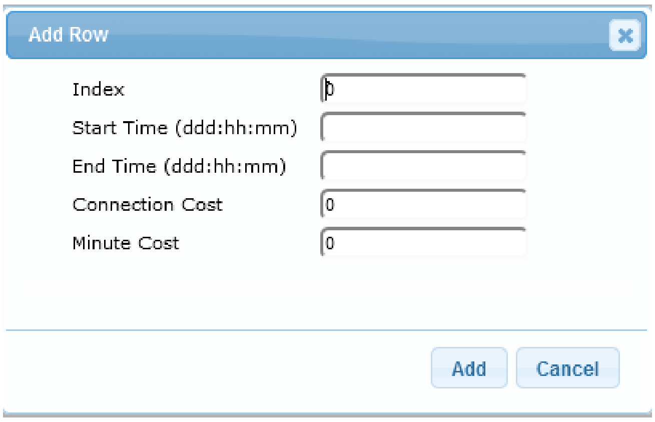
The Time Band table lets you configure Time Bands per Cost Group. A Time Band defines a day and time range (e.g., from Saturday 05:00 to Sunday 24:00), as well as the fixed call connection charge and call rate per minute for this interval. You can configure up to 70 Time Bands, where up to 21 Time Bands can be assigned to each Cost Group.

Note: You cannot configure overlapping Time Bands.

The following procedure describes how to configure Time Bands per Cost Group through the Web interface. You can also configure it through ini file (CostGroupTimebands) or CLI (configure voip >services least-cost-routing cost-group-time-bands).

➢ To configure a Time Band per Cost Group:

1. Open the Cost Group table (**Configuration** tab > **VoIP** menu > **Services** > **Least Cost Routing** > **Cost Group Table**).
2. Select a Cost Group for which you want to assign Time Bands, and then click the **Time Band** link located below the table; the Time Band table for the selected Cost Group appears.
3. Click **Add**; the following dialog box appears:



Add Row

Index	0
Start Time (ddd:hh:mm)	
End Time (ddd:hh:mm)	
Connection Cost	0
Minute Cost	0

4. Configure a Time Band according to the parameters described in the table below.
5. Click **Add**, and then save ("burn") your settings to flash memory.

Table 15-18: Time Band Table Description

Parameter	Description
Index `timeband-index` [CostGroupTimebands_TimebandIndex]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Start Time `start-time` [CostGroupTimebands_StartTime]	Defines the day and time of day from when this time band is applicable. The format is DDD:hh:mm, where: ▪ *DDD* is the day of the week, represented by the first three letters of the day in upper case (i.e., SUN, MON, TUE, WED, THU, FRI, or SAT). ▪ *hh* and *mm* denote the time of day, where *hh* is the hour (00-23) and *mm* the minutes (00-59) For example, SAT:22:00 denotes Saturday at 10 pm.
End Time `end-time` [CostGroupTimebands_EndTime]	Defines the day and time of day until when this time band is applicable. For a description of the valid values, see the parameter above.
Connection Cost `connection-cost` [CostGroupTimebands_ConnectionCost]	Defines the call connection cost during this time band. This is added as a fixed charge to the call. The valid value range is 0-65533. The default is 0. **Note:** The entered value must be a whole number (i.e., not a decimal).

Parameter	Description
Minute Cost `minute-cost` [CostGroupTimebands_MinuteCost]	Defines the call cost per minute charge during this timeband. The valid value range is 0-65533. The default is 0. **Note:** The entered value must be a whole number (i.e., not a decimal).

15.5.2.3 Assigning Cost Groups to Routing Rules

To use your configured Cost Groups, you need to assign them to routing rules:

- Gateway application: Tel-to-IP Routing table - see Configuring Tel-to-IP Routing Rules on page 467
- SBC application: IP-to-IP Routing table - see Configuring SBC IP-to-IP Routing Rules on page 578

15.6 HTTP-based Remote Services

15.6.1 Configuring HTTP Services

The HTTP Remote Services table lets you configure up to seven HTTP-based services provided by third-party remote hosts (e.g., routing server). The following types of services can be offered by the remote host:

- **Routing:** Call routing service, whereby the remote host (e.g., routing server) determines the next hop of an incoming call on the path to the final destination. For more information on employing a third-party, remote routing server, see ''Centralized Third-Party Routing Server or ARM'' on page 272.
- **Call Status:** Call status of calls processed by the device. The call status is provided to the remote host through CDRs sent by the device.
- **Topology Status:** Status of device configuration (add, edit and delete). The device sends topology status to the HTTP host, using the REST TopologyStatus API command. To enable the functionality, configure the 'Topology Status' (RoutingServerGroupStatus) parameter to **Enable**. The parameter is located below the table.

  Topology status includes the following:

  - IP Groups: status is reported when the keep-alive mechanism (enabled for the associated Proxy Set) detects that the IP Group is unavailable, or when CAC thresholds (configured in the Admission Control table) are crossed.
  - Trunk Groups: status is reported when the trunk's physical state indicates that the trunk is unavailable.
  - Status is reported when IP Groups, Trunk Groups or SIP Interfaces that are configured to be used by HTTP-based services (i.e., the UsedByRoutingServer parameter is set to 1 - Used) are created or deleted. If you subsequently change the settings of the UsedByRoutingServer parameter or the 'Name' parameter, the device reports the change as a creation or deletion of the corresponding configuration entity.
- **Capture:** Recording of signaling and RTP packets, and Syslog. The remote host can be, for example, a Syslog server or AudioCodes SEM.

Notes:

- You can configure only **one** HTTP Remote Service entry for Routing, for Call Status, and for Topology. However, you can configure up to four HTTP Remote Services for Capture.
- The Routing service also includes the Call Status and Topology Status services.
- Currently, the Capture service is not supported.
- The device supports HTTP redirect responses (3xx) only during connection establishment with the host. Upon receipt of a redirect response, the device attempts to open a new socket with the host and if this is successful, closes the current connection.

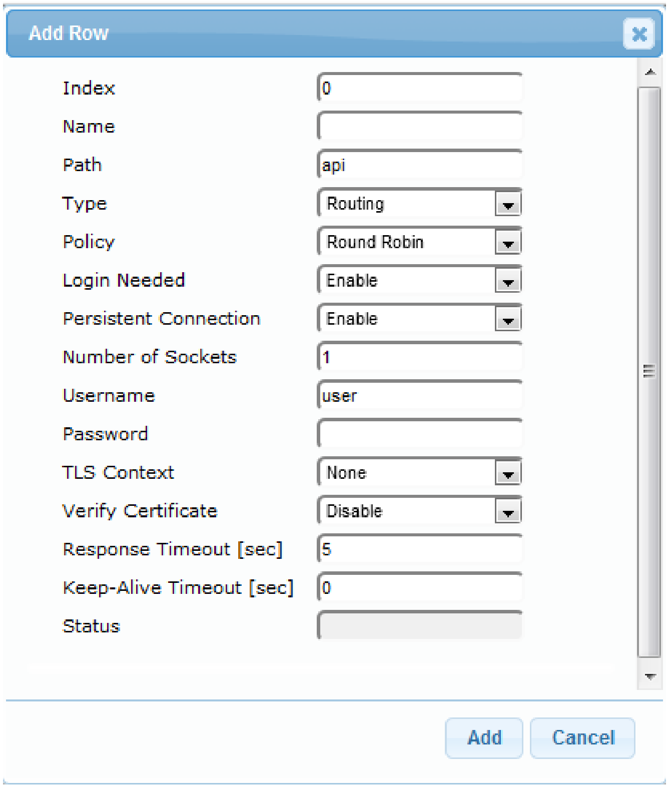
The following procedure describes how to configure HTTP Remote Services through the Web interface. You can also configure it through ini file (HTTPRemoteServices).

➢ **To configure an HTTP-based service**

1. Open the HTTP Remote Services table (**Configuration** tab > **VoIP** menu > **Services** > **HTTP Services** > **HTTP Remote Services**).

2. Click **Add**; the following dialog box appears:

Figure 15-34: HTTP Remote Services Table - Add Row Dialog Box



Index	0
Name	
Path	api
Type	Routing
Policy	Round Robin
Login Needed	Enable
Persistent Connection	Enable
Number of Sockets	1
Username	user
Password	
TLS Context	None
Verify Certificate	Disable
Response Timeout [sec]	5
Keep-Alive Timeout [sec]	0
Status	

3. Configure an HTTP remote service according to the parameters described in the table below.
4. Click **Add**, and then save ("burn") your settings to flash memory.

Table 15-19: HTTP Remote Services Table Parameter Descriptions

Parameter	Description
Index [HTTPRemoteServices_Index]	Defines an index number for the new table row. **Notes:** ▪ Each row must be configured with a unique index. ▪ The parameter is mandatory.
Name [HTTPRemoteServices_Name]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. **Notes:** ▪ Each row must be configured with a unique name. ▪ The parameter is mandatory.
Path [HTTPRemoteServices_Path]	Defines the path (prefix) to the REST APIs. The valid value is a string of up to 80 characters. The default is "api".

Parameter	Description
Type [HTTPRemoteServices_HTTPType]	Defines the type of service provided by the HTTP remote host: ▪ [0] Routing (default) = Routing service (also includes Call Status and Topology Status). ▪ [1] Call Status = Call status service. ▪ [2] Topology Status = Topology status service (e.g., change in configuration). ▪ [3] Capture = Recording of signaling and RTP packets, which can be sent to a remote host, for example, to a Syslog server or AudioCodes SEM. **Notes:** ▪ You can only configure one HTTP service for each of the following service types: Routing, Call Status and Topology Status. ▪ For the **Topology Status** option to be functional, you must enable the RoutingServerGroupStatus parameter. ▪ Currently, the **Capture** option is not supported.
Policy [HTTPRemoteServices_Policy]	Defines the mode of operation when you have configured multiple remote hosts (in the HTTP Remote Hosts table) for a specific HTTP service. ▪ [0] Round Robin = (Default) Load balancing of traffic across all configured hosts. Every consecutive message is sent to the next available host. ▪ [1] Sticky Primary = Device always attempts to send traffic to the first (primary) host. If the host does not respond, the device sends the traffic to the next available host. If the primary host becomes available again, the device sends the traffic to the primary host. ▪ [2] Sticky Next = Similar to **Sticky Primary**, but if the primary host does not respond, the device sends the traffic to the next available host and continues sending traffic to this host even if the primary host becomes available again.
Login Needed [HTTPRemoteServices_LoginNeeded]	Enables the use of proprietary REST API Login and Logout commands for connecting to the remote host. The commands verify specific information (e.g., software version) before allowing connectivity with the device. ▪ [0] Disable = Commands are not used. ▪ [1] Enable (default)
Persistent Connection [HTTPRemoteServices_PersistentConnection]	Defines whether the HTTP connection with the host remains open or is only opened per request. ▪ [0] Disable = Connection is not persistent and closes when the device detects inactivity. The device uses HTTP keep-alive messages to detect inactivity. ▪ [1] Enable = (Default) Connection remains open (persistent) even during inactivity. The device uses HTTP keep-alive / HTTP persistent connection messages to keep the connection open.
Number of Sockets [HTTPRemoteServices_NumOfSockets]	Defines how many sockets (connection) are established per remote host. The valid value is 1 to 10. The default is 1.

Parameter	Description
Username [HTTPRemoteServices_AuthUserName]	Defines the username for HTTP authentication. The valid value is a string of up to 80 characters. The default is "user".
Password [HTTPRemoteServices_AuthPassword]	Defines the password for HTTP authentication. The valid value is a string of up to 80 characters. The default is "password".
TLS Context [HTTPRemoteServices_TLSContext]	Assigns a TLS Context for the connection with the HTTP service. By default, no value is defined (**None**). For configuring TLS Contexts, see "Configuring TLS Certificate Contexts" on page 107. **Note:** The parameter is applicable only if the connection is HTTPS.
Verify Certificate [HTTPRemoteServices_VerifyCertificate]	Enables certificate verification when the connection with the host is based on HTTPS. ▪ [0] Disable (default) = No certificate verification is done. ▪ [1] Enable = The device verifies the authentication of the certificate received from the HTTPS peer. The device authenticates the certificate against the trusted root certificate store associated with the associated TLS Context (see 'TLS Context' parameter above) and if ok, allows communication with the HTTPS peer. If authentication fails, the device denies communication (i.e., handshake fails). The device can also authenticate the certificate by querying with an Online Certificate Status Protocol (OCSP) server whether the certificate has been revoked. This is also configured for the associated TLS Context. ▪ **Note:** The parameter is applicable only if the connection is HTTPS.
Response Timeout [HTTPRemoteServices_TimeOut]	Defines the TCP response timeout (in seconds) from the remote host. If one of the remote hosts does not respond to a request within the specified timeout, the device closes the corresponding socket and attempts to connect to the next remote host. The valid value is 1 to 65535. The default is 5.
Keep-Alive Timeout [HTTPRemoteServices_KeepAliveTimeOut]	Defines the duration/timeout (in seconds) in which HTTP-REST keep-alive messages are sent by the device if no other messages are sent. Keep-alive messages may be required for HTTP services that expire upon inactive sessions. The valid value is 0 to 65535. The default is 0 (i.e., no keep-alive messages are sent). **Note:** The parameter is applicable only if the 'Persistent Connection' parameter (in the table) is configured to **Enable**.
Topology Status [HTTPRemoteServices_ServiceStatus]	Indicates the status of the host. ▪ "Connected": at least one of the hosts is connected. ▪ "Disconnected": all hosts are disconnected. ▪ "Not In Service": Configuration of the service is invalid.

15.6.2 Configuring Remote HTTP Hosts

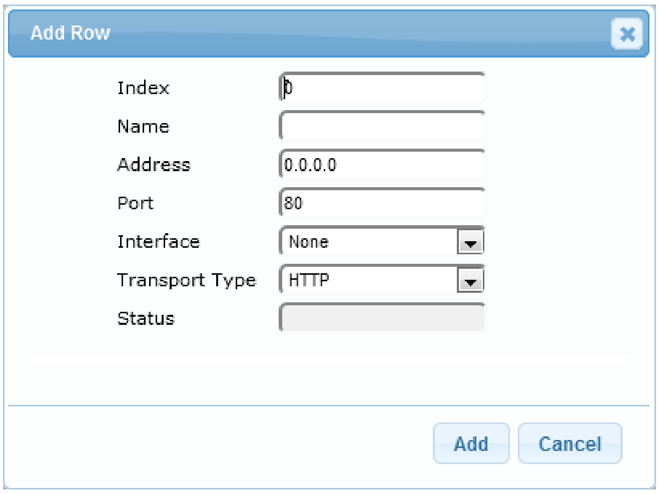
The HTTP Remote Hosts table lets you configure up to 10 remote HTTP hosts per HTTP Remote Service. The HTTP Remote Hosts table is a "child" of the HTTP Remote Services table (configured in "Configuring HTTP Services" on page 267).

The following procedure describes how to configure HTTP Remote hosts through the Web interface. You can also configure it through ini file (HTTPRemoteServices).

➢ **To configure an HTTP-based service**

1. Open the HTTP Remote Services table (**Configuration** tab > **VoIP** menu > **Services** > **HTTP Services** > **HTTP Remote Services**).
2. In the table, select the required HTTP Remote Service index row, and then click the **HTTP Remote Hosts** button, located below the table; the HTTP Remote Hosts page appears.
3. Click **Add**; the following dialog box appears:

Figure 15-35: HTTP Remote Hosts Table - Add Row Dialog Box



Index: 0
Address: 0.0.0.0
Port: 80
Interface: None
Transport Type: HTTP

4. Configure an HTTP remote host according to the parameters described in the table below.
5. Click **Add**, and then save ("burn") your settings to flash memory.

Table 15-20: HTTP Remote Hosts Table Parameter Descriptions

Parameter	Description
Index [HTTPRemoteHosts_RemoteHostindex]	Defines an index number for the new table row. **Notes:** ▪ Each row must be configured with a unique index. ▪ The parameter is mandatory.
Name [HTTPRemoteHosts_Name]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. By default, no value is defined. **Notes:** ▪ Each row must be configured with a unique name. ▪ The parameter is mandatory.
Address [HTTPRemoteHosts_Address]	Defines the address (IP address or FQDN) of the host. The valid value is a string of up to 80 characters. **Notes:**

Parameter	Description
	- An IPv6 address can only be configured if the interface is a CONTROL type. - If the address is an FQDN and the DNS resolution results in multiple IP addresses, the device device attempts to establish multiple connections (sessions) for each IP address. Only the first 10 resolved IP addresses are used regardless of the number of hosts. - FQDN resolution is also performed (immediately) when connection is subsequently "closed" (by timeout or by the remote host) and connections are updated accordingly. In addition, the device periodically (every 15 minutes) performs DNS name resolution to ensure that the list of resolved IP addresses has not changed. If a change is detected, the device updates its' list of IP addresses and re-establishes connections accordingly. - In addition to multiple HTTP sessions, the device establishes multiple (TCP) connections per session, thereby enhancing data exchange capabilities with the host.
Port [HTTPRemoteHosts_Port]	Defines the port of the host. The valid value is 0 to 65535. The default is 80.
Interface [HTTPRemoteHosts_Interface]	Assigns one of the device's IP network interfaces through which communication with the remote host is done. By default, no value is defined and the OAMP interface is used.
Transport Type [HTTPRemoteHosts_HTTPTransportType]	Defines the protocol used for communicating with the host: - [0] HTTP (default) - [1] HTTPS
Status	Read-only field displaying the status of the connection. - "Connected": The hosts is connected. - "Disconnected": The host is disconnected. - "Not In Service": Configuration of the host is invalid.

15.6.3 Centralized Third-Party Routing Server or ARM

You can employ a remote, third-party Routing server or AudioCodes Routing Manager (ARM) routing server to handle call routing decisions in deployments consisting of multiple AudioCodes devices. The routing server can be used to handle SBC, Tel-to-IP, and IP-to-Tel calls. Employing a routing server replaces the need for the device's routing tables (IP-to-IP Routing table for SBC calls, and Tel-to-IP Routing table and IP-to-Tel Routing table for Tel-to-IP and IP-to-Tel calls respectively) to determine call destination.

For SBC calls: When the device receives an incoming call (SIP INVITE, NOTIFY or MESSAGE), it searches the IP-to-IP Routing table for a matching routing rule that is also configured to use a routing server. If found, the device requests the routing server for an appropriate destination. For Gateway calls: When the device receives an incoming call (SIP INVITE, NOTIFY or MESSAGE), it disregards the routing tables and instead immediately requests the routing server for an appropriate destination. The request is sent to the routing server using an HTTP Get Route message. The request contains information about the call (SIP message and for IP-to-Tel calls, the source IP Group based on the associated Proxy Set).

The routing server uses its own algorithms and logic in determining the best route path. The routing server manages the call route between devices in "hops", which may be spread over different geographical locations. The destination to each hop (device) can be by IP address (with port), IP Group, or Trunk Group. If the destination is an IP address, even though the destination type (in the IP-to-IP Routing table) is an IP Group, the device only uses the IP Group for profiling (i.e., associated IP Profile etc.). If multiple devices exist in the call routing path, the routing server sends the IP address only to the last device ("node") in the path.

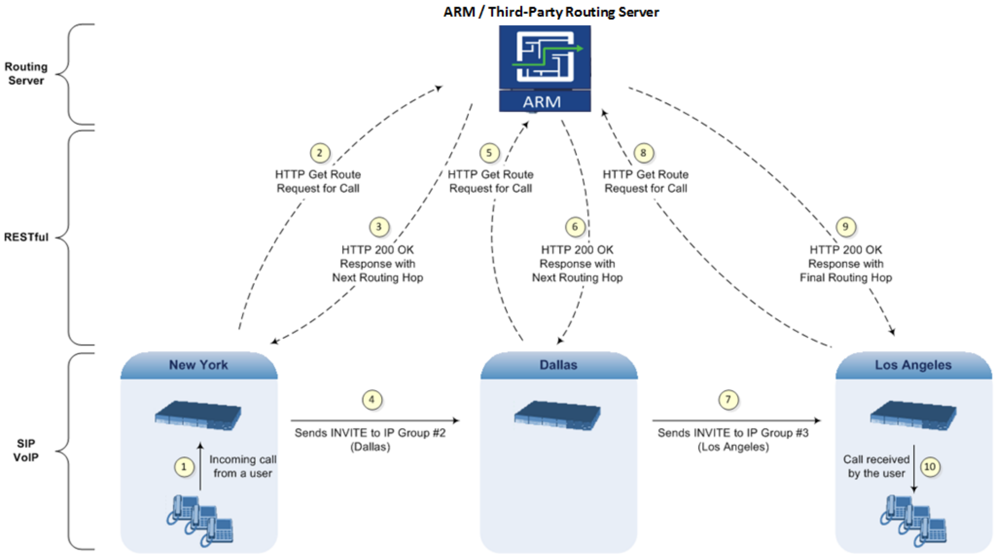
Once the device receives the resultant destination hop from the routing server, it sends the call to that destination. The routing server can provide the device with an appropriate route or reject the call. However, if for the initial request (first sent Get Route request for the call) the routing server cannot find an appropriate route for the call or it does not respond, for example, due to connectivity loss (i.e., the routing server sends an HTTP 404 "Not Found" message), the device routes the call using its routing tables. If the Get Route request is not the first one sent for the call (e.g., in call forwarding or alternative routing) and the routing server responds with an HTTP 404 "Not Found" message, the device rejects the call.

This HTTP request-response transaction for the routing path occurs between routing server and each device in the route path (hops) as the call traverses the devices to its final destination. Each device in the call path connects to the routing server, which responds with the next hop in the route path. Each device considers the call as an incoming call from an IP Group or Trunk Group. The session ID (SID) is generated by the first device in the path and then passed unchanged down the route path, enabling the routing server to uniquely identify requests belonging to the same call session.

Communication between the device and the routing server is through the device's embedded Representational State Transfer (RESTful) API. The RESTful API is used to manage the routing-related information exchanged between the routing server (RESTful server) and the device (RESTful client). When you have configured the device with connection settings of the Routing sever and the device starts-up, it connects to the routing server and activates the RESTful API, which triggers the routing-related API commands.

The following figure provides an example of information exchange between devices and a routing server for routing calls:

Figure 15-36: Example of Call Routing Information Exchange between Devices and Routing Server


ARM / Third-Party Routing Server
ARM
Routing Server
RESTful
SIP VoIP
2
HTTP Get Route Request for Call
3
HTTP 200 OK Response with Next Routing Hop
5
HTTP Get Route Request for Call
6
HTTP 200 OK Response with Next Routing Hop
8
HTTP Get Route Request for Call
9
HTTP 200 OK Response with Final Routing Hop
New York
1
Incoming call from a user
4
Sends INVITE to IP Group #2 (Dallas)
Dallas
7
Sends INVITE to IP Group #3 (Los Angeles)
Los Angeles
Call received by the user
10

The routing server can also manipulate call data such as calling name, if required. It can also create new IP Groups and associated configuration entities, if necessary for routing. Multiple routing servers can also be employed, whereby each device in the chain path can use a specific routing server. Alternatively, a single routing server can be employed and used for all devices ("stateful" routing server).

The device automatically updates (sends) the routing server with its' configuration topology regarding SIP routing-related entities (Trunk Groups, SRDs, SIP Interfaces, and IP Groups) that have been configured for use by the routing server. For example, if you add a new IP Group and enable it for use by the routing server, the device sends this information to the routing server. Routing of calls associated with routing-related entities that are disabled for use by the routing server (default) are handled only by the device (not the routing server).

In addition to regular routing, the routing server functionality also supports the following:

- **Alternative Routing:** If a call fails to be established, the device "closest" to the failure and configured to send "additional" routing requests (through REST API - "additionalRoute" attribute in HTTP Get Route request) to the routing server, sends a new routing request to the routing server. The routing server may respond with a new route destination, thereby implementing alternative routing. Alternatively, it may enable the device to return a failure response to the previous device in the route path chain and respond with an alternative route to this device. Therefore, alternative routing can be implemented at any point in the route path. If the routing server sends an HTTP 404 "Not Found" message for an alternative route request, the device rejects the call. If the routing server is configured to handle alternative routing, the device does not make any alternative routing decisions based on its alternative routing tables.
- **Call Status:** The device can report call status to the routing server to indicate whether a call has successfully been established and/or failed (disconnected). The device can also report when an IP Group (Proxy Set) is unavailable, detected by the keep-alive mechanism, or when the CAC thresholds permitted per IP Group have been crossed. For Trunk Groups, the device reports when the trunk's physical state indicates that the trunk is unavailable.
- **Credentials for Authentication:** The Routing Server can provide user (e.g., IP Phone caller) credentials (username-password) in the Get Route response, which can be used by the device to authenticate outbound SIP requests if challenged by the outbound peer, for example, Microsoft Skype for Business (per RFC 2617 and RFC 3261). If multiple devices exist in the call routing path, the routing server sends the credentials only to the last device ("node") in the path.

➢ To configure routing based on routing server:

1. For each configuration entity (e.g., IP Group) that you want routing done by the routing server, configure the entity's 'Used By Routing Server' parameter to **Used**.
2. Configure an additional Security Administrator user account in the Local Users table (see "Configuring Web User Accounts" on page 70), which is used by the routing server (REST client) to log in to the device's management interface.
3. Configure the address and connection settings of the routing server, referred to as a Remote Web Service and HTTP remote host (see HTTP-based Remote Services on page 267). You must configure the 'Type' parameter of the Remote Web Service to **Routing**.
4. SBC Calls: In the IP-to-IP Routing table, configure the 'Destination Type' parameter of the routing rule to **Routing Server** (see Configuring SBC IP-to-IP Routing Rules on page 578).
5. Gateway Calls: Enable routing based on routing server, by configuring the GWRoutingServer parameter to 1.

15.7 HTTP-based Proxy Services

You can configure the device for the following HTTP-based proxy services:

- **HTTP Reverse Proxy for Managing Equipment behind NAT:**

  You can configure the device to function as a reverse HTTP proxy server. This functionality is required to enable administrators to manage communication equipment (such as IP Phones) over HTTP when the equipment is located behind NAT (e.g., in the LAN) and the administrator is located in a public domain (e.g., in the WAN). Thus, this functionality resolves NAT issues, enabling the administrator to access the IP Phone's management interface (e.g., embedded Web server).

  To support the functionality, the following configuration is required:

  1. Enable the HTTP Proxy application (see 'Enabling the HTTP Proxy Application' on page 276).
  2. Define a local, listening HTTP interface for the leg interfacing with the administrator (see 'Configuring HTTP Interfaces' on page 276).

> **Note:** It is recommended **not** to use port 80 as this is the default port used by IP Phones for their Web-based management interface.

  3. Define each HTTP-based managed equipment:
     a. Define the URL prefix for accessing the equipment's management interface (see 'Configuring HTTP Proxy Services' on page 277). To access the equipment's management interface, the administrator needs to enter the following URL in a Web browser:

        http://<device's WAN IP address:port>/url prefix/

     b. Define the IP address of the managed equipment (see 'Configuring HTTP Proxy Hosts' on page 279).

> **Note:** For this feature, no special configuration is required on the managed equipment.

- **HTTP-based EMS Services for AudioCodes Equipment behind NAT:**

  You can configure the device to act as an HTTP Proxy that enables AudioCodes EMS to manage AudioCodes equipment (such as IP Phones) over HTTP when the equipment is located behind NAT (e.g., in the LAN) and EMS is located in a public domain (e.g., in the WAN). Thus, the feature resolves NAT traversal issues. The IP Phones register with the device in order to allow communication between the IP Phones and the EMS.

  To support the functionality, the following configuration is required:

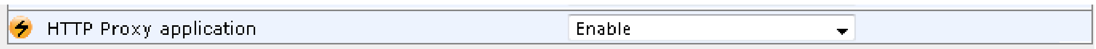
  1. Enable the HTTP Proxy application (see 'Enabling the HTTP Proxy Application' on page 276).
  2. Configure two local, listening HTTP interfaces - one for the EMS and one for the IP Phones (see 'Configuring HTTP Interfaces' on page 276).
  3. Configure the address of the EMS server (see 'Configuring an HTTP-based EMS Service' on page 281).

15.7.1 Enabling the HTTP Proxy Application

Before you can configure HTTP-based proxy services, you must enable the HTTP Proxy application, as described in the following procedure. Once enabled, the Web interface displays menus in the Navigation pane that are relevant to the HTTP Proxy application.

- **To enable the HTTP Proxy application:**

1. Open the Applications Enabling page (**Configuration** tab > **VoIP** menu > **Applications Enabling** > **Applications Enabling**).



HTTP Proxy application: Enable

2. From the 'HTTP Proxy Application' drop-down list, select **Enable**.
3. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

15.7.2 Configuring HTTP Interfaces

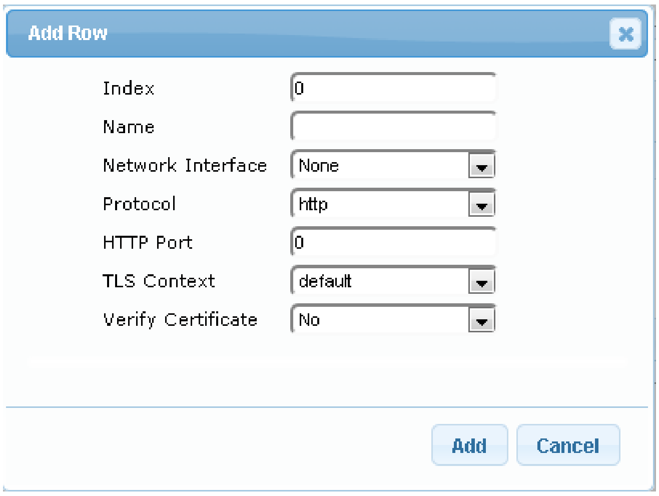
The HTTP Interfaces table lets you configure up to 10 HTTP Interfaces. An HTTP Interface represents a local, listening interface for receiving HTTP/S requests from HTTP-based (Web) clients such as managed equipment (e.g., IP Phones) and/or the EMS management tool for HTTP/S-based services.

The following procedure describes how to configure HTTP Interfaces through the Web interface. You can also configure it through ini file (HTTPInterface) or CLI (configure system > http-proxy > http-interface).

- **To configure an HTTP Interface:**

1. Open the HTTP Interfaces table (**Configuration** tab > **VoIP** menu > **Services** > **HTTP Proxy** > **HTTP Interfaces**).
2. Click **Add**; the following dialog box appears:

Figure 15-37: HTTP Interfaces Table - Add Row Dialog Box



Add Row

Index: 0
Name:
Network Interface: None
Protocol: http
HTTP Port: 0
TLS Context: default
Verify Certificate: No

3. Configure an HTTP Interface according to the parameters described in the table below.
4. Click **Add**, and then save ("burn") your settings to flash memory.

Table 15-21: HTTP Interfaces Table Parameter Descriptions

Parameter	Description
Index	Defines an index number for the new table row.

Parameter	Description
[HTTPInterface_Index]	**Notes:** - Each row must be configured with a unique index. - The parameter is mandatory.
Name `interface-name` [HTTPInterface_InterfaceName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. By default, no value is defined. **Notes:** - Each row must be configured with a unique name. - The parameter is mandatory.
Network Interface `network-interface` [HTTPInterface_NetworkInterface]	Assigns a local, network interface to the HTTP interface. By default, no value is defined (**None**). For configuring network interfaces, see Configuring IP Network Interfaces on page 133. **Note:** The parameter is mandatory.
Protocol `protocol` [HTTPInterface_Protocol]	Defines the protocol type. - [0] HTTP (default) - [1] HTTPS
HTTP Port `http-port` [HTTPInterface_Port]	Defines the local, listening HTTP port. The valid value is 0 to 65534. The default is 0. **Note:** The parameter is mandatory.
TLS Context `tls-context` [HTTPInterface_TLSContext]	Assigns a TLS Context for the connection with the HTTP Proxy service. By default, the default TLS Context (Index 0) is assigned. For configuring TLS Contexts, see Configuring TLS Certificate Contexts on page 107. **Note:** The parameter is applicable only if the connection protocol is HTTPS (defined using the 'Protocol' parameter, above).
Verify Certificate `verify-cert` [HTTPInterface_VerifyCert]	Enables TLS certificate verification when the connection with the proxy service is based on HTTPS. - [0] No = (Default) No certificate verification is done. - [1] Yes = The device verifies the authentication of the certificate received from the HTTPS peer. The device authenticates the certificate against the trusted root certificate store associated with the associated TLS Context (see 'TLS Context' parameter above) and if ok, allows communication with the HTTPS peer. If authentication fails, the device denies communication (i.e., handshake fails). The device can also authenticate the certificate by querying with an Online Certificate Status Protocol (OCSP) server whether the certificate has been revoked. This is also configured for the associated TLS Context. - **Note:** The parameter is applicable only if the connection protocol is HTTPS (defined using the 'Protocol' parameter, above).

15.7.3 Configuring HTTP Proxy Services

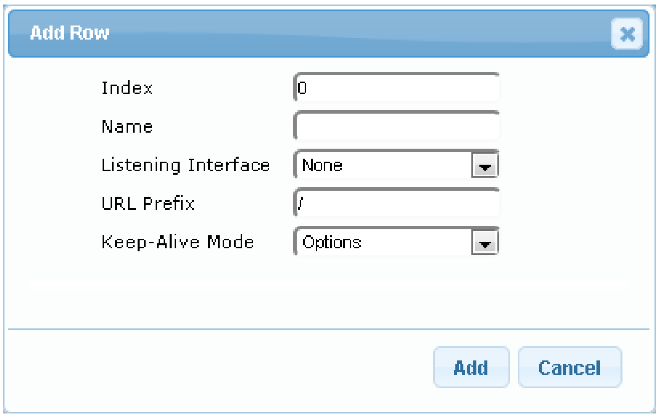
The HTTP Proxy Services table lets you configure up to 10 HTTP Proxy Services.

The following procedure describes how to configure HTTP Proxy Services through the Web interface. You can also configure it through ini file (HTTPProxyService) or CLI (configure system > http-proxy > http-proxy-serv).

➢ **To configure an HTTP Proxy Service:**

1. Open the HTTP Proxy Services table (**Configuration** tab > **VoIP** menu > **Services** > **HTTP Proxy** > **HTTP Proxy Services**).
2. Click **Add**; the following dialog box appears:

Figure 15-38: HTTP Proxy Services Table - Add Row Dialog Box



Add Row

Index: 0

Name:

Listening Interface: None

URL Prefix: /

Keep-Alive Mode: Options

3. Configure an HTTP Proxy service according to the parameters described in the table below.
4. Click **Add**, and then save ("burn") your settings to flash memory.

Table 15-22: HTTP Proxy Services Table Parameter Descriptions

Parameter	Description
Index [HTTPProxyService_Index]	Defines an index number for the new table row. **Notes:** ▪ Each row must be configured with a unique index. ▪ The parameter is mandatory.
Name `service-name` [HTTPProxyService_ServiceName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. By default, no value is defined. **Notes:** ▪ Each row must be configured with a unique name. ▪ The parameter is mandatory.
Listening Interface `listening-int` [HTTPProxyService_ListeningInterface]	Assigns an HTTP Interface to the HTTP Proxy service. To configure HTTP Interfaces, see 'Configuring HTTP Interfaces' on page 276. **Note:** The parameter is mandatory.

Parameter	Description
URL Prefix `url-prefix` [HTTPProxyService_URLPrefix]	Defines the URL prefix that is used to access the managed equipment's embedded Web server. The URL prefix is matched against the target of the HTTP requests sent by the client (such as GET and POST). If a match is located in the table, the device removes the prefix from the request and then forwards the HTTP request to the managed equipment without the prefix. For example, for the URL of GET /home/index.html HTTP/1.1 (which is part of the URL http://10.20.30.40/home/index.html), a URL prefix of “/home” can be configured. To match all URLs, configure the parameter to "/" (default).
Keep-Alive Mode `keep-alive-mode` [HTTPProxyService_KeepAliveMode]	Enables a keep-alive mechanism with the managed equipment: ▪ [0] Disable ▪ [1] Options = (Default) Enables keep-alive by sending HTTP OPTIONS messages. If no response is received from the HTTP host, the device stops forwarding HTTP requests to the host and raises an SNMP alarm (acHTTPProxyServiceAlarm). If you configured the address of the host as an FQDN (see 'Configuring HTTP Proxy Hosts' on page 279) and the DNS resolution results in multiple IP addresses, when no response is received from the keep-alive, the device checks connectivity with the next resolved IP address and so on, until a response is received.

15.7.4 Configuring HTTP Proxy Hosts

The HTTP Proxy Hosts table lets you configure up to 50 HTTP Proxy hosts (up to 5 per HTTP Proxy Service). The table is a "child" of the HTTP Proxy Services table (see 'Configuring HTTP Proxy Services' on page 277). An HTTP Proxy Host represents the HTTP-based managed equipment (e.g., IP Phone).

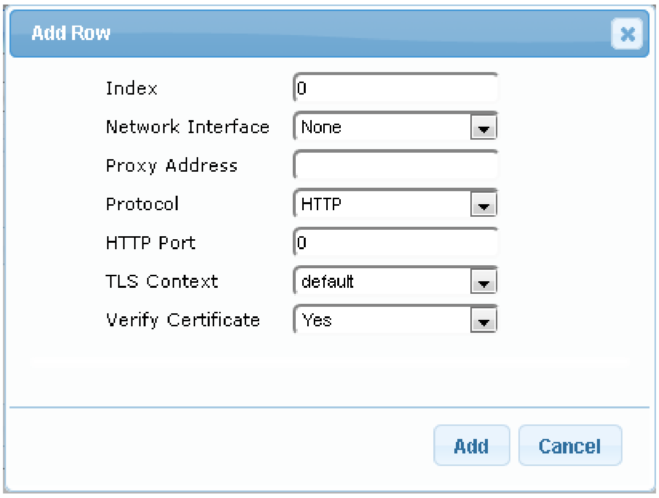
The following procedure describes how to configure HTTP Remote hosts through the Web interface. You can also configure it through ini file (HTTPProxyHost) or CLI (configure system > http-proxy > http-proxy-host).

➢ To configure an HTTP Proxy Host:

1. Open the HTTP Proxy Services table (**Configuration** tab > **VoIP** menu > **Services** > **HTTP Proxy** > **HTTP Proxy Services**).
2. In the table, select the required HTTP Proxy Service index row, and then click the **HTTP Proxy Hosts** link, located below the table; the HTTP Proxy Hosts table appears.

3. Click **Add**; the following dialog box appears:

Figure 15-39: HTTP Proxy Hosts Table - Add Row Dialog Box



Add Row

Index: 0
Network Interface: None
Proxy Address:
Protocol: HTTP
HTTP Port: 0
TLS Context: default
Verify Certificate: Yes

4. Configure an HTTP Proxy Host according to the parameters described in the table below.
5. Click **Add**, and then save ("burn") your settings to flash memory.

Table 15-23: HTTP Proxy Hosts Table Parameter Descriptions

Parameter	Description
Index	Defines an index number for the new table row. **Notes:** ▪ Each row must be configured with a unique index. ▪ The parameter is mandatory.
Network Interface `network-interface` [HTTPProxyHost_NetworkInterface]	Assigns a local, network interface to the HTTP Proxy Host. By default, no value is defined (**None**). For configuring network interfaces, see Configuring IP Network Interfaces on page 133. **Note:** The parameter is mandatory.
Proxy Address `proxy-address` [HTTPProxyHost_IpAddress]	Defines the address of the managed equipment (host). The valid value is an IP address in dotted-decimal notation or an FQDN (up to 100 characters). If the address is an FQDN, the device uses DNS to resolve it into an IP address. If the DNS resolution results in multiple IP addresses, the device uses the first available address (i.e., that responds to the keep-alive).
Protocol `protocol` [HTTPProxyHost_Protocol]	Defines the protocol type. ▪ [0] HTTP (default) ▪ [1] HTTPS
HTTP Port `http-port` [HTTPProxyHost_Port]	Defines the port of the managed equipment. The default is 0. **Note:** The parameter is mandatory.

Parameter	Description
TLS Context `tls-context` [HTTPProxyHost_TLSContext]	Assigns a TLS Context for the TLS connection with the HTTP Proxy host. By default, the default TLS Context (Index 0) is assigned. For configuring TLS Contexts, see Configuring TLS Certificate Contexts on page 107. **Note:** The parameter is applicable only if the connection protocol is HTTPS (defined using the 'Protocol' parameter, above).
Verify Certificate `verify-cert` [HTTPProxyHost_VerifyCert]	Enables TLS certificate verification when the connection with the host is based on HTTPS. ▪ [0] No = No certificate verification is done. ▪ [1] Yes = (Default) The device verifies the authentication of the certificate received from the HTTPS peer. The device authenticates the certificate against the trusted root certificate store associated with the associated TLS Context (see 'TLS Context' parameter above) and if ok, allows communication with the HTTPS peer. If authentication fails, the device denies communication (i.e., handshake fails). The device can also authenticate the certificate by querying with an Online Certificate Status Protocol (OCSP) server whether the certificate has been revoked. This is also configured for the associated TLS Context) **Note:** The parameter is applicable only if the connection protocol is HTTPS (defined using the 'Protocol' parameter, above).

15.7.5 Configuring an HTTP-based EMS Service

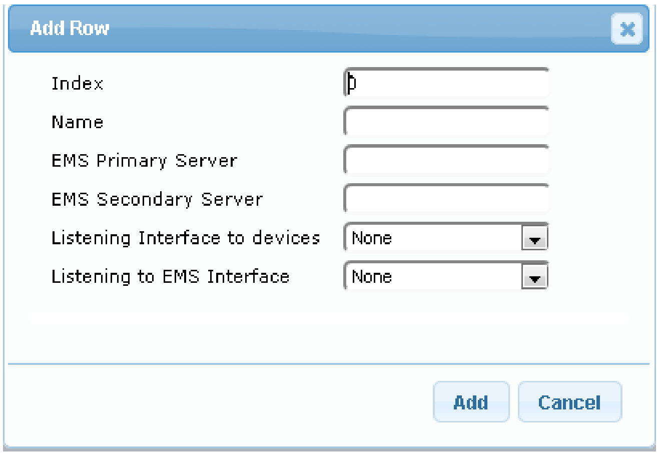
The EMS Services table lets you configure a single HTTP-based EMS service. For a description of the EMS service, see 'HTTP-based Proxy Services' on page 275.

The following procedure describes how to configure an EMS Service through the Web interface. You can also configure it through ini file (EMSService) or CLI (configure system > http-proxy > ems-serv).

➢ **To configure an EMS Service:**

1. Open the EMS Services table (**Configuration** tab > **VoIP** menu > **Services** > **HTTP Proxy** > **EMS Services**).
2. Click **Add**; the following dialog box appears:

Figure 15-40: EMS Services Table - Add Row Dialog Box



Index	0
Name	
EMS Primary Server	
EMS Secondary Server	
Listening Interface to devices	None
Listening to EMS Interface	None

3. Configure an EMS Service according to the parameters described in the table below.

4. Click **Add**, and then save ("burn") your settings to flash memory.

Table 15-24: EMS Services Table Parameter Descriptions

Parameter	Description
Index [EMSService_Index]	Defines an index number for the new table row. **Notes:** ▪ Each row must be configured with a unique index. ▪ The parameter is mandatory.
Name `service-name` [EMSService_ServiceName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. By default, no value is defined. **Notes:** ▪ Each row must be configured with a unique name. ▪ The parameter is mandatory.
EMS Primary Server `primary-server` [EMSService_PrimaryServer]	Defines the address of the primary EMS server. **Note:** The parameter is mandatory.
EMS Secondary Server `secondary-server` [EMSService_SecondaryServer]	Defines the address of the secondary EMS server.
Listening Interface to devices `dev-login-int` [EMSService_DeviceLoginInterface]	Assigns an HTTP Interface (local, listening HTTP interface:port) for communication with the client. To configure HTTP Interfaces, see 'Configuring HTTP Interfaces' on page 276. By default, no value is defined (**None**). **Note:** The parameter is mandatory.
Listening to EMS Interface `ems-int` [EMSService_EMSInterface]	Assigns an HTTP Interface (local, listening HTTP interface:port) for communication with the EMS. To configure HTTP Interfaces, see 'Configuring HTTP Interfaces' on page 276. By default, no value is defined (**None**). **Note:** The parameter is mandatory.

15.8 Configuring Call Setup Rules

The Call Setup Rules table lets you configure up to 64 Call Setup rules. Call Setup rules define various sequences that are run upon the receipt of an incoming call (dialog) at call setup, before the device routes the call to its destination. Call Setup rules can be configured for any call direction (SBC, Tel-to-IP, or IP-to-Tel). Call Setup rules provides you with full flexibility in implementing simple or complex script-like rules that can be used for Lightweight Directory Access Protocol (LDAP) based routing as well as other advanced routing logic requirements such as manipulation. These Call Setup rules are assigned to routing rules.

Below is a summary of functions for which you can employ Call Setup rules:

- LDAP query rules: LDAP is used by the device to query Microsoft’s Active Directory (AD) server for specific user details for routing, for example, office extension number, mobile number, private number, OCS (Lync) address, and display name. Call Setup rules provides full flexibility in AD-lookup configuration to suite just about any customer deployment requirement:
  - Routing based on query results.
  - Queries based on any AD attribute.
  - Queries based on any attribute value (alphanumeric), including the use of the asterisk (*) wildcard as well as the source number, destination number, redirect number, and SBC SIP messages. For example, the following Call Setup rule queries the attribute "proxyAddresses" for the record value "WOW:" followed by source number: "proxyAddresses=WOW:12345*"
  - Conditional LDAP queries, for example, where the query is based on two attributes (&(telephoneNumber=4064)(company=ABC).
  - Conditions for checking LDAP query results.
  - Manipulation of call parameters such as source number, destination number, and redirect number and SBC SIP messages, while using LDAP query results.
  - Multiple LDAP queries.
- Manipulation (similar to the Message Manipulations table) of call parameters (such as source number, destination number, and redirect number) and SBC SIP messages.
- Conditions for routing, for example, if the source number equals a specific value, then use the call routing rule.

You configure Call Setup rules with a Set ID, similar to the Message Manipulations table, where multiple rules can be associated with the same Set ID. This lets you perform multiple Call Setup rules on the same call setup dialog.

To use your Call Setup rule(s), you need to assign the Call Setup Rules Set ID to the relevant routing rule. This is done using the 'Call Setup Rules Set ID' field in the routing table:

- SBC IP-to-IP routing - see Configuring SBC IP-to-IP Routing Rules on page 578
- Tel-to-IP routing rules - see Configuring Tel-to-IP Routing Rules on page 467
- IP-to-Tel routing rules - see "Configuring IP-to-Trunk Group Routing Rules" on page 476

If an incoming call matches the characteristics of a routing rule, the device **first** runs the assigned Call Setup Rules Set ID before routing the call according to the rule. The device uses the routing rule to route the call, depending on the result of the Call Setup Rules Set ID:

- **Rule's condition is met:** The device performs the rule's action and then runs the next rule in the Set ID until the last rule or until a rule with an **Exit** Action Type. If the **Exit** rule is configured with a "True" Action Value, the device uses the current routing rule. If the **Exit** rule is configured with a "False" Action Value, the device moves to the next routing rule. If an **Exit** Action Type is not configured and the device has run all the rules in the Set ID, the default Action Value of the Set ID is "True" (i.e., use the current routing rule).
- **Rule's condition is not met:** The device runs the next rule in the Set ID. When the

device reaches the end of the Set ID and no **Exit** was performed, the Set ID ends with a "True" result.

You can also configure a Call Setup rule that determines whether the device must discontinue with the Call Setup Rules Set ID and route the call accordingly. This is done using the **Exit** optional value of the ‘Action Type’ parameter. When used, the ‘Action Value’ parameter can be configured to one of the following strings:

- “true”: Indicates that if the condition is met, the device routes the call according to the selected routing rule. Note that if the condition is not met, the device also uses the selected routing rule, unless the next Call Setup rule in the Set ID has an **Exit** option configured to “false” for an empty condition.
- “false”: Indicates that if the condition is met, the device attempts to route the call to the next matching routing rule (if configured). If the condition is not met, the device routes the call according to the selected routing rule.

As the default result of a Call Setup rule is always “true”, please adhere to the following guidelines when configuring the ‘Action Type’ field to **Exit**: If, for example, you want to exit the Call Setup Rule Set ID with "true" when LDAP query result is found and "false" when LDAP query result is not found:

- Incorrect -this rule will always exit with result = True:

  Condition: ldap.found exists **Action Type:** Exit **Action Value:** True
- Correct:
  - Single rule:

    Condition: ldap.found !exists **Action Type**: Exit **Action Value:** False
  - Set of rules:

    Condition: ldap.found exists **Action Type:** Exit **Action Value:** True

    Condition: <leave it blank> **Action Type:** Exit **Action Value:** False

Note: If the source and/or destination numbers are manipulated by the Call Setup rules, they revert to their original values if the device moves to the next routing rule.

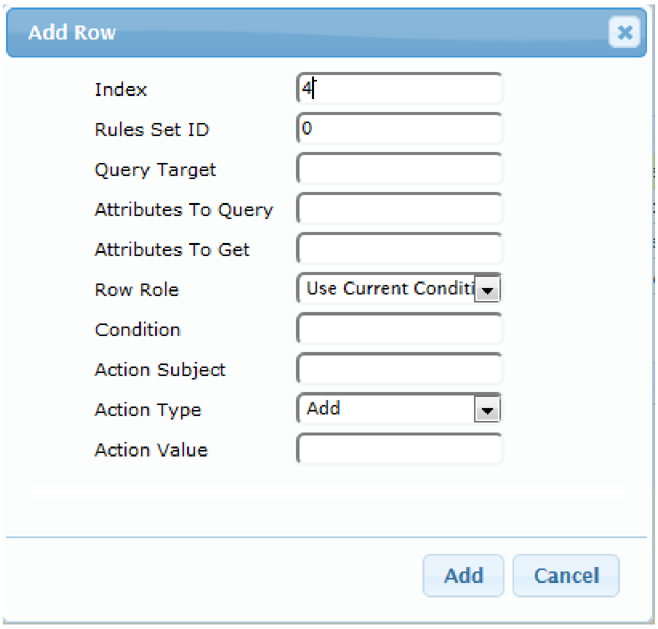
The following procedure describes how to configure Call Setup Rules through the Web interface. You can also configure it through ini file (CallSetupRules) or CLI (configure voip > services call-setup-rules).

➢ **To configure a Call Setup rule:**

1. Open the Call Setup Rules table (**Configuration** tab > **VoIP** menu > **Services** > **LDAP** > **Call Setup Rules**).

2. Click **Add**; the following dialog box appears:

Figure 15-41: Call Setup Rules Table - Add Row Dialog Box



Add Row

Index: 4

Rules Set ID: 0

Query Target:

Attributes To Query:

Attributes To Get:

Row Role: Use Current Conditi

Condition:

Action Subject:

Action Type: Add

Action Value:

3. Configure a Call Setup rule according to the parameters described in the table below.
4. Click **Add**, and then save ("burn") your settings to flash memory.

Table 15-25: Call Setup Rules Parameter Descriptions

Parameter	Description
Index [CallSetupRules_Index]	Defines an index number for the new table record. **Note:** Each rule must be configured with a unique index.
Rules Set ID `rules-set-id` [CallSetupRules_RulesSetID]	Defines a Set ID for the rule. You can define the same Set ID for multiple rules to create a group of rules. You can configure up to 32 Set IDs, where each Set ID can include up to 10 rules. The Set ID is used to assign the Call Setup rules to a routing rule in the routing table. The valid value is 0 to 31. The default is 0.
Query Target query-target [CallSetupRules_QueryTarget]	Specifies an LDAP server (LDAP Server Group) on which to perform an LDAP query. To configure LDAP Server Groups, see Configuring LDAP Server Groups on page 234.
Attributes To Query `attr-to-query` [CallSetupRules_AttributesToQuery]	Defines the query string that the device sends to the LDAP server. The valid value is a string of up to 100 characters. Combined strings and values can be configured like in the Message Manipulations table, using the '+' operator. Single quotes (') can be used for specifying a constant string (e.g., '12345'). For example: ▪ 'mobile=' + param.call.dst.user (searches for the AD attribute, "mobile" that has the value of the destination user part of the incoming call) ▪ 'telephoneNumber=' + param.call.redirect + '*' (searches for the AD attribute, "telephoneNumber" that has a redirect number)

Parameter	Description
Attributes To Get `attr-to-get` [CallSetupRules_AttributesToGet]	Defines the attributes of the queried LDAP record that the device must handle (e.g., retrieve value). The valid value is a string of up to 100 characters. Up to five attributes can be defined, each separated by a comma (e.g., msRTCSIP-PrivateLine,msRTCSIP-Line,mobile). **Note:** The device saves the retrieved attributes' values for future use in other rules, until the next LDAP query or until the call is connected. Thus, the device does not need to re-query the same attributes.
Row Role `row-role` [CallSetupRules_RowRole]	Determines which condition must be met in order for this rule to be performed. ▪ **[0]** Use Current Condition = The Condition configured for this rule must be matched in order to perform the configured action (default). ▪ **[1]** Use Previous Condition = The Condition configured for the rule located directly above this rule in the Call Setup table must be matched in order to perform the configured action. This option lets you configure multiple actions for the same Condition.
Condition `condition` [CallSetupRules_Condition]	Defines the condition that must exist for the device to perform the action. The valid value is a string of up to 200 characters (case-insensitive). Regular Expression (regex) can also be used, for example: ▪ ldap.attr.mobile exists (attribute "mobile" exists in AD) ▪ param.call.dst.user == ldap.attr.msRTCSIP-PrivateLine (called number is the same as the number in the attribute "msRTCSIP-PrivateLine") ▪ ldap.found !exists (LDAP record not found) ▪ ldap.err exists (LDAP error exists)
Action Subject `action-subject` [CallSetupRules_ActionSubject]	Defines the element (header, parameter, or body) upon which you want to perform the action. The valid value is a string of up to 100 characters (case-insensitive). Examples: ▪ header.from contains '1234' (SBC calls only) ▪ param.call.dst.user (called number) ▪ param.call.src.user (calling number) ▪ param.call.src.name (calling name) ▪ param.call.redirect (redirect number) ▪ param.call.src.host (source host) ▪ param.call.dst.host (destination host)

Parameter	Description
Action Type `action-type` [CallSetupRules_ActionType]	Defines the type of action to perform. ▪ **[0]** Add (default) = Adds new message header, parameter or body elements. ▪ **[1]** Remove = Removes message header, parameter, or body elements. ▪ **[2]** Modify = Sets element to the new value (all element types). ▪ **[3]** Add Prefix = Adds value at the beginning of the string (string element only). ▪ **[4]** Add Suffix = Adds value at the end of the string (string element only). ▪ **[5]** Remove Suffix = Removes value from the end of the string (string element only). ▪ **[6]** Remove Prefix = Removes value from the beginning of the string (string element only). ▪ **[20]** Run Rules Set = Performs a different Rule Set ID, specified in the 'Action Value' parameter (below). ▪ **[21]** Exit = Stops the Rule Set ID and returns a result ("True" or "False").
Action Value `action-value` [CallSetupRules_ActionValue]	Defines a value that you want to use in the action. The valid value is a string of up to 300 characters (case-insensitive). Examples: ▪ '+9723976'+ldap.attr.alternateNumber ▪ '9764000' ▪ ldap.attr.displayName ▪ true (if the 'Action Type' is set to **Exit**) ▪ false (if the 'Action Type' is set to **Exit**)

15.8.1 Call Setup Rule Examples

Below are configuration examples for using Call Setup Rules.

- **Example 1:** This example configures the device to replace (manipulate) the incoming call's source number with a number retrieved from the AD by an LDAP query. The device queries the AD server for the attribute record, "telephoneNumber" whose value is the same as the received source number (e.g., "telephoneNumber =4064"). If such an attribute is found, the device retrieves the number of the attribute record, "alternateNumber" and uses this number as the source number.
    - **Call Setup Rules table configuration:**
        - 'Rules Set ID': **1**
        - 'Attributes to Query': **'telephoneNumber=' + param.call.src.user**
        - 'Attributes to Get': **alternateNumber**
        - 'Row Role': **Use Current Condition**
        - 'Condition': **ldap.attr. alternateNumber exists**
        - 'Action Subject': **param.call.src.user**
        - 'Action Type': **Modify**
        - 'Action Value': **ldap.attr. alternateNumber**
    - **Routing table configuration:** A single routing rule is assigned the Call Setup Rule Set ID.
        - Index 1:
            - ✓ 'Call Setup Rules Set Id': **1**
- **Example 2:** This example configures the device to replace (manipulate) the incoming call's calling name (caller ID) with a name retrieved from the AD by an LDAP query. The device queries the AD server for the attribute record, "telephoneNumber" whose value is the same as the received source number (e.g., "telephoneNumber =5098"). If such an attribute is found, the device retrieves the name from the attribute record, "displayName" and uses this as the calling name in the incoming call.
    - **Call Setup Rules table configuration:**
        - 'Rules Set ID': **2**
        - 'Attributes to Query': **'telephoneNumber=' + param.call.src.user**
        - 'Attributes to Get': **displayName**
        - 'Row Role': **Use Current Condition**
        - 'Condition': **ldap.attr. displayName exists**
        - 'Action Subject': **param.call.src.name**
        - 'Action Type': **Modify**
        - 'Action Value': **ldap.attr. displayName**
    - **Routing table configuration:** A single routing rule is assigned the Call Setup Rule Set ID.
        - Index 1:
            - ✓ 'Call Setup Rules Set Id': **2**
- **Example 3:** This example configures the device to route the incoming call according to whether or not the source number of the incoming call also exists in the AD server. The device queries the AD server for the attribute record, "telephoneNumber" whose value is the same as the received source number (e.g., telephoneNumber=4064"). If such an attribute is found, the device sends the call to the Lync server; if the query fails, the device sends the call to the PBX.
    - **Call Setup Rules table configuration:**
        - 'Rules Set ID': **3**
        - 'Attributes to Query': **'telephoneNumber=' + param.call.src.user**
        - 'Attributes to Get': **telephoneNumber**

    - 'Row Role': **Use Current Condition**
    - 'Condition': **ldap.found !exists**
    - 'Action Subject': -
    - 'Action Type': **Exit**
    - 'Action Value': **false**

  If the attribute record is found (i.e., condition is not met), the rule ends with a default exit result of true and uses the first routing rule (Lync). If the attribute record does not exist (i.e., condition is met), the rule exits with a false result and uses the second routing rule (PBX).

- **Routing table configuration:** Two routing rules are assigned with the same matching characteristics. Only the main routing rule is assigned a Call Setup Rules Set ID.
    - Index 1:
        - ✓ 'Call Setup Rules Set Id': **3**
        - ✓ 'Destination IP Group ID': **3** (IP Group for Lync)
    - Index 2:
        - ✓ 'Destination IP Group ID': **4** (IP Group of PBX)

15.9 Enhanced 9-1-1 Support for Lync Server

The Enhanced 9-1-1 (E9-1-1) service is becoming the mandatory emergency service required in many countries around the world. The E9-1-1 service, based on its predecessor 911, enables emergency operators to pinpoint the location (granular location) of callers who dial the 9-1-1 emergency telephone number.

Today, most enterprises implement an IP-based infrastructure providing a VoIP network with fixed and nomadic users, allowing connectivity anywhere with any device. This, together with an often deployed multi-line telephone system (MLTS) poses a challenge for E9-1-1 due to the difficulty in accurately locating the E9-1-1 caller.

This section describes the E9-1-1 solution provided by Microsoft Lync Server (hereafter referred to as *Lync Server*) and AudioCodes' device's ELIN interworking capabilities, which provides the SIP Trunk or ISDN (or CAMA) connectivity to the E9-1-1 emergency service provider. This section also describes the configuration of the device for interoperating between the Lync Server environment and the E9-1-1 emergency provider.

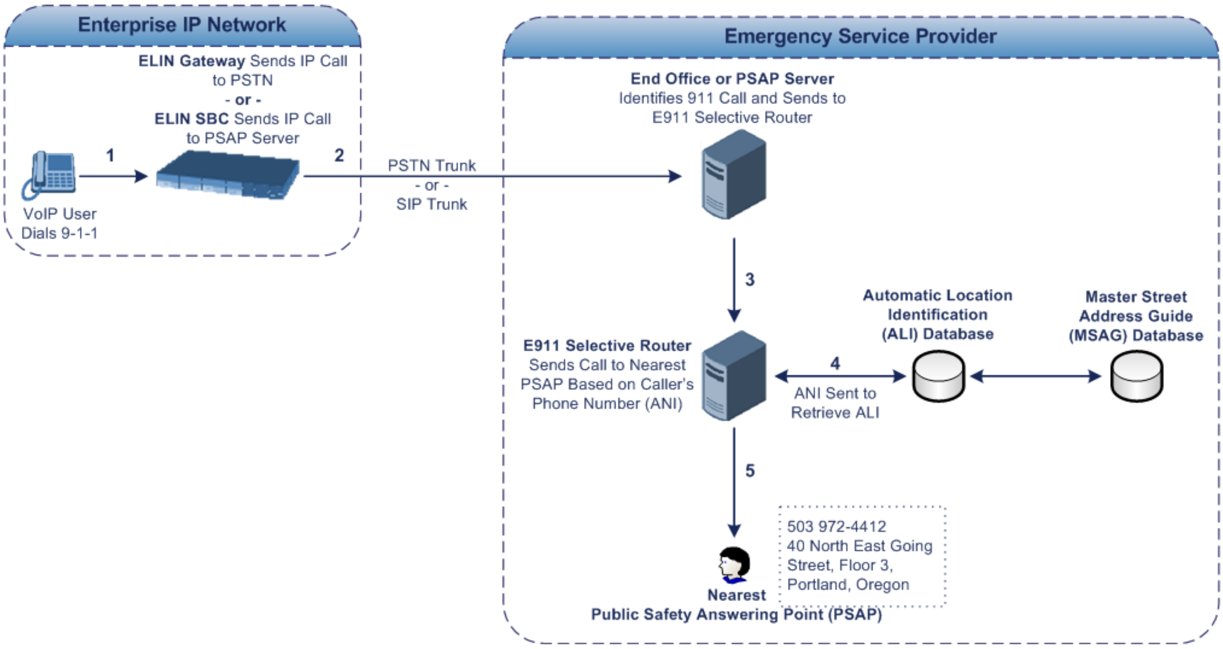
Notes:

- The ELIN feature for E9-1-1 is a license-dependent feature and is available only if it is included in the Software License Key installed on the device. For ordering the feature, please contact your AudioCodes sales representative. For installing a new Software License Key, see Software License Key on page 668.
- The E9-1-1 for Lync support is applicable to the SBC application and Gateway application for digital PSTN interfaces.

15.9.1 About E9-1-1 Services

E9-1-1 is a national emergency service for many countries, enabling E9-1-1 operators to automatically identify the geographical location and phone number of a 911 caller. In E9-1-1, the 911 caller is routed to the nearest E9-1-1 operator, termed *public safety answering point* (PSAP) based on the location of the caller. Automatic identification of the caller's location and phone number reduces the time spent on requesting this information from the 911 caller. Therefore, the E9-1-1 service enables the PSAP to quickly dispatch the relevant emergency services (for example, fire department or police) to the caller's location. Even if the call prematurely disconnects, the operator has sufficient information to call back the 911 caller.

The figure below illustrates the routing of an E9-1-1 call to the PSAP:


Enterprise IP Network
ELIN Gateway Sends IP Call to PSTN
- or -
ELIN SBC Sends IP Call to PSAP Server
1
2
VoIP User Dials 9-1-1
PSTN Trunk
- or -
SIP Trunk
Emergency Service Provider
End Office or PSAP Server Identifies 911 Call and Sends to E911 Selective Router
3
E911 Selective Router Sends Call to Nearest PSAP Based on Caller's Phone Number (ANI)
4
ANI Sent to Retrieve ALI
Automatic Location Identification (ALI) Database
Master Street Address Guide (MSAG) Database
5
503 972-4412
40 North East Going Street, Floor 3, Portland, Oregon
Nearest
Public Safety Answering Point (PSAP)

1. The VoIP user dials 9-1-1.
2. AudioCodes' ELIN device sends the call to the emergency service provider over the PSTN or SIP Trunk (PSAP server).
3. The emergency service provider identifies the call is an emergency call and sends it to an E9-1-1 Selective Router in the Emergency Services provider's network.
4. The E9-1-1 Selective Router determines the geographical location of the caller by requesting this information from an Automatic Location Identification (ALI) database based on the phone number or Automatic Number Identifier (ANI) of the 911 caller. Exact location information is also supplied by the Master Street Address Guide (MSAG) database, which is a companion database to the ALI database. Phone companies and public safety agencies collaborate beforehand to create master maps that match phone numbers, addresses and cross streets to their corresponding PSAP. This MSAG is the official record of valid streets (with exact spelling), street number ranges, and other address elements with which the service providers are required to update their ALI databases.
5. The E9-1-1 Selective Router sends the call to the appropriate PSAP based on the retrieved location information from the ALI.
6. The PSAP operator dispatches the relevant emergency services to the E9-1-1 caller.

15.9.2 Microsoft Lync Server and E9-1-1

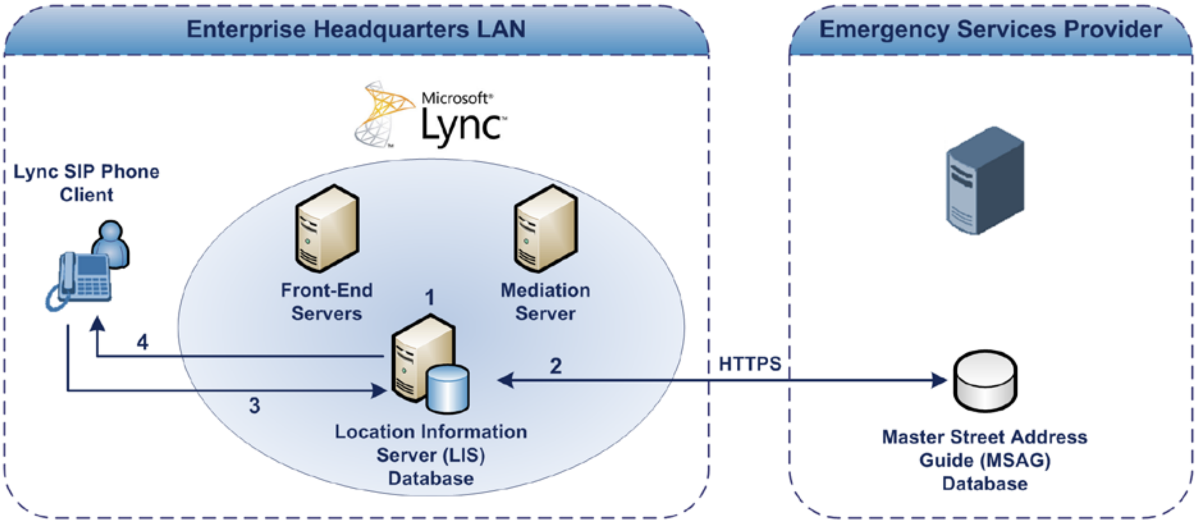
Microsoft Lync Server enables Enterprise voice users to access its unified communications platform from virtually anywhere and through many different devices. This, together with a deployed MLTS, poses a challenge for E9-1-1 due to the difficulty in accurately locating the E9-1-1 caller. However, Lync Server offers an innovative solution to solving Enterprises E9-1-1 location problems.

15.9.2.1 Gathering Location Information of Lync Clients for 911 Calls

When a Microsoft® Lync™ client (hereafter referred to as *Lync client*) is enabled for E9-1-1, the location data that is stored on the client is sent during an emergency call. This stored location information is acquired automatically from the Microsoft Location Information Server (LIS). The LIS stores the location of each network element in the enterprise. Immediately after the Lync client registration process or when the operating system detects a network connection change, each Lync client submits a request to the LIS for a location. If the LIS is able to resolve a location address for the client request, it returns the address in a location response. Each client then caches this information. When the Lync client dials 9-1-1, this location information is then included as part of the emergency call and used by the emergency service provider to route the call to the correct PSAP.

The gathering of location information in the Lync Server network is illustrated in the figure below:

Figure 15-42: Microsoft Lync Server 2010 Client Acquiring Location Information


Enterprise Headquarters LAN
Emergency Services Provider
Microsoft Lync
Lync SIP Phone Client
Front-End Servers
Mediation Server
1
2
3
4
HTTPS
Location Information Server (LIS) Database
Master Street Address Guide (MSAG) Database

1. The Administrator provisions the LIS database with the location of each network element in the Enterprise. The location is a civic address, which can include contextual in-building and company information. In other words, it associates a specific network entity (for example, a WAP) with a physical location in the Enterprise (for example, Floor 2, Wing A, and the Enterprise's street address). For more information on populating the LIS database, see "Adding ELINs to the Location Information Server" on page 292.
2. The Administrator validates addresses with the emergency service provider's MSAG – a companion database to the ALI database. This ensures that the civic address is valid as an official address (e.g., correct address spelling).
3. The Lync client initiates a location request to the LIS under the following circumstances:
   - Immediately after startup and registering the user with Lync Server
   - Approximately every four hours after initial registration
   - Whenever a network connection change is detected (such as roaming to a new WAP)

   The Lync client includes in its location request the following known network connectivity information:

   - Always included:
     - ♦ IPv4 subnet

    - Media Access Control (MAC) address
  - Depends on network connectivity:
    - Wireless access point (WAP) Basic Service Set Identifier (BSSID)
    - Link Layer Discovery Protocol-Media Endpoint Discovery (LLDP-MED) chassis ID and port ID

   For a Lync client that moves inside the corporate network such as a soft phone on a laptop that connects wirelessly to the corporate network, Lync Server can determine which subnet the phone belongs to or which WAP / SSID is currently serving the soft-client.

4. The LIS queries the published locations for a location and if a match is found, returns the location information to the client. The matching order is as follows:
   - WAP BSSID
   - LLDP switch / port
   - LLDP switch
   - Subnet
   - MAC address

   This logic ensures that for any client that is connected by a wireless connection, a match is first attempted based on the hardware address of its connected access point. The logic is for the match to be based on the most detailed location. The subnet generally provides the least detail. If no match is found in the LIS for WAP BSSID, LLDP switch / port, LLDP switch, or subnet, the LIS proxies the MAC address to an integrated Simple Network Management Protocol (SNMP) scanning application. Using SNMP may benefit some organizations for the following reasons:
   - LLDP is not supported by Lync Server so this provides a mechanism for soft phones to acquire detailed location information.
   - Installed Layer-2 switches may not support LLDP.

If there is no match and the LIS cannot determine the location, the user may be prompted to manually enter the location. For example, the client may be located in an undefined subnet, at home, in a coffee shop or anywhere else outside the network. When a user manually provides a location, the location is mapped based on the MAC address of the default gateway of the client's network and stored on the client. When the client returns to any previously stored location, the client is automatically set to that location. A user can also manually select any location stored in the local users table and manage existing entries.

15.9.2.2 Adding ELINs to the Location Information Server

As mentioned in the previous section, the administrator needs to populate the Location Information Server (LIS) database with a network wire map, which maps the Enterprise's network elements to civic addresses. Once done, it can automatically locate clients within a network. You can add addresses individually to the LIS or in a batch using a comma-separated value (CSV) file containing the column formats listed in the table below.

Table 15-26: Columns in the LIS Database

Network Element	Columns
Wireless access point	<BSSID>,<Description>,<Location>,<**CompanyName**>,<HouseNumber>,<HouseNumberSuffix>,<PreDirectional>,…<StreetName>,<StreetSuffix>,<PostDirectional>,<City>,<State>,<PostalCode>,<Country>
Subnet	<Subnet>,<Description>,<Location>,<CompanyName>,<HouseNumber>,<HouseNumberSuffix>,<PreDirectional>,…<StreetName>,<StreetSuffix>,<PostDirectional>,<City>,<State>,<PostalCode>,<Country>
Port	<ChassisID>,<PortIDSubType>,<PortID>,<Description>,<Location>,<CompanyName>,<HouseNumber>,<HouseNumberSuffix>,…<PreDirectional>,<StreetName>,<StreetSuffix>,<PostDirectional>,<City>,<State>,<PostalCode>,<Country>
Switch	<ChassisID>,<Description>,<Location>,<CompanyName>,<HouseNumber>,<HouseNumberSuffix>,<PreDirectional>,…<StreetName>,<StreetSuffix>,<PostDirectional>,<City>,<State>,<PostalCode>,<Country>

For the ELIN number to be included in the SIP INVITE (XML-based PIDF-LO message) sent by the Mediation Server to the ELIN device, the administrator must add the ELIN number to the <CompanyName> column (shown in the table above in **bold** typeface). As the ELIN device supports up to five ELINs per PIDF-LO, the <CompanyName> column can be populated with up to this number of ELINs, each separated by a semicolon. The digits of each ELIN can be separated by hyphens (xxx-xxx-xxx) or they can be adjacent (xxxxxxxxx).

When the ELIN device receives the SIP INVITE, it extracts the ELINs from the NAM field in the PIDF-LO (e.g., <ca:NAM>1111-222-333; 1234567890 </ca:NAM>), which corresponds to the <CompanyName> column of the LIS.

If you do not populate the location database, and the Lync Server location policy, Location Required is set to **Yes** or **Disclaimer**, the user will be prompted to enter a location manually.

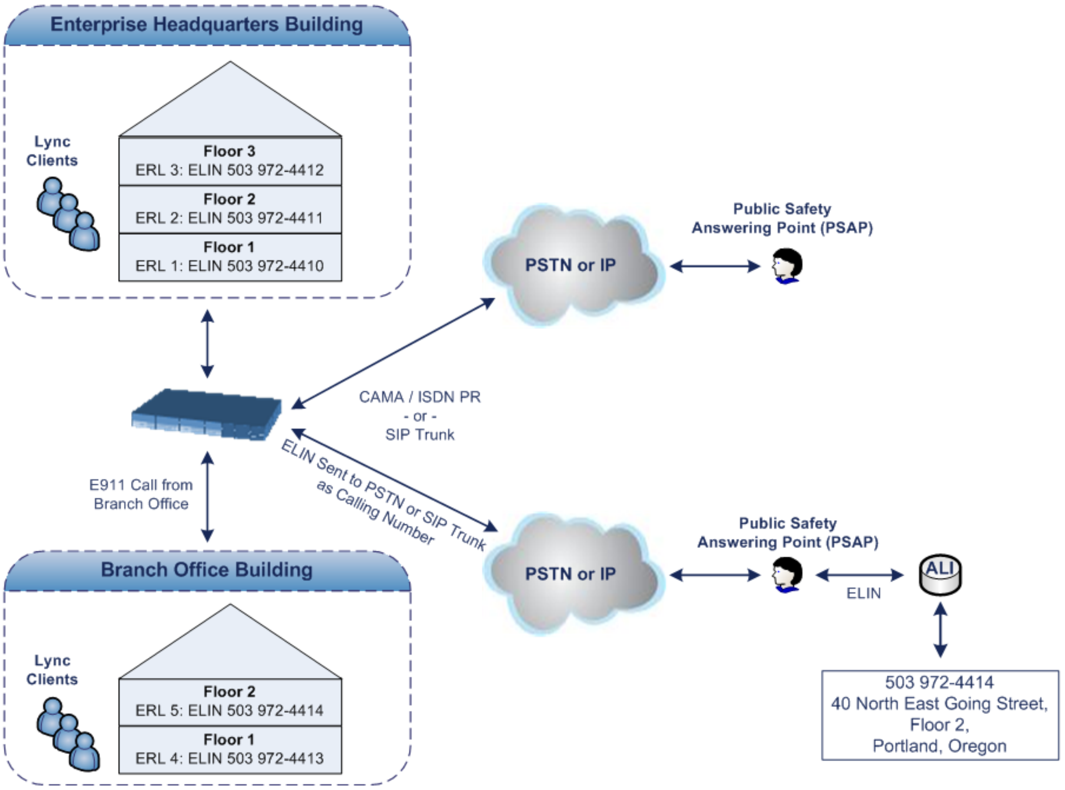
15.9.2.3 Passing Location Information to the PSTN Emergency Provider

When a Lync client, enabled for E9-1-1 emergency services, dials 9-1-1, the location data and callback information stored on the client is sent with the call through the Mediation Server to a SIP Trunk-based or PSTN-based emergency service provider. The emergency service provider then routes the call to the nearest and most appropriate PSAP based on the location information contained within the call.

Lync Server passes the location information of the Lync client in an IETF-standard format - Presence Information Data Format - Location Object (PIDF-LO)–in a SIP INVITE message. However, this content cannot be sent on the SIP Trunk or PSTN network since they do no support such a content. To overcome this, Enterprises deploying the device can divide their office space into Emergency Response Locations (ERLs) and assign a dedicated Emergency Location Identification Number (ELIN) to each ERL (or zone). When Lync Server sends a SIP INVITE message with the PIDF-LO to the device, it can parse the content and translate the calling number to an appropriate ELIN. The device then sends the call to the SIP Trunk or PSTN with the ELIN number as the calling number. The ELIN number is sent to the emergency service provider, which sends it on to the appropriate PSAP according to the ELIN address match in the ALI database lookup.

The ERL defines a specific location at a street address, for example, the floor number of the building at that address. The geographical size of an ERL is according to local or national regulations (for example, less than 7000 square feet per ERL). Typically, you would have an ERL for each floor of the building. The ELIN is used as the phone number for 911 callers within this ERL.

The figure below illustrates the use of ERLs and ELINs, with an E9-1-1 call from floor 2 at the branch office:



The table below shows an example of designating ERLs to physical areas (floors) in a building and associating each ERL with a unique ELIN.

Table 15-27: Designating ERLs and Assigning to ELINs

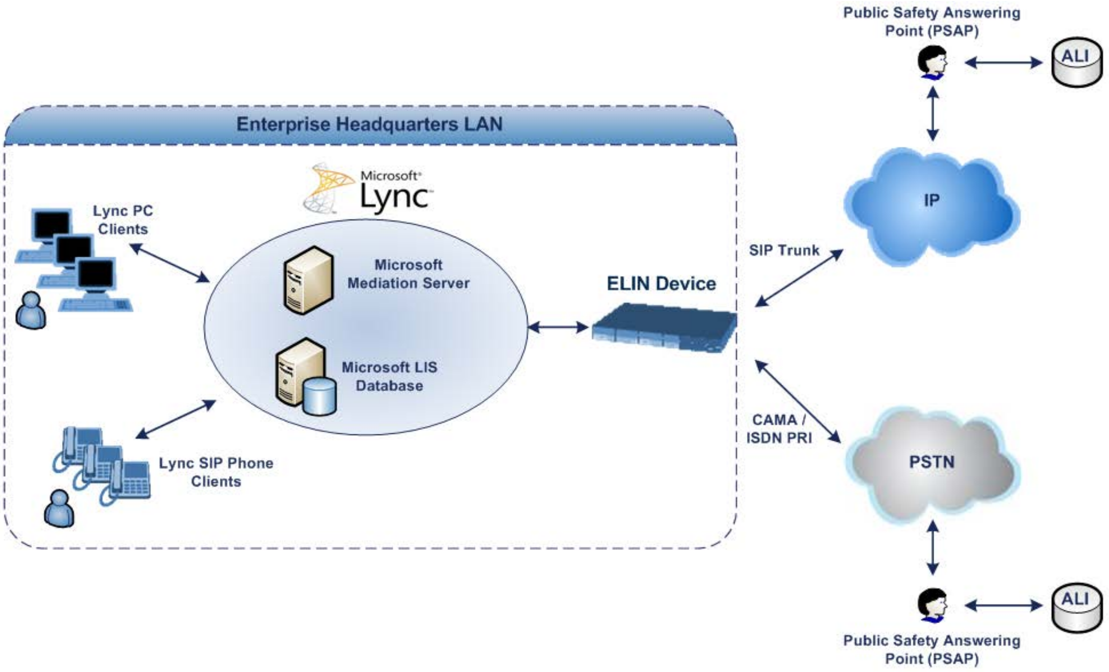
ERL Number	Physical Area	IP Address	ELIN
1	Floor 1	10.13.124.xxx	503 972-4410
2	Floor 2	10.15.xxx.xxx	503 972-4411
3	Floor 3	10.18.xxx.xxx	503 972-4412

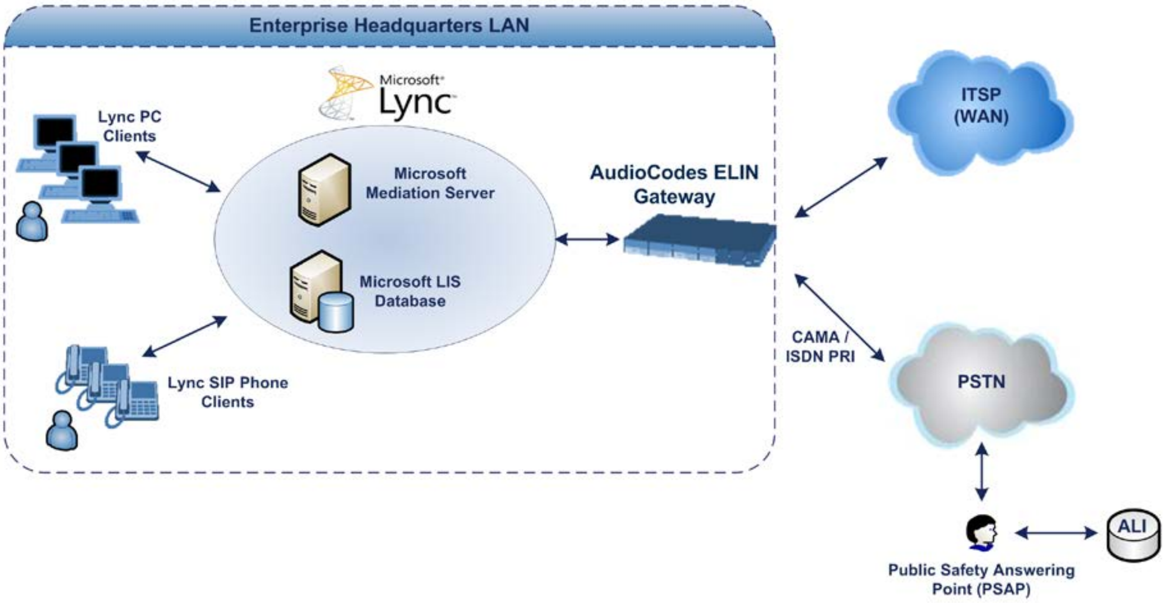
In the table above, a unique IP subnet is associated per ERL. This is useful if you implement different subnets between floors. Therefore, IP phones, for example, on a specific floor are in the same subnet and therefore, use the same ELIN when dialing 9-1-1.

15.9.3 AudioCodes ELIN Device for Lync Server E9-1-1 Calls to PSTN

Microsoft Mediation Server sends the location information of the E9-1-1 caller in the XML-based PIDF-LO body contained in the SIP INVITE message. However, this content cannot be sent on the SIP Trunk or PSTN network since they do not support such content. To solve this issue, Lync Server requires a device (*ELIN* SBC or Gateway) to send the E9-1-1 call to the SIP Trunk or PSTN. When Lync Server sends the PIDF-LO to the device, it parses the content and translates the calling number to an appropriate ELIN. This ensures that the call is routed to an appropriate PSAP, based on ELIN-address match lookup in the emergency service provider's ALI database.

The figure below illustrates an AudioCodes ELIN device deployed in the Lync Server environment for handling E9-1-1 calls between the Enterprise and the emergency service provider.


Public Safety Answering Point (PSAP)
ALI
Enterprise Headquarters LAN
Microsoft Lync
Lync PC Clients
Microsoft Mediation Server
Microsoft LIS Database
ELIN Device
SIP Trunk
IP
CAMA / ISDN PRI
PSTN
Lync SIP Phone Clients
ALI
Public Safety Answering Point (PSAP)



15.9.3.1 Detecting and Handling E9-1-1 Calls

The ELIN device identifies E9-1-1 calls and translates their incoming E9-1-1 calling numbers into ELIN numbers, sent toward the PSAP. The device handles the received E9-1-1 calls as follows:

1. The device identifies E9-1-1 calls if the incoming SIP INVITE message contains a PIDF-LO XML message body. This is indicated in the SIP *Content-Type* header, as shown below:

   ```
   Content-Type: application/pidf+xml
   ```

2. The device extracts the ELIN number(s) from the "NAM" field in the XML message. The

"NAM" field corresponds to the <CompanyName> column in the Location Information Server (LIS). The device supports up to five ELIN numbers per XML message. The ELINs are separated by a semicolon. The digits of the ELIN number can be separated by hyphens (xxx-xxx-xxx) or they can be adjacent (xxxxxxxxx), as shown below:

```
<ca:NAM>1111-222-333; 1234567890 </ca:NAM>
```

3. The device saves the *From* header value of the SIP INVITE message in its ELIN database table (**Call From** column). The ELIN table is used for PSAP callback, as discussed later in "PSAP Callback to Lync Clients for Dropped E9-1-1 Calls" on page 298. The ELIN table also stores the following information:
   - **ELIN:** ELIN number
   - **Time:** Time at which the original E9-1-1 call was terminated with the PSAP
   - **Count:** Number of E9-1-1 calls currently using the ELIN

   An example of the ELIN database table is shown below:

ELIN	Time	Count	Index	Call From
4257275678	22:11:52	0	2	4258359333
4257275999	22:11:57	0	3	4258359444
4257275615	22:12:03	0	0	4258359555
4257275616	22:11:45	0	1	4258359777

   The ELIN table stores this information for a user-defined period (see "Configuring the E9-1-1 Callback Timeout" on page 300), starting from when the E9-1-1 call, established with the PSAP, terminates. After this time expires, the table entry with its ELIN is disregarded and no longer used (for PSAP callback). Therefore, table entries of only the most recently terminated E9-1-1 callers are considered in the ELIN table. The maximum entries in the ELIN table is 100.

4. The device uses the ELIN number as the E9-1-1 calling number and sends it in the SIP INVITE or ISDN Setup message (as an ANI / Calling Party Number) to the SIP Trunk or PSTN.

An example of a SIP INVITE message received from an E9-1-1 caller is shown below. The SIP *Content-Type* header indicating the PIDF-LO, and the NAM field listing the ELINs are shown in **bold** typeface.

```
INVITE sip:911;phone-context=Redmond@192.168.1.12;user=phone
SIP/2.0
From:
"voip_911_user1"<sip:voip_911_user1@contoso.com>;epid=1D19090AED;t
ag=d04d65d924
To: <sip:911;phone-context=Redmond@192.168.1.12;user=phone>
CSeq: 8 INVITE
Call-ID: e6828be1-1cdd-4fb0-bdda-cda7faf46df4
VIA: SIP/2.0/TLS 192.168.0.244:57918;branch=z9hG4bK528b7ad7
CONTACT:
<sip:voip_911_user1@contoso.com;opaque=user:epid:R4bCDaUj51a06PUbk
raS0QAA;gruu>;text;audio;video;image
PRIORITY: emergency
CONTENT-TYPE: multipart/mixed; boundary= ------
=_NextPart_000_4A6D_01CAB3D6.7519F890
geolocation: <cid:voip_911_user1@contoso.com>;inserted-
by="sip:voip_911_user1@contoso .com"
Message-Body:
------=_NextPart_000_4A6D_01CAB3D6.7519F890
```

```
Content-Type: application/sdp ; charset=utf-8
v=0
o=- 0 0 IN IP4 Client
s=session
c=IN IP4 Client
t=0 0
m=audio 30684 RTP/AVP 114 111 112 115 116 4 3 8 0 106 97
c=IN IP4 172.29.105.23
a=rtcp:60423
a=label:Audio
a=rtpmap:3 GSM/8000/1
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=ptime:20

------=_NextPart_000_4A6D_01CAB3D6.7519F890
Content-Type: application/pidf+xml
Content-ID: <voip_911_user1@contoso.com>
<?xml version="1.0" encoding="utf-8"?>
<presence xmlns="urn:ietf:params:xml:ns:pidf"
xmlns:gp="urn:ietf:params:xml:ns:pidf:geopriv10"
xmlns:bp="urn:ietf:params:xml:ns:pidf:geopriv10:basicPolicy"
xmlns:ca="urn:ietf:params:xml:ns:pidf:geopriv10:civicAddr"
xmlns:ms="urn:schema:Rtc.LIS.msftE911PidfExtn.2008"
entity="sip:voip_911_user1@contoso.com"><tuple
id="0"><status><gp:geopriv><gp:location-
info><ca:civicAddress><ca:country>US</ca:country><ca:A1>WA</ca:A1>
<ca:A3>Redmond</ca:A3><ca:RD>163rd</ca:RD><ca:STS>Ave</ca:STS><ca:
POD>NE</ca:POD><ca:HNO>3910</ca:HNO><ca:LOC>40/4451</ca:LOC>
<ca:NAM>1111-222-333; 1234567890 </ca:NAM>
<ca:PC>98052</ca:PC></ca:civicAddress></gp:location-
info><gp:usage-rules><bp:retransmission-
allowed>true</bp:retransmission-allowed></gp:usage-
rules></gp:geopriv><ms:msftE911PidfExtn><ms:ConferenceUri>sip:+142
55550199@contoso.com;user=phone</ms:ConferenceUri><ms:ConferenceMo
de>twoway</ms:ConferenceMode><LocationPolicyTagID
xmlns="urn:schema:Rtc.Lis.LocationPolicyTagID.2008">user-
tagid</LocationPolicyTagID
></ms:msftE911PidfExtn></status><timestamp>1991-09-
22T13:37:31.03</timestamp></tuple></presence>
------=_NextPart_000_4A6D_01CAB3D6.7519F890--
```

15.9.3.2 Pre-empting Existing Calls for E9-1-1 Calls

If the ELIN device receives an E9-1-1 call from the IP network and there are unavailable channels (for example, all busy), the device immediately terminates one of the non-E9-1-1 calls (arbitrary) and accepts the E9-1-1 call on the freed channel:

- SBC: The preemption is done only on a call pertaining to the same source IP Group from which the E9-1-1 call is received, or the same destination IP Group (i.e., PSAP Server).
- Gateway: The preemption is done only on a channel pertaining to the same Trunk Group for which the E9-1-1 call was initially destined. For example, if an E9-1-1 call is destined for Trunk Group #2 and all the channels belonging to this group are busy, the

device terminates one of the calls in this group to free a channel for accepting the E9-1-1 call.

This feature is initiated only if the received SIP INVITE message contains a *Priority* header set to "emergency", as shown below:

```
PRIORITY: emergency
```

15.9.3.3 PSAP Callback to Lync Clients for Dropped E9-1-1 Calls

As the E9-1-1 service automatically provides all the contact information of the E9-1-1 caller to the PSAP, the PSAP operator can call back the E9-1-1 caller. This is especially useful in cases where the caller disconnects prematurely. However, as the Enterprise sends ELINs to the PSAP for E9-1-1 calls, a callback can only reach the original E9-1-1 caller using the device to translate the ELIN number back into the E9-1-1 caller's extension number.

In the ELIN table of the device, the temporarily stored *From* header value of the SIP INVITE message originally received from the E9-1-1 caller is used for PSAP callback. When the PSAP makes a callback to the E9-1-1 caller, the device translates the called number (i.e., ELIN) received from the PSAP to the corresponding E9-1-1 caller's extension number as matched in the ELIN table.

The handling of PSAP callbacks by the device is as follows:

1. When the device receives a call from the emergency service provider, it searches the ELIN table for an ELIN that corresponds to the received called party number in the incoming message.
2. If a match is found in the ELIN table, it routes the call to the Mediation Sever by sending a SIP INVITE, where the values of the *To* and *Request-URI* are taken from the value of the original *From* header that is stored in the ELIN table (in the **Call From** column).
3. The device updates the Time in the ELIN table. (The Count is not affected).

The PSAP callback can be done only within a user-defined period (see "Configuring the E9-1-1 Callback Timeout" on page 300), started from after the original E9-1-1 call established with the PSAP is terminated. After this time expires, the table entry with its ELIN is disregarded and no longer used (for PSAP callback). Therefore, table entries of only the most recently terminated E9-1-1 callers are considered in the ELIN table. If the PSAP callback is done after this timeout expires, the device is unable to route the call to the E9-1-1 caller and instead, either sends it as a regular call or most likely, rejects it if there are no matching routing rules. However, if another E9-1-1 caller has subsequently been processed with the same ELIN number, the PSAP callback is routed to this new E9-1-1 caller.

In scenarios where the same ELIN number is used by multiple E9-1-1 callers, upon receipt of a PSAP callback, the device sends the call to the most recent E9-1-1 caller. For example, if the ELIN number "4257275678" is being used by three E9-1-1 callers, as shown in the table below, then when a PSAP callback is received, the device sends it to the E9-1-1 caller with phone number "4258359555".

Table 15-28: Choosing Caller of ELIN

ELIN	Time	Call From
4257275678	11:00	4258359333
4257275678	11:01	4258359444
4257275678	11:03	**4258359555**

15.9.3.4 Selecting ELIN for Multiple Calls within Same ERL

The device supports the receipt of up to five ELIN numbers in the XML message of each incoming SIP INVITE message. As discussed in the preceding sections, the device sends the ELIN number as the E9-1-1 calling number to the emergency service provider. If the XML message contains more than one ELIN number, the device chooses the ELIN according to the following logic:

- If the first ELIN in the list is not being used by other active calls, it chooses this ELIN.
- If the first ELIN in the list is being used by another active call, the device skips to the next ELIN in the list, and so on until it finds an ELIN that is not being used and sends this ELIN.
- If all the ELINs in the list are in use by active calls, the device selects the ELIN number as follows:
  1. The ELIN with the lowest count (i.e., lowest number of active calls currently using this ELIN).
  2. If the count between ELINs is identical, the device selects the ELIN with the greatest amount of time passed since the original E9-1-1 call using this ELIN was terminated with the PSAP. For example, if E9-1-1 caller using ELIN 4257**2**75678 was terminated at **11:01** and E9-1-1 caller using ELIN 425727567**0** was terminated at **11:03**, then the device selects ELIN 425727567**8**.

  In this scenario, multiple E9-1-1 calls are sent with the same ELIN.

15.9.3.5 Location Based Emergency Routing

The device supports location-based emergency routing (E-911) in Lync Server. This ensures that E-911 calls from remote branches are routed to emergency providers that are relevant to the geographical area in which the remote branch callers are physically located.

To support this, the device enables routing and SIP header / number manipulation of such emergency calls based on the geographical location of the caller. The device manipulates the received destination number (i.e., 911) from the remote branch callers, into a destination number of an emergency provider that is relevant to the geographical area in which the remote branch office is located.

For an example on location-based emergency call routing, see "Configuring Location-Based Emergency Routing" on page 301.

Note: Location-based emergency routing is applicable only to the Gateway application.

15.9.4 Configuring AudioCodes ELIN Device

This section describes E9-1-1 configuration of the AudioCodes ELIN Gateway deployed in the Lync Server environment.

15.9.4.1 Enabling the E9-1-1 Feature

By default, the ELIN device feature for E9-1-1 emergency call handling in a Lync environment is disabled.

➢ **To enable the ELIN feature for the SBC application:**

- Configure the 'SBC PSAP Mode' parameter to **Enable** for the IP Group through which you want to communicate with the public-safety answering point (PSAP). For more information on IP Groups, see "Configuring IP Groups" on page 340.

➢ **To enable the ELIN feature for the Gateway application:**

1. Open the Advanced Parameters page (Configuration tab > VoIP menu > SIP Definitions > Advanced Parameters).

2. From the 'E911 Gateway' drop-down list (E911Gateway), select NG911 Callback Gateway.

15.9.4.2 Configuring the E9-1-1 Callback Timeout

The PSAP can use the ELIN to call back the E9-1-1 caller within a user-defined time interval (in minutes) from when the initial call established with the PSAP has been terminated. By default, an ELIN can be used for PSAP callback within 30 minutes after the call is terminated. You can change this to any value between 0 and 60:

➢ **To configure the E9-1-1 callback timeout**

1. Open the Advanced Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Advanced Parameters**).
2. In the 'E911 Callback Timeout' field (E911CallbackTimeout), enter the required callback timeout.

15.9.4.3 Configuring the SIP Release Cause Code for Failed E9-1-1 Calls

When a Lync client makes an emergency call, the call is routed through the Microsoft Mediation Server to the ELIN device, which sends it on to the SIP Trunk or PSTN. In some scenarios, the call may not be established due to either the destination (for example, busy or not found) or the ELIN device (for example, lack of resources or an internal error). In such a scenario, the Mediation Server requires that the ELIN device "reject" the call with the SIP release cause code 503 "Service Unavailable" instead of the designated release call. Such a release cause code enables the Mediation Server to issue a failover to another entity (for example, another ELIN device), instead of retrying the call or returning the release call to the user.

To support this requirement, you can configure the ELIN device to send a 503 "Service Unavailable" release cause code instead of SIP 4xx if an emergency call cannot be established:

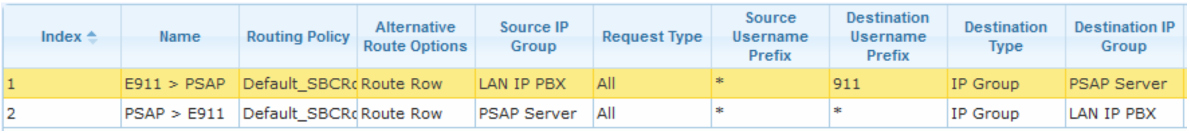
➢ **To enable SIP response 503 upon failed E911:**

1. Open the Advanced Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Advanced Parameters**).
2. From the 'Emergency Special Release Cause' drop-down list (EmergencySpecialReleaseCause), select **Enable**.

15.9.4.4 Configiuring SBC IP-to-IP Routing Rule for E9-1-1

To route incoming E9-1-1 calls to the emergency service provider's PSAP server, you need to configure routing rules in the IP-to-IP Routing table for routing between the emergency callers' IP Group and the PSAP server's IP Group. The only special configuration is to define the emergency number (e.g., 911) in the 'Destination Username Prefix' parameter of the IP Group belonging to the E9-1-1 callers. The following example shows IP-to-IP routing rules for E9-1-1 in a Lync environment:

Figure 15-43: Example of IP-to-IP Routing Rules for Lync E9-1-1



Index	Name	Routing Policy	Alternative Route Options	Source IP Group	Request Type	Source Username Prefix	Destination Username Prefix	Destination Type	Destination IP Group
1	E911 > PSAP	Default_SBCRc	Route Row	LAN IP PBX	All	*	911	IP Group	PSAP Server
2	PSAP > E911	Default_SBCRc	Route Row	PSAP Server	All	*	*	IP Group	LAN IP PBX

15.9.4.5 Configuring Location-Based Emergency Routing

The device identifies the geographical location of emergency callers by their ELIN numbers, which is present in the PIDF-LO XML body of received SIP INVITE messages. Therefore, you need to configure the device to route emergency calls to a destination (i.e., emergency center such as police) that is appropriate to the caller's ELIN number. As the destination of incoming calls is the emergency number (e.g., 999), the device needs to manipulate the destination number to a number that represents the caller's **local** emergency center (e.g., +4420999 for London police).

Note: Location-based emergency routing is applicable only to the Gateway application.

To add manipulation rules for location-based emergency routing, you need to use the Destination Phone Number Manipulation Table for IP-to-Tel Calls table. In this table, you need to use the ELIN number (e.g., 5000) as the source prefix, with the "ELIN" string value added in front of it (e.g., ELIN5000) which is used by the device to identify the number as an ELIN number (and **not** used for any other routing processes etc.). For each corresponding ELIN source number prefix entry, you need to configure the manipulation action required on the destination number so that the call is routed to the appropriate destination.

Following is an example of how to configure location-based emergency routing:

- **Assumptions:**
  - Company with offices in different cities -- London and Manchester.
  - Each city has its local police department.
  - In an emergency, users need to dial 999.
  - Company employs Microsoft Lync for communication between employers, and between employers and the external telephone network (PSTN). In other words, all employers are seemingly (virtual) in the same location in respect to the IP network.
  - ELIN numbers are used to identify the geographical location of emergency calls dialed by users:
    - London ELIN is 5000.
    - Manchester ELIN is 3000.

- **Configuration Objectives:**
    - Emergency calls received from London office users are routed by the device to the London police department (+4420999).
    - Emergency calls received from Manchester office users are routed by the device to the Manchester police department (+44161999).

    The international code, +44 for England is used for IP routing considerations, but can be omitted depending on your specific deployment.

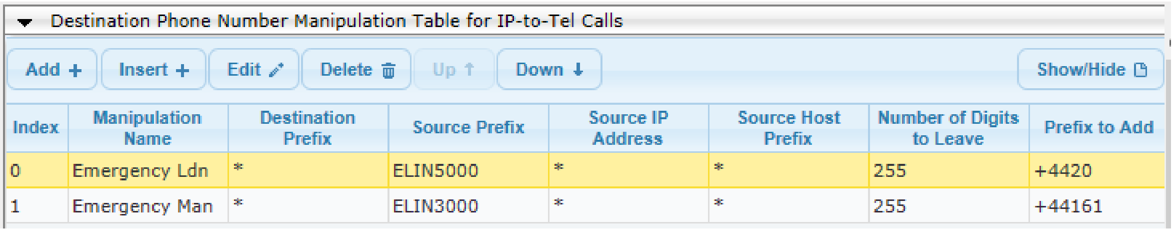
The above scenario is configured as follows:

1. Enable location-based emergency routing, by loading an ini file to the device with the following parameter setting:

```
E911Gateway = 2
```

2. In the Destination Phone Number Manipulation Table for IP-to-Tel Calls (**Configuration** tab > **VoIP** menu > **Gateway** > **Manipulations** > **Dest Number IP->Tel**), add the following two rules for manipulating the destination number of incoming emergency calls, based on ELIN numbers:

Figure 15-44: Destination Number Manipulation Rules for Location-Based Emergency Routing


Destination Phone Number Manipulation Table for IP-to-Tel Calls
Add +
Insert +
Edit
Delete
Up ↑
Down ↓
Show/Hide

Index	Manipulation Name	Destination Prefix	Source Prefix	Source IP Address	Source Host Prefix	Number of Digits to Leave	Prefix to Add
0	Emergency Ldn	*	ELIN5000	*	*	255	+4420
1	Emergency Man	*	ELIN3000	*	*	255	+44161

Index 0 manipulates the destination number for London emergency callers; Index 1 manipulates the destination number for Manchester emergency callers.

15.9.4.6 Viewing the ELIN Table

To view the ELIN table:

- CLI

```
# show voip e911
ELIN            Time      Count Index Call From
-----------------------------------------------------------
4257275678      22:11:52  0     2     4258359333
4257275999      22:11:57  0     3     4258359444
4257275615      22:12:03  0     0     4258359555
4257275616      22:11:45  0     1     4258359777
------------ Current Time: 22:12:40
```

- Using Syslog, by invoking the following Web command shell:

```
SIP / GateWay / E911Dump
```

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16 Quality of Experience

This chapter describes how to configure the Quality of Experience feature.

16.1 Reporting Voice Quality of Experience to SEM

The device can be configured to report voice (media) Quality of Experience (QoE) to AudioCodes' Session Experience Manager (SEM) server, a plug-in for AudioCodes EMS. The reports include real-time metrics of the quality of the actual call experience, which are then processed by the SEM.

SEM is a VoIP-quality monitoring and analysis tool. SEM provides comprehensive details on voice traffic quality, allowing system administrators to quickly identify, fix and prevent issues that could affect the voice calling experience in enterprise and service provider VoIP networks. IT managers and administrators can employ SEM in their VoIP networks to guarantee effective utilization, smooth performance, reliable QoS levels, and SLA fulfillment.

⚠ **Note:** For information on the SEM server, refer to the *SEM User's Manual*.

16.1.1 Configuring the SEM Server

The device can be configured to report QoE voice metrics to a single SEM server or to two SEM servers deployed in a Geographic Redundancy, High-Availability (HA) mode. Geographic Redundancy is when each SEM/EMS server is located in a different network subnet and has its own IP address. Thus, for the device to report QoE to both servers, you need to configure the IP address of each server. For normal HA mode, when both SEM/EMS servers are located in the same subnet, a single SEM/EMS server (global, virtual) IP address is used for all network components (EMS clients and managed devices). Thus, in such a setup, you need to configure only this IP address.

You can also configure the device to use a TLS connection with the SEM server. Before you can do this, configure a TLS Context (certificates) in the TLS Contexts table (see "Configuring TLS Certificate Contexts" on page 107). If no TLS Context is specified, the device uses the default TLS Context (ID 0).

You can also configure at what stage of the call the device must send the report to the SEM server. The report can be sent during the call or only at the end of the call. Reporting at the end of the call may be beneficial when network congestion occurs, as this reduces bandwidth usage over time.

⚠ **Note:** If a QoE traffic overflow is experienced between SEM and the device, the device sends the QoE data only at the end of the call, regardless of your settings.

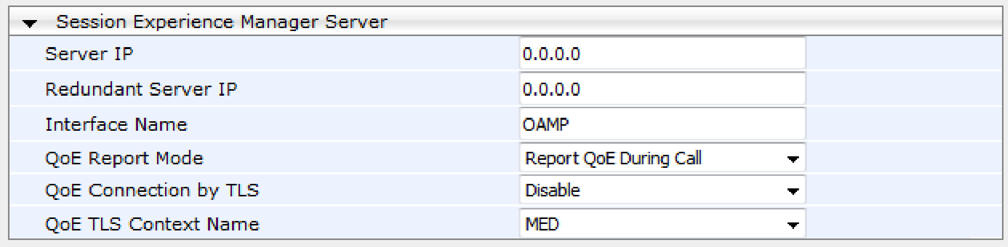
For a detailed description of the SEM parameters, see "Quality of Experience Parameters" on page 860.

➢ **To configure the SEM server address and other related features:**

1. Open the Session Experience Manager Server page (**Configuration** tab > **VoIP** menu

> **Quality of Experience** > **Session Experience Manager Server**).

Figure 16-1: Session Experience Manager Server Page



Session Experience Manager Server

Server IP	0.0.0.0
Redundant Server IP	0.0.0.0
Interface Name	OAMP
QoE Report Mode	Report QoE During Call
QoE Connection by TLS	Disable
QoE TLS Context Name	MED

2. Configure the address of the SEM server:
   a. In the 'Server IP' field, enter the primary SEM server's IP address.
   b. If Geographical-Redundancy HA mode exists, in the 'Redundant Server IP' field, enter the secondary SEM server's IP address.
   c. In the 'Interface Name' field, enter the device's IP network interface from which the device sends the reports to the SEM server.
3. From the 'QoE Report Mode' drop-down list, select when you want the device to send reports of a call to the SEM.
4. (Optional) Configure a TLS connection with the SEM server:
   a. From the 'QOE Connection by TLS' drop-down list, select **Enable**.
   b. From the 'Qoe TLS Context Name' drop-down list, select the desired TLS Context, which defines the TLS settings (e.g., certificates).
5. Click **Submit**, and then save ("burn") your settings to flash memory.

16.1.2 Configuring Clock Synchronization between Device and SEM

To ensure accurate call quality statistics and analysis by the SEM server, you must configure the device and the SEM server with the same clock source for clock synchronization. In other words, you need to configure them with the same NTP server.

The NTP server can be one of the following:

- AudioCodes EMS server (also acting as an NTP server)
- Third-party, external NTP server

Once you have determined the NTP server, all the elements--device, SEM, and EMS--must be configured with the same NTP server address.

To configure, the NTP server's address on the device, see "Configuring Automatic Date and Time using SNTP" on page 121.

16.1.3 Enabling RTCP XR Reporting to SEM

In order for the device to be able to send voice metric reports to the SEM, you need to enable the RTP Control Protocol Extended Reports (RTCP XR) VoIP management protocol. RTCP XR defines a set of voice metrics that contain information for assessing VoIP call quality and diagnosing problems. Enabling RTCP XR means that the device can send RTCP XR messages, containing the call-quality metrics, to the SEM server.

For enabling RTCP XR reporting, see "Configuring RTCP XR" on page 733. For configuring what to report to the SEM, see "Configuring Quality of Experience Profiles" on page 307.

16.2 Configuring Quality of Experience Profiles

The Quality of Experience feature lets you monitor the quality of voice calls traversing the device in your network. Voice-metric monitoring profiles (Quality of Experience Profiles) can be configured and applied to specific network links, including IP Groups (see "Configuring IP Groups" on page 340), Media Realms (see "Configuring Media Realms" on page 317), and Remote Media Subnets (see "Configuring Remote Media Subnets" on page 320).

The monitored voice metrics include the following:

- **Mean Opinion Score (MOS):** MOS is the average grade on a quality scale, expressed as a single number in the range of 1 to 5, where 1 is the lowest audio quality and 5 the highest audio quality.
- **Delay (or latency):** Time it takes for information to travel from source to destination (round-trip time).
- **Packet Loss:** Lost packets are RTP packets that are not received by the voice endpoint. Packet loss can result in choppy voice transmission.
- **Jitter:** Jitter can result from uneven delays between received voice packets. To space evenly, the device's jitter buffer adds delay. The higher the measurement, the greater the impact of the jitter buffer's delay on audio quality.
- **Residual Echo Return Loss (RERL):** An echo is a reflection of sound arriving at the listener at some time after the sound was initiated (often by the listener). Echo is typically caused by delay.

At any given time during a call, a voice metric can be in one of the following color-coded quality states:

- **Green:** Indicates good call quality
- **Yellow:** Indicates medium call quality
- **Red:** Indicates poor call quality

Quality of Experience Profiles let you configure quality thresholds per monitored voice metric. These are based on the following color-coded quality thresholds:

- **Green-Yellow threshold:** Lower threshold that indicates changes from Green to Yellow or vice versa when the threshold is crossed.
- **Yellow-Red threshold:** Higher threshold that indicates changes from Yellow to Red or vice versa when the threshold is crossed.

Hysteresis is also used to configure the threshold. This defines the amount of fluctuation from a threshold in order for the threshold to be considered as crossed (i.e., change in color state). Hysteresis is used to avoid false reports being sent by the device.

Each time a configured voice metric threshold is crossed (i.e., color changes), the device can do the following, depending on configuration:

- Report the change in the measured metrics to AudioCodes' Session Experience Manager (SEM) server. The SEM displays this call quality status for the associated SEM link (IP Group, Media Realm, or Remote Media Subnet). For configuring the SEM server's address, see "Configuring the SEM Server" on page 305.
- Determine access control and media enhancements based on measured metrics. Depending on the crossed threshold type, you can configure the device to accept or reject calls, or use an alternative IP Profile for the IP Group to which the call belongs. For more information, see "Configuring Media Enhancement Profiles" on page 313.
- Alternative routing based on measured metrics. If a call is rejected because of a crossed threshold, the device generates a SIP 806 response. You can configure this SIP response code as a reason for alternative routing (see "Configuring SIP Response Codes for Alternative Routing Reasons" on page 588).

Note: For your convenience, the device provides pre-configured Quality of Experience Profiles. One of these pre-configured profiles is the default Quality of Experience Profile. Therefore, if you do not configure a Quality of Experience Profile, this default is used.

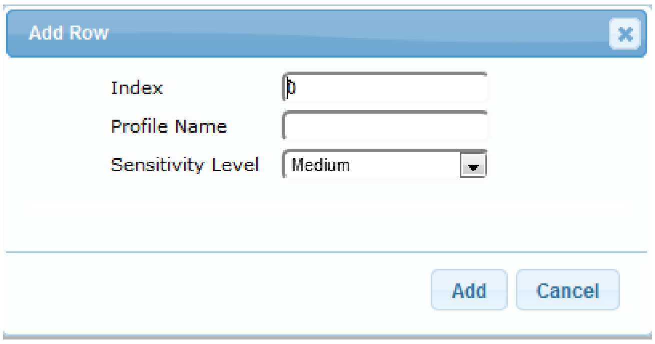
The following procedure describes how to configure Quality of Experience Profiles through the Web interface. You can also configure it through other management platforms:

- **Quality of Experience Profile table:** *ini* file (QoEProfile) or CLI (configure voip/qoe qoe-profile)
- **Quality of Experience Color Rules table:** *ini* file (QOEColorRules) or CLI (configure voip/qoe qoe-profile qoe-color-rules)

➢ To configure a QoE Profile:

1. Open the Quality of Experience Profile page (**Configuration** tab > **VoIP** menu > **Quality of Experience** > **Quality of Experience Profile**).
2. Click **Add**; the following dialog box appears:

Figure 16-2: Quality of Experience Profile Table - Add Row Dialog Box



Index: 0
Profile Name:
Sensitivity Level: Medium

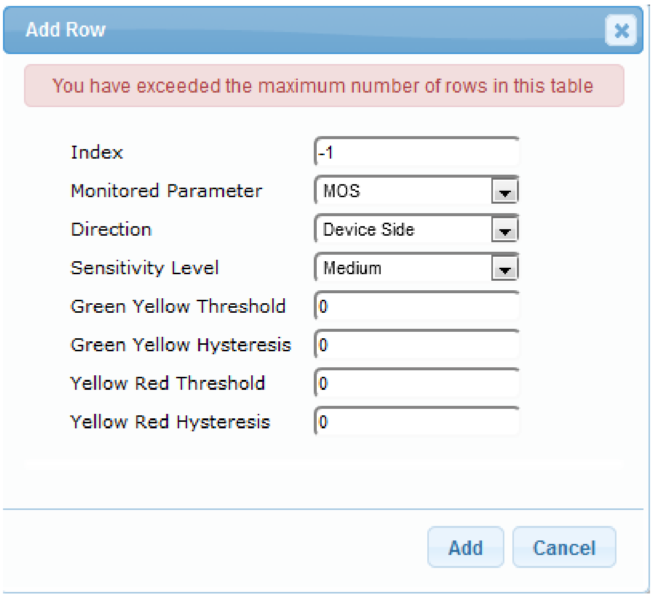
3. Configure a QoE Profile according to the parameters described in the table below.
4. Click **Add**.

Table 16-1: Quality of Experience Profile Table Parameter Descriptions

Parameter	Description
Index [QOEProfile_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Profile Name `name` [QOEProfile_Name]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 20 characters.
Sensitivity Level `sensitivity-level` [QOEProfile_SensitivityLevel]	Defines the pre-configured threshold profile to use. ▪ **[0]** User Defined = Need to define thresholds per monitored parameter in the Quality of Experience Color Rules table. ▪ **[1]** Low = Pre-configured low sensitivity thresholds. ▪ **[2]** Medium = (Default) Pre-configured medium sensitivity thresholds. ▪ **[3]** High = Pre-configured high sensitivity thresholds. Reporting is done for small fluctuations in parameter values.

5. In the Quality of Experience Profile page, select the QoE Profile index row for which you want to configure QoE thresholds, and then click the **Quality of Experience Color Rules** link located below the table; the Quality of Experience Color Rules page appears.
6. Click **Add**; the following dialog box appears:

Figure 16-3: Quality of Experience Table - Add Row Dialog Box



Add Row

You have exceeded the maximum number of rows in this table

Index: -1

Monitored Parameter: MOS

Direction: Device Side

Sensitivity Level: Medium

Green Yellow Threshold: 0

Green Yellow Hysteresis: 0

Yellow Red Threshold: 0

Yellow Red Hysteresis: 0

The figure above shows a configuration example where if the MOS value changes by 0.1 (hysteresis) to 3.3 or 3.5, the Green-Yellow threshold is crossed. The device considers a change to 3.3 as a Yellow state (i.e., medium quality) and a change to 3.5 as a Green state.

7. Configure a QoE Color rule according to the parameters described in the table below.
8. Click **Add**, and then save ("burn") your settings to flash memory.

Table 16-2: Quality of Experience Color Rules Table Parameter Descriptions

Parameter	Description
Index `index` [QOEColorRules_ColorRuleIndex]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Monitored Parameter `monitored-parameter` [QOEColorRules_monitoredParam]	Defines the parameter to monitor and report. ▪ **[0]** MOS (default) ▪ **[1]** Delay ▪ **[2]** Packet Loss ▪ **[3]** Jitter ▪ **[4]** RERL [Echo]
Direction `direction` [QOEColorRules_direction]	Defines the monitoring direction. ▪ **[0]** Device Side (default) ▪ **[1]** Remote Side

Parameter	Description
Sensitivity Level `sensitivity-level` [QOEColorRules_profile]	Defines the sensitivity level of the thresholds. - **[0]** User Defined = Need to define the thresholds in the parameters described below. - **[1]** Low = Pre-configured low sensitivity threshold values. Thus, reporting is done only if changes in parameters' values are significant. - **[2]** Medium = (Default) Pre-configured medium sensitivity threshold values. - **[3]** High = Pre-configured high sensitivity threshold values. Thus, reporting is done for small fluctuations in parameter values.
Green Yellow Threshold `green-yellow-threshold` [QOEColorRules_GreenYellowThreshold]	Defines the parameter threshold values between Green (good quality) and Yellow (medium quality) states. The valid threshold values are as follows: - MOS values are in multiples of 10. For example, to denote a MOS of 3.2, the value 32 (i.e., 3.2*10) must be entered. - Delay values are in msec. - Packet Loss values are in percentage (%). - Jitter is in msec. - Echo measures the Residual Echo Return Loss (RERL) in dB.
Green Yellow Hysteresis `green-yellow-hysteresis` [QOEColorRules_GreenYellowHysteresis]	Defines the fluctuation (change) from the value configured for the Green-Yellow threshold. When the threshold is exceeded by this hysteresis, the device sends a report to the SEM indicating this change. **Note:** If the monitored parameter crosses two thresholds at once (e.g., from Green to Red), the device ignores the hysteresis value and reports the call state change to the SEM.
Yellow Red Threshold `yellow-red-threshold` [QOEColorRules_YellowRedThreshold]	Defines the parameter threshold values between Yellow (medium quality) and Red (poor quality) states. The valid threshold values are as follows: - MOS values are in multiples of 10. For example, to denote a MOS of 3.2, the value 32 (i.e., 3.2*10) must be entered. - Delay values are in msec. - Packet Loss values are in percentage (%). - Jitter is in msec. - Echo measures the Residual Echo Return Loss (RERL) in dB.
Yellow Red Hysteresis `yellow-red-hysteresis` [QOEColorRules_YellowRedHysteresis]	Defines the fluctuation (change) from the value configured for the Yellow-Red threshold. When the threshold is exceeded by this hysteresis value, the device sends a report to the SEM indicating this change. **Note:** If the monitored parameter crosses two thresholds at once (e.g., from Green to Red), the device ignores the hysteresis value and reports the call state change to the SEM.

16.3 Configuring Bandwidth Profiles

Bandwidth Profiles enhance the device's monitoring of bandwidth utilization. A Bandwidth Profile defines bandwidth utilization thresholds for audio and/or video traffic (incoming and outgoing). Bandwidth Profiles can be assigned to IP Groups (see "Configuring IP Groups" on page 340), Media Realms (see "Configuring Media Realms" on page 317), and Remote Media Subnets (see "Configuring Remote Media Subnets" on page 320).

Each time a configured bandwidth threshold is crossed, the device can do the following, depending on configuration:

- Determine access control and media enhancements based on bandwidth utilization. Depending on the crossed threshold type, you can configure the device to accept or reject calls, or use an alternative IP Profile for the IP Group to which the call belongs. For more information, see "Configuring Media Enhancement Profiles" on page 313.
- Alternative routing based on bandwidth utilization. If a call is rejected because of a crossed threshold, the device generates a SIP 806 response. You can configure this SIP response code as a reason for alternative routing (see "Configuring SIP Response Codes for Alternative Routing Reasons" on page 588).
- Send an SNMP alarm (acMediaRealmBWThresholdAlarm). The device clears the alarm when bandwidth utilization returns to normal (within the thresholds).

The thresholds of Bandwidth Profiles use the same color-coding as the Quality of Experience Profile:

- **Green-Yellow threshold:** Lower threshold that indicates that the bandwidth exceeded a user-defined percentage of the configured threshold. This is referred to as a "Warning" alarm (i.e., warning you that bandwidth is nearing the threshold). When bandwidth goes over the threshold, the device considers it as a Yellow state; when it goes below the threshold, it considers it as a Green state.
- **Yellow-Red threshold:** Indicates that bandwidth has exceeded the configured threshold. When bandwidth goes over the threshold, the device considers it as a Red state; when it goes below the threshold, it considers it as a Yellow state.

Hysteresis is also used to configure the threshold. This defines the amount of fluctuation from a threshold in order for the threshold to be considered as crossed (i.e., change in color state). Hysteresis is used to avoid false reports.

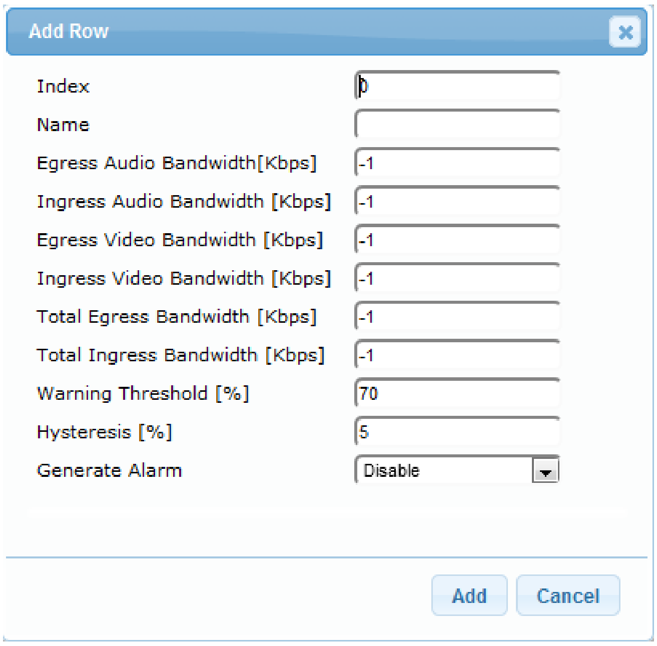
The following procedure describes how to configure Bandwidth Profiles through the Web interface. You can also configure it through ini file (BWProfile) or CLI (configure voip > qoe bw-profile).

➢ **To configure Bandwidth Profiles:**

1. Open the Bandwidth Profile page (**Configuration** tab > **VoIP** menu > **Quality of Experience** > **Bandwidth Profile**).

2. Click **Add**; the following dialog box appears:

Figure 16-4: Bandwidth Profile Table - Add Row Dialog Box



Index: 0
Name:
Egress Audio Bandwidth[Kbps]: -1
Ingress Audio Bandwidth [Kbps]: -1
Egress Video Bandwidth [Kbps]: -1
Ingress Video Bandwidth [Kbps]: -1
Total Egress Bandwidth [Kbps]: -1
Total Ingress Bandwidth [Kbps]: -1
Warning Threshold [%]: 70
Hysteresis [%]: 5
Generate Alarm: Disable

The figure above shows a configuration example where if the outgoing voice traffic threshold of 64,000 increases by 80% (70% warning threshold plus 10% hysteresis) to 115,200 (64,000 plus 51,200), a Yellow state occurs and an alarm is sent. If the threshold increases by 10%, a Red state occurs and an alarm is sent.

3. Configure a Bandwidth Profile according to the parameters described in the table below.
4. Click **Add**, and then reset the device with a save ("burn") to flash memory.

Table 16-3: Bandwidth Profile Table Parameter Descriptions

Parameter	Description
Index [BWProfile_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `name` [BWProfile_Name]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 20 characters.
Egress Audio Bandwidth `egress-audio-bandwidth` [BWProfile_EgressAudioBandwidth]	Defines the outgoing audio traffic threshold (in Kbps).
Ingress Audio Bandwidth `ingress-audio-bandwidth` [BWProfile_IngressAudioBandwidth]	Defines the incoming audio traffic threshold (in Kbps).
Egress Video Bandwidth `egress-video-bandwidth` [BWProfile_EgressVideoBandwidth]	Defines the outgoing video traffic threshold (in Kbps).
Ingress Video Bandwidth `ingress-video-bandwidth` [BWProfile_IngressVideoBandwidth]	Defines the incoming video traffic threshold (in Kbps).

Parameter	Description
Total Egress Bandwidth `total-egress-bandwidth` [BWProfile_TotalEgressBandwidth]	Defines the total (video and audio) outgoing bandwidth threshold (in Kbps).
Total Ingress Bandwidth `total-ingress-bandwidth` [BWProfile_TotalIngressBandwidth]	Defines the total (video and audio) incoming bandwidth threshold (in Kbps).
Warning Threshold `warning-threshold` [BWProfile_WarningThreshold]	Defines the threshold (in percentage) of the bandwidth thresholds that if exceeded is considered a Warning alarm (Green-Yellow threshold). This applies to any of the configured bandwidth thresholds. The Hysteresis is also added to this Warning threshold. For example, if set to 70% and the Hysteresis to 10%, when the current outgoing voice traffic exceeds 80% of the configured threshold, the Yellow state occurs and a Warning threshold alarm is sent if 'Generate Alarm' is set to **Enable**.
Hysteresis `hysteresis` [BWProfile_hysteresis]	Defines the bandwidth fluctuation (change) from the bandwidth threshold value (in percentage). The threshold is considered crossed if bandwidth exceeds the configured threshold plus this hysteresis, and a Red state occurs. For example, assume the parameter is set to 10% and the configured bandwidth threshold is set to 64000 Kbps. If current bandwidth reaches 70,400 Kbps (additional 10%), the threshold is considered crossed.
Generate Alarm `generate-alarms` [BWProfile_GenerateAlarms]	Enables the generation of an SNMP alarm if the threshold (with the hysteresis) is crossed. ▪ **[0]** Disable (default) ▪ **[1]** Enable If enabled, an alarm is sent if one of the following scenarios occurs: ▪ Warning threshold is exceeded (Warning severity - Yellow threshold). ▪ Any configured bandwidth threshold is exceeded (Major severity - Red threshold).

16.4 Configuring Media Enhancement Profiles

Media Enhancement Profiles provides support for access control and media quality enhancements based on call quality measurements (configured in "Configuring Quality of Experience Profiles" on page 307) and bandwidth utilization (configured in "Configuring Bandwidth Profiles" on page 311). These profiles contain color-coded thresholds that are used to trigger access control and/or media enhancements.

The Media Enhancement Profile table lets you configure any one of the following actions when a specific color-coded threshold (Green-Yellow or Yellow-Red) is crossed for a specific monitored voice metrics (e.g., MOS) or bandwidth (e.g., Egress Audio Bandwidth):

- Reject new calls until the voice metrics or bandwidth returns to below the threshold. This can be used, for example, to reject new calls when bandwidth threshold is exceeded.
- Use a different IP Profile. For example, if packet loss is detected, the IP Group (to which the Media Enhancement Rule is later assigned) can switch to an IP Profile configured with a higher RTP redundancy level. The ability to use a different IP Profile

when call quality or bandwidth thresholds are crossed provides a wide range of options for media enhancement and traffic shaping. For example, it may be used to:

- switch to a low bit-rate coder,
- negotiate different p-time (and perform transrating if required),
- increase RTP redundancy level,
- or block video calls.

- Accept calls

A Media Enhancement Profile can later be assigned to an IP Group (in the IP Group table). However, when the device analyzes the call and determines whether Media Enhancement Profile should be applied or not, it searches for the "most relevant" Quality of Experience Profile or Bandwidth Profile in the following order: 1) Remote Media Subnet, 2) Media Realm, and then 3) IP Group. Thus, a Media Enhancement Profile associated with a specific IP Group may actually "respond" to Quality of Experience or bandwidth thresholds crossed at the Media Realm or Remote Media Subnet level.

Notes:

- The color-coded threshold is first calculated for the IP Group and only then for the Media Realm. The device uses the "worst" color-coded threshold crossing. For example, if a Media Realm crossed a Green-Yellow threshold and an IP Group a Yellow-Red threshold, the action defined for the Red color state is used.
- The device applies Media Enhancements Profiles on new calls **only**, based on the information gathered from previous and/or currently established calls.

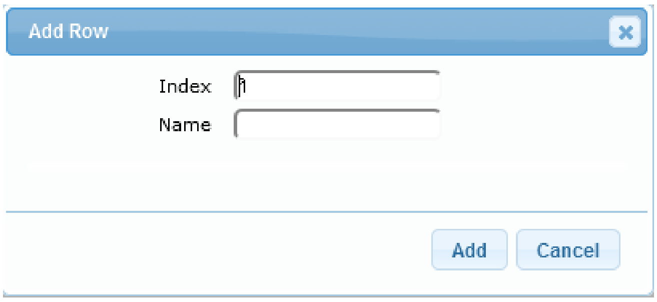
The following procedure describes how to configure Media Enhancement Profiles through the Web interface. You can also configure it through other management platforms:

- **Media Enhancement Profile table:** *ini* file (MediaEnhancementProfile) or CLI (configure voip/qoe media-enhancement)
- **Media Enhancement Rules table:** *ini* file (MediaEnhancementRules) or CLI (configure voip/qoe media-enhancement-rules)

➢ To configure a Media Enhancement Profile:

1. Open the Media Enhancement Profile page (**Configuration** tab > **VoIP** menu > **Quality of Experience** > **Media Enhancement Profile**).
2. Click **Add**; the following dialog box appears:

Figure 16-5: Media Enhancement Profile Table - Add Row Dialog Box



Add Row

Index: 1

Name:

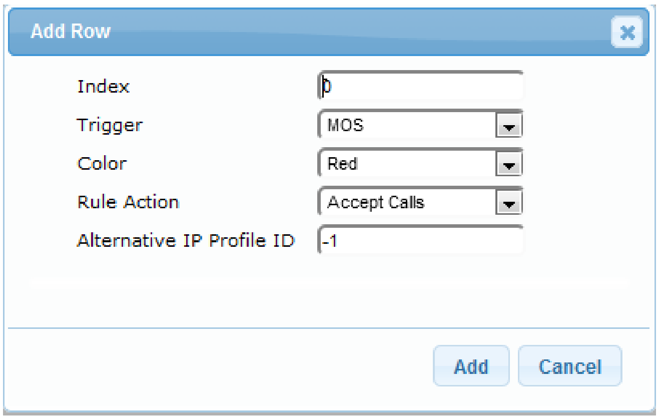
3. Configure a Media Enhancement Profile according to the parameters described in the table below.
4. Click **Add**.

Table 16-4: Media Enhancement Profile Table Parameter Descriptions

Parameter	Description
Index [MediaEnhancementProfile_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `profile-name` [MediaEnhancementProfile_ProfileName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 20 characters.

5. In the Media Enhancement Profile table, select the required Media Enhancement Profile index row, and then click the **Media Enhancement Rules** link located below the table; the Media Enhancement Rules page appears.
6. Click **Add**; the following dialog box appears:

Figure 16-6: Media Enhancement Rules Table - Add Row Dialog Box


Add Row

Index	0
Trigger	MOS
Color	Red
Rule Action	Accept Calls
Alternative IP Profile ID	-1

7. Configure a Media Enhancement Rule according to the parameters described in the table below.
8. Click **Add**, and then reset the device with a save ("burn") to flash memory.

Table 16-5: Media Enhancement Rules Table Parameter Descriptions

Parameter	Description
Index `rule-index` [MediaEnhancementRules_RuleIndex]	Defines the index of the table row entry.
Trigger `trigger` [MediaEnhancementRules_Trigger]	Defines the monitored metrics parameter or bandwidth associated with this rule. ▪ **[0]** MOS (default) ▪ **[1]** Delay ▪ **[2]** Packet Loss ▪ **[3]** Jitter ▪ **[4]** Bandwidth
Color `color` [MediaEnhancementRules_Color]	Defines the color-coded threshold change of the monitored metrics or bandwidth (configured in the 'Trigger' parameter) for which this rule is done. ▪ **[0]** Red (default) = Yellow-to-Red threshold is crossed. ▪ **[1]** Yellow = Green-to-Yellow threshold is crossed.

Parameter	Description
Rule Action `action-rule` [MediaEnhancementRules_ActionRule]	Defines the action that the device performs when the color-coded threshold is crossed: ▪ **[0]** Accept Calls (default) ▪ **[1]** Reject Calls ▪ **[2]** Alternative IP Profile = An alternative IP Profile is used, as configured in the 'Value' field (below). **Notes:** ▪ If the parameter is set to a restrictive action (i.e., **Reject Calls** or **Alternative IP Profile**) for Yellow and no action is set for Red, the device also applies the Yellow action to Red, if this color-coded threshold occurs. ▪ If the parameter is set to a permissive action (i.e., **Accept Calls**) for Red and no action is set for Yellow, the device applies the same action to Yellow, if this color-coded threshold occurs.
Alternative IP Profile ID `value` [MediaEnhancementRules_ActionValue]	Defines an alternative IP Profile ID for the IP Group that is associated with this rule, if this rule is applied. The parameter is applicable only if the 'Rule Action' parameter is set to **Alternative IP Profile**.

17 Control Network

This section describes configuration of the network at the SIP control level.

17.1 Configuring Media Realms

The Media Realm table lets you configure a pool of up to 64 SIP media interfaces, termed *Media Realms*. Media Realms lets you divide a Media-type interface (configured in the Interface table) into several media realms, where each realm is specified by a UDP port range. Media Realms also define the maximum number of permitted media sessions.

Once configured, to apply Media Realms to specific calls, you need to assign them to any of the following configuration entities:

- IP Groups (see "Configuring IP Groups" on page 340)
- SIP Interfaces (see "Configuring SIP Interfaces" on page 333)

You can also apply the device's Quality of Experience feature to Media Realms:

- **Quality of Experience Profile:** Call quality monitoring based on thresholds for voice metrics (e.g., MOS) can be applied per Media Realm. For example, if MOS is considered poor, calls on this Media Realm can be rejected. For configuring Quality of Experience Profiles, see "Configuring Quality of Experience Profiles" on page 307.
- **Bandwidth Profile:** Bandwidth utilization thresholds can be applied per Media Realm. For example, if bandwidth thresholds are crossed, the device can reject any new new calls on this Media Realm. For configuring Bandwidth Profiles, see "Configuring Bandwidth Profiles" on page 311.

The Media Realm table provides sub-tables ("child" tables) that let you configure the following:

- Remote Media Subnets: Defines remote destination subnets per Media Realm and assigns each subnet a Quality of Experience Profile and Bandwidth Profile. For more information, see "Configuring Remote Media Subnets" on page 320.
- Media Realm Extensions: Defines port ranges for multiple Media-type interfaces per Media Realm. For more information, see "Configuring Media Realm Extensions" on page 322.

Notes:

- The Media Realm assigned to an IP Group overrides any other Media Realm assigned to any other configuration entity associated with the call.
- If you modify a Media Realm that is currently being used by a call, the device does not perform Quality of Experience for the call.
- If you delete a Media Realm that is currently being used by a call, the device maintains the call until the call parties end the call.

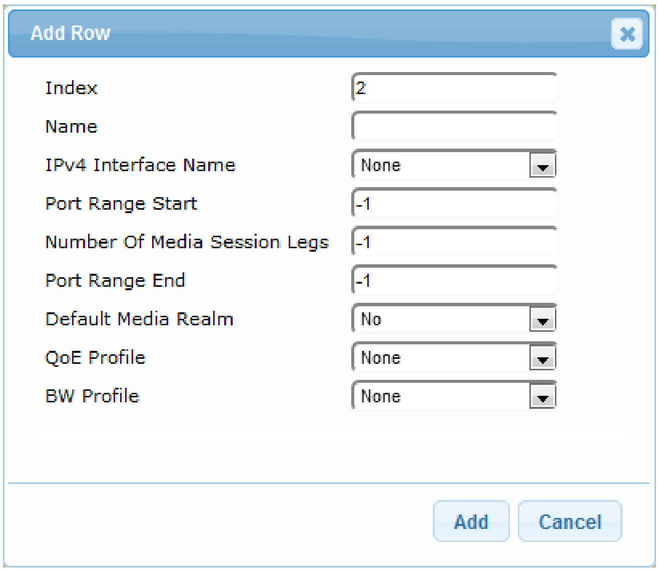
The following procedure describes how to configure Media Realms through the Web interface. You can also configure it through ini file (CpMediaRealm) or CLI (configure voip > voip-network realm).

➢ **To configure a Media Realm:**

1. Open the Media Realm table (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Media Realm Table**).

2. Click **Add**; the following dialog box appears:

Figure 17-1: Media Realm Table - Add Row Dialog Box



Add Row

Index: 2
Name:
IPv4 Interface Name: None
Port Range Start: -1
Number Of Media Session Legs: -1
Port Range End: -1
Default Media Realm: No
QoE Profile: None
BW Profile: None

3. Configure the Media Realm according to the parameters described in the table below.
4. Click **Add**.

Table 17-1: Media Realm Table Parameter Descriptions

Parameter	Description
Index [CpMediaRealm_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `name` [CpMediaRealm_MediaRealmName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. **Notes:** ▪ The parameter is mandatory. ▪ Each row must be configured with a unique name.
IPv4 Interface Name `ipv4` [CpMediaRealm_IPv4IF]	Assigns an IPv4 network interface to the Media Realm. This is the name of the interface as configured in the 'Interface Name' parameter in the Interface table. By default, no value is defined (**None**).
IPv6 Interface Name `ipv6if` [CpMediaRealm_IPv6IF]	Assigns an IPv6 network interface to the Media Realm. This is the name of the interface as configured for the 'Interface Name' parameter in the Interface table. By default, no value is defined (None).
Port Range Start `port-range-start` [CpMediaRealm_PortRangeStart]	Defines the starting port for the range of media interface UDP ports. By default, no value is defined. **Notes:** ▪ You must either configure all your Media Realms with port ranges or all without; not some with and some without. ▪ The available UDP port range is according to the BaseUDPport parameter. For more information, see "Configuring RTP Base UDP Port" on page 191. ▪ The base UDP port number (BaseUDPPort parameter) must be greater than the highest UDP port configured for

Parameter	Description
	a SIP Interface (see Configuring SIP Interfaces on page 333). For example, if your highest configured UDP port for a SIP Interface is 6060, you must configure the BaseUDPPort parameter to any value greater than 6060. ▪ The port must be different from ports configured for SIP traffic (i.e., ports configured for SIP Interfaces). For example, if the RTP port range is 6000 to 6999, the SIP port can be less than 6000 or greater than 6999.
Number of Media Session Legs `session-leg` [CpMediaRealm_MediaSessionLeg]	Defines the number of media sessions for the configured port range. By default, no value is defined.

Port Range End
`port-range-end`
[CpMediaRealm_PortRangeEnd]
	(Read-only field) Displays the ending port for the range of media interface UDP ports. The device automatically populates the parameter with a value, calculated by the summation of the 'Port Range Start' parameter and 'Number of Media Session Legs' parameter (multiplied by the port spacing) minus 1:
	`start port + (sessions * port spacing) - 1`
	For example, a port starting at 6,000, 5 sessions and 10 port spacing:
	`6,000 + (5 * 10) - 1 = 6,000 + (50) - 1 = 6,000 + 49 = 6,049`
	The device allocates the UDP ports for RTP, RTCP and T.38 in "jumps" (spacing) of 10 (default. For example, if the port range starts at 6000 and the UDP port spacing is 10, the available ports include 6000, 6010, 6020, 6030, and so on (depending on number of media sessions).
	For RTCP and T.38 traffic, the port offset from the RTP port used for the voice session (channel) is one and two, respectively. For example, if the voice session uses RTP port 6000, the RTCP port and T.38 port for the session is 6001 and 6002, respectively. However, you can configure the device to use the same port for RTP and T.38 packets, by setting the T38UseRTPPort parameter to 1.

Default Media Realm `is-default` [CpMediaRealm_IsDefault]	Defines the Media Realm as the default Media Realm. The default Media Realm is used for SIP Interfaces and IP Groups for which you have not assigned a Media Realm. ▪ **[0]** No (default) ▪ **[1]** Yes **Notes:** ▪ You can configure the parameter to **Yes** for only **one** Media Realm; all the other Media Realms must be configured to **No**. ▪ If you do not configure the parameter (i.e., the parameter is **No** for all Media Realms), the device uses the first Media Realm in the table as the default. ▪ If the table is not configured, the default Media Realm includes all configured media interfaces.

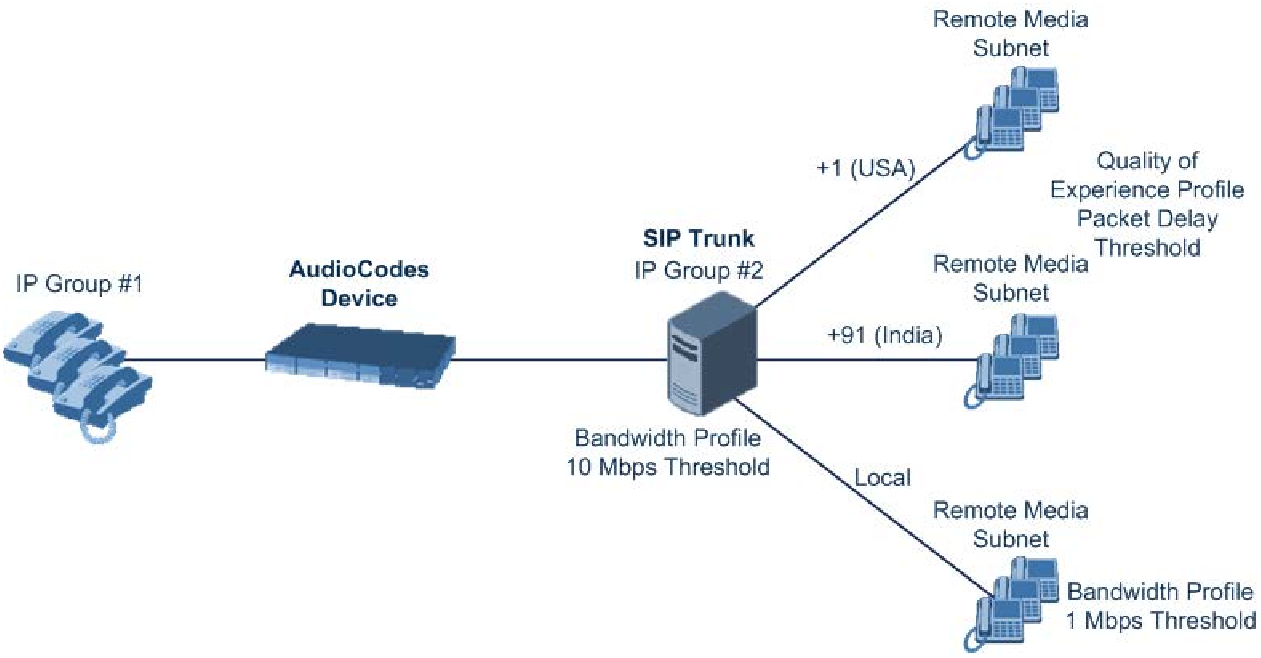
Parameter	Description
QoE Profile `qoe-profile` [CpMediaRealm_QoeProfile]	Assigns a QoE Profile to the Media Realm. By default, no value is defined (**None**). For configuring QoE Profiles, see "Configuring Quality of Experience Profiles" on page 307.
BW Profile `bw-profile` [CpMediaRealm_BWProfile]	Assigns a Bandwidth Profile to the Media Realm. By default, no value is defined (**None**). For configuring Bandwidth Profiles, see "Configuring Bandwidth Profiles" on page 311.

17.1.1 Configuring Remote Media Subnets

Remote Media Subnets define destination subnets for media (RTP/SRTP) traffic on a specific Media Realm. Each Remote Media Subnet can be assigned different call quality (Quality of Experience Profile) and bandwidth utilization (Bandwidth Profile) profiles. These profiles are configured in "Configuring Quality of Experience Profiles" on page 307 and "Configuring Bandwidth Profiles" on page 311, respectively. Thus, you can apply these profiles to remote media subnets instead of Media Realms or IP Groups. You can configure up to five Remote Media Subnets per Media Realm.

The figure below illustrates an example for implementing Remote Media Subnets. IP Group #2 represents a SIP Trunk which routes international (USA and India) and local calls. As international calls are typically more prone to higher delay than local calls, different Quality of Experience Profiles are assigned to them. This is done by creating Remote Media Subnets for each of these call destinations and assigning each Remote Media Subnet a different Quality of Experience Profile. A Quality of Experience Profile that defines a packet delay threshold is assigned to the international calls, which if crossed, a different IP Profile is used that defines higher traffic priority to voice over other traffic. In addition, IP Group #2 has a 10-Mbps bandwidth threshold and a "tighter" bandwidth limitation (e.g., 1 Mbps) is allocated to local calls. If this limit is exceeded, the device rejects new calls to this Remote Media Subnet.

Figure 17-2: Remote Media Subnets Example

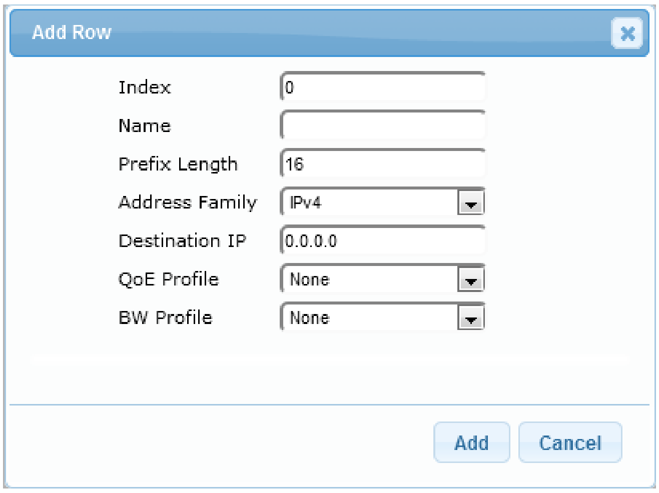


The following procedure describes how to configure Remote Media Subnets through the Web interface. You can also configure it through ini file (RemoteMediaSubnet) or CLI (configure voip > voip-network realm remote-media-subnet).

➢ **To configure a Remote Media Subnet:**

1. Open the Media Realm table (see "Configuring Media Realms" on page 317).
2. Select the Media Realm row for which you want to add Remote Media Subnets, and then click the **Remote Media Subnet** link located below the table; the Remote Media Subnet table appears.
3. Click **Add**; the following dialog box appears:

Figure 17-3: Remote Media Subnet Table - Add Row Dialog Box



Add Row

Index: 0

Name:

Prefix Length: 16

Address Family: IPv4

Destination IP: 0.0.0.0

QoE Profile: None

BW Profile: None

4. Configure the Remote Media Subnet according to the parameters described in the table below.
5. Click **Add**.

Table 17-2: Remote Media Subnet Table Parameter Descriptions

Parameter	Description
Index [RemoteMediaSubnet_RemoteMediaSubnetIndex]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name name [RemoteMediaSubnet_RemoteMediaSubnetName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 20 characters. **Note:** Each row must be configured with a unique name.
Prefix Length `prefix-length` [RemoteMediaSubnet_PrefixLength]	Defines the subnet mask in Classless Inter-Domain Routing (CIDR) notation. For example, 16 denotes 255.255.0.0. The default is 16.
Address Family `address-family` [RemoteMediaSubnet_AddressFamily]	Defines the IP address protocol. ▪ **[2]** IPv4 (default) ▪ **[10]** IPv6

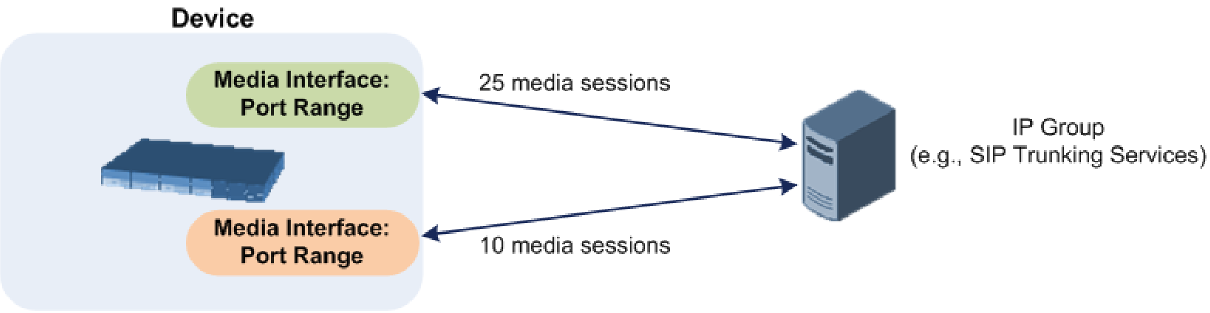
Parameter	Description
Destination IP `dst-ip-address` [RemoteMediaSubnet_DstIPAddress]	Defines the IP address of the destination. The default is 0.0.0.0.
QOE Profile Name `qoe-profile` [RemoteMediaSubnet_QOEProfileName]	Assigns a Quality of Experience Profile to the Remote Media Subnet. By default, no value is defined (**None**). For configuring QoE Profiles, see "Configuring Quality of Experience Profiles" on page 307.
BW Profile Name `bw-profile` [RemoteMediaSubnet_BWProfileName]	Assigns a Bandwidth Profile to the Remote Media Subnet. By default, no value is defined (**None**). For configuring Bandwidth Profiles, see "Configuring Bandwidth Profiles" on page 311.

17.1.2 Configuring Media Realm Extensions

The Media Realm Extension table lets you configure 2 Media Realm Extensions. A Media Realm Extension is associated with a specific Media Realm and defines a port range and the number of media sessions for a specific Media-type network interface (configured in the IP Interfaces table). Therefore, a Media Realm Extension enhances a Media Realm by allowing you to define different port ranges, media sessions, and network interface than is defined by the associated Media Realm (i.e., the Media Realm is distributed across multiple interfaces).

Media Realm Extensions can be useful, for example, to overcome limitations of the maximum number of media ports supported per interface. Instead of configuring only a single Media Realm in the Media Realm table (see "Configuring Media Realms" on page 317), you can also configure additional "Media Realms" in the Media Realm Extensions table associated with the single Media Realm. An IP Group that is associated with a Media Realm configured with Media Realm Extensions, allocates its media sessions / ports between the different interfaces, as configured by the Media Real and its associated Media Realm Extensions. For example, two Media Realm Extensions could be configured, whereby one allocates 25 media sessions on interface "LAN-1" and another, 25 sessions on interface "LAN-2". The Media Realm associated with these Media Realm Extensions would be assigned to the relevant IP Group.

Figure 17-4: Example of Implementation of Media Realm Extensions


Device
Media Interface: Port Range
Media Interface: Port Range
25 media sessions
10 media sessions
IP Group
(e.g., SIP Trunking Services)

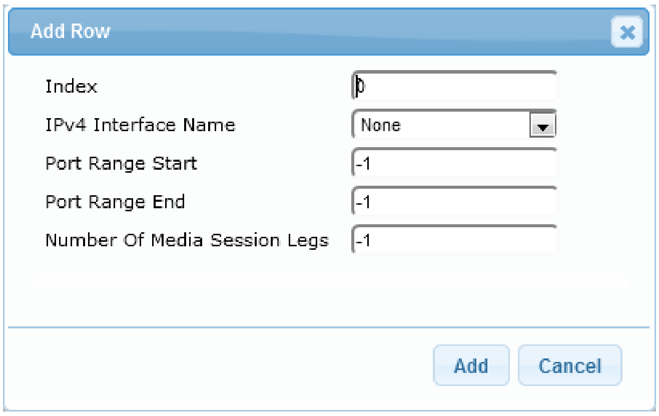
The following procedure describes how to configure Media Realm Extensions through the Web interface. You can also configure it through ini file (MediaRealmExtension).

➢ **To configure a Media Realm Extension:**

1. Open the Media Realm table (see "Configuring Media Realms" on page 317).

2. Select the Media Realm row for which you want to add Remote Media Extensions, and then click the **Media Realm Extension** link located below the table; the Media Realm Extension table appears.
3. Click **Add**; the following dialog box appears:

Figure 17-5: Media Realm Extension Table - Add Row Dialog Box



Index: 0
IPv4 Interface Name: None
Port Range Start: -1
Port Range End: -1
Number Of Media Session Legs: -1

4. Configure the Media Realm Extension according to the parameters described in the table below.
5. Click **Add**.

Table 17-3: Media Realm Extension Table Parameter Descriptions

Parameter	Description
Index [MediaRealmExtension_ExtensionIndex]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
IPv4 Interface Name [MediaRealmExtension_IPv4IF]	Assigns an IPv4 network interface (configured in the Interface table) to the Media Realm Extension. By default, no value is defined (**None**). For configuring IP network interfaces, see "Configuring IP Network Interfaces" on page 133. **Note:** ▪ The parameter is mandatory. ▪ You must configure the Media Realm Extension with an IP network interface that has the same IP version(s) as the Media Realm to which the Media Realm Extension is associated. If the associated Media Realm is assigned both an IPv4 and IPv6 network interface, you also need to assign the Media Realm Extension with both an IPv4 and IPv6 network interface. For example, if the associated Media Realm is assigned only an IPv4 network interface, you also need to assign the Media Realm Extension with an IPv4 network interface.

Parameter	Description
IPv6 Interface Name [MediaRealmExtension_IPv6IF]	Assigns an IPv6 network interface (configured in the Interface table) to the Media Realm Extension. By default, no value is defined (None). **Note:** ▪ The parameter is mandatory. ▪ You must configure the Media Realm Extension with an IP network interface that has the same IP version(s) as the Media Realm to which the Media Realm Extension is associated. If the associated Media Realm is assigned both an IPv4 and IPv6 network interface, you also need to assign the Media Realm Extension with both an IPv4 and IPv6 network interface. For example, if the associated Media Realm is assigned only an IPv6 network interface, you also need to assign the Media Realm Extension with an IPv6 network interface.
Port Range Start [MediaRealmExtension_PortRangeStart]	Defines the first (lower) port in the range of media UDP ports for the Media Realm Extension. By default, no value is defined. **Notes:** ▪ You must either configure all your Media Realms with port ranges or all without; not some with and some without. ▪ The available UDP port range is according to the BaseUDPport parameter. For more information, see "Configuring RTP Base UDP Port" on page 191.
Port Range End [MediaRealmExtension_PortRangeEnd]	Defines the last (upper) port in the range of media UDP ports for the Media Realm Extension. **Note:** It is unnecessary to configure the parameter. The device automatically populates the parameter with a value, calculated by the summation of the 'Number of Media Session Legs' parameter (multiplied by the port chunk size) and the 'Port Range Start' parameter. After you have added the Media Realm Extension row to the table, the parameter is displayed with the calculated value.
Number Of Media Session Legs [MediaRealmExtension_MediaSessionLeg]	Defines the number of media sessions for the port range. For example, 100 ports correspond to 10 media sessions, since ports are allocated in chunks of 10. By default, no value is defined. **Note:** The parameter is mandatory.

17.2 Configuring SRDs

The SRD table lets you configure up to 41 signaling routing domains (SRD) for Gateway and SBC calls. The SRD is a logical representation of an entire SIP-based VoIP network (Layer 5) consisting of groups of SIP users and servers. The SRD is associated with all the configuration entities (e.g., SIP Interfaces and IP Groups) required for routing calls within the network. Typically, only a **single** SRD is required (recommended) for most deployments. Multiple SRDs are only required for multi-tenant deployments, where the physical device is "split" into multiple logical devices. For more information on multi-tenant architecture, see "Multiple SRDs for Multi-tenant Deployments" on page 329.

As the device is shipped with a default SRD ("DefaultSRD" at Index 0), if your deployment requires only one SRD, you can use the default SRD instead of creating a new one. When only one SRD is employed and you create other related configuration entities (e.g., SIP Interfaces), the default SRD is automatically assigned to the new configuration entity. Therefore, when employing a single-SRD configuration topology, there is no need to handle SRD configuration (i.e., transparent).

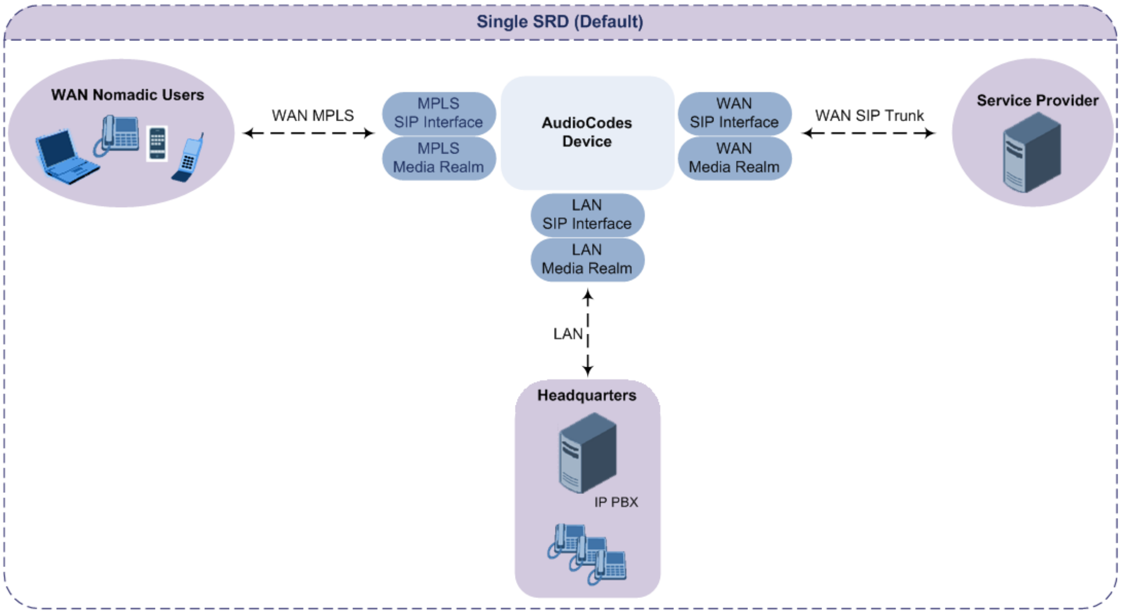
SRDs are associated with the following configuration entities:

- SIP Interface (mandatory) - see "Configuring SIP Interfaces" on page 333
- IP Group (mandatory) - see "Configuring IP Groups" on page 340
- Proxy Set (mandatory) - see "Configuring Proxy Sets" on page 352
- Admission Control rule - see Configuring Admission Control Table on page 561
- Classification rule - see Configuring Classification Rules on page 569

As mentioned previously, if you use only a single SRD, the device automatically assigns it to the above-listed configuration entities.

As each SIP Interface defines a different Layer-3 network (see "Configuring SIP Interfaces" on page 333 for more information) on which to route or receive calls and as you can assign multiple SIP Interfaces to the same SRD, for most deployment scenarios (even for multiple Layer-3 network environments), you only need to employ a single SRD to represent your VoIP network (Layer 5). For example, if your VoIP deployment consists of an Enterprise IP PBX (LAN), a SIP Trunk (WAN), and far-end users (WAN), you would only need a single SRD. The single SRD would be assigned to three different SIP Interfaces, where each SIP Interface would represent a specific Layer-3 network (IP PBX, SIP Trunk, or far-end users) in your environment. The following figure provides an example of such a deployment:

Figure 17-6: Deployment using a Single SRD



Notes:

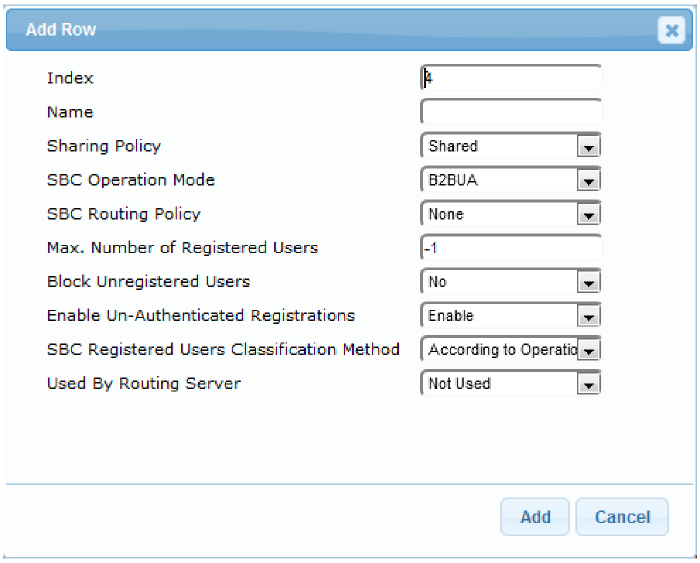
- It is recommended to use a single-SRD configuration topology, unless you are deploying the device in a multi-tenant environment, in which case, multiple SRDs are required.
- Each SIP Interface, Proxy Set, and IP Group can be associated with only one SRD.
- If you have upgraded your device to Version 7.0 and your device was configured with multiple SRDs but not operating in a multi-tenant environment, it is recommended to gradually change your configuration to a single SRD topology.
- If you upgrade the device from an earlier release to Version 7.0, your previous SRD configuration is fully preserved regarding functionality. The same number of SRDs is maintained, but the configuration elements are changed to reflect the configuration topology of Version 7.0. Below are the main changes in configuration topology when upgrading to Version 7.0:
    - √ The SIP Interface replaces the associated SRD in several tables (due to support for multiple SIP Interfaces per SRD).
    - √ Some fields in the SRD table were duplicated or moved to the SIP Interface table.
    - √ Indices used for associating configuration entities in tables are changed to row pointers (using the entity's name).
    - √ Some tables are now associated (mandatory) with an SRD (SIP Interface, IP Group, Proxy Set, and Classification).
    - √ Some fields used for associating configuration entities in tables now have a value of **Any** to distinguish between **Any** and **None** (deleted entity or not associated).

The following procedure describes how to configure SRDs through the Web interface. You can also configure it through ini file (SRD) or CLI (configure voip > voip-network srd).

➢ To configure an SRD:

1. Open the SRD table (**Configuration** tab > **VoIP** menu > **VoIP Network** > **SRD Table**).
2. Click **Add**; the following dialog box appears:

Figure 17-7: SRD Table - Add Row Dialog Box



Add Row

Index	4
Name	
Sharing Policy	Shared
SBC Operation Mode	B2BUA
SBC Routing Policy	None
Max. Number of Registered Users	-1
Block Unregistered Users	No
Enable Un-Authenticated Registrations	Enable
SBC Registered Users Classification Method	According to Operatio
Used By Routing Server	Not Used

3. Configure an SRD according to the parameters described in the table below.
4. Click **Add**.

Table 17-4: SRD Table Parameter Descriptions

Parameter	Description
Index [SRD_Index]	Defines an index for the new table row. **Note:** Each row must be configured with a unique index.
Name `name` [SRD_Name]	Defines an arbitrary name to easily identify the row. The valid value can be a string of up to 40 characters. **Notes:** ▪ The parameter is mandatory. ▪ Each row must be configured with a unique name.
Sharing Policy `type` [SRD_SharingPolicy]	Defines the sharing policy of the SRD, which determines whether the SRD shares its SIP resources (SIP Interfaces, Proxy Sets, and IP Groups) with all other SRDs (Shared and Isolated). ▪ [0] Shared = (Default) SRD shares its resources with other SRDs (Isolated and Shared) and calls can thus be routed between the SRD and other SRDs. ▪ [1] Isolated = SRD does not share its resources with other SRDs and calls cannot be routed between the SRD and other Isolated SRDs. However, calls can be routed between the SRD and other Shared SRDs. For more information on SRD Sharing Policy, see Multiple SRDs for Multi-tenant Deployments on page 329.
SBC Operation Mode `sbc-operation-mode` [SRD_SBCOperationMode]	Defines the device's operational mode for the SRD. ▪ [0] B2BUA = (Default) Device operates as a back-to-back user agent (B2BUA), changing the call identifiers and headers between the inbound and outbound legs. ▪ [1] Call Stateful Proxy = Device operates as a Stateful Proxy, passing the SIP message transparently between inbound and outbound legs. In other words, the same SIP dialog identifiers (tags, Call-Id and CSeq) occur on both legs (as long as no other configuration disrupts the CSeq compatibleness). For more information on B2BUA and Stateful Proxy modes, see B2BUA and Stateful Proxy Operating Modes on page 528. **Notes:** ▪ The settings of the parameter also determines the default behavior of related parameters in the IP Profile table (SBCRemoteRepresentationMode, SBCKeepVIAHeaders, SBCKeepUserAgentHeader, SBCKeepRoutingHeaders, SBCRemoteMultipleEarlyDialogs). ▪ If the 'SBC Operation Mode' parameter is configured in the IP Group table, the 'SBC Operation Mode' parameter in the SRD table is ignored.

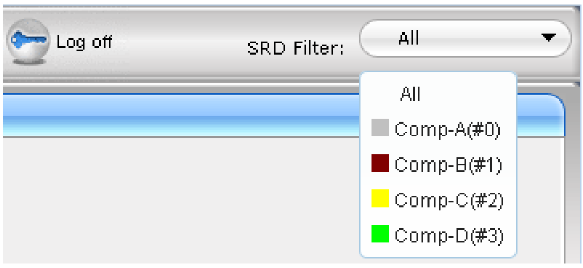
Parameter	Description
SBC Routing Policy `sbc-routing-policy-name` [SRD_SBCRoutingPolicyName]	Assigns an SBC Routing Policy to the SRD. By default, no value is defined (None) if you have configured multiple SBC Routing Policies. If you have configured only one SBC Routing Policy, the device assigns it to the SRD by default. For more information on SBC Routing Policies, see Configuring SBC Routing Policy Rules on page 590. **Notes:** ▪ If you have assigned an SBC Routing Policy to a Classification rule that is associated with the SRD, the SBC Routing Policy assigned to the SRD is ignored. ▪ You can assign the same Routing Policy to multiple SRDs.
Max. Number of Registered Users `max-reg-users` [SRD_MaxNumOfRegUsers]	Defines the maximum number of users belonging to the SRD that can register with the device. The default is -1, which means that the number of allowed user registrations is unlimited.
Block Unregistered Users `block-un-reg-users` [SRD_BlockUnRegUsers]	Enables the device to block (reject) incoming calls (INVITE requests) from unregistered users belonging to the SRD. ▪ [0] No = (Default) Calls from unregistered users are allowed. ▪ [1] Yes = Calls from unregistered users are blocked. **Notes:** ▪ When the device blocks a call, it sends a SIP 500 "Server Internal Error" response to the remote end. ▪ The parameter applies to calls belonging to a User-type IP Group. ▪ When the corresponding parameter in the SIP Interface table (SIPInterface_BlockUnRegUsers) is configured to Yes or No for a SIP Interface that is associated with the SRD, the parameter in the SRD table is ignored for calls belonging to the SIP Interface.
Enable Un-Authenticated Registrations `enable-un-auth-registrs` [SRD_EnableUnAuthenticatedRegistrations]	Enables the device to accept REGISTER requests and register them in its registration database from new users that have not been authenticated by a proxy/registrar server (due to proxy down) and thus, re-routed to a User-type IP Group. In normal operation scenarios in which the proxy server is available, the device forwards the REGISTER request to the proxy and if authenticated by the proxy (i.e., device receives a success response), the device adds the user to its registration database. The routing to the proxy is according to the SBC IP-to-IP Routing table where the destination is the proxy's IP Group. However, when the proxy is unavailable (e.g., due to network connectivity loss), the device can accept REGISTER requests from new users if a matching alternative routing rule exists in the SBC IP-to-IP Routing table where the destination is the user's User-type IP Group (i.e., call survivability scenarios) and if the parameter is enabled. ▪ [0] Disable = The device rejects REGISTER requests from new users that were not authenticated by a proxy server. ▪ [1] Enable = (Default) The device accepts REGISTER requests from new users even if they were not authenticated by a proxy server, and registers the user in its registration database. **Notes:**

Parameter	Description
	- Regardless of the parameter, the device always accepts registration refreshes from users that are already registered in its database. - For a SIP Interface that is associated with the SRD, if the corresponding parameter in the SIP Interface table (SIPInterface_EnableUnAuthenticatedRegistrations) is configured to Disable or Enable, the parameter in the SRD is ignored for calls belonging to the SIP Interface.
Used By Routing Server `used-by-routing-server` [SIPInterface_UsedByRoutingServer]	Enables the SRD to be used by a third-party routing server for call routing decisions. - [0] Not Used (default) - [1] Used For more information on the third-party routing server feature, see "Centralized Third-Party Routing Server or ARM" on page 272.

17.2.1 Filtering Tables in Web Interface by SRD

When your configuration includes multiple SRDs, you can filter tables in the Web interface by a specific SRD. The filter is configured in the SRD Filter drop-down list, located on the Web interface's toolbar. The filter is applied throughout the GUI. The following figure shows the SRD Filter with an example of configured SRDs in its' drop-down list.

Figure 17-8: SRD Filter



When you select an SRD for filtering, the Web interface displays only table rows associated with the filtered SRD. In addition, if you add a new row to a table, the filtered SRD is automatically selected as the associated SRD (in the 'SRD' parameter of the Add Row dialog box). For example, if your SRD filter is set to "Comp-A" and you then add a new Proxy Set, the Proxy Set is automatically associated with SRD "Comp-A" (i.e., the 'SRD' parameter is set to "Comp-A"). All other parameters in the Add Row dialog box are also automatically set to values associated with the filtered SRD.

SRD filtering is especially useful in multi-tenant setups where multiple SRDs may be configured. In such a setup, SRD filtering eliminates configuration clutter by "hiding" SRDs that are irrelevant to the current configuration and facilitates configuration by automatically associating the filtered SRD, and other configuration elements associated with the filtered SRD, wherever applicable.

17.2.2 Multiple SRDs for Multi-tenant Deployments

The device can be deployed in a multi-tenant architecture, serving multiple customers (tenants) from a single, shared physical entity. The device's multi-tenant feature is fully scalable, offering almost “non-bleeding” partition per tenant, whereby users of one tenant can’t infringe on the space of users of another tenant. The device provides per tenant configuration, monitoring, reporting, analytics, alarms and interfacing. The device is a real-time multi-tenant system that provides each tenant with optimal real-time performance, as each session received by the device is classified and processed only through the tenant's "orbit".

While some enterprises are large enough to justify a dedicated standalone device, many enterprises require only a fraction of the device's capacity and capabilities. Service providers offering SIP Trunking services can funnel multiple enterprises into a single device and thereby, reap significant cost improvements over a device-per-customer model. Tenant size in a multi-tenant architecture can vary and therefore, the instance CPU, memory and interface allocations should be optimized so as not to waste resources for small-sized tenants on the one hand, and not to allocate too many instances for a single tenant/customer on the other. For example, it would be a waste to allocate a capacity of 100 concurrent sessions to a small tenant for which 10 concurrent sessions suffice.

In a multi-tenant deployment, each tenant is represented by a dedicated SRD. The different Layer-3 networks (e.g., LAN IP-PBX users, WAN SIP Trunk, and WAN far-end users) of the tenant are represented by SIP Interfaces, which are all associated with the tenant's SRD. As related configuration entities (SIP Interfaces, IP Groups, Proxy Sets, Classification rules, and IP-to-IP Routing rules) are associated with the specific SRD, each SRD has its own logically separated configuration tables (although configured in the same tables). Therefore, full logical separation (on the SIP application layer) between tenants is achieved by SRD.

To create a multi-tenant configuration topology that is as non-bleeding as possible, you can configure an SRD (tenant) as *Isolated* and *Shared*:

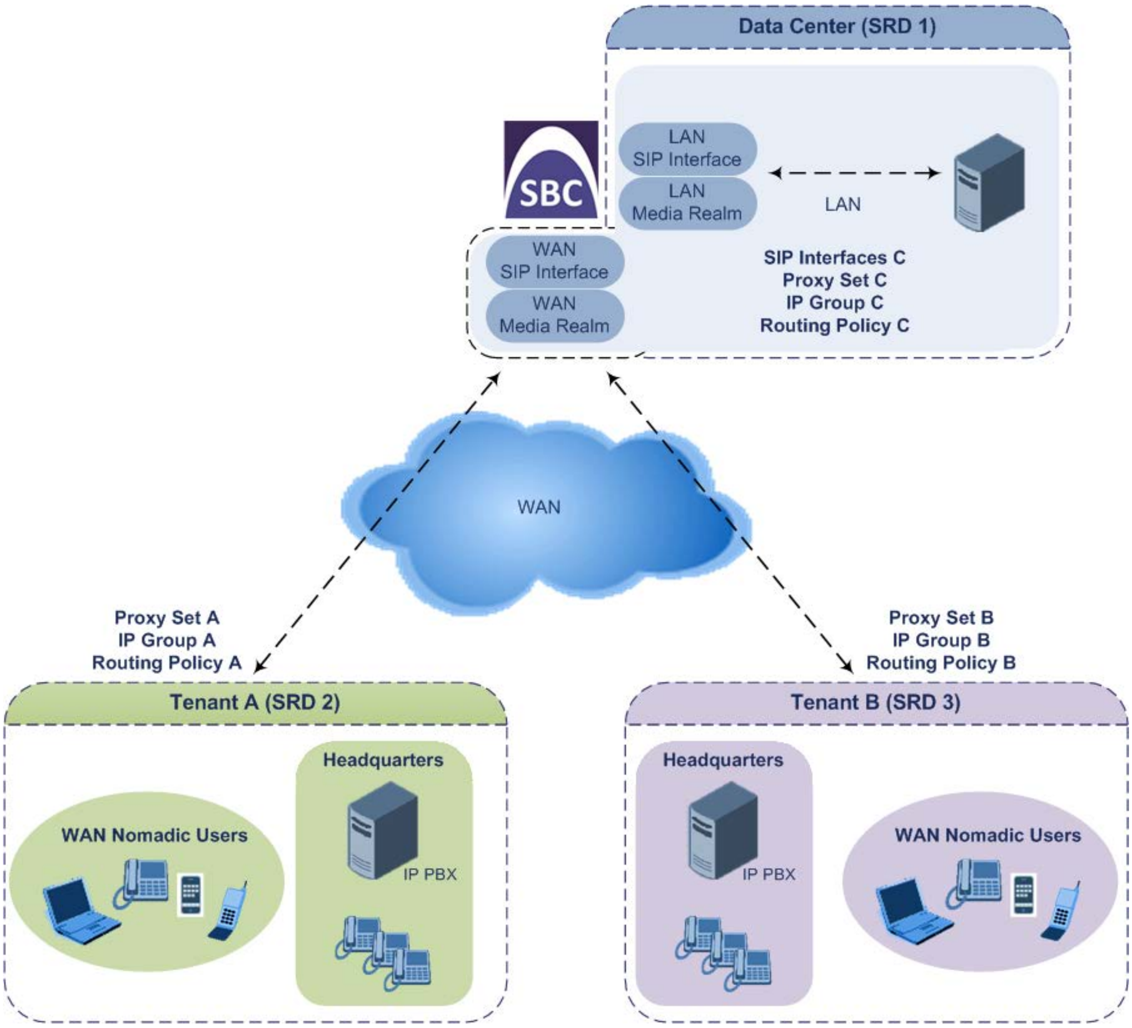
- **Isolated SRD:** An Isolated SRD has its own dedicated SIP resources (SIP Interfaces, Proxy Sets, and IP Groups). No other SRD can use the SIP resources of an Isolated SRD. Thus, call traffic of an Isolated SRD is kept separate from other SRDs (tenants), preventing any risk of traffic "leakage" with other SRDs.

  Isolated SRDs are more relevant when each tenant needs its own separate (dedicated) routing "table" for non-bleeding topology. Separate routing tables are implemented using Routing Policies. In such a non-bleeding topology, routing between Isolated SRDs is not possible. This enables accurate and precise routing per SRD, eliminating any possibility of erroneous call routing between SRDs, restricting routing to each tenant's (SRD's) sphere. Configuring only one Routing Policy that is shared between Isolated SRDs is not best practice for non-bleeding environments, since it allows routing between these SRDs.

- **Shared SRD:** Isolated SRDs have their own dedicated SIP resources (SIP Interfaces, Proxy Sets, and IP Groups). This may not be possible in some deployments. For example, in deployments where all tenants use the same SIP Trunking service, or use the same SIP Interface due to limited SIP interface resources (e.g., multiple IP addresses cannot be allocated and SIP port 5060 must be used). In contrast to Isolated SRDs, a Shared SRD can share its' SIP resources with all other SRDs (Shared and Isolated). This is typically required when tenants need to use common resources. In the SIP Trunk example, the SIP Trunk would be associated with a Shared SRD, enabling all tenants to route calls with the SIP Trunk.

Another configuration entity that can be used for multi-tenant deployments is the Routing Policy. Routing Policies allow each SRD (or tenant) to have its own routing rules, manipulation rules, Least Cost Routing (LCR) rules, and/or LDAP-based routing configuration. However, not all multi-tenant deployments need multiple Routing Policies and typically, their configuration is not required. Isolated SRDs are more relevant only when each tenant requires its own dedicated Routing Policy to create separate, dedicated routing "tables"; for all other scenarios, SRDs can be Shared. For more information on Routing Policies, see "Configuring SBC Routing Policy Rules" on page 590.

The figure below illustrates a multi-tenant architecture with Isolated SRD tenants ("A" and "B") and a Shared SRD tenant ("Data Center") serving as a SIP Trunk:



To facilitate multi-tenant configuration through CLI, you can access a specific tenant "view". Once in a specific tenant view, all configuration commands apply only to the currently viewed tenant. Only table rows (indexes) belonging to the viewed tenant can be modified. New table rows are automatically associated with the viewed tenant (i.e., SRD name). The display of tables and show running-configuration commands display only rows relevant to the viewed tenant (and shared tenants). The show commands display only information relevant to the viewed tenant. To support this CLI functionality, use the following commands:

- To access a specific tenant view:

  ```
  # srd-view <SRD name>
  ```

  Once accessed, the tenant's name (i.e., SRD name) forms part of the CLI prompt, for example:

  ```
  # srd-view datacenter
  (srd-datacenter)#
  ```

- To exit the tenant view:

  ```
  # no srd-view
  ```

17.2.3 Cloning SRDs

You can clone (duplicate) existing SRDs. This is especially useful when operating in a multi-tenant environment and you need to add new tenants (SRDs). The new tenants can quickly and easily be added by simply cloning one of the existing SRDs. Once cloned, all you need to do is tweak configuration entities associated with the SRD clone.

When an SRD is cloned, the device adds the new SRD clone to the next available index row in the SRD table. The SRD clone is assigned a unique name in the following syntax format: <unique ID>_<original SRD index>_CopyOf_<name or index if no name of original SRD>. For example, if you clone SRD "SIP-Trunk" at index 2, the new SRD clone is assigned the name, "36454371_2_CopyOf_SIP-Trunk".

The SRD clone has identical settings as the original SRD. In addition, all configuration entities associated with the original SRD are also cloned and these clones are associated with the SRD clone. The naming convention of these entities is the same as the SRD clones (see above) and all have the same unique clone ID ("36454371" in the example above) as the cloned SRD. These configuration entities include IP Groups, SIP Interfaces, Proxy Sets (without addresses), Classification rules, and Admission Control rules. If the Routing Policy associated with the original SRD is not associated with any other SRD, the Routing Policy is also cloned and its' clone is associated with the SRD clone. All configuration entities associated with the original Routing Policy are also cloned and these clones are associated with the Routing Policy clone. These configuration entities include IP-to-IP Routing rules, Inbound Manipulation rules, and Outbound Manipulation rules.

When any configuration entity is cloned (e.g., an IP-to-IP Routing rule) as a result of a cloned SRD, all fields of the entity's row which "point" to other entities (e.g., SIP Interface, Source IP Group, and Destination IP Group) are replaced by their corresponding clones.

> **Note:** For some cloned entities such as SIP Interfaces, some parameter values may change. This occurs in order to avoid the same parameter having the same value in more than one table row (index), which would result in invalid configuration. For example, a SIP Interface clone will have an empty Network Interface setting. After the clone process finishes, you thus need to update the Network Interface for valid configuration.

➢ **To clone an SRD:**

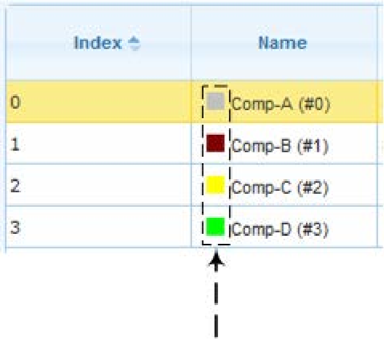
- Web interface: In the SRD table, select an SRD to clone, and then click the **Clone** button.
- CLI:

```
(config-voip)# voip-network srd clone <SRD index that you want cloned>
```

17.2.4 Color-Coding of SRDs in Web Interface

To easily identify your configured SRDs, the Web interface displays each SRD in a unique color. The color is automatically and randomly assigned to new SRDs, and is displayed in a box alongside the name of the SRD, in tables where the SRD is configured or assigned. This is applied throughout the Web interface's GUI. The following example shows SRDs assigned with unique color codes.

Figure 17-9: Color-Coding of SRDs



Index	Name
0	⬜ Comp-A (#0)
1	🟥 Comp-B (#1)
2	🟨 Comp-C (#2)
3	🟩 Comp-D (#3)

17.2.5 Automatic Configuration based on SRD

To facilitate configuration and eliminate possible flaws in configuration due to invalid associations between configuration entities, the Web interface automatically configures configuration entities based on SRD:

- If you delete an SRD (in the SRD table) that is associated with other configuration entities in other tables, the device automatically deletes the associated table rows. For example, if you delete an SRD that is associated with a Proxy Set, the device automatically deletes the Proxy Set.
- If you associate an SRD with a configuration entity in another table (i.e., other than the SRD table), the device automatically configures certain parameters of the configuration entity according to the SRD or associated SRD. For example, if you add a rule in the IP-to-IP Routing table and you select a Routing Policy, the 'Source IP Group' and 'Destination IP Group' parameters list only IP Groups that re associated with the SRD to which the Routing Policy is assigned (and IP Groups belonging to a Shared SRD, if exists).
- If your configuration setup includes only a single SRD, the device automatically selects the SRD when adding related configuration entities. For example, when adding an IP Group, the single SRD is automatically selected in the Add Row dialog box.

17.3 Configuring SIP Interfaces

The SIP Interface table lets you configure up to 82 SIP Interfaces. A SIP Interface represents a Layer-3 network in your deployment environment, by defining a local, listening port number and type (e.g., UDP), and assigning an IP network interface for SIP signaling traffic. For example, if your deployment consists of an IP PBX in the LAN, a SIP Trunk in the WAN, and remote far-end users in the WAN, you would need to configure a SIP Interface for each of these SIP entities. You can configure the SIP Interfaces for the different types of applications - SBC, Gateway, and SAS. Gateway SIP Interfaces are required for Tel-to-IP and IP-to-Tel calls. You can also configure various optional features for the SIP Interface such as assigning it a Media Realm, blocking calls received on the SIP Interface from users not registered with the device, and enabling direct media.

Each SIP Interface can be associated with only one SRD. As the SRD configuration entity represents your VoIP deployment SIP network (Layer 5), you need to associate your SIP Interfaces with a specific SRD in order to represent your Layer-3 networks. For most deployments (except multi-tenant deployments), your SRD represents your entire network and thus, only one SRD is required. The device provides a default SRD and in such scenarios where only a single SRD is required, your SIP Interfaces are automatically assigned to the default SRD. Therefore, there is no need to even handle SRD configuration entity.

Once configured, you can apply SIP Interfaces to calls, by assigning them to the following configuration entities in their respective tables:

- (Mandatory) Proxy Set to specify the SIP Interface for communication with the proxy server (i.e., IP Group). For more information, see "Configuring Proxy Sets" on page 352.
- IP-to-IP Routing rules for specifying the destination SIP Interface to where you want to route the call. For more information, see "Configuring SBC IP-to-IP Routing Rules" on page 578.
- Classification rules for specifying the SIP Interface as a matching characteristic of the incoming call. This is especially useful for the single SRD-configuration topology, where each SIP Interface represents a Layer-3 network (SIP entity). Therefore, classification of calls to IP Groups (SIP entities) can be based on SIP Interface. For more information, see "Configuring Classification Rules" on page 569.
- Admission Control rules to apply call admission control per SIP Interface. For more information, see "Configuring Admission Control" on page 561.
- Tel-to-IP Routing rules for specifying the destination SIP Interface to where you want to route Tel-to-IP calls. For more information, see Configuring Tel-to-IP Routing Rules on page 467.
- IP-to-Trunk Group Routing rules for specifying the SIP Interface as a matching characteristics for the incoming IP call.
- Intrusion Detection System (IDS) for applying the IDS policy to a specific SIP Interface. For more information, see "Configuring IDS Policies" on page 164.

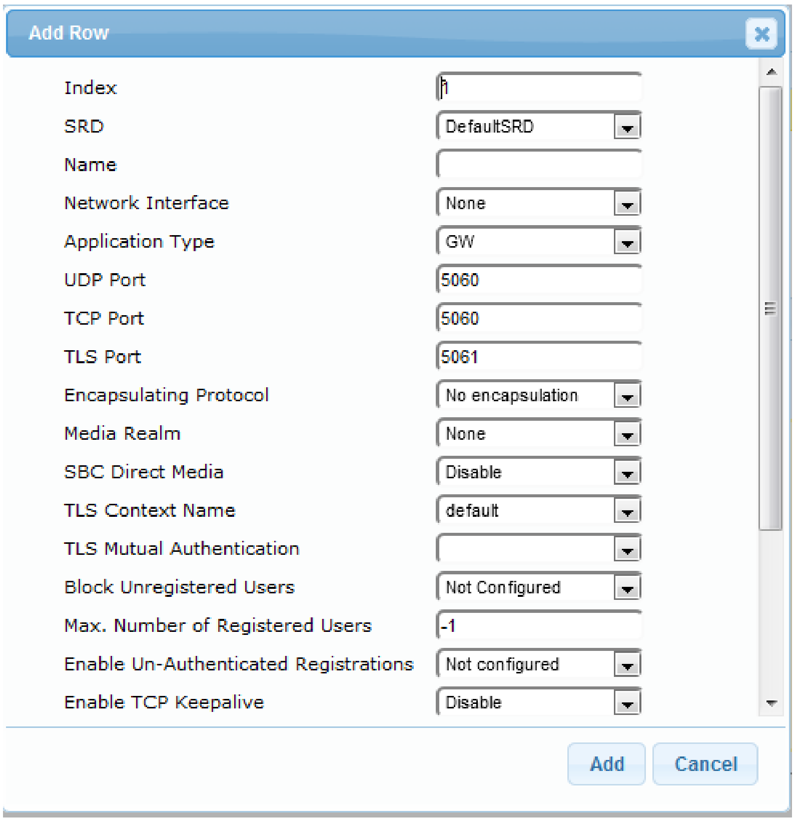
Note: The device terminates active calls associated with a SIP Interface in the following scenarios:

- If you delete the associated SIP Interface.
- If you edit any of the following fields of the associated SIP Interface: 'Application Type', 'UDP Port, 'TCP Port', 'TLS Port' or 'SRD' fields.
- If you edit or delete a network interface in the Interface table that is associated with the SIP Interface.

The following procedure describes how to configure SIP interfaces through the Web interface. You can also configure it through ini file (SIPInterface) or CLI (configure voip > voip-network sip-interface).

➢ To configure a SIP Interface:

1. Open the SIP Interface table (**Configuration** tab > **VoIP** menu > **VoIP Network** > **SIP Interface Table**).
2. Click **Add**; the following dialog box appears:



Add Row

Index	
SRD	DefaultSRD
Name	
Network Interface	None
Application Type	GW
UDP Port	5060
TCP Port	5060
TLS Port	5061
Encapsulating Protocol	No encapsulation
Media Realm	None
SBC Direct Media	Disable
TLS Context Name	default
TLS Mutual Authentication	
Block Unregistered Users	Not Configured
Max. Number of Registered Users	-1
Enable Un-Authenticated Registrations	Not configured
Enable TCP Keepalive	Disable

3. Configure a SIP Interface according to the parameters described in the table below.
4. Click **Add**.

Table 17-5: SIP Interface Table Parameter Descriptions

Parameter	Description
Index [SIPInterface_Index]	Defines an index for the new table row. **Note:** Each row must be configured with a unique index.
SRD `srd` [SIPInterface_SRDName]	Assigns an SRD to the SIP Interface. If only one SRD is configured in the SRD table, the SRD is assigned to the SIP Interface by default. If multiple SRDs are configured in the SRD table, no value is defined. For configuring SRDs, see "Configuring SRDs" on page 325. **Notes:** ▪ The parameter is mandatory. ▪ You can assign the same SRD to multiple SIP Interfaces (SBC and Gateway). ▪ For the SAS application, use only SRD 0.
Name `interface-name` [SIPInterface_InterfaceName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. By default, if you do not configure a name, the device automatically assigns the name "SIPInterface_<row index>" (e.g., "SIPInterface_1" when added to Index 1).
Network Interface `network-interface` [SIPInterface_NetworkInterface]	Assigns a Control-type IP network interface to the SIP Interface. By default, no value is defined (**None**). For configuring network interfaces, see "Configuring IP Network Interfaces" on page 133.

Parameter	Description
	Note: The parameter is mandatory.
Application Type `application-type` [SIPInterface_ApplicationType]	Defines the application for which the SIP Interface is used. ▪ [0] GW = (Default) Gateway application. ▪ [1] SAS = Stand-Alone Survivability (SAS) application. ▪ [2] SBC = SBC application.
UDP Port `udp-port` [SIPInterface_UDPPort]	Defines the device's listening and source port for SIP signaling traffic over UDP. The valid range is 1 to 65534. The default is 5060. **Notes:** ▪ The port **must** be different from ports configured for RTP traffic (i.e., ports configured for Media Realms). For example, if the RTP port range is 6000 to 6999, the SIP port can either be less than 6000 or greater than 6999. ▪ The base UDP port number (BaseUDPPort parameter) for RTP traffic must be greater than the highest UDP port configured for a SIP Interface. For example, if your highest configured UDP port for a SIP Interface is 6060, you must configure the BaseUDPPort parameter to any value greater than 6060. For more information on base UDP port, see Configuring RTP Base UDP Port on page 192. ▪ Each SIP Interface must have a unique signaling port (i.e., no two SIP Interfaces can share the same port - no port overlapping).
TCP Port `tcp-port` [SIPInterface_TCPPort]	Defines the device's listening port for SIP signaling traffic over TCP. The valid range is 1 to 65534. The default is 5060. **Notes:** ▪ The port **must** be different from ports configured for RTP traffic (i.e., ports configured for Media Realms). For example, if the RTP port range is 6000 to 6999, the SIP port can either be less than 6000 or greater than 6999. ▪ Each SIP Interface must have a unique signaling port (i.e., no two SIP Interfaces can share the same port - no port overlapping).
TLS Port `tls-port` [SIPInterface_TLSPort]	Defines the device's listening port for SIP signaling traffic over TLS. The valid range is 1 to 65534. The default is 5061. **Notes:** ▪ The port **must** be different from ports configured for RTP traffic (i.e., ports configured for Media Realms). For example, if the RTP port range is 6000 to 6999, the SIP port can either be less than 6000 or greater than 6999. ▪ Each SIP Interface must have a unique signaling port (i.e., no two SIP Interfaces can share the same port - no port overlapping).
Media Realm `media-realm-name` [SIPInterface_MediaRealm]	Assigns a Media Realm to the SIP Interface. By default, no value is defined (**None**). For configuring Media Realms, see "Configuring Media Realms" on page 317. **Note:** If you later delete the assigned Media Realm in the Media Realm table, this value becomes **None**.

Parameter	Description
SBC Direct Media `intra-srd-media-anchoring` [SIPInterface_SBCDirectMedia]	Enables direct media (RTP/SRTP) flow (i.e., no Media Anchoring) between endpoints associated with the SIP Interface. - [0] Disable = (Default) Media Anchoring is employed, whereby the media stream traverses the device (and each leg uses a different coder or coder parameters). - [1] Enable = No Media Anchoring. Media stream flows directly between endpoints (i.e., does not traverse the device - no Media Anchoring). - [2] Enable when Same NAT = No Media Anchoring. Media stream flows directly between endpoints if they are located behind the same NAT. **Notes:** - If the parameter is enabled for direct media and the two endpoints belong to the same SIP Interface, calls cannot be established if the following scenario exists: **a.** One of the endpoints is defined as a foreign user (for example, “follow me service”) **b.** and one endpoint is located on the WAN and the other on the LAN. The reason for the above is that in direct media, the device does not interfere in the SIP signaling such as manipulation of IP addresses, which is necessary for calls between LAN and WAN. - To enable direct media for all calls, use the global parameter SBCDirectMedia. If enabled, even if the SIP Interface is disabled for direct media, direct media is employed for calls belonging to the SIP Interface. - If you enable direct media for the SIP Interface, make sure that your Media Realm provides sufficient ports, as media may traverse the device for mid-call services (e.g., call transfer). - For more information on direct media, see Direct Media on page 538.
TLS Context Name `tls-context-name` [SIPInterface_TLSContext]	Assigns a TLS Context (SSL/TLS certificate) to the SIP Interface. The default TLS Context ("default" at Index 0) is assigned to the SIP Interface by default. **Notes:** - For incoming calls: The assigned TLS Context is used if no TLS Context is configured for the Proxy Set associated with the call or classification to an IP Group based on Proxy Set fails. - For outgoing calls: The assigned TLS Context is used if no TLS Context is configured for the Proxy Set associated with the call. For more information on TLS Contexts, see "Configuring SSL/TLS Certificates" on page 107.
TLS Mutual Authentication `tls-mutual-auth` [SIPInterface_TLSMutualAuthentication]	Enables TLS mutual authentication for the SIP Interface (when the device acts as a server). - **[-1]** Not Configured = (Default) The SIPSRequireClientCertificate global parameter setting is applied. - **[0]** Disable = Device does not request the client certificate for TLS connection on the SIP Interface. - **[1]** Enable = Device requires receipt and verification of the client certificate to establish the TLS connection on the SIP Interface.

Parameter	Description
Block Unregistered Users `block-un-reg-users` [SIPInterface_BlockUnRegUsers]	Enables the device to block (reject) incoming calls (INVITE requests) from unregistered users belonging to the SIP Interface. ▪ [-1] Not Configured = (Default) The corresponding parameter in the SRD table (SRD_BlockUnRegUsers) of the SRD that is associated with the SIP Interface is applied. ▪ [0] No = Calls from unregistered users are allowed. ▪ [1] Yes = Calls from unregistered users are blocked. **Notes:** ▪ When the device blocks a call, it sends a SIP 500 "Server Internal Error" response to the remote end. ▪ The parameter applies to calls belonging to a User-type IP Group. ▪ If configured to Yes or No, the parameter overrides the 'Block Unregistered Users' parameter of the associated SRD in the SRD table.
Max. Number of Registered Users `max-reg-users` [SIPInterface_MaxNumOfRegUsers]	Defines the maximum number of users belonging to the SIP Interface that can register with the device. By default, no value is defined (i.e., the number of allowed user registrations is unlimited).
Enable Un-Authenticated Registrations `enable-un-auth-registrs` [SIPInterface_EnableUnAuthenticatedRegistrations]	Enables the device to accept REGISTER requests and register them in its registration database from new users that have not been authenticated by a proxy/registrar server (due to proxy down) and thus, re-routed to a User-type IP Group. In normal operation scenarios in which the proxy server is available, the device forwards the REGISTER request to the proxy and if authenticated by the proxy (i.e., device receives a success response), the device adds the user to its registration database. The routing to the proxy is according to the SBC IP-to-IP Routing table where the destination is the proxy's IP Group. However, when the proxy is unavailable (e.g., due to network connectivity loss), the device can accept REGISTER requests from new users if a matching alternative routing rule exists in the SBC IP-to-IP Routing table where the destination is the user's User-type IP Group (i.e., call survivability scenarios) and if the parameter is enabled. ▪ **[-1]** Not Configured = (Default) The corresponding parameter in the SRD table (SRD_EnableUnAuthenticatedRegistrations) of the SRD associated with the SIP Interface is applied. ▪ **[0]** Disable = The device rejects REGISTER requests from new users that were not authenticated by a proxy server. ▪ **[1]** Enable = The device accepts REGISTER requests from new users even if they were not authenticated by a proxy server, and registers the user in its registration database. **Notes:** ▪ Regardless of the parameter, the device always accepts registration refreshes from users that are already registered in its database. ▪ If configured to Disable or Enable, the parameter overrides the 'Enable Un-Authenticated Registrations' parameter settings of the SRD (in the SRD table) that is associated with the SIP Interface.

Parameter	Description
Enable TCP Keepalive `tcp-keepalive-enable` [SIPInterface_TCPKeepaliveEnable]	Enables the TCP Keep-Alive mechanism with the IP entity on this SIP Interface. TCP keep-alive can be used, for example, to keep a NAT entry open for clients located behind a NAT server, or simply to check that the connection to the IP entity is available. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** For configuring TCP keepalive, use the following ini file parameters: TCPKeepAliveTime, TCPKeepAliveInterval, and TCPKeepAliveRetry.
Classification Failure Response Type `classification_fail_response_type` [SIPInterface_ClassificationFailureResponseType]	Defines the SIP response code that the device sends if a received SIP request (OPTIONS, REGISTER, or INVITE) fails the SBC Classification process. The valid value can be a SIP response code from 400 through 699, or it can be set to 0 to not send any response at all. The default response code is 500 (Server Internal Error). This feature is important for preventing Denial of Service (DoS) attacks, typically initiated from the WAN. Malicious attackers can use SIP scanners to detect ports used by SIP devices. These scanners scan devices by sending UDP packets containing a SIP request to a range of specified IP addresses, listing those that return a valid SIP response. Once the scanner finds a device that supports SIP, it extracts information from the response and identifies the type of device (IP address and name) and can execute DoS attacks. A way to defend the device against such attacks is to not send a SIP reject response to these unclassified "calls" so that the attacker assumes that no device exists at such an IP address and port. **Note:** The parameter is applicable only if the device is set to reject unclassified calls. This is configured using the 'Unclassified Calls' parameter on the General Settings page (Configuration tab > VoIP menu > SBC > General Settings).
Pre Classification Manipulation Set ID `preclassification-manset` [SIPInterface_PreClassificationManipulationSet]	Assigns a Message Manipulation Set ID to the SIP Interface. This lets you apply SIP message manipulation rules on incoming SIP initiating-dialog request messages (not in-dialog), received on this SIP Interface, prior to the Classification process. By default, no Message Manipulation Set ID is defined. For configuring Message Manipulation rules, see Configuring SIP Message Manipulation on page 370. **Notes:** ▪ The Message Manipulation Set assigned to a SIP Interface that is associated with an outgoing call, is ignored. Only the Message Manipulation Set assigned to the associated IP Group is applied to the outgoing call. ▪ If both the SIP Interface and IP Group associated with the incoming call are assigned a Message Manipulation Set, the one assigned to the SIP Interface is applied first. ▪ The parameter is applicable only to SBC calls.
Message Policy `message-policy` [SIPInterface_MessagePolicyName]	Assigns a SIP message policy to the SIP interface. For configuring SIP Message Policy rules, see "Configuring SIP Message Policy Rules".

Parameter	Description
Used By Routing Server `used-by-routing-server` [SIPInterface_UsedByRoutingServer]	Enables the SIP Interface to be used by a third-party routing server for call routing decisions. ▪ [0] Not Used (default) ▪ [1] Used For more information on the third-party routing server feature, see Centralized Third-Party Routing Server or ARM on page 272.

17.4 Configuring IP Groups

The IP Group table lets you configure up to 102 IP Groups. An IP Group represents a SIP entity in the network with which the device communicates. This can be a server (e.g., IP PBX or ITSP) or a group of users (e.g., LAN IP phones). For servers, the IP Group is typically used to define the server's IP address by associating it with a Proxy Set (see Configuring Proxy Sets on page 352).

You can use IP Groups for the following:

- SBC application: Classification of incoming SIP dialog-initiating requests (e.g., INVITE messages) to IP Groups based on Proxy Set. If the source address of the incoming SIP dialog is defined for a Proxy Set, the device assigns ("bonds") the SIP dialog to the IP Group associated with the Proxy Set. The feature is configured using the IP Group table's 'Classify by Proxy Set' parameter. For more information and recommended security guidelines, see the parameter's description, later in this section.
- SBC application: IP Groups are used for configuring IP-to-IP routing rules where they represent the source and destination of the call (see Configuring SBC IP-to-IP Routing Rules on page 578).
- SIP dialog registration and authentication (digest user/password) of specific IP Groups (Served IP Group, e.g., corporate IP-PBX) with other IP Groups (Serving IP Group, e.g., ITSP). This is configured in the Account table (see "Configuring Registration Accounts" on page 363).
- Gateway application: Call routing rules:
  - Tel-to-IP calls: The IP Group is used as the destination of the outgoing IP call and is used in Tel-to-IP call routing rules (see Configuring Tel-to-IP Routing Rules on page 467).
  - IP-to-Tel calls: The IP Group identifies the source of the IP call and is used in IP-to-Tel call routing rules (see Configuring IP-to-Trunk Group Routing Rules on page 476).
  - Number manipulation: The IP Group can be associated with a number manipulation rule (see Configuring Number Manipulation Tables on page 441).
- Included in routing decisions by a third-party routing server. If deemed necessary for routing, the routing server can even create an IP Group. For more information, see Centralized Third-Party Routing Server or ARM on page 272.

You can also apply the device's Quality of Experience feature to IP Groups:

- **Quality of Experience Profile:** Call quality monitoring based on thresholds for voice metrics (e.g., MOS) can be applied per IP Group. For example, if MOS is considered poor, calls belonging to this IP Group can be rejected. For configuring Quality of Experience Profiles, see "Configuring Quality of Experience Profiles" on page 307.
- **Bandwidth Profile:** Bandwidth utilization thresholds can be applied per IP Group. For example, if bandwidth thresholds are crossed, the device can reject any new calls on this IP Group. For configuring Bandwidth Profiles, see "Configuring Bandwidth Profiles" on page 311.

Notes:

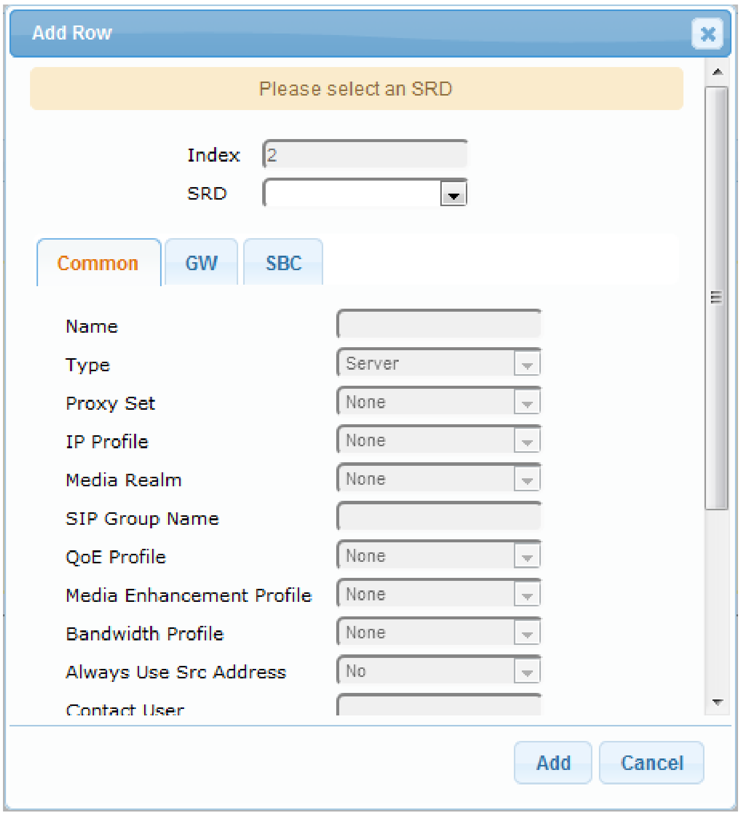
- For the Gateway application: IP Group ID 0 cannot be associated with Proxy Set ID 0.
- If you delete an IP Group or modify the 'Type' or 'SRD' parameters, the device immediately terminates currently active calls associated with the IP Group. In addition, all users belonging to this IP Group are removed from the device's users database.

The following procedure describes how to configure IP Groups through the Web interface. You can also configure it through ini file (IPGroup) or CLI (configure voip > control-network ip-group).

➢ **To configure an IP Group:**

1. Open the IP Group table (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).
2. Click **Add**; the following dialog box appears:

Figure 17-10: IP Group Table - Add Row Dialog Box



Add Row

Please select an SRD

Index	2
SRD	

Common | GW | SBC

Name	
Type	Server
Proxy Set	None
IP Profile	None
Media Realm	None
SIP Group Name	
QoE Profile	None
Media Enhancement Profile	None
Bandwidth Profile	None
Always Use Src Address	No
Contact User	

3. Configure an IP Group according to to the parameters described in the table below.
4. Click **Add**.

Table 17-6: IP Group Table Parameter Descriptions

Parameter	Description
Common Parameters	
Index [IPGroup_Index]	Defines an index for the new table row. **Note:** Each row must be configured with a unique index.
SRD `srd-name` [IPGroup_SRDName]	Assigns an SRD to the IP Group. If only one SRD is configured in the SRD table, the SRD is assigned by default. If multiple SRDs are configured in the SRD table, no value is assigned by default. For configuring SRDs, see Configuring SRDs on page 325. **Notes:** ▪ The parameter is mandatory. ▪ For the parameter to take effect, a device reset is required.
Name `name` [IPGroup_Name]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. **Note:** Each row must be configured with a unique name.
Type `type` [IPGroup_Type]	Defines the type of IP Group: ▪ [0] Server = Applicable when the destination address of the IP Group (e.g., ITSP, Proxy, IP-PBX, or Application server) is known. The address is configured by the Proxy Set that is associated with the IP Group. ▪ [1] User = Represents a group of users such as IP phones and softphones where their location is dynamically obtained by the device when REGISTER requests and responses traverse (or are terminated) by the device. These users are considered remote (far-end). Typically, this IP Group is configured with a Serving IP Group that represents an IP-PBX, Application or Proxy server that serves this User-type IP Group. Each SIP request sent by a user of this IP Group is proxied to the Serving IP Group. For registrations, the device updates its registration database with the AOR and contacts of the users. Digest authentication using SIP 401/407 responses (if needed) is performed by the Serving IP Group. The device forwards these responses directly to the SIP users. To route a call to a registered user, a rule must be configured in the Tel-to-IP Routing table or SBC IP-to-IP Routing table. The device searches the dynamic database (by using the Request-URI) for an entry that matches a registered AOR or Contact. Once an entry is found, the IP destination is obtained from this entry and a SIP request is sent to the destination. The device also supports NAT traversal for the SIP clients located behind NAT. In this case, the device must be defined with a global IP address. ▪ [2] Gateway = (Applicable only to the SBC application.) In scenarios where the device receives requests to and from a gateway representing multiple users. This IP Group type is necessary for any of the following scenarios: ✓ The IP Group cannot be defined as a Server-type since its address is initially unknown and therefore, a Proxy Set cannot be configured for it.

Parameter	Description
	✓ The IP Group cannot be defined as a User since the SIP Contact header of the incoming REGISTER does not represent a specific user. The Request-URI user part can change and therefore, the device is unable to identify an already registered user and therefore, adds an additional record to the database. The IP address of the Gateway IP Group is obtained dynamically from the host part of the Contact header in the REGISTER request received from the IP Group (i.e., IP Group is registered with the device). Therefore, routing to this IP Group is possible only once a REGISTER request is received. If a REGISTER refresh request arrives, the device updates the new location (i.e., IP address) of the IP Group. If the REGISTER fails, no update is performed. If an UN-REGISTER request arrives, the IP address associated with the IP Group is deleted and therefore, no routing to the IP Group is done. You can view the registration status of the Gateway-type IP Group in the 'GW Group Registered Status' field, and view the IP address of the IP Group in the 'GW Group Registered IP Address' field if it is registered with the device.
Proxy Set `proxy-set-id` [IPGroup_ProxySetName]	Assigns a Proxy Set to the IP Group. All INVITE messages destined to the IP Group are sent to the IP address configured for the Proxy Set. For configuring Proxy Sets, see "Configuring Proxy Sets" on page 352. **Notes:** ▪ For the Gateway application: IP Group ID 0 cannot be associated with Proxy Set ID 0. ▪ The Proxy Set must be associated with the same SRD as that assigned to the IP Group. ▪ You can assign the same Proxy Set to multiple IP Groups. ▪ For the SBC application: Proxy Sets are used for Server-type IP Groups, but may in certain scenarios also be used for User-type IP Groups. For example, this is required in deployments where the device mediates between an IP PBX and a SIP Trunk, and the SIP Trunk requires SIP registration for each user that requires service. In such a scenario, the device must register all the users to the SIP Trunk on behalf of the IP PBX. This is done by using the User Info table where each user is associated with the source IP Group (i.e., the IP PBX). For configuring the User Info table, see SBC User Information for SBC User Database on page 664. ▪ For the Gateway application: Proxy Sets are applicable only to Sever-type IP Groups.
IP Profile `ip-profile-name` [IPGroup_ProfileName]	Assigns an IP Profile to the IP Group. By default, no value is defined (**None**). For configuring IP Profiles, see "Configuring IP Profiles" on page 387.
Media Realm Name `media-realm-name` [IPGroup_MediaRealm]	Assigns a Media Realm to the IP Group. The Media Realm determines the UDP port range and maximum sessions on a specific interface for media traffic associated with the IP Group. By default, no value is defined (None).

Parameter	Description
	For configuring Media Realms, see Configuring Media Realms on page 317. **Notes:** - For the parameter to take effect, a device reset is required. - If you delete a Media Realm from the Media Realm table that is assigned to the IP Group, the parameter value reverts to None.
SIP Group Name `sip-group-name` [IPGroup_SIPGroupName]	Defines the SIP Request-URI host name in INVITE and REGISTER messages sent to this IP Group, or the host name in the From header of INVITE messages received from this IP Group. In other words, it replaces the original host name. The valid value is a string of up to 100 characters. By default, no value is defined. **Notes:** - If the parameter is not configured, the value of the global parameter, ProxyName is used instead (see "Configuring Proxy and Registration Parameters" on page 367). - The parameter overrides inbound message manipulation rules that manipulate the host name in Request-URI, To, and/or From SIP headers. If you configure the parameter and you want to manipulate the host name in any of these SIP headers, you must apply your manipulation rule (Manipulation Set ID) to the IP Group as an Outbound Message Manipulation Set (see the IPGroup_OutboundManSet parameter), when the IP Group is the destination of the call. If you apply the Manipulation Set as an Inbound Message Manipulation Set (see the IPGroup_InboundManSet parameter), when the IP Group is the source of the call, the manipulation rule is overridden by the SIP Group Name parameter. - If the IP Group is of User type, the parameter is used internally as a host name in the Request-URI for Tel-to-IP initiated calls. For example, if an incoming call from the device's trunk is routed to a User-type IP Group, the device first creates the Request-URI (<destination_number>@<SIP Group Name>), and then it searches the registration database for a match.
UUI Format uui-format **[IPGroup_UUIFormat]**	Enables the generation of the Avaya UCID value, adding it to the outgoing INVITE sent to this IP Group. - [0] Disabled (default) - [1] Enabled This provides support for interworking with Avaya equipment by generating Avaya's UCID value in outgoing INVITE messages sent to Avaya's network. The device adds the UCID in the User-to-User SIP header. Avaya's UCID value has the following format (in hexadecimal): 00 + FA + 08 + node ID (2 bytes) + sequence number (2 bytes) + timestamp (4 bytes) This is interworked in to the SIP header as follows: `User-to-User: 00FA080019001038F725B3;encoding=hex`

Parameter	Description
	Note: To define the Network Node Identifier of the device for Avaya UCID, use the 'Network Node ID' (NetworkNodeId) parameter.
QoE Profile `qoe-profile` [IPGroup_QOEProfile]	Assigns a Quality of Experience Profile rule. By default, no value is defined (**None**). For configuring Quality of Experience Profiles, see "Configuring Quality of Experience Profiles" on page 307.
Media Enhancement Profile `media-enhancement-profile` [IPGroup_MediaEnhancementProfile]	Assigns a Media Enhancement Profile rule. By default, no value is defined (**None**). For configuring Media Enhancement Profiles, see "Configuring Media Enhancement Profiles" on page 313.
Bandwidth Profile `bandwidth-profile` [IPGroup_BWProfile]	Assigns a Bandwidth Profile rule. By default, no value is defined (**None**). For configuring Bandwidth Profiles, see "Configuring Bandwidth Profiles" on page 311.
Always Use Src Address `always-use-source-addr` [IPGroup_AlwaysUseSourceAddr]	Enables the device to always send SIP requests and responses, within a SIP dialog, to the source IP address received in the previous SIP message packet. This feature is especially useful in scenarios where the IP Group endpoints are located behind a NAT firewall (and the device is unable to identify this using its regular NAT mechanism). ▪ **[0]** No = (Default) The device sends SIP requests according to the settings of the global parameter, SIPNatDetection. ▪ **[1]** Yes = The device sends SIP requests and responses to the source IP address received in the previous SIP message packet. For more information on NAT traversal, see "Remote UA behind NAT" on page 153.
Contact User `contact-user` [IPGroup_ContactUser]	Defines the user part of the From, To, and Contact headers of SIP REGISTER messages, and the user part of the Contact header of INVITE messages received from this IP Group and forwarded by the device to another IP Group. By default, no value is defined. **Notes:** ▪ The parameter is applicable only to Server-type IP Groups. ▪ The parameter is overridden by the 'Contact User' parameter in the Account table (see "Configuring Registration Accounts" on page 363).
Local Host Name `local-host-name` [IPGroup_ContactName]	Defines the host name (string) that the device uses in the SIP message's Via and Contact headers. This is typically used to define an FQDN as the host name. The device uses this string for Via and Contact headers in outgoing INVITE messages sent to a specific IP Group, and the Contact header in SIP 18x and 200 OK responses for incoming INVITE messages received from a specific IP Group. The Inbound IP Routing table can be used to identify the source IP Group from where the INVITE message was received.

Parameter	Description
	If the parameter is not configured, these headers are populated with the device's dotted-decimal IP address of the network interface on which the message is sent. By default, no value is defined. **Note:** To ensure proper device handling, the parameter should be a valid FQDN.
Used By Routing Server `used-by-routing-server` [IPGroup_UsedByRoutingServer]	Enables the IP Group to be used by a third-party routing server for call routing decisions. ▪ [0] Not Used (default) ▪ [1] Used For more information on the third-party routing server feature, see Centralized Third-Party Routing Server or ARM on page 272.
Created By Routing Server [IPGroup_CreatedByRoutingServer]	(Read-only) Indicates whether the IP Group was created by a third-party routing server: ▪ [0] No ▪ [1] Yes For more information on the third-party routing server feature, see Centralized Third-Party Routing Server or ARM on page 272.
GW Tab (Gateway Application)	
SIP Re-Routing Mode `re-routing-mode` [IPGroup_SIPReRoutingMode]	Defines the routing mode after a call redirection (i.e., a 3xx SIP response is received) or transfer (i.e., a SIP REFER request is received). ▪ [-1] Not Configured (Default) ▪ [0] Standard = INVITE messages that are generated as a result of Transfer or Redirect are sent directly to the URI, according to the Refer-To header in the REFER message or Contact header in the 3xx response. ▪ [1] Proxy = Sends a new INVITE to the Proxy. This is applicable only if a Proxy server is used and the parameter AlwaysSendtoProxy is set to 0. ▪ [2] Routing Table = Uses the Routing table to locate the destination and then sends a new INVITE to this destination. **Notes:** ▪ When the parameter is set to [1] and the INVITE sent to the Proxy fails, the device re-routes the call according to the Standard mode [0]. ▪ When the parameter is set to [2] and the INVITE fails, the device re-routes the call according to the Standard mode [0]. If DNS resolution fails, the device attempts to route the call to the Proxy. If routing to the Proxy also fails, the Redirect / Transfer request is rejected. ▪ When the parameter is set to [2], the XferPrefix parameter can be used to define different routing rules for redirected calls. ▪ The parameter is ignored if the parameter AlwaysSendToProxy is set to 1.
Always Use Route Table `always-use-route-table` [IPGroup_AlwaysUseRouteTable]	Defines the Request-URI host name in outgoing INVITE messages. ▪ [0] No (default). ▪ [1] Yes = The device uses the IP address (or domain name) defined in the Tel-to-IP Routing table (see Configuring the Tel

Parameter	Description
	to IP Routing on page 467) as the Request-URI host name in outgoing INVITE messages, instead of the value configured in the 'SIP Group Name' field. **Note:** The parameter is applicable only to Server-type IP Groups.
SBC Tab (SBC Application)	
SBC Operation Mode `sbc-operation-mode` [IPGroup_SBCOperationMode]	Defines the device's operational mode for the IP Group. ▪ [-1] Not Configured = (Default) ▪ [0] B2BUA = Device operates as a back-to-back user agent (B2BUA), changing the call identifiers and headers between the inbound and outbound legs. ▪ [1] Call Stateful Proxy = Device operates as a Stateful Proxy, passing the SIP message transparently between inbound and outbound legs. In other words, the same SIP dialog identifiers (tags, Call-Id and CSeq) occur on both legs (as long as no other configuration disrupts the CSeq compatibleness). For more information on B2BUA and Stateful Proxy modes, see B2BUA and Stateful Proxy Operating Modes on page 528. **Note:** If configured, the parameter overrides the 'SBC Operation Mode' parameter in the SRD table.
Classify By Proxy Set `classify-by-proxy-set` [IPGroup_ClassifyByProxySet]	Enables classification of incoming SIP dialogs (INVITEs) to Server-type IP Groups based on Proxy Set (assigned using the IPGroup_ProxySetName parameter). ▪ [0] Disable ▪ [1] Enable = (Default) The device searches the Proxy Set table for a Proxy Set that is configured with the same source IP address as that of the incoming INVITE (if host name, then according to the dynamically resolved IP address list). If such a Proxy Set is found, the device classifies the INVITE as belonging to the IP Group associated with the Proxy Set. **Note:** ▪ The parameter is applicable only to Server-type IP Groups. ▪ For security, it is recommended to classify SIP dialogs based on Proxy Set only if the IP address of the IP Group is unknown. In other words, if the Proxy Set associated with the IP Group is configured with an FQDN. In such cases, the device classifies incoming SIP dialogs to the IP Group based on the DNS-resolved IP address. If the IP address is known, it is recommended to use a Classification rule instead (and disable the Classify by Proxy Set feature), where the rule is configured with not only the IP address, but also with SIP message characteristics to increase the strictness of the classification process (see Configuring Classification Rules on page 569). The reason for preferring classification based on Proxy Set when the IP address is unknown is that IP address forgery (commonly known as IP spoofing) is more difficult than malicious SIP message tampering and therefore, using a Classification rule without an IP address offers a weaker form of security. When classification is based on Proxy Set, the Classification table for the specific IP Group is ignored. ▪ If you have assigned the same Proxy Set to multiple IP Groups, disable the parameter and instead, use Classification

Parameter	Description
	rules to classify incoming SIP dialogs to these IP Groups. If the parameter is enabled, the device is unable to correctly classify incoming INVITEs to their appropriate IP Groups. - Classification by Proxy Set occurs only if classification based on the device's registration database fails (i.e., the INVITE is not from a registered user).
SBC Client Forking Mode `enable-sbc-client-forking` [IPGroup_EnableSBCClientForking]	Defines call forking of INVITE messages to up to five separate SIP outgoing legs for User-type IP Groups. This occurs if multiple contacts are registered under the same AOR in the device's registration database. - [0] Sequential = (Default) Sequentially sends the INVITE to each contact. If there is no answer from the first contact, it sends the INVITE to the second contact, and so on until a contact answers. If no contact answers, the call fails or is routed to an alternative destination, if configured. - [1] Parallel = Sends the INVITE simultaneously to all contacts. The call is established with the first contact that answers. - [2] Sequential Available Only = Sequentially sends the INVITE only to available contacts (i.e., not busy). If there is no answer from the first available contact, it sends the INVITE to the second contact, and so on until a contact answers. If no contact answers, the call fails or is routed to an alternative destination, if configured. **Note:** The device can also fork INVITE messages received for a Request-URI of a specific contact (user) registered in the database to all other users located under the same AOR as the specific contact. This is configured using the SBCSendInviteToAllContacts parameter.
Inbound Message Manipulation Set `inbound-mesg-manipulation-set` [IPGroup_InboundManSet]	Assigns a Message Manipulation Set (rule) to the IP Group for SIP message manipulation on the inbound leg. For configuring Message Manipulation rules, see Configuring SIP Message Manipulation on page 370. **Note:** The IPGroup_SIPGroupName parameter overrides inbound message manipulation rules (assigned to the IPGroup_InboundManSet parameter) that manipulate the host name in Request-URI, To, and/or From SIP headers. If you want to manipulate the host name using message manipulation rules in any of these SIP headers, you must apply your manipulation rule (Manipulation Set ID) to the IP Group as an Outbound Message Manipulation Set (see the IPGroup_OutboundManSet parameter), when the IP Group is the destination of the call.
Outbound Message Manipulation Set `outbound-mesg-manipulation-set` [IPGroup_OutboundManSet]	Assigns a Message Manipulation Set (rule) to the IP Group for SIP message manipulation on the outbound leg. For configuring Message Manipulation rules, see Configuring SIP Message Manipulation on page 370. **Note:** If you assign a Message Manipulation Set ID that includes rules for manipulating the host name in the Request-URI, To, and/or From SIP headers, the parameter overrides the IPGroup_SIPGroupName parameter.

Parameter	Description
Msg Man User Defined String1 `msg-man-user-defined-string1` [IPGroup_MsgManUserDef1]	Defines a value for the SIP user part that can be used in Message Manipulation rules configured in the Message Manipulations table. The Message Manipulation rule obtains this value from the IP Group, by using the following syntax: param.ipg.<src|dst>.user-defined.<0>. The valid value is a string of up to 30 characters. By default, no value is defined. For configuring Message Manipulation rules, see Configuring SIP Message Manipulation on page 370.
Msg Man User Defined String2 `msg-man-user-defined-string2` [IPGroup_MsgManUserDef2]IPGroup_MsgManUserDef2]	Defines a value for the SIP user part that can be used in Message Manipulation rules configured in the Message Manipulations table. The Message Manipulation rule obtains this value from the IP Group, by using the following syntax: param.ipg.<src|dst>.user-defined.<1>. The valid value is a string of up to 30 characters. By default, no value is defined. For configuring Message Manipulation rules, see Configuring SIP Message Manipulation on page 370.
Registration Mode `registration-mode` [IPGroup_RegistrationMode]	Defines the registration mode for the IP Group: ▪ [0] User Initiates Registration (default) ▪ [1] SBC Initiates Registration = Used when the device serves as a client (e.g., with an IP PBX). This functions only with the User Info file. ▪ [2] Registrations not Needed = The device adds users to its database in active state.
Max. Number of Registered Users `max-num-of-reg-users` [IPGroup_MaxNumOfRegUsers]	Defines the maximum number of users in this IP Group that can register with the device. The default is -1, meaning that no limitation exists for registered users. **Note:** The parameter is applicable only to User-type IP Groups.
Authentication Mode `authentication-mode` [IPGroup_AuthenticationMode]	Defines the authentication mode. ▪ [0] User Authenticates = (Default) The device does not handle the authentication, but simply forwards the authentication messages between the SIP user agents. ▪ [1] SBC as Client = The device authenticates as a client. It receives the 401/407 response from the proxy requesting for authentication. The device sends the proxy the authorization credentials (i.e., username and password) according to one of the following: 1)Account configured in the Account table (only if authenticating Server-type IP Group), 2) global username and password parameters (only if authenticating Server-type IP Group), 3) User Information file, or 4) sends request to users requesting credentials (only if authenticating User-type IP Group). For more information on Accounts, see Configuring Registration Accounts on page 363. ▪ [2] SBC as Server = The device acts as an Authentication server: ✓ Authenticates SIP clients, using the usernames and passwords in the User Information table (see SBC User Information for SBC User Database on page 664). This is applicable only to User-type IP Groups.

Parameter	Description
	✓ Authenticates SIP severs. This is applicable only to Server-type IP Groups.
Authentication Method List `authentication-method-list` [IPGroup_MethodList]	Defines SIP methods received from the IP Group that must be challenged by the device when the device acts as an Authentication server. If no methods are configured, the device doesn't challenge any methods. By default, no value is defined. To define multiple SIP methods, use the backslash (\ ) to separate each method (e.g., INVITE\REGISTER). **Note:** The parameter is applicable only if the 'Authentication Mode' parameter is set to SBC as Server [2].
Username `username` [IPGroup_Username]	Defines the shared username for authenticating the IP Group, when the device acts as an Authentication server. The valid value is a string of up to 51 characters. By default, no username is defined. **Notes:** ▪ The parameter is applicable only to Server-type IP Groups and when the 'Authentication Mode' parameter is set to SBC as Server (i.e., authentication of servers). ▪ To specify the SIP request types (e.g., INVITE) that must be challenged by the device, use the 'Authentication Method List' parameter.
Password `password` IPGroup_Password]	Defines the shared password for authenticating the IP Group, when the device acts as an Authentication server. The valid value is a string of up to 51 characters. By default, no password is defined. **Notes:** ▪ The parameter is applicable only to Server-type IP Groups and when the 'Authentication Mode' parameter is set to SBC as Server (i.e., authentication of servers). ▪ To specify the SIP request types (e.g., INVITE) that must be challenged by the device, use the 'Authentication Method List' parameter.
Source URI Input `src-uri-input` [IPGroup_SourceUriInput]	Defines the SIP header in the incoming INVITE that is used for call matching characteristics based on source URIs. ▪ [-1] Not Configured (default) ▪ [0] From ▪ [1] To ▪ [2] Request-URI ▪ [3] P-Asserted - First Header ▪ [4] P-Asserted - Second Header ▪ [5] P-Preferred ▪ [6] Route ▪ [7] Diversion ▪ [8] P-Associated-URI ▪ [9] P-Called-Party-ID ▪ [10] Contact ▪ [11] Referred-by **Notes:**

Parameter	Description
	▪ The parameter is applicable only when classification is done according to the Classification table. ▪ If the configured SIP header does not exist in the incoming INVITE message, the classification of the message to a source IP Group fails. ▪ If the device receives an INVITE as a result of a REFER request or a 3xx response, then the incoming INVITE is routed according to the Request-URI. The device identifies such INVITEs according to a specific prefix in the Request-URI header, configured by the SBCXferPrefix parameter. Therefore, in this scenario, the device ignores the parameter setting.
Destination URI Input `dst-uri-input` [IPGroup_DestUriInput]	Defines the SIP header in the incoming INVITE to use as a call matching characteristic based on destination URIs. The parameter is used for classification and routing purposes. The device first uses the parameter's settings as a matching characteristic (input) to classify the incoming INVITE to an IP Group (source IP Group) in the Classification table. Once classified, the device uses the parameter for routing the call. For example, if set to To, the URI in the To header of the incoming INVITE is used as a matching characteristic for classifying the call to an IP Group in the Classification table. Once classified, the device uses the URI in the To header as the destination. ▪ [-1] Not Configured (default) ▪ [0] From ▪ [1] To ▪ [2] Request-URI ▪ [3] P-Asserted - First Header ▪ [4] P-Asserted - Second Header ▪ [5] P-Preferred ▪ [6] Route ▪ [7] Diversion ▪ [8] P-Associated-URI ▪ [9] P-Called-Party-ID ▪ [10] Contact ▪ [11] Referred-by **Notes:** ▪ The parameter is applicable only when classification is done according to the Classification table. ▪ If the configured SIP header does not exist in the incoming INVITE message, the classification of the message to a source IP Group fails. ▪ If the device receives an INVITE as a result of a REFER request or a 3xx response, the incoming INVITE is routed according to the Request-URI. The device identifies such INVITEs according to a specific prefix in the Request-URI header, configured by the SBCXferPrefix parameter. Therefore, in this scenario, the device ignores the parameter setting.

Parameter	Description
SIP Connect `sip-connect` [IPGroup_SIPConnect]	Defines the IP Group as a registered server that represents multiple users. The device saves registrations received from the IP Group, with the IP address as a key in its registration database. The device classifies incoming SIP dialog requests (e.g., INVITEs) from the IP Group according to the received IP address. For requests routed to the IP Group users, the device replaces the Request-URI header with the incoming To header (which contains the remote phone number). ▪ [0] No (default) ▪ [1] Yes **Note:** The parameter is applicable only to User-type IP Groups.
SBC PSAP Mode `sbc-psap-mode` [IPGroup_SBCPSAPMode]	Enables E9-1-1 emergency call routing in a Microsoft Lync Server environment. ▪ [0] Disable (default) ▪ [1] Enable For more information, see Enhanced 9-1-1 Support for Lync Server on page 289.
DTLS Context `dtls-context` [IPGroup_DTLSContext]	Assigns a TLS Context (certificate) to the IP Group, which is used for DTLS sessions (handshakes) with the IP Group. By default, no value is defined (None). For configuring TLS Contexts, see Configuring TLS Certificate Contexts on page 107.
Route Using Request URI Port `use-requri-port` [IPGroup_SBCRouteUsingRequestURIPort]	Enables the device to use the port indicated in the Request-URI of the incoming message as the destination port when routing the message to the IP Group. The device uses the IP address (and not port) that is configured for the Proxy Set associated with the IP Group. The parameter thus allows the device to route calls to the same server (IP Group), but different port. ▪ [0] Disable = (Default) The port configured for the associated Proxy Set is used as the destination port. ▪ [1] Enable = The port indicated in the Request-URI of the incoming message is used as the destination port.
GW Group Registered IP Address	(Read-only field) Displays the IP address of the IP Group entity (gateway) if registered with the device; otherwise, the field is blank. **Note:** The field is applicable only to Gateway-type IP Groups (i.e., the 'Type' parameter is configured to Gateway).
GW Group Registered Status	(Read-only field) Displays whether the IP Group entity (gateway) is registered with the device ("Registered" or "Not Registered"). **Note:** The field is applicable only to Gateway-type IP Groups (i.e., the 'Type' parameter is configured to Gateway).

17.5 Configuring Proxy Sets

The Proxy Sets table lets you configure up to 102 Proxy Sets. A Proxy Set defines the address and transport type (e.g., UDP or TCP) of a SIP server (e.g., SIP proxy and SIP registrar server). The Proxy Set represents the destination (address) of the IP Group configuration entity. Each Proxy Set can be configured with up to 10 addresses configured as an IP address and/or DNS host name (FQDN), enabling you to implement load balancing and redundancy (Proxy Hot-Swap feature) between multiple servers. If you configure the address as an FQDN, you can configure the method (A-record DNS, SRV, or NAPTR) for resolving the domain name to an IP address. The device supports up to 30 DNS-resolved IP addresses. (If the DNS resolution provides more than this number, the device uses the first 30 IP addresses in the received list and ignores the rest.) Each Proxy Set can be assigned a specific SSL/TLS certificate the (TLS Context), enabling you to use different TLS certificates per SIP entity (IP Group). In addition, each Proxy Set must be assigned a SIP Interface (and SRD), which determines, amongst others, the device's local network interface through which communication with the Proxy Set is done.

You can enable the device's keep-alive feature per Proxy Set, which determines whether proxies (addresses) configured for the Proxy Set are online or offline. If offline, the device will not route the call to the specific proxy. You can configure the device to send either SIP OPTIONS or REGISTER messages for the keep-alive. The keep-alive feature is required when using the proxy load-balancing or redundancy feature. For load-balancing, the device performs keep-alive on all proxies. For Parking-type redundancy, the device performs keep-alive only on the currently active proxy. For Homing-type redundancy, the device performs keep-alive on the current proxy as well as the "main" proxy. When using SIP OPTIONS, you can configure the device to consider the proxy as offline if specific SIP response codes are received from the keep-alive messages. To ensure that a previously offline proxy is now online, you can configure the number of required consecutive successful keep-alive messages (SIP OPTIONS only) before the device considers the proxy as being online. This mechanism avoids the scenario in which the device falsely detects a proxy as being online when it is actually offline, resulting in call routing failure. To view the connectivity status of Proxy Sets, see Viewing Proxy Set Status on page 728.

You can also enable the device to classify incoming SBC SIP dialogs to IP Groups, based on Proxy Set. If the source address of the incoming SIP dialog is the same as the address of a Proxy Set, the device classifies the SIP dialog as belonging to the IP Group that is associated with the Proxy Set.

To use a configured Proxy Set, you need to assign it to an IP Group in the IP Group table (see "Configuring IP Groups" on page 340). When the device sends INVITE messages to an IP Group, it sends it to the address configured for the Proxy Set. You can assign the same Proxy Set to multiple IP Groups (belonging to the same SRD).

Notes:

- It is recommended to classify incoming SIP dialogs to IP Groups, based on the Classification table (see Configuring Classification Rules on page 569) instead of based on Proxy Set.
- You can view the device's connectivity status with proxy servers in the Tel-to-IP Routing table, for Tel-to-IP routing rules whose destination is an IP Group that is associated with a Proxy Set. The status is only displayed for Proxy Sets enabled with the Proxy Keep-Alive feature.

The Proxy Set is configured using two tables, one a "child" of the other:

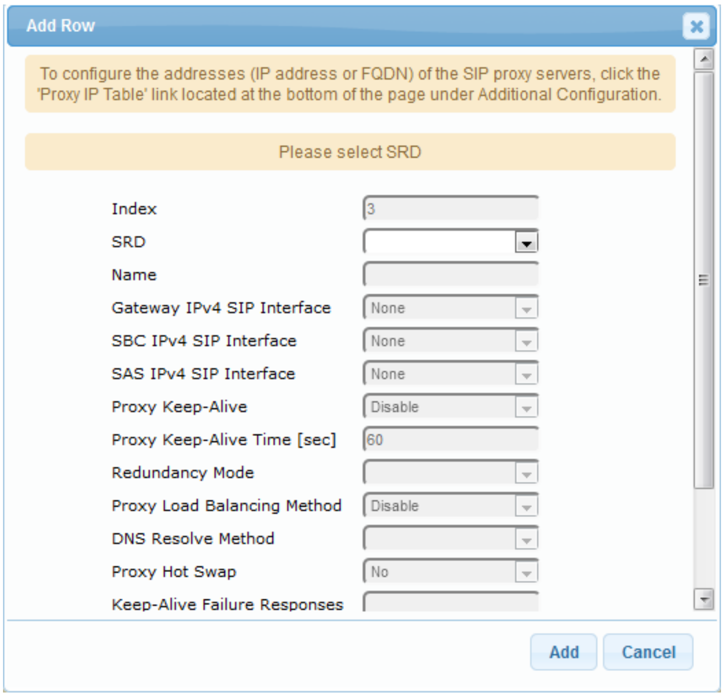
- Proxy Sets table: Defines the attributes of the Proxy Set such as associated SIP Interface and redundancy features - ini file parameter, ProxySet or CLI command, configure voip > voip-network proxy-set

- Proxy Set Address table ("child"): Defines the addresses of the Proxy Set - table ini file parameter, ProxyIP or CLI command, configure voip > voip-network proxy-ip > proxy-set-id

➢ To configure a Proxy Set:

1. Open the Proxy Sets table (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table**).
2. Click **Add**; the following dialog box appears:

Figure 17-11: Proxy Sets Table - Add Row Dialog Box



Add Row

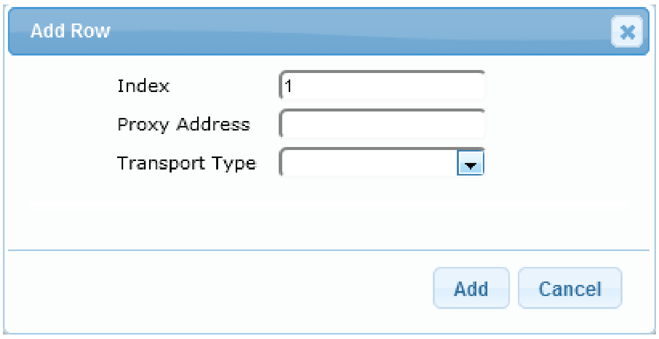
To configure the addresses (IP address or FQDN) of the SIP proxy servers, click the 'Proxy IP Table' link located at the bottom of the page under Additional Configuration.

Please select SRD

Index	3
SRD	
Name	
Gateway IPv4 SIP Interface	None
SBC IPv4 SIP Interface	None
SAS IPv4 SIP Interface	None
Proxy Keep-Alive	Disable
Proxy Keep-Alive Time [sec]	60
Redundancy Mode	
Proxy Load Balancing Method	Disable
DNS Resolve Method	
Proxy Hot Swap	No
Keep-Alive Failure Responses	

3. Configure a Proxy Set according to the parameters described in the table below.
4. Click **Add**.
5. Select the index row of the Proxy Set that you added, and then click the **Proxy Address Table** link located below the table; the Proxy Address table opens.
6. Click **Add**; the following dialog box appears:

Figure 17-12: Proxy Address Table - Add Row Dialog Box



Add Row

Index	1
Proxy Address	
Transport Type	

7. Configure the address of the Proxy Set according to the parameters described in the table below.
8. Click **Add**.

Table 17-7: Proxy Sets Table and Proxy Address Table Parameter Description

Parameter	Description
Proxy Sets Table	
Index `configure voip > voip-network proxy-set` [ProxySet_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
SRD `voip-network proxy-set > srd-id` [ProxySet_SRDName]	Assigns an SRD to the Proxy Set. **Notes:** ▪ The parameter is mandatory and must be configured first before you can configure the other parameters in the table. ▪ To configure SRDs, see Configuring SRDs on page 325.
Name `proxy-name` [ProxySet_ProxyName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. **Note:** ▪ Each row must be configured with a unique name. ▪ The value cannot include a "/" forward slash.
Gateway IPv4 SIP Interface `gwipv4-sip-int-name` [ProxySet_GWIPv4SIPInterfaceName]	Assigns an IPv4-based SIP Interface for Gateway calls to the Proxy Set. **Notes:** ▪ At least one SIP Interface must be assigned to the Proxy Set. ▪ The parameter appears only if you have configured a network interface with an IPv4 address in the Interface table (see Configuring IP Network Interfaces on page 133). ▪ To configure SIP Interfaces, see Configuring SIP Interfaces on page 333.
SBC IPv4 SIP Interface `sbcipv4-sip-int-name` [ProxySet_SBCIPv4SIPInterfaceName]	Assigns an IPv4-based SIP Interface for SBC calls to the Proxy Set. **Notes:** ▪ At least one SIP Interface must be assigned to the Proxy Set. ▪ The parameter appears only if you have configured a network interface with an IPv4 address in the Interface table (see Configuring IP Network Interfaces on page 133). ▪ To configure SIP Interfaces, see "Configuring SIP Interfaces" on page 333.
SAS IPv4 SIP Interface `sasipv4-sip-int-name` [ProxySet_SASIPv4SIPInterfaceName]	Assigns an IPv4-based SIP Interface for SAS calls to the Proxy Set. **Notes:** ▪ At least one SIP Interface must be assigned to the Proxy Set. ▪ The parameter appears only if you have configured a network interface with an IPv4 address in the Interface table.

Parameter	Description
Gateway IPv6 SIP Interface `gwipv6-sip-int-name` [ProxySet_GWIPv6SIPInterfaceName]	Assigns an IPv6-based SIP Interface for Gateway calls to the Proxy Set. **Notes:** ▪ At least one SIP Interface must be assigned to the Proxy Set. ▪ The parameter appears only if you have configured a network interface with an IPv6 address in the Interface table.
SBC IPv6 SIP Interface `sbcipv6-sip-int-name` [ProxySet_SBCIPv6SIPInterfaceName]	Assigns an IPv6-based SIP Interface for SBC calls to the Proxy Set. **Notes:** ▪ At least one SIP Interface must be assigned to the Proxy Set. ▪ The parameter appears only if you have configured a network interface with an IPv6 address in the Interface table.
SAS IPv6 SIP Interface `sasipv6-sip-int-name` [ProxySet_SASIPv6SIPInterfaceName]	Assigns an IPv6-based SIP Interface for SAS calls to the Proxy Set. **Notes:** ▪ At least one SIP Interface must be assigned to the Proxy Set. ▪ The parameter appears only if you have configured a network interface with an IPv6 address in the Interface table.
Proxy Keep-Alive `proxy-enable-keep-alive` [ProxySet_EnableProxyKeepAlive]	Enables the device's Proxy Keep-Alive feature, which checks communication with the proxy server. ▪ **[0]** Disable (default). ▪ **[1]** Using OPTIONS = Enables the Proxy Keep-Alive feature using SIP OPTIONS messages. The device sends an OPTIONS message every user-defined interval, configured by the 'Proxy Keep-Alive Time' parameter (in this table). If the device receives a SIP response code that is configured in the 'Keep-Alive Failure Responses' parameter (in this table), the device considers the proxy as down. You can also configure whether to use the device's IP address or string name ("gateway name") in the OPTIONS message (see the UseGatewayNameForOptions parameter). ▪ **[2]** Using REGISTER = Enables the Proxy Keep-Alive feature using SIP REGISTER messages. The device sends a REGISTER message every user-defined interval, configured by the RegistrationTime parameter (Gateway application) or SBCProxyRegistrationTime parameter (SBC application). Any SIP response from the proxy - success (200 OK) or failure (4xx response) - is considered as if the proxy is "alive". If the proxy does not respond to INVITE messages sent by the device, the proxy is considered as down (offline). **Notes:**

Parameter	Description
	▪ Proxy keep-alive using REGISTER messages (**Using REGISTER** option) is applicable only to the Parking redundancy mode ('Redundancy Mode' parameter configured to **Parking**). ▪ For Survivability mode for User-type IP Groups, you must enable this Proxy Keep-Alive feature. ▪ If you enable this Proxy Keep-Alive feature and the proxy uses the TCP/TLS transport type, you can enable CRLF Keep-Alive feature, using the UsePingPongKeepAlive parameter. ▪ If you enable this Proxy Keep-Alive feature, the device can operate with multiple proxy servers (addresses) for redundancy and load balancing (see the 'Proxy Load Balancing Method' parameter).
Proxy Keep-Alive Time `proxy-keep-alive-time` [ProxySet_ProxyKeepAliveTime]	Defines the interval (in seconds) between keep-alive messages sent by the device when the Proxy Keep-Alive feature is enabled (see the 'Proxy Keep-Alive' parameter in this table). The valid range is 5 to 2,000,000. The default is 60. **Note:** The parameter is applicable only if the 'Proxy Keep-Alive' parameter is set to **Using Options**.
Success Detection Retries success-detect-retries [ProxySet_SuccessDetectionRetries]	Defines the minimum number of consecutive, successful keep-alive messages that the device sends to an offline proxy, before the device considers the proxy as being online. The valid range is 1 to 10. The default is 1. Note: The parameter is applicable only if the 'Proxy Keep-Alive' parameter is set to Using Options.
Success Detection Interval success-detect-int [ProxySet_SuccessDetectionInterval]	Defines the interval (in seconds) between each keep-alive retries (as configured by the 'Success Detection Retries' parameter) that the device performs for offline proxies. The valid range is 1 to 30. The default is 10. Note: The parameter is applicable only if the 'Proxy Keep-Alive' parameter is set to Using Options.
Failure Detection Retransmissions fail-detect-rtx [ProxySet_FailureDetectionRetransmissions]	Defines the maximum number of UDP retransmissions that the device sends to an offline proxy, before the device considers the proxy as being offline. The valid range is -1 to 255. The default is -1 (i.e., the settings of the global parameter SIPMaxRtxis applied). Note: The parameter is applicable only if the 'Proxy Keep-Alive' parameter is set to Using Options.
Redundancy Mode `proxy-redundancy-mode` [ProxySet_ProxyRedundancyMode]	Determines whether the device switches from a redundant proxy to the primary proxy when the primary proxy becomes available again. ▪ **[-1]** Not configured = (Default) Proxy redundancy method is according to the settings of the global parameter, ProxyRedundancyMode.

Parameter	Description
	▪ **[0]** Parking = The device continues operating with the redundant (now active) proxy even if the primary proxy returns to service. If the redundant proxy subsequently becomes unavailable, the device operates with the next configured redundant proxy. ▪ **[1]** Homing = The device always attempts to operate with the primary proxy. The device switches back to the primary proxy whenever it becomes available. **Notes:** ▪ To enable this functionality, you must also enable the Proxy Keep-Alive feature (see the 'Proxy Keep-Alive' parameter in this table). ▪ The **Homing** option can only be used if the 'Proxy Keep-Alive' parameter is set to **Using Options.**
Proxy Load Balancing Method `proxy-load-balancing-method` [ProxySet_ProxyLoadBalancingMethod]	Enables load balancing between proxy servers of the Proxy Set. ▪ **[0]** Disable = (Default) Disables proxy load balancing. ▪ **[1]** Round Robin = A list of all possible proxy IP addresses is compiled. This list includes all IP addresses of the Proxy Set after DNS resolutions (including NAPTR and SRV, if configured). After this list is compiled, the Proxy Keep-Alive feature (enabled by the 'Proxy Keep-Alive' and 'Proxy Keep-Alive Time' parameters in this table) tags each entry as "offline" or "online". Load balancing is only performed on proxy servers that are tagged as "online". All outgoing messages are equally distributed across the list of IP addresses. REGISTER messages are also distributed unless a RegistrarIP is configured. The IP address list is refreshed every user-defined interval (see the ProxyIPListRefreshTime parameter). If a change in the order of the IP address entries in the list occurs, all load statistics are erased and balancing starts over again. ▪ **[2]** Random Weights = The outgoing requests are not distributed equally among the Proxies. The weights are received from the DNS server, using SRV records. The device sends the requests in such a fashion that each proxy receives a percentage of the requests according to its' assigned weight. A single FQDN should be configured as a proxy IP address. Random Weights Load Balancing is not used in the following scenarios: ✓ More than one IP address has been configured for the Proxy Set. ✓ The proxy address is not configured as an FQDN (only IP address). ✓ SRV is disabled (see the DNSQueryType parameter).

Parameter	Description
	✓ The SRV response includes several records with a different Priority value.
Min. Active Servers for Load Balancing min-active-serv-lb [ProxySet_MinActiveServersLB]	Defines the minimum number of proxies in the Proxy Set that must be online for the device to consider the Proxy Set as online, when proxy load balancing is used. The valid value is 1 to 15. The default is 1. **Note:** The parameter is applicable only if proxy load balancing is enabled (see the 'Proxy Load Balancing Method' parameter, above).
DNS Resolve Method `dns-resolve-method` [ProxySet_DNSResolveMethod]	Defines the DNS query record type for resolving the proxy server's host name (FQDN) into an IP address(es). ▪ **[-1]** = DNS resolution method is according to the settings of the global parameter, ProxyDNSQueryType. ▪ **[0]** A-Record = (Default) DNS A-record query is used to resolve DNS to IP addresses. ▪ **[1]** SRV = If the proxy address is configured with a domain name without a port (e.g., domain.com), an SRV query is done. The SRV query returns the host names (and their weights). The device then performs DNS A-record queries per host name (according to the received weights). If the configured proxy address contains a domain name with a port (e.g., domain.com:5080), the device performs a regular DNS A-record query. ▪ **[2]** NAPTR = NAPTR query is done. If successful, an SRV query is sent according to the information received in the NAPTR response. If the NAPTR query fails, an SRV query is done according to the configured transport type. If the configured proxy address contains a domain name with a port (e.g., domain.com:5080), the device performs a regular DNS A-record query. If the transport type is configured for the proxy address, a NAPTR query is not performed. ▪ **[3]** MS-Lync = SRV query as required by Microsoft when the device is deployed in a Microsoft Lync environment. The device sends a special SRV query to the DNS server according to the transport protocol configured in the 'Transport Type' parameter (described later in this section): ✓ TLS: "_sipinternaltls_tcp.<domain>" and "_sip_tls.<domain>". For example, if the configured domain name (in the 'Proxy Address' parameter) is "ms-server.com", the device queries for "_sipinternaltls_tcp.ms-server.com" and "_sip_tls.ms-server.com". ✓ TCP: "_sipinternal._tcp.<domain>" and "_sip_tcp.<domain>". ✓ Undefined: "_sipinternaltls_tcp.<domain>", "_sipinternal_tcp.<domain>",

Parameter	Description
	"_sip_tls.<domain>" and "_sip_tcp.<domain>". The SRV query returns the host names (and their weights). The device then performs DNS A-record queries per host name (according to the received weights) to resolve into IP addresses. **Note:** An SRV query can return up to four host names. For each host name, the subsequent DNS A-record query can resolve into up to 15 IP addresses. However, the device supports up to 30 DNS-resolved IP addresses. If the device receives more than this number of IP addresses, it uses the first 30 IP addresses in the received list and ignores the rest.
Proxy Hot Swap `is-proxy-hot-swap` [ProxySet_IsProxyHotSwap]	Enables the Proxy Hot-Swap feature, whereby the device switches to a redundant proxy upon a failure in the primary proxy (no response is received). ▪ **[0]** No = (Default) Disables the Proxy Hot-Swap feature. If a failure occurs in te primary proxy, the device does not connect with any other address (proxy) configured for the Proxy Set. ▪ **[1]** Yes = The device sends SIP INVITE/REGISTER messages to the first address listed in the Proxy Address table that is configured for the Proxy Set. If a SIP response is received and this response code is configured in the Alternative Routing Reasons table (see Configuring SIP Response Codes for Alternative Routing Reasons on page 588) for SBC, or in the Reasons for Tel-to-IP Alternative Routing table (see Alternative Routing Based on SIP Responses on page 588) for Gateway, the device assumes that the proxy is down and sends the message to the next available proxy (address) in the list.
Keep-Alive Failure Responses `keepalive-fail-resp` [ProxySet_KeepAliveFailureResp]	Defines SIP response codes that if any is received in response to a keep-alive message using SIP OPTIONS, the device considers the proxy as down. Up to three response codes can be configured, where each code is separated by a comma (e.g., 407,404). By default, no response code is defined. If no response code is configured, or if response codes received are not those configured, the proxy is considered "alive". **Note:** The SIP 200 response code is not supported for this feature.
Classification Input `classification-input` [ProxySet_ClassificationInput]	Defines how the device classifies incoming IP calls to the Proxy Set. ▪ [0] IP Only = (Default) Classifies calls to the Proxy Set according to IP address only. ▪ [1] IP + Port + Transport = Classifies calls to the Proxy Set according to IP address, port, and transport type. **Notes:**

Parameter	Description
	▪ The parameter is applicable only if the IP Group table's parameter, 'Classify by Proxy Set' is set to Enable (see Configuring IP Groups on page 340). ▪ The parameter is applicable only to the SBC application.
TLS Context Index `tls-context-index` [ProxySet_TLSContextName]	Assigns a TLS Context (SSL/TLS certificate) to the Proxy Set. By default, no TLS Context is assigned. If you assign a TLS Context, the TLS Context is used as follows: ▪ **Incoming calls:** If the 'Transport Type' parameter (in this table) is set to **TLS** and the incoming call is successfully classified to an IP Group based on the Proxy Set, this TLS Context is used. If the 'Transport Type' parameter is set to **UDP** or classification to this Proxy Set fails, the TLS Context is not used. Instead, the device uses the TLS Context configured for the SIP Interface (see "Configuring SIP Interfaces" on page 333) used for the call; otherwise, the default TLS Context (ID 0) is used. ▪ **Outgoing calls:** If the 'Transport Type' parameter is set to **TLS** and the outgoing call is sent to an IP Group that is associated with this Proxy Set, this TLS Context is used. Instead, the device uses the TLS Context configured for the SIP Interface used for the call; otherwise, the default TLS Context (ID 0) is used. If the 'Transport Type' parameter is set to **UDP**, the device uses UDP to communicate with the proxy and no TLS Context is used. For configuring TLS Contexts, see "Configuring TLS Certificate Contexts" on page 107.
Proxy Address Table `configure voip > voip-network proxy-ip > proxy-set-id`	
Index `proxy-ip-index` [ProxyIp_ProxyIpIndex]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Proxy Address `proxy-address` [ProxyIp_IpAddress]	Defines the address of the proxy. Up to 10 addresses can be configured per Proxy Set. The address can be defined as an IP address in dotted-decimal notation (e.g., 201.10.8.1) or FQDN. You can also specify the port using the following format: ▪ IPv4 address: <IP address>:<port> (e.g., 201.10.8.1:5060) ▪ IPv6 address: <[IPV6 address]>:<port> (e.g., [2000::1:200:200:86:14]:5060) **Note:** For the SBC application: You can configure the device to use the port indicated in the Request-URI of the incoming message, instead of the port configured for the parameter. To enable this, use the IPGroup_SBCRouteUsingRequestURIPort

Parameter	Description
	parameter for the IP Group that is associated with the Proxy Set (Configuring IP Groups on page 340).
Transport Type `transport-type` [ProxyIp_TransportType]	Defines the transport type for communicating with the proxy. ▪ **[0]** UDP ▪ **[1]** TCP ▪ **[2]** TLS ▪ **[-1]** = (Default) The transport type is according to the settings of the global parameter, SIPTransportType.

18 SIP Definitions

This section describes configuration of various SIP-related functionalities.

18.1 Configuring SIP Parameters

Many of the stand-alone SIP parameters associated with various features can be configured in the following pages:

- **SIP General Parameters page:** Provides SIP parameters for configuring general SIP features. To access this page, use the following path: **Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**.
- **SIP Advanced Parameters page:** Provides SIP parameters for configuring advanced SIP features. To access this page, use the following path: **Configuration** tab > **VoIP** menu > **SIP Definitions** > **Advanced Parameters**.

For a description of these parameters, refer to the section corresponding to the feature or see "Configuration Parameters Reference" on page 813.

18.2 Configuring Registration Accounts

The Account table lets you configure up to 102 Accounts. An Account defines registration information for registering and authenticating (digest) Trunk Groups (e.g., PBX) or IP Groups (e.g., IP PBX) with a "serving" IP Group (e.g., ITSP).

The device initiates registration with a "serving" IP Group on behalf of the "served" Trunk Group or IP Group. Therefore, Accounts are typically required when the "served" Trunk Group or IP Group is unable to register or authenticate itself for whatever reason. Registration information includes username, password, host name (AOR), and contact user name (AOR). The device includes this information in the REGISTER message sent to the serving IP Group. Up to 10 Accounts can be configured per "served" Trunk Group or IP Group. A Trunk Group or IP Group can register to more than one IP Group (e.g., multiple ITSPs). This is done by configuring multiple entries in the Account table for the same served Trunk Group or IP Group, but with different serving IP Groups, username/password, host name, and contact user values.

Authentication is typically required for INVITE messages sent to the "serving" IP Group. If the device receives a SIP 401 (Unauthorized) in response to a sent INVITE, the device checks for a matching "serving" and "served" entry in the Account table. If a matching row exists, the device authenticates the INVITE by providing the corresponding MD5 authentication username and password to the "serving" IP Group.

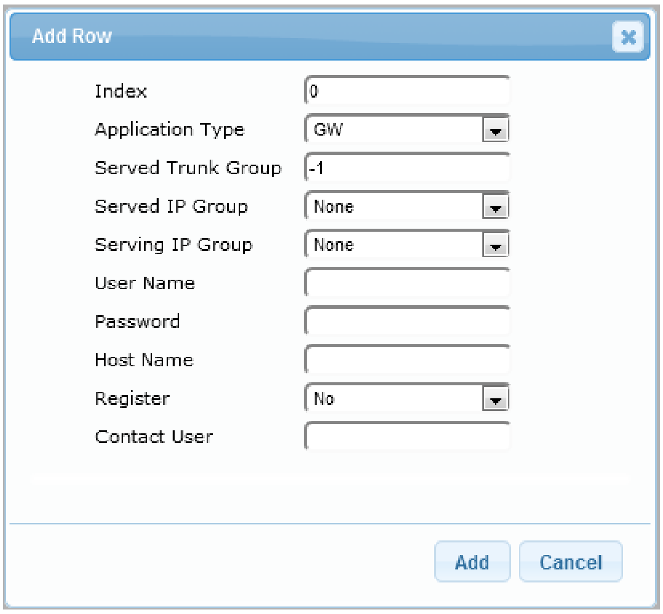
Note: If no match is found in the Account table for incoming or outgoing calls, the username and password is taken from:

-
- 'UserName' and 'Password' parameters on the Proxy & Registration page

The following procedure describes how to configure Accounts through the Web interface. You can also configure it through ini file (Account) or CLI (configure voip > sip-definition account).

➢ **To configure an Account:**

1. Open the Account table (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Account Table**).
2. Click **Add**; the following dialog box appears:



Index: 0
Application Type: GW
Served Trunk Group: -1
Served IP Group: None
Serving IP Group: None
User Name:
Password:
Host Name:
Register: No
Contact User:

3. Configure an account according to the parameters described in the table below.
4. Click **Add**.

Once you have configured Accounts, you can register or un-register them, as described below:

➢ To register or un-register an Account:

1. In the table, select the required Account entry row.
2. From the **Action** drop-down list, choose one of the following commands:
   - **Register** to register the Account.
   - **Un-Register** to un-register an Account.

To view Account registration status, see "Viewing Registration Status" on page 728.

If all trunks belonging to the Trunk Group are down, the device un-registers them. If any trunk belonging to the Trunk Group is returned to service, the device registers them again. This ensures, for example, that the Proxy does not send INVITEs to trunks that are out of service.

If registration with an IP Group fails for all accounts of a specific Trunk Group that includes all the channels in the Trunk Group, the Trunk Group is set to Out-Of-Service if the OOSOnRegistrationFail parameter is set to 1 (see Proxy & Registration Parameters on page 367).

Table 18-1: Account Table Parameter Descriptions

Parameter	Description
Index	Defines an index for the new table row. **Note:** Each row must be configured with a unique index.
Served Trunk Group `served-trunk-group` [Account_ServedTrunkGroup]	Defines the Trunk Group ID that you want to register and/or authenticate. ▪ For Tel-to-IP calls, the served Trunk Group is the source Trunk Group from where the call originated. ▪ For IP-to-Tel calls, the served Trunk Group is the Trunk Group ID to where the call is sent. **Note:** The parameter is applicable only to the Gateway application.
Served IP Group	Defines the IP Group (e.g., IP-PBX) that you want to register and/or authenticate upon its behalf.

Parameter	Description
`served-ip-group-name` [Account_ServedIPGroupName]	**Note:** - The parameter is applicable only to the SBC application. - By default, all IP Groups are displayed. However, if you filter the Web display by SRD (using the SRD Filter box), only IP Groups associated with the filtered SRD are displayed.
Serving IP Group `serving-ip-group-name` [Account_ServingIPGroupName]	Defines the IP Group (*Serving IP Group*) to where the device sends the SIP REGISTER requests (if enabled) for registration and authentication (of the Served IP Group). - Tel-to-IP calls: The serving IP Group is the destination IP Group configured in the Trunk Group Settings table or Tel-to-IP Routing table (see Configuring Tel-to-IP Routing Rules on page 467). - IP-to-Tel calls: The serving IP Group is the 'Source IP Group ID' configured in the IP to Trunk Group Routing table (see "Configuring IP-to-Trunk Group Routing Rules" on page 476). **Note:** By default, only IP Groups associated with the SRD to which the Served IP Group is associated are displayed, as well as IP Groups of Shared SRDs. However, if you filter the Web display by SRD (using the SRD Filter box), only IP Groups associated with the filtered SRD are displayed, as well as IP Groups of Shared SRDs.
User Name `user-name` [Account_Username]	Defines the digest MD5 Authentication username. The valid value is a string of up to 50 characters.
Password `password` [Account_Password]	Defines the digest MD5 Authentication password. The valid value is a string of up to 50 characters.
Host Name `host-name` [Account_HostName]	Defines the Address of Record (AOR) host name. The host name appears in SIP REGISTER From/To headers as ContactUser@HostName. For a successful registration, the host name is also included in the URI of the INVITE From header. The valid value is a string of up to 49 characters. **Note:** If the parameter is not configured or if registration fails, the 'SIP Group Name' parameter value configured in the IP Group table is used instead.
Register `register` [Account_Register]	Enables registration. - **[0]** No= (Default) The device only performs authentication (not registration). Authentication is typically done for INVITE messages sent to the "serving" IP Group. If the device receives a SIP 401 (Unauthorized) in response to a sent INVITE, the device checks for a matching "serving" and "served" entry in the table. If a matching row exists, the device authenticates the INVITE by providing the corresponding MD5 authentication username and password to the "serving" IP Group. - **[1]** Regular = Regular registration process. For more information, see "Regular Registration Mode" on page 366. - **[2]** GIN = Registration for legacy PBXs, using Global Identification Number (GIN). For more information, see "Single Registration for Multiple Phone Numbers using GIN" on page 366.

Parameter	Description
	Notes: ▪ Gateway application: To enable registration, you also need to set the 'Registration Mode' parameter to Per Account in the Trunk Group Settings table (see Configuring Trunk Group Settings on page 435). ▪ The account registration is not affected by the IsRegisterNeeded parameter.
Contact User `contact-user` [Account_ContactUser]	Defines the AOR username. This appears in REGISTER From/To headers as ContactUser@HostName, and in INVITE/200 OK Contact headers as ContactUser@<device's IP address>. **Notes:** ▪ If the parameter is not configured, the 'Contact User' parameter in the IP Group table is used instead. ▪ If registration fails, the user part in the INVITE Contact header contains the source party number.
Application Type `application-type` [Account_ApplicationType]	Defines the application type: ▪ [0] GW = (Default) Gateway application. ▪ [2] SBC = SBC application.

18.2.1 Regular Registration Mode

When you configure the registration mode in the Account table to **Regular**, the device sends REGISTER requests to the Serving IP Group. The host name (in the SIP From/To headers) and contact user (user in From/To and Contact headers) are taken from the configured Account table upon successful registration. See the example below:

```
REGISTER sip:xyz SIP/2.0
Via: SIP/2.0/UDP 10.33.37.78;branch=z9hG4bKac1397582418
From: <sip:ContactUser@HostName>;tag=1c1397576231
To: <sip: ContactUser@HostName >
Call-ID: 1397568957261200022256@10.33.37.78
CSeq: 1 REGISTER
Contact: <sip:ContactUser@10.33.37.78>;expires=3600
Expires: 3600
User-Agent: Sip-Gateway/v.6.80A.227.005
Content-Length: 0
```

18.2.2 Single Registration for Multiple Phone Numbers using GIN

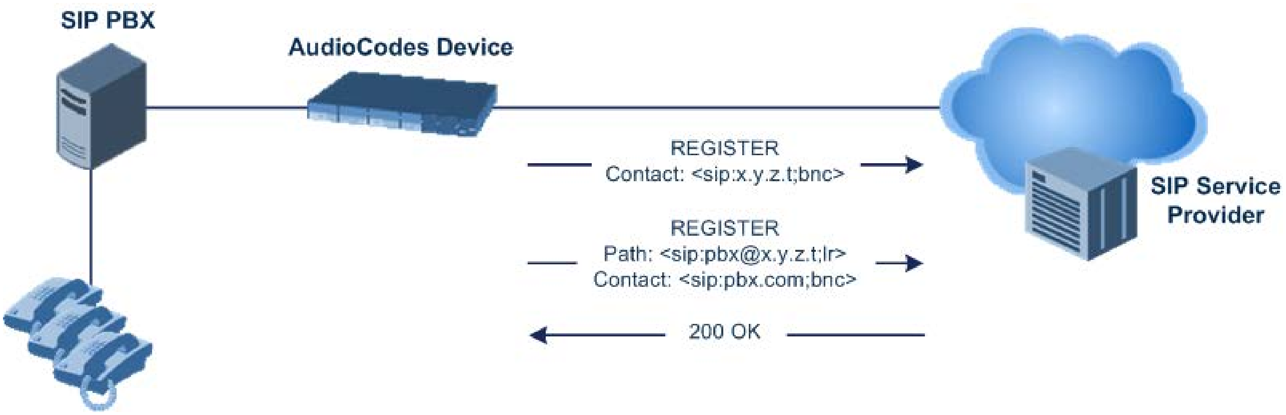
When you configure the registration mode in the Account table to **GIN**, the Global Identifiable Number (GIN) registration method is used, according to RFC 6140. The device performs GIN-based registration of users to a SIP registrar on behalf of a SIP PBX. In effect, the PBX registers with the service provider, just as a directly hosted SIP endpoint would register. However, because a PBX has multiple user agents, it needs to register a contact address on behalf of each of these. Rather than performing a separate registration procedure for each user agents, GIN registration mode does multiple registrations using a single REGISTER transaction.

According to this mechanism, the PBX delivers to the service provider in the Contact header field of a REGISTER request a template from which the service provider can construct contact URIs for each of the AORs assigned to the PBX and thus, can register these contact URIs within its location service. These registered contact URIs can then be used to deliver to the PBX inbound requests targeted at the AORs concerned. The mechanism can be used with AORs comprising SIP URIs based on global E.164 numbers and the service provider's domain name or sub-domain name.

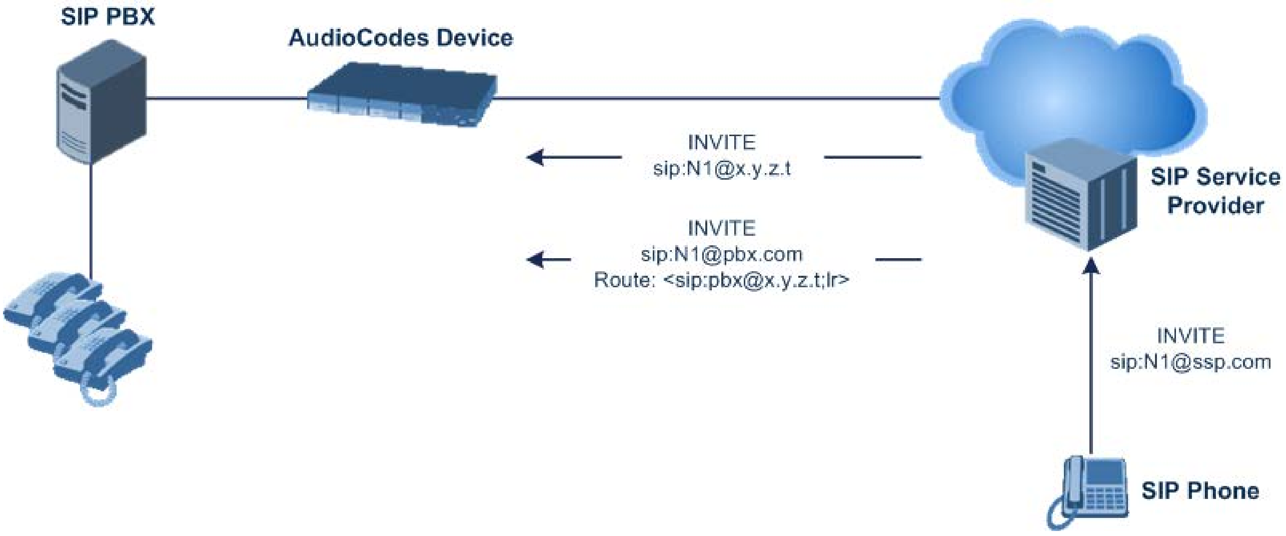
The SIP REGISTER request sent by the device for GIN registration with a SIP server provider contains the Require and Proxy-Require headers. These headers contain the token 'gin'. The Supported header contains the token 'path' and the URI in the Contact header contains the parameter 'bnc' without a user part:

```
Contact: <sip:198.51.100.3;bnc>;
```

The figure below illustrates the GIN registration process:


SIP PBX
AudioCodes Device
SIP Service Provider
REGISTER
Contact: <sip:x.y.z.t;bnc>
REGISTER
Path: <sip:pbx@x.y.z.t;lr>
Contact: <sip:pbx.com;bnc>
200 OK

The figure below illustrates an incoming call using GIN:



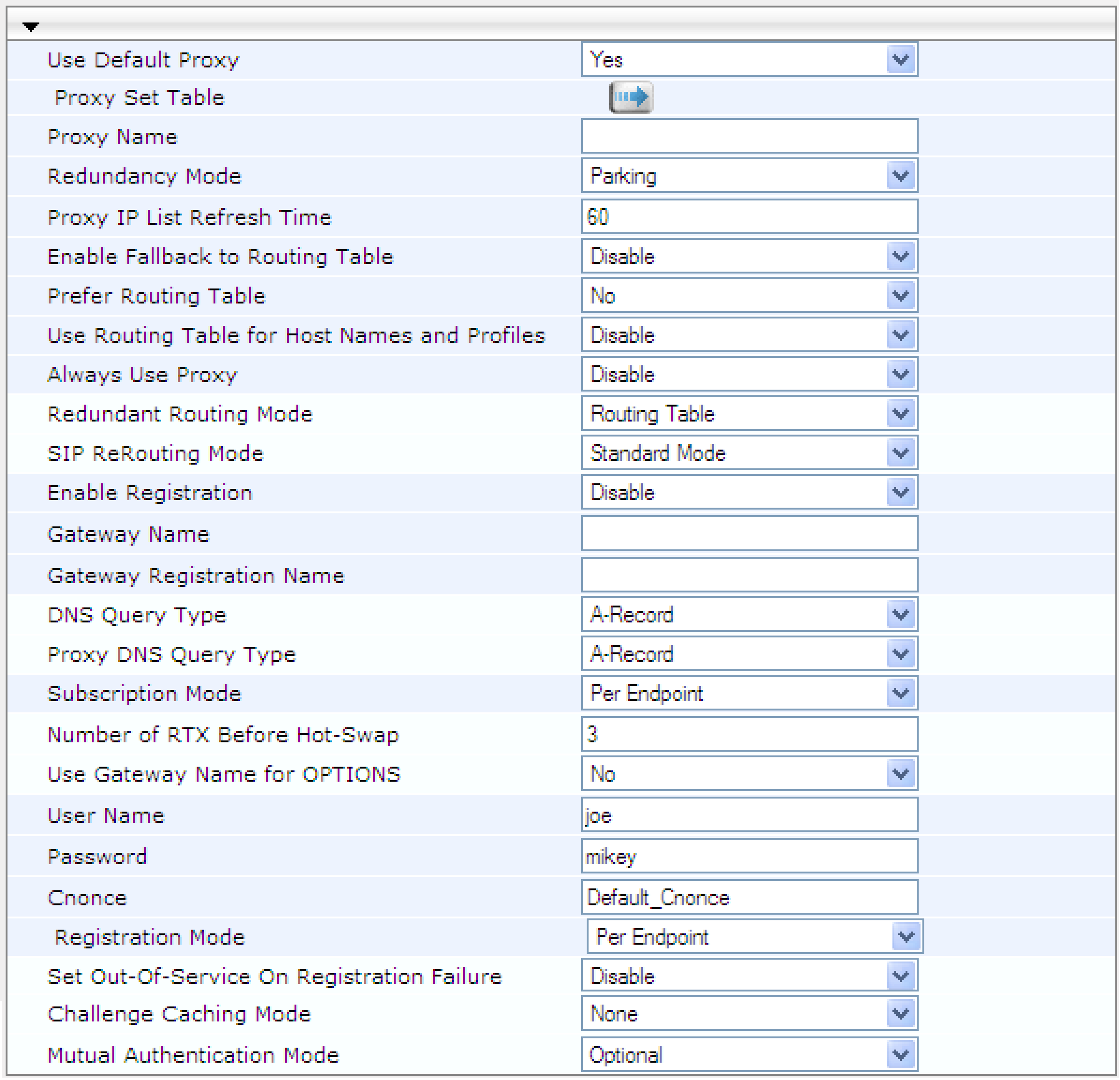
18.3 Configuring Proxy and Registration Parameters

The Proxy & Registration page allows you to configure the Proxy server and registration parameters. For a description of the parameters appearing on this page, see "Configuration Parameters Reference" on page 813.

Note: To view the registration status of endpoints with a SIP Registrar/Proxy server, see "Viewing Registration Status" on page 728.

➢ To configure the Proxy and registration parameters:

1. Open the Proxy & Registration page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Proxy & Registration**).



Use Default Proxy	Yes
Proxy Set Table	➡
Proxy Name	
Redundancy Mode	Parking
Proxy IP List Refresh Time	60
Enable Fallback to Routing Table	Disable
Prefer Routing Table	No
Use Routing Table for Host Names and Profiles	Disable
Always Use Proxy	Disable
Redundant Routing Mode	Routing Table
SIP ReRouting Mode	Standard Mode
Enable Registration	Disable
Gateway Name	
Gateway Registration Name	
DNS Query Type	A-Record
Proxy DNS Query Type	A-Record
Subscription Mode	Per Endpoint
Number of RTX Before Hot-Swap	3
Use Gateway Name for OPTIONS	No
User Name	joe
Password	mikey
Cnonce	Default_Cnonce
Registration Mode	Per Endpoint
Set Out-Of-Service On Registration Failure	Disable
Challenge Caching Mode	None
Mutual Authentication Mode	Optional

2. Configure the parameters as required.
3. Click **Submit**.

➢ To register or un-register the device to a Proxy/Registrar:

- Click the **Register** button to register.
- Click **Un-Register** button to un-register.

Instead of registering the entire device, you can register specific entities as listed below by using the **Register** button located on the page in which these entities are configured:

- BRI endpoints, Trunk Groups - Trunk Group table (see Configuring Trunk Groups on page 433)
- Accounts - Account table (see "Configuring Registration Accounts" on page 363)

Click the **Proxy Set Table** ➡ button to Open the Proxy Sets table to configure groups of proxy addresses. Alternatively, you can open this page from the **Proxy Sets Table** page item (see "Configuring Proxy Sets" on page 352 for a description of this page).

18.3.1 SIP Message Authentication Example

The device supports basic and digest (MD5) authentication types, according to SIP RFC 3261. A proxy server might require authentication before forwarding an INVITE message. A Registrar/Proxy server may also require authentication for client registration. A proxy replies to an unauthenticated INVITE with a 407 Proxy Authorization Required response, containing a Proxy-Authenticate header with the form of the challenge. After sending an ACK for the 407, the user agent can then re-send the INVITE with a Proxy-Authorization header containing the credentials.

User agents, Redirect or Registrar servers typically use the SIP 401 Unauthorized response to challenge authentication containing a WWW-Authenticate header, and expect the re-INVITE to contain an Authorization header.

The following example shows the Digest Authentication procedure, including computation of user agent credentials:

1. The REGISTER request is sent to a Registrar/Proxy server for registration:

```
REGISTER sip:10.2.2.222 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.200
From: <sip: 122@10.1.1.200>;tag=1c17940
To: <sip: 122@10.1.1.200>
Call-ID: 634293194@10.1.1.200
User-Agent: Sip-Gateway/Mediant 500 E-SBC/v.6.80A.227.005
CSeq: 1 REGISTER
Contact: sip:122@10.1.1.200:
Expires:3600
```

2. Upon receipt of this request, the Registrar/Proxy returns a 401 Unauthorized response:

```
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 10.2.1.200
From: <sip:122@10.2.2.222 >;tag=1c17940
To: <sip:122@10.2.2.222 >
Call-ID: 634293194@10.1.1.200
Cseq: 1 REGISTER
Date: Mon, 30 Jul 2012 15:33:54 GMT
Server: Columbia-SIP-Server/1.17
Content-Length: 0
WWW-Authenticate: Digest realm="audiocodes.com",
nonce="11432d6bce58ddf02e3b5e1c77c010d2",
stale=FALSE,
algorithm=MD5
```

3. According to the sub-header present in the WWW-Authenticate header, the correct REGISTER request is created.

4. Since the algorithm is MD5:
   - The username is equal to the endpoint phone number "122".
   - The realm return by the proxy is "audiocodes.com".
   - The password from the *ini* file is "AudioCodes".
   - The equation to be evaluated is "122:audiocodes.com:AudioCodes". According to the RFC, this part is called A1.
   - The MD5 algorithm is run on this equation and stored for future usage.
   - The result is "a8f17d4b41ab8dab6c95d3c14e34a9e1".

5. The par called A2 needs to be evaluated:
   - The method type is "REGISTER".
   - Using SIP protocol "sip".
   - Proxy IP from *ini* file is "10.2.2.222".
   - The equation to be evaluated is "REGISTER:sip:10.2.2.222".

- The MD5 algorithm is run on this equation and stored for future usage.
- The result is "a9a031cfddcb10d91c8e7b4926086f7e".

6. Final stage:
    - A1 result: The nonce from the proxy response is "11432d6bce58ddf02e3b5e1c77c010d2".
    - A2 result: The equation to be evaluated is "A1:11432d6bce58ddf02e3b5e1c77c010d2:A2".
    - The MD5 algorithm is run on this equation. The outcome of the calculation is the response needed by the device to register with the Proxy.
    - The response is "b9c45d0234a5abf5ddf5c704029b38cf".

    At this time, a new REGISTER request is issued with the following response:

```
REGISTER sip:10.2.2.222 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.200
From: <sip: 122@10.1.1.200>;tag=1c23940
To: <sip: 122@10.1.1.200>
Call-ID: 654982194@10.1.1.200
Server: Audiocodes-Sip-Gateway/Mediant 500 E-
SBC/v.6.80A.227.005
CSeq: 1 REGISTER
Contact: sip:122@10.1.1.200:
Expires:3600
Authorization: Digest, username: 122,
realm="audiocodes.com”,
nonce="11432d6bce58ddf02e3b5e1c77c010d2",
uri=”10.2.2.222”,
response=“b9c45d0234a5abf5ddf5c704029b38cf”
```

7. Upon receiving this request and if accepted by the Proxy, the Proxy returns a 200 OK response, completing the registration transaction:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.1.1.200
From: <sip: 122@10.1.1.200>;tag=1c23940
To: <sip: 122@10.1.1.200>
Call-ID: 654982194@10.1.1.200
Cseq: 1 REGISTER
Date: Thu, 26 Jul 2012 09:34:42 GMT
Server: Columbia-SIP-Server/1.17
Content-Length: 0
Contact: <sip:122@10.1.1.200>; expires="Thu, 26 Jul 2012
10:34:42 GMT"; action=proxy; q=1.00
Contact: <122@10.1.1.200:>; expires="Tue, 19 Jan 2038 03:14:07
GMT"; action=proxy; q=0.00
Expires: Thu, 26 Jul 2012 10:34:42 GMT
```

18.4 Configuring SIP Message Manipulation

The Message Manipulations table lets you configure up to 100 Message Manipulation rules. A Message Manipulation rule defines a manipulation sequence for SIP messages. SIP message manipulation enables the normalization of SIP messaging fields between communicating network segments. For example, it allows service providers to design their own policies on the SIP messaging fields that must be present before a SIP call enters their network. Similarly, enterprises and small businesses may have policies for the information that can enter or leave their networks for policy or security reasons from a service provider. SIP message manipulations can also be implemented to resolve incompatibilities between SIP devices inside the enterprise network.

Each Message Manipulation rule is configured with a Manipulation Set ID. You can create groups (sets) of Message Manipulation rules by assigning each of the relevant Message Manipulation rules to the same Manipulation Set ID. The Manipulation Set ID is then used to assign the rules to specific calls:

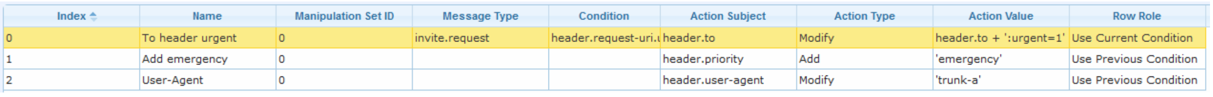
- SBC application: Message manipulation rules can be applied pre- or post-classification:
  - Pre-classification Process: Message manipulation can be done on incoming SIP dialog-initiating messages (e.g., INVITE) prior to the classification process. You configure this by assigning the Manipulation Set ID to the SIP Interface on which the call is received (see Configuring SIP Interfaces on page 333).
  - Post-classification Process: Message manipulation can be done on inbound and/or outbound SIP messages after the call has been successfully classified. You configure this by assigning the Manipulation Set ID to the relevant IP Group in the IP Group table (see Configuring IP Groups on page 340).
- Gateway application: Message Manipulation rules are applied to calls as follows:
  - Manipulating Inbound SIP INVITE Messages: Message manipulation can be applied only to all inbound calls (not specific calls). This is done by assigning a Manipulation Set ID to the "global" ini file parameter, GWInboundManipulationSet.
  - Manipulating Outbound SIP INVITE Messages:
    a. Message manipulation can be done for specific calls, by assigning a Manipulation Set ID to an IP Group in the IP Group table, using the 'Outbound Message Manipulation Set' parameter.
    b. Message manipulation can be applied to all outbound calls (except for IP Groups that have been assigned a Manipulation Set ID). This is done by assigning a Manipulation Set ID to the "global" ini file parameter, GWOutboundManipulationSet.

The device also supports a built-in SIP message normalization feature that can be enabled per Message Manipulation rule. The normalization feature removes unknown SIP message elements before forwarding the message. These elements can include SIP headers, SIP header parameters, and SDP body fields.

The SIP message manipulation feature supports the following:

- Manipulation on SIP message type (Method, Request/Response, and Response type)
- Addition of new SIP headers
- Removal of SIP headers ("black list")
- Modification of SIP header components such as values, header values (e.g., URI value of the P-Asserted-Identity header can be copied to the From header), call's parameter values
- Deletion of SIP body (e.g., if a message body is not supported at the destination network this body is removed)
- Translating one SIP response code to another
- Topology hiding (generally present in SIP headers such as Via, Record Route, Route and Service-Route).
- Configurable identity hiding (information related to identity of subscribers, for example, P-Asserted-Identity, Referred-By, Identity and Identity-Info)
- Apply conditions per rule - the condition can be on parts of the message or call’s parameters
- Multiple manipulation rules on the same SIP message
- Multiple manipulation rules using the same condition. The following figure shows a configuration example where rules 1 and 2 ('Row Rule' configured to **Use Previous Condition**) use the condition configured for rule 0 ('Row Rule' configured to **Use Current Condition**). For more information, see the description of the 'Row Rule' parameter in this section.

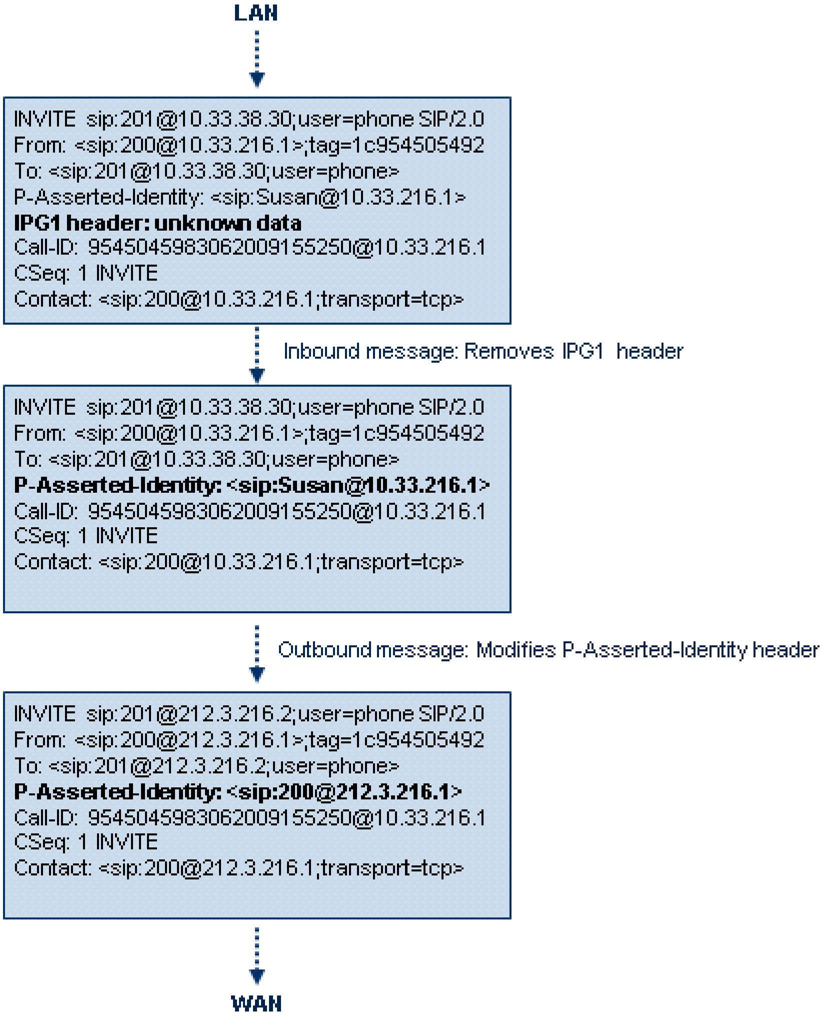
Figure 18-1: Configuration Example of Message Manipulation Rules uising Same Condition



Index	Name	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value	Row Role
0	To header urgent	0	invite.request	header.request-uri.(	header.to	Modify	header.to + ':urgent=1'	Use Current Condition
1	Add emergency	0			header.priority	Add	'emergency'	Use Previous Condition
2	User-Agent	0			header.user-agent	Modify	'trunk-a'	Use Previous Condition

The figure below illustrates a SIP message manipulation example:

Figure 18-2: SIP Header Manipulation Example


LAN
Inbound message: Removes IPG1 header
Outbound message: Modifies P-Asserted-Identity header
WAN

```
INVITE  sip:201@10.33.38.30;user=phone SIP/2.0
From: <sip:200@10.33.216.1>;tag=1c954505492
To: <sip:201@10.33.38.30;user=phone>
P-Asserted-Identity: <sip:Susan@10.33.216.1>
IPG1 header: unknown data
Call-ID: 954504598306200915525O@10.33.216.1
CSeq: 1 INVITE
Contact: <sip:200@10.33.216.1;transport=tcp>
```

```
INVITE  sip:201@10.33.38.30;user=phone SIP/2.0
From: <sip:200@10.33.216.1>;tag=1c954505492
To: <sip:201@10.33.38.30;user=phone>
P-Asserted-Identity: <sip:Susan@10.33.216.1>
Call-ID: 954504598306200915525O@10.33.216.1
CSeq: 1 INVITE
Contact: <sip:200@10.33.216.1;transport=tcp>
```

```
INVITE  sip:201@212.3.216.2;user=phone SIP/2.0
From: <sip:200@212.3.216.1>;tag=1c954505492
To: <sip:201@212.3.216.2;user=phone>
P-Asserted-Identity: <sip:200@212.3.216.1>
Call-ID: 954504598306200915525O@10.33.216.1
CSeq: 1 INVITE
Contact: <sip:200@212.3.216.1;transport=tcp>
```

Notes:

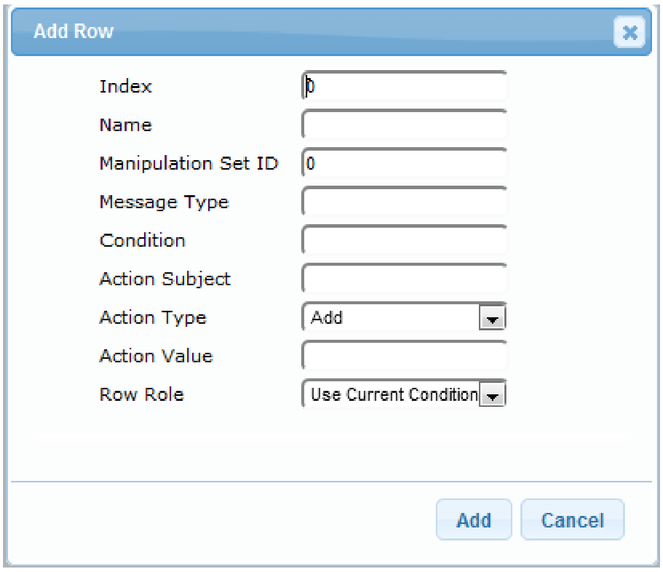
- For a detailed description of the syntax used for configuring Message Manipulation rules, refer to the *SIP Message Manipulations Quick Reference Guide*.
- For the SBC application, Inbound message manipulation is done only after the Classification, inbound/outbound number manipulations, and routing processes.
- Each message can be manipulated twice - on the source leg and on the destination leg (i.e., source and destination IP Groups).
- Unknown SIP parts can only be added or removed.
- SIP manipulations do not allow you to remove or add mandatory SIP headers. They can only be modified and only on requests that initiate new dialogs. Mandatory SIP headers include To, From, Via, CSeq, Call-Id, and Max-Forwards.
- The SIP Group Name (IPGroup_SIPGroupName) parameter overrides inbound message manipulation rules that manipulate the host name in Request-URI, To, and/or From SIP headers. If you configure a SIP Group Name for the IP Group (see Configuring IP Groups on page 340) and you want to manipulate the host name in any of these SIP headers, you must apply your manipulation rule (Manipulation Set ID) to the IP Group as an Outbound Message Manipulation Set (IPGroup_OutboundManSet), when the IP Group is the destination of the call. If you apply the Manipulation Set as an Inbound Message Manipulation Set (IPGroup_InboundManSet), when the IP Group is the source of the call, the manipulation rule will be overridden by the SIP Group Name.

The following procedure describes how to configure Message Manipulation rules through the Web interface. You can also configure it through ini file (MessageManipulations) or CLI (configure voip > sbc manipulations message-manipulations).

➢ To configure SIP message manipulation rules:

1. Open the Message Manipulations page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Msg Policy & Manipulation** > **Message Manipulations**).
2. Click **Add**; the following dialog box appears:

Figure 18-3: Message Manipulations Table - Add Row Dialog Box



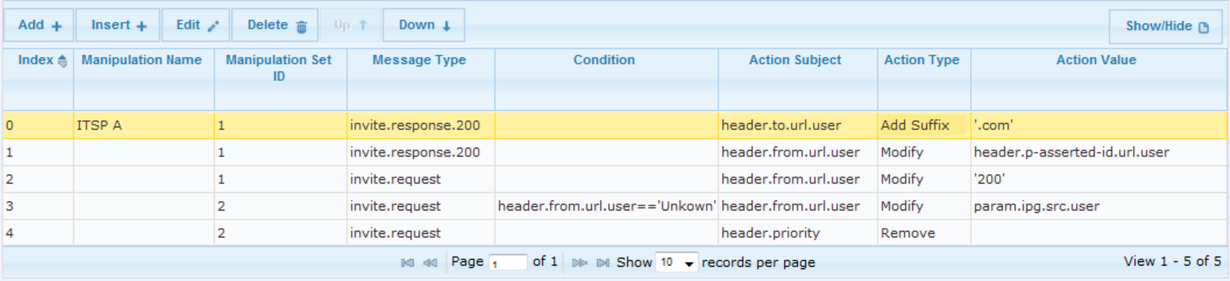
Add Row

Index	0
Name	
Manipulation Set ID	0
Message Type	
Condition	
Action Subject	
Action Type	Add
Action Value	
Row Role	Use Current Condition

3. Configure a Message Manipulation rule according to the parameters described in the table below.
4. Click **Add**.

An example of configured message manipulation rules are shown in the figure below:

Figure 18-4: Message Manipulations Page



Index	Manipulation Name	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value
0	ITSP A	1	invite.response.200		header.to.url.user	Add Suffix	'.com'
1		1	invite.response.200		header.from.url.user	Modify	header.p-asserted-id.url.user
2		1	invite.request		header.from.url.user	Modify	'200'
3		2	invite.request	header.from.url.user=='Unkown'	header.from.url.user	Modify	param.ipg.src.user
4		2	invite.request		header.priority	Remove	

- Index 0: Adds the suffix ".com" to the host part of the To header.
- Index 1: Changes the user part of the From header to the user part of the P-Asserted-ID.
- Index 2: Changes the user part of the SIP From header to "200".
- Index 3: If the user part of the From header equals "unknown", then it is changed according to the srcIPGroup call's parameter.
- Index 4: Removes the Priority header from an incoming INVITE message.

Table 18-2: Message Manipulations Parameter Descriptions

Parameter	Description
Index [MessageManipulations_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `manipulation-name` [MessageManipulations_ManipulationName]	Defines an arbitrary name to easily identify the rule. The valid value is a string of up to 16 characters.
Manipulation Set ID `manipulation-set-id` [MessageManipulations_ManSetID]	Defines a Manipulation Set ID for the rule. You can define the same Manipulation Set ID for multiple rules to create a group of rules. The Manipulation Set ID is used to assign the manipulation rules to an IP Group (in the IP Group table) for inbound and/or outbound messages. The valid value is 0 to 19. The default is 0.
Matching Characteristics	
Message Type `message-type` [MessageManipulations_MessageType]	Defines the SIP message type that you want to manipulate. The valid value is a string (case-insensitive) denoting the SIP message. For example: ▪ Empty = rule applies to all messages ▪ Invite = rule applies to all INVITE requests and responses ▪ Invite.Request = rule applies to INVITE requests ▪ Invite.Response = rule applies to INVITE responses ▪ subscribe.response.2xx = rule applies to SUBSCRIBE confirmation responses **Note:** Currently, SIP 100 Trying messages cannot be manipulated.

Parameter	Description
Condition `condition` [MessageManipulations_Condition]	Defines the condition that must exist for the rule to apply. The valid value is a string (case-insensitive). For example: ▪ header.from.url.user== '100' (indicates that the user part of the From header must have the value "100") ▪ header.contact.param.expires > '3600' ▪ header.to.url.host contains 'domain' ▪ param.call.dst.user != '100'
Operation	
Action Subject `action-subject` [MessageManipulations_ActionSubject]	Defines the SIP header upon which the manipulation is performed. The valid value is a string (case-insensitive).
Action Type `action-type` [MessageManipulations_ActionType]	Defines the type of manipulation. ▪ **[0]** Add (default) = Adds new header/param/body (header or parameter elements). ▪ **[1]** Remove = Removes header/param/body (header or parameter elements). ▪ **[2]** Modify = Sets element to the new value (all element types). ▪ **[3]** Add Prefix = Adds value at the beginning of the string (string element only). ▪ **[4]** Add Suffix = Adds value at the end of the string (string element only). ▪ **[5]** Remove Suffix = Removes value from the end of the string (string element only). ▪ **[6]** Remove Prefix = Removes value from the beginning of the string (string element only). ▪ **[7]** Normalize = Removes unknown SIP message elements before forwarding the message.
Action Value `action-value` [MessageManipulations_ActionValue]	Defines a value that you want to use in the manipulation. The default value is a string (case-insensitive) in the following syntax: ▪ string/<message-element>/<call-param> + ▪ string/<message-element>/<call-param> For example: ▪ 'itsp.com' ▪ header.from.url.user ▪ param.call.dst.user ▪ param.call.dst.host + '.com' ▪ param.call.src.user + '<' + header.from.url.user + '@' + header.p-asserted-id.url.host + '>' **Note:** Only single quotation marks must be used.
Row Role `row-role` [MessageManipulations_RowRole]	Determines which message manipulation condition (configured by the 'Condition' parameter) to use for the rule.

Parameter	Description
	- [0] Use Current Condition = (Default) The condition configured in the table row of the rule is used. - [1] Use Previous Condition = The condition configured in the first table row above the rule that is configured to **Use Current Condition** is used. For example, if Index 3 is configured to **Use Current Condition** and Index 4 and 5 are configured to **Use Previous Condition**, Index 4 and 5 use the condition configured for Index 3. A configuration example is shown in the beginning of this section. The option allows you to use the same condition for multiple manipulation rules. **Notes:** - When configured to **Use Previous Condition**, the 'Message Type' and 'Condition' parameters are not applicable and if configured are ignored. - When multiple manipulation rules apply to the same header, the next rule applies to the resultant string of the previous rule.

18.5 Configuring SIP Message Policy Rules

The Message Policy table lets you configure up to 20 SIP Message Policy rules. SIP Message Policy rules are used to block (blacklist) unwanted incoming SIP messages or permit (whitelist) receipt of desired SIP messages. You can configure legal and illegal characteristics of a SIP message. This feature is helpful against VoIP fuzzing (also known as robustness testing), which sends different types of packets to its "victims" for finding bugs and vulnerabilities. For example, the attacker might try sending a SIP message containing either an oversized parameter or too many occurrences of a parameter.

To apply SIP Message Policy rules, you need to assign them to SIP Interfaces associated with the relevant IP Groups (see "Configuring SIP Interfaces" on page 333).

Each Message Policy rule can be configured with the following:

- Maximum message length
- Maximum header length
- Maximum message body length
- Maximum number of headers
- Maximum number of bodies
- Option to send 400 "Bad Request" response if message request is rejected
- Blacklist and whitelist for defined methods (e.g., INVITE)
- Blacklist and whitelist for defined bodies

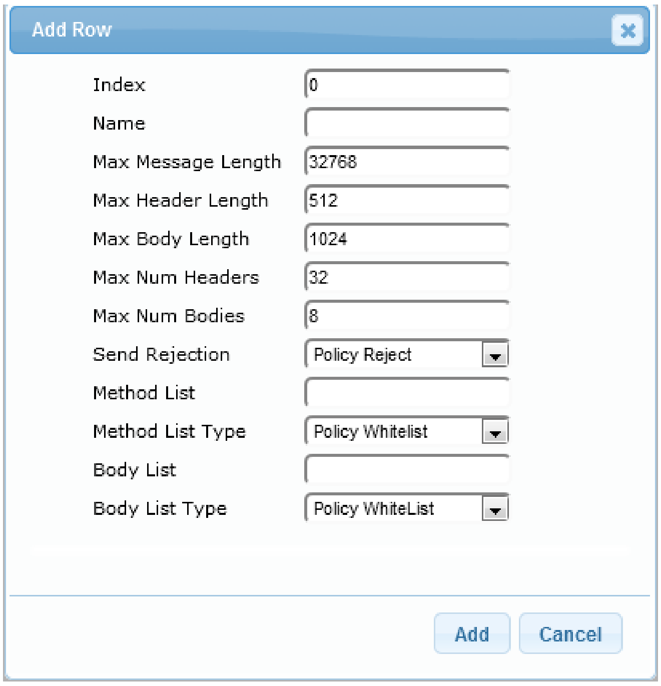
The following procedure describes how to configure Message Policy rules through the Web interface. You can also configure it through ini file (MessagePolicy) or CLI (configure voip > sbc message-policy).

➢ **To configure SIP Message Policy rules:**

1. Open the Message Policy table (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Msg Policy & Manipulation** > **Message Policy Table**).

2. Click **Add**; the following dialog box appears:

Figure 18-5: Message Policy Table - Add Row Dialog Box



Index: 0
Name:
Max Message Length: 32768
Max Header Length: 512
Max Body Length: 1024
Max Num Headers: 32
Max Num Bodies: 8
Send Rejection: Policy Reject
Method List:
Method List Type: Policy Whitelist
Body List:
Body List Type: Policy WhiteList

3. Configure a Message Policy rule according to the parameters described in the table below.
4. Click **Add**.

Table 18-3: Message Policy Table Parameter Descriptions

Parameter	Description
Index [MessagePolicy_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `name` [MessagePolicy_Name]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. **Note:** Each row must be configured with a unique name.
Max Message Length `max-message-length` [MessagePolicy_MaxMessageLength]	Defines the maximum SIP message length. The valid value is up to 32,768 characters. The default is 32,768.
Max Header Length `max-header-length` [MessagePolicy_MaxHeaderLength]	Defines the maximum SIP header length. The valid value is up to 512 characters. The default is 512.
Max Body Length `max-body-length` [MessagePolicy_MaxBodyLength]	Defines the maximum SIP message body length. This is the value of the Content-Length header. The valid value is up to 1,024 characters. The default is 1,024.
Max Num Headers `max-num-headers` [MessagePolicy_MaxNumHeaders]	Defines the maximum number of SIP headers. The valid value is any number up to 32. The default is 32. **Note:** The device supports up to 20 SIP Record-Route headers that can be received in a SIP INVITE request or a 200 OK response. If it receives more than this, it responds with a SIP 513 'Message Too Large' response.

Parameter	Description
Max Num Bodies `max-num-bodies` [MessagePolicy_MaxNumBodies]	Defines the maximum number of bodies (e.g., SDP) in the SIP message. The valid value is any number up to 8. The default is 8.
Send Rejection `send-rejection` [MessagePolicy_SendRejection]	Determines whether the device sends a 400 "Bad Request" response if a message request is rejected. ▪ **[0]** Policy Reject = (Default) If the message is a request, the device sends a response to reject the request. ▪ **[1]** Policy Drop = The device ignores the message without sending any response.
SIP Method Blacklist-Whitelist Policy	
Method List `method-list` [MessagePolicy_MethodList]	Defines SIP methods (e.g., INVITE\BYE) to blacklist or whitelist. Multiple methods are separated by a backslash (\). The method values are case-insensitive.
Method List Type `method-list-type` [MessagePolicy_MethodListType]	Defines the policy (blacklist or whitelist) for the SIP methods specified in the 'Method List' parameter (above). ▪ **[0]** Policy Blacklist = The specified methods are rejected. ▪ **[1]** Policy Whitelist = (Default) Only the specified methods are allowed; the others are rejected.
SIP Body Blacklist-Whitelist Policy	
Body List `body-list` [MessagePolicy_BodyList]	Defines the SIP body type (i.e., value of the Content-Type header) to blacklist or whitelist. For example, application/sdp. The values of the parameter are case-sensitive.
Body List Type `body-list-type` [MessagePolicy_BodyListType]	Defines the policy (blacklist or whitelist) for the SIP body specified in the 'Body List' parameter (above). ▪ **[0]** Policy Blacklist =The specified SIP body is rejected. ▪ **[1]** Policy Whitelist = (Default) Only the specified SIP body is allowed; the others are rejected.

19 Coders and Profiles

This section describes configuration of the coders and SIP profiles parameters.

19.1 Configuring Default Coders

The Coders table lets you configure up to 11 voice coders for the device. This is the default Coder Group, which is used by the device for all calls, unless a different Coder Group, configured in the Coder Group Settings table (see "Configuring Coder Groups" on page 382) is assigned to specific calls, using Tel or IP Profiles.

Each coder can be configured with packetization time (ptime), bit rate, payload type, and silence suppression. The first coder configured in the table has the highest priority and is used by the device whenever possible. If the remote side cannot use the first coder, the device attempts to use the next coder in the table, and so on.

Notes:

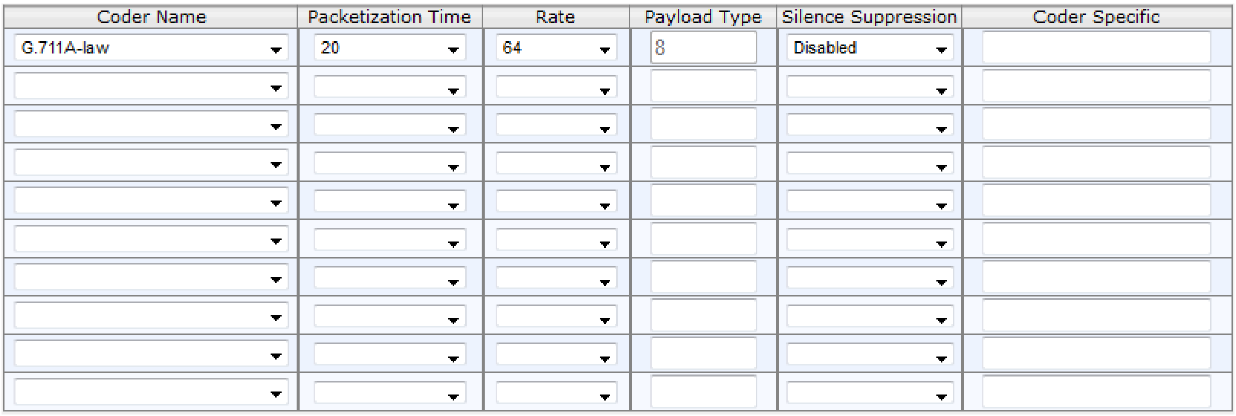
- Some coders are license-dependent and are available only if purchased from AudioCodes and included in the Software License Key installed on your device. For more information, contact your AudioCodes sales representative.
- Only the packetization time of the first coder in the coder list is declared in INVITE/200 OK SDP, even if multiple coders are defined. The device always uses the packetization time requested by the remote side for sending RTP packets. If not specified, the packetization time is assigned the default value.
- The value of several fields is hard-coded according to common standards (e.g., payload type of G.711 U-law is always 0). Other values can be set dynamically. If no value is specified for a dynamic field, a default value is assigned. If a value is specified for a hard-coded field, the value is ignored.
- The G.722 coder provides Packet Loss Concealment (PLC) capabilities, ensuring higher voice quality.
- Opus coder:
  - √ For SBC calls: If one leg uses a narrowband coder (e.g., G.711) and the other leg uses the Opus coder, the device maintains the narrowband coder flavor by using the narrowband Opus coder. Alternatively, if one leg uses a wideband coder (e.g., G.722) and the other leg uses the Opus coder, the device maintains the wideband coder flavor by using the wideband Opus coder.
  - √ Gateway calls always use the narrowband Opus coder.
- For information on V.152 and implementation of T.38 and VBD coders, see "Supporting V.152 Implementation" on page 187.

The following procedure describes how to configure the Coders table through the Web interface. You can also configure it through ini file (CodersGroup) or CLI (configure voip > coders-and-profiles coders-group).

➢ To configure coders:

1. Open the Coders page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **Coders**).

Figure 19-1: Coders Table Page



Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression	Coder Specific
G.711A-law	20	64	8	Disabled	

2. Configure coders according to the parameters described in the table below.
3. Click **Submit**, and then reset the device with a save ("burn") to flash memory.

Table 19-1: Coders Table Parameter Descriptions

Parameter	Description
Coder Name `name` [CodersGroup0_Name]	Defines the coder. **Note:** Each coder type (e.g., G.729) can be configured only once in the table.
Packetization Time `p-time` [CodersGroup0_pTime]	Defines the packetization time (in msec) for the coder. The packetization time determines how many coder payloads are combined into a single RTP packet.
Rate `rate` [CodersGroup0_rate]	Defines the bit rate (in kbps) for the coder.
Payload Type `payload-type` [CodersGroup0_PayloadType]	Defines the payload type if the payload type (i.e., format of the RTP payload) for the coder is dynamic.
Silence Suppression `silence-suppression` [CodersGroup0_Sce]	Enables silence suppression for the coder. ▪ **[0]** Disable (Default) ▪ **[1]** Enable ▪ **[2]** Enable w/o Adaptation =Applicable only to G.729. **Notes:** ▪ If G.729 is configured with silence suppression disabled, the device includes 'annexb=no' in the SDP of the relevant SIP messages. If silence suppression is enabled or set to **Enable w/o Adaptations**, 'annexb=yes' is included. For the Gateway application, an exception to this logic is when the remote gateway is Cisco equipment (IsCiscoSCEMode).

Parameter	Description
Coder Specific `coder-specific` [CodersGroup0_CoderSpecific]	Defines additional settings specific to the coder. Currently, the parameter is applicable only to the AMR coder and is used to configure the payload format type. ▪ [0] 0 = Bandwidth Efficient ▪ [1] 1 = Octet Aligned (default) **Note:** The AMR payload type can be configured globally using the AmrOctetAlignedEnable parameter. However, the Coder Group configuration overrides the global parameter.

The table below lists the supported coders:

Table 19-2: Supported Coders

Coder Name	Packetization Time (msec)	Rate (kbps)	Payload Type	Silence Suppression
G.711 A-law **[g711Alaw64k]**	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	64	8	▪ **[0]** Disable ▪ **[1]** Enable
G.711 U-law **[g711Ulaw64k]**	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	64	0	▪ **[0]** Disable ▪ **[1]** Enable
G.711A-law_VBD **[g711AlawVbd]**	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	64	8 or Dynamic (default 118)	N/A
G.711U-law_VBD **[g711UlawVbd]**	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	64	0 or Dynamic (default 110	N/A
G.722 **[g722]**	20 (default), 40, 60, 80, 100, 120	64 (default)	9	N/A
G.723.1 **[g7231]**	30 (default), 60, 90, 120, 150	▪ **[0]** 5.3 (default) ▪ **[1]** 6.3	4	▪ **[0]** Disable ▪ **[1]** Enable
G.726 **[g726]**	10, 20 (default), 30, 40, 50, 60, 80	▪ **[0]** 16 ▪ **[1]** 24 ▪ **[2]** 32 (default) ▪ **[3]** 40	Dynamic (default 2	▪ **[0]** Disable ▪ **[1]** Enable
G.729 **[g729]**	10, 20 (default), 30, 40, 50, 60, 80, 100	8	18	▪ **[0]** Disable ▪ **[1]** Enable ▪ **[2]** Enable w/o Adaptations

Coder Name	Packetization Time (msec)	Rate (kbps)	Payload Type	Silence Suppression
AMR [Amr]	20 (default)	▪ [0] 4.75 ▪ [1] 5.15 ▪ [2] 5.90 ▪ [3] 6.70 ▪ [4] 7.40 ▪ [5] 7.95 ▪ [6] 10.2 ▪ [7] 12.2 (default)	Dynamic	▪ [0] Disable ▪ [1] Enable
AMR-WB [Amr-WB]	20 (default)	▪ [0] 6.6 ▪ [1] 8.85 ▪ [2] 12.65 ▪ [3] 14.25 ▪ [4] 15.85 ▪ [5] 18.25 ▪ [6] 19.85 ▪ [7] 23.05 ▪ [8] 23.85 (default)	Dynamic	▪ [0] Disable ▪ [1] Enable
iLBC [iLBC]	20 (default), 40, 60, 80, 100, 120	15 (default)	Dynamic (default 65)	▪ [0] Disable ▪ [1] Enable
	30 (default), 60, 90, 120	13		
silk-nb [Silk-8Khz]	20 (default), 40, 60, 80, and 100	8	Dynamic (default 76)	N/A
silk-wb [Silk-16Khz]	20 (default), 40, 60, 80, and 100	16	Dynamic (default 77)	N/A
T.38 **[t38fax]**	N/A	N/A	N/A	N/A
T.38 Version 3 [t38fax]	-	-	-	-
T.38 Over RTP	N/A	N/A	Dynamic (default 106)	N/A
OPUS [Opus]	20 (default), 40, 60, 80, 120	N/A	Dynamic (default 111)	N/A

19.2 Configuring Coder Groups

The Coder Group Settings table lets you configure up to 11 *Coder Groups.* A Coder Group is a set of configured coders (coder type, packetization time, rate, payload type, and silence suppression). Each Coder Group can include up to 10 coders.

The first coder in the Coder Group has the highest priority and is used by the device whenever possible. If the remote side cannot use the first coder, the device attempts to use the next coder in the Coder Group, and so on.

To define coders for specific calls, you can configure a Coder Group with the necessary coders and then assign the Coder Group to the calls using Tel Profiles (see Configuring Tel Profiles on page 384) or IP Profiles (see "Configuring IP Profiles" on page 387).

Notes:

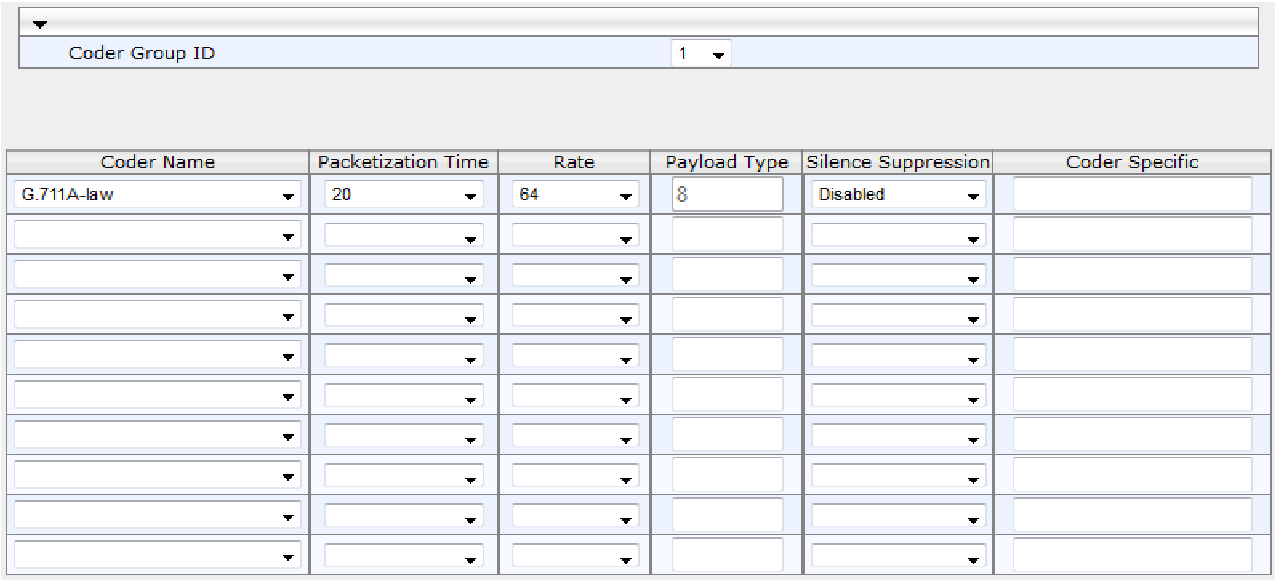
- To define coders for calls that are not assigned a specific Coder Group using Tel Profiles or IP Profiles, see "Configuring Default Coders" on page 379. This group of coders is termed the *Default Coder Group*.
- For a list of supported coders, see "Configuring Default Coders" on page 379.

The following procedure describes how to configure the Coders table through the Web interface. You can also configure it through ini file (CodersGroup) or CLI (configure voip > coders-and-profiles coders-group).

➢ To configure a Coder Group:

1. Open the Coder Group Settings page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **Coders Group Settings**).

Figure 19-2: Coder Group Settings Page



Coder Group ID: 1

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression	Coder Specific
G.711A-law	20	64	8	Disabled	

2. Configure the Coder Group according to the parameters described in the table below.
3. Click **Add**, and then reset the device with a save ("burn") to flash memory.

Table 19-3: Coder Group Settings Table Parameter Descriptions

Parameter	Description
Coder Group ID [CodersGroupX_Index]	Defines an ID for the Coder Group.
Coder Name `name` [CodersGroupX_Name]	Defines the coder type. **Note:** Each coder type (e.g., G.729) can be configured only once in the table.
Packetization Time `p-time` [CodersGroupX_pTime]	Defines the packetization time (in msec) for the coder. The packetization time determines how many coder payloads are combined into a single RTP packet.

Parameter	Description
Rate `rate` [CodersGroupX_rate]	Defines the bit rate (in kbps) for the coder.
Payload Type `payload-type` [CodersGroupX_PayloadType]	Defines the payload type if the payload type (i.e., format of the RTP payload) for the coder is dynamic.
Silence Suppression `silence-suppression` [CodersGroupX_Sce]	Enables silence suppression for the coder. ▪ **[0]** Disable (Default) ▪ **[1]** Enable ▪ **[2]** Enable w/o Adaptation =Applicable only to G.729. ▪
Coder Specific `coder-specific` [CodersGroupX_CoderSpecific]	Defines additional settings specific to the coder. Currently, the parameter is applicable only to the AMR coder and is used to configure the payload format type. ▪ [0] 0 = Bandwidth Efficient ▪ [1] 1 = Octet Aligned (default) **Note:** The AMR payload type can be configured globally using the AmrOctetAlignedEnable parameter. However, the Coder Group configuration overrides the global parameter.

19.3 Configuring Tel Profile

The Tel Profile Settings table lets you configure up to nine *Tel Profiles*. A Tel Profile is a set of parameters with specific settings which can be assigned to specific calls. The Tel Profile Settings table includes a wide range of parameters for configuring the Tel Profile. Each of these parameters has a corresponding "global" parameter, which when configured applies to all calls. The main difference, if any, between the Tel Profile parameters and their corresponding global parameters are their default values.

Tel Profiles provide high-level adaptation when the device interworks between different equipment and protocols (at both the Tel and IP sides), each of which may require different handling by the device. For example, if specific channels require the use of the G.711 coder, you can configure a Tel Profile with this coder and assign it to these channels.

To use your Tel Profile for specific calls, you need to assign it to specific channels (trunks) in the Trunk Group table (see Configuring Trunk Groups on page 433)).

The following procedure describes how to configure Tel Profiles through the Web interface. You can also configure it through ini file (TelProfile) or CLI (configure voip/coders-and-profiles tel-profile).

➤ **To configure a Tel Profile:**

1. Open the Tel Profile Settings page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **Tel Profile Settings**).
2. Click **Add**; the following dialog box appears:
3. Configure a Tel Profile according to the parameters described in the table below. For a description of each parameter, refer to the corresponding "global" parameter.
4. Click **Add**.

Table 19-4: Tel Profile Table Parameter Descriptions

Parameter	Description
Common	
Index [TelProfile_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Profile Name `profile-name` [TelProfile_ProfileName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters.
Profile Preference `tel-preference` [TelProfile_TelPreference]	Defines the priority of the Tel Profile, where **1** is the lowest priority and **20** the highest priority. **Notes:** 1. If both the IP Profile and Tel Profile apply to the same call, the coders and common parameters of the Preferred profile are applied to the call. 2. If the Preference of the Tel Profile and IP Profile are identical, the Tel Profile parameters are applied. 3. If the coder lists of both the IP Profile and Tel Profile apply to the same call, only the coders common to both are used. The order of the coders is determined by the preference.
Fax Signaling Method `fax-sig-method` [TelProfile_IsFaxUsed]	IsFaxUsed
Enable Digit Delivery `digit-delivery` [TelProfile_EnableDigitDelivery]	EnableDigitDelivery
Time For Reorder Tone `time-for-reorder-tone` [TelProfile_TimeForReorderTone]	TimeForReorderTone
Disconnect Call on Detection of Busy Tone `disconnect-on-busy-tone` [TelProfile_DisconnectOnBusyTone]	DisconnectOnBusyTone
Enable Voice Mail Delay `enable-voice-mail-delay` [TelProfile_EnableVoiceMailDelay]	VoiceMailInterface This is useful for disabling voice mail services per Trunk Group to eliminate the phenomenon of call delay on Trunks that do not implement voice mail when voice mail is enabled using the global parameter.
Dial Plan Index `dial-plan-index` [TelProfile_DialPlanIndex]	DialPlanIndex
Swap Tel To IP Phone Numbers `swap-teltoip-phone-numbers` [TelProfile_SwapTelToIpPhoneNumbers]	SwapTEl2IPCalled&CallingNumbers
Call Priority Mode `call-priority-mode` [TelProfile_CallPriorityMode]	CallPriorityMode
IP Related	
Coders Group ID `coders-group-id` [TelProfile_CodersGroupID]	CodersGroup0
RTP IP DiffServ `rtp-ip-diffserv` [TelProfile_IPDiffServ]	PremiumServiceClassMediaDiffServ

Parameter	Description
Signaling DiffServ `signaling-diffserv` [TelProfile_SigIPDiffServ]	PremiumServiceClassControlDiffServ
Enable Early Media `early-media` [TelProfile_EnableEarlyMedia]	EnableEarlyMedia
Progress Indicator to IP `prog-ind-to-ip` [TelProfile_ProgressIndicator2IP]	ProgressIndicator2IP
Channel	
Dynamic Jitter Buffer Minimum Delay `jitter-buffer-minimum-delay` [TelProfile_JitterBufMinDelay]	DJBufMinDelay
Dynamic Jitter Buffer Optimization Factor `jitter-buffer-optimization-factor` [TelProfile_JitterBufOptFactor]	DJBufOptFactor
DTMF Volume `dtmf-volume` [TelProfile_DtmfVolume]	DTMFVolume
Input Gain `input-gain` [TelProfile_InputGain]	InputGain
Voice Volume `voice-volume` [TelProfile_VoiceVolume]	VoiceVolume
Echo Canceler `echo-canceller` [TelProfile_EnableEC]	EnableEchoCanceller
Enable AGC `enable-agc` [TelProfile_EnableAGC]	EnableAGC
EC NLP Mode `echo-canceller-nlp-mode` [TelProfile_ECNlpMode]	ECNLPMode

19.4 Configuring IP Profiles

The IP Profile Settings table lets you configure up to 20 IP Profiles. An IP Profile is a set of parameters with user-defined settings relating to signaling (e.g., SIP message terminations such as REFER) and media (e.g., coder type). An IP Profile can later be assigned to specific IP calls (inbound and/or outbound). Thus, IP Profiles provide high-level adaptation when the device interworks between different IP entities, each of which may require different handling by the device. For example, if a specific IP entity uses the G.711 coder only, you can configure an IP Profile with G.711 for this IP entity.

To use your IP Profile for specific calls, you need to assign it to any of the following:

- IP Groups - see "Configuring IP Groups" on page 340
- Gateway application: Tel-to-IP routing rules - see Configuring Tel-to-IP Routing Rules on page 467
- Gateway application: IP-to-Tel routing rules - see Configuring IP-to-Trunk Group Routing Rules on page 476

For the Gateway application: The device selects the IP Profile as follows:

- If you assign different IP Profiles (not default) to the same specific calls in all of the above-mentioned tables, the device uses the IP Profile that has the highest preference level (as set in the 'Profile Preference' parameter). If these IP Profiles have the same preference level, the device uses the IP Profile that you assigned in the IP Group table.
- If you assign different IP Profiles to all of the above-mentioned tables and one table is set to the default IP Profile, the device uses the IP Profile that is not the default.

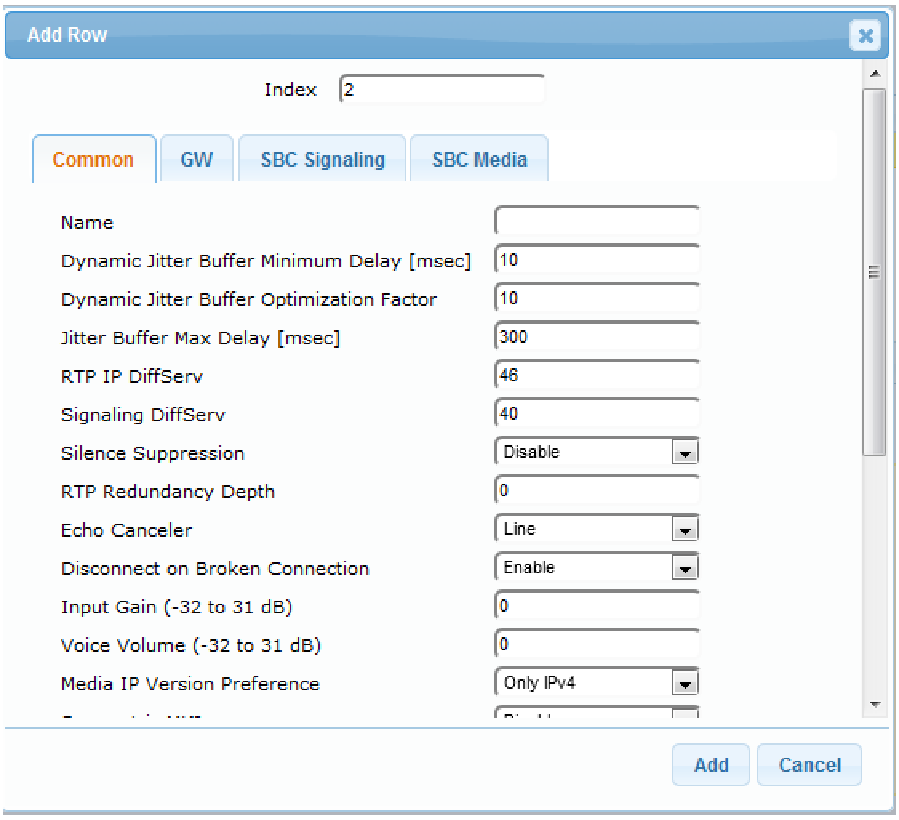
Many of the parameters in the IP Profile table have a corresponding "global" parameter. For calls that are not associated with any IP Profile, the settings of the "global" parameters are applied.

Note: IP Profiles can also be implemented when using a Proxy server (when the AlwaysUseRouteTable parameter is set to 1).

The following procedure describes how to configure IP Profiles through the Web interface. You can also configure it through ini file (IPProfile) or CLI (configure voip > coders-and-profiles ip-profile).

➢ To configure an IP Profile:

1. Open the IP Profile Settings table (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **IP Profile Settings**).
2. Click **Add**; the following dialog box appears:



Index: 2

Name:

Dynamic Jitter Buffer Minimum Delay [msec]: 10

Dynamic Jitter Buffer Optimization Factor: 10

Jitter Buffer Max Delay [msec]: 300

RTP IP DiffServ: 46

Signaling DiffServ: 40

Silence Suppression: Disable

RTP Redundancy Depth: 0

Echo Canceler: Line

Disconnect on Broken Connection: Enable

Input Gain (-32 to 31 dB): 0

Voice Volume (-32 to 31 dB): 0

Media IP Version Preference: Only IPv4

3. Configure an IP Profile according to the parameters described in the table below.
4. Click **Add**.

Table 19-5: IP Profile Settings Table Parameter Descriptions

Parameter	Description
Common	
Index [IpProfile_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `profile-name` [IpProfile_ProfileName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters.
Dynamic Jitter Buffer Minimum Delay `jitter-buffer-minimum-delay` [IpProfile_JitterBufMinDelay]	Defines the minimum delay (in msec) of the device's dynamic Jitter Buffer. The valid range is 0 to 150. The default delay is 10. For more information on Jitter Buffer, see Dynamic Jitter Buffer Operation on page 187. **Note:** The corresponding global parameter is DJBufMinDelay.
Dynamic Jitter Buffer Optimization Factor `jitter-buffer-optimization-factor` [IpProfile_JitterBufOptFactor]	Defines the Dynamic Jitter Buffer frame error/delay optimization factor. The valid range is 0 to 12. The default factor is 10. For more information on Jitter Buffer, see Dynamic Jitter Buffer Operation on page 187. **Notes:** ▪ For data (fax and modem) calls, set the parameter to 12. ▪ The corresponding global parameter is DJBufOptFactor.

Parameter	Description
Jitter Buffer Max Delay `jitter-buffer-max-delay` [IpProfile_JitterBufMaxDelay]	Defines the maximum delay and length (in msec) of the Jitter Buffer. The valid range is 150 to 2,000. The default is 250.
RTP IP DiffServ `rtp-ip-diffserv` [IpProfile_IPDiffServ]	Defines the DiffServ value for Premium Media class of service (CoS) content. The valid range is 0 to 63. The default is 46. **Note:** The corresponding global parameter is PremiumServiceClassMediaDiffServ.
Signaling DiffServ `signaling-diffserv` [IpProfile_SigIPDiffServ]	Defines the DiffServ value for Premium Control CoS content (Call Control applications). The valid range is 0 to 63. The default is 40. **Note:** ▪ The corresponding global parameter is PremiumServiceClassControlDiffServ.
RTP Redundancy Depth `rtp-redundancy-depth` [IpProfile_RTPRedundancyDepth]	Enables the device to generate RFC 2198 redundant packets. This can be used for packet loss where the missing information (audio) can be reconstructed at the receiver's end from the redundant data that arrives in subsequent packets. This is required, for example, in wireless networks where a high percentage (up to 50%) of packet loss can be experienced. ▪ **[0]** 0 = (Default) Disable. ▪ **[1]** 1 = Enable - previous voice payload packet is added to current packet. **Notes:** ▪ When enabled, you can configure the payload type, using the RFC2198PayloadType parameter. ▪ For the Gateway application only: The RTP redundancy dynamic payload type can be included in the SDP, by using the EnableRTPRedundancyNegotiation parameter. ▪ The corresponding global parameter is RTPRedundancyDepth.
Echo Canceler `echo-canceller` [IpProfile_EnableEchoCanceller]	Enables the device's Echo Cancellation feature (i.e., echo from voice calls is removed). ▪ [0] Disable ▪ [1] Line (default) For a detailed description of the Echo Cancellation feature, see Configuring Echo Cancellation on page 174. **Note:** The corresponding global parameter is EnableEchoCanceller.
Broken Connection Mode `disconnect-on-broken-connection` [IpProfile_DisconnectOnBrokenConnection]	Defines the device's handling of calls when RTP packets (media) are not received within a user-defined timeout (configured by the BrokenConnectionEventTimeout parameter). ▪ [0] Ignore = The call is maintained despite no media and is released when signaling ends the call (i.e., SIP BYE). ▪ [1] Disconnect = (Default) The device ends the call. ▪ [2] Reroute = (SBC application only) The device ends the call and searches the IP-to-IP Routing table for a matching rule and if found, generates a new INVITE to the corresponding destination (i.e., alternative routing). You can configure a routing rule whose matching characteristics is explicitly for

Parameter	Description
	calls with broken RTP connections. This is done using the Call Trigger parameter, as described in Configuring SBC IP-to-IP Routing Rules. **Note:** ▪ The device can only detect a broken RTP connection if silence compression is disabled for the RTP session. ▪ If during a call the source IP address (from where the RTP packets are received by the device) is changed without notifying the device, the device rejects these RTP packets. To overcome this, configure the DisconnectOnBrokenConnection parameter to 0. By this configuration, the device doesn't detect RTP packets arriving from the original source IP address and switches (after 300 msec) to the RTP packets arriving from the new source IP address. ▪ The corresponding global parameter is DisconnectOnBrokenConnection.
Input Gain `input-gain` [IpProfile_InputGain]	Defines the pulse-code modulation (PCM) input gain control (in decibels). For the Gateway application: Defines the level of the received signal for Tel-to-IP calls. The valid range is -32 to 31 dB. The default is 0 dB. **Note:** The corresponding global parameter is InputGain.
Voice Volume `voice-volume` [IpProfile_VoiceVolume]	Defines the voice gain control (in decibels). For the Gateway application: Defines the level of the transmitted signal for IP-to-Tel calls. The valid range is -32 to 31 dB. The default is 0 dB. **Note:** The corresponding global parameter is VoiceVolume.

Media IP Version Preference
media-ip-version-preference
[IpProfile_MediaIPVersionPreference]
	Defines the preferred RTP media IP addressing version for outgoing SIP calls (according to RFC 4091 and RFC 4092). The RFCs concern Alternative Network Address Types (ANAT) semantics in the SDP to offer groups of network addresses (IPv4 and IPv6) and the IP address version preference to establish the media stream. The IP address is indicated in the "c=" field (Connection) of the SDP.
	▪ [0] Only IPv4 = (Default) SDP offer includes only IPv4 media IP addresses.
	▪ [1] Only IPv6 = SDP offer includes only IPv6 media IP addresses.
	▪ [2] Prefer IPv4 = SDP offer includes IPv4 and IPv6 media IP addresses, but the first (preferred) media is IPv4.
	▪ [3] Prefer IPv6 = SDP offer includes IPv4 and IPv6 media IP addresses, but the first (preferred) media is IPv6.
	To indicate ANAT support, the device uses the SIP Allow header or to enforce ANAT it uses the Require header:
	`Require: sdp-anat`
	In the outgoing SDP, each 'm=' field is associated with an ANAT group. This is done using the 'a=mid:' and 'a=group:ANAT' fields. Each 'm=' field appears under a unique 'a=mid:' number, for example:
	`a=mid:1`
	`m=audio 63288 RTP/AVP 0 8 18 101`

Parameter	Description
	c=IN IP6 3000::290:8fff:fe40:3e21 The 'a=group:ANAT' field shows the 'm=' fields belonging to it, using the number of the 'a=mid:' field. In addition, the ANAT group with the preferred 'm=' fields appears first. For example, the preferred group includes 'm=' fields under 'a=mid:1' and 'a=mid3': a=group:ANAT 1 3 a=group:ANAT 2 4 If you configure the parameter to a "prefer" option, the outgoing SDP offer contains two medias which are the same except for the "c=" field. The first media is the preferred address type (and this type is also on the session level "c=" field), while the second media has its "c=" field with the other address type. Both medias are grouped by ANAT. For example, if the incoming SDP contains two medias, one secured and the other non-secured, the device sends the outgoing SDP with four medias: ▪ Two secured medias grouped in the first ANAT group, one with IPv4 and the other with IPv6. The first is the preferred type. ▪ Two non-secured medias grouped in the second ANAT group, one with IPv4 and the other with IPv6. The first is the preferred type. **Note:** ▪ The parameter is applicable only when the device offers an SDP. ▪ The IP addressing version is determined according to the first SDP "m=" field. ▪ The feature is applicable to any type of media (e.g., audio and video) that has an IP address. ▪ The corresponding global parameter is MediaIPVersionPreference.

Symmetric MKI
`enable-symmetric-mki`
[IpProfile_EnableSymmetricMKI]
	Enables symmetric MKI negotiation.
	▪ **[0]** Disable = (Default) The device includes the MKI in its SIP 200 OK response according to the SRTPTxPacketMKISize parameter (if set to 0, it is not included; if set to any other value, it is included with this value).
	▪ **[1]** Enable = The answer crypto line contains (or excludes) an MKI value according to the selected crypto line in the offer. For example, assume that the device receives an INVITE containing the following two crypto lines in SDP:
	`a=crypto:2 AES_CM_128_HMAC_SHA1_80 inline:TAaxNnQt8/qLQMnDuG4vxYfWl6K7eBK/ufk04pR4|2^31|1:1`
	`a=crypto:3 AES_CM_128_HMAC_SHA1_80 inline:bnuYZnMxSfUiGitviWJZmzr7OF3AiRO0l5Vnh0kH|2^31`
	The first crypto line includes the MKI parameter "1:1". In the 200 OK response, the device selects one of the crypto lines (i.e., '2' or '3'). Typically, it selects the first line that supports the crypto suite. However, for SRTP-to-SRTP in SBC sessions, it can be determined by the remote side on the outgoing leg. If the device selects crypto line '2', it includes the MKI parameter in its answer SDP, for example:

Parameter	Description
	`a=crypto:2 AES_CM_128_HMAC_SHA1_80 inline:R1VyA1xV/qwBjkEklu4kSJyl3wCtYeZLq1/QFuxw|2^31|1:1` If the device selects a crypto line that does not contain the MKI parameter, then the MKI parameter is not included in the crypto line in the SDP answer (even if the SRTPTxPacketMKISize parameter is set to any value other than 0). **Note:** The corresponding global parameter is EnableSymmetricMKI.
MKI Size `mki-size` [IpProfile_MKISize]	Defines the size (in bytes) of the Master Key Identifier (MKI) in SRTP Tx packets. The valid value is 0 to 4. The default is 0 (i.e., new keys are generated without MKI). **Notes:** ▪ Gateway application: The device only initiates the MKI size. ▪ SBC application: The device can forward MKI size as is for SRTP-to-SRTP flows or override the MKI size during negotiation. This can be done on the inbound or outbound leg. ▪ The corresponding global parameter is SRTPTxPacketMKISize.
Reset SRTP Upon Re-key `reset-srtp-upon-re-key` [IpProfile_ResetSRTPStateUponRekey]	Enables synchronization of the SRTP state between the device and a server when a new SRTP key is generated upon a SIP session expire. This feature ensures that the roll-over counter (ROC), one of the parameters used in the SRTP encryption/decryption process of the SRTP packets is synchronized on both sides for transmit and receive packets. ▪ **[0]** Disable = (Default) ROC is not reset on the device side. ▪ **[1]** Enable = If the session expires causing a session refresh through a re-INVITE, the device or server generates a new key and the device resets the ROC index (and other SRTP fields) as done by the server, resulting in a synchronized SRTP. **Notes:** ▪ If this feature is disabled and the server resets the ROC upon a re-key generation, one-way voice may occur. ▪ The corresponding global parameter is ResetSRTPStateUponRekey.
Generate SRTP Keys Mode generate-srtp-keys **[IpProfile_GenerateSRTPKeys]**	Enables the device to generate a new SRTP key upon receipt of a re-INVITE with the SIP entity associated with the IP Profile. ▪ **[0]** Only If Required= (Default) The device generates an SRTP key only if necessary. ▪ **[1]** Always = The device always generates a new SRTP key.
AMD Sensitivity Parameter Suite `amd-sensitivity-parameter-suit` [IpProfile_AMDSensitivityParameterSuit]	Defines the AMD Parameter Suite to use for the Answering Machine Detection (AMD) feature. ▪ [0] 0 = (Default) Parameter Suite 0 based on North American English with standard detection sensitivity resolution (8 sensitivity levels, from 0 to 7). This AMD Parameter Suite is provided by the AMD Sensitivity file, which is shipped pre-installed on the device.

Parameter	Description
	▪ [1] 1 = Parameter Suite based 1 on North American English with high detection sensitivity resolution (16 sensitivity levels, from 0 to 15). This AMD Parameter Suite is provided by the AMD Sensitivity file, which is shipped pre-installed on the device. ▪ [2] 2 to [7]7 = Optional Parameter Suites that you can create based on any language (16 sensitivity levels, from 0 to 15). This requires a customized AMD Sensitivity file that needs to be installed on the device. For more information, contact your AudioCodes sales representative. **Notes:** ▪ To configure the detection sensitivity level, use the 'AMD Sensitivity Level' parameter. ▪ For more information on the AMD feature, see Answering Machine Detection (AMD) on page 198. ▪ The corresponding global parameter is AMDSensitivityParameterSuit.
AMD Sensitivity Level `amd-sensitivity-level` [IpProfile_AMDSensitivityLevel]	Defines the AMD detection sensitivity level of the selected AMD Parameter Suite (using the 'AMD Sensitivity Parameter Suite' parameter). For Parameter Suite 0, the valid range is 0 to 7, where 0 is for best detection of an answering machine and 7 for best detection of a live call. For any Parameter Suite other than 0, the valid range is 0 to 15, where 0 is for best detection of an answering machine and 15 for best detection of a live call. **Note:** The corresponding global parameter is AMDSensitivityLevel.
AMD Max Greeting Time `amd-max-greeting-time` [IpProfile_AMDMaxGreetingTime]	Defines the maximum duration (in 5-msec units) that the device can take to detect a greeting message. The valid range value is 0 to 51132767. The default is 300. **Note:** The corresponding global parameter is AMDMaxGreetingTime.
AMD Max Post Silence Greeting Time `amd-max-post-silence-greeting-time` [IpProfile_AMDMaxPostSilenceGreetingTime]	Defines the maximum duration of silence from after the greeting time is over (configured by AMDMaxGreetingTime) until the device's AMD decision. **Note:** The corresponding global parameter is AMDMaxPostGreetingSilenceTime.
GW (Gateway Calls)	
Profile Preference `ip-preference` [IpProfile_IpPreference]	Defines the priority of the IP Profile, where 20 is the highest priority and 1 the lowest priority. **Notes:** ▪ If an IP Profile and a Tel Profile apply to the same call, the coders and other common parameters of the profile with the highest preference are applied to the call. If the preference of the profiles is identical, the Tel Profile parameters are applied. ▪ If the coder lists of both an IP Profile and a Tel Profile apply to the same call, only the coders common to both are used. The order of the coders is determined by the preference. ▪ The parameter is applicable only to the Gateway application.

Parameter	Description
Coders `coders-group-id` [IpProfile_CodersGroupID]	Assigns a Coders Group, which defines coders supported by the SIP entity associated with the IP Profile. The value, Default Coders Group represents the coders configured in the Coders table (see Configuring Coders on page 379). All other optional values (e.g., Coders Group 1), represent the coders defined for the specific Coder Group configured in the Coder Group Settings table (see Configuring Coder Groups on page 382).
Media Security Mode `media-security-behaviour` [IpProfile_MediaSecurityBehaviour]	Defines the handling of SRTP for the SIP entity associated with the IP Profile. ▪ [-1] Not Configured = Applies the settings of the corresponding global parameter, MediaSecurityBehaviour. ▪ [0] Preferable = (Default) The device initiates encrypted calls to this SIP entity. However, if negotiation of the cipher suite fails, an unencrypted call is established. The device accepts incoming calls received from the SIP entity that don't include encryption information. ▪ [1] Mandatory = The device initiates encrypted calls to this SIP entity, but if negotiation of the cipher suite fails, the call is terminated. The device rejects incoming calls received from the SIP entity that don't include encryption information. ▪ [2] Disable = This SIP entity does not support encrypted calls (i.e., SRTP). ▪ [3] Preferable - Single Media = The device sends SDP with a single media ('m=') line only (e.g., m=audio 6000 RTP/AVP 4 0 70 96) with RTP/AVP and crypto keys. The SIP entity can respond with SRTP or RTP parameters: ✓ If the SIP entity does not support SRTP, it uses RTP and ignores the crypto lines. ✓ If the device receives an SDP offer with a single media (as shown above) from the SIP entity, it responds with SRTP (RTP/SAVP) if the EnableMediaSecurity parameter is set to 1. If SRTP is not supported (i.e., EnableMediaSecurity is set to 0), it responds with RTP. ✓ If two 'm=' lines are received in the SDP offer, the device prefers the SAVP (secure audio video profile), regardless of the order in the SDP. **Notes:** ▪ The parameter is applicable only when the EnableMediaSecurity parameter is set to 1. ▪ The corresponding global parameter is MediaSecurityBehaviour.
Is DTMF Used [IpProfile_IsDTMFUsed]	Enables DTMF signaling. ▪ [0] Disable = (Default) ▪ [1] Enable
First Tx DTMF Option `first-tx-dtmf-option` [IpProfile_FirstTxDtmfOption]	Defines the first preferred transmit DTMF negotiation method. ▪ [0] Not Supported = No negotiation - DTMF digits are sent according to the parameters DTMFTransportType and RFC2833PayloadType (for transmit and receive). ▪ [1] INFO (Nortel) = Sends DTMF digits according to IETF Internet-Draft draft-choudhuri-sip-info-digit-00.

Parameter	Description
	▪ [2] NOTIFY = Sends DTMF digits according to IETF Internet-Draft draft-mahy-sipping-signaled-digits-01. ▪ [3] INFO (Cisco) = Sends DTMF digits according to the Cisco format. ▪ [4] RFC 2833 = (Default) The device: ✓ negotiates RFC 2833 payload type using local and remote SDPs. ✓ sends DTMF packets using RFC 2833 payload type according to the payload type in the received SDP. ✓ expects to receive RFC 2833 packets with the same payload type as configured by the parameter RFC2833PayloadType. ✓ removes DTMF digits in transparent mode (as part of the voice stream). ▪ [5] INFO (Korea) = Sends DTMF digits according to the Korea Telecom format. **Notes:** ▪ When out-of-band DTMF transfer is used ([1], [2], [3], or [5]), the DTMFTransportType parameter is automatically set to 0 (DTMF digits are removed from the RTP stream). ▪ If an ISDN phone user presses digits (e.g., for interactive voice response / IVR applications such as retrieving voice mail messages), ISDN Information messages received by the device for each digit are sent in the voice channel to the IP network as DTMF signals, according to the settings of the parameter. ▪ The corresponding global parameter is FirstTxDTMFOption.
Second Tx DTMF Option `second-tx-dtmf-option` [IpProfile_SecondTxDtmfOption]	Defines the second preferred transmit DTMF negotiation method. For a description, see IpProfile_FirstTxDtmfOption (above). **Note:** The corresponding global parameter is SecondTxDTMFOption.
Rx DTMF Option `rx-dtmf-option` [IpProfile_RxDTMFOption]	Enables the device to declare the RFC 2833 'telephony-event' parameter in the SDP. ▪ [0] Not Supported ▪ [3] Supported (default) The device is always receptive to RFC 2833 DTMF relay packets. Thus, it is always correct to include the 'telephony-event' parameter by default in the SDP. However, some devices use the absence of the 'telephony-event' in the SDP to decide to send DTMF digits in-band using G.711 coder. If this is the case, set the parameter to 0. **Note:** The corresponding global parameter is RxDTMFOption.
Fax Signaling Method `fax-sig-method` [IpProfile_IsFaxUsed]	Defines the SIP signaling method for establishing and transmitting a fax session when the device detects a fax. ▪ [0] No Fax = (Default) No fax negotiation using SIP signaling. The fax transport method is according to the FaxTransportMode parameter. ▪ [1] T.38 Relay = Initiates T.38 fax relay. ▪ [2] G.711 Transport = Initiates fax/modem using the coder G.711 A-law/Mu-law with adaptations (see Note below).

Parameter	Description
	▪ [3] Fax Fallback = Initiates T.38 fax relay. If the T.38 negotiation fails, the device re-initiates a fax session using the coder G.711 A-law/Mu-law with adaptations (see the Note below). **Notes:** ▪ Fax adaptations (for options 2 and 3): ✓ Echo Canceller = On ✓ Silence Compression = Off ✓ Echo Canceller Non-Linear Processor Mode = Off ✓ Dynamic Jitter Buffer Minimum Delay = 40 ✓ Dynamic Jitter Buffer Optimization Factor = 13 ▪ If the device initiates a fax session using G.711 (option 2 or 3), a 'gpmd' attribute is added to the SDP in the following format: ✓ For A-law: 'a=gpmd:8 vbd=yes;ecan=on' ✓ For Mu-law: 'a=gpmd:0 vbd=yes;ecan=on' ▪ When the parameter is set to 1, 2, or 3, the parameter FaxTransportMode is ignored. ▪ When the parameter is set to 0, T.38 might still be used without the control protocol's involvement. To completely disable T.38, set FaxTransportMode to a value other than 1. ▪ For more information on fax transport methods, see Fax/Modem Transport Modes on page 175. ▪ The corresponding global parameter is IsFaxUsed.
CNG Detector Mode `cng-mode` [IpProfile_CNGmode]	Enables the detection of the fax calling tone (CNG) and defines the detection method. ▪ [0] Disable = (Default) The originating fax does not detect CNG; the device passes the CNG signal transparently to the remote side. ▪ [1] Relay = The originating fax detects CNG. The device sends CNG packets to the remote side according to T.38 (if IsFaxUsed is set to 1) and the fax session is started. A SIP Re-INVITE message is not sent and the fax session starts by the terminating fax. This option is useful, for example, when the originating fax is located behind a firewall that blocks incoming T.38 packets on ports that have not yet received T.38 packets from the internal network (i.e., originating fax). To also send a Re-INVITE message upon detection of a fax CNG tone in this mode, set the parameter FaxCNGMode to 1 or 2. ▪ [2] Event Only = The originating fax detects CNG and a fax session is started by the originating fax, using the Re-INVITE message. Typically, T.38 fax session starts when the preamble signal is detected by the answering fax. Some SIP devices do not support the detection of this fax signal on the answering fax and thus, in these cases, it is possible to configure the device to start the T.38 fax session when the CNG tone is detected by the originating fax. However, this mode is not recommended. **Note:** The corresponding global parameter is CNGDetectorMode.
Vxx Modem Transport Type `vxx-transport-type`	Defines the modem transport type.

Parameter	Description
[IpProfile_VxxTransportType]	▪ [-1] = (Not Configured) The settings of the global parameters are used: ✓ V21ModemTransportType ✓ V22ModemTransportType ✓ V23ModemTransportType ✓ V32ModemTransportType ✓ V34ModemTransportType ▪ [0] Disable = Transparent. ▪ [2] Enable Bypass (Default) ▪ [3] Events Only = Transparent with Events. For a detailed description of the parameter per modem type, see the relevant global parameter (listed above).
NSE Mode `nse-mode` [IpProfile_NSEMode]	Enables Cisco's compatible fax and modem bypass mode, Named Signaling Event (NSE) packets. ▪ [0] Disable (Default) ▪ [1] Enable In NSE bypass mode, the device starts using G.711 A-Law (default) or G.711□Law, according to the FaxModemBypassCoderType parameter. The payload type for these G.711 coders is a standard one (8 for G.711 A-Law and 0 for G.711 □Law). The parameters defining payload type for the 'old' Bypass mode FaxBypassPayloadType and ModemBypassPayloadType are not used with NSE Bypass. The bypass packet interval is configured according to the FaxModemBypassBasicRtpPacketInterval parameter. The SDP contains the following line: 'a=rtpmap:100 X-NSE/8000'. **Notes:** ▪ When enabled, the following conditions must also be met: ✓ The Cisco gateway must include the following definition: 'modem passthrough nse payload-type 100 codec g711alaw'. ✓ Set the Modem transport type to Bypass mode (VxxModemTransportType is set to 2) for all modems. ✓ Set the NSEPayloadType parameter to 100. ▪ The corresponding global parameter is NSEMode.
Play RB Tone to IP `play-rbt-to-ip` [IpProfile_PlayRBTone2IP]	Enables the device to play a ringback tone to the IP side for IP-to-Tel calls. ▪ [0] Disable (Default) ▪ [1] Enable = Plays a ringback tone after a SIP 183 session progress response is sent. **Notes:** ▪ ▪ To enable the device to send a 183/180+SDP responses, set the EnableEarlyMedia parameter to 1. ▪ If the EnableDigitDelivery parameter is set to 1, the device doesn't play a ringback tone to IP and doesn't send 183 or 180+SDP responses. ▪ If the parameter is enabled and EnableEarlyMedia is set to 1, the device plays a ringback tone according to the following:

Parameter	Description
	✓ ISDN: If a Progress or an Alerting message with PI (1 or 8) is received from the ISDN, the device opens a voice channel, sends a 183+SDP or 180+SDP response, but doesn't play a ringback tone to IP. If PI (1 or 8) is received from the ISDN, the device assumes that ringback tone is played by the ISDN switch; otherwise, the device plays a ringback tone to IP after receiving an Alerting message from the ISDN. It sends a 180+SDP response, signaling to the calling party to open a voice channel to hear the played ringback tone. ▪ The corresponding global parameter is PlayRBTone2IP.
Early Media `early-media` [IpProfile_EnableEarlyMedia]	Enables the Early Media feature for sending media (e.g., ringing) before the call is established. ▪ [0] Disable (default) ▪ [1] Enable ✓ The device sends a SIP 18x response with SDP, allowing the media stream to be established before the call is answered. **Notes:** ▪ The inclusion of the SDP in the 18x response depends on the ISDN Progress Indicator (PI). The SDP is sent only if PI is set to 1 or 8 in the received Proceeding, Alerting, or Progress messages. See also the ProgressIndicator2IP parameter, which if set to 1 or 8, the device behaves as if it received the ISDN messages with the PI. ✓ ISDN: Sending a 183 response depends on the ISDN PI. It is sent only if PI is set to 1 or 8 in the received Proceeding or Alerting messages. Sending 183 response also depends on the ReleaseIP2ISDNCallOnProgressWithCause parameter, which must be set to any value other than 2. ▪ See also the IgnoreAlertAfterEarlyMedia parameter. The parameter allows, for example, to interwork Alert with PI to SIP 183 with SDP instead of 180 with SDP. ▪ You can also configure early SIP 183 response immediately upon the receipt of an INVITE, using the EnableEarly183 parameter. ▪ The corresponding global parameter is EnableEarlyMedia.
Progress Indicator to IP `prog-ind-to-ip` [IpProfile_ProgressIndicator2IP]	Defines the progress indicator (PI) sent to the IP. ▪ [-1] = (Default) Not configured: ✓ Digital ISDN: The PI received in ISDN Proceeding, Progress, and Alerting messages is used, as described in the options below. ▪ [0] No PI = ✓ Digital: For IP-to-Tel calls, the device sends 180 Ringing response to the IP after receiving an ISDN Alerting. ▪ [1] PI = 1: ✓ Digital: For IP-to-Tel calls, if the parameter EnableEarlyMedia is set to 1, the device sends 180 Ringing with SDP in response to an ISDN Alerting or it sends a 183 Session Progress message with SDP in response to only the first received ISDN Proceeding or

Parameter	Description
	Progress message after a call is placed to PBX/PSTN over the trunk. ▪ [8] PI = 8: same as PI = 1. **Note:** The corresponding global parameter is ProgressIndicator2IP.
Early 183 `enable-early-183` [IpProfile_EnableEarly183]	Enables the device to send SIP 183 responses with SDP to the IP upon receipt of INVITE messages. The parameter is applicable to IP-to-Tel (ISDN) calls and applies to all calls. ▪ [0] Disable (default) ▪ [1] Enable = By sending the 183 response, the device opens an RTP channel before receiving the "progress" tone from the ISDN side. The device sends RTP packets immediately upon receipt of an ISDN Progress, Alerting with Progress indicator, or Connect message according to the initial negotiation without sending the 183 response again, thereby saving response time and avoiding early media clipping. **Notes:** ▪ To enable this feature, set the EnableEarlyMedia parameter to 1. ▪ When the BChannelNegotiation parameter is set to Preferred or Any, the EnableEarly183 parameter is ignored and a SIP 183 is not sent upon receipt of an INVITE. In such a case, you can set the ProgressIndicator2IP parameter to 1 (PI = 1) for the device to send a SIP 183 upon receipt of an ISDN Call Proceeding message. ▪ The corresponding global parameter is EnableEarly183.
Early Answer Timeout early-answer-timeout [IpProfile_EarlyAnswerTimeout]	Defines the duration (in seconds) that the device waits for an ISDN Connect message from the called party (Tel side), started from when it sends a Setup message. If this timer expires, the call is answered by sending a SIP 200 OK message (to the IP side). The valid range is 0 to 2400. The default is 0 (i.e., disabled). **Note:** The corresponding global parameter is EarlyAnswerTimeout.
Hold `enable-hold` [IpProfile_EnableHold]	Enables the interworking of the Hold/Retrieve supplementary service from ISDN to SIP. ▪ [0] Disable ▪ [1] Enable (default) **Notes:** ▪ To interwork the Hold/Retrieve supplementary service from SIP to ISDN (Euro ISDN), set the EnableHold2ISDN parameter to 1. ▪ The corresponding global parameter is EnableHold.
Add IE In Setup `add-ie-in-setup` [IpProfile_AddIEInSetup]	Defines an optional Information Element (IE) data (in hex format) which is added to ISDN Setup messages. For example, to add IE '0x20,0x02,0x00,0xe1', enter the value "200200e1". **Notes:** ▪ This IE is sent from the Trunk Group IDs that are defined by the SendIEonTG parameter . ▪ You can configure different IE data for Trunk Groups by configuring the parameter for different IP Profiles and then

Parameter	Description
	assigning the required IP Profile in the IP to Trunk Group Routing table (PSTNPrefix). ▪ The feature is similar to that of the EnableISDNTunnelingIP2Tel parameter. If both parameters are configured, the EnableISDNTunnelingIP2Tel parameter takes precedence. ▪ The corresponding global parameter is AddIEinSetup.
QSIG Tunneling enable-qsig-tunneling [IpProfile_EnableQSIGTunneling]	Enables QSIG tunneling-over-SIP for this SIP entity. This is according to IETF Internet-Draft draft-elwell-sipping-qsig-tunnel-03 and ECMA-355 and ETSI TS 102 345. ▪ [0] Disable (default). ▪ [1] Enable = Enables QSIG tunneling from QSIG to SIP, and vice versa. All QSIG messages are sent as raw data in corresponding SIP messages using a dedicated message body. **Notes:** ▪ QSIG tunneling must be enabled on originating and terminating devices. ▪ To enable this function, set the ISDNDuplicateQ931BuffMode parameter to 128 (i.e., duplicate all messages). ▪ To define the format of encapsulated QSIG messages, use the QSIGTunnelingMode parameter. ▪ Tunneling according to ECMA-355 is applicable to all ISDN variants (in addition to the QSIG protocol). ▪ For more information on QSIG tunneling, see QSIG Tunneling. ▪ The corresponding global parameter is EnableQSIGTunneling.
Copy Destination Number to Redirect Number `copy-dst-to-redirect-number` [IpProfile_CopyDest2RedirectNumber]	Enables the device to copy the called number, received in the SIP INVITE message, to the redirect number in the outgoing Q.931 Setup message, for IP-to-Tel calls. Thus, even if there is no SIP Diversion or History header in the incoming INVITE message, the outgoing Q.931 Setup message will contain a redirect number. ▪ [0] Disable (default). ▪ [1] After Manipulation = Copies the called number after manipulation. The device first performs IP-to-Tel destination phone number manipulation, and only then copies the manipulated called number to the redirect number sent in the Q.931 Setup message to the Tel. Thus, the called and redirect numbers are the same. ▪ [2] Before Manipulation = Copies the called number before manipulation. The device first copies the original called number to the SIP Diversion header, and then performs IP-to-Tel destination phone number manipulation. Thus, the called (i.e., SIP To header) and redirect (i.e., SIP Diversion header) numbers are different. **Note:** The corresponding global parameter is CopyDest2RedirectNumber.
Number of Calls Limit `call-limit` [IpProfile_CallLimit]	Defines the maximum number of concurrent calls (incoming and outgoing) for the SIP entity associated with the IP Profile. If the

Parameter	Description
	number of concurrent calls reaches this limit, the device rejects any new incoming and outgoing calls belonging to this IP Profile. The parameter can also be set to the following: ▪ [-1] = (Default) No limitation on calls. ▪ [0] = All calls are rejected.
SBC Signaling Tab	
PRACK Mode `sbc-prack-mode` [IpProfile_SbcPrackMode]	Defines the device's handling of SIP PRACK messages for the SIP entity associated with the IP Profile. ▪ [1] Optional = PRACK is optional. If required, the device performs the PRACK process on behalf of the SIP entity. ▪ [2] Mandatory = PRACK is required for this SIP entity. Calls from endpoints that do not support PRACK are rejected. Calls destined to these endpoints are also required to support PRACK. ▪ [3] Transparent (default) = The device does not intervene with the PRACK process and forwards the request as is.
P-Asserted-Identity Header Mode `sbc-assert-identity` [IpProfile_SBCAssertIdentity]	Defines the device's handling of the SIP P-Asserted-Identity header for the SIP entity associated with the IP Profile. This header indicates how the outgoing SIP message asserts identity. ▪ [0] As Is = (Default) P-Asserted Identity header is not affected and the device uses the same P-Asserted-Identity header (if present) in the incoming message for the outgoing message. ▪ [1] Add = Adds a P-Asserted-Identity header. The header's values are taken from the source URL. ▪ [2] Remove = Removes the P-Asserted-Identity header. **Notes:** ▪ The parameter affects only the initial INVITE request. ▪ The corresponding global parameter is SBCAssertIdentity.
Diversion Header Mode `sbc-diversion-mode` [IpProfile_SBCDiversionMode]	Defines the device’s handling of the SIP Diversion header for the SIP entity associated with the IP Profile. ▪ [0] As Is = (Default) Diversion header is not handled. ▪ [1] Add = History-Info header is converted to a Diversion header. ▪ [2] Remove = Removes the Diversion header and the conversion to the History-Info header depends on the SBCHistoryInfoMode parameter. For more information on interworking of the History-Info and Diversion headers, see Interworking SIP Diversion and History-Info Headers on page 546. **Note:** If the Diversion header is used, you can specify the URI type (e.g., "tel:") to use in the header, using the SBCDiversionUriType parameter.
History-Info Header Mode `sbc-history-info-mode` [IpProfile_SBCHistoryInfoMode]	Defines the device’s handling of the SIP History-Info header for the SIP entity associated with the IP Profile. ▪ [0] As Is = (Default) History-Info header is not handled. ▪ [1] Add = Diversion header is converted to a History-Info header. ▪ [2] Remove = History-Info header is removed from the SIP dialog and the conversion to the Diversion header depends on the SBCDiversionMode parameter.

Parameter	Description
	For more information on interworking of the History-Info and Diversion headers, see Interworking SIP Diversion and History-Info Headers on page 546.
Session Expires Mode `sbc-session-expires-mode` [IpProfile_SBCSessionExpiresMode]	Defines the required session expires mode for the SIP entity associated with the IP Profile. - [0] Transparent = (Default) The device does not interfere with the session expires negotiation. - [1] Observer = If the SIP Session-Expires header is present, the device does not interfere, but maintains an independent timer for each leg to monitor the session. If the session is not refreshed on time, the device disconnects the call. - [2] Not Supported = The device does not allow a session timer with this SIP entity. - [3] Supported = The device enables the session timer with this SIP entity. If the incoming SIP message does not include any session timers, the device adds the session timer information to the sent message. You can configure the value of the Session-Expires and Min-SE headers, using the SBCSessionExpires and SBCMinSE parameters, respectively.
Remote Update Support `sbc-rmt-update-supp` [IpProfile_SBCRemoteUpdateSupport]	Defines whether the SIP UPDATE message is supported by the SIP entity associated with the IP Profile. - [0] Not Supported = UPDATE message is not supported. - [1] Supported Only After Connect = UPDATE message is supported only after the call is connected. - [2] Supported = (Default) UPDATE message is supported during call setup and after call establishment.
Remote re-INVITE `sbc-rmt-re-invite-supp` [IpProfile_SBCRemoteReinviteSupport]	Defines whether the destination UA of the re-INVITE request supports re-INVITE messages and if so, whether it supports re-INVITE with or without SDP. - [0] Not Supported = re-INVITE is not supported and the device does not forward re-INVITE requests. The device sends a SIP response to the re-INVITE request, which can either be a success or a failure, depending on whether the device can bridge the media between the endpoints. - [1] Supported only with SDP = re-INVITE is supported, but only with SDP. If the incoming re-INVITE arrives without SDP, the device creates an SDP and adds it to the outgoing re-INVITE. - [2] Supported = (Default) re-INVITE is supported with or without SDP.
Remote Delayed Offer Support `sbc-rmt-delayed-offer` [IpProfile_SBCRemoteDelayedOfferSupport]	Defines whether the remote endpoint supports delayed offer (i.e., initial INVITEs without an SDP offer). - [0] Not Supported = Initial INVITE requests without SDP are not supported. - [1] Supported = (Default) Initial INVITE requests without SDP are supported.
User Registration Time `sbc-usr-reg-time`	Defines the registration time (in seconds) that the device responds to SIP REGISTER requests from users belonging to the SIP entity associated with the IP Profile. The registration

Parameter	Description
[IpProfile_SBCUserRegistrationTime]	time is inserted in the Expires header in the outgoing response sent to the user. The Expires header determines the lifespan of the registration. For example, a value of 3600 means that the registration will timeout in one hour and at that point, the user will not be able to make or receive calls. The valid range is 0 to 2,000,000. The default is 0. If configured to 0, the Expires header's value received in the user's REGISTER request remains unchanged. If no Expires header is received in the REGISTER message and the parameter is set to 0, the Expires header's value is set to 180 seconds, by default. **Note:** The corresponding global parameter is SBCUserRegistrationTime.
NAT UDP Registration Time `sbc-usr-udp-nat-reg-time` [IpProfile_SBCUserBehindUdpNATRegistrationTime]	Defines the registration time (in seconds) that the device includes in register responses, in response to SIP REGISTER requests from users belonging to the SIP entity associated with the IP Profile. The parameter applies only to users that are located behind NAT and whose communication type is UDP. The registration time is inserted in the Expires header in the outgoing response sent to the user. The Expires header determines the lifespan of the registration. For example, a value of 3600 means that the registration will timeout in one hour, unless the user sends a refresh REGISTER before the timeout. Upon timeout, the device removes the user's details from the registration database, and the user will not be able to make or receive calls through the device. The valid value is 0 to 2,000,000. If configured to 0, the Expires header's value received in the user's REGISTER request remains unchanged. By default, no value is defined (-1). **Note:** If the parameter is not configured, the registration time is according to the global parameter SBCUserRegistrationTime or IP Profile parameter IpProfile_SBCUserRegistrationTime.
NAT TCP Registration Time `sbc-usr-tcp-nat-reg-time` [IpProfile_SBCUserBehindTcpNATRegistrationTime]	Defines the registration time (in seconds) that the device includes in register responses, in response to SIP REGISTER requests from users belonging to the SIP entity associated with the IP Profile. The parameter applies only to users that are located behind NAT and whose communication type is TCP. The registration time is inserted in the Expires header in the outgoing response sent to the user. The Expires header determines the lifespan of the registration. For example, a value of 3600 means that the registration will timeout in one hour, unless the user sends a refresh REGISTER before the timeout. Upon timeout, the device removes the user's details from the registration database, and the user will not be able to make or receive calls through the device. The valid value is 0 to 2,000,000. If configured to 0, the Expires header's value received in the user's REGISTER request remains unchanged. By default, no value is defined (-1). **Note:** If the parameter is not configured, the registration time is according to the global parameter SBCUserRegistrationTime or IP Profile parameter IpProfile_SBCUserRegistrationTime.

Parameter	Description
Remote REFER Mode `sbc-rmt-refer-behavior` [IpProfile_SBCRemoteReferBehavior]	Defines the device's handling of REFER requests for the SIP entity associated with the IP Profile. ▪ [0] Regular = (Default) Refer-To header is unchanged and the device forwards the REFER as is. ▪ [1] Database URL = Changes the Refer-To header so that the re-routed INVITE is sent through the SBC: **a.** Before forwarding the REFER request, the device changes the host part to the device's IP address and adds a special prefix ("T~&R_") to the Contact user part. **b.** The incoming INVITE is identified as a REFER-resultant INVITE according to this special prefix. **c.** The device replaces the host part in the Request-URI with the host from the REFER contact. The special prefix remains in the user part for regular classification, manipulation, and routing. The special prefix can also be used for specific routing rules for REFER-resultant INVITEs. **d.** The special prefix is removed before the resultant INVITE is sent to the destination. ▪ [2] IP Group Name = Sets the host part in the REFER message to the name defined for the IP Group (in the IP Group table). ▪ [3] Handle Locally = Handles the incoming REFER request itself without forwarding the REFER. The device generates a new INVITE to the alternative destination according to the rules in the IP-to-IP Routing table (the 'Call Trigger' field must be set to REFER). **Note:** The corresponding global parameter is SBCReferBehavior.
Remote Replaces Mode `sbc-rmt-replaces-behavior` [IpProfile_SBCRemoteReplacesBehavior]	Enables the device to handle incoming INVITEs containing the Replaces header for the SIP entity (which does not support the header) associated with the IP Profile. The Replaces header is used to replace an existing SIP dialog with a new dialog such as in call transfer or call pickup. ▪ [0] Standard = (Default) The SIP entity supports INVITE messages containing Replaces headers. The device forwards the INVITE message containing the Replaces header to the SIP entity. The device may change the value of the Replaces header to reflect the call identifiers of the leg. ▪ [1] Handle Locally = The SIP entity does not support INVITE messages containing Replaces headers. The device terminates the received INVITE containing the Replaces header and establishes a new call between the SIP entity and the new call party. It then disconnects the call with the initial call party, by sending it a SIP BYE request. ▪ [2] Keep as is = The SIP entity supports INVITE messages containing Replaces headers. The device forwards the Replaces header as is in incoming REFER and outgoing INVITE messages from/to the SIP entity (i.e., Replaces header's value is unchanged). For example, assume that the device establishes a call between A and B. If B initiates a call transfer to C, the device receives an INVITE with the Replaces header from C. If A supports the Replaces header, the device simply forwards the INVITE as is to

Parameter	Description
	A; a new call is established between A and C and the call between A and B is disconnected. However, if A does not support the Replaces header, the device uses this feature to terminate the INVITE with Replaces header and handles the transfer for A. The device does this by connecting A to C, and disconnecting the call between A and B, by sending a SIP BYE request to B. Note that if media transcoding is required, the device sends an INVITE to C on behalf of A with a new SDP offer.
Play RBT To Transferee `sbc-play-rbt-to-xferee` [IpProfile_SBCPlayRBTToTransferee]	Enables the device to play a ringback tone to the transferred party (transferee) during a blind call transfer, for the SIP entity associated with the IP Profile (which does not support such a tone generation during call transfer). The ringback tone indicates to the transferee of the ringing of the transfer target (to where the transferee is being transferred). ▪ [0] No (Default) ▪ [1] Yes Typically, the transferee hears a ringback tone only if the transfer target sends it early media. However, if the transferee is put on-hold before being transferred, no ringback tone is heard. When this feature is enabled, the device generates a ringback tone to the transferee during call transfer in the following scenarios: ▪ Transfer target sends a SIP 180 (Ringing) to the device. ▪ For non-blind transfer, if the call is transferred while the transfer target is ringing and no early media occurs. ▪ The 'Remote Early Media RTP Behavior parameter is set to Delayed (used in the Lync environment), and transfer target sends a 183 Session Progress with SDP offer. If early media from the transfer target has already been detected, the transferee receives RTP stream from the transfer target. If it has not been detected, the device generates a ringback tone to the transferee and stops the tone generation once RTP has been detected from the transfer target. For any of these scenarios, if the transferee is put on-hold by the transferor, the device retrieves the transferee from hold, sends a re-INVITE if necessary, and then plays the ringback tone. **Note:** For the device to play the ringback tone, it must be loaded with a Prerecorded Tones (PRT) file. For more information, see Prerecorded Tones File on page 652.
Remote 3xx Mode `sbc-rmt-3xx-behavior` [IpProfile_SBCRemote3xxBehavior]	Defines the device's handling of SIP 3xx redirect responses for the SIP entity associated with the IP Profile. By default, the device's handling of SIP 3xx responses is to send the Contact header unchanged. However, some SIP entities may support different versions of the SIP 3xx standard while others may not even support SIP 3xx. When enabled, the device handles SIP redirections between different subnets (e.g., between LAN and WAN sides). This is required when the new address provided by the redirector (Redirect sever) may not be reachable by the far-end user (FEU) located in another subnet. For example, a far-end user (FEU) in the WAN sends a SIP request via the device to a Redirect server in the LAN, and the Redirect server replies with a SIP 3xx response to a PBX in the LAN in the Contact header. If the

Parameter	Description
	device sends this response as is (i.e., with the original Contact header), the FEU is unable to reach the new destination. ▪ [0] Transparent = (Default) The device forwards the received SIP 3xx response as is, without changing the Contact header (i.e.,transparent handling). ▪ [1] Database URL = The device changes the Contact header so that the re-route request is sent through the device. The device changes the URI in the Contact header of the received SIP 3xx response to its own URI and adds a special user prefix ("T~&R_”), which is then sent to the FEU. The FEU then sends a new INVITE to the device, which the device then sends to the correct destination. ▪ [2] Handle Locally = The device handles SIP 3xx responses on behalf of the dialog-initiating UA and retries the request (e.g., INVITE) using one or more alternative URIs included in the 3xx response. The device sends the new request to the alternative destination according to the IP-to-IP Routing table (the 'Call Trigger' field must be set to 3xx). **Notes:** ▪ When the parameter is changed from 1 to 0, new 3xx Contact headers remain unchanged. However, requests with the special prefix continue using the device's database to locate the new destination. ▪ Only one database entry is supported for the same host, port, and transport combination. For example, the following URLs cannot be distinguished by the device: ✓ sip:10.10.10.10:5060;transport=tcp;param=a ✓ sip:10.10.10.10:5060;transport=tcp;param=b ▪ The database entry expires two hours after the last use. ▪ The maximum number of destinations (i.e., database entries) is 50. ▪ The corresponding global parameter is SBC3xxBehavior.
Remote Early Media `sbc-rmt-early-media-supp` [IpProfile_SBCRemoteEarlyMediaSupport]	Defines whether the remote side can accept early media or not. ▪ [0] Not Supported = Early media is not supported. ▪ [1] Supported = (Default) Early media is supported.
Remote Multiple 18x `sbc-rmt-mltple-18x-supp` [IpProfile_SBCRemoteMultiple18xSupport]	Defines whether multiple 18x responses including 180 Ringing, 181 Call is Being Forwarded, 182 Call Queued, and 183 Session Progress are forwarded to the caller, for the SIP entity associated with the IP Profile. ▪ [0] Not Supported = Only the first 18x response is forwarded to the caller. ▪ [1] Supported = (Default) Multiple 18x responses are forwarded to the caller.
Remote Early Media Response Type `sbc-rmt-early-media-resp` [IpProfile_SBCRemoteEarlyMediaResponseType]	Defines the SIP provisional response type - 180 or 183 - for forwarding early media to the caller, for the SIP entity associated with the IP Profile. ▪ [0] Transparent = (Default) All early media response types are supported; the device forwards all responses as is (unchanged). ▪ [1] 180 = Early media is sent as 180 response only. ▪ [2] 183 = Early media is sent as 183 response only.

Parameter	Description
Remote Multiple Early Dialogs `sbc-multi-early-diag` [IpProfile_SBCRemoteMultipleEarlyDialogs]	Defines the device's handling of To-header tags in call forking responses (i.e., multiple SDP answers) sent to the SIP entity associated with the IP Profile. When the SIP entity initiates an INVITE that is subsequently forked (for example, by a proxy server) to multiple endpoints, the endpoints respond with a SIP 183 containing an SDP answer. Typically, each endpoint's response has a different To-header tag. For example, a call initiated by the SIP entity (100@A) is forked and two endpoints respond with ringing, each with a different tag: - Endpoint "tag 2": SIP/2.0 180 Ringing From: <sip:100@A>;tag=tag1 To: sip:200@B;tag=tag2 Call-ID: c2 - Endpoint "tag 3": SIP/2.0 180 Ringing From: <sip:100@A>;tag=tag1 To: sip:200@B;tag=tag3 Call-ID: c2 In non-standard behavior (when the parameter is configured to Disable), the device forwards all the SDP answers with the same tag. In the example, endpoint "tag 3" is sent with the same tag as endpoint "tag 2" (i.e., To: sip:200@B;tag=tag2). - [-1] According to Operation Mode = (Default) Depends on the setting of the 'Operation Mode' parameter in the IP Group or SRD table: ✓ B2BUA: Device operates as if the parameter is set to Disable [0]. ✓ Call State-full Proxy: Device operates as if the parameter is set to Enable [1]. In addition, the device preserves the From tags and Call-IDs of the endpoints in the SDP answer sent to the SIP entity. - [0] Disable = Device sends the multiple SDP answers with the same To-header tag, to the SIP entity. In other words, this option is relevant if the SIP entity does not support multiple dialogs (and multiple tags). However, non-standard, multiple answer support may still be configured by the SBCRemoteMultipleAnswersMode parameter. - [1] Enable = Device sends the multiple SDP answers with different To-header tags, to the SIP entity. In other words, the SIP entity supports standard multiple SDP answers (with different To-header tags). In this case, the SBCRemoteMultipleAnswersMode parameter is ignored. **Note:** If the parameter and the SBCRemoteMultipleAnswersMode parameter are disabled, multiple SDP answers are not reflected to the SIP entity (i.e., the device sends the same SDP answer in multiple 18x and 200 responses).
Remote Multiple Answers Mode `sbc-multi-answers` [IpProfile_SBCRemoteMultipleAnswersMode]	Enables interworking multiple SDP answers within the same SIP dialog (non-standard). The parameter enables the device to forward multiple answers to the SIP entity associated with the IP Profile. The parameter is applicable only when the IpProfile_SBCRemoteMultipleEarlyDialogs parameter is disabled.

Parameter	Description
	- [0] Disable = (Default) Device always sends the same SDP answer, which is based on the first received answer that it sent to the SIP entity, for all forked responses (even if 'Forking Handling Mode' is Sequential), and thus, may result in transcoding. - [1] Enable = If the 'Forking Handling Mode' parameter is configured to Sequential, the device sends multiple SDP answers.
Remote Early Media RTP Detection Mode `sbc-rmt-early-media-rtp` [IpProfile_SBCRemoteEarlyMediaRTP]	Defines whether the destination UA sends RTP immediately after it sends a 18x response. - [0] By Signaling = (Default) Remote client sends RTP immediately after it sends 18x response with early media. The device forwards 18x and RTP as is. - [1] By Media = After sending 18x response, the remote client waits before sending RTP (e.g., Microsoft Lync environment). For the device's handling of this remote UA support, see Interworking SIP Early Media on page 547.
Remote RFC 3960 Support `sbc-rmt-rfc3960-supp` [IpProfile_SBCRemoteSupportsRFC3960]	Defines whether the destination UA is capable of receiving 18x messages with delayed RTP. - [0] Not Supported = (Default) UA does not support receipt of 18x messages with delayed RTP. For the device's handling of this remote UA support, see Interworking SIP Early Media on page 547. - [1] Supported = UA is capable of receiving 18x messages with delayed RTP.
Remote Can Play Ringback `sbc-rmt-can-play-ringback` [IpProfile_SBCRemoteCanPlayRingback]	Defines whether the destination UA can play a local ringback tone. - [0] No = UA does not support local ringback tone. The device sends 18x with delayed SDP to the UA. - [1] Yes = (Default) UA supports local ringback tone. For the device's handling of this remote UA support, see Interworking SIP Early Media on page 547.
Reliable Held Tone Source `reliable-heldtone-source` [IPProfile_ReliableHoldToneSource]	Enables the device to consider the received call-hold request (re-INVITE/UPDATE) with SDP containing 'a=sendonly', as genuine. - [0] No = (Default) Even if the received SDP contains 'a=sendonly', the device plays a held tone to the held party. This is useful in cases where the initiator of the call hold does not support the generation of held tones. - [1] Yes = If the received SDP contains 'a=sendonly', the device does not play a held tone to the held party (and assumes that the initiator of the call hold plays the held tone). **Note:** The device plays a held tone only if the 'SBC Play Held Tone' parameter is set to Yes.
Play Held Tone `play-held-tone` [IpProfile_SBCPlayHeldTone]	Enables the device to play a held tone to the held party. This is useful if the held party does not support playing a local held tone, or for IP entities initiating call hold that do not support the generation of held tones. - [0] No (default) - [1] Yes

Parameter	Description
	Note: If the parameter is set to Yes, the device plays the tone only if the 'SBC Remote Hold Format' parameter is set to send-only, send only 0.0.0.0, or not supported.
Remote Hold Format `remote-hold-Format` [IPProfile_SBCRemoteHoldFormat]	Defines the format of the SDP in the re-INVITE for call hold that the device sends to the held party. ▪ [0] Transparent = (Default) Device forwards SDP as is. ▪ [1] Send Only = Device sends SDP with 'a=sendonly'. ▪ [2] Send Only Zero ip = Device sends SDP with 'a=sendonly' and 'c=0.0.0.0'. ▪ [3] Inactive = Device sends SDP with 'a=inactive'. ▪ [4] Inactive Zero ip = Device sends SDP with 'a=inactive' and 'c=0.0.0.0'. ▪ [5] Not Supported = Used when remote side cannot identify a call-hold message. The device terminates the received call-hold message (re-INVITE / UPDATE) and sends a 200 OK to the initiator of the call hold. The device plays a held tone to the held party if the 'SBC Play Held Tone' parameter is set to Yes.
Remote Representation Mode `sbc-rmt-rprsntation` [IpProfile_SBCRemoteRepresentationMode]	Enables interworking SIP in-dialog, Contact and Record-Route headers between SIP entities. The parameter defines the device's handling of in-dialog, Contact and Record-Route headers for messages sent to the SIP entity associated with the IP Profile. ▪ [-1] According to Operation Mode = (Default) Depends on the setting of the 'Operation Mode' parameter in the IP Group or SRD table: ✓ B2BUA: Device operates as if the parameter is set to Replace Contact [0]. ✓ Call State-full Proxy: Device operates as if the parameter is set to Add Routing Headers [1]. ▪ [0] Replace Contact = Device replaces the address in the Contact header, received in incoming messages from the other side, with its own address in the outgoing message sent to the SIP entity. ▪ [1] Add Routing Headers = Device adds a Record-Route header for itself to outgoing messages (requests\responses) sent to the SIP entity in dialog-setup transactions. The Contact header remains unchanged. ▪ [2] Transparent = Device doesn't change the Contact header and doesn't add a Record-Route header for itself. Instead, it relies on its' own inherent mechanism to remain in the route of future requests in the dialog (for example, relying on the way the endpoints are set up or on TLS as the transport type).
Keep Incoming Via Headers `sbc-keep-via-headers` [IpProfile_SBCKeepVIAHeaders]	Enables interworking SIP Via headers between SIP entities. The parameter defines the device's handling of Via headers for messages sent to the SIP entity associated with the IP Profile. ▪ [-1] According to Operation Mode = Depends on the setting of the 'Operation Mode' parameter in the IP Group table or SRD table: ✓ B2BUA: Device operates as if the parameter is set to Disable [0].

Parameter	Description
	✓ Call State-full Proxy: Device operates as if the parameter is set to Enable [1]. ▪ [0] Disable = Device removes all Via headers received in the incoming SIP request from the other leg and adds a Via header identifying only itself, in the outgoing message sent to the SIP entity. ▪ [1] Enable = Device retains the Via headers received in the incoming SIP request and adds itself as the top-most listed Via header in the outgoing message sent to the SIP entity.
Keep Incoming Routing Headers `sbc-keep-routing-headers` [IpProfile_SBCKeepRoutingHeaders]	Enables interworking SIP Record-Route headers between SIP entities. The parameter defines the device's handling of Record-Route headers for request/response messages sent to the the SIP entity associated with the IP Profile. ▪ [-1] According to Operation Mode = (Default) Depends on the setting of the 'Operation Mode' in the IP Group or SRD table: ✓ B2BUA: Device operates as if the parameter is set to Disable [0]. ✓ Call State-full Proxy: Device operates as if the parameter is set to Enable [1]. ▪ [0] Disable = Device removes the Record-Route headers received in requests and responses from the other side, in the outgoing SIP message sent to the SIP entity. The device creates a route set for that side of the dialog based on these headers, but doesn't send them to the SIP entity. ▪ [1] Enable = Device retains the incoming Record-Route headers received in requests and non-failure responses from the other side, in the following scenarios: ✓ The message is part of a SIP dialog-setup transaction. ✓ The messages in the setup and previous transaction didn't include the Record-Route header, and therefore hadn't set the route set. Note: Record-Routes are kept only for SIP INVITE, UPDATE, SUBSCRIBE and REFER messages.
Keep User-Agent Header `sbc-keep-user-agent` [IpProfile_SBCKeepUserAgentHeader]	Enables interworking SIP User-Agent headers between SIP entities. The parameter defines the device's handling of User-Agent headers for response/request messages sent to the SIP entity associated with the IP Profile. ▪ [-1] According to Operation Mode = (Default) Depends on the setting of the 'Operation Mode' parameter in the IP Group or SRD table: ✓ B2BUA: Device operates as if this parameter is set to Disable [0]. ✓ Call State-full Proxy: Device operates as if this parameter is set to Enable [1]. ▪ [0] Disable = Device removes the User-Agent/Server headers received in the incoming message from the other side, and adds its' own User-Agent header in the outgoing message sent to the SIP entity. ▪ [1] Enable = Device retains the User-Agent/Server headers received in the incoming message and sends the headers as is in the outgoing message to the SIP entity.
Handle X-Detect `sbc-handle-xdetect`	Enables the detection and notification of events (AMD, CPT, and fax), using the X-Detect SIP header.

Parameter	Description
[IpProfile_SBCHandleXDetect]	▪ [0] No (default) ▪ [1] Yes For more information, see Event Detection and Notification using X-Detect Header on page 193.
SBC Media Tab	
Allowed Coders `sbc-allowed-coders-group-id` [IpProfile_SBCAllowedCodersGroupID]	Assigns an Allowed Audio Coders Group. This defines audio (voice) coders that can be used for the SIP entity associated with the IP Profile. To configure Allowed Audio Coder Groups, see Configuring Allowed Audio Coder Groups on page 565. For a description of the Allowed Coders feature, see "Restricting Coders" on page 540.
Allowed Coders Mode `sbc-allowed-coders-mode` [IpProfile_SBCAllowedCodersMode]	Defines the mode of the Allowed Coders feature for the SIP entity associated with the IP Profile. ▪ [0] Restriction = In the incoming SDP offer, the device uses only Allowed coders; the rest are removed from the SDP offer (i.e., only coders common between those in the received SDP offer and the Allowed coders are used). ▪ [1] Preference = The device re-arranges the priority (order) of the coders in the incoming SDP offer according to their order of appearance in the Allowed Coders Group or Allowed Video Coders. The coders in the original SDP offer are listed after the Allowed coders. ▪ [2] Restriction and Preference = Performs both Restriction and Preference. **Notes:** ▪ The parameter is applicable only if Allowed coders are assigned to the IP Profile (using the 'Allowed Coders' or 'Allowed Video Coders Group ID' parameters). ▪ For more information on the Allowed Coders feature, see Restricting Coders on page 540.
Allowed Video Coders `sbc-allowed-video-coders-group-id` [IPProfile_SBCAllowedVideoCodersGroupID]	Assigns an Allowed Video Coders Group. This defines permitted video coders when forwarding video streams to the SIP entity associated with the IP Profile. The video coders are listed in the "video" media type in the SDP (i.e., 'm=video' line). For this SIP entity, the device uses only video coders that appear in both the SDP offer and the Allowed Video Coders Group ID. By default, no Allowed Video Coders Group is assigned (i.e., all video coders are allowed). To configure Allowed Video Coders Groups, see Configuring Allowed Video Coder Groups on page 566.
Allowed Media Types `sbc-allowed-media-types` [IpProfile_SBCAllowedMediaTypes]	Defines media types permitted for the SIP entity associated with the IP Profile. The media type appears in the SDP 'm=' line (e.g., 'm=audio'). The device permits only media types that appear in both the SDP offer and this configured list. If no common media types exist between the SDP offer and this list, the device drops the call. The valid value is a string of up to 64 characters. To configure multiple media types, separate the strings with a comma, e.g., "audio, text" (without quotes). By default, no media types are configured (i.e., all media types are permitted).

Parameter	Description
SBC Media Security Mode `sbc-media-security-behaviour` [IpProfile_SBCMediaSecurityBehaviour]	Defines the handling of RTP and SRTP for the SIP entity associated with the IP Profile. ▪ [0] As is = (Default) No special handling for RTP\SRTP is done. ▪ [1] SRTP = SBC legs negotiate only SRTP media lines, and RTP media lines are removed from the incoming SDP offer-answer. ▪ [2] RTP = SBC legs negotiate only RTP media lines, and SRTP media lines are removed from the incoming offer-answer. ▪ [3] Both = Each offer-answer is extended (if not already) to two media lines - one RTP and the other SRTP. If two SBC legs (after offer-answer negotiation) use different security types (i.e., one RTP and the other SRTP), the device performs RTP-SRTP transcoding. To transcode between RTP and SRTP, the following prerequisites must be met: ▪ At least one supported SDP "crypto" attribute and parameters. ▪ EnableMediaSecurity must be set to 1. If one of the above transcoding prerequisites is not met, then: ▪ any value other than “As is” is discarded. ▪ if the incoming offer is SRTP, force transcoding, coder transcoding, and DTMF extensions are not applied.
Media Security Method `sbc-media-security-method` [IpProfile_SBCMediaSecurityMethod]	Defines the media security protocol for SRTP, for the SIP entity associated with the IP Profile. ▪ [0] SDES = (Default) The device secures RTP using the Session Description Protocol Security Descriptions (SDES) protocol to negotiate the cryptographic keys (RFC 4568). The keys are sent in the SDP body ('a=crypto') of the SIP message and are typically secured using SIP over TLS (SIPS). The encryption of the keys is in plain text in the SDP. SDES implements TLS over TCP. ▪ [1] DTLS = The device uses Datagram Transport Layer Security (DTLS) protocol to secure UDP-based media streams (RFCs 5763 and 5764). For more information on DTLS, see SRTP using DTLS Protocol on page 204. **Notes:** ▪ To support DTLS, you must also configure the following for the SIP entity: ✓ TLS Context for DTLS (see Configuring TLS Certificate Contexts on page 107). The server cipher ('Cipher Server') must be configured to All. ✓ IpProfile_SBCMediaSecurityBehaviourMedia configured to SRTP or Both. ✓ IpProfile_SBCRTCPMux configured to Supported. The setting is required as the DTLS handshake is done for the port used for RTP. Therefore, RTCP and RTP should be multiplexed over the same port. ▪ The device does not support forwarding of DTLS transparently between endpoints (SIP entities).
Enforce MKI Size `sbc-enforce-mki-size`	Enables negotiation of the Master Key Identifier (MKI) length for SRTP-to-SRTP flows between SIP networks (i.e., IP Groups). This includes the capability of modifying the MKI length on the

Parameter	Description
[IpProfile_SBCEnforceMKISize]	inbound or outbound SBC call leg for the SIP entity associated with the IP Profile. ▪ [0] Don't enforce = (Default) Device forwards the MKI size as is. ▪ [1] Enforce = Device changes the MKI length according to the settings of the IP Profile parameter, MKISize.
SDP Remove Crypto LifeTime sbc-sdp-remove-crypto-lifetime [IpProfile_SBCRemoveCryptoLifetimeInSDP]	Defines the handling of the lifetime field in the 'a=crypto' attribute of the SDP for the SIP entity associated with the IP Profile. The SDP field defines the lifetime of the master key as measured in maximum number of SRTP or SRTCP packets using the master key. ▪ [0] No = (Default) The device retains the lifetime field (if present) in the SDP. ▪ [1] Yes = The device removes the lifetime field from the 'a=crypto' attribute. **Note:** If you configure the parameter to Yes, the following IP Profile parameters must be configured as follows: ▪ IpProfile_EnableSymmetricMKI configured to Enable [1]. ▪ IpProfile_MKISize configured to 0. ▪ IpProfile_SBCEnforceMKISize configured to Enforce [1].
RFC 2833 DTMF Payload Type sbc-2833dtmf-payload [IpProfile_SBC2833DTMFPayloadType]	Defines the payload type of DTMF digits for the SIP entity associated with the IP Profile. This enables the interworking of the DTMF payload type for RFC 2833 between different SBC call legs. For example, if two entities require different DTMF payload types, the SDP offer received by the device from one entity is forwarded to the destination entity with its payload type replaced with the configured payload type, and vice versa. The value range is 0 to 200. The default is 0 (i.e., the device forwards the received payload type as is).
Adapt RFC2833 BW to Voice coder BWsbc-adapt-rfc2833-bw-voice-bw [IpProfile_SBCAdaptRFC2833BWToVoiceCoderBW]	Defines the 'telephone-event' type (8000 or 16000) in the SDP that the device sends in the outgoing SIP 200 OK message for DTMF payload negotiation (sampling rate). ▪ [0] Disable = (Default) The device always sends the 'telephone-event' as 8000 in the outgoing SIP 200 OK, even if the SDP of the incoming INVITE contains multiple telephone-event types (e.g., 8000 and 16000). ▪ [1] Enable = The type of 'telephone-event' that the device sends in the outgoing SIP 200 OK message is according to the coder type (narrowband or wideband). If narrowband, it sends the 'telephone-event' as 8000; if wideband, it sends it as 16000. An example when the parameter is configured to **Enable** is shown below, whereby the 'telephone-event' is "16000" in the outgoing message due to the wideband coder: **SDP in incoming INVITE:**

Parameter	Description

	a=rtpmap:97 AMR-WB/16000/1
	a=fmtp:97 mode-change-capability=2
	a=rtpmap:98 AMR-WB/16000/1
	a=fmtp:98 octet-align=1; mode-change-capability=2
	a=rtpmap:100 AMR/8000/1
	a=fmtp:100 mode-change-capability=2
	a=rtpmap:99 telephone-event/16000/1
	a=fmtp:99 0-15
	a=rtpmap:102 telephone-event/8000/1
	a=fmtp:102 0-15
	SDP in outgoing 200 OK:
	m=audio 6370 RTP/AVP 97 99
	a=rtpmap:99 telephone-event/**16000**/1
	a=fmtp:99 0-15
	a=sendrecv
	a=ptime:20
	a=maxptime:120
	a=rtpmap:97 AMR-WB/16000
	a=fmtp:97 mode-change-capability=2;mode-set=0,1,2,3,4,5,6,7,
	Note: The parameter is applicable only to the SBC application.

SDP Ptime Answer `sbc-sdp-ptime-ans` [IpProfile_SBCSDPPtimeAnswer]	Defines the packetization time (ptime) of the coder in RTP packets for the SIP entity associated with the IP Profile. This is useful when implementing transrating. ▪ [0] Remote Answer = (Default) Use ptime according to SDP answer. ▪ [1] Original Offer = Use ptime according to SDP offer. ▪ [2] Preferred Value= Use the ptime according to the 'Preferred Ptime' parameter (see below) if it is configured to a non-zero value. **Note:** Regardless of the settings of this parameter, if a non-zero value is configured for the 'Preferred Ptime' parameter (see below), it is used as the ptime in the SDP offer.
Preferred Ptime `sbc-preferred-ptime` [IpProfile_SBCPreferredPTime]	Defines the packetization time (ptime) in msec for the SIP entity associated with the IP Profile, in the outgoing SDP offer. If the 'SDP Ptime Answer' parameter (see above) is configured to **Preferred Value** [2] and the 'Preferred Ptime' parameter is configured to a non-zero value, the configured ptime is used (enabling ptime transrating if the other side uses a different ptime). If the 'SDP Ptime Answer' parameter is configured to **Remote Answer** [0] or **Original Offer** [1] and the 'Preferred Ptime' parameter is configured to a non-zero value, the configured value is used as the ptime in the SDP offer. The valid range is 0 to 200. The default is 0 (i.e., a preferred ptime is not used).
RTP Redundancy Mode `sbc-rtp-red-behav` [IpProfile_SBCRTPRedundancyBehavior]	Enables interworking RTP redundancy negotiation support between SIP entities in the SDP offer-answer exchange (according to RFC 2198). The parameter defines the device's handling of RTP redundancy for the SIP entity associated with the IP Profile. According to the RTP redundancy SDP offer/answer negotiation, the device uses or discards the RTP redundancy packets. The parameter enables asymmetric RTP

Parameter	Description
	redundancy, whereby the device can transmit and receive RTP redundancy packets to and from a specific SIP entity, while transmitting and receiving regular RTP packets (no redundancy) for the other SIP entity involved in the voice path. The device can identify the RTP redundancy payload type in the SDP for indicating that the RTP packet stream includes redundant packets. RTP redundancy is indicated in SDP using the "red" coder type, for example: a=rtpmap:<payload type> red/8000/1 RTP redundancy is useful when there is packet loss; the missing information may be reconstructed at the receiver side from the redundant packets. ▪ [0] As Is = (Default) The device does not interfere in the RTP redundancy negotiation and forwards the SDP offer/answer (incoming and outgoing calls) as is without interfering in the RTP redundancy negotiation. ▪ [1] Enable = The device always adds RTP redundancy capabilities in the outgoing SDP offer sent to the SIP entity. Whether RTP redundancy is implemented depends on the subsequent incoming SDP answer from the SIP entity. The device does not modify the incoming SDP offer received from the SIP entity, but if RTP redundancy is required, it will be supported. Select the option if the SIP entity requires RTP redundancy. ▪ [2] Disable = The device removes the RTP redundancy payload (if present) from the SDP offer/answer for calls received from or sent to the SIP entity. Select the option if the SIP entity does not support RTP redundancy. **Notes:** ▪ To enable the device to generate RFC 2198 redundant packets, use the IPProfile_RTPRedundancyDepth parameter. ▪ To configure the payload type in the SDP offer for RTP redundancy, use the RFC2198PayloadType.
RTCP Mode `sbc-rtcp-mode` [IPProfile_SBCRTCPMode]	Defines how the device handles RTCP packets during call sessions for the SIP entity associated with the IP Profile. This is useful for interworking RTCP between SIP entities. For example, this may be necessary when incoming RTCP is not compatible with the destination SIP entity's (this IP Profile) RTCP support. In such a scenario, the device can generate the RTCP and send it to the SIP entity. ▪ [0] Transparent = (Default) RTCP is forwarded as is. ▪ [1] Generate Always = Generates RTCP packets during active and inactive (e.g., during call hold) RTP periods (i.e., media is 'a=recvonly' or 'a=inactive' in the INVITE SDP). ▪ [2] Generate only if RTP Active = Generates RTCP packets only during active RTP periods. In other words, the device does not generate RTCP when there is no RTP traffic (such as when a call is on hold). **Note:** The corresponding global parameter is SBCRTCPMode.
Jitter Compensation `sbc-jitter-compensation`	Enables the on-demand jitter buffer for SBC calls. The jitter buffer is useful when incoming packets are received at inconsistent intervals (i.e., packet delay variation). The jitter

Parameter	Description
[IpProfile_SBCJitterCompensation]	buffer stores the packets and sends them out at a constant rate (according to the coder's settings). - [0] Disable (default) - [1] Enable **Note:** The jitter buffer parameters, 'Dynamic Jitter Buffer Minimum Delay' (DJBufMinDelay) and 'Dynamic Jitter Buffer Optimization Factor' (DJBufOptFactor) can be used to configure minimum packet delay only when transcoding is employed.
ICE Mode `ice-mode` [IPProfile_SBCIceMode]	Enables Interactive Connectivity Establishment (ICE) Lite for the SIP entity associated with the IP Profile. ICE is a methodology for NAT traversal, employing the Session Traversal Utilities for NAT (STUN) and Traversal Using Relays around NAT (TURN) protocols to provide a peer with a public IP address and port that can be used to connect to a remote peer. - [0] None (default) - [1] Lite For more information on ICE Lite, see ICE Lite.
SDP Handle RTCP `sbc-sdp-handle-rtcp` [IpProfile_SBCSDPHandleRTCPAttribute]	Enables the interworking of the RTCP attribute, 'a=rtcp' (RTCP) in the SDP, for the SIP entity associated with the IP Profile. The RTCP attribute is used to indicate the RTCP port for media when that port is not the next higher port number following the RTP port specified in the media line ('m='). The parameter is useful for SIP entities that either require the attribute or do not support the attribute. For example, Google Chrome and Web RTC do not accept calls without the RTCP attribute in the SDP. In Web RTC, Chrome (SDES) generates the SDP with 'a=rtcp', for example: m=audio 49170 RTP/AVP 0 a=rtcp:53020 IN IP6 2001:2345:6789:ABCD:EF01:2345:6789:ABCD - [0] Don't Care = (Default) The device forwards the SDP as is without interfering in the RTCP attribute (regardless if present or not). - [1] Add = The device adds the 'a=rtcp' attribute to the outgoing SDP offer sent to the SIP entity if the attribute was not present in the original incoming SDP offer. - [2] Remove = The device removes the 'a=rtcp' attribute, if present in the incoming SDP offer received from the other SIP entity, before sending the outgoing SDP offer to the SIP entity.
RTCP Mux `sbc-rtcp-mux` [IPProfile_SBCRTCPMux]	Enables interworking of multiplexing of RTP and RTCP onto a single local port, between SIP entities. The parameter enables multiplexing of RTP and RTCP traffic onto a single local port, for the SIP entity associated with the IP Profile. Multiplexing of RTP data packets and RTCP packets onto a single local UDP port is done for each RTP session (according to RFC 5761). If multiplexing is not enabled, the device uses different (but adjacent) ports for RTP and RTCP packets. With the increased use of NAT and firewalls, maintaining multiple NAT bindings can be costly and also complicate firewall administration since multiple ports must be opened to allow RTP traffic. To reduce these costs and session setup times, support

Parameter	Description
	for multiplexing RTP data packets and RTCP packets onto a single port is advantageous. For multiplexing, the initial SDP offer must include the "a=rtcp-mux" attribute to request multiplexing of RTP and RTCP onto a single port. If the SDP answer wishes to multiplex RTP and RTCP, it must also include the "a=rtcp-mux" attribute. If the answer does not include the attribute, the offerer must not multiplex RTP and RTCP packets. If both ICE and multiplexed RTP-RTCP are used, the initial SDP offer must also include the "a=candidate:" attribute for both RTP and RTCP along with the "a=rtcp:" attribute, indicating a fallback port for RTCP in case the answerer does not support RTP and RTCP multiplexing. ▪ [0] Not Supported = (Default) RTP and RTCP packets use different ports. ▪ [1] Supported = Device multiplexes RTP and RTCP packets onto a single port.
RTCP Feedback sbc-rtcp-feedback [IPProfile_SBCRTCPFeedback]	Enables RTCP-based feedback indication in outgoing SDPs sent to the SIP entity associated with the IP Profile. The parameter supports indication of RTCP-based feedback, according to RFC 5124, during RTP profile negotiation between two communicating SIP entities. RFC 5124 defines an RTP profile (S)AVPF for (secure) real-time communications to provide timely feedback from the receivers to a sender. For more information on RFC 5124, see http://tools.ietf.org/html/rfc5124. Some SIP entities may require RTP secure-profile feedback negotiation (AVPF/SAVPF) in the SDP offer/answer exchange, while other SIP entities may not support it. The device indicates whether or not feedback is supported on behalf of the SIP entity. It does this by adding an "F" or removing the "F" from the SDP media line ('m=') for AVP and SAVP. For example, the following shows "AVP" appended with an "F", indicating that the SIP entity is capable of receiving feedback `m=audio 49170 RTP/SAVPF 0 96` ▪ [0] Disable = (Default) The device does not send the feedback flag ("F") in SDP offers/answers that are sent to the SIP entity. If the SDP 'm=' attribute of an incoming message that is destined to the SIP entity includes the feedback flag, the device removes it before sending the message to the SIP entity. ▪ [1] Enable = The device includes the feedback flag ("F") in the SDP offer that is sent to the SIP entity. The device includes the feedback flag in the SDP answer sent to the SIP entity only if it was present in the SDP offer received from the other SIP entity.
Direct Media Tag `sbc-dm-tag` [IPProfile_SBCDirectMediaTag]	Defines an identification tag for enabling direct media (no Media Anchoring) for the SIP entity associated with the IP Profile. Direct media occurs between all endpoints whose IP Profiles have the same tag value (non-empty value). For example, if you set the parameter to "direct-rtp" for two IP Profiles "IP-PBX-1" and "IP-PBX-2", the device employs direct media for calls amongst endpoints associated with IP Profile "IP-PBX-1", for calls amongst endpoints associated with IP Profile "IP-PBX-2", and for calls between endpoints associated with IP Profile "IP-PBX-1" and IP Profile "IP-PBX-2".

Parameter	Description
	The valid value is a string of up to 16 characters. By default, no value is defined. For more information on direct media, see Direct Media on page 538. **Note:** If you enable direct media for the IP Profile, make sure that your Media Realm provides sufficient ports, as media may traverse the device for mid-call services (e.g., call transfer).

Part V

Gateway Application

20 Introduction

This section describes configuration of the Gateway application. The Gateway application refers to IP-to-Tel (PSTN for digital interfaces) and Tel-to-IP call routing. For digital interfaces, Tel refers to the PSTN.

Notes:

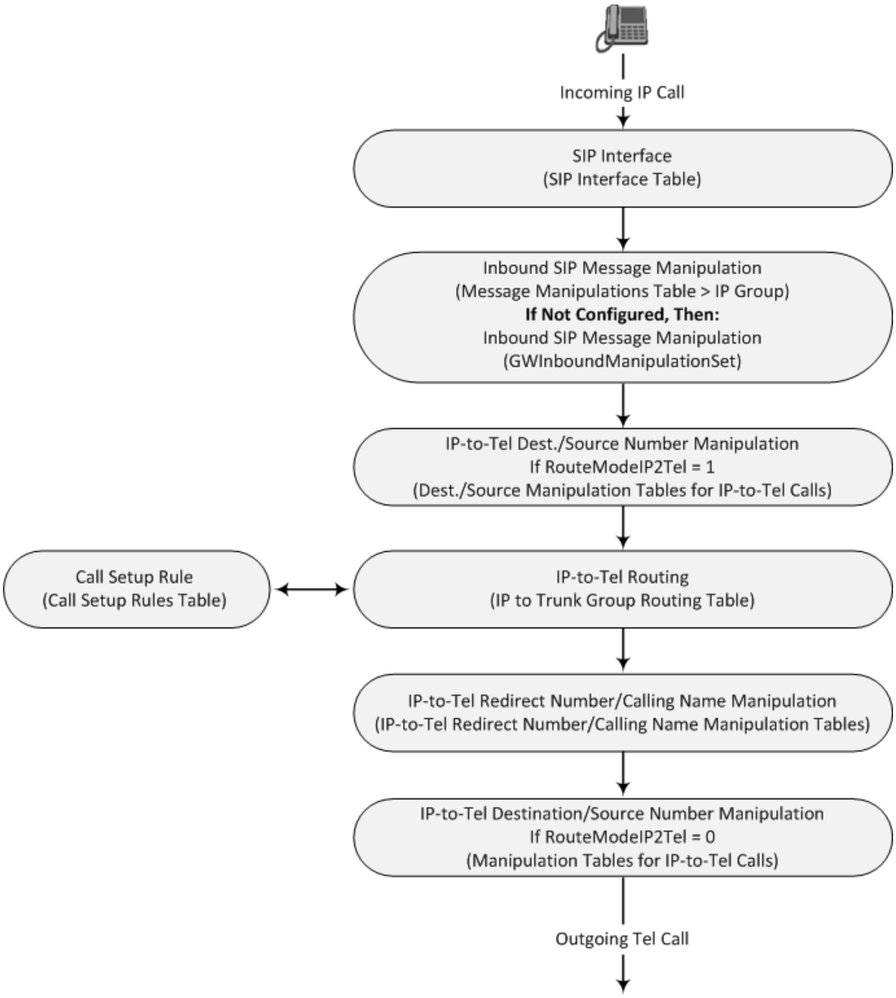
- In some areas of the Web interface, the term "GW" refers to the Gateway application.
- The terms *IP-to-Tel* and *Tel-to-IP* refer to the direction of the call relative to the device. IP-to-Tel refers to calls received from the IP network and destined to the PSTN/PBX (i.e., telephone connected directly or indirectly to the device); Tel-to-IP refers to calls received from the PSTN/PBX, and destined for the IP network.

20.1 Call Processing Summary

The device's call processing for Gateway calls is shown in the following flowcharts.

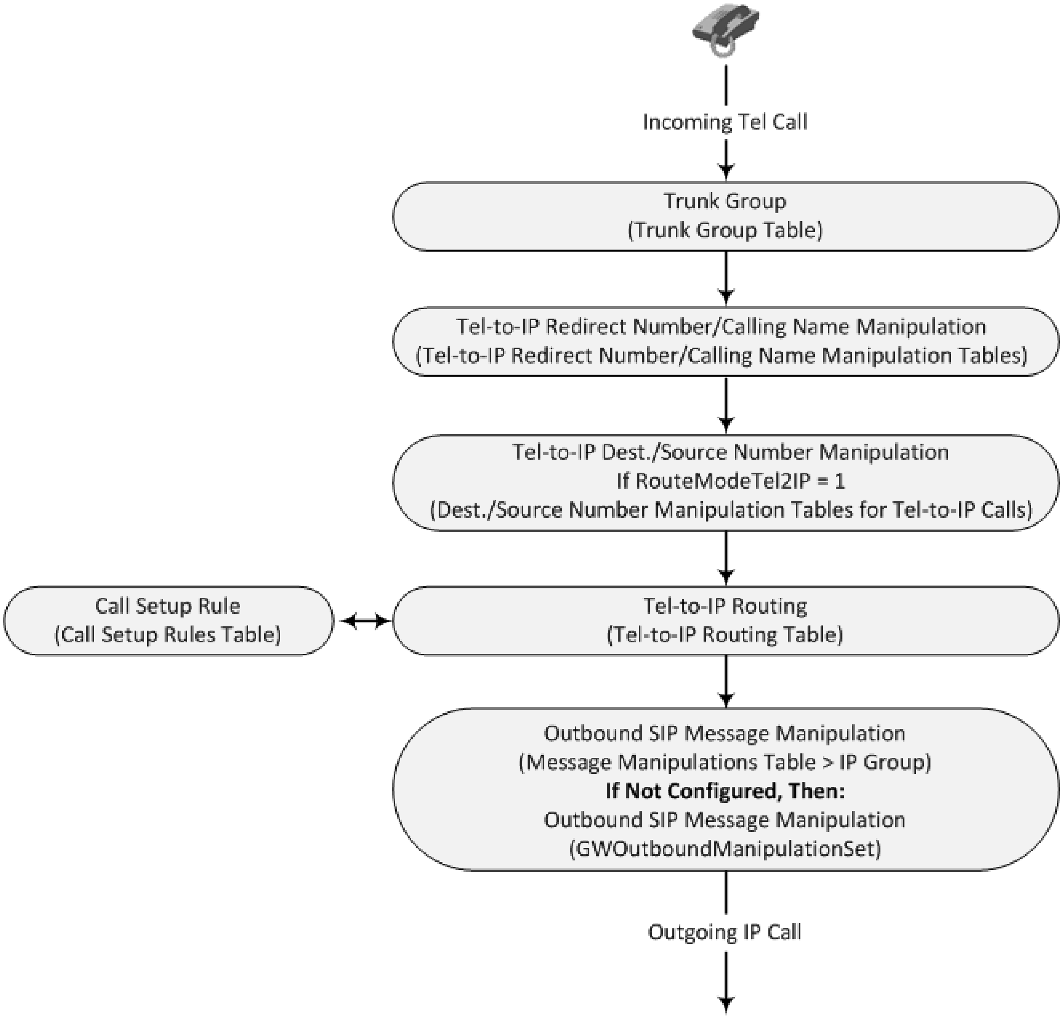
- **IP-to-Tel Call:**

Figure 20-1: IP-to-Tel Call Processing Flowchart


Incoming IP Call
SIP Interface
(SIP Interface Table)
Inbound SIP Message Manipulation
(Message Manipulations Table > IP Group)
If Not Configured, Then:
Inbound SIP Message Manipulation
(GWInboundManipulationSet)
IP-to-Tel Dest./Source Number Manipulation
If RouteModeIP2Tel = 1
(Dest./Source Manipulation Tables for IP-to-Tel Calls)
Call Setup Rule
(Call Setup Rules Table)
IP-to-Tel Routing
(IP to Trunk Group Routing Table)
IP-to-Tel Redirect Number/Calling Name Manipulation
(IP-to-Tel Redirect Number/Calling Name Manipulation Tables)
IP-to-Tel Destination/Source Number Manipulation
If RouteModeIP2Tel = 0
(Manipulation Tables for IP-to-Tel Calls)
Outgoing Tel Call

- **Tel-to-IP Call:**

Figure 20-2: Tel-to-IP Call Processing Flowchart


Incoming Tel Call
Trunk Group
(Trunk Group Table)
Tel-to-IP Redirect Number/Calling Name Manipulation
(Tel-to-IP Redirect Number/Calling Name Manipulation Tables)
Tel-to-IP Dest./Source Number Manipulation
If RouteModeTel2IP = 1
(Dest./Source Number Manipulation Tables for Tel-to-IP Calls)
Call Setup Rule
(Call Setup Rules Table)
Tel-to-IP Routing
(Tel-to-IP Routing Table)
Outbound SIP Message Manipulation
(Message Manipulations Table > IP Group)
If Not Configured, Then:
Outbound SIP Message Manipulation
(GWOutboundManipulationSet)
Outgoing IP Call

21 Digital PSTN

This section describes the configuration of the public switched telephone network (PSTN) related parameters.

21.1 Configuring Trunk Settings

The Trunk Settings page allows you to configure the device's trunks. This includes selecting the PSTN protocol and configuring related parameters.

This page also enables the following maintenance procedures:

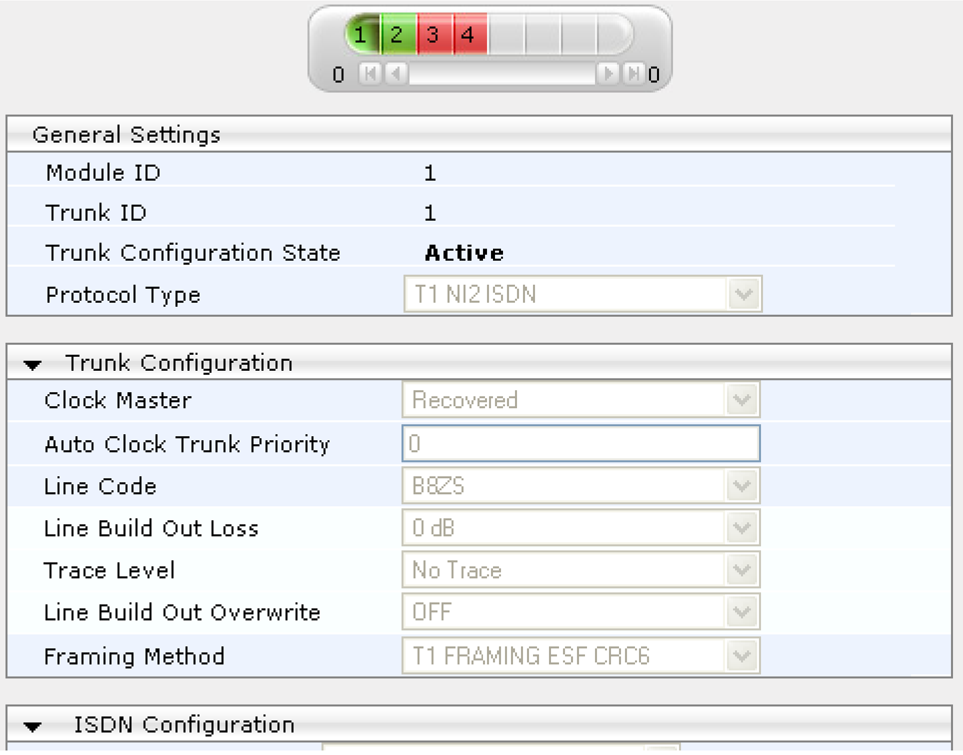
- **Taking a Trunk Out of Service:** Some parameters can be configured when the trunk is in service, while others require you to take the trunk out of service. This is done by clicking the **Stop** ■ button. Once you have "stopped" a trunk, all current calls are dropped and no new calls can be made on the trunk.
- **Deactivating a Trunk:** You can deactivate a trunk for maintenance. This is done by clicking the **Deactivate** Deactivate button. Deactivation temporarily disconnects (logically) the trunk from the PSTN network. Upon trunk deactivation, the device generates an AIS alarm on the trunk to the far-end. As a result, an RAI alarm signal may be received by the device. A subsequent trunk activation, done by clicking the **Activate** Activate button, reconnects the trunk to the PSTN network and clears the AIS alarm. Trunk deactivation is typically used for maintenance such as checking the trunk's physical integrity.
- **Creating a Loopback Line:** You can create (and remove) remote loopback for DS1 lines. This is done by clicking the **Create Loopback** Create Loopback button. To remove the loopback, click the **Remove Loopback** Remove Loopback button.

Notes:

- To delete a configured trunk, set the 'Protocol Type' parameter to **NONE**.
- For a description of the trunk parameters, see "PSTN Parameters" on page 944.
- During trunk deactivation, you cannot configure trunks.
- You cannot activate or deactivate a stopped trunk.
- If the trunk can’t be stopped because it provides the device’s clock (assuming the device is synchronized with the BRI clock), assign a different trunk to provide the device’s clock or enable ‘TDM Bus PSTN Auto Clock’ in the TDM Bus Settings page (see "TDM and Timing" on page 426).
- The ISDN BRI North American variants (NI-2, DMS-100, and 5ESS) are partially supported by the device. Please contact your AudioCodes sales representative before implementing this protocol.
- The displayed parameters depend on the protocol selected.

➢ To configure trunks:

1. Open the Trunk Settings page (**Configuration** tab > **VoIP** menu > **PSTN** > **Trunk Settings**).



General Settings

Module ID	1
Trunk ID	1
Trunk Configuration State	**Active**
Protocol Type	T1 NI2 ISDN

Trunk Configuration

Clock Master	Recovered
Auto Clock Trunk Priority	0
Line Code	B8ZS
Line Build Out Loss	0 dB
Trace Level	No Trace
Line Build Out Overwrite	OFF
Framing Method	T1 FRAMING ESF CRC6

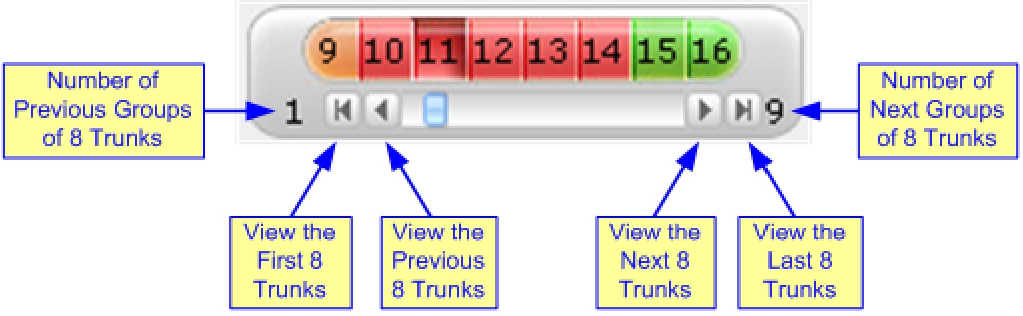
ISDN Configuration

On the top of the page, a bar with Trunk number icons displays the status of each trunk, according to the following color codes:

- **Grey:** Disabled
- **Green:** Active
- **Yellow:** RAI alarm (also appears when you deactivate a Trunk by clicking the **Deactivate** button)
- **Red:** LOS/LOF alarm
- **Blue:** AIS alarm
- **Orange:** D-channel alarm (ISDN only)

2. Select the trunk that you want to configure by clicking the desired Trunk number icon. The bar initially displays the first eight trunk number icons (i.e., trunks 1 through 8). To scroll through the trunk number icons (i.e., view the next/last or previous/first group of eight trunks), see the figure below:

Figure 21-1: Trunk Scroll Bar (Used Only as an Example)



9
10
11
12
13
14
15
16
1
9
Number of Previous Groups of 8 Trunks
Number of Next Groups of 8 Trunks
View the First 8 Trunks
View the Previous 8 Trunks
View the Next 8 Trunks
View the Last 8 Trunks

Note: If the Trunk scroll bar displays all available trunks, the scroll bar buttons are unavailable.

After you have selected a trunk, the following is displayed:

- The read-only 'Module ID' field displays the module number to which the trunk belongs.
- The read-only 'Trunk ID' field displays the selected trunk number.
- The read-only ‘Trunk Configuration State’ displays the state of the trunk ('Active' or 'Inactive').
- The displayed parameters pertain to the selected trunk only.

3. Click the **Stop Trunk** ■ button (located at the bottom of the page) to take the trunk out of service so that you can configure the currently grayed out (unavailable) parameters. (Skip this step if you want to configure parameters that are available when the trunk is active). The stopped trunk is indicated by the following:
   - The ‘Trunk Configuration State’ field displays ‘Inactive’.
   - The **Stop Trunk** button is replaced by the **Apply Trunk Settings** ✓ button.

     When all trunks are stopped, the **Apply to All Trunks** ✓ button also appears.
   - All the parameters are available and can be modified.
4. Configure the trunk parameters as required.
5. Click the **Apply Trunk Settings** button to apply the changes to the selected trunk (or click **Apply to All Trunks** to apply the changes to all trunks); the **Stop Trunk** button replaces **Apply Trunk Settings** and the ‘Trunk Configuration State’ displays 'Active'.
6. To save the changes to flash memory, see "Saving Configuration" on page 643.
7. To reset the device, see "Resetting the Device" on page 641.

21.2 TDM and Timing

This section describes the configuration of the TDM and clock timing parameters.

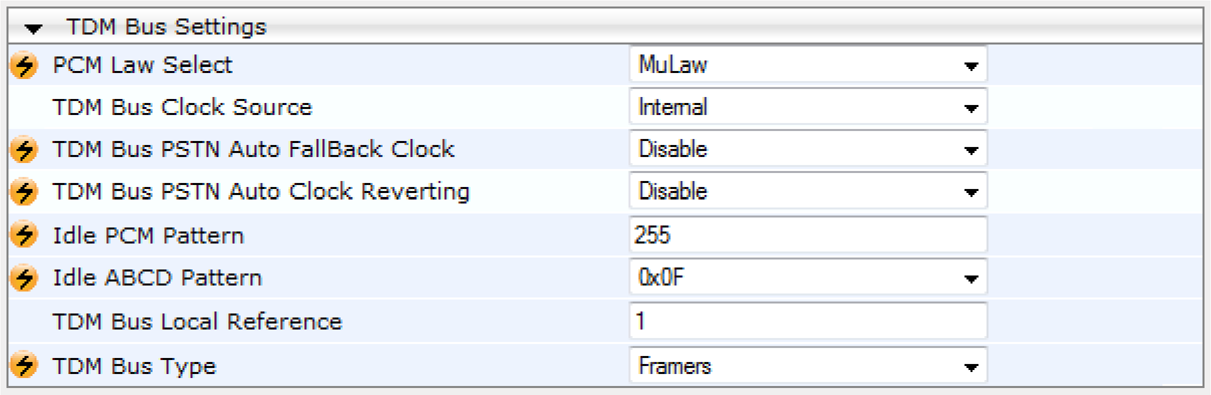
21.2.1 Configuring TDM Bus Settings

The TDM page allows you to configure the device's Time-Division Multiplexing (TDM) bus settings. For a description of these parameters, see "PSTN Parameters" on page 944.

➢ **To configure the TDM Bus settings:**

1. Open the TDM page (**Configuration** tab > **VoIP** menu > TDM > TDM Bus Settings).

Figure 21-2: TDM Bus Settings Page



▼ TDM Bus Settings

PCM Law Select	MuLaw
TDM Bus Clock Source	Internal
TDM Bus PSTN Auto FallBack Clock	Disable
TDM Bus PSTN Auto Clock Reverting	Disable
Idle PCM Pattern	255
Idle ABCD Pattern	0x0F
TDM Bus Local Reference	1
TDM Bus Type	Framers

2. Configure the parameters as required.
3. Click **Submit**.
4. Save the changes to flash memory, see "Saving Configuration" on page 643.

21.2.2 Clock Settings

In a traditional TDM service network such as PSTN, both ends of the TDM connection must be synchronized. If synchronization is not achieved, voice frames are either dropped (to prevent a buffer overflow condition) or inserted (to prevent an underflow condition). In both cases, connection quality and reliability is affected.

- PSTN line clock (see "Recovering Clock from PSTN Line" on page 427)
- Internal clock (see "Configuring Internal Clock as Clock Source" on page 427)

Note: When the device is used in a ‘non-span’ configuration, the internal device clock must be used (as explained above).

21.2.2.1 Recovering Clock from PSTN Line Interface

This section provides a brief description for configuring synchronization based on recovering clock from the PSTN line interface. For a full description of the clock parameters, see "PSTN Parameters" on page 944.

➢ **To configure synchronization based on clock from PSTN line:**

1. In the TDM Bus Settings page, do the following:
   a. Set the 'TDM Bus Clock Source' parameter (TDMBusClockSource) to **Network** to recover the clock from the line interface.
   b. Select the trunk from which the clock is derived, using the 'TDM Bus Local Reference' parameter (TDMBusLocalReference).

> **Note:** The BRI trunk should be configured as an ISDN user-side.

   c. Enable automatic switchover to the next available "slave" trunk if the device detects that the local-reference trunk is no longer capable of supplying the clock to the system:
      a. Set the 'TDM Bus PSTN Auto FallBack Clock' parameter (TDMBusPSTNAutoClockEnable) to **Enable**.
      b. Enable the device to switch back to a previous trunk that returns to service if it has higher switchover priority, using the 'TDM Bus PSTN Auto Clock Reverting' parameter (TDMBusPSTNAutoClockRevertingEnable).
      c. In the Trunk Settings page, configure the priority level of the trunk for taking over as a local-reference trunk, using the 'Auto Clock Trunk Priority' parameter (AutoClockTrunkPriority). A value of 100 means that it never uses the trunk as local reference.

21.2.2.2 Configuring Internal Clock as Clock Source

This section describes how to configure the device to use its internal clock source. The internal clock source is a stratum 4E-compliant clock source. When the device has no line interfaces, the device should be configured in this mode.

➢ **To configure internal clock as clock source:**

1. Set the clock source to be from the device's internal oscillator. In the TDM Bus Settings page, set the 'TDM Bus Clock Source' parameter (TDMBusClockSource) to **Internal**.

21.3 Configuring Digital Gateway Parameters

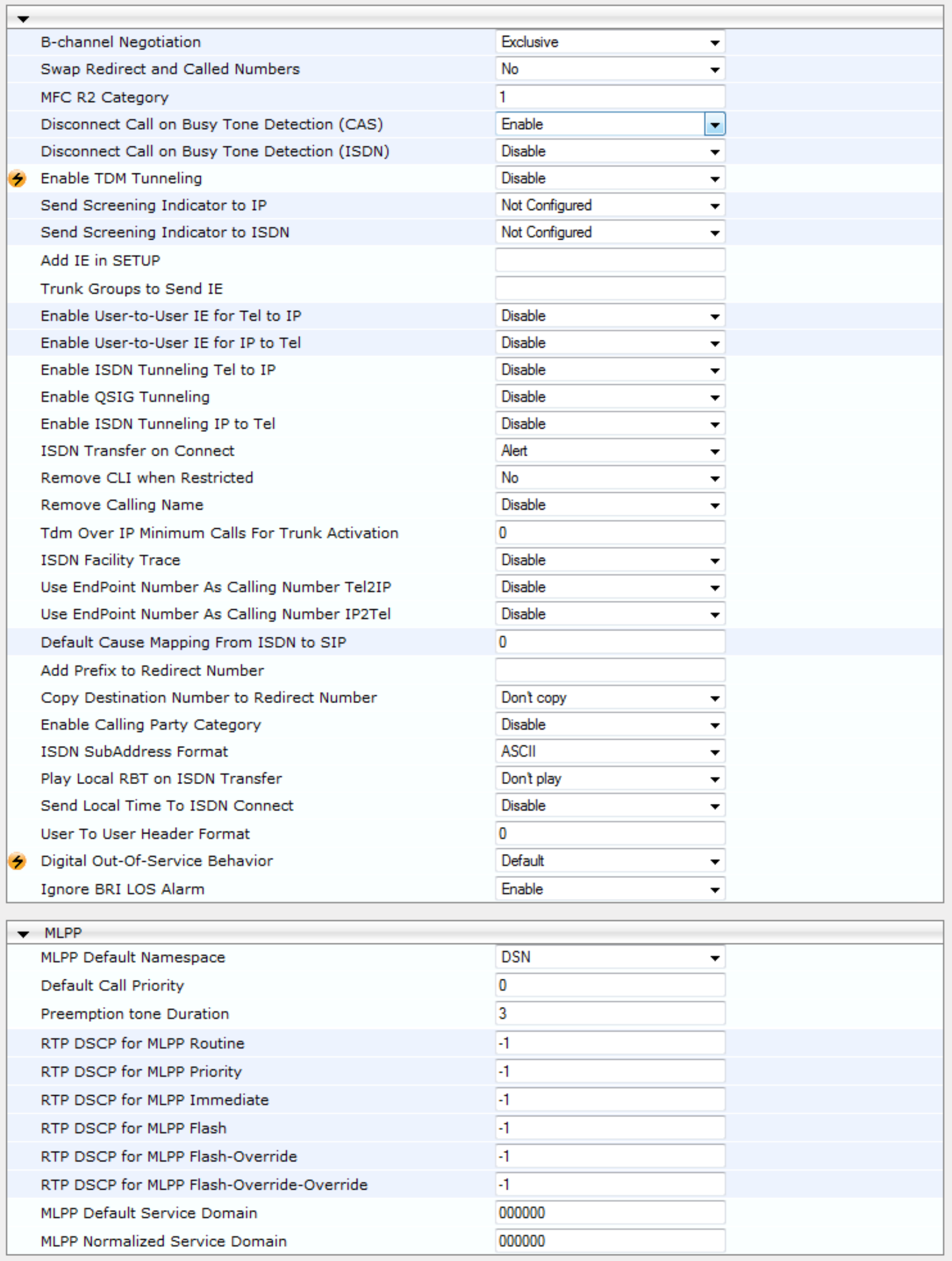
The Digital Gateway Parameters page allows you to configure miscellaneous digital parameters. For a description of these parameters, see "Configuration Parameters Reference" on page 813.

➢ **To configure the digital gateway parameters:**

1. Open the Digital Gateway Parameters page (**Configuration** tab > **VoIP** menu >

Gateway > **Digital Gateway** > **Digital Gateway Parameters**).

Figure 21-3: Digital Gateway Parameters Page



B-channel Negotiation	Exclusive
Swap Redirect and Called Numbers	No
MFC R2 Category	1
Disconnect Call on Busy Tone Detection (CAS)	Enable
Disconnect Call on Busy Tone Detection (ISDN)	Disable
⚡ Enable TDM Tunneling	Disable
Send Screening Indicator to IP	Not Configured
Send Screening Indicator to ISDN	Not Configured
Add IE in SETUP	
Trunk Groups to Send IE	
Enable User-to-User IE for Tel to IP	Disable
Enable User-to-User IE for IP to Tel	Disable
Enable ISDN Tunneling Tel to IP	Disable
Enable QSIG Tunneling	Disable
Enable ISDN Tunneling IP to Tel	Disable
ISDN Transfer on Connect	Alert
Remove CLI when Restricted	No
Remove Calling Name	Disable
Tdm Over IP Minimum Calls For Trunk Activation	0
ISDN Facility Trace	Disable
Use EndPoint Number As Calling Number Tel2IP	Disable
Use EndPoint Number As Calling Number IP2Tel	Disable
Default Cause Mapping From ISDN to SIP	0
Add Prefix to Redirect Number	
Copy Destination Number to Redirect Number	Don't copy
Enable Calling Party Category	Disable
ISDN SubAddress Format	ASCII
Play Local RBT on ISDN Transfer	Don't play
Send Local Time To ISDN Connect	Disable
User To User Header Format	0
⚡ Digital Out-Of-Service Behavior	Default
Ignore BRI LOS Alarm	Enable

MLPP

MLPP Default Namespace	DSN
Default Call Priority	0
Preemption tone Duration	3
RTP DSCP for MLPP Routine	-1
RTP DSCP for MLPP Priority	-1
RTP DSCP for MLPP Immediate	-1
RTP DSCP for MLPP Flash	-1
RTP DSCP for MLPP Flash-Override	-1
RTP DSCP for MLPP Flash-Override-Override	-1
MLPP Default Service Domain	000000
MLPP Normalized Service Domain	000000

2. Configure the parameters as required.
3. Click Subm**it**.
4. To save the changes to flash memory, see "Saving Configuration" on page 643.

21.4 Tunneling Applications

This section discusses the device's support for VoIP tunneling applications.

21.4.1 QSIG Tunneling

The device supports QSIG tunneling over SIP, according to IETF Internet-Draft draft-elwell-sipping-qsig-tunnel-03 ("Tunnelling of QSIG over SIP") and ECMA-355/ISO/IEC 22535. This is applicable to all ISDN variants. QSIG tunneling can be applied to all calls or to specific calls using IP Profiles.

QSIG tunneling sends all QSIG messages as raw data in corresponding SIP messages using a dedicated message body. This is used, for example, to enable two QSIG subscribers connected to the same or different QSIG PBX to communicate with each other over an IP network. Tunneling is supported in both directions (Tel-to-IP and IP-to-Tel).

The term tunneling means that messages are transferred ‘as is’ to the remote side without being converted (QSIG > SIP > QSIG). The advantage of tunneling over QSIG-to-SIP interworking is that by using interworking, QSIG functionality can only be partially achieved. When tunneling is used, all QSIG capabilities are supported and the tunneling medium (the SIP network) does not need to process these messages.

QSIG messages are transferred in SIP messages in a separate Multipurpose Internet Mail Extensions (MIME) body. Therefore, if a message contains more than one body (e.g., SDP and QSIG), multipart MIME must be used. The Content-Type of the QSIG tunneled message is ‘application/QSIG’. The device also adds a Content-Disposition header in the following format:

```
Content-Disposition: signal; handling=required.
```

QSIG tunneling is done as follows:

- **Call setup (originating device):** The QSIG Setup request is encapsulated in the SIP INVITE message without being altered. After the SIP INVITE request is sent, the device does not encapsulate the subsequent QSIG message until a SIP 200 OK response is received. If the originating device receives a 4xx, 5xx, or 6xx response, it disconnects the QSIG call with a ‘no route to destination’ cause.
- **Call setup (terminating device):** After the terminating device receives a SIP INVITE request with a 'Content-Type: application/QSIG', it sends the encapsulated QSIG Setup message to the Tel side and sends a 200 OK response (no 1xx response is sent) to IP. The 200 OK response includes an encapsulated QSIG Call Proceeding message (without waiting for a Call Proceeding message from the Tel side). If tunneling is disabled and the incoming INVITE includes a QSIG body, a 415 response is sent.
- **Mid-call communication:** After the SIP connection is established, all QSIG messages are encapsulated in SIP INFO messages.
- **Call tear-down:** The SIP connection is terminated once the QSIG call is complete. The Release Complete message is encapsulated in the SIP BYE message that terminates the session.

➢ **To enable QSIG tunneling:**

1. In the Digital Gateway Parameters page (**Configuration** tab > **VoIP** menu > **Gateway** > **Digital Gateway** > **Digital Gateway Parameters**), set the 'Enable QSIG Tunneling' parameter (EnableQSIGTunneling) to **Enable** on the originating and terminating devices.
2. Configure the QSIGTunnelingMode parameter for defining the format of encapsulated QSIG message data in the SIP message MIME body (0 for ASCII presentation; 1 for binary encoding).
3. Set the ISDNDuplicateQ931BuffMode parameter to 128 to duplicate all messages.
4. Set the ISDNInCallsBehavior parameter to 4096.

5. Set the ISDNRxOverlap parameter to 0 for tunneling of QSIG overlap-dialed digits (see below for description).

The configuration of the ISDNInCallsBehavior and ISDNRxOverlap parameters allows tunneling of QSIG overlap-dialed digits (Tel to IP). In this configuration, the device **delays** the sending of the QSIG Setup Ack message upon receipt of the QSIG Setup message. Instead, the device sends the Setup Ack message to QSIG only when it receives the SIP INFO message with Setup Ack encapsulated in its MIME body. The PBX sends QSIG Information messages (to complete the Called Party Number) only after it receives the Setup Ack. The device relays these Information messages encapsulated in SIP INFO messages to the remote party.

21.5 ISDN Overlap Dialing

Overlap dialing is a dialing scheme used by several ISDN variants to send and/or receive called number digits one after the other (or several at a time). This is in contrast to en-bloc dialing in which a complete number is sent in one message.

The device supports the following ISDN overlap dialing methods:

- Collects ISDN called party number digits and then sends the SIP INVITE to the IP side with the complete destination number (see "Collecting ISDN Digits and Sending Complete Number in SIP" on page 430)
- Interworks ISDN overlap dialing with SIP, according to RFC 3578 (see "Interworking ISDN Overlap Dialing with SIP According to RFC 3578" on page 431)

21.5.1 Collecting ISDN Digits and Sending Complete Number in SIP

The device can support an overlap dialing mode whereby the device collects the called party number digits from ISDN Q.931 Information messages or DTMF signals, and then sends a SIP INVITE message to the IP side containing the complete destination number.

ISDN overlap dialing for incoming ISDN calls can be configured for the entire device or per ISDN trunk. This is configured using the global, ISDNRxOverlap parameter or the ISDNRxOverlap_x parameter (where *x* denotes the trunk number), respectively.

By default (see the ISDNINCallsBehavior parameter), the device plays a dial tone to the ISDN user side when it receives an empty called number from the ISDN. In this scenario, the device includes the Progress Indicator in the SetupAck ISDN message that it sends to the ISDN side.

The device can also mute in-band DTMF detection until it receives the complete destination number from the ISDN. This is configured using the MuteDTMFInOverlap parameter. The Information digits can be sent in-band in the voice stream, or out-of-band using Q.931 Information messages. If Q.931 Information messages are used, the DTMF in-band detector must be disabled. Note that when at least one digit is received in the ISDN Setup message, the device stops playing a dial tone.

The device stops collecting digits (from the ISDN) upon the following scenarios:

- The device receives a Sending Complete IE in the ISDN Setup or Information messages, indicating no more digits.
- The timeout between received digits expires (configured by the TimeBetweenDigits parameter).
- The maximum number of received digits has been reached (configured by the MaxDigits parameter).
- A match is found with the defined digit map (configured by the DigitMapping parameter).

Relevant parameters (described in "PSTN Parameters" on page 944):

- ISDNRxOverlap_x = 1 (can be configured per trunk)
- TimeBetweenDigits
- MaxDigits
- MuteDTMFInOverlap
- DigitMapping

For configuring ISDN overlap dialing using the Web interface, see "Configuring Trunk Settings" on page 423.

21.5.2 Interworking ISDN Overlap Dialing with SIP According to RFC 3578

With overlap dialing disabled, the device expects to receive the digits all at once (enbloc) or with very little delay between digits and then sends the complete number in a single message. Overlap signaling sends portions of the number in separate messages as it collects the digits from the sender. The interval between receiving the digits (time between digits) is relatively long. However, overlap dialing allows the device to begin call setup (routing) even before all digits have been collected. For example, if the dialled (destination) number is "3312418", the device first receives the digits "331" and then routes the call based on these digits. It then delivers the remaining 4 digits "2418" in overlap mode. The device supports the interworking of ISDN overlap dialing to SIP and vice versa, according to RFC 3578.

- **Interworking ISDN overlap dialing to SIP (Tel to IP):** The device sends the first digits (e.g., "331") received from the ISDN Setup message to the IP side in the initial SIP INVITE message. Each time it receives additional (collected) digits, which are received from subsequent Q.931 Information messages, it sends them to the IP side in SIP re-INVITE or SIP INFO messages. You can use the following parameters to configure overlap dialing for Tel-to-IP calls:
    - ISDNRxOverlap: Enables Tel-to-IP overlap dialing and defines how the device sends the collected digits to the IP side - in SIP re-INVITE [2] or INFO messages [3].
    - MinOverlapDigitsForRouting: Defines the minimum number of overlap digits to collect from the Tel side before the device can send the first SIP message (INVITE) for routing the call to the IP side.
    - MaxDigits: Defines the maximum number of collected digits that can be received from the Tel side (if ISDN Sending Complete IE is not received). When the number of collected digits reaches the maximum, the device uses these digits for the called destination number.
    - TimeBetweenDigits: Defines the maximum time (in seconds) that the device waits between digits received from the Tel side. When the time expires, the device uses the collected digits to dial the called destination number.
    - MuteDTMFInOverlap: Enables the device to ignore in-band DTMF digits received during overlap dialing.

Note: If the device receives SIP 4xx responses during the overlap dialing (while collecting digits), it does not release the call.

- **Interworking SIP to ISDN overlap dialing (IP to Tel):** The device sends the first digits (e.g., "331") received from the initial SIP INVITE message to the Tel side in an ISDN Setup message. Each time it receives additional (collected) digits for the same dialog, which are received from subsequent SIP re-INVITE messages or SIP INFO messages, it sends them to the Tel side in SIP Q.931 Information messages. For each subsequent re-INVITE or SIP INFO message received, the device sends a SIP 484 "Address Incomplete" response to the IP side to maintain the current dialog session and to receive additional digits from subsequent re-INVITE or INFO messages. You can use the following parameters to configure overlap dialing for IP-to-Tel calls:

- ISDNTxOverlap: Enables IP-to-Tel overlap dialing and defines how the device receives the collected digits from the IP side - in SIP re-INVITE [1] or INFO messages [2].
- TimeBetweenDigits: Defines the maximum time (in seconds) that the device waits between digits received from the IP side. When the time expires, the device uses the collected digits to dial the called destination number.

Note: For IP-to-Tel overlap dialing, to send ISDN Setup messages without including the Sending Complete IE, you must configure the ISDNOutCallsBehavior parameter to USER SENDING COMPLETE [2].

For more information on the above mentioned parameters, see PSTN Parameters on page 944. For configuring ISDN overlap dialing using the Web interface, see "Configuring Trunk Settings" on page 423.

21.6 Redirect Number and Calling Name (Display)

The following tables define the device's redirect number and calling name (Display) support for various ISDN variants according to NT (Network Termination) / TE (Termination Equipment) interface direction:

Table 21-1: Calling Name (Display)

NT/TE Interface	DMS-100	NI-2	4/5ESS	Euro ISDN	QSIG
NT-to-TE	Yes	Yes	Yes	Yes	Yes
TE-to-NT	Yes	Yes	Yes	No	Yes

Table 21-2: Redirect Number

NT/TE Interface	DMS-100	NI-2	4/5ESS	Euro ISDN	QSIG
NT-to-TE	Yes	Yes	Yes	Yes	Yes
TE-to-NT	Yes	Yes	Yes	Yes*	Yes

***** When using ETSI DivertingLegInformation2 in a Facility IE (not Redirecting Number IE).

22 Trunk Groups

This section describes the configuration of the device's channels, which includes assigning them to Trunk Groups.

22.1 Configuring Trunk Groups

The Trunk Group table lets you configure up to 120 Trunk Groups. A Trunk Group is a logical group of physical trunks and channels. A Trunk Group can include multiple trunks and ranges of channels. To enable and activate the channels of the device, Trunk Groups need to be configured and assigned with telephone numbers. Channels that are not configured in this table are disabled.

Once you have configured your Trunk Groups, you need to use them for routing incoming IP calls to the Tel side, which is represented by a specific Trunk Group (ID). For configuring IP-to-Tel routing rules, see "Configuring IP-to-Trunk Group Routing Rules" on page 476. You can also use Trunk Groups for routing Tel calls to the IP side. For configuring Tel-to-IP routing rules, see "Configuring Tel-to-IP Routing Rules" on page 467.

The following procedure describes how to configure Trunk Groups through the Web interface. You can also configure it through ini file (TrunkGroup_x) or CLI (configure voip > gw hunt-or-trunk-group trunk-group).

➢ **To configure a Trunk Group:**

1. Open the Trunk Group table (**Configuration** tab > **VoIP** menu > **Gateway** > **Trunk Group** > **Trunk Group**).
2. Configure a Trunk Group according to the parameters described in the table below.
3. Click **Submit**.

You can also register all your Trunk Groups. The registration method per Trunk Group is configured by the 'Registration Mode' parameter in the Trunk Group Settings page (see "Configuring Trunk Group Settings" on page 435).

- To register the Trunk Groups, click the **Register** button, located below the Trunk Group table.
- To unregister the Trunk Groups, click the **Unregister** button, located below the Trunk Group table.

Table 22-1: Trunk Group Table Parameter Descriptions

Parameter	Description
Module `module` [TrunkGroup_Module]	Defines the telephony interface module for which you want to define the Trunk Group.
From Trunk `first-trunk-id` [TrunkGroup_FirstTrunkId]	Defines the starting physical Trunk number in the Trunk Group. The number of listed Trunks depends on the device's hardware configuration.
To Trunk `last-trunk-id` [TrunkGroup_LastTrunkId]	Defines the ending physical Trunk number in the Trunk Group. The number of listed Trunks depends on the device's hardware configuration.

Channels
`first-b-channel`
[TrunkGroup_FirstBChannel]
`last-b-channel`
	Defines the device's Trunk B-channels. To enable channels, enter the channel numbers.
	You can enter a range of channels by using the syntax *n-m*, where *n* represents the lower channel number and *m* the higher channel number. For example, "1-4" specifies channels 1

Parameter	Description
[TrunkGroup_LastBChannel]	through 4. To represent all the Trunk's B-channels, enter a single asterisk (*). **Note:** The number of defined channels must not exceed the maximum number of the Trunk's B-channels.
Phone Number `first-phone-number` [TrunkGroup_FirstPhoneNumber]	Defines the telephone number(s) of the channels. The valid value can be up to 50 characters. For a range of channels, enter only the first telephone number. Subsequent channels are assigned the next consecutive telephone number. For example, if you enter 400 for channels 1 to 4, then channel 1 is assigned phone number 400, channel 2 is assigned phone number 401, and so on. These numbers are also used for channel allocation for IP-to-Tel calls if the Trunk Group's 'Channel Select Mode' parameter is set to **By Dest Phone Number**. **Notes:** ▪ If this field includes alphabetical characters and the phone number is defined for a range of channels (e.g., 1-4), then the phone number must end with a number (e.g., 'user1'). ▪ This field is optional. The logical numbers defined in this field are used when an incoming Tel call doesn't contain the calling number or called number (the latter being determined by the ReplaceEmptyDstWithPortNumber parameter). These numbers are used to replace them. ▪ This field is ignored if routing of IP-to-Tel calls is done according to the Supplementary Services table, where multiple line extension numbers are configured per port (see Configuring Multi-Line Extensions and Supplementary Services). For this routing method, the 'Channel Select Mode' must be set to **Select Trunk By Supplementary Services Table** in the Trunk Group Settings table (see "Configuring Trunk Group Settings" on page 435).
Trunk Group ID `trunk-group-id` [TrunkGroup_TrunkGroupNum]	Defines the Trunk Group ID for the specified channels. The same Trunk Group ID can be assigned to more than one group of channels. If an IP-to-Tel call is assigned to a Trunk Group, the IP call is routed to the channel(s) pertaining to that Trunk Group ID. The valid value can be 0 to 119.
Tel Profile Name `tel-profile-id` [TrunkGroup_ProfileName]	Assigns a Tel Profile to the Trunk Group. For configuring Tel Profiles, see "Configuring Tel Profiles" on page 384.

22.2 Configuring Trunk Group Settings Table

The Trunk Group Settings table lets you configure various settings per Trunk Group, which are configured in the Trunk Group table. The main configuration includes the following:

- Channel select method, which defines how the device allocates IP-to-Tel calls to the channels of a Trunk Group.
- Registration method for registering Trunk Groups to remote IP servers (*Serving IP Group*).

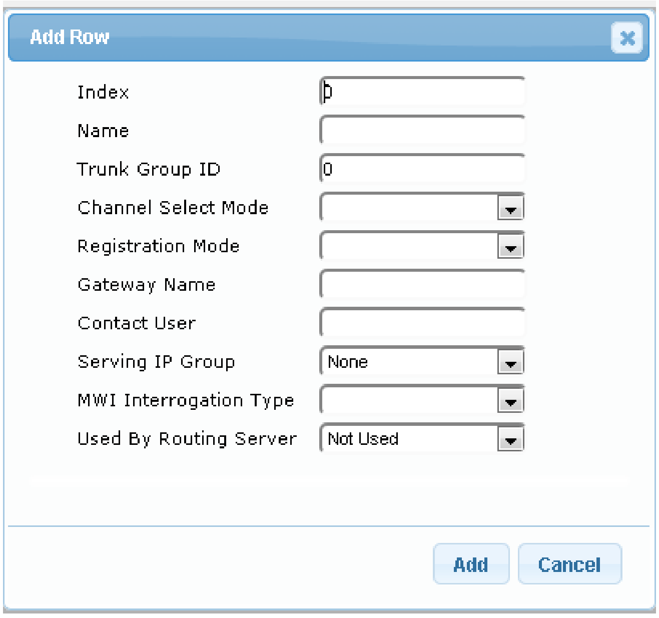
The Trunk Group Settings table also provides an **Action** drop-down button with commands that let you perform various actions per configured Trunk Group:

- **Lock / Unlock:** Locks (blocks) a Trunk Group in order to take its member trunks out-of-service. For more information, see 'Locking and Unlocking Trunk Groups' on page 645.
- **Register / Un-Register:** Initiates a registration request for the Trunk Group with a Serving IP Group. For more information, see the description of the 'Registration Mode' parameter of the Trunk Group Settings table in this section.

The following procedure describes how to configure settings for Trunk Groups through the Web interface. You can also configure it through ini file (TrunkGroupSettings) or CLI (configure voip/gw hunt-or-trunk-group trunk-group-setting).

➢ To configure Trunk Group settings:

1. Open the Trunk Group Settings table (**Configuration** tab > **VoIP** menu > **Gateway** > **Trunk Group** > **Trunk Group Settings**).
2. Click **Add**; the following dialog box appears:



Add Row

Index	0
Name	
Trunk Group ID	0
Channel Select Mode	
Registration Mode	
Gateway Name	
Contact User	
Serving IP Group	None
MWI Interrogation Type	
Used By Routing Server	Not Used

3. Configure a Trunk Group according to the parameters described in the table below.
4. Click **Add**.

Table 22-2: Trunk Group Settings Table Parameter Descriptions

Parameter	Description
Index [TrunkGroupSettings_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.

Parameter	Description
Name `trunk-group-name` [TrunkGroupSettings_TrunkGroupName]	Defines an arbitrary name to easily identify the row. The name represents the Trunk Group in the SIP 'tgrp' parameter in outgoing INVITE messages (according to RFC 4904). For example: `sip:+16305550100;tgrp=`**`TG-1`**`;trunk-context=+1-630@isp.example.net;user=phone` The valid value can be a string of up to 40 characters. By default, no name is configured. **Notes:** ▪ Each row must be configured with a unique name. ▪ If the parameter is not configured, the Trunk Group decimal number is used in the SIP 'tgrp' parameter. ▪ The feature is enabled by any of the following parameters: ✓ UseSIPtgrp ✓ UseBroadsoftDTG
Trunk Group ID trunk-group-id [TrunkGroupSettings_TrunkGroupId]	Defines the Trunk Group ID that you want to configure.
Channel Select Mode `channel-select-mode` [TrunkGroupSettings_ChannelSelectMode]	Defines the method by which IP-to-Tel calls are assigned to the channels of the Trunk Group. ▪ **[0]** By Dest Phone Number = The channel is selected according to the called (destination) number. If the number is not located, the call is released. If the channel is unavailable (e.g., busy), the call is put on call waiting (if call waiting is enabled and no other call is on call waiting); otherwise, the call is released. ▪ **[1]** Cyclic Ascending = The next available channel in the Trunk Group, in ascending cyclic order is selected. After the device reaches the highest channel number in the Trunk Group, it selects the lowest channel number in the Trunk Group, and then starts ascending again. ▪ **[2]** Ascending = The lowest available channel in the Trunk Group is selected, and if unavailable, the next higher channel is selected. ▪ **[3]** Cyclic Descending = The next available channel in descending cyclic order is selected. The next lower channel number in the Trunk Group is always selected. When the device reaches the lowest channel number in the Trunk Group, it selects the highest channel number in the Trunk Group, and then starts descending again. ▪ **[4]** Descending = The highest available channel in the Trunk Group is selected, and if unavailable, the next lower channel is selected. ▪ **[5]** Dest Number + Cyclic Ascending = The channel is selected according to the called number. If the called number isn't found, the next available channel in ascending cyclic order is selected. **Note:** If the called number is located, but the port associated with the number is busy, the call is released. ▪ **[6]** By Source Phone Number = The channel is selected according to the calling number.

Parameter	Description
	▪ [7] Trunk Cyclic Ascending = The channel from the first channel of the next trunk (adjacent to the trunk from which the previous channel was selected) is selected. ▪ [8] Trunk & Channel Cyclic Ascending = The device implements the Trunk Cyclic Ascending and Cyclic Ascending methods to select the channel. This method selects the next physical trunk in the Trunk Group, and then selects the B-channel of this trunk according to the Cyclic Ascending method (i.e., selects the channel after the last allocated channel). For example, if the Trunk Group includes two physical trunks, 0 and 1: ✓ For the first incoming call, the first channel of Trunk 0 is selected. ✓ For the second incoming call, the first channel of Trunk 1 is selected. ✓ For the third incoming call, the second channel of Trunk 0 is selected. ▪ [10] Select Trunk by Supplementary Services Table = The BRI port/module is selected according to the settings in the ISDN Supplementary Services table (see Configuring Multi-Line Extensions and Supplementary Services on page 513), allowing the routing of IP-to-Tel calls to specific BRI endpoints according to extension number. This option is applicable only to BRI interfaces. ▪ **[11]** Dest Number + Ascending = The device allocates a channels to incoming IP-to-Tel calls as follows: a. The device attempts to route the call to the channel that is associated with the destination (called) number. If located, the call is sent to that channel. b. If the number is not located or the channel is unavailable (e.g., busy), the device searches in ascending order for the next available channel in the Trunk Group. If located, the call is sent to that channel. c. If all the channels are unavailable, the call is released. **Note:** If the parameter is not configured, the Trunk Group's channel select method is according to the global parameter, ChannelSelectMode.
Registration Mode `registration-mode` [TrunkGroupSettings_RegistrationMode]	Defines the registration method of the Trunk Group: ▪ **[1]** Per Gateway = (Default) Single registration for the entire device. This is applicable only if a default Proxy or Registrar IP is configured and Registration is enabled (i.e., parameter IsRegisterUsed is set to 1). In this mode, the SIP URI user part in the From, To, and Contact headers is set to the value of the global registration parameter, GWRegistrationName or username if GWRegistrationName is not configured. ▪ **[0]** Per Endpoint = Each channel in the Trunk Group registers individually. The registrations are sent to the 'Serving IP Group ID' if defined in the table, otherwise, it is sent to the default Proxy, and if no default Proxy, then to the Registrar IP.

Parameter	Description

	▪ **[4]** Don't Register = No registrations are sent by endpoints pertaining to the Trunk Group. For example, if the device is configured globally to register all its endpoints (using the parameter ChannelSelectMode), you can exclude some endpoints from being registered by assigning them to a Trunk Group and configuring the Trunk Group registration mode to 'Don't Register'.
	▪ **[5]** Per Account = Registrations are sent (or not) to an IP Group, according to the settings in the Account table (see "Configuring Registration Accounts" on page 363).
	An example is shown below of a REGISTER message for registering endpoint "101" using the registration Per Endpoint mode:
	`REGISTER sip:SipGroupName SIP/2.0`
	`Via: SIP/2.0/UDP`
	`10.33.37.78;branch=z9hG4bKac862428454`
	`From:`
	`<sip:101@GatewayName>;tag=1c862422082`
	`To: <sip:101@GatewayName>`
	`Call-ID:`
	`9907977062512000232825@10.33.37.78`
	`CSeq: 3 REGISTER`
	`Contact:`
	`<sip:101@10.33.37.78>;expires=3600`
	`Expires: 3600`
	`User-Agent: Sip-Gateway/v.6.80A.227.005`
	`Content-Length: 0`
	The "SipGroupName" in the Request-URI is configured in the IP Group table (see "Configuring IP Groups" on page 340).
	Notes:
	▪ If the parameter is not configured, the registration is performed according to the global registration parameter, ChannelSelectMode.
	▪ To enable Trunk Group registration, set the global parameter, IsRegisterNeeded to 1. This is unnecessary for 'Per Account' registration mode.
	▪ If the device is configured globally to register Per Endpoint and an channel group includes four channels to register Per Gateway, the device registers all channels except the first four channels. The group of these four channels sends a single registration request.

Gateway Name `gateway-name` [TrunkGroupSettings_GatewayName]	Defines the host name for the SIP From header in INVITE messages, and the From and To headers in REGISTER requests. **Note:** If the parameter is not configured, the global parameter, SIPGatewayName is used.
Contact User `contact-user` [TrunkGroupSettings_ContactUser]	Defines the user part for the SIP Contact URI in INVITE messages, and the From, To, and Contact headers in REGISTER requests. **Notes:**

Parameter	Description
	▪ The parameter is applicable only if the 'Registration Mode' parameter is set to 'Per Account' and registration through the Account table is successful. ▪ If registration fails, the user part in the INVITE Contact header is set to the source party number. ▪ The 'Contact User' parameter in the Account table overrides this parameter (see "Configuring Registration Accounts" on page 363).
Serving IP Group `serving-ip-group` [TrunkGroupSettings_ServingIPGroupName]	Assigns an IP Group to where the device sends INVITE messages for calls received from the Trunk Group. The actual destination to where the INVITE messages are sent is according to the Proxy Set associated with the IP Group. The Request-URI host name in the INVITE and REGISTER messages (except for 'Per Account' registration modes) is set to the value of the 'SIP Group Name' parameter configured in the IP Group table (see "Configuring IP Groups" on page 340). **Notes:** ▪ If the parameter is not configured, the INVITE messages are sent to the default Proxy or according to the Tel-to-IP Routing table (see "Configuring Tel-to-IP Routing Rules" on page 467). ▪ If the PreferRouteTable parameter is set to 1 (see "Configuring Proxy and Registration Parameters" on page 367), the routing rules in the Tel-to-IP Routing table take precedence over the selected Serving IP Group ID.
MWI Interrogation Type `mwi-interrogation-type` [TrunkGroupSettings_MWIInterrogationType]	Defines message waiting indication (MWI) QSIG-to-IP interworking for interrogating MWI supplementary services: ▪ [255] Not Configured ▪ [0] None = Disables the feature. ▪ [1] Use Activate Only = MWI Interrogation messages are not sent and only "passively" responds to MWI Activate requests from the PBX. ▪ [2] Result Not Used = MWI Interrogation messages are sent, but the result is not used. Instead, the device waits for MWI Activate requests from the PBX. ▪ [3] Use Result = MWI Interrogation messages are sent, its results are used, and the MWI Activate requests are used. MWI Activate requests are interworked to SIP NOTIFY MWI messages. The SIP NOTIFY messages are sent to the IP Group defined by the NotificationIPGroupID parameter. **Note:** The parameter appears in the table only if the VoiceMailInterface parameter is set to 3 (QSIG) (see Configuring Voice Mail on page 521).
Used By Routing Server `used-by-routing-server` [TrunkGroupSettings_UsedByRoutingServer]	Enables the use of the Trunk Group by a routing server for routing decisions. ▪ [0] Not Used (default) ▪ [1] Used For more information, see Centralized Third-Party Routing Server or ARM on page 272.

Parameter	Description
Admin State	(Read-only) Displays the administrators state: ▪ "Locked": The **Lock** command has been chosen from the **Action** drop-down button. ▪ "Unlocked": The **Unlock** command has been chosen from the **Action** drop-down button.
Status	(Read-only) Displays the current status of the trunks/channels in the Trunk Group: ▪ "In Service": Indicates that all channels in the Trunk Group are in service, for example, when the Trunk Group is unlocked or Busy Out state cleared (see the EnableBusyOut parameter for more information). ▪ "Going Out Of Service": Appears as soon as you choose the **Lock** command and indicates that the device is starting to lock the Trunk Group and take channels out of service. ▪ "Going Out Of Service (<duration remaining of graceful period> sec / <number of calls still active> calls)": Appears when the device is locking the Trunk Group and indicates the number of buys channels and the time remaining until the graceful period ends, after which the device locks the channels regardless of whether the call has ended or not. "Out Of Service": All fully configured trunks in the Trunk Group are out of service, for example, when the Trunk Group is locked or in Busy Out state (see the EnableBusyOut parameter).

23 Manipulation

This section describes the configuration of various manipulation processes.

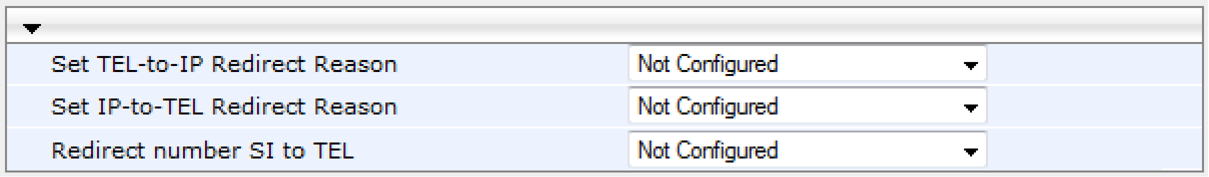
23.1 Configuring General Settings

The General Settings page allows you to configure general manipulation parameters. For a description of the parameters appearing on this page, see "Configuration Parameters Reference" on page 813.

➢ **To configure the general manipulation parameters:**

1. Open the General Settings page (**Configuration** tab > **VoIP** menu > **Gateway** > **Manipulations** >**General Settings**).

Figure 23-1: General Settings Page



Set TEL-to-IP Redirect Reason: Not Configured
Set IP-to-TEL Redirect Reason: Not Configured
Redirect number SI to TEL: Not Configured

2. Configure the parameters as required.
3. Click **Submit**.

23.2 Configuring Source/Destination Number Manipulation Rules

The number manipulation tables let you configure rules for manipulating source and destination telephone numbers for IP-to-Tel and Tel-to-IP calls. The number manipulation tables include the following:

- **Tel-to-IP calls:**
  - Source Phone Number Manipulation Table for Tel-to-IP Calls (up to 120 entries)
  - Destination Phone Number Manipulation Table for Tel-to-IP Calls (up to 120 entries)
- **IP-to-Tel calls:**
  - Source Phone Number Manipulation Table for IP-to-Tel Calls (up to 120 entries)
  - Destination Phone Number Manipulation Table for IP-to-Tel Calls (up to 120 entries)

Configuration of number manipulation rules includes two areas:

- **Rule:** Defines the matching characteristics of the incoming call (e.g., prefix of destination number).
- **Action:** Defines the action that is done if the incoming call matches the characteristics of the rule (e.g., removes a user-defined number of digits from the left of the number).

The device searches the table from top to bottom for the first rule that matches the characteristics of the incoming call. If it finds a matching rule, it applies the manipulation configured for that rule. In other words, a rule at the top of the table takes precedence over a rule defined lower down in the table. Therefore, define more specific rules above more generic rules. For example, if you configure the source prefix number as "551" for rule index 1 and "55" for rule index 2, the device uses rule index 1 for numbers that start with 551 and uses rule index 2 for numbers that start with 550, 552, 553, and so on until 559. However, if you configure the source prefix number as "55" for rule index 1 and "551" for rule index 2, the device applies rule index 1 to all numbers that start with 55, including numbers that start with 551. If the device doesn't find a matching rule, no manipulation is done on the call. You can perform a second "round" (additional) of source and destination number manipulations for IP-to-Tel calls, on an already manipulated number. The initial and additional number manipulation rules are both configured in the number manipulation tables for IP-to-Tel calls. The additional manipulation is performed on the initially manipulated number. Thus, for complex number manipulation schemes, you only need to configure relatively few manipulation rules in these tables (that would otherwise require many rules). To enable this additional manipulation, use the following parameters:

- Source number manipulation - PerformAdditionalIP2TELSourceManipulation
- Destination number manipulation - PerformAdditionalIP2TELDestinationManipulation

Telephone number manipulation can be useful, for example, for the following:

- Stripping or adding dialing plan digits from or to the number, respectively. For example, a user may need to first dial 9 before dialing the phone number to indicate an external line. This number 9 can then be removed by number manipulation before the call is setup.
- Allowing or blocking Caller ID information according to destination or source prefixes.
- Assigning Numbering Plan Indicator (NPI) and Type of Numbering (TON) to IP-to-Tel calls. The device can use a single global setting for NPI/TON classification or it can use the setting in the manipulation tables on a call-by-call basis.

Notes:

- Number manipulation can be performed before or after a routing decision is made. For example, you can route a call to a specific Trunk Group according to its original number, and then you can remove or add a prefix to that number before it is routed. To determine when number manipulation is performed, use the 'IP to Tel Routing Mode' parameter (RouteModeIP2Tel) and 'Tel to IP Routing Mode' parameter (RouteModeTel2IP).
- The device manipulates the number in the following order: 1) strips digits from the left of the number, 2) strips digits from the right of the number, 3) retains the defined number of digits, 4) adds the defined prefix, and then 5) adds the defined suffix.

The following procedure describes how to configure number manipulation rules through the Web interface. You can also configure this using the following management tools:

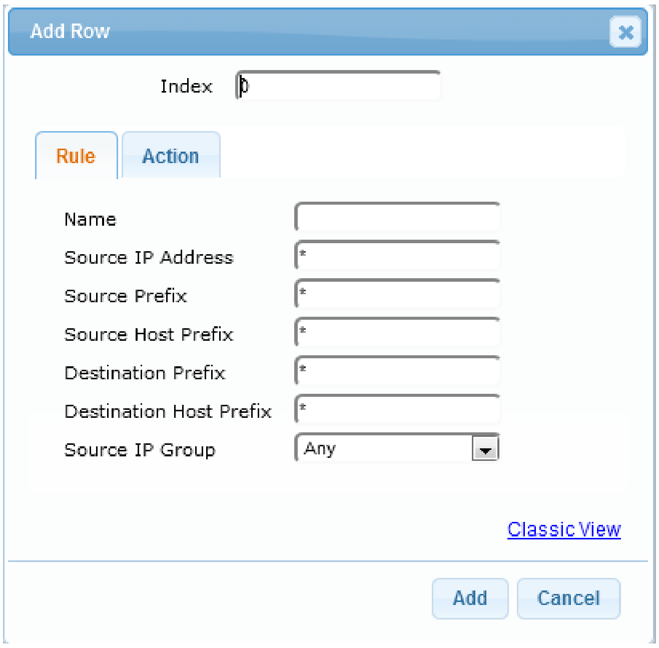
- **Destination Phone Number Manipulation Table for IP-to-Tel Calls table:** ini file table parameter, NumberMapIP2Tel or CLI command, configure voip/gw manipulations dst-number-map-ip2tel
- **Destination Phone Number Manipulation Table for Tel-to-IP Calls table:** ini file table parameter, NumberMapTel2IP or CLI command, configure voip/gw manipulations dst-number-map-tel2ip
- **Source Phone Number Manipulation Table for IP-to-Tel Calls table:** ini file table parameter, SourceNumberMapIP2Tel or CLI command, configure voip/gw manipulations src-number-map-ip2tel
- **Source Phone Number Manipulation Table for Tel-to-IP Calls table:** ini file table parameter, SourceNumberMapTel2IP or CLI command, configure voip/gw manipulations src-number-map-tel2ip

➢ To configure a number manipulation rule:

1. Open the required Number Manipulation page (**Configuration** tab > **VoIP** menu > **Gateway** > **Manipulations** > **Dest Number IP->Tel**, **Dest Number Tel->IP**, **Source Number IP->Tel**, or **Source Number Tel->IP**); the relevant Manipulation table page is displayed.

2. Click **Add**; the following dialog box appears:

Figure 23-2: Number Manipulation Table (Example) - Add Row Dialog Box



Index: 0

Name:
Source IP Address: *
Source Prefix: *
Source Host Prefix: *
Destination Prefix: *
Destination Host Prefix: *
Source IP Group: Any

3. Configure a number manipulation rule according to the parameters described in the table below.
4. Click **Add**.

The table below shows configuration examples of Tel-to-IP source phone number manipulation rules, where:

- **Rule 1:** When the destination number has the prefix 03 (e.g., 035000), source number prefix 201 (e.g., 20155), and from source IP Group ID 2, the source number is changed to, for example, 97120155.
- **Rule 2:** When the source number has prefix 1001 (e.g., 1001876), it is changed to 587623.
- **Rule 3:** When the source number has prefix 123451001 (e.g., 1234510012001), it is changed to 20018.
- **Rule 4:** When the source number has prefix from 30 to 40 and a digit (e.g., 3122), it is changed to 2312.
- **Rule 5:** When the destination number has the prefix 6, 7, or 8 (e.g., 85262146), source number prefix 2001, it is changed to 3146.

Parameter	Rule 1	Rule 2	Rule 3	Rule 4	Rule 5
Destination Prefix	03		*	*	[6,7,8]
Source Prefix	201	1001	123451001#	[30-40]x	2001
Stripped Digits from Left	-	4	-	-	5
Stripped Digits from Right	-	-	-	1	-
Prefix to Add	971	5	-	2	3
Suffix to Add	-	23	8	-	-

Parameter	Rule 1	Rule 2	Rule 3	Rule 4	Rule 5
Number of Digits to Leave	-	-	4	-	-
Presentation	Allowed	Restricted	-	-	-

Table 23-1: Number Manipulation Tables Parameter Descriptions

Parameter	Description
Index [_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name [_ManipulationName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 20 characters. By default, no value is defined.
Rule (Matching Characteristics)	
Source IP Address `src-ip-address` [_SourceAddress]	Defines the source IP address of the caller. This is obtained from the Contact header in the INVITE message. The default is the asterisk (*) wildcard (i.e., any address). **Notes:** ▪ The parameter is applicable only to the number manipulation tables for IP-to-Tel calls. ▪ The source IP address can include the 'x' wildcard to represent single digits. For example: 10.8.8.xx represents all IP addresses between 10.8.8.10 to 10.8.8.99. ▪ The source IP address can include the asterisk (*) wildcard to represent any number between 0 and 255. For example, 10.8.8.* represents all IP addresses between 10.8.8.0 and 10.8.8.255.
Destination IP Group `dst-ip-group-id` [_DestIPGroupID]	Defines the IP Group to where the call is sent. The default is Any (i.e., any IP Group). **Note:** The parameter is applicable only to the Destination Phone Number Manipulation Table for Tel -> IP Calls.
Source Trunk Group `src-trunk-group-id` [_SrcTrunkGroupID]	Defines the source Trunk Group ID for Tel-to-IP calls. The default is -1 (i.e., any Trunk Group). **Note:** The parameter is applicable only to the number manipulation tables for Tel-to-IP calls.
Source Prefix `src-prefix` [_SourcePrefix]	Defines the source (calling) telephone number prefix and/or suffix. You can use special notations for denoting the prefix. For example, [100-199](100,101,105) denotes a number that starts with 100 to 199 and ends with 100, 101 or 105. You can also use the $ sign to denote calls without a calling number. For a description of available notations, see "Dialing Plan Notation for Routing and Manipulation Tables" on page 809. The default is the asterisk (*) wildcard (i.e., any prefix).

Parameter	Description
Source Host Prefix `src-host-prefix` [_SrcHost]	Defines the URI host name prefix of the incoming SIP INVITE message in the From header. The default is the asterisk (*) wildcard (i.e., any prefix). **Notes:** ▪ The parameter is applicable only to the number manipulation tables for IP-to-Tel calls. ▪ If the P-Asserted-Identity header is present in the incoming INVITE message, then the value of the parameter is compared to the P-Asserted-Identity URI host name (instead of the From header).
Destination Prefix `dst-prefix` [_DestinationPrefix]	Defines the destination (called) telephone number prefix and/or suffix. You can use special notations for denoting the prefix. For example, [100-199](100,101,105) denotes a number that starts with 100 to 199 and ends with 100, 101 or 105. You can also use the $ sign to denote calls without a called number. For a description of available notations, see "Dialing Plan Notation for Routing and Manipulation Tables" on page 809. The default is the asterisk (*) wildcard (i.e., any prefix).
Destination Host Prefix `dst-host-prefix` [_DestHost]	Defines the Request-URI host name prefix of the incoming SIP INVITE message. The default is the asterisk (*) wildcard (i.e., any prefix). **Note:** The parameter is applicable only to the number manipulation tables for IP-to-Tel calls.
Source IP Group `src-ip-group-id` [_SrcIPGroupID]	Defines the IP Group from where the IP call originated. Typically, the IP Group of an incoming INVITE is determined by the Inbound IP Routing table. The default is Any (i.e., any IP Group). **Note:** The parameter is applicable only to the number manipulation tables for IP-to-Tel calls.
Operation (Action)	
Stripped Digits From Left `remove-from-left` [_RemoveFromLeft]	Defines the number of digits to remove from the left of the telephone number prefix. For example, if you enter 3 and the phone number is 5551234, the new phone number is 1234.
Stripped Digits From Right `remove-from-right` [RemoveFromRight]	Defines the number of digits to remove from the right of the telephone number prefix. For example, if you enter 3 and the phone number is 5551234, the new phone number is 5551.
Number of Digits to Leave `num-of-digits-to-leave` [LeaveFromRight]	Defines the number of digits that you want to keep from the right of the phone number. For example, if you enter 4 and the phone number is 00165751234, then the new number is 1234.
Prefix to Add `prefix-to-add` **[Prefix2Add]**	Defines the number or string that you want added to the front of the telephone number. For example, if you enter 9 and the phone number is 1234, the new number is 91234.
Suffix to Add `suffix-to-add` [Suffix2Add]	Defines the number or string that you want added to the end of the telephone number. For example, if you enter 00 and the phone number is 1234, the new number is 123400.

Parameter	Description
TON ton [NumberType]	Defines the Type of Number (TON). - If you selected 'Unknown' for the NPI, you can select Unknown [0]. - If you selected 'Private' for the NPI, you can select Unknown [0], Level 2 Regional [1], Level 1 Regional [2], PISN Specific [3] or Level 0 Regional (Local) [4]. - If you selected 'E.164 Public' for the NPI, you can select Unknown [0], International [1], National [2], Network Specific [3], Subscriber [4] or Abbreviated [6]. The default is 'Unknown'. **Notes:** - The parameter is applicable only to number manipulation tables for IP-to-Tel calls. - TON can be used in the SIP Remote-Party-ID header by using the EnableRPIHeader and AddTON2RPI parameters. - For more information on available NPI/TON values, see Numbering Plans and Type of Number on page 465.
NPI npi [NumberPlan]	Defines the Numbering Plan Indicator (NPI). - [0] Unknown (default) - [9] Private - [1] E.164 Public - [-1] Not Configured = value received from PSTN/IP is used **Notes:** - The parameter is applicable only to number manipulation tables for IP-to-Tel calls. - NPI can be used in the SIP Remote-Party-ID header by using the EnableRPIHeader and AddTON2RPI parameters. - For more information on available NPI/TON values, see Numbering Plans and Type of Number on page 465.
Presentation is-presentation-restricted [IsPresentationRestricted]	Enables caller ID. - - **[0]** Allowed = Sends Caller ID information when a call is made using these destination/source prefixes. - **[1]** Restricted = Restricts Caller ID information for these prefixes. **Notes:** - This field is applicable only to number manipulation tables for source phone number manipulation. - If this field is set to **Restricted** and the 'Asserted Identity Mode' (AssertedIdMode) parameter is set to **Add P-Asserted-Identity**, the From header in the INVITE message includes the following: From: 'anonymous' <sip: anonymous@anonymous.invalid> and 'privacy: id' header.

23.3 Manipulating Number Prefix

The device supports a notation for adding a prefix where part of the prefix is first extracted from a user-defined location in the original destination or source number. This notation is entered in the 'Prefix to Add' field in the Number Manipulation tables (see "Configuring Source/Destination Number Manipulation" on page 441): x[n,l]y...

where,

- *x* = any number of characters/digits to add at the beginning of the number (i.e. first digits in the prefix).
- *[n,l]* = defines the location in the original destination or source number where the digits *y* are added:
  - *n* = location (number of digits counted from the left of the number) of a specific string in the original destination or source number.
  - *l* = number of digits that this string includes.
- y = prefix to add at the specified location.

For example, assume that you want to manipulate an incoming IP call with destination number +5492028888888 (area code 202 and phone number 8888888) to the number 0202158888888. To perform such a manipulation, the following configuration is required in the Number Manipulation table:

1. The following notation is used in the 'Prefix to Add' field:

   0[5,3]15

   where,

   - 0 is the number to add at the beginning of the original destination number.
   - [5,3] denotes a string that is located after (and including) the fifth character (i.e., the first '2' in the example) of the original destination number, and its length being three digits (i.e., the area code 202, in the example).
   - 15 is the number to add immediately after the string denoted by [5,3] - in other words, 15 is added after (i.e. to the right of) the digits 202.
2. The first seven digits from the left are removed from the original number, by entering "7" in the 'Stripped Digits From Left' field.

Table 23-2: Example of Configured Rule for Manipulating Prefix using Special Notation

Parameter	Rule 1
Destination Prefix	+5492028888888
Source Prefix	*
Source IP Address	*
Stripped Digits from Left	7
Prefix to Add	0[5,3]15

In this configuration example, the following manipulation process occurs:

1. The prefix is calculated as 020215.
2. The first seven digits from the left are removed from the original number, thereby changing the number to 8888888.
3. The prefix that was previously calculated is then added.

23.4 SIP Calling Name Manipulations

The calling name manipulation tables lets you configure up to 120 manipulation rules for manipulating the calling name (i.e., caller ID) in SIP messages, for IP-to-Tel and Tel-to-IP calls. Manipulation includes modifying or removing the calling name. The calling name manipulation tables include the following:

- Calling Name Manipulation Table for IP-to-Tel Calls table
- Calling Name Manipulation Table for Tel-to-IP Calls table

For example, assume that an incoming SIP INVITE message includes the following header:

```
P-Asserted-Identity: "company:john" sip:6666@78.97.79.104
```

Using the Calling Name Manipulation Table for IP-to-Tel table, the text "company" can be changed to "worker" in the outgoing INVITE, as shown below:

```
P-Asserted-Identity: "worker:john" sip:996666@10.13.83.10
```

Configuration of calling name manipulation rules includes two areas:

- **Rule:** Defines the matching characteristics of an incoming call (e.g., prefix of destination number).
- **Action:** Defines the action that is done if the incoming call matches the characteristics of the rule (e.g., removes a user-defined number of digits from the left of the calling name).

The device searches the table from top to bottom for the first rule that matches the characteristics of the incoming call. If it finds a matching rule, it applies the manipulation configured for that rule.

Note: For using the Calling Name Manipulation Table for Tel-to-IP Calls table for retrieving the calling name (display name) from an Active Directory using LDAP queries, see Querying the AD for Calling Name on page 260.

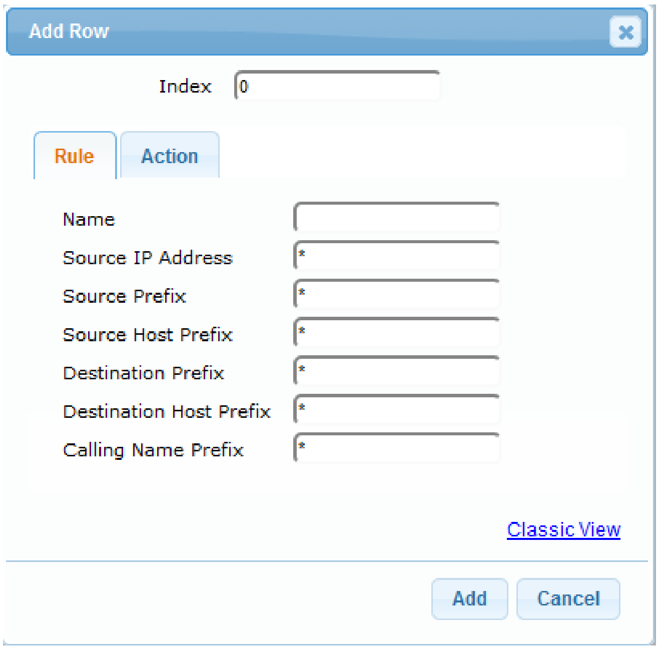
The following procedure describes how to configure calling name manipulation rules through the Web interface. You can also configure these rules using the the following management tools:

- Calling Name Manipulation Table for Tel-to-IP Calls table: *ini* file (CallingNameMapTel2Ip) or CLI (configure voip/gw manipulations calling-name-map-tel2ip)
- Calling Name Manipulation Table for IP-to-Tel Calls table: *ini* file (CallingNameMapIp2Tel) or CLI (configure voip/gw manipulations calling-name-map-ip2tel)

➢ **To configure calling name manipulation rules:**

1. Open the required calling name manipulations page (**Configuration** tab > **VoIP** menu > **Gateway** > **Manipulations** > **Calling Name IP->Tel** or **Calling Name Tel->IP**).
2. Click **Add**; the following dialog box appears:

Figure 23-3: Calling Name Manipulation Table (Example) - Add Row Dialog Box



Index	0
Name	
Source IP Address	*
Source Prefix	*
Source Host Prefix	*
Destination Prefix	*
Destination Host Prefix	*
Calling Name Prefix	*

3. Configure a manipulation rule according to the parameters described in the table below.
4. Click **Add**.

Table 23-3: Calling Name Manipulation Tables Parameter Descriptions

Parameter	Description
Index [_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `manipulation-name` [_ManipulationName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 20 characters.
Rule (Matching Characteristics)	
Destination Prefix `dst-prefix` [_DestinationPrefix]	Defines the destination (called) telephone number prefix and/or suffix. You can use special notations for denoting the prefix. For example, [100-199](100,101,105) denotes a number that starts with 100 to 199 and ends with 100, 101 or 105. You can also use the $ sign to denote calls without a called number. For a description of available notations, see "Dialing Plan Notation for Routing and Manipulation Tables" on page 809. The default value is the asterisk (*) symbol (i.e., any destination prefix).
Source Prefix `src-prefix` [_SourcePrefix]	Defines the source (calling) telephone number prefix and/or suffix. You can use special notations for denoting the prefix. For example, [100-199](100,101,105) denotes a number that starts with 100 to 199 and ends with 100, 101 or 105. You can also use the $ sign to denote calls without a calling number. For a description of available notations,

Parameter	Description
	see "Dialing Plan Notation for Routing and Manipulation Tables" on page 809. The default value is the asterisk (*) symbol (i.e., any source prefix).
Calling Name Prefix `calling-name-prefix` [_CallingNamePrefix]	Defines the caller name (i.e., caller ID) prefix. You can use special notations for denoting the prefix. For example, to denote calls without a calling name, use the $ sign. For a description of available notations, see "Dialing Plan Notation for Routing and Manipulation Tables" on page 809. The default value is the asterisk (*) symbol (i.e., any calling name prefix).
Source Trunk Group ID `src-trunk-group-id` [_SrcTrunkGroupID]	Defines the source Trunk Group ID from where the Tel-to-IP call was received. The default value is -1, which denotes any Trunk Group. **Note:** The parameter is applicable only to the Calling Name Manipulation Table for Tel-to-IP Calls table.
Source IP Address `src-ip-address` [_SourceAddress]	Defines the source IP address of the caller, for IP-to-Tel calls. The source IP address appears in the SIP Contact header in the INVITE message. The default value is the asterisk (*) symbol (i.e., any IP address). The source IP address can include the following wildcards: ▪ "x" wildcard: represents single digits. For example, 10.8.8.xx represents all IP addresses between 10.8.8.10 to 10.8.8.99. ▪ "*" (asterisk) wildcard: represents any number between 0 and 255. For example, 10.8.8.* represents all IP addresses between 10.8.8.0 and 10.8.8.255. **Note:** The parameter is applicable only to the Calling Name Manipulation Table for IP-to-Tel Calls table.
Source Host Prefix `src-host-prefix` [_SrcHost]	Defines the URI host name prefix of the incoming SIP INVITE message in the From header. The default value is the asterisk (*) symbol (i.e., any source host prefix). **Notes:** ▪ The parameter is applicable only to the Calling Name Manipulation Table for IP-to-Tel Calls table. ▪ If the P-Asserted-Identity header is present in the incoming INVITE message, the value of the parameter is compared to the P-Asserted-Identity URI host name (instead of the From header).
Destination Host Prefix `dst-host-prefix` [_DestHost]	Defines the Request-URI host name prefix of the incoming SIP INVITE message. The default value is the asterisk (*) symbol (i.e., any destination host prefix). **Note:** The parameter is applicable only to the Calling Name Manipulation Table for IP-to-Tel Calls table.
Operation (Action)	
Stripped Characters From Left `remove-from-left` [_RemoveFromLeft]	Defines the number of characters to remove from the left of the calling name. For example, if you enter 3 and the calling name is "company:john", the new calling name is "pany:john".

Parameter	Description
Stripped Characters From Right `remove-from-right` [_RemoveFromRight]	Defines the number of characters to remove from the right of the calling name. For example, if you enter 3 and the calling name is "company:name", the new name is "company:n".
Number of Characters to Leave `num-of-digits-to-leave` [LeaveFromRight]	Defines the number of characters that you want to keep from the right of the calling name. For example, if you enter 4 and the calling name is "company:name", the new name is "name".
Prefix to Add `prefix-to-add` [_Prefix2Add]	Defines the number or string to add at the front of the calling name. For example, if you enter ITSP and the calling name is "company:name", the new name is ITSPcompany:john".
Suffix to Add `suffix-to-add` [_Suffix2Add]	Defines the number or string to add at the end of the calling name. For example, if you enter 00 and calling name is "company:name", the new name is "company:name00".

23.5 Configuring Redirect Number IP to Tel

The redirect number manipulation tables let you configure rules for manipulating the redirect number received in SIP messages. The redirect number manipulation tables include:

- Redirect Number IP-to-Tel table: This table defines IP-to-Tel redirect number manipulation. You can manipulate the value of the received SIP Diversion, Resource-Priority, or History-Info headers, which is then added to the Redirecting Number Information Element (IE) in the ISDN Setup message sent to the Tel side. This also includes the reason for the call redirection. This is configured in the Redirect Number IP-to-Tel table.
- Redirect Number Tel to IP table: This table defines Tel-to-IP redirect number manipulation. You can manipulate the prefix of the redirect number, received from the Tel side, in the outgoing SIP Diversion, Resource-Priority, or History-Info headers sent to the IP side. This is configured in the Redirect Number Tel-to-IP table.

Configuration of redirect number manipulation rules includes two areas:

- Rule: Defines the matching characteristics of an incoming call (e.g., prefix of redirect number).
- Action: Defines the action that is done if the incoming call matches the characteristics of the rule (e.g., removes a user-defined number of digits from the left of the redirect number).
- The device searches the table from top to bottom for the first rule that matches the characteristics of the incoming call. If it finds a matching rule, it applies the manipulation configured for that rule..

Notes:

- If the device copies the received destination number to the outgoing SIP redirect number (enabled by the CopyDest2RedirectNumber parameter), no redirect number Tel-to-IP manipulation is done.
- The manipulation rules are done in the following order: Stripped Digits From Left, Stripped Digits From Right, Number of Digits to Leave, Prefix to Add, and then Suffix to Add.
- The device uses the 'Redirect Prefix' parameter before it manipulates the prefix.

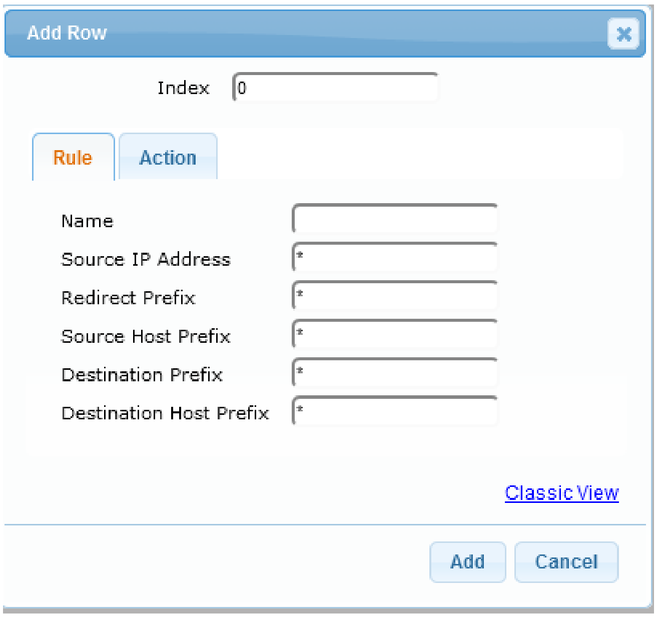
The following procedure describes how to configure redirect number manipulation rules through the Web interface. You can also configure these rules using the following management tools:

- Redirect Number IP to Tel table: ini file (RedirectNumberMapIp2Tel) or CLI (configure voip/gw manipulations redirect-number-map-ip2tel)
- Redirect Number Tel to IP table: ini file (RedirectNumberMapTel2Ip) or CLI (configure voip/gw manipulations redirect-number-map-tel2ip)

➢ To configure a redirect number manipulation rule:

1. Open the redirect number manipulation table (**Configuration** tab > **VoIP** menu > **Gateway** > **Manipulations** > **Redirect Number Tel > IP** or Redirect Number IP > Tel).
2. Click **Add**; the following dialog box appears (e.g., Redirect Number Tel-to-IP table):

Figure 23-4: Redirect Number Manipulation Table (Example) - Add Row Dialog Box



Index: 0

Name:

Source IP Address: *

Redirect Prefix: *

Source Host Prefix: *

Destination Prefix: *

Destination Host Prefix: *

3. Configure a manipulation rule according to the parameters described in the table below.
4. Click **Add**.

Table 23-4: Redirect Number Manipulation Tables Parameter Description

Parameter	Description
Index [_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.

Parameter	Description
Name `manipulation-name` [_ManipulationName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 20 characters.
Rule (Matching Characteristics)	
Destination Prefix `dst-prefix` [_DestinationPrefix]	Defines the destination (called) telephone number prefix. The default value is the asterisk (*) symbol (i.e., any number). For manipulating the diverting and redirected numbers for call diversion, you can use the strings "DN" and "RN" to denote the destination prefix of these numbers. For more information, see Manipulating Redirected and Diverted Numbers for Call Diversion on page 455.
Redirect Prefix `redirect-prefix` [_RedirectPrefix]	Defines the redirect telephone number prefix. The default value is the asterisk (*) symbol (i.e., any number prefix).
Source Trunk Group ID `src-trunk-group-id` [_SrcTrunkGroupID]	Defines the Trunk Group from where the Tel call is received. To denote any Trunk Group, leave this field empty. The value -1 indicates that this field is ignored in the rule. **Note:** The parameter is applicable only to the Redirect Number Tel-to-IP table.
Source IP Address `src-ip-address` [_SourceAddress]	Defines the IP address of the caller. The IP address appears in the SIP Contact header of the incoming INVITE message. The default value is the asterisk (*) symbol (i.e., any IP address). The value can include the following wildcards: ▪ "x": represents single digits, for example, 10.8.8.xx denotes all addresses between 10.8.8.10 and 10.8.8.99. ▪ "*": represents any number between 0 and 255, for example, 10.8.8.* denotes all addresses between 10.8.8.0 and 10.8.8.255. **Note:** The parameter is applicable only to the Redirect Number IP-to-Tel table.
Source Host Prefix `src-host-prefix` [_SrcHost]	Defines the URI host name prefix of the caller. The host name appears in the SIP From header of the incoming SIP INVITE message. The default value is the asterisk (*) symbol (i.e., any host name prefix). **Notes:** ▪ The parameter is applicable only to the Redirect Number IP-to-Tel table. ▪ If the P-Asserted-Identity header is present in the incoming INVITE message, the value of the parameter is compared to the P-Asserted-Identity URI host name (instead of to the From header).
Destination Host Prefix `dst-host-prefix` [_DestHost]	Defines the Request-URI host name prefix, which appears in the incoming SIP INVITE message. The default value is the asterisk (*) symbol (i.e., any prefix). **Note:** The parameter is applicable only to the Redirect Number IP-to-Tel table.
Operation (Action)	

Parameter	Description
Stripped Digits From Left `remove-from-left` [_RemoveFromLeft]	Defines the number of digits to remove from the left of the redirect number prefix. For example, if you enter 3 and the redirect number is 5551234, the new number is 1234.
Stripped Digits From Right `remove-from-right` [_RemoveFromRight]	Defines the number of digits to remove from the right of the redirect number prefix. For example, if you enter 3 and the redirect number is 5551234, the new number is 5551.
Number of Digits to Leave `num-of-digits-to-leave` [_LeaveFromRight]	Defines the number of digits that you want to retain from the right of the redirect number.
Prefix to Add `prefix-to-add` [_Prefix2Add]	Defines the number or string that you want added to the front of the redirect number. For example, if you enter 9 and the redirect number is 1234, the new number is 91234.
Suffix to Add `suffix-to-add` [_Suffix2Add]	Defines the number or string that you want added to the end of the redirect number. For example, if you enter 00 and the redirect number is 1234, the new number is 123400.
TON `ton` [_NumberType]	Defines the Type of Number (TON). The default is Not Configured [-1]. ▪ If NPI is set to Unknown, you can set TON to Unknown [0]. ▪ If NPI is set to Private, you can set TON to Unknown [0], International [1], National [2], Network Specific [3] or Subscriber [4]. ▪ If NPI is set to E.164 Public, you can set TON to Unknown [0], International [1], National [2], Network Specific [3], Subscriber [4] or Abbreviated [6]. For more information on available NPI/TON values, see Numbering Plans and Type of Number on page 465.
NPI `npi` [_NumberPlan]	Defines the Numbering Plan Indicator (NPI). ▪ [-1] Not Configured = (Default) Value received from PSTN/IP is used ▪ [0] Unknown ▪ [1] E.164 Public ▪ [9] Private For more information on available NPI/TON values, see Numbering Plans and Type of Number on page 465.
Presentation `is-presentation-restricted` [_IsPresentationRestricted]	Enables caller ID. ▪ ▪ **[0]** Allowed = Sends Caller ID information when a call is made using these destination / source prefixes. ▪ **[1]** Restricted = Restricts Caller ID information for these prefixes. **Note:** If the parameter is set to **Restricted** and the 'AssertedIdMode' parameter is set to **Add P-Asserted-Identity,** the From header in the INVITE message includes the following: `From: 'anonymous' <sip:anonymous@anonymous.invalid> and 'privacy: id' header.`

23.6 Manipulating Redirected and Diverted Numbers for Call Diversion

You can configure manipulation rules to manipulate the Diverted-to and Diverting numbers received in the incoming Call Redirection Facility message for call diversion, which is interworked to outgoing SIP 302 responses. This feature is applicable to the Euro ISDN and QSIG variants, and to IP-to-Tel calls.

The incoming redirection Facility message includes, among other parameters, the Diverted-to number and Diverting number. The Diverted-to number (i.e., new destination) is mapped to the user part in the Contact header of the SIP 302 response. The Diverting number is mapped to the user part in the Diversion header of the SIP 302 response.

These two numbers can be manipulated by entering the following special strings in the 'Destination Prefix' field of the Redirect Number Tel-to-IP table:

- "RN" - used in the rule to manipulate the Redirected number (i.e., originally called number or Diverting number).
- "DN" - used in the rule to manipulate the Diverted-to number (i.e., the new called number or destination). This manipulation is done on the user part in the Contact header of the SIP 302 response.

For example, assume the following required manipulation:

- Manipulate Redirected number 6001 (originally called number) to 6005
- Manipulate Diverted-to number 8002 (the new called number or destination) to 8005

The configuration in the Redirect Number Tel-to-IP table is as follows:

Table 23-5: Redirect Number Configuration Example

Parameter	Rule 1	Rule 2
Destination Prefix	RN	DN
Redirect Prefix	6	8
Stripped Digits From Right	1	1
Suffix to Add	5	5
Number of Digits to Leave	5	-

After the above manipulation is done, the device sends the following outgoing SIP 302 response:

```
SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/TLS 10.33.45.68;branch=z9hG4bKac54132643;alias
From: "MP118 1" <sip:8001@10.33.45.68>;tag=1c54119560
To: <sip:6001@10.33.45.69;user=phone>;tag=1c664560944
Call-ID: 541189832710201115142@10.33.45.68
CSeq: 1 INVITE
Contact: <sip:8005@10.33.45.68;user=phone>
Supported: em,timer,replaces,path,early-session,resource-priority
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUB
SCRIBE,UPDATE
Diversion: <tel:6005>;reason=unknown;counter=1
Server: Audiocodes-Sip-Gateway-IPmedia 260_UN/v.6.80A.227.005
Reason: SIP ;cause=302 ;text="302 Moved Temporarily"
Content-Length: 0
```

23.7 Mapping NPI/TON to SIP Phone-Context

The Phone Context table lets you configure rules for mapping the Numbering Plan Indication (NPI) and Type of Number (TON) to the SIP 'phone-context' parameter, and vice versa. The 'phone-context' parameter appears in the standard SIP headers where a phone number is used (i.e., Request-URI, To, From, and Diversion). When a call is received from the ISDN side, the NPI and TON are compared against the table and the matching 'phone-context' value is used in the outgoing SIP INVITE message. The same mapping occurs when an INVITE with a 'phone-context' parameter is received.

For example, for a Tel-to-IP call with NPI/TON set as E164 National (values 1/2), the device can send the following SIP INVITE URI:

```
sip:12365432;phone-context= na.e.164.nt.com
```

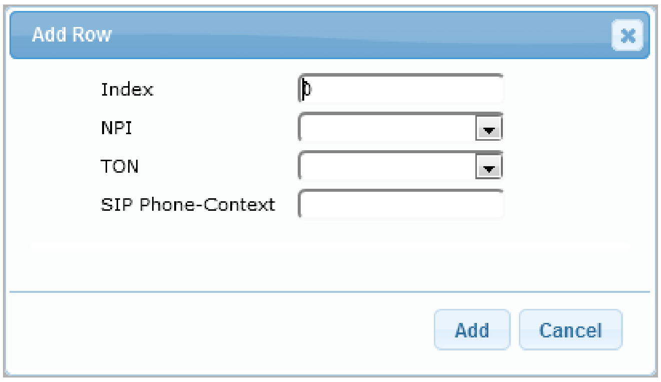
For an IP-to-Tel call, if the incoming INVITE contains this 'phone-context' (e.g. "phone-context= na.e.164.nt.com"), the NPI/TON of the called number in the outgoing Setup message is changed to E164 National.

The following procedure describes how to configure NPI/TON-SIP phone-context mapping rules through the Web interface. You can also configure it through ini file (PhoneContext) or CLI (configure voip > gw manipulations phone-context-table).

➢ To configure NPI/TON-SIP phone-context mapping rules:

1. Open the Phone Context table (**Configuration** tab > **VoIP** menu > **Gateway** > **Manipulations** > **Phone Context**).
2. Click **Add**; the following dialog box appears:

Figure 23-5: Phone Context Table - Add Row Dialog Box



Index	0
NPI	
TON	
SIP Phone-Context	

3. Configure a mapping rule according to the parameters described in the table below.
4. To add the incoming SIP 'phone-context' parameter as a prefix to the outgoing ISDN Setup message with called and calling numbers, from the 'Add Phone Context As Prefix' drop-down list (AddPhoneContextAsPrefix), select **Enable**.
5. Click **Add**.

Note: You can configure multiple rows with the same NPI/TON or same SIP 'phone-context'. In such a configuration, a Tel-to-IP call uses the first matching rule in the table.

Table 23-6: Phone Context Table Parameter Description

Parameter	Description
Index [PhoneContext_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.

Parameter	Description
NPI `npi` [PhoneContext_Npi]	Defines the Number Plan Indicator (NPI). ▪ **[0]** Unknown (default) ▪ **[1]** E.164 Public ▪ **[9]** Private For a detailed list of the available NPI/TON values, see Numbering Plans and Type of Number on page 465.
TON `ton` [PhoneContext_Ton]	Defines the Type of Number (TON). ▪ If you selected Unknown as the NPI, you can select Unknown **[0]**. ▪ If you selected Private as the NPI, you can select one of the following: ✓ **[0]** Unknown ✓ **[1]** Level 2 Regional ✓ **[2]** Level 1 Regional ✓ **[3]** PSTN Specific ✓ **[4]** Level 0 Regional (Local) ▪ If you selected E.164 Public as the NPI, you can select one of the following: ✓ **[0]** Unknown ✓ **[1]** International ✓ **[2]** National ✓ **[3]** Network Specific ✓ **[4]** Subscriber ✓ **[6]** Abbreviated
SIP Phone Context `context` [PhoneContext_Context]	Defines the SIP 'phone-context' URI parameter.

23.8 Configuring Release Cause Mapping

When a call is disconnected, the reason for the disconnection (or call failure) is sent by the side (IP or Tel) on which the call disconnection occurred. From the IP side, a SIP response is sent (e.g., 406); from the Tel side, an ISDN cause code is sent (e.g., 6). You can configure ISDN-SIP release cause mapping rules, as discussed in this section.

23.8.1 SIP-to-ISDN Release Cause Mapping

23.8.1.1 Configuring SIP-to-ISDN Release Cause Mapping

The Release Cause Mapping from SIP to ISDN table lets you configure up to 12 SIP response code to ISDN ITU-T Q.850 release cause code (call failure) mapping rules. The table lets you override the default SIP-to-ISDN release cause mappings, listed in "Fixed Mapping of SIP Response to ISDN Release Reason" on page 458. When the device receives a SIP response from the IP side, it searches the table for a matching SIP response. If found, the device sends the corresponding Q.850 Release Cause to the PSTN. If the SIP response is not configured in the table, the default, fixed SIP-to-ISDN release reason mapping is used.

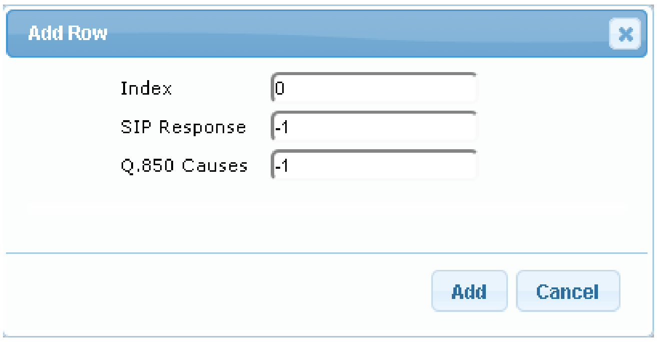
Note: For Tel-to-IP calls, you can also map less commonly used SIP responses to a single, default ISDN release cause code, using the DefaultCauseMapISDN2IP parameter. The parameter defines a default ISDN cause code that is always used, except when the following Release Causes are received: Normal Call Clearing (16), User Busy (17), No User Responding (18) or No Answer from User (19).

The following procedure describes how to configure SIP-to-ISDN release cause mapping through the Web interface. You can also configure it through ini file (CauseMapSIP2ISDN) or CLI (configure voip > gw manipulations cause-map-sip2isdn).

➢ **To configure a SIP-to-ISDN release cause mapping rule:**

1. Open the Release Cause Mapping from SIP to ISDN table (**Configuration** tab > **VoIP** menu > **Gateway** > **Manipulations** > **Release Cause SIP > ISDN**).
2. Click **Add**; the following dialog box appears:

Figure 23-6: Release Cause Mapping from SIP to ISDN Table - Add Row Dialog Box



Index: 0
SIP Response: -1
Q.850 Causes: -1

3. Configure a mapping rule according to the parameters described in the table below.
4. Click **Add**.

Table 23-7: Release Cause Mapping from SIP to ISDN Table Parameter Descriptions

Parameter	Description
Index [CauseMapSip2Isdn_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
SIP Response `sip-response` [CauseMapSip2Isdn_SipResponse]	Defines the SIP response code. For example, you can enter "406" (without apostrophes) to represent the SIP 406 Not Acceptable response.
Q.850 Causes `q850-causes` [CauseMapSip2Isdn_IsdnReleaseCause]	Defines the ISDN Q.850 cause code. For example, you can enter "6" (without apostrophes) to represent Cause Code 6 Channel Unacceptable.

23.8.1.2 Fixed Mapping of SIP Response to ISDN Release Reason

The following table describes the mapping of SIP response to ISDN release reason.

Table 23-8: Mapping of SIP Response to ISDN Release Reason

SIP Response	Description	ISDN Release Reason	Description
400*	Bad request	31	Normal, unspecified

SIP Response	Description	ISDN Release Reason	Description
401	Unauthorized	21	Call rejected
402	Payment required	21	Call rejected
403	Forbidden	21	Call rejected
404	Not found	1	Unallocated number
405	Method not allowed	63	Service/option unavailable
406	Not acceptable	79	Service/option not implemented
407	Proxy authentication required	21	Call rejected
408	Request timeout	102	Recovery on timer expiry
409	Conflict	41	Temporary failure
410	Gone	22	Number changed w/o diagnostic
411	Length required	127	Interworking
413	Request entity too long	127	Interworking
414	Request URI too long	127	Interworking
415	Unsupported media type	79	Service/option not implemented
417	Unknown Resource Priority	127	Interworking
420	Bad extension	127	Interworking
480	Temporarily unavailable	18	No user responding
481*	Call leg/transaction doesn't exist	127	Interworking
482*	Loop detected	127	Interworking
483	Too many hops	127	Interworking
484	Address incomplete	28	Invalid number format
485	Ambiguous	1	Unallocated number
486	Busy here	17	User busy
488	Not acceptable here	31	Normal, unspecified
489	Bad Event	31	Normal, unspecified
491	Request Pending	31	Normal, unspecified
500	Server internal error	41	Temporary failure
501	Not implemented	38	Network out of order
502	Bad gateway	38	Network out of order
503	Service unavailable	41	Temporary failure
504	Server timeout	102	Recovery on timer expiry
505*	Version not supported	127	Interworking
600	Busy everywhere	17	User busy
603	Decline	21	Call rejected

SIP Response	Description	ISDN Release Reason	Description
604	Does not exist anywhere	1	Unallocated number
606*	Not acceptable	38	Network out of order

***** Messages and responses were created because the ‘ISUP to SIP Mapping’ draft does not specify their cause code mapping.

23.8.2 ISDN-to-SIP Release Cause Mapping

23.8.2.1 Configuring ISDN-to-SIP Release Cause Mapping

The Release Cause Mapping from ISDN to SIP table lets you configure up to 12 ISDN ITU-T Q.850 release cause code (call failure) to SIP response code mapping rules. The table lets you override the default ISDN-to-SIP release cause mappings, listed in "Fixed Mapping of ISDN Release Reason to SIP Response" on page 461. When the device receives an ISDN cause code from the PSTN side, it searches the table for a matching ISDN cause code. If found, the device sends the corresponding SIP response to the IP. If the ISDN cause code is not configured in the table, the default, fixed ISDN-to-SIP release reason mapping is used.

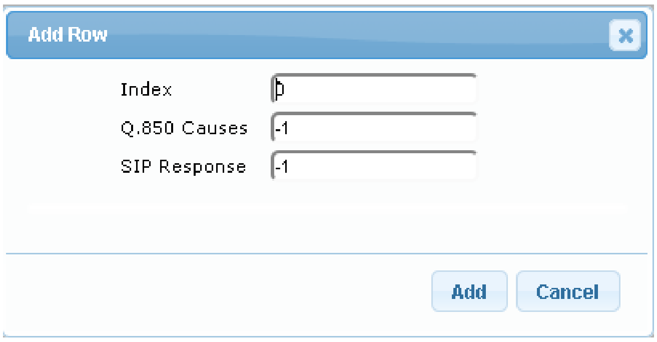
Note: You can change the originally received ISDN cause code to any other ISDN cause code, using the Release Cause ISDN to ISDN table (see "Configuring ISDN-to-ISDN Release Cause Mapping" on page 463). If the originally received ISDN cause code appears in both the Release Cause ISDN to ISDN table and the Release Cause Mapping ISDN to SIP table, the mapping rule in the Release Cause Mapping ISDN to SIP table is ignored. The device only uses a mapping rule that matches the new ISDN cause code.

The following procedure describes how to configure ISDN-to-SIP release cause mapping through the Web interface. You can also configure it through ini file (CauseMapISDN2SIP) or CLI (configure voip > gw manipulations cause-map-isdn2sip).

➢ **To configure a ISDN-to-SIP release cause mapping rule:**

1. Open the Release Cause Mapping from ISDN to SIP table (**Configuration** tab > **VoIP** menu > **Gateway** > **Manipulations** > **Release Cause ISDN > SIP**).
2. Click **Add**; the following dialog box appears:

Figure 23-7: Release Cause Mapping from ISDN to SIP Table - Add Row Dialog Box



Add Row

Index: 0

Q.850 Causes: -1

SIP Response: -1

3. Configure a mapping rule according to the parameters described in the table below.

4. Click **Add**.

Table 23-9: Release Cause Mapping from ISDN to SIP Table Parameter Descriptions

Parameter	Description
Index [CauseMapIsdn2Sip_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Q.850 Causes `q850-causes` [CauseMapIsdn2Sip_IsdnReleaseCause]	Defines the ISDN Q.850 cause code. For example, you can enter "6" (without apostrophes) to represent Cause Code 6 Channel Unacceptable.
SIP Response `sip-response` [CauseMapIsdn2Sip_SipResponse]	Defines the SIP response code. For example, you can enter "406" (without apostrophes) to represent the SIP 406 Not Acceptable response.

23.8.2.2 Fixed Mapping of ISDN Release Reason to SIP Response

The following table describes the mapping of ISDN release reason to SIP response.

Table 23-10: Mapping of ISDN Release Reason to SIP Response

ISDN Release Reason	Description	SIP Response	Description
1	Unallocated number	404	Not found
2	No route to network	404	Not found
3	No route to destination	404	Not found
4	Send Special Information Tone	400	Bad Request
5	Misdialed Trunk Prefix	400	Bad Request
6	Channel unacceptable	406*	Not acceptable
7	Call awarded and being delivered in an established channel	500	Server internal error
8	Preemption	480	Temporarily unavailable
9	Preemption - Circuit Reserved for Reuse	488	Not Acceptable Here
16	Normal call clearing	-*	BYE
17	User busy	486	Busy here
18	No user responding	408	Request timeout
19	No answer from the user	480	Temporarily unavailable
21	Call rejected	403	Forbidden
22	Number changed w/o diagnostic	410	Gone
23	Redirection	400	Bad Request
25	Exchange Routing Error	400	Bad Request
26	Non-selected user clearing	404	Not found
27	Destination out of order	502	Bad gateway

ISDN Release Reason	Description	SIP Response	Description
28	Address incomplete	484	Address incomplete
29	Facility rejected	501	Not implemented
30	Response to status enquiry	501*	Not implemented
31	Normal unspecified	480	Temporarily unavailable
32	Circuit Congestion	500	Server internal error
33	User Congestion	500	Server internal error
34	No circuit available	503	Service unavailable
38	Network out of order	503	Service unavailable
39	Permanent Frame Mode Connection Out-of-Service	503	Service unavailable
40	Permanent Frame Mode Connection Operational	503	Service unavailable
41	Temporary failure	503	Service unavailable
42	Switching equipment congestion	503	Service unavailable
43	Access information discarded	502*	Bad gateway
44	Requested channel not available	503*	Service unavailable
46	Precedence Call Blocked	488	Not Acceptable Here
47	Resource unavailable	503	Service unavailable
49	QoS unavailable	503*	Service unavailable
50	Facility not subscribed	503*	Service unavailable
53	Outgoing Calls Barred within CUG	488	Not Acceptable Here
55	Incoming calls barred within CUG	403	Forbidden
57	Bearer capability not authorized	403	Forbidden
58	Bearer capability not presently available	503	Service unavailable
62	Inconsistency In Outgoing Information Element	503	Service unavailable
63	Service/option not available	503*	Service unavailable
65	Bearer capability not implemented	501	Not implemented
66	Channel type not implemented	480*	Temporarily unavailable
69	Requested facility not implemented	503*	Service unavailable
70	Only restricted digital information bearer capability is available	503*	Service unavailable
79	Service or option not implemented	501	Not implemented
81	Invalid call reference value	502*	Bad gateway
82	Identified channel does not exist	502*	Bad gateway
83	Suspended call exists, but this call identity does not	503*	Service unavailable

ISDN Release Reason	Description	SIP Response	Description
84	Call identity in use	503*	Service unavailable
85	No call suspended	503*	Service unavailable
86	Call having the requested call identity has been cleared	408*	Request timeout
87	User not member of CUG	503	Service unavailable
88	Incompatible destination	503	Service unavailable
90	Non-Existent CUG	503	Service unavailable
91	Invalid transit network selection	502*	Bad gateway
95	Invalid message	503	Service unavailable
96	Mandatory information element is missing	409*	Conflict
97	Message type non-existent or not implemented	480*	Temporarily not available
98	Message not compatible with call state or message type non-existent or not implemented	409*	Conflict
99	Information element non-existent or not implemented	480*	Not found
100	Invalid information elements contents	501*	Not implemented
101	Message not compatible with call state	503*	Service unavailable
102	Recovery of timer expiry	408	Request timeout
103	Parameter Non-Existent Or Not Implemented - Passed On	400	Bad Request
111	Protocol error	500	Server internal error
112	Uknown Error	400	Bad Request
127	Interworking unspecified	500	Server internal error

***** Messages and responses were created because the ‘ISUP to SIP Mapping’ draft doesn’t specify their cause code mapping.

23.8.3 Configuring ISDN-to-ISDN Release Cause Mapping

The Release Cause ISDN to ISDN table lets you configure up to 10 ISDN ITU-T Q.850 release cause code (call failure) to ISDN ITU-T Q.850 release cause code mapping rules. In other words, it lets you change the originally received ISDN cause code to a different ISDN cause code. For example, the PSTN may indicate disconnected calls (hang up) by sending cause code 127. However, you can change the cause code to 16, which is a more typical cause code for such call scenarios. When the device receives an ISDN cause code from the PSTN side, it searches the table for a matching ISDN cause code. If found, the device changes the cause code to the corresponding ISDN cause code. If the ISDN cause code is not configured in the table, the originally received ISDN cause code is used. If the new ISDN cause code also appears in the Release Cause Mapping ISDN to SIP table (see "Configuring

ISDN-to-SIP Release Cause Mapping" on page 460), the device maps it to the corresponding SIP response code, which it sends to the IP side.

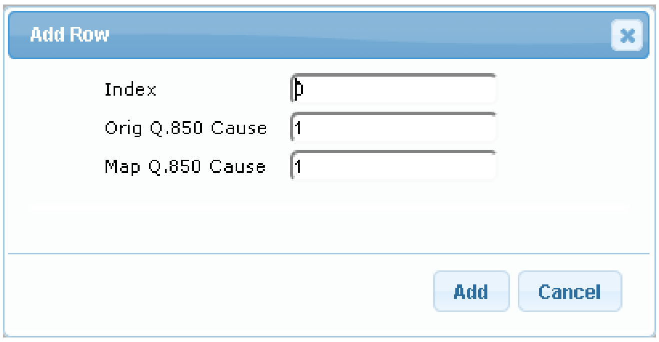
Note: If the originally received ISDN cause code is configured in both the Release Cause ISDN to ISDN table and the Release Cause Mapping ISDN to SIP table, the mapping rule with the originally received code in the Release Cause Mapping ISDN to SIP table is ignored; the device uses only the mapping rule in the Release Cause Mapping ISDN to SIP table that matches the new ISDN cause code. For example, if you configure a mapping rule in the Release Cause ISDN to ISDN table to change a received 127 code to 16, the device searches for a rule in the Release Cause Mapping ISDN to SIP table for an ISDN code of 16 (ignoring any entry with code 127).

The following procedure describes how to configure ISDN-to-ISDN release cause mapping through the Web interface. You can also configure it through ini file (CauseMapIsdn2Isdn) or CLI (configure voip > gw manipulations cause-map-isdn2isdn).

➢ **To configure a ISDN-to-ISDN release cause mapping rule:**

1. Open the Release Cause Mapping from ISDN to ISDN table (**Configuration** tab > **VoIP** menu > **Gateway** > **Manipulations** > **Release Cause ISDN > ISDN**).
2. Click **Add**; the following dialog box appears:

Figure 23-8: Release Cause ISDN to ISDN Table - Add Row Dialog Box



Index: 0
Orig Q.850 Cause: 1
Map Q.850 Cause: 1

3. Configure a mapping rule according to the parameters described in the table below.
4. Click **Add**.

Table 23-11: Release Cause Mapping ISDN to ISDN Table Parameter Descriptions

Parameter	Description
Index [CauseMapIsdn2Isdn_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Orig. Q.850 Causes `q850-causes` [CauseMapIsdn2Isdn_OrigIsdnReleaseCause]	Defines the originally received ISDN Q.850 cause code. For example, you can enter "127" (without apostrophes) to represent cause code 127 Interworking, Unspecified. The valid value (cause code) is 1 to 127.
Map Q.850 Causes `q850-causes` [CauseMapIsdn2Isdn_MapIsdnReleaseCause]	Defines the ISDN Q.850 cause code to which you want to change the originally received cause code. For example, you can enter "16" (without apostrophes) to represent cause code 16 Normal Call Clearing. The valid value (cause code) is 1 to 127.

23.8.4 Reason Header

The device supports the SIP Reason header according to RFC 3326. The Reason header describes the disconnection cause of a call:

- **Sending Reason header:** If a call is disconnected from the Tel side (ISDN), the Reason header contains the value of the received Q.850 cause in the appropriate message (BYE/CANCEL/final failure response) and sent to the IP side. If the call is disconnected because of a SIP reason, the Reason header is set to the appropriate SIP response.
- **Receiving Reason header:** If a call is disconnected from the IP side and the SIP message includes the Reason header, it is sent to the Tel side according to the following logic:
    - If the Reason header includes a Q.850 cause, it is sent as is.
    - If the Reason header includes a SIP response:
        - If the message is a final response, the response status code is translated to Q.850 format and passed to ISDN.
        - If the message isn’t a final response, it is translated to a Q.850 cause.
    - When the Reason header is received twice (i.e., SIP Reason and Q.850), the Q.850 takes precedence over the SIP reason and is sent to the Tel side.

23.9 Numbering Plans and Type of Number

The IP-to-Tel destination or source number manipulation tables allow you to classify numbers by their Numbering Plan Indication (NPI) and Type of Number (TON). The device supports all NPI/TON classifications used in the ETSI ISDN variant, as shown in the table below:

Table 23-12: NPI/TON Values for ETSI ISDN Variant

NPI	TON	Description
Unknown [0]	Unknown [0]	A valid classification, but one that has no information about the numbering plan.
E.164 Public [1]	Unknown [0]	A public number in E.164 format, but no information on what kind of E.164 number.
	International [1]	A public number in complete international E.164 format, e.g., 16135551234.
	National [2]	A public number in complete national E.164 format, e.g., 6135551234.
	Network Specific [3]	The type of number "network specific number" is used to indicate administration / service number specific to the serving network, e.g., used to access an operator.
	Subscriber [4]	A public number in complete E.164 format representing a local subscriber, e.g., 5551234.
	Abbreviated [6]	The support of this code is network dependent. The number provided in this information element presents a shorthand representation of the complete number in the specified numbering plan as supported by the network.
Private [9]	Unknown [0]	A private number, but with no further information about the numbering plan.

NPI	TON	Description
	Level 2 Regional [1]	
	Level 1 Regional [2]	A private number with a location, e.g., 3932200.
	PISN Specific [3]	
	Level 0 Regional (local) [4]	A private local extension number, e.g., 2200.

For NI-2 and DMS-100 ISDN variants, the valid combinations of TON and NPI for calling and called numbers include (Plan/Type):

- 0/0 - Unknown/Unknown
- 1/1 - International number in ISDN/Telephony numbering plan
- 1/2 - National number in ISDN/Telephony numbering plan
- 1/4 - Subscriber (local) number in ISDN/Telephony numbering plan
- 9/4 - Subscriber (local) number in Private numbering plan

24 Routing

This section describes the configuration of call routing rules.

24.1 Configuring General Routing Parameters

The Routing General Parameters page allows you to configure general routing parameters. For a description of these parameters, see "Configuration Parameters Reference" on page 813.

➢ **To configure general routing parameters:**

1. Open the Routing General Parameters page (**Configuration** tab > **VoIP** menu > **Gateway** > **Routing** > **General Parameters**).
2. Configure the parameters as required.
3. Click **Submit**.

24.2 Configuring Tel-to-IP Routing Rules

The Tel-to-IP Routing table lets you configure up to 180 Tel-to-IP routing rules. Tel-to-IP routing rules are used to route calls from the Tel side to an IP destination.

Configuration of Tel-to-IP routing rules includes two areas:

- **Rule:** Defines the characteristics of the incoming Tel call (e.g., Trunk Group on which the call is received).
- **Action:** Defines the action that is done if the incoming call matches the characteristics of the rule (i.e., routes the call to the specified IP destination).

The device searches the table from top to bottom for the first rule that matches the characteristics of the incoming call. If it finds a matching rule, it sends the call to the IP destination configured for that rule. If it doesn't find a matching rule, it rejects the call. .

You can configure the routing rule with one or more of the following incoming Tel characteristics:

- Source Trunk Group (from where the call is received)
- Source (calling) and destination (called) telephone number prefix and suffix

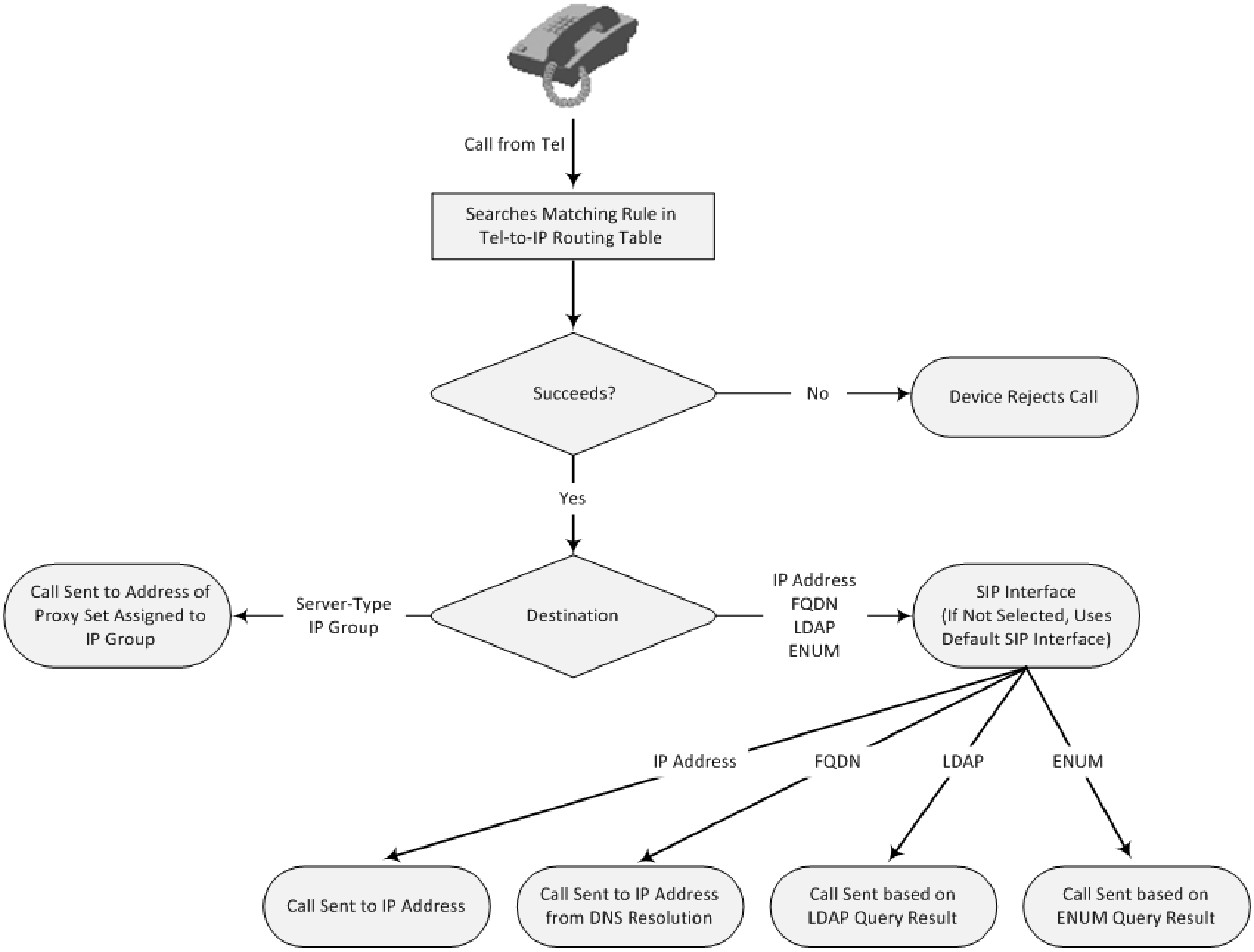
You can configure the IP destination to one of the following:

- IP address.
- FQDN.
- E.164 Telephone Number Mapping (ENUM service).
- Lightweight Directory Access Protocol (LDAP). For more information, see LDAP-based Management and SIP Services on page 232 and Active Directory-based Routing for Microsoft Lync on page 254.
- IP Group. When an IP Group is selected, the device sends the call to the IP address configured for the Proxy Set that is associated with the IP Group (configured in "Configuring IP Groups" on page 340). The SRD associated with the IP Group determines the:
  - SIP Interface (SIP port and control network interface) - important when using multiple SIP control VLANs
  - Media Realm (port and network interface for media / RTP voice)
  - SRD-related features on which the call is routed

If you configure the routing rule to send the call to any destination other than an IP Group (e.g., an IP address), you need to select a SIP Interface for the call. If no SIP Interface is selected, the device uses the SIP Interface associated with the default SRD (Index 0). If you have deleted this SRD or SIP Interface, for whatever reason, the device drops the call. The SIP Interface determines many attributes for the destination:

- Device's logical SIP port and network interface through which the call signaling is sent
- Device's logical RTP port and network interface through which the media is sent (Media Realm)
- Other features that can be configured for the SIP Interface
- SRD. As one of the attributes of a SIP Interface is an SRD and as you can configure multiple SIP Interfaces per SRD, the specific SIP Interface not only determines the above-mentioned attributes, but also the SRD for routing the call.

Figure 24-1: Locating SRD


Call from Tel
Searches Matching Rule in Tel-to-IP Routing Table
Succeeds?
No
Device Rejects Call
Yes
Destination
Server-Type IP Group
Call Sent to Address of Proxy Set Assigned to IP Group
IP Address
FQDN
LDAP
ENUM
SIP Interface (If Not Selected, Uses Default SIP Interface)
IP Address
FQDN
LDAP
ENUM
Call Sent to IP Address
Call Sent to IP Address from DNS Resolution
Call Sent based on LDAP Query Result
Call Sent based on ENUM Query Result

Note: When using a proxy server, it is unnecessary to configure routing rules in the Tel-to-IP Routing table, unless you require one of the following:

- Fallback (alternative) routing when communication with the proxy server fails.
- IP security, whereby the device routes only received calls whose source IP addresses are configured in the table. IP security is enabled using the SecureCallsFromIP parameter.
- Filter Calls to IP feature: the device checks the table before a call is routed to the proxy server. However, if the number is not allowed, i.e., the number is not specified in the table or a Call Restriction (see below) routing rule is configured, the call is rejected.
- Obtain different SIP URI host names (per called number).
- Assign IP Profiles to calls.
- For the table to take precedence over a proxy server for routing calls, you need to configure the PreferRouteTable parameter to 1. The device checks the 'Destination IP Address' field in the table for a match with the outgoing call; a proxy is used only if a match is not found.

In addition to normal Tel-to-IP routing, this table supports the following features:

- **Least Cost Routing (LCR):** If the LCR feature is enabled, the device searches the routing table for matching routing rules and then selects the one with the lowest call cost. The call cost of the routing rule is done by assigning it a Cost Group. For configuring Cost Groups, see "Least Cost Routing" on page 261. If two routing rules have identical costs, the rule appearing higher up in the table (i.e., first-matched rule) is used. If a selected route is unavailable, the device uses the next least-cost routing rule. However, even if a matched rule is not assigned a Cost Group, the device can select it as the preferred route over other matched routing rules with Cost Groups, according to the optional, default LCR settings configured by the Gateway Routing Policy (see "Configuring a Gateway Routing Policy Rule" on page 482).
- **Call Forking:** If the Tel-to-IP Call Forking feature is enabled, the device can send a Tel call to multiple IP destinations. An incoming Tel call with multiple matched routing rules (e.g., all with the same source prefix numbers) can be sent (forked) to multiple IP destinations if all these rules are configured with a Forking Group. The call is established with the first IP destination that answers the call.
- **Call Restriction:** Calls whose matching routing rule is configured with the destination IP address of 0.0.0.0 are rejected.
- **Always Use Routing Table:** Even if a proxy server is used, the SIP Request-URI host name in the outgoing INVITE message is obtained from this table. Using this feature, you can assign a different SIP URI host name for different called and/or calling numbers. This feature is enabled using the AlwaysUseRouteTable parameter.
- **IP Profiles:** IP Profiles can be assigned to destination addresses (also when a proxy is used).
- **Alternative Routing (when a proxy isn't used):** An alternative IP destination (alternative routing rule) can be configured for specific calls ("main" routing rule). When the "main" route fails (e.g., busy), the device can send the call to the alternative route. You must configure the alternative routing rules in table rows (indices) that are located anywhere **below** the "main" routing rule. For example, if you configure a "main" routing rule in Index 4, the alternative routing rule can be configured in Index 6. In addition, you must configure the alternative routing rules with identical matching characteristics (e.g., destination prefix number) as the "main" routing rule, but assigned with different destination IP addresses. Instead of an IP address, you can use an FQDN to resolve into two IP addresses. For more information on alternative routing, see "Alternative Routing for Tel-to-IP Calls" on page 485.
- **Advice of Charge (AOC):** AOC is a pre-billing feature that tasks the rating engine with calculating the cost of using a service (Tel-to-IP call) and relaying that information

to the customer. AOC, which is configured in the Charge Codes table, can be applied per Tel-to-IP routing rule.

Notes:

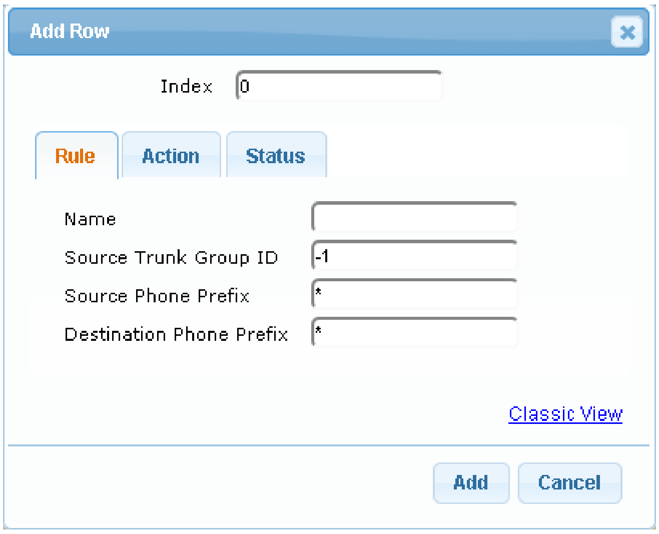
- Instead of using the table for Tel-to-IP routing, you can employ a routing server to handle the routing decisions. For more information, see Centralized Third-Party Routing Server or ARM on page 272.
- You can configure up to three alternative routing rules per "main" routing rule in the Tel-to-IP Routing table.
- By default, the device applies telephone number manipulation (if configured) only after processing the routing rule. You can change this and apply number manipulation before processing the routing rule (see the RouteModeTel2IP parameter).

The following procedure describes how to configure Tel-to-IP routing rules through the Web interface. You can also configure it through ini file (Prefix) or CLI (configure voip > gw routing tel2ip-routing).

➢ To configure Tel-to-IP routing rules:

1. Open the Tel-to-IP Routing table (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Routing** > **Tel to IP Routing**).
2. Click **Add**; the following dialog box appears:

Figure 24-2: Tel-to-IP Routing Table - Add Row Dialog Box



Index: 0

Name:
Source Trunk Group ID: -1
Source Phone Prefix: *
Destination Phone Prefix: *

3. Configure a routing rule according to the parameters described in the table below.
4. Click **Add**.

The following table shows configuration examples of Tel-to-IP routing rules, where:

- **Rules 1 and 2 (Least Cost Routing):** For both rules, the called (destination) phone number prefix is 10, the caller's (source) phone number prefix is 100, and the call is assigned IP Profile "ABC". However, Rule 1 is assigned a cheaper Cost Group than Rule 2, and therefore, the call is sent to the destination IP address (10.33.45.63) associated with Rule 1.
- **Rule 3 (IP Group destination):** For all callers (*), if the called phone number prefix is 20, the call is sent to IP Group "ITSP-ZA".
- **Rule 4 (domain name destination):** For called phone number prefixes 5, 7, 8, or 9, and the caller belongs to Trunk Group ID 4, the call is sent to the domain "itsp.com".

- **Rule 5 (block):** For all callers (*), if the called phone number prefix is 00, the call is rejected (IP address 0.0.0.0).
- **Rule 6, 7, and 8 (Forking Group):** For all callers (*), if the called phone number prefix is 100, the call is sent to Rule 7 and 9 (belonging to Forking Group "1"). If their destinations are unavailable and alternative routing is enabled, the call is sent to Rule 8 (Forking Group "2").

Table 24-1: Example of Tel-to-IP Routing Rules

Parameter	Rule 1	Rule 2	Rule 3	Rule 4	Rule 5	Rule 6	Rule 7	Rule 8
Matching Characteristics of Incoming Call								
Source Trunk Group ID				4		*	*	*
Source Phone Prefix	100	100	*	*	*	*	*	*
Destination Phone Prefix	10	10	20	[5,7-9]	00	100	100	100
Action								
Destination IP Group			ITSP-ZA					
Destination IP Address	10.33.45.63	10.33.45.50		itsp.com	0.0.0.0	10.33.45.68	10.33.45.67	domain.com
IP Profile	ABC	ABC						
Forking Group						1	2	1
Cost Group ID	Weekend-Low	Weekend_High						

Table 24-2: Tel-to-IP Routing table Parameter Descriptions

Parameter	Description
Index [PREFIX_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `route-name` [PREFIX_RouteName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. By default, no value is defined. **Note:** Each row must be configured with a unique name.
Rule Tab - Matching Call Characteristics	
Source Trunk Group ID `src-trunk-group-id` [PREFIX_SrcTrunkGroupID]	Defines the Trunk Group from where the call is received. To denote any Trunk Group, use the asterisk (*) symbol. By default, no Trunk Group is defined (-1).

Parameter	Description
Source Phone Prefix `src-phone-prefix` [PREFIX_SourcePrefix]	Defines the prefix and/or suffix of the calling (source) telephone number. You can use special notations for denoting the prefix. For example, [100-199](100,101,105) denotes a number that starts with 100 to 199 and ends with 100, 101 or 105. To denote any prefix, use the asterisk (*) symbol (default) or to denote calls without a calling number, use the $ sign. For a description of available notations, see "Dialing Plan Notation for Routing and Manipulation Tables" on page 809. The number can include up to 50 digits.
Destination Phone Prefix `dst-phone-prefix` [PREFIX_DestinationPrefix]	Defines the prefix and/or suffix of the called (destination) telephone number. The suffix is enclosed in parenthesis after the suffix value. You can use special notations for denoting the prefix. For example, [100-199](100,101,105) denotes a number that starts with 100 to 199 and ends with 100, 101 or 105. To denote any prefix, use the asterisk (*) symbol (default) or to denote calls without a called number, use the $ sign. For a description of available notations, see "Dialing Plan Notation for Routing and Manipulation Tables" on page 809. The number can include up to 50 digits. **Notes:** ▪ For LDAP-based routing, enter the LDAP query keyword as the prefix number to denote the IP domain: ✓ "PRIVATE" = Private number ✓ "OCS" = Lync / OCS client number ✓ "PBX" = PBX / IP PBX number ✓ "MOBILE" = Mobile number ✓ "LDAP_ERR" = LDAP query failure For more information, see Active Directory-based Routing for Microsoft Lync on page 254. ▪ If you want to configure re-routing of ISDN Tel-to-IP calls to fax destinations, enter the value string "FAX" (case-sensitive) as the destination phone prefix. For more information, see the FaxReroutingMode parameter.
Action Tab - IP Destination	
Destination IP Group `dst-ip-group-id` [PREFIX_DestIPGroupName]	Defines the IP Group to where you want to route the call. The SIP INVITE message is sent to the IP address configured for the Proxy Set that is associated with the IP Group. **Notes:** ▪ If you select an IP Group, you do not need to configure a destination IP address. However, if both parameters are configured in this table, the INVITE message is sent only to the IP Group. ▪ If the destination is a User-type IP Group, the device searches for a match of the Request-URI in the received INVITE to an AOR registration record in the device's database. The INVITE is then sent to the IP address of the registered contact. ▪ If the AlwaysUseRouteTable parameter is set to 1 (see "Configuring IP Groups" on page 340), the Request-URI host name in the INVITE message is set to the value configured for the 'Destination IP Address' parameter (in this table); otherwise, if no IP address is defined, it is set to the value of

Parameter	Description
	the 'SIP Group Name' parameter (configured in the IP Group table). ▪ The parameter is used as the 'Serving IP Group' in the Account table for acquiring authentication username/password for this call (see "Configuring Registration Accounts" on page 363). ▪ For defining Proxy Sets, see "Configuring Proxy Sets" on page 352.
Destination SIP Interface `dest-sip-interface-name` [PREFIX_DestSIPInterfaceName]	Assigns a SIP Interface to the routing rule. The call is sent to its' destination through this SIP interface. For configuring SIP Interfaces, see "Configuring SIP Interfaces" on page 333. **Note:** If a SIP Interface is not assigned, the device uses the SIP Interface associated with the default SRD (Index 0). If, for whatever reason, you have deleted the default SRD and there are no SRDs, the call is rejected.
Destination IP Address `dst-ip-address` [PREFIX_DestAddress]	Defines the IP address (in dotted-decimal notation or FQDN) to where the call is sent. If an FQDN is used (e.g., domain.com), DNS resolution is done according to the DNSQueryType parameter. The IP address can include the following wildcards: ▪ "x": represents single digits. For example, 10.8.8.xx denotes all addresses between 10.8.8.10 and 10.8.8.99. ▪ "*": represents any number between 0 and 255. For example, 10.8.8.* denotes all addresses between 10.8.8.0 and 10.8.8.255. For ENUM-based routing, enter the string "ENUM". The device sends an ENUM query containing the destination phone number to an external DNS server, configured in the Interface table. The ENUM reply includes a SIP URI which is used as the Request-URI in the subsequent outgoing INVITE and for routing (if a proxy is not used). To configure the type of ENUM service (e.g., e164.arpa), see the EnumService parameter. For LDAP-based routing, enter the string value, "LDAP" for denoting the IP address of the LDAP server. For more information, see Active Directory-based Routing for Microsoft Lync on page 254. **Notes:** ▪ The parameter is ignored if you have configured a destination IP Group in the 'Destination IP Group' field (in this table). ▪ To reject calls, enter the IP address 0.0.0.0. For example, if you want to prohibit international calls, then in the 'Destination Phone Prefix' field, enter 00 and in the 'Destination IP Address' field, enter 0.0.0.0. ▪ For routing calls between phones connected to the device (i.e., local routing), enter the device's IP address. If the device's IP address is unknown (e.g., when DHCP is used), enter IP address 127.0.0.1. ▪ When using domain names, enter the DNS server's IP address or alternatively, configure these names in the Internal DNS table (see "Configuring the Internal DNS Table" on page 147).

Parameter	Description
Destination Port `dst-port` [PREFIX_DestPort]	Defines the destination port to where you want to route the call.
Transport Type `transport-type` [PREFIX_TransportType]	Defines the transport layer type used for routing the call: ▪ **[-1]** = (Default) Not defined - transport type is according to the settings of the global parameter, SIPTransportType. ▪ **[0]** UDP ▪ **[1]** TCP ▪ **[2]** TLS
IP Profile `ip-profile-id` [PREFIX_ProfileName]	Assigns an IP Profile to the routing rule in the outgoing direction. The IP Profile allows you to assign various configuration attributes (e.g., voice coder) per routing rule. To configure IP Profiles, see "Configuring IP Profiles" on page 387.
Call Setup Rules Set ID `call-setup-rules-set-id` [PREFIX_CallSetupRulesSetId]	Assigns a Call Setup Rule Set ID to the routing rule. The device performs the Call Setup rules of this Set ID if the incoming call matches the characteristics of the routing rule. The device routes the call to the destination according to the routing rule's configured action only after it has performed the Call Setup rules. By default, no value is defined. For configuring Call Setup rules, see "Configuring Call Setup Rules" on page 283.
Forking Group `forking-group` [PREFIX_ForkingGroup]	Defines a Forking Group number for the routing rule. This enables forking of incoming Tel calls to multiple IP destinations. The device sends simultaneous INVITE messages and handles multiple SIP dialogs until one of the calls is answered. When a call is answered, the other calls are dropped. Each Forking Group can contain up to 10 members. In other words, up to 10 routing rules can be configured with the same Forking Group number. By default, no value is defined. If all matched routing rules belong to the same Forking Group number, the device sends an INVITE to all the destinations belonging to this group. If matched routing rules belong to different Forking Groups, the device sends the call to the Forking Group of the first matched routing rule. If the call cannot be established with any of the destinations associated with the Forking Group and alternative routing is enabled, the device forks the call to the Forking Group of the next matched routing rules, as long as the Forking Group is defined with a **higher** number than the previous Forking Group. For example: ▪ Table index entries 1 and 2 are defined with Forking Group "1", and index entries 3 and 4 with Forking Group "2": The device first sends the call according to index entries 1 and 2, and if unavailable and alternative routing is enabled, sends the call according to index entries 3 and 4. ▪ Table index entry 1 is defined with Forking Group "2", and index entries 2, 3, and 4 with Forking Group "1": The device sends the call according to index entry 1 only and ignores the other index entries even if the destination is unavailable and alternative routing is enabled. This is because the

Parameter	Description
	subsequent index entries are defined with a Forking Group number that is lower than that of index entry 1. ▪ Table index entry 1 is defined with Forking Group "1", index entry 2 with Forking Group "2", and index entries 3 and 4 with Forking Group "1": The device first sends the call according to index entries 1, 3, and 4 (all belonging to Forking Group "1"), and if the destination is unavailable and alternative routing is enabled, the device sends the call according to index entry 2. ▪ Table index entry 1 is defined with Forking Group "1", index entry 2 with Forking Group "3", index entry 3 with Forking Group "2", and index entry 4 with Forking Group "1": The device first sends the call according to index entries 1 and 4 (all belonging to Forking Group "1"), and if the destination is unavailable and alternative routing is enabled, the device sends the call according to index entry 2 (Forking Group "3"). Even if index entry 2 is unavailable and alternative routing is enabled, the device ignores index entry 3 because it belongs to a Forking Group that is lower than index entry 2. **Notes:** ▪ To enable Tel-to-IP call forking, set the 'Tel2IP Call Forking Mode' (*Tel2IPCallForkingMode*) parameter to **Enable**. ▪ You can configure the device to immediately send the INVITE message to the first member of the Forking Group (as in normal operation) and then only after a user-defined interval, send the INVITE messages simultaneously to the other members. If the device receives a SIP 4xx or 5xx in response to the first INVITE, it immediately sends INVITEs to all the other members, regardless of the interval. To configure this feature, see the ForkingDelayTimeForInvite ini file parameter. ▪ You can implement Forking Groups when the destination is an LDAP server or a domain name using DNS. In such scenarios, the INVITE is sent to all the queried LDAP or resolved IP addresses, respectively. You can also use LDAP routing rules with standard routing rules for Forking Groups. ▪ When the UseDifferentRTPportAfterHold parameter is enabled, every forked call is sent with a different RTP port. Thus, ensure that the device has sufficient available RTP ports for these forked calls.
Cost Group `cost-group-id` [PREFIX_CostGroup]	Assigns a Cost Group to the routing rule for determining the cost of the call (i.e., Least Cost Routing or LCR). By default, no value is defined (**None**). To configure Cost Groups, see "Configuring Cost Groups" on page 263. **Note:** To implement LCR and its Cost Groups, you must enable LCR ▪ To implement LCR and its Cost Groups, the Gateway Routing Policy must be enabled for LCR (see "Configuring a Gateway Routing Policy Rule" on page 482). If LCR is disabled, the device ignores the parameter. ▪ The Routing Policy also determines whether matched routing rules that are **not** assigned Cost Groups are considered as a higher or lower cost route compared to matching routing

Parameter	Description
	rules that are assigned Cost Groups. For example, if the 'Default Call Cost' parameter in the Routing Policy is configured to **Lowest Cost**, even if the device locates matching routing rules that are assigned Cost Groups, the first-matched routing rule without an assigned Cost Group is considered as the lowest cost route and thus, chosen as the preferred route.
Charge Code `charge-code` [PREFIX_MeteringCode]	Assigns a Charge Code to the routing rule for generating metering pulses (Advice of Charge). By default, no value is defined (**None**). To configure Charge Codes, see "Configuring Charge Codes" on page 520.
Status Tab	
Connectivity Status	(Read-only field) Displays the connectivity status of the routing rule's destination. The destination can be an IP address or an IP Group, as configured in the 'Destination IP Address' and 'Destination IP Group' fields respectively. For IP Groups, the status indicates the connectivity with the SIP proxy server's address configured for the Proxy Set that is associated with the IP Group. For the status to be displayed, the Proxy Keep-Alive feature, which monitors the connectivity with proxy servers per Proxy Set, must be enabled for the Proxy Set (see "Configuring Proxy Sets" on page 352). If a Proxy Set is configured with multiple proxies for redundancy, the status may change according to the proxy server with which the device attempts to verify connectivity. For example, if there is no response from the first configured proxy address, the status displays "No Connectivity". However, if there is a response from the next proxy server in the list, the status changes to "OK". If there is connectivity with the destination, the field displays "OK" and the device uses the routing rule if required. The routing rule is not used if any of the following is displayed: ▪ "n/a" = IP Group is unavailable. ▪ "No Connectivity" = No connection with the destination (no response to the SIP OPTIONS). ▪ "QoS Low" = Poor Quality of Service (QoS) of the destination. ▪ "DNS Error" = No DNS resolution. This status is applicable only when a domain name is used (instead of an IP address). ▪ "Not Available" = Destination is unreachable due to networking issues.

24.3 Configuring IP-to-Trunk Group Routing Rules

The IP to Trunk Group Routing table lets you configure up to 120 IP-to-Trunk Group routing rules. IP-to-Trunk Group routing rules are used to route incoming IP calls to Trunk Groups. The specific channel pertaining to the Trunk Group to which the call is routed is determined according to the Trunk Group's channel selection mode. The channel selection mode can be configured per Trunk Group (see "Configuring Trunk Group Settings" on page 435) or for all Trunk Groups using the global parameter ChannelSelectMode.

Configuration of IP-to-Trunk Group routing rules includes two areas:

- **Rule:** Defines the characteristics of the incoming IP call (e.g., source IP address from which the call is received).
- **Action:** Defines the action that is done if the incoming call matches the characteristics of the rule (i.e., routes the call to the specified Tel/Trunk Group destination).

The device searches the table from top to bottom for the first rule that matches the characteristics of the incoming call. If it finds a matching rule, it sends the call to the Tel destination configured for that rule. If it doesn't find a matching rule, it rejects the call. :

If an IP-to-Tel call cannot be routed to the Trunk Group, the device can route it to an alternative destination:

- **Routing to an Alternative Trunk Group:** If the device sends the IP call to the Tel destination and a subsequent call release reason (cause) code (e.g., 17 for User Busy) is received from the Tel side, and you have configured this release reason code in the Reasons for IP-to-Tel Alternative Routing table, the device re-routes the call to an alternative Trunk Group if an alternative routing rule has been configured in the table. The alternative routing rules must be configured in table rows (indices) that are located anywhere below the "main" routing rule. For example, if you configure a "main" routing rule in Index 4, the alternative routing rule can be configured in Index 6. In addition, you must configure the alternative routing rules with identical matching characteristics (e.g., destination prefix number) to the "main" routing rule, but assigned with different destinations (i.e., Trunk Groups). For more information on IP-to-Tel alternative routing and for configuring call release reasons for alternative routing, see "Alternative Routing to Trunk upon Q.931 Call Release Cause Code" on page 492.
- **Routing to an IP Destination (i.e., Call Redirection):** The device can re-route the IP-to-Tel call to an alternative IP destination, using SIP 3xx responses. For more information, see "Alternative Routing to IP Destinations upon Busy Trunk" on page 494.
- **Routing to an Alternative Physical** Trunk **within Same Trunk Group:** The device can re-route an IP-to-Tel call to a different physical trunk if the destined trunk within the same Trunk Group is out of service (e.g., physically disconnected). When the physical trunk is disconnected, the device sends the SNMP trap, GWAPP_TRAP_BUSYOUT_LINK notifying of the out-of-service state for the specific trunk number. When the physical trunk is physically reconnected, this trap is sent notifying of the back-to-service state.

Notes:

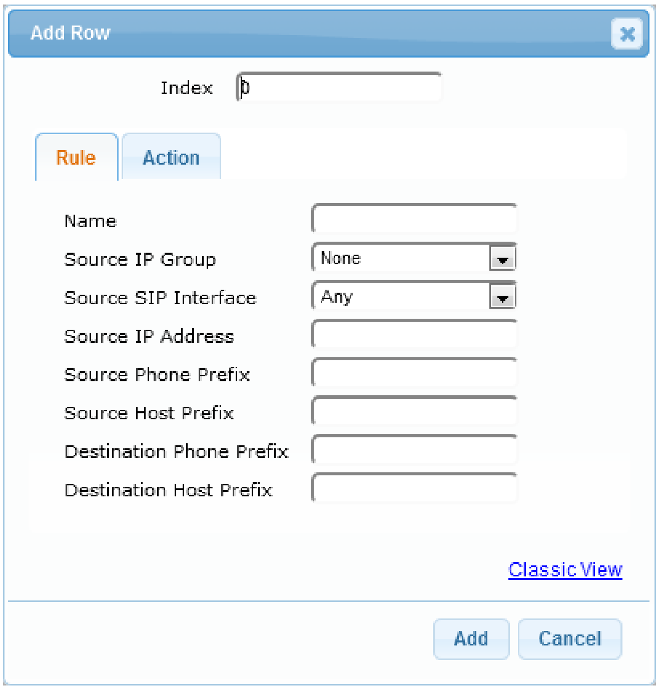
- Instead of using the table for IP-to-Tel routing, you can employ a routing server to handle the routing decisions. For more information, see Centralized Third-Party Routing Server or ARM on page 272.
- You can configure up to three alternative routing rules per "main" routing rule in the table.
- If your deployment includes calls of many different called (source) and/or calling (destination) numbers that need to be routed to the same destination, you can employ user-defined prefix tags to represent these numbers. Thus, instead of configuring many routing rules, you need to configure only one routing rule using the prefix tag as the source and destination number matching characteristics, and a destination for the calls. For more information on prefix tags, see "Dial Plan Prefix Tags for IP-to-Tel Routing" on page 655.
- By default, the device applies destination telephone number manipulation (if configured) only after processing the routing rule. You can change this and apply number manipulation before processing the routing rule (see the RouteModeIP2Tel parameter). For configuring number manipulation, see "Configuring Source/Destination Number Manipulation" on page 441.

The following procedure describes how to configure Inbound IP Routing rules through the Web interface. You can also configure it through ini file (PSTNPrefix) or CLI (configure voip > gw routing ip2tel-routing).

➢ To configure IP-to-Tel routing rules:

1. Open the IP to Trunk Group Routing table (**Configuration** tab > **VoIP** menu > **Gateway** > **Routing** > **IP to Trunk Group Routing**).
2. Click **Add**; the following dialog box appears:

Figure 24-3: IP to Trunk Group Table - Add Row Dialog Box



Add Row

Index: 0

Rule | Action

Name:
Source IP Group: None
Source SIP Interface: Any
Source IP Address:
Source Phone Prefix:
Source Host Prefix:
Destination Phone Prefix:
Destination Host Prefix:

3. Configure a routing rule according to the parameters described in the table below.
4. Click **Add**.

The following table shows configuration examples of Tel-to-IP routing rules, where:

- **Rule 1:** If the incoming IP call destination phone prefix is between 10 and 19, the call is assigned settings configured for IP Profile "ITSP-A" and routed to Trunk Group ID 3.
- **Rule 2:** If the incoming IP call destination phone prefix is between 501 and 502 and source phone prefix is 101, the call is assigned settings configured for IP Profile "ITSP-B" and routed to Trunk Group ID 2.
- **Rule 3:** If the incoming IP call has a From URI host prefix as abcd.com, the call is routed to Trunk Group ID 4.

Table 24-3: Example of IP-to-Trunk Group Routing Rules

Parameter	Rule 1	Rule 2	Rule 3
Source Host Prefix			abcd.domain
Destination Phone Prefix	1x	[501-502]	
Source Phone Prefix		101	
Source IP Address			
Trunk Group ID	3	2	4
IP Profile	ITSP-A	ITSP-B	

Table 24-4: IP to Trunk Group Table Parameter Description

Parameter	Description
Index [PstnPrefix_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `route-name` [PstnPrefix_RouteName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. By default, no value is defined. **Note:** Each row must be configured with a unique name.
Rule (Matching Characteristics)	
Source SIP Interface `src-sip-interface-name` [PstnPrefix_SrcSIPInterfaceName]	Defines the SIP Interface on which the incoming IP call is received. The default is Any (i.e., any SIP Interface). For configuring SIP Interfaces, see Configuring SIP Interfaces on page 333. **Note:** If the incoming INVITE is received on the specified SIP Interface and the SIP Interface associated with the specified IP Group in the 'Source IP Group' parameter (in this table) is different, the incoming SIP call is rejected. If the 'Source IP Group' parameter is not defined, the SIP Interface associated with the default SRD (Index 0) is used. If there is no valid source IP Group, the call is rejected.
Source IP Address `src-ip-address` [PstnPrefix_SourceAddress]	Defines the source IP address of the incoming IP call. The IP address must be configured in dotted-decimal notation (e.g., 10.8.8.5); not as an FQDN. By default, no value is defined. **Notes:** ▪ The source IP address is obtained from the Contact header in the INVITE message. ▪ You can configure from where the source IP address is obtained, using the SourceIPAddressInput parameter. ▪ The source IP address can include the following wildcards: ✓ "x": denotes single digits. For example, 10.8.8.xx represents all the addresses between 10.8.8.10 and 10.8.8.99. ✓ "*": denotes any number between 0 and 255. For example, 10.8.8.* represents all addresses between 10.8.8.0 and 10.8.8.255.
Source Phone Prefix `src-phone-prefix` [PstnPrefix_SourcePrefix]	Defines the prefix or suffix of the calling (source) telephone number. The prefix can include up to 49 digits. You can use special notations for denoting the prefix. For example, [100-199](100,101,105) denotes a number that starts with 100 to 199 and ends with 100, 101 or 105. To denote any prefix, use the asterisk (*) symbol. To denote calls without a calling number, use the $ sign. For a description of available notations, see "Dialing Plan Notation for Routing and Manipulation Tables" on page 809. By default, no value is defined. **Note:** If the P-Asserted-Identity header is present in the incoming INVITE message, the value of the parameter is

Parameter	Description
	compared to the P-Asserted-Identity URI host name (and not the From header).
Source Host Prefix `src-host-prefix` [PstnPrefix_SrcHostPrefix]	Defines the prefix of the URI host name in the From header of the incoming INVITE message. By default, no value is defined. To denote any prefix, use the asterisk (*) wildcard. **Note:** If the P-Asserted-Identity header is present in the incoming INVITE message, the value of the parameter is compared to the P-Asserted-Identity URI host name (and not the From header).
Destination Phone Prefix `dst-host-prefix` [PstnPrefix_DestHostPrefix]	Defines the prefix or suffix of the called (destined) telephone number. You can use special notations for denoting the prefix. For example, [100-199](100,101,105) denotes a number that starts with 100 to 199 and ends with 100, 101 or 105. To denote any prefix, use the asterisk (*) symbol or to denote calls without a called number, use the $ sign. For a description of available notations, see "Dialing Plan Notation for Routing and Manipulation Tables" on page 809. By default, no value is defined. The prefix can include up to 49 digits.
Destination Host Prefix `dst-phone-prefix` [PstnPrefix_DestPrefix]	Defines the Request-URI host name prefix of the incoming INVITE message. By default, no value is defined. To denote any prefix, use the asterisk (*) wildcard.
Action Tab (Tel Destination)	
Destination Type `dst-type` [PstnPrefix_DestType]	Defines the type of Tel destination: ▪ [0] Trunk Group (default) ▪ [1] Trunk
Trunk Group ID `trunk-group-id` [PstnPrefix_TrunkGroupId]	Defines the Trunk Group ID to where the incoming SIP call is sent.
Trunk ID `trunk-id` [PstnPrefix_TrunkId]	Defines the Trunk to where the incoming SIP call is sent. **Notes:** ▪ If both 'Trunk Group ID' and 'Trunk ID' parameters are configured in the table, the routing is done according to the 'Trunk Group ID' parameter. ▪ To configure the method for selecting the trunk's channel to which the IP call is sent, see the global parameter, ChannelSelectMode.
Source IP Group `src-ip-group-id` [PstnPrefix_SrcIPGroupName]	Defines the IP Group associated with the incoming IP call. This is the IP Group from where the SIP message (INVITE) is received. By default, no value is defined. The IP Group can later be used as the 'Serving IP Group' in the Account table for obtaining authentication username/password for this call. For configuring registration accounts, see "Configuring Account Table" on page 363.

Parameter	Description
IP Profile `ip-profile-id` [PstnPrefix_ProfileName]	Assigns an IP Profile (configured in "Configuring IP Profiles" on page 387) to the call.
Call Setup Rules Set ID `call-setup-rules-set-id` [PstnPrefix_CallSetupRulesSetId]	Assigns a Call Setup Rule Set ID to the routing rule. The device performs the Call Setup rules of this Set ID if the incoming call matches the characteristics of the routing rule. The device routes the call to the destination according to the routing rule's configured action, only after it has performed the Call Setup rules. For configuring Call Setup rules, see "Configuring Call Setup Rules" on page 283.

24.4 Configuring a Gateway Routing Policy Rule

The Gateway Routing Policy table lets you edit the default Gateway Routing Policy rule. The Routing Policy is used for Gateway call routing and defines the following:

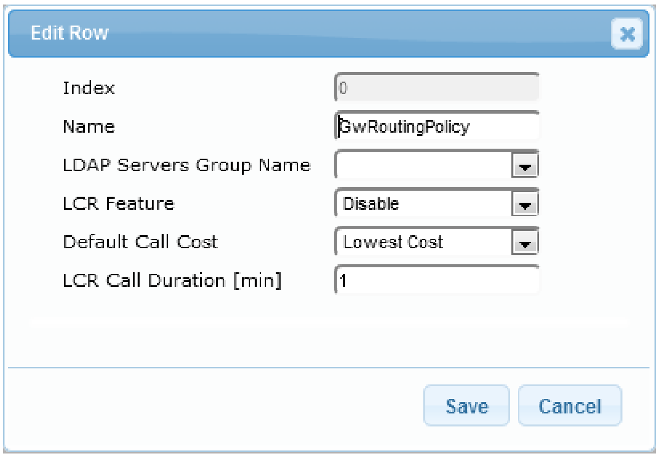
- LDAP server (LDAP Server Group) for LDAP-based call routing (LDAP and/or Call Setup Rules queries). LDAP-based routing is applicable to Tel-to-IP routing ("Configuring Tel-to-IP Routing Rules" on page 467) and IP-to-Tel routing ("Configuring IP-to-<trunkgroupM5SBC> Routing Rules" on page 476).
- Enables Least Cost Routing (LCR), and defines default call cost (highest or lowest) and average call duration for Tel-to-IP routing rules that are not assigned LCR Cost Groups. The default call cost determines whether matched routing rules that are not assigned a Cost Group are considered as a higher or lower cost route compared to other matching routing rules that are assigned Cost Groups. If you disable LCR, the device ignores the Cost Groups assigned to Tel-to-IP routing rules in the Tel-to-IP Routing table. LCR is applicable only to Tel-to-IP routing.

The following procedure describes how to configure Gateway Routing Policy rules through the Web interface. You can also configure it through ini file (GwRoutingPolicy) or CLI (configure voip > gw routing gw-routing-policy).

➢ To edit the Gateway Routing Policy rule:

1. Open the Gateway Routing Policy table (**Configuration** tab > **VoIP** menu > **Gateway** > **Routing** > **Gateway Routing Policy**).
2. Click **Add**; the following dialog box appears:

Figure 24-4: Gateway Routing Policy Table - Edit Row Dialog Box



Index: 0
Name: GwRoutingPolicy
LDAP Servers Group Name:
LCR Feature: Disable
Default Call Cost: Lowest Cost
LCR Call Duration [min]: 1

3. Configure the Gateway Routing Policy rule according to the parameters described in the table below.
4. Click **Add**.

Table 24-5: Gateway Routing Policy Table Parameter Descriptions

Parameter	Description
Index [GwRoutingPolicy_Index]	(Read-only) Displays the index number of the table row.
Name `name` [GWRoutingPolicy_Name]	Defines an arbitrary name to easily identify the row. The default value is "GwRoutingPolicy".
LDAP Servers Group Name `ldap-srv-group-name` [GWRoutingPolicy_LdapServersGroupName]	Assigns an LDAP Server Group to the Routing Policy. IP-to-Tel and Tel-to-IP routing rules that require LDAP-based routing (and/or Call Setup Rules) use the LDAP server(s) assigned to the LDAP Server Group. The valid value is a string of up to 40 characters. By default, no value is defined (**None**). For more information on LDAP Server Groups, see "Configuring LDAP Server Groups" on page 234.
LCR Feature `lcr-enable` [GWRoutingPolicy_LCREnable]	Enables the Least Cost Routing (LCR) feature for the Routing Policy. ▪ [0] Disable (default) ▪ [1] Enable For more information on LCR, see "Least Cost Routing" on page 261. **Note:** LCR is applicable only to Tel-to-IP routing.

Parameter	Description
Default Call Cost `lcr-default-cost` [GWRoutingPolicy_LCRDefaultCost]	Defines whether routing rules in the Tel-to-IP Routing table that are not assigned a Cost Group are considered a higher cost or lower cost route compared to other matched routing rules that are assigned Cost Groups. ▪ [0] Lowest Cost = (Default) The device considers a matched routing rule that is not assigned a Cost Group as the lowest cost route. Therefore, it uses the routing rule. ▪ [1] Highest Cost = The device considers a matched routing rule that is not assigned a Cost Group as the highest cost route. Therefore, it is only used if the other matched routing rules that are assigned Cost Groups are unavailable.
LCR Call Duration `lcr-call-length` [GWRoutingPolicy_LCRAverageCallLength]	Defines the average call duration (in minutes) and is used to calculate the variable portion of the call cost. This is useful, for example, when the average call duration spans over multiple time bands. The LCR is calculated as follows: cost = call connect cost + (minute cost * average call duration) The valid value is 0-65533. The default is 1. For example, assume the following Cost Groups: ▪ "Weekend A": call connection cost is 1 and charge per minute is 6. Therefore, a call of 1 minute cost 7 units. ▪ "Weekend B": call connection cost is 6 and charge per minute is 1. Therefore, a call of 1 minute cost 7 units. Therefore, for calls under one minute, "Weekend A" carries the lower cost. However, if the average call duration is more than one minute, "Weekend B" carries the lower cost.

24.5 IP Destinations Connectivity Feature

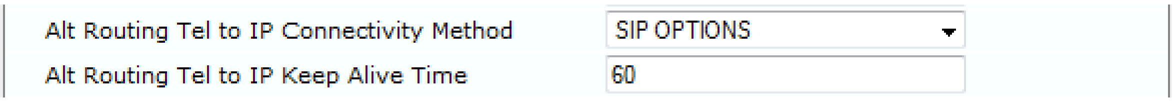
The device can be configured to check the integrity of the connectivity to IP destinations of Tel-to-IP routing rules in the Tel-to-IP Routing table. The IP Connectivity feature can be used for the Alternative Routing feature, whereby the device attempts to re-route calls from unavailable Tel-to-IP routing destinations to available ones (see "Alternative Routing Based on IP Connectivity" on page 486).

The device supports the following methods for checking the connectivity of IP destinations:

- **Network Connectivity:** The device checks the network connectivity of the IP destination configured by the 'Alt Routing Tel to IP Connectivity Method' parameter:
  - **SIP OPTIONS:** The device sends "keep-alive" SIP OPTIONS messages to the IP destination. If the device receives a SIP 200 OK in response, it considers the destination as available. If the destination does not respond to the OPTIONS message, then it is considered unavailable. You can configure the time interval for sending these OPTIONS messages, using the 'Alt Routing Tel to IP Keep Alive Time' parameter.

These parameters are configured in the Routing General Parameters page (**Configuration** tab > **VoIP** menu > **Gateway** > **Routing** > **General Parameters**), as shown below:

Figure 24-5: IP Connectivity Method in Routing General Parameters Page



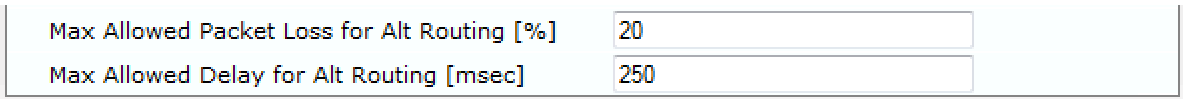
Alt Routing Tel to IP Connectivity Method: SIP OPTIONS

Alt Routing Tel to IP Keep Alive Time: 60

- **Quality of Service (QoS):** You can enable the device to check the QoS of IP destinations. The device measures the QoS according to RTCP statistics of previously established calls with the IP destination. The RTCP includes packet delay (in milliseconds) and packet loss (in percentage). If these measured statistics exceed a user-defined threshold, the destination is considered unavailable. Note that if call statistics is not received within two minutes, the QoS data is reset. These thresholds are configured using the following parameters:
  - 'Max Allowed Packet Loss for Alt Routing' (IPConnQoSMaxAllowedPL): defines the threshold value for packet loss after which the IP destination is considered unavailable.
  - 'Max Allowed Delay for Alt Routing' (IPConnQoSMaxAllowedDelay): defines the threshold value for packet delay after which the IP destination is considered unavailable

  These parameters are configured in the Routing General Parameters page, as shown below:

Figure 24-6: IP QoS Thresholds in Routing General Parameters Page



Max Allowed Packet Loss for Alt Routing [%]: 20

Max Allowed Delay for Alt Routing [msec]: 250

- **DNS Resolution:** When a host name (FQDN) is used (instead of an IP address) for the IP destination, it is resolved into an IP address by a DNS server. The device checks network connectivity and QoS of the resolved IP address. If the DNS host name is unresolved, the device considers the connectivity of the IP destination as unavailable.

You can view the connectivity status of IP destinations in the following Web interface pages:

- **Tel-to-IP Routing table:** The connectivity status of the IP destination per routing rule is displayed in the 'Status' column. For more information, see "Configuring Tel-to-IP Routing Rules" on page 467.
- **IP Connectivity:** This page displays a more informative connectivity status of the IP destinations used in Tel-to-IP routing rules in the Tel-to-IP Routing table. For viewing this page, see "Viewing IP Connectivity" on page 731.

24.6 Alternative Routing for Tel-to-IP Calls

The device supports various alternative Tel-to-IP call routing methods, as described in this section.

24.6.1 Alternative Routing Based on IP Connectivity

You can configure the device to route Tel-to-IP calls to an alternative IP destination when the connectivity state of an IP destination is unavailable. The alternative routing rules are configured in the Tel-to-IP Routing table. These rules must be configured anywhere below the "main" routing rule and with identical matching characteristics (e.g., destination prefix number) to the "main" routing rule. The device uses the first alternative route that is available. For more information on configuring alternative Tel-to-IP routing rules in the Tel-to-IP Routing table, see "Configuring Tel-to-IP Routing Rules" on page 467.

Note:

- Alternative routing based on IP connectivity is applicable only when a proxy server is **not** used.
- You can also enable the Busy Out feature, whereby the device can take specified actions if all IP destinations of matching routing rules in the Tel-to-IP Routing table do not respond to connectivity checks. For more information, see the EnableBusyOut parameter.
- If the AltRoutingTel2IPEnable parameter is enabled, the Busy Out feature does not function with the Proxy Set keep-alive mechanism (see Alternative Routing Based on SIP Responses on page 487). To use the Busy Out feature with the Proxy Set keep-alive mechanism (for IP Groups), disable the AltRoutingTel2IPEnable parameter.

The device searches for an alternative routing rule (IP destination) when any of the following connectivity states are detected with the IP destination of the "main" routing rule:

- No response received from SIP OPTIONS messages. This depends on the chosen method for checking IP connectivity.
- Poor QoS according to the configured thresholds for packet loss and delay.
- No response from a DNS-resolved IP address, where the domain name (FQDN) is configured for the IP destination. If the device sends the INVITE message to the first IP address and receives no response, the device makes a user-defined number of attempts (configured by the HotSwapRtx parameter) to send it again (re-transmit). If there is still no response after all the attempts, it sends it to the next DNS-resolved IP address, and so on. For example, if you configure the parameter to "3" and the device receives no response from the first IP address, it attempts up to three times to send the INVITE to the first IP address and if unsuccessful, it attempts to send the call to the next DNS-resolved IP address, and so on.
- No response for in-dialog request from a DNS-resolved IP address, where the domain name is received in the Contact header of an incoming setup or target refresh SIP message (e.g., 200 OK). If no response is received from the first IP address, the device tries to send it again for up to a user-defined number of attempts (configured by the HotSwapRtx parameter). If there is still no response, it attempts to send the SIP request to the next DNS-resolved IP address, and so on.

The connectivity status of the IP destination is displayed in the 'Status' column of the Tel-to-IP Routing table per routing rule. If it displays a status other than "ok", the device considers the IP destination as unavailable and attempts to re-route the call to an alternative destination. For more information on the IP connectivity methods and on viewing IP connectivity status, see "IP Destinations Connectivity Feature" on page 484.

The table below shows an example of alternative routing where the device uses an available alternative routing rule in the Tel-to-IP Routing table to re-route the initial Tel-to-IP call.

Table 24-6: Alternative Routing based on IP Connectivity Example

	Destination Phone Prefix	IP Destination	IP Connectivity Status	Rule Used?
Main Route	40	10.33.45.68	"No Connectivity"	No
Alternative Route #1	40	10.33.45.70	"QoS Low"	No
Alternative Route #2	40	10.33.45.72	"ok"	Yes

The steps for configuring alternative Tel-to-IP routing based on IP connectivity are summarized below.

➢ **To configure alternative Tel-to-IP routing based on IP connectivity:**

1. In the Tel-to-IP Routing table, add alternative Tel-to-IP routing rules for specific calls.
2. In the Routing General Parameters page (**Configuration** tab > **VoIP** menu > **Gateway** > **Routing** > **General Parameters**), do the following:
   a. Enable alternative routing based on IP connectivity, by setting the 'Enable Alt Routing Tel to IP AltRouting' (AltRoutingTel2IPEnable) parameter to **Enable**.
   b. Configure the IP connectivity reason for triggering alternative routing, by setting the 'Alt Routing Tel to IP Mode' parameter (AltRoutingTel2IPMode) to one of the following:
      - SIP OPTIONS failure
      - Poor QoS
      - SIP OPTIONS failure, poor QoS, or unresolved DNS

24.6.2 Alternative Routing Based on SIP Responses

The device can perform alternative routing based on the received SIP response code (i.e., 4xx, 5xx, 6xx, or 8xx). If you have configured this response code in the Reasons for Tel-to-IP Alternative Routing table, the device attempts to re-route the call to an alternative destination, if configured. You can configure up to 10 SIP response codes in the Reasons for Tel-to-IP Alternative Routing table.

Typically, the device performs alternative routing when there is no response at all to an INVITE message. This is done after a user-defined number of INVITE re-transmissions, configured by the SIPMaxRtx parameter. In such a scenario, the device issues itself the SIP response code 408 (Request Timeout). You can also configure the device to perform alternative routing for the following proprietary response codes that are issued by the device itself:

- **805 IP Profile Call Limit:** The device generates this response code when Call Admission Control (CAC) limits are exceeded for an IP Group. The CAC rules are configured in the IP Profile table (see "Configuring IP Profiles" on page 387). When this occurs, the device sends a SIP 480 (Temporarily Unavailable) response to the SIP entity.
- **806 Media Limits Exceeded:** The device generates this response code when the call is terminated due to crossed thresholds of QoE metrics such as MOS, packet delay, and packet loss (configured in the Quality of Experience Profile table) and/or media bandwidth (configured in the Bandwidth profile table). When this occurs, the device sends a SIP 480 (Temporarily Unavailable) response to the SIP entity. This is configured by 1) assigning an IP Group a QoE and/or Bandwidth profile that rejects calls if the threshold is crossed, 2) configuring 806 in the Reasons for Tel-to-IP Alternative Routing table and 3) configuring an alternative routing rule.

Depending on configuration, the alternative routing is done using one of the following configuration entities:

- **Tel-to-IP Routing Rules:** You configure alternative routing rules for a specific routing rule in the Tel-to-IP Routing table. If the destination of the "main" routing rule is unavailable, the device searches the table for the next matching rule (e.g., destination phone number), and if available attempts to re-route the call to the IP destination configured for this alternative routing rule. For more information on configuring alternative Tel-to-IP routing rules, see "Configuring Tel-to-IP Routing Rules" on page 467. The table below shows an example of alternative routing where the device uses the first available alternative routing rule to re-route the initial, unsuccessful Tel-to-IP call destination.

Table 24-7: Alternative Routing based on SIP Response Code Example

	Destination Phone Prefix	IP Destination	SIP Response	Rule Used?
Main Route	40	10.33.45.68	408 Request Timeout	No
Alternative Route #1	40	10.33.45.70	486 Busy Here	No
Alternative Route #2	40	10.33.45.72	200 OK	Yes

- **Proxy Sets:** Proxy Sets are used for Server-type IP Groups (e.g., an IP PBX or proxy), which define the address (IP address or FQDN) of the server (see "Configuring Proxy Sets" on page 352). As you can configure multiple IP destinations per Proxy Set, the device supports proxy redundancy, which works together with the alternative routing feature. If the destination of a routing rule in the Tel-to-IP Routing table is an IP Group, the device routes the call to the IP destination configured for the Proxy Set associated with the IP Group. If the first IP destination of the Proxy Set is unavailable, the device attempts to re-route the call to the next proxy destination, and so on until an available IP destination is located. To enable the Proxy Redundancy feature for a Proxy Set, set the IsProxyHotSwap parameter to 1 and the EnableProxyKeepAlive parameter to 1.

  When the Proxy Redundancy feature is enabled, the device continually monitors the connection with the proxies by using keep-alive messages (SIP OPTIONS). The device sends these messages every user-defined interval (ProxyKeepAliveTime parameter). If the first (primary) proxy in the list replies with a SIP response code that you have also configured by the 'Keep-Alive Failure Responses' parameter, the device considers the Proxy as down; otherwise, the device considers the proxy as "alive". If the proxy is still considered down after a user-defined number of re-transmissions (configured by the HotSwapRtx parameter), the device attempts to communicate (using the same INVITE) with the next configured (redundant) proxy in the list, and so on until an available redundant proxy is located. Once an available proxy is located, the device can operate in one of the following modes (configured by the ProxyRedundancyMode parameter):

  - **Parking mode:** The device continues operating with the redundant proxy (now active) until the next failure occurs, after which it switches to the next redundant proxy.
  - **Homing mode:** The device always attempts to operate with the primary proxy. In other words, it switches back to the primary proxy whenever it's available again.

  If none of the proxy servers respond, the device goes over the list again.

Note:

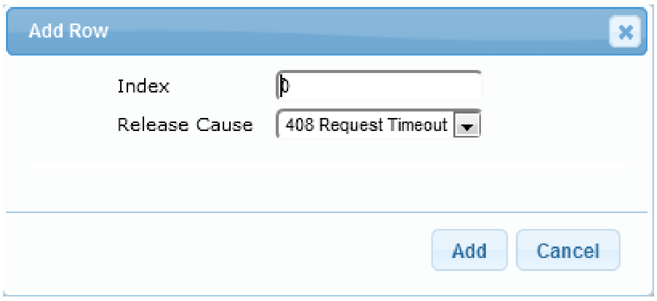
- The device assumes that all the proxy servers belonging to the Proxy Set are synchronized with regards to registered users. Thus, when the device locates an available proxy using the Hot Swap feature, it does not re-register the users; new registration (refresh) is done as normal.
- You can also enable the Busy Out feature, whereby the device can take specified actions if all Proxy Sets of associated destination IP Groups of matching routing rules in the Tel-to-IP Routing table do not respond to connectivity checks. For more information, see the EnableBusyOut parameter.
- If the AltRoutingTel2IPEnable parameter is enabled for the IP Connectivity feature (see Alternative Routing Based on IP Connectivity on page 486), the Busy Out feature does not function with the Proxy Set keep-alive mechanism (see below). To use the Busy Out feature with the Proxy Set keep-alive mechanism (for IP Groups), disable the AltRoutingTel2IPEnable parameter.

The steps for configuring alternative Tel-to-IP routing based on SIP response codes are summarized below.

➢ **To configure alternative Tel-to-IP routing based on SIP response codes:**

1. Configure SIP response codes (call failure reasons) that invoke alternative Tel-to-IP routing:
   a. Open the Reasons for Tel-to-IP Alternative Routing page (**Configuration** tab > **VoIP** menu > **Gateway** > **Routing** > **Alternative Reasons** > **Reasons for Tel-to-IP**).
   b. Click **Add**; the following dialog box appears:

Figure 24-7: Reasons for Tel-to-IP Alternative Routing Table - Add Row Dialog Box



Index: 0
Release Cause: 408 Request Timeout

   c. Configure a SIP response code for alternative routing according to the parameters described in the table below.
   d. Click **Add**.

Table 24-8: Reasons for Tel-to-IP Alternative Routing Table Parameter Descriptions

Parameter	Description
Index [AltRouteCauseTel2Ip_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Release Cause `rel-cause` [AltRouteCauseTel2Ip_ReleaseCause]	Defines a SIP response code that if received, the device attempts to route the call to an alternative destination (if configured).

2. Enable alternative routing based on SIP responses, by setting the 'Redundant Routing Mode' parameter in the Proxy & Registration page to one of the following:
   - **Routing Table:** Tel-to-IP Routing table is used for alternative routing.
   - **Proxy:** Proxy Set redundancy feature is used for alternative routing.
3. If you are using the Tel-to-IP Routing table, configure alternative routing rules with identical call matching characteristics, but with different IP destinations.
4. If you are using the Proxy Set, configure redundant proxies.

24.6.3 Alternative Routing upon SIP 3xx with Multiple Contacts

You can configure how the device handles received SIP 3xx responses that contain multiple alternative contacts. The 3xx response indicates that the original destination is unavailable (e.g., 301 Moved Permanently – user cannot be found) and that the call can be redirected to alternative destinations specified in the SIP Contact headers.

Configured by the '3xx Use Alt Route Reasons' parameter, the device can handle the receipt of 3xx responses using one of the following methods:

- The device tries each contact sequentially, listed in the Contact headers, until a successful destination is found. If a contact responds with a SIP 486 or 600, the device does not try to redirect the call to the next contact and drops the call.
- The device tries each contact sequentially, listed in the Contact headers. If a SIP 6xx Global Failure response is received during this process (e.g., 600 Busy Everywhere), the device does not try to redirect the call to the next contact and drops the call.
- The device redirects the call to the first contact listed in the Contact header. If the contact responds with a SIP response that is configured in the Reasons for Tel-to-IP Alternative Routing table (see "Alternative Routing Based on SIP Responses" on page 487), the device tries to redirect the call to the next contact, and so on. If a contact responds with a response that is not configured in the table, the device does not try to redirect the call to the next contact and drops the call.

Note: If a SIP 401 or 407 response is received from a contact, the device does not try to redirect the call to the next contact. Instead, the device continues with the regular authentication process, as indicated by these response types.

24.6.4 PSTN Fallback

The PSTN Fallback feature enables the device to re-route a Tel-to-IP call to the legacy PSTN using one of its trunks if the IP destination is unavailable. For example, if poor voice quality is detected over the IP network, the device attempts to re-route the call to the PSTN.

The steps for configuring alternative Tel-to-IP routing to the legacy PSTN are summarized below.

➢ **To configure alternative Tel-to-IP routing to the legacy PSTN:**

1. Configure an alternative routing rule in the Tel-to-IP Routing table with the same call matching characteristics (e.g., phone number destination), but where the destination is the IP address of the device itself.
2. Configure an IP-to-Tel routing rule in the Inbound IP Routing table to route calls received from the device (i.e., its IP address) to a specific Trunk Group connected to the PSTN. This configuration is necessary as the re-routed call is now considered an IP-to-Tel call. For configuring IP-to-Tel routing rules, see "Configuring IP to Trunk Group Routing Rules" on page 476.

Note: The PSTN Fallback feature is applicable only to digital interfaces.

24.7 Alternative Routing for IP-to-Tel Calls

The device supports alternative IP-to-Tel call routing, as described in this section.

24.7.1 Alternative Routing to Trunk upon Q.931 Call Release Cause Code

You can configure up to 10 ISDN Q.931 release cause codes, which if received from the Tel side, the device routes the IP-to-Tel call to an alternative Trunk Group, if configured. Alternative IP-to-Tel routing rules are configured in the Inbound IP Routing table. These rules must be configured anywhere below the "main" routing rule and with identical matching characteristics (e.g., destination prefix number) to the "main" routing rule. The device uses the first alternative route that is available. For more information on configuring alternative IP-to-Tel routing rules in the Inbound IP Routing table, see "Configuring IP-to-Trunk Group Routing Rules" on page 476.

A release cause code indicates that the IP-to-Tel call has been rejected or disconnected on the Tel side. The release cause codes are configured in the Reasons for IP-to-Tel Alternative Routing table. For example, you can configure alternative IP-to-Tel routing for scenarios where the initial Tel destination is busy and a Q.931 Cause Code No. 17 is received (or for other call releases that issue the default Cause Code No. 3).

You can configure a default release cause code that the device issues itself upon the following scenarios:

- The device initiates a call release whose cause is unknown.
- No free channels (i.e., busy) in the Trunk Group.
- No appropriate routing rule located in the Inbound IP Routing table.
- Phone number is not located in the Inbound IP Routing table.

This default release code is set to Cause Code No. 3 (No Route to Destination).You can change the code number using the 'Default Release Cause' parameter, located on the Advanced Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Advanced Parameters**).

⚠ **Notes:**

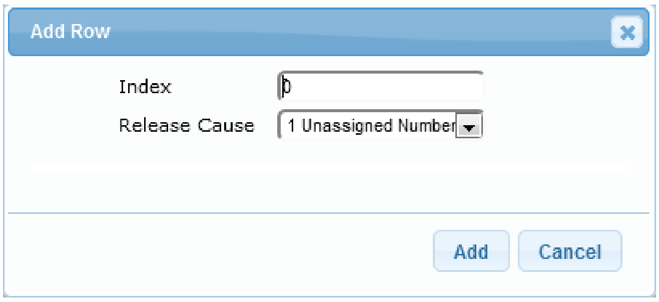
- If a Trunk is disconnected or not synchronized, the device issues itself the internal Cause Code No. 27. This cause code is mapped (by default) to SIP 502.
- The default release cause is described in the Q.931 notation and translated to corresponding SIP 40x or 50x values (e.g., Cause Code No. 3 to SIP 404, and Cause Code No. 34 to SIP 503).
- For mapping SIP-to-Q.931 and Q.931-to-SIP release causes, see Configuring Release Cause Mapping on page 457.

The following procedure describes how to configure alternative routing reasons for IP-to-Tel calls through the Web interface. You can also configure it through ini file (AltRouteCauseIP2Tel) or CLI (configure voip/gw routing alt-route-cause-ip2tel).

➢ **To configure alternative Trunk Group routing based on Q.931 cause codes:**

1. In the Proxy & Registration page, set the 'Redundant Routing Mode' parameter to **Routing Table** so that the device uses the Inbound IP Routing table for alternative routing.
2. In the Inbound IP Routing table, configure alternative routing rules with the same call matching characteristics, but with different Trunk Group destinations.
3. Configure Q.931 cause codes that invoke alternative IP-to-Tel routing:
   a. Open the Reasons for IP-to-Tel Alternative Routing table (**Configuration** tab > **VoIP** menu > **Gateway** > **Routing** > **Alternative Routing Reasons** > **Reasons for IP-to-Tel**).
   b. Click **Add**; the following dialog box appears:

Figure 24-8: Reasons for IP-to-Tel Alternative Routing Table - Add Row Dialog Box



Index: 0
Release Cause: 1 Unassigned Number

   c. Configure a Q.931 release cause code for alternative routing according to the parameters described in the table below.
   d. Click **Add**.

Table 24-9: Reasons for IP-to-Tel Alternative Routing Table Parameter Descriptions

Parameter	Description
Index [AltRouteCauseIP2Tel_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Release Cause `rel-cause` [AltRouteCauseIP2Tel_ReleaseCause]	Defines a Q.931 release code that if received, the device attempts to route the call to an alternative destination (if configured).

24.7.2 Alternative Routing to an IP Destination upon a Busy Trunk

The Forward on Busy Trunk Destination table lets you configure alternative routing rules for forwarding (i.e., call redirection) IP-to-Tel calls to an alternative IP destination using SIP 3xx responses. These rules are used upon the following:

- Trunk Group has no free channels (i.e., “busy”).

This feature is configured per Trunk Group. The alternative destination can be defined as a host name or as a SIP Request-URI user name and host part (i.e., user@host). For example, the below configuration forwards IP-to-Tel calls to destination user “112” at host IP address 10.13.4.12, port 5060, using transport protocol TCP, if Trunk Group ID 2 is unavailable:

```
ForwardOnBusyTrunkDest 1 = 2, 112@10.13.4.12:5060;transport=tcp;
```

When configured with user@host, the original destination number is replaced by the user part.

The device forwards calls using this table only if no alternative IP-to-Tel routing rule has been configured in the Inbound IP Routing table or alternative routing fails and the following reason(s) in the SIP Diversion header of 3xx messages exists:

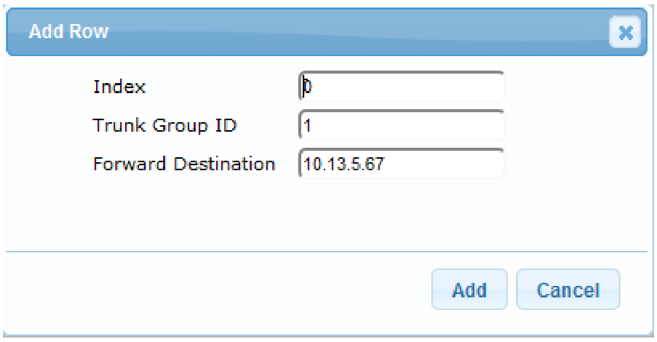
- “out-of-service” - all trunks are unavailable/disconnected
- "unavailable":
    - All trunks are busy or unavailable

The following procedure describes how to configure Forward on Busy Trunks through the Web interface. You can also configure it through ini file (ForwardOnBusyTrunkDest) or CLI (configure voip/gw routing fwd-on-bsy-trk-dst).

➢ To configure a Forward on Busy Trunk Destination rule:

1. Open the Forward on Busy Trunk Destination table (**Configuration** tab > **VoIP** menu > **Gateway** > **Routing** > **Forward on Busy Trunk**).
2. Click **Add**; the following dialog box appears:

Figure 24-9: Forward on Busy Trunk Destination Table - Add Row Dialog Box



Index: 0
Trunk Group ID: 1
Forward Destination: 10.13.5.67

The figure above displays a configuration that forwards IP-to-Tel calls destined for Trunk Group ID 1 to destination IP address 10.13.5.67 if the conditions mentioned earlier exist.

3. Configure a rule according to the parameters described in the table below.
4. Click **Add**, and then reset the device with a burn-to-flash for your settings to take effect.

Table 24-10: Forward on Busy Trunk Destination Parameter Descriptions

Parameter	Description
Index [ForwardOnBusyTrunkDest_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Trunk Group ID	Defines the Trunk Group ID to which the IP call is destined to.

Parameter	Description
`trunk-group-id` [ForwardOnBusyTrunkDest_TrunkGroupId]	
Forward Destination `forward-dst` [ForwardOnBusyTrunkDest_ForwardDestination]	Defines the alternative IP destination for the call used if the Trunk Group is busy or unavailable. The valid value can be an IP address in dotted-decimal notation, an FQDN, or a SIP Request-URI user name and host part (i.e., user@host). The following syntax can also be used: host:port;transport=xxx (i.e., IP address, port and transport type). **Note:** • If you do not specify a port, the device uses UDP port 5060. • When configured with a user@host, the original destination number is replaced by the user part.

24.7.3 Alternative Routing upon ISDN Disconnect

You can configure when the device sends a call to an alternative route if it receives an ISDN Q.931 Disconnect message with a Progress Indicator (PI) IE from the Tel side for IP-to-Tel calls. The Disconnect message indicates that the call cannot be established due to, for example, a busy state on the Tel side. Using the DisconnectCallwithPIifAlt ini file parameter, you can configure the following modes of operation:

- The device does not immediately disconnect the call. Instead, it waits for any subsequent media from the Tel side (e.g., "this number is currently busy") and forwards it to the IP side (SIP 183 for early media). Only when it receives a Q.931 Release message, does the device disconnect the call (sends a SIP BYE message to the IP side). If you have configured an alternative route, the device sends the IP call to the alternative route.
- The device immediately sends the IP call to an alternative route, if you have configured one. If no alternative route has been configured and the Disconnect message is received with PI, the device forwards the subsequent early media to the IP side. The device disconnects the IP call only upon receipt of the subsequent Release message.

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25 Configuring DTMF and Dialing

The DTMF & Dialing page is used to configure parameters associated with dual-tone multi-frequency (DTMF) and dialing. For a description of the parameters appearing on this page, see "Configuration Parameters Reference" on page 813.

➢ **To configure the DTMF and dialing parameters:**

1. Open the DTMF & Dialing page (**Configuration** tab > **VoIP** menu > **Gateway** > **DTMF & Supplementary** > **DTMF & Dialing**).
2. Configure the parameters as required.
3. Click **Submit**.
4. To save the changes to flash memory, see "Saving Configuration" on page 643.

25.1 Dialing Plan Features

This section describes various dialing plan features supported by the device.

25.1.1 Digit Mapping

Digit map pattern rules are used for Tel-to-IP ISDN overlap dialing (by setting the ISDNRxOverlap parameter to 1) to reduce the dialing period. For more information on digit maps for ISDN overlapping, see ISDN Overlap Dialing. The device collects digits until a match is found in the user-defined digit pattern (e.g., for closed numbering schemes). The device stops collecting digits and starts sending the digits (collected number) upon any of the following scenarios:

- Maximum number of digits is received. You can define (using the MaxDigits parameter) the maximum number of collected destination number digits that can be received from the Tel side by the device. When the number of collected digits reaches the maximum (or a digit map pattern is matched), the device uses these digits for the called destination number.
- Inter-digit timeout expires (e.g., for open numbering schemes). This is defined using the TimeBetweenDigits parameter. This is the time that the device waits between each received digit. When this inter-digit timeout expires, the device uses the collected digits to dial the called destination number.
- Digit string (i.e., dialed number) matches one of the patterns defined in the digit map.

Digit map (pattern) rules are defined using the DigitMapping parameter. The digit map pattern can contain up to 52 options (rules), each separated by a vertical bar ("|"). The maximum length of the entire digit pattern is 152 characters. The available notations are described in the table below:

Table 25-1: Digit Map Pattern Notations

Notation	Description
[n-m]	Range of numbers (not letters).
.	(single dot) Repeat digits until next notation (e.g., T).
x	Any single digit. **Note:** This notation does not apply to some scenarios when using the star (*) or hash (#) key. For example, the key sequence of ** must be presented in the dial plan as *x.s (instead of xx).
T	Dial timeout (configured by the TimeBetweenDigits parameter).

Notation	Description
S	Short timer (configured by the TimeBetweenDigits parameter; default is two seconds) that can be used when a specific rule is defined after a more general rule. For example, if the digit map is 99|998, then the digit collection is terminated after the first two 9 digits are received. Therefore, the second rule of 998 can never be matched. But when the digit map is 99s|998, then after dialing the first two 9 digits, the device waits another two seconds within which the caller can enter the digit 8.

Below is an example of a digit map pattern containing eight rules:

```
DigitMapping = 11xS|00[1-
7]xxx|8xxxxxxx|#xxxxxxx|*xx|91xxxxxxxxxx|9011x|xx.T
```

In the example, the rule "00[1-7]xxx" denotes dialed numbers that begin with 00, and then any digit from 1 through 7, followed by three digits (of any number). Once the device receives these digits, it does not wait for additional digits, but starts sending the collected digits (dialed number) immediately.

Notes:

- If you want the device to accept/dial any number, ensure that the digit map contains the rule "xx.T"; otherwise, dialed numbers not defined in the digit map are rejected.
- If you are using an external Dial Plan file for dialing plans (see "Dialing Plans for Digit Collection" on page 653), the device first attempts to locate a matching digit pattern in the Dial Plan file, and if not found, then attempts to locate a matching digit pattern in the Digit Map (configured by the DigitMapping parameter).
- It may be useful to configure both Dial Plan file and Digit Maps. For example, the Digit Map can be used for complex digit patterns (which are not supported by the Dial Plan) and the Dial Plan can be used for long lists of relatively simple digit patterns. In addition, as timeout between digits is not supported by the Dial Plan, the Digit Map can be used to define digit patterns (MaxDigits parameter) that are shorter than those defined in the Dial Plan, or left at default. For example, “xx.T” Digit Map instructs the device to use the Dial Plan and if no matching digit pattern, it waits for two more digits and then after a timeout (TimeBetweenDigits parameter), it sends the collected digits. Therefore, this ensures that calls are not rejected as a result of their digit pattern not been completed in the Dial Plan.

25.1.2 External Dial Plan File

The device can be loaded with a Dial Plan file with user-defined dialing plans. For more information, see "Dial Plan File" on page 653.

25.2 Interworking Keypad DTMFs for SIP-to-ISDN Calls

The device can interwork DTMF tones received from the IP to the PSTN, using the ISDN Keypad Facility information element (IE) in Q.931 INFORMATION messages. This feature is applicable only to the Euro ISDN variant (User side).

If the device receives from the IP side an INVITE message whose called party number (To header) contains the asterisk (*) or pound (#) character, or a SIP NOTIFY or SIP INFO message that contains these characters (e.g., 123**#456**), the device sends the character and the digits positioned to its right, as Keypad IE in the INFORMATION message. The device sends only the digits positioned before the character to the PSTN (in SETUP message) as the called party number. For example, if the device receives the below INVITE, it sends "123" to the PSTN as the called party number and #456 as Keypad IE in the INFORMATION message:

```
INVITE sip:%7B54443994-BDFF-413C-AE4F-
D039B0FFB134%7D@192.168.100.214:5064;transport=tcp;rinstance=9f25c
4452eff4acb SIP/2.0
To: sip:123#456@192.168.100.214;user=phone;x-type=unknown;x-
plan=unknown;x-pres=allowed
```

The destination number can be manipulated when this feature is enabled. Note that if manipulation before routing is required, the * and # characters should not be used, as the device will handle them according to the above keypad protocol. For example, a manipulation rule should not be configured to add #456 to the destination number. If manipulation after routing is required, the destination number to be manipulated will not include the keypad part. For example, if you configure a manipulation rule to add the suffix 888 and the received INVITE contains the number 123#456, only 123 is manipulated and the number dialed toward the PSTN is 123888; #456 is sent as keypad.

To enable this feature, use the ISDNKeypadMode parameter.

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26 Configuring Supplementary Services

This section describes SIP supplementary services that can enhance your telephone service.

Notes:

- All call participants must support the specific supplementary service that is used.
- When working with certain application servers (such as BroadSoft's BroadWorks) in client server mode (the application server controls all supplementary services and keypad features by itself), the device's supplementary services must be disabled.

The Supplementary Services page is used to configure many of the discussed supplementary services parameters. For a description of the parameters appearing on this page, see "Configuration Parameters Reference" on page 813.

➢ **To configure supplementary services parameters:**

1. Open the Supplementary Services page (**Configuration** tab > **VoIP** menu > **Gateway** > **DTMF & Supplementary** > **Supplementary Services**).
2. Configure the parameters as required.
3. Click **Submit**, or click the **Subscribe to MWI** or **Unsubscribe to MWI** buttons to save your changes and to subscribe / unsubscribe to the MWI server.
4. To save the changes to flash memory, see "Saving Configuration" on page 643.

26.1 Call Hold and Retrieve

Call Hold and Retrieve:

- The party that initiates the hold is called the *holding* party; the other party is called the *held* party.
- After a successful Hold, the holding party hears a dial tone (HELD_TONE defined in the device's Call Progress Tones file).
- After a successful retrieve, the voice is connected again.
- The hold and retrieve functionalities are implemented by re-INVITE messages. The IP address 0.0.0.0 as the connection IP address or the string 'a=inactive' in the received re-INVITE SDP cause the device to enter Hold state and to play the held tone (configured in the device) to the PBX/PSTN. If the string 'a=sendonly' is received in the SDP message, the device stops sending RTP packets, but continues to listen to the incoming RTP packets. Usually, the remote party plays, in this scenario, Music on Hold (MOH) and the device forwards the MOH to the held party.

You can also configure the device to keep a call on-hold for a user-defined time after which the call is disconnected, using the HeldTimeout parameter.

26.2 BRI Suspend and Resume

The device supports call suspend and resume services initiated by ISDN BRI phones connected to the device. During an ongoing call, the BRI phone user can suspend the call by typically pressing the phone's "P" button or a sequence of keys (depending on the phone), and then on-hooking the handset. To resume the call, the phone user typically presses the same keys or button again and then off-hooks the phone. During the suspended state, the device plays a howler tone to the remote party. This service is also supported when instead of pressing the call park button(s), the phone cable is disconnected (suspending the call) and then reconnected again (resuming the call).

If the phone user does not resume the call within a user-defined interval (configured using the HeldTimeout parameter), the device releases the call.

> **Note:** Only one call can be suspended per trunk. If another suspend request is received from a BRI phone while there is already a suspended call (even if done by another BRI phone connected to the same trunk), the device rejects this suspend request.

26.3 Call Transfer

This section describes the device's support for call transfer types.

26.3.1 Consultation Call Transfer

The device supports Consultation Call Transfer. The common method to perform a consultation transfer is described in the following example, which assumes three call parties:

- Party A = transferring
- Party B = transferred
- Party C = transferred to

1. A Calls B.
2. B answers.
3. A presses the hook-flash button and places B on-hold (party B hears a hold tone).
4. A dials C.
5. After A completes dialing C, A can perform the transfer by on-hooking the A phone.
6. After the transfer is complete, B and C parties are engaged in a call.

The transfer can be initiated at any of the following stages of the call between A and C:

- Just after completing dialing C phone number - transfer from setup
- While hearing ringback – transfer from alert
- While speaking to C - transfer from active

The device also supports attended (consultation) call transfer for BRI phones (user side) connected to the device and using the Euro ISDN protocol. BRI call transfer is according to ETSI TS 183 036, Section G.2 (Explicit Communication Transfer – ECT). Call transfer is enabled using the EnableTransfer and EnableHoldtoISDN parameters.

The Explicit Call Transfer (ECT, according to ETS-300-367, 368, 369) supplementary service is supported for BRI trunks. This service provides the served user who has two calls to ask the network to connect these two calls together and release its connection to both parties. The two calls can be incoming or outgoing calls. This service is similar to NI-2 Two B-Channel Transfer (TBCT) Supplementary Service. The main difference is that in ECT one of the calls must be in HELD state. The ECT standard defines two methods - Implicit and Explicit. In implicit method, the two calls must be on the same trunk. BRI uses the implicit mechanism.

.

26.3.2 Consultation Transfer for QSIG Path Replacement

The device can interwork consultation call transfer requests for ISDN QSIG-to-IP calls. When the device receives a request for a consultation call transfer from the PBX, the device sends a SIP REFER message with a Replaces header to the SIP UA to transfer it to another SIP UA. Once the two SIP UA parties are successfully connected, the device requests the PBX to disconnect the ISDN call, thereby freeing resources on the PBX.

For example, assume legacy PBX user "A" has two established calls connected through the device – one with remote SIP UA "B" and the other with SIP UA "C". In this scenario, user "A" initiates a consultation call transfer to connect "B" with "C". The device receives the consultation call transfer request from the PBX and then connects "B" with "C", by sending "B" a REFER message with a Replaces header (i.e., replace caller "A" with "C"). Upon receipt of a SIP NOTIFY 200 message in response to the REFER, the device sends a Q.931 Disconnect messages to the PBX, notifying the PBX that it can disconnect the ISDN calls (of user "A").

This feature is enabled by the QSIGPathReplacementMode parameter.

26.3.3 Blind Call Transfer

Blind call transfer is done (using SIP REFER messages) after a call is established between call parties A and B, and party A decides to immediately transfer the call to C without first speaking to C. The result of the transfer is a call between B and C (similar to consultation transfer, but skipping the consultation stage).

You can also use the ManipulateIP2PSTNReferTo parameter to manipulate the destination number according to the number received in the SIP Refer-To header. This is applicable to all types of blind transfers to the PSTN (e.g., TBCT, ECT, RLT, QSIG). During blind transfer, the device initiates a new call to the PSTN and the destination number of this call can be manipulated if the parameter is enabled. The following is an example of such a blind transfer:

1. IP phone "A" calls PSTN phone "B", and the call is established.
2. "A" performs a blind transfer to PSTN phone "C". It does this as follows:
   a. "A" sends a SIP REFER message (with the phone number of "C" in the Refer-To header) to the device.
   b. The device sends a Q.931 Setup message to "C". This feature enables manipulating the called party number in this outgoing Setup message.

The manipulation is done as follows:

1. If you configure a value for the xferPrefix parameter, then this value (string) is added as a prefix to the number in the Refer-To header.
2. This called party number is then manipulated using the IP-to-Tel Destination Phone Number Manipulation table.
3. The source number of the transferred call is taken from the original call, according to its initial direction:
   - Tel-to-IP call: source number of the original call.
   - IP-to-Tel call: destination number of the original call.
   - If the UseReferredByForCallingNumber parameter is set to 1, the source number is taken from the SIP Referred-By header if included in the received SIP REFER message.

   This source number can also be used as the value for the 'Source Prefix' field in the IP-to-Tel Destination Phone Number Manipulation table. The local IP address is used as the value for the 'Source IP Address' field.

Note: Manipulation using the ManipulateIP2PSTNReferTo parameter does not affect IP-to-Trunk Group routing rules.

26.4 Call Forward

The device supports Call Deflection (ETS-300-207-1) for Euro ISDN and QSIG (ETSI TS 102 393) for Network and User sides, which provides IP-ISDN interworking of call forwarding (call diversion) when the device receives a SIP 302 response.

Call forward performed by the SIP side: Upon receipt of a Facility message with Call Rerouting IE from the PSTN, the device initiates a SIP transfer process by sending a SIP 302 (including the Call Rerouting destination number) to the IP in response to the remote SIP entity's INVITE message. The device then responds with a Disconnect message to the PSTN side.

Call forward performed by the PSTN side: When the device sends the INVITE message to the remote SIP entity and receives a SIP 302 response, the device sends a Facility message with the same IE mentioned above to the PSTN, and waits for the PSTN side to disconnect the call. This is configured using the CallReroutingMode.

Notes:

- When call forward is initiated, the device sends a SIP 302 response with a contact that contains the phone number from the forward table and its corresponding IP address from the routing table (or when a proxy is used, the proxy’s IP address).
- For receiving call forward, the device handles SIP 3xx responses for redirecting calls with a new contact.

26.5 Remote Handling of BRI Call Forwarding

The device supports call forwarding (CF) services initiated by ISDN Basic BRI phones connected to it. Upon receipt of an ISDN Facility message for call forward from the BRI phone, the device sends a SIP INVITE to the softswitch with a user-defined code in the SIP To header, representing the reason for the call forward.

The codes for the call forward can be defined using the following parameters:

- SuppServCodeCFU - Call Forward Unconditional
- SuppServCodeCFUDeact - Call Forward Unconditional Deactivation
- SuppServCodeCFB - Call Forward on Busy
- SuppServCodeCFBDeact - Call Forward on Busy Deactivation
- SuppServCodeCFNR - Call Forward on No Reply
- SuppServCodeCFNRDeact - Call Forward on No Reply Deactivation

Note: These codes must be defined according to the settings of the softswitch (i.e., the softswitch must recognize them).

Below is an example of an INVITE message sent by the device indicating an unconditional call forward (“*72”) to extension number 100. This code is defined using the SuppServCodeCFU parameter.

```
INVITE sip:*72100@10.33.8.53;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.33.2.5:5060;branch=z9hG4bKWDSUKUHWFEXQSVOUVJGM
From: <sip:400@10.33.2.5;user=phone>;tag=DUOROSXSOYJJLNBFRQTG
To: <sip:*72100@10.33.8.53;user=phone>
Call-ID: GMNOVQRRXUUCYCQSFAHS@10.33.2.5
```

```
CSeq: 1 INVITE
Contact: <sip:400@10.33.2.5:5060>
Supported: em,100rel,timer,replaces
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUB
SCRIBE
User-Agent: Sip Message Generator V1.0.0.5
User-to-User: 31323334;pd=4
Content-Type: application/sdp
Content-Length: 155
```

You can also enable the device to indicate the type of CF service in the Request-URI of the outgoing SIP INVITE message. Upon receipt of an ISDN Facility message for call forward (Diversion) from the BRI phone, the device indicates the call forwarding service in the Request-URI header using a proprietary parameter “facility=<call forward service>”, where call forward service can be one of the following:

- “cfu-activate”: Call Forwarding Unconditional activated
- “cfu-deactivate”: Call Forwarding Unconditional deactivated
- “cfb-activate”: Call Forward on Busy activated
- “cfb-deactivate”: Call Forward on Busy deactivated
- “cfnr-activate”: Call Forward on No Reply activated
- “cfnr-deactivate”: Call Forward on No Reply deactivated

For example:

```
INVITE sip:400@10.33.2.48;user=phone;facility=cfu-activate SIP/2.0
```

To enable the feature, configure the UseFacilityInRequest ini file parameter to 1.

26.5.1 Local Handling of BRI Call Forwarding

You can configure the device to handle BRI call forwarding per BRI extension line, using the Supplementary Services table.

Upon receipt of an ISDN Facility message from the BRI phone user, the device retrieves and stores the user's call forwarding information. This includes whether the user has activated or de-actived call forwarding as well as the type of call forwarding service (Call Forward Busy, Call Forward No Reply, and Call Forward Unconditional). When the device receives a call (INVITE message) for a user whose phone number is configured in the 'Local Phone Number' field of the Supplementary Services table and call forwarding has been activated by the user, the device replies to the calling party with a SIP 302 (Moved Temporarily) response containing the configured call forwarding number corresponding to the call forwarding type. The call forwarding type and original called number (user's phone number) is sent in the SIP Diversion header, for example:

```
Diversion: <sip:401>;reason=unconditional;counter=1
```

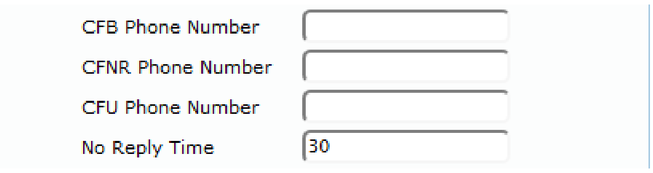
The call forwarding number is sent in the SIP Contact header, for example:

```
Contact: sip:567@10.33.77.17;user=phone
```

When call forwarding is activated and the user off-hooks the phone, the device plays the stutter dial tone (Tone Type #15) as a reminder to the user that call forwarding is activated.

➢ **To configure BRI call forwarding:**

1. Open the Supplementary Services table (see Configuring Multi-Line Extensions and Supplementary Services on page 513), and then configure BRI line extensions with the required call forwarding parameters:



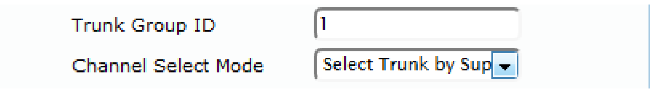
CFB Phone Number	
CFNR Phone Number	
CFU Phone Number	
No Reply Time	30

For more information, see Configuring Multi-Line Extensions and Supplementary Services on page 513.

2. Enable BRI call forwarding, by using the following ini file parameter setting:

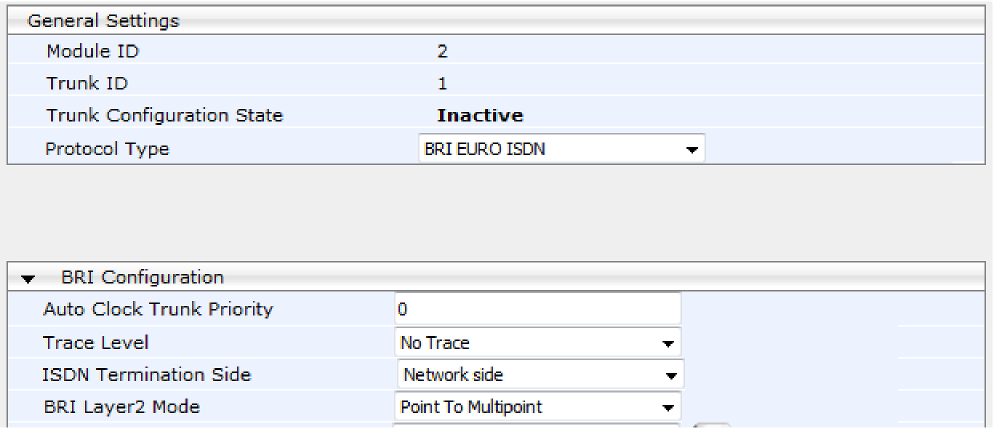
```
BRICallForwardHandling=1
```

3. Open the Trunk Group table (Configuring Trunk Groups on page 433), and then configure a Trunk Group (e.g., 1) for the BRI ports.
4. Open the Trunk Group Settings table (see Configuring Trunk Group Settings on page 435), and then for the Trunk Group ID to which the BRI ports belong, set the 'Channel Select Mode' parameter to **Select Trunk by Supp-Serv Table**:



Trunk Group ID	1
Channel Select Mode	Select Trunk by Sup

5. Open the Trunk Settings page (see Configuring Trunk Settings on page 423), and then make sure that you configure the BRI ports with the following settings:
   - 'Protocol Type': **BRI EURO ISDN**
   - 'ISDN Termination Side': **Network Side**
   - 'BRI Layer2 Mode': **Point to Multipoint**



General Settings

Module ID	2
Trunk ID	1
Trunk Configuration State	**Inactive**
Protocol Type	BRI EURO ISDN

BRI Configuration

Auto Clock Trunk Priority	0
Trace Level	No Trace
ISDN Termination Side	Network side
BRI Layer2 Mode	Point To Multipoint

26.6 Message Waiting Indication

The device supports Message Waiting Indication (MWI) according to IETF RFC 3842. The device also supports subscribing to an MWI server (using SIP SUBSCRIBE messages).

Note: For more information on configuring IP-based voice mail, refer to the *IP Voice Mail CPE Configuration Guide*.

To configure MWI, use the following parameters:

- EnableMWI
- MWIServerIP, or MWISubscribeIPGroupID and ProxySet
- EnableMWISubscription
- MWIExpirationTime
- SubscribeRetryTime
- SubscriptionMode

- VoiceMailInterface
- EnableVMURI

The device supports the following digital PSTN-based MWI features:

- ISDN BRI: The device supports MWI for its BRI phones, using the Euro ISDN BRI variant. When this feature is activated and a voice mail message is recorded to the mail box of a BRI extension, the softswitch sends a notification to the device. In turn, the device notifies the BRI extension and a red light flashes on the BRI extension’s phone. Once the voice message is retrieved, the MWI light on the BRI phone turns off. This is configured by setting the VoiceMailInterface parameter to 8 (“ETSI”) and enabled by the EnableMWI parameter.
- Euro-ISDN MWI: The device supports Euro-ISDN MWI for IP-to-Tel calls. The device interworks SIP MWI NOTIFY messages to Euro-ISDN Facility information element (IE) MWI messages. This is configured by setting the VoiceMailInterface parameter to 8.
- QSIG MWI: The device supports the interworking of QSIG MWI to IP (in addition to interworking of SIP MWI NOTIFY to QSIG Facility MWI messages). This provides interworking between an ISDN PBX with voice mail capabilities and a softswitch, which requires information on the number of messages waiting for a specific user. This support is configured using the TrunkGroupSettings_MWIInterrogationType parameter (in the Trunk Group Settings table), which determines the device's handling of MWI Interrogation messages. The process for sending the MWI status upon request from a softswitch is as follows:

  1. The softswitch sends a SIP SUBSCRIBE message to the device.
  2. The device responds by sending an empty SIP NOTIFY to the softswitch, and then sending an ISDN Setup message with Facility IE containing an MWI Interrogation request to the PBX.
  3. The PBX responds by sending to the device an ISDN Connect message containing Facility IE with an MWI Interrogation result, which includes the number of voice messages waiting for the specific user.
  4. The device sends another SIP NOTIFY to the softswitch, containing this MWI information.
  5. The SIP NOTIFY messages are sent to the IP Group defined by the NotificationIPGroupID parameter.

  When a change in the status occurs (e.g., a new voice message is waiting or the user has retrieved a message from the voice mail), the PBX initiates an ISDN Setup message with Facility IE containing an MWI Activate request, which includes the new number of voice messages waiting for the user. The device forwards this information to the softswitch by sending a SIP NOTIFY.

  Depending on PBX support, the MWIInterrogationType parameter can be configured to handle these MWI Interrogation messages in different ways. For example, some PBXs support only the MWI Activate request (and not MWI Interrogation request). Some support both these requests. Therefore, the device can be configured to disable this feature or enable it with one of the following support:

  - Responds to MWI Activate requests from the PBX by sending SIP NOTIFY MWI messages (i.e., does not send MWI Interrogation messages).
  - Send MWI Interrogation message, but don't use its result. Instead, wait for MWI Activate requests from the PBX.
  - Send MWI Interrogation message, use its result, and use the MWI Activate requests.

26.7 Three-Way Conferencing

The device supports three-way conference calls. Multiple, concurrent three-way conference calls are also supported. The device supports the following conference modes:

- **Conference Managed by External, AudioCodes Conferencing (Media) Server:**

The conference-initiating INVITE sent by the device uses the ConferenceID concatenated with a unique identifier as the Request-URI. This same Request-URI is set as the Refer-To header value in the REFER messages that are sent to the two remote parties. This mode is configured by setting the 3WayConferenceMode parameter to 0 (default.)

To join a conference, the Request-URI includes the Conference ID string preceded by the number of the participants in the conference and terminated by a unique number. INVITE messages with the same URI join the same conference. For example:

```
INVITE sip:4conf1234@10.1.10.10
```

- **Local, On-board Conferencing:** The conference is established on the device without the need for an external Conferencing server. The device supports up to four simultaneous, on-board, three-way conference calls. This feature includes local mixing and transcoding of the 3-Way Call legs on the device, and even allowing multi-codec conference calls. This mode is configured by setting the 3WayConferenceMode parameter to 2.

Notes:

- Each on-board, three-way conference call utilizes the resources of two DSP channels, thereby reducing overall channel capacity by two. For example, for two three-way conference calls, four DSP channels are utilized, thereby reducing channel capacity from eight (four BRI ports * two) to four.
- Instead of using the flash-hook button to establish a three-way conference call, you can dial a user-defined hook-flash code (e.g., "*1"), configured by the HookFlashCode parameter.
- Three-way conferencing support for the BRI phones connected to the device complies with ETS 300 185.

The following example demonstrates three-way conferencing using the device's local, on-board conferencing feature. In this example, telephone "A" connected to the device establishes a three-way conference call with two remote IP phones, "B" and "C":

1. A establishes a regular call with B.
2. A places B on hold, by pressing the telephone's flash-hook button and the number "1" key.
3. A hears a dial tone and then makes a call to C.
4. C answers the call.
5. A establishes a three-way conference call with B and C, by pressing the flash-hook button and the number "3" key.

To configure local, on-board three-way conferencing:

1. Open the Supplementary Services page.
2. Set 'Enable 3-Way Conference' to **Enable** (Enable3WayConference = 1).
3. Set '3-Way Conference Mode' to **On Board** (3WayConferenceMode = 2).

Set 'Flash Keys Sequence Style' to **Sequence 1** or **Sequence 2** (FlashKeysSequenceStyle = 1 or 2).

26.8 Emergency E911 Phone Number Services

This section describes the device's support for emergency phone number services.

26.8.1 Pre-empting Existing Calls for E911 IP-to-Tel Calls

If the device receives an E911 call from the IP network destined to the Tel, and there are unavailable channels (e.g., all busy), the device terminates one of the calls (arbitrary) and then sends the E911 call to that channel. The preemption is done only on a channel pertaining to the same Trunk Group for which the E911 call was initially destined and if the channel select mode (configured by the ChannelSelectMode parameter) is set to a value other than “By Dest Phone Number” (0).

The preemption is done only if the incoming IP-to-Tel call is identified as an emergency call. The device identifies emergency calls by one of the following:

- The destination number of the IP call matches one of the numbers defined by the EmergencyNumbers parameter. For E911, you must defined the parameter with the value "911".
- The Priority header of the incoming SIP INVITE message contains the “emergency” value.

Emergency pre-emption of calls can be enabled for all calls, using the global parameter CallPriorityMode, or for specific calls using the Tel Profile parameter CallPriorityMode.

Notes:

- For Trunk Groups configured with call preemption, all must be configured to MLPP [1] or all configured to Emergency [2]. In other words, you cannot set some trunks to [1] and some to [2].
- The global parameter must be set to the same value as that of the Tel Profile parameter; otherwise, the Tel Profile parameter is not applied.
- If you configure call preemption using the global parameter and a new Tel Profile is subsequently added, the TelProfile_CallPriorityMode parameter automatically acquires the same setting as well.

26.9 Multilevel Precedence and Preemption

The device supports Multilevel Precedence and Preemption (MLPP) service. MLPP is a call priority scheme, which does the following:

- Assigns a precedence level (priority level) to specific phone calls or messages.
- Allows higher priority calls (*precedence call*) and messages to preempt lower priority calls and messages (i.e., terminates existing lower priority calls) that are recognized within a user-defined domain (*MLPP domain ID*). The domain specifies the collection of devices and resources that are associated with an MLPP subscriber. When an MLPP subscriber that belongs to a particular domain places a precedence call to another MLPP subscriber that belongs to the same domain, MLPP service can preempt the existing call that the called MLPP subscriber is on for a higher-precedence call. MLPP service availability does not apply across different domains.

MLPP is typically used in the military where, for example, high-ranking personnel can preempt active calls during network stress scenarios such as a national emergency or degraded network situations.

MLPP can be enabled for all calls, using the global parameter, CallPriorityMode, or for specific calls using the Tel Profile parameter, CallPriorityMode.

Notes:

- The device provides MLPP interworking between SIP and ISDN (both directions).
- For Trunk Groups configured with call preemption, all must be configured to MLPP [1] or all configured to Emergency [2]. In other words, you cannot set some trunks to [1] and some to [2].
- The global parameter must be set to the same value as that of the Tel Profile parameter; otherwise, the Tel Profile parameter is not applied.
- If you configure call preemption using the global parameter and a new Tel Profile is subsequently added, the TelProfile_CallPriorityMode parameter automatically acquires the same setting as well.

The Resource Priority value in the Resource-Priority SIP header can be any one of those listed in the table below. A default MLPP call Precedence Level (configured by the SIPDefaultCallPriority parameter) is used if the incoming SIP INVITE or ISDN Setup message contains an invalid priority or Precedence Level value respectively. For each MLPP call priority level, the Multiple Differentiated Services Code Points (DSCP) can be set to a value from 0 to 63.

Table 26-1: MLPP Call Priority Levels (Precedence) and DSCP Configuration Parameters

MLPP Precedence Level	Precedence Level in Resource-Priority SIP Header	DSCP Configuration Parameter
0 (lowest)	routine	MLPPRoutineRTPDSCP
2	priority	MLPPPriorityRTPDSCP
4	immediate	MLPPImmediateRTPDSCP
6	flash	MLPPFlashRTPDSCP
8	flash-override	MLPPFlashOverRTPDSCP
9 (highest)	flash-override-override	MLPPFlashOverOverRTPDSCP

The device automatically interworks the network identity digits (NI) in the ISDN Q.931 Precedence Information Element (IE) to the network domain subfield of the INVITE's Resource-Priority header, and vice versa. The SIP Resource-Priority header contains two fields, namespace and priority. The namespace is subdivided into two subfields, network-domain and precedence-domain. Below is an example of a Resource-Priority header whose network-domain subfield is "uc", r-priority field is "priority" (2), and precedence-domain subfield is "000000":

```
Resource-Priority: uc-000000.2
```

The MLPP Q.931 Setup message contains the Precedence IE. The NI digits are presented by four nibbles found in octets 5 and 6. The device checks the NI digits according to the translation table of the Department of Defense (DoD) Unified Capabilities (UC) Requirements (UCR 2008, Changes 3) document, as shown below:

Table 26-2: NI Digits in ISDN Precedence

Level IE	Network Domain in SIP Resource-Priority Header
0000	uc
0001	cuc
0002	dod
0003	nato

Notes:

- If the received ISDN message contains NI digits that are not listed in the translation table, the device sets the network-domain to "uc" in the outgoing SIP message.
- If the received SIP message contains a network-domain value that is not listed in the translation table, the device sets the NI digits to "0000" in the outgoing ISDN message.
- If the received ISDN message does not contain a Precedence IE, you can configure the namespace value - dsn (default), dod, drsn, uc, or cuc - in the SIP Resource-Priority header of the outgoing INVITE message. This is done using the MLPPDefaultNamespace parameter. You can also configure up to 32 user-defined namespaces, using the table ini file parameter, ResourcePriorityNetworkDomains. Once defined, you need to set the MLPPDefaultNamespace parameter value to the desired table row index.

By default, the device maps the received Resource-Priority field of the SIP Resource-Priority header to the outgoing ISDN Precedence Level (priority level) field as follows:

- If the network-domain field in the Resource-Priority header is "uc", then the device sets the Precedence Level field in the ISDN Precedence Level IE according to Table 5.3.2.12-4 (Mapping of RPH r-priority Field to ISDN Precedence Level Value):

Table 26-3: Mapping of SIP Resource-Priority Header to ISDN Precedence Level for MLPP

MLPP Precedence Level	ISDN Precedence Level	SIP Resource-Priority Header Field
Routine	4	0
Priority	3	2
Immediate	2	4
Flash	1	6
Flash Override	0	8

- If the network-domain field in the Resource-Priority header is any value other than "uc", then the device sets the Precedence Level field to "0 1 0 0" (i.e., "routine").

This can be modified using the EnableIp2TelInterworkingtable field of the ini file parameter, ResourcePriorityNetworkDomains.

Notes:

- If required, you can exclude the "resource-priority" tag from the SIP Require header in INVITE messages for Tel-to-IP calls when MLPP priority call handling is used. This is configured using the RPRequired parameter.
- For a complete list of the MLPP parameters, see "MLPP and Emergency Call Parameters" on page 938.

26.9.1 MLPP Preemption Events in SIP Reason Header

The device sends the SIP Reason header (as defined in RFC 4411) to indicate the reason and type of a preemption event. The device sends a SIP BYE or CANCEL request, or SIP 480, 486, 488 response (as appropriate) with a Reason header whose Reason-params can includes one of the following preemption cause classes:

- Reason: preemption ;cause=1 ;text=”UA Preemption”
- Reason: preemption ;cause=2 ;text=”Reserved Resources Preempted”
- Reason: preemption ;cause=3 ;text=”Generic Preemption”
- Reason: preemption ;cause=4 ;text=”Non-IP Preemption”

  This Reason cause code indicates that the session preemption has occurred in a non-IP portion of the infrastructure. The device sends this code in the following scenarios:

  - The device performs a network preemption of a busy call (when a high priority call is received), the device sends a SIP BYE or CANCEL request with this Reason cause code.
  - The device performs a preemption of a B-channel for a Tel-to-IP outbound call request from the softswitch for which it has not received an answer response (e.g., Connect), and the following sequence of events occurs:

    a. The device sends a Q.931 DISCONNECT over the ISDN MLPP to the partner switch to preempt the remote end instrument.
    b. The device sends a 488 (Not Acceptable Here) response with this Reason cause code.

- Reason: preemption; cause=5; text=”Network Preemption”

  This Reason cause code indicates preempted events in the network. Within the Defense Switched Network (DSN) network, the following SIP request messages and response codes for specific call scenarios have been identified for signaling this preemption cause:

  - SIP:BYE - If an active call is being preempted by another call
  - CANCEL - If an outgoing call is being preempted by another call
  - 480 (Temporarily Unavailable), 486 (User Busy), 488 (Not Acceptable Here) - Due to incoming calls being preempted by another call.

  The device receives SIP requests with preemption reason cause=5 in the following cases:

  - The softswitch performs a network preemption of an active call - the following sequence of events occurs:

    a. The softswitch sends the device a SIP BYE request with this Reason cause code.
    b. The device initiates the release procedures for the B-channel associated with the call request and maps the preemption cause to ISDN Cause = #8 ‘Preemption’. This value indicates that the call is being preempted. For ISDN, it also indicates that the B-channel is not reserved for reuse.
    c. The device sends a SIP 200 OK in response to the received BYE, before the SIP end instrument can proceed with the higher precedence call.

  - The softswitch performs a network preemption of an outbound call request for the device that has not received a SIP 2xx response - the following sequence of events occur:

    a. The softswitch sends the device a SIP 488 (Not Acceptable Here) response code with this Reason cause code. The device initiates the release procedures for the B-channel associated with the call request and maps the preemption cause to ISDN Cause = #8 ‘Preemption’.
    b. The device deactivates any user signaling (e.g., ringback tone) and when the call is terminated, it sends a SIP ACK message to the softswitch.

26.9.2 Precedence Ring Tone

You can configure the duration for which the device plays a preemption tone to the Tel and IP sides if a call is preempted, using the PreemptionToneDuration parameter.

26.10 Configuring Multi-Line Extensions and Supplementary Services

The Supplementary Services table lets you configure up to 100 supplementary services for endpoints connected to the device. These endpoints include Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) phones.

The table can be used for the following functionalities:

- Configuring multiple phone line extension numbers per port, supporting point-to-multipoint configuration of several phone numbers per channel.
- Registration of each line extension (endpoint), using a user-defined user ID and password, to a third-party softswitch for authentication and/or billing. For each line extension, the device sends a SIP REGISTER to the softswitch, using the global number in the From/To headers. If authentication is necessary for registration, the device sends the endpoint’s user ID and password in the SIP MD5 Authorization header. For viewing registration status, see "Viewing Registration Status" on page 728.
- BRI call forwarding services for point-to-multipoint configurations (according to ETSI 300 207-1) - Call Forward Busy (CFB), Call Forward No Reply (CFNR), and Call Forward Unconditional (CFU).
- Caller ID name per line extension, which is displayed to the called party (if enabled).
- Enabling receipt by the line extension of caller ID from incoming calls.
- Routing IP-to-Tel calls (including voice and fax) to specific endpoints based on called line extension number (local number). To enable this functionality, in the Trunk Group Settings table, set the 'Channel Select Mode' field to **Select Trunk by Supplementary Services Table** for the Trunk Group to which the port belongs (see "Configuring Hunt Group Settings" on page 435).
- Mapping local numbers (line extension number) with global phone numbers (E.164). The endpoint can be configured with two numbers – *local* and *global*. The local number represents the endpoint’s line extension number (e.g., PBX extension number); the global number represents the corresponding E.164 number used for the IP side in the SIP message:
  - IP-to-Tel calls: Maps the called global number in the user part of the SIP Request-URI in the incoming SIP message to the local number sent to the Tel side. For example, the device receives an incoming IP call with a destination (called) that is a global number 638002 and then sends the call to the Tel side with the destination number manipulated to the corresponding local number of 402.
  - Tel-to-IP Calls: Maps the calling (source) local number of the Tel side to the global number sent to the IP side (in the From and To headers of the outgoing SIP message). For example, if the device receives a Tel call from line extension local number 402, it changes this calling number to 638002 and then sends the call to the IP side with this calling number. This functionality in effect, validates the calling number.

Notes:

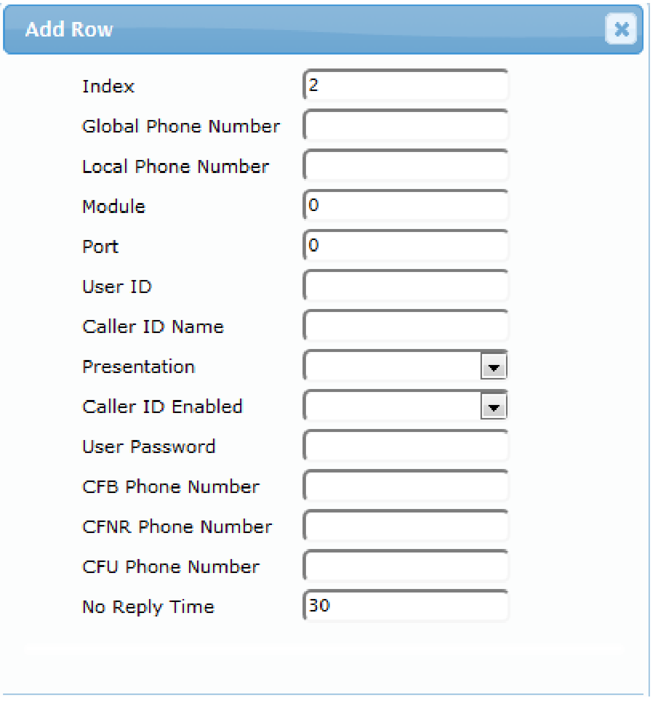
- If you have configured regular Tel-to-IP or IP-to-Tel manipulation rules (see "Configuring Source/Destination Number Manipulation Rules" on page 441), the device applies them before applying the local-global mapping rules configured in the table.
- To allow the end-user to hear a dial tone when picking up the BRI phone, it is recommended to set the Progress Indicator in the Setup Ack bit (0x10000=65536). Therefore, the recommended value is 0x10000 + 0 x1000 = 65536 + 4096 = 69632 (i.e., set the ISDNInCallsBehavior parameter to 69632).

The following procedure describes how to configure the Supplementary Services table through the Web interface. You can also configure it through ini file (ISDNSuppServ) or CLI (configure voip > gw digitalgw isdn-supp-serv).

➢ **To configure endpoint supplementary services:**

1. Open the Supplementary Services table (**Configuration** tab > **VoIP** menu > **Gateway** > **DTMF and Supplementary** > **Supp Services Table**).
2. Click **Add**; the following dialog box appears:

Figure 26-1: Supplementary Services Table - Add Row Dialog Box



Add Row

Index	2
Global Phone Number	
Local Phone Number	
Module	0
Port	0
User ID	
Caller ID Name	
Presentation	
Caller ID Enabled	
User Password	
CFB Phone Number	
CFNR Phone Number	
CFU Phone Number	
No Reply Time	30

3. Configure a supplementary service according to the parameters described in the table below.
4. Click **Add**.

You can register and un-register an endpoint configured in the table. The registration method is according to the 'Registration Mode' parameter located in the Trunk Group Settings page (see "Configuring Trunk Group Settings" on page 435).

➢ **To register or un-register an endpoint:**

1. Select the required table row in which the endpoint is configured.
2. From the 'Action' drop-down list, select **Register**. To unregister the endpoint, select **Un-Register**.

Table 26-4: Supplementary Services Table Parameter Description

Parameter	Description
Index [ISDNSuppServ_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.

Parameter	Description
Global Phone Number `phone-number` [ISDNSuppServ_PhoneNumber]	Defines a global telephone extension number for the endpoint. The global number is used for the following functionalities: ▪ Endpoint registration ▪ IP-to-Tel routing ▪ Mapping between local and global (E.164) numbers between Tel and IP sides respectively
Local Phone Number `local-phone-number` [ISDNSuppServ_LocalPhoneNumber]	Defines a local telephone extension number for the endpoint (e.g., the PBX extension number). The local number is used for the following functionalities: ▪ Validation of source (calling) number for Tel-to-IP calls ▪ Mapping between local and global (E.164) numbers between Tel and IP sides respectively
Module `module` [ISDNSuppServ_Module]	Defines the device's module number to which the endpoint is connected.
Port `port` [ISDNSuppServ_Port]	Defines the port number on the module to which the endpoint is connected.
User ID `user-id` [ISDNSuppServ_UserId]	Defines the User ID for registering the endpoint to a third-party softswitch for authentication and/or billing.
Caller ID Name `caller-id-number` [ISDNSuppServ_CallerID]	Defines the caller ID name of the endpoint (sent to the IP side). The valid value is a string of up to 18 characters.
Presentation `presentation-restricted` [ISDNSuppServ_IsPresentationRestricted]	Determines whether the endpoint sends its Caller ID information to the IP when a call is made. ▪ **[0]** Allowed = The device sends the string defined in the 'Caller ID' field when this endpoint makes a Tel-to-IP call. ▪ **[1]** Restricted = The string defined in the 'Caller ID' field is not sent.
Caller ID Enabled `caller-id-enable` [ISDNSuppServ_IsCallerIDEnabled]	Enables the receipt of Caller ID. ▪ **[0]** Disabled = The device does not send Caller ID information to the endpoint. ▪ **[1]** Enabled = The device sends Caller ID information to the endpoint.
User Password `user-password` [ISDNSuppServ_UserPassword]	Defines the user password for registering the endpoint to a third-party softswitch for authentication and/or billing. **Note:** For security, the password is displayed as an asterisk (*).

Parameter	Description
CFB Phone Number cfb-to_phone-number [ISDNSuppServ_CFB2PhoneNumber]	Defines the phone number for BRI Call Forward Busy (CFB) services. If the BRI extension is currently in use, the device forwards the call to this number. Note: ▪ The parameter is applicable only to BRI interfaces. ▪ To enable BRI call forwarding services, see the BRICallForwardHandling parameter. ▪ For more information on configuring local handling of BRI call forwarding, see Local Handling of BRI Call Forwarding on page 505.
CFNR Phone Number cfnr-to_phone-number [ISDNSuppServ_CFNR2PhoneNumber]	Defines the phone number for BRI Call Forward No Reply (CFNR) services. If the BRI extension does not answer the call within a user-defined timeout (see the 'No Reply Time' parameter below), the device forwards the call to this number. Note: ▪ The parameter is applicable only to BRI interfaces. ▪ To enable BRI call forwarding services, see the BRICallForwardHandling parameter. ▪ For more information on configuring local handling of BRI call forwarding, see Local Handling of BRI Call Forwarding on page 505.
CFU Phone Number cfu-to_phone-number [ISDNSuppServ_CFU2PhoneNumber]	Defines the phone number for BRI Call Forward Unconditional (CFU) services. The device always forwards the call to this number. Note: ▪ The parameter is applicable only to BRI interfaces. ▪ To enable BRI call forwarding services, see the BRICallForwardHandling parameter. ▪ For more information on configuring local handling of BRI call forwarding, see Local Handling of BRI Call Forwarding on page 505.
No Reply Time no-reply-time [ISDNSuppServ_NoReplyTime]	Defines the timeout (in seconds) that if the BRI extension does not answer before it expires, the device forwards the call to the phone number as defined by the 'CFNR Phone Number' parameter (see above). The default is 30. Note: ▪ The parameter is applicable only to BRI interfaces. ▪ To enable BRI call forwarding services, see the BRICallForwardHandling parameter.

Parameter	Description
	▪ For more information on configuring local handling of BRI call forwarding, see Local Handling of BRI Call Forwarding on page 505.

26.11 Detecting Collect Calls

The device detects collect calls (reverse charge calls) using any of the following information elements (IE) in the received Q.931 ISDN Setup message for Tel-to-IP calls:

- Reverse Charging Indication IE
- Facility IE

When the device detects a collect call, it adds a proprietary header (*X-Siemens-Call-Type: collect call*) to the outgoing SIP INVITE message.

This support does not require any configuration and is applicable to the Euro ISDN protocol variant.

26.12 Advice of Charge Services for Euro ISDN

Advice of charge (AOC) is a pre-billing function that tasks the rating engine with calculating the cost of using a service and relaying that information back to the customer (caller). This allows users to obtain call charging information during the call (AOC-D) or at the end of the call (AOC-E).

The AOC messages are sent in the EURO ISDN Facility Information Element (IE) message. The device interworks these ISDN messages with SIP by converting the AOC messages into SIP INFO (during call) and BYE messages (end of call) using AudioCodes proprietary SIP AOC header, and vice versa. The device supports both currency (monetary units) and pulse (non-monetary units) AOC messages.

This feature can typically be implemented in the hotel industry, where external calls made by guests can be billed accurately. In such a setup, the device is connected on one side to a PBX through an ISDN line (Euro ISDN), and on the other side to a SIP trunk provided by an ITSP. When a call is made by a guest, the device first sends an AOC-D Facility message to the PBX indicating the connection charge unit, and then sends subsequent AOC-D messages every user-defined interval to indicate the charge unit during the call. When the call ends, the device sends an AOC-E Facility message to the PBX indicating the total number of charged units.

The device supports various methods for AOC:

- **Tel-to-IP Direction:** The device converts the AOC messages received in the EURO ISDN Facility IE messages into SIP INFO and BYE messages using AudioCodes proprietary SIP AOC header.
- **Device Generation of AOC to Tel:** The device generates the metering tones according to user-defined pulses and intervals, configured in the Charge Code table (see "Configuring Charge Codes" on page 520). These include:
  - 'Pulses On Answer' - number of charging units in the first generated AOC-D Facility message.
  - 'Pulse Interval' - time between every sent AOC-D Facility message.
  - 'End Time' - time at which the charge code ends.
- **IP-to-Tel Direction:**
  - SIP-to-Tel interworking: The device uses the AOC header from the IP side and sends to Tel in EURO ISDN Facility IE messages. Below shows the SIP AOC header:

    ```
    AOC: charged; <parameters>
    ```

    Where parameters can be:

    - state="active" or "terminated"
    - charging-info="currency" or "pulse"

If "currency", the following parameters are available:

- ♦ currency=<string>
- ♦ currency-type="iso4217-a" or <string>
- ♦ amount=<number>
- ♦ multiplier=("0.001","0.01","0.1","1","10","100","1000")

If "pulse", the following parameter is available:

- ♦ recorded-units=<number>

- TELES proprietary method
- Cirpack proprietary methods

For more information on the proprietary methods, see the PayPhoneMeteringMode parameter in "Metering Tone Parameters" on page 975.

➢ To configure AOC:

1. Make sure that the PSTN protocol for the trunk line is Euro ISDN and set to network side.
2. Make sure that the date and time of the device is correct. For accuracy, it is recommended to use an NTP server to obtain the date and time.
3. Configure the required AOC method:
   - **Device Generation of AOC to Tel:**
     a. On the Supplementary Services page (**Configuration** tab > **VoIP** menu > **GW** > **DTMF and Supplementary** > **Supplementary Services**), set the 'Generate Metering Tones' parameter (PayPhoneMeteringMode) to **Internal Table**.
     b. Configure the Charge Codes in the Charge Codes table ("Configuring Charge Codes" on page 520).
     c. Assign the Charge Code index to the relevant Tel-to-IP routing rule in the Tel-to-IP Routing table (see "Configuring Tel-to-IP Routing Rules" on page 467).
   - **AOC in the Tel-to-IP Direction:** On the Supplementary Services page (**Configuration** tab > **VoIP** menu > **GW** > **DTMF and Supplementary** > **Supplementary Services**), set 'AoC Support' to **Enable**, to enable the AOC service for sending AOC to IP.
   - **AOC in the IP-to-Tel Direction:** On the Supplementary Services page (**Configuration** tab > **VoIP** menu > **GW** > **DTMF and Supplementary** > **Supplementary Services**), configure the 'Generate Metering Tones' parameter (PayPhoneMeteringMode) to the required method (**SIP Interval Provided**, **SIP RAW Data Provided**, **SIP RAW Data Incremental Provided**, or **SIP 2 TEL INTERWORKING**).

Note: This feature is applicable to Euro ISDN BRI only.

26.13 Configuring Charge Codes

The Charge Codes table lets you configure metering tones Advice of Charge (AOC) services for Euro ISDN trunks (see Advice of Charge Services for Euro ISDN on page 518).

You can configure up to 25 different Charge Codes, where each table row represents a Charge Code. Each Charge Code can include up to four different time periods in a day (24 hours). The device selects the time period by comparing the device's current time to the end time of each time period of the selected Charge Code. The device generates the number of pulses on answer once the call is connected, and from that point on, it generates a pulse for each pulse interval. If a call starts at a certain time period and crosses to the next, the information of the next time period is used.

To assign a Charge Code to an outgoing Tel-to-IP call, use the Tel-to-IP Routing table.

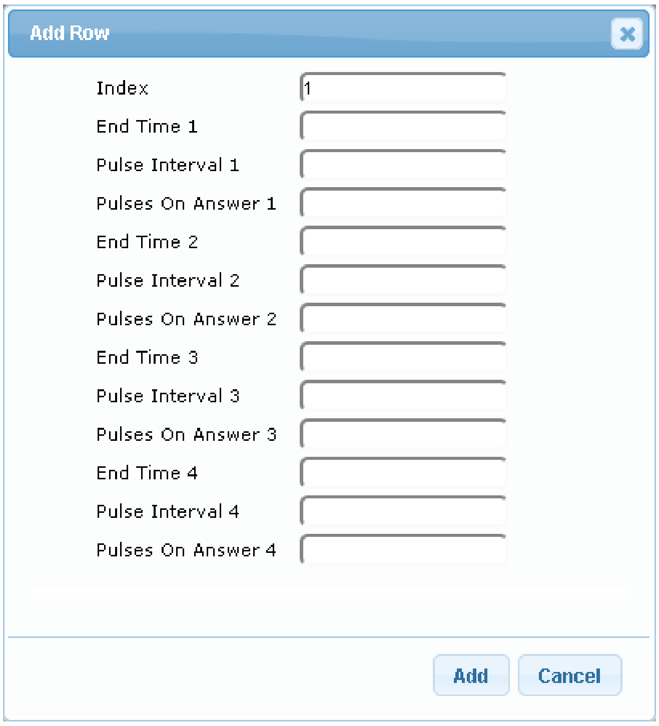
Note: The Charge Codes table is applicable only to Euro ISDN BRI interfaces.

The following procedure describes how to configure Charge Codes through the Web interface. You can also configure it through ini file (ChargeCode) or CLI (configure voip > gw analoggw charge-code).

➢ To configure a Charge Code:

1. Open the Charge Codes table (**Configuration** tab > **VoIP** menu > **Services** > **Least Cost Routing** > **Charge Codes**).
2. Click **Add**; the following dialog box appears:

Figure 26-2: Charge Codes Table - Add Row Dialog Box



Add Row

Index: 1
End Time 1:
Pulse Interval 1:
Pulses On Answer 1:
End Time 2:
Pulse Interval 2:
Pulses On Answer 2:
End Time 3:
Pulse Interval 3:
Pulses On Answer 3:
End Time 4:
Pulse Interval 4:
Pulses On Answer 4:

3. Configure a Charge Code according to the parameters described in the table below.
4. Click **Add**.

Table 26-5: Charge Codes Table Parameter Descriptions

Parameter	Description
Index [ChargeCode_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
End Time (1 - 4) `end-time-<1-4>` [ChargeCode_EndTime<1-4>]	Defines the end of the time period in a 24 hour format, *hh*. For example, "04" denotes 4 A.M. **Notes:** ▪ The first time period always starts at midnight (00). ▪ It is mandatory that the last time period of each rule end at midnight (00). This prevents undefined time frames in a day.
Pulse Interval (1 - 4) `pulse-interval-<1-4>` [ChargeCode_PulseInterval<1-4>]	Defines the time interval between pulses (in tenths of a second). Once the call is established, the device generates a pulse for each pulse interval.
Pulses On Answer (1 - 4) `pulses-on-answer-<1-4>` [ChargeCode_PulsesOnAnswer<1-4>]	Defines the number of pulses that the device generates upon call answer.

26.14 Converting Accented Characters from IP to Tel

The Char Conversion table lets you configure up to 40 Character Conversion rules. A Character Conversion rule maps (converts) accented characters (Unicode / UTF-8) received from the IP side into simple ASCII characters (ISO-8859) for sending to the Tel side. Typically, the device receives the caller ID and calling name in Unicode characters (in the SIP INVITE message). Unicode characters consist of two bytes, while ASCII characters consist of one byte. Accented characters are used in various languages such as German. An example of such a character is the umlaut (or diaeresis), which consists of two dots placed over a letter, as in ä. The importance of this conversion feature is that it allows PSTN entities that do not support accented characters, to receive ASCII characters. For example, the device can convert the Unicode character ä into the ASCII character "ae".

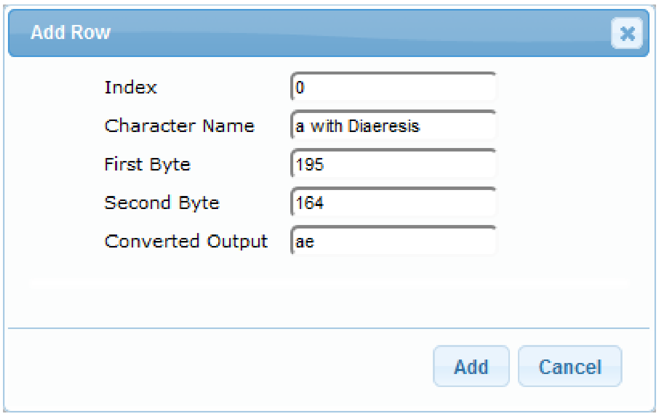
Note: The table works in conjunction with the ISO8859CharacterSet parameter. When the parameter is set to [0] (Latin only), it converts accented characters into ASCII (e.g., ä to "a"). However, the table can be used to overwrite these "basic" conversions and customize them (e.g., ä to "ae" instead of the default "a").

The following procedure describes how to configure Character Conversion rules through the Web interface. You can also configure it through ini file (CharConversion) or CLI (configure voip > gw dtmf-and-suppl dtmf-and-dialing > char-conversion).

➢ To configure a Character Conversion rule:

1. Open the Char Conversion table (**Configuration** tab > **VoIP** menu > **Gateway** > **DTMF & Supplementary** > **Char Conversion**).
2. Click **Add**; the following dialog box appears:

Figure 26-3: Char Conversion Table - Add Row Dialog Box



Add Row

Index: 0
Character Name: a with Diaeresis
First Byte: 195
Second Byte: 164
Converted Output: ae

The figure above shows a configuration example where ä is converted to ae.

3. Configure a Character Conversion rule according to the parameters described in the table below.
4. Click **Add**.

Table 26-6: Char Conversion Table Parameter Descriptions

Parameter	Description
Index [CharConversion_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Character Name `char-name`	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters.

Parameter	Description
[CharConversion_CharName]	**Note:** Each row must be configured with a unique name.
First Byte `first-byte` [CharConversion_FirstByte]	Defines the first byte of the Unicode character (e.g., 195). The default is 194.
Second Byte `second-byte` [CharConversion_SecondByte]	Defines the second byte of the Unicode character (e.g., 164). The default is 128.
Converted Output `converted-output` [CharConversion_ConvertedOutput]	Defines the ASCII character (e.g., "ae") to which the Unicode character must be converted. The valid value is a string of up to four characters. The valid value is up to four ASCII characters. This can include any ASCII character - alphanumerical (e.g., a, A, 6) and/or symbols (e.g., !, ?, _, &).

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Part VI

Session Border Controller Application

27 SBC Overview

This section provides an overview of the device's SBC application.

Notes:

- For guidelines on how to deploy your SBC device, refer to the *SBC Design Guide* document.
- The SBC feature is available only if the device is installed with a Software License Key that includes this feature. For installing a Software License Key, see "Software License Key" on page 668.
- For the maximum number of supported SBC sessions, and SBC users than can be registered in the device's registration database, see "Technical Specifications" on page 1035.

27.1 Feature List

The SBC application supports the following main features:

- NAT traversal: The device supports NAT traversal, allowing, for example, communication with ITSPs with globally unique IP addresses and with far-end users located behind NAT on the WAN. The device supports this by:
  - Continually registering far-end users with its users registration database.
  - Maintaining remote NAT binding state by frequent registrations and thereby, off-loading far-end registrations from the LAN IP PBX.
  - Using Symmetric RTP (RFC 4961) to overcome bearer NAT traversal.
- VoIP firewall and security for signaling and media:
  - SIP signaling:
    - Deep and stateful inspection of all SIP signaling packets.
    - SIP dialog initiations may be rejected based on values of incoming SIP INVITE message and other Layer-3 characteristics.
    - Packets not belonging to an authorized SIP dialog are discarded.
  - RTP:
    - Deep packet inspection of all RTP packets.
    - Late rogue detection - if a SIP session was gracefully terminated and someone tries to "ride on it" with rogue traffic from the already terminated RTP and SIP context, the VoIP Firewall prevents this from occurring.
    - Disconnects call (after user-defined time) if RTP connection is broken.
    - Black/White lists for both Layer-3 firewall and SIP classification.
- Stateful Proxy Operation Mode: The device can act as a Stateful Proxy by enabling SIP messages to traverse it transparently (with minimal interference) between the inbound and outbound legs.
- B2BUA and Topology Hiding: The device intrinsically supports topology hiding, limiting the amount of topology information displayed to external parties. For example, IP addresses of ITSPs' equipment (e.g. proxies, gateways, and application servers) can be hidden from outside parties. The device's topology hiding is provided by implementing back-to-back user agent (B2BUA) leg routing:
  - Strips all incoming SIP Via header fields and creates a new Via value for the outgoing message.
  - Each leg has its own Route/Record Route set.
  - User-defined manipulation of SIP To, From, and Request-URI host names.

- Generates a new SIP Call-ID header value (different between legs).
- Changes the SIP Contact header and sets it to the device's address.
- Layer-3 topology hiding by modifying source IP address in the SIP IP header.

- SIP normalization: The device supports SIP normalization, whereby the SBC application can overcome interoperability problems between SIP user agents. This is achieved by the following:
    - Manipulation of SIP URI user and host parts.
    - Connection to ITSP SIP trunks on behalf of an IP-PBX - the device can register and utilize user and password to authenticate for the IP-PBX.
- Survivability:
    - Routing calls to alternative routes such as the PSTN.
    - Routing calls between user agents in the local network using a dynamic database (built according to registrations of SIP user agents).
- Routing:
    - IP-to-IP routing translations of SIP, UDP, TCP, TLS (when extensive transcoding is not required).
    - Load balancing and redundancy of SIP servers.
    - Routing according to Request-URI\Specific IP address\Proxy\FQDN.
    - Alternative routing.
    - Routing between different Layer-3 networks (e.g., LAN and WAN).
- Load balancing\redundancy of SIP servers.
- ITSP accounts.
- SIP URI user and host name manipulations.

27.2 B2BUA and Stateful Proxy Operating Modes

The device can operate in one or both of the following SBC modes:

- **Back-to-Back User Agent (B2BUA):** Maintains independent sessions toward the endpoints, processing an incoming request as a user agent server (UAS) on the inbound leg, and processing the outgoing request as a user agent client (UAC) on the outbound leg. SIP messages are modified regarding headers between the legs and all the device's interworking features may be applied.
- **Stateful Proxy Server:** SIP messages traverse the device transparently (with minimal interference) between the inbound and outbound legs, for connecting SIP endpoints.

By default, the device's B2BUA mode changes SIP dialog identifiers and topology data in SIP messages traversing through it:

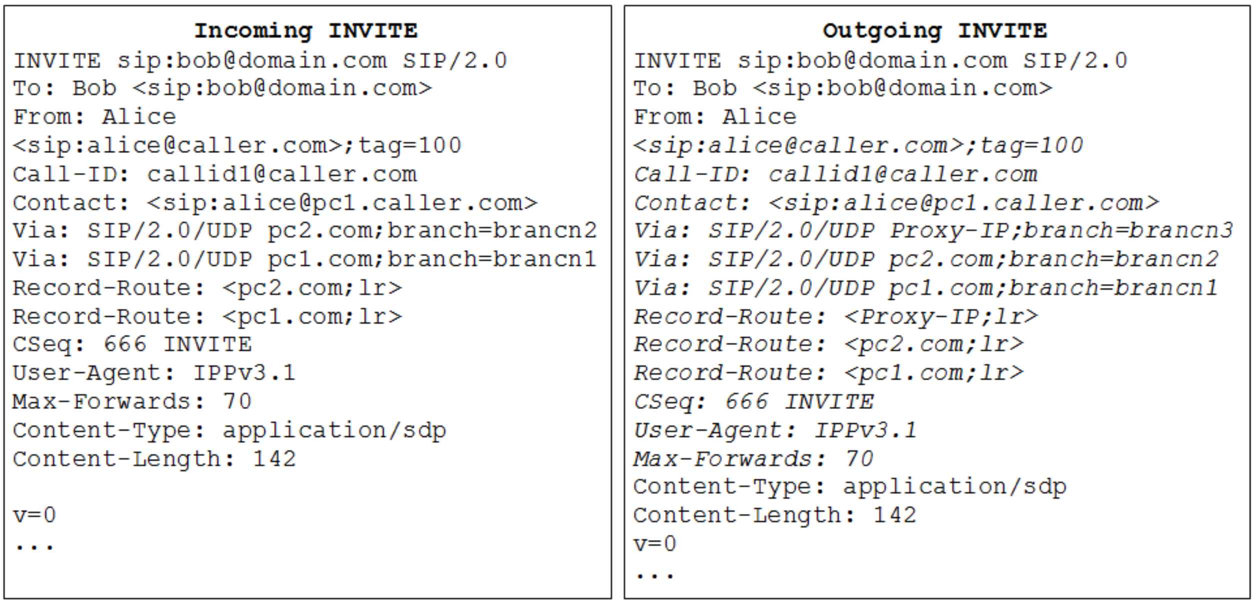
- Call identifiers: Replaces the From-header tag and Call-ID header so that they are different for each leg (inbound and outbound).
- Routing headers:
    - Removes all Via headers in incoming requests and sends the outgoing message with its own Via header.
    - Doesn't forward any Record-Route headers from the inbound to outbound leg, and vice versa.
    - Replaces the address of the Contact header in the incoming message with its own address in the outgoing message.
- Replaces the User-Agent/ Server header value in the outgoing message, and replaces the original value with itself in the incoming message.

In contrast, when the device operates in Stateful Proxy mode, the device by default forwards SIP messages transparently (unchanged) between SIP endpoints (from inbound to outbound legs). The device retains the SIP dialog identifiers and topology headers received in the incoming message and sends them as is in the outgoing message. The device handles the above mentioned headers transparently (i.e., they remain unchanged) or according to configuration (enabling partial transparency), and only adds itself as the top-most Via header and optionally, to the Record-Route list. For configuring the handling of these headers for partial transparency, use the following IP Profile parameters (see "Configuring IP Profiles" on page 387):

- IpProfile_SBCRemoteRepresentationMode: Contact and Record-Route headers
- IpProfile_SBCKeepVIAHeaders: Via headers
- IpProfile_SBCKeepUserAgentHeader: User-Agent headers
- IpProfile_SBCKeepRoutingHeaders: Record-Route headers
- IpProfile_SBCRemoteMultipleEarlyDialogs: To-header tags

Thus, the Stateful Proxy mode provides full SIP transparency (no topology hiding) or asymmetric topology hiding. Below is an example of a SIP dialog-initiating request when operating in Stateful Proxy mode for full transparency, showing all the incoming SIP headers retained in the outgoing INVITE message.

Figure 27-1: Example of SIP Message Handling in Stateful Proxy Mode



```
Incoming INVITE
INVITE sip:bob@domain.com SIP/2.0
To: Bob <sip:bob@domain.com>
From: Alice
<sip:alice@caller.com>;tag=100
Call-ID: callid1@caller.com
Contact: <sip:alice@pc1.caller.com>
Via: SIP/2.0/UDP pc2.com;branch=brancn2
Via: SIP/2.0/UDP pc1.com;branch=brancn1
Record-Route: <pc2.com;lr>
Record-Route: <pc1.com;lr>
CSeq: 666 INVITE
User-Agent: IPPv3.1
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 142

v=0
...
```

```
Outgoing INVITE
INVITE sip:bob@domain.com SIP/2.0
To: Bob <sip:bob@domain.com>
From: Alice
<sip:alice@caller.com>;tag=100
Call-ID: callid1@caller.com
Contact: <sip:alice@pc1.caller.com>
Via: SIP/2.0/UDP Proxy-IP;branch=brancn3
Via: SIP/2.0/UDP pc2.com;branch=brancn2
Via: SIP/2.0/UDP pc1.com;branch=brancn1
Record-Route: <Proxy-IP;lr>
Record-Route: <pc2.com;lr>
Record-Route: <pc1.com;lr>
CSeq: 666 INVITE
User-Agent: IPPv3.1
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 142
v=0
...
```

Some of the reasons for implementing Stateful Proxy mode include:

- B2BUA typically hides certain SIP headers for topology hiding. In specific setups, some SIP servers require the inclusion of these headers to know the history of the SIP request. In such setups, the requirement may be asymmetric topology hiding, whereby SIP traffic toward the SIP server must expose these headers whereas SIP traffic toward the users must not expose these headers.
- B2BUA changes the call identifiers between the inbound and outbound SBC legs and therefore, call parties may indicate call identifiers that are not relayed to the other leg. Some SIP functionalities are achieved by conveying the SIP call identifiers either in SIP specific headers (e.g., Replaces) or in the message bodies (e.g. Dialog Info in an XML body).
- In some setups, the SIP client authenticates using a hash that is performed on one or more of the headers that B2BUA changes (removes). Therefore, implementing B2BUA would cause authentication to fail.
- For facilitating debugging procedures, some administrators require that the value in the Call-ID header remains unchanged between the inbound and outbound SBC legs.

As B2BUA changes the Call-ID header, such debugging requirements would fail.

The operating mode can be configured per the following configuration entities:

- SRDs in the SRD table (see "Configuring SRDs" on page 325)
- IP Groups in the IP Group table (see "Configuring IP Groups" on page 340)

If the operation mode is configured in both tables, the operation mode of the IP Group is applied. Once configured, the device uses default settings in the IP Profile table for handling the SIP headers, as mentioned previously. However, you can change the default settings to enable partial transparency.

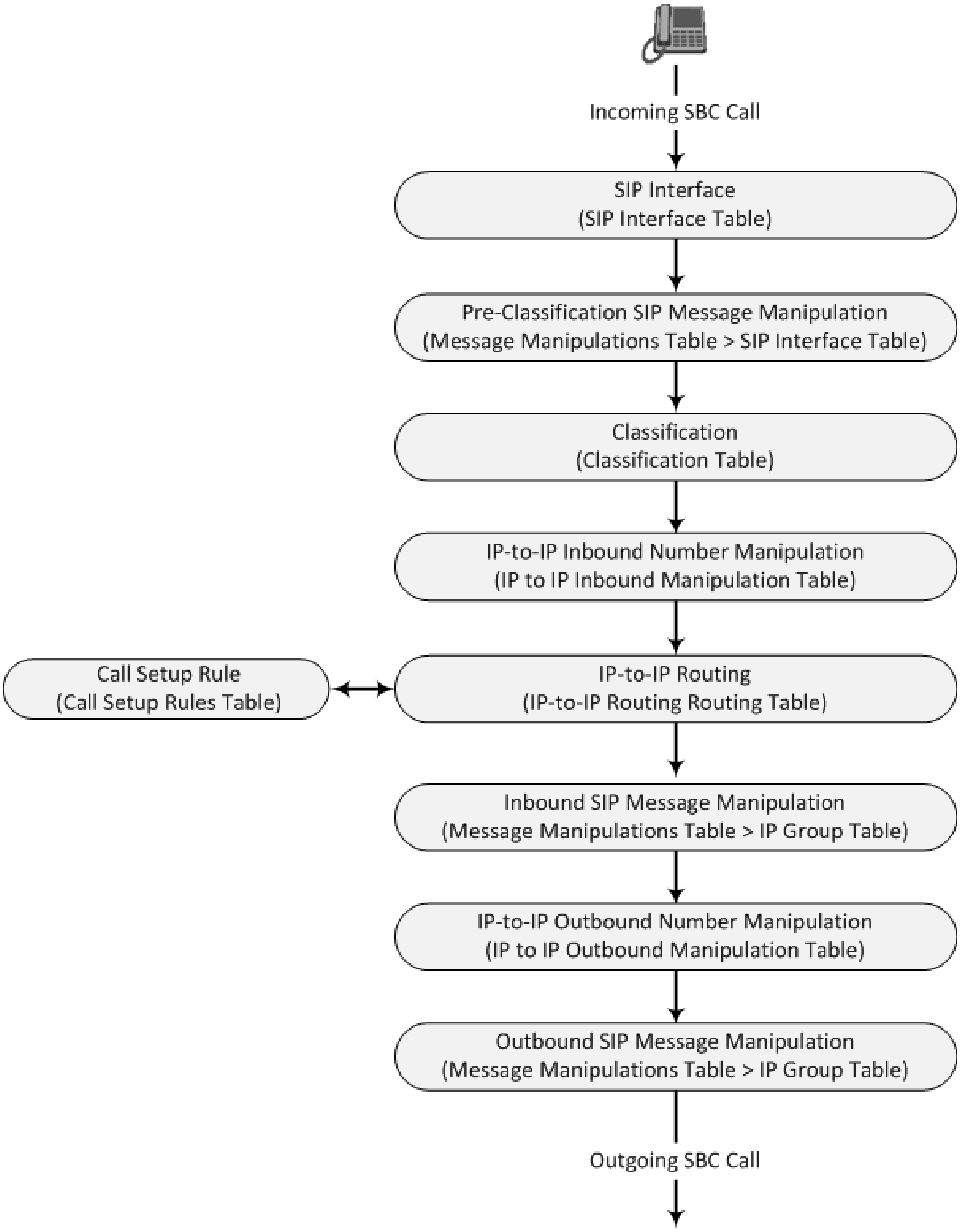
Notes:

- The To-header tag remains the same for inbound and outbound legs of the dialog, regardless of operation mode.
- If the Operation Mode of the SRD\IP Group of one leg of the dialog is set to 'Call Stateful Proxy', the device also operates in this mode on the other leg with regards to the dialog identifiers (Call-ID header, tags, CSeq header).
- It is recommended to implement the B2BUA mode, unless one of the reasons mentioned previously is required. B2BUA supports all the device's feature-rich offerings, while Stateful Proxy may offer only limited support. The following features are not supported when in Stateful Proxy mode:
  - √ Alternative routing
  - √ Call forking
  - √ Terminating REFER/3xx
- If Stateful Proxy mode is enabled and any one of the unsupported features is enabled, the device disables the Stateful Proxy mode and operates in B2BUA mode.
- You can configure the device to operate in both B2BUA and Stateful Proxy modes for the same users. This is typically implemented when users need to communicate with different SIP entities (IP Groups). For example, B2BUA mode for calls destined to a SIP Trunk and Stateful Proxy mode for calls destined to an IP PBX. The configuration is done using IP Groups and SRDs.
- If Stateful Proxy mode is used only due to the debugging benefits, it is recommended to configure the device to only forward the Call-ID header unchanged.

27.3 Call Processing of SIP Dialog Requests

The device processes incoming SIP dialog requests (SIP methods) such as INVITE, SUBSCRIBE, OPTIONS, REFER, INFO, UNSOLICITED NOTIFY, MESSAGE, and REGISTER. The process is summarized in the following figure and subsequently described:

Figure 27-2: SBC Call Processing


Incoming SBC Call
SIP Interface
(SIP Interface Table)
Pre-Classification SIP Message Manipulation
(Message Manipulations Table > SIP Interface Table)
Classification
(Classification Table)
IP-to-IP Inbound Number Manipulation
(IP to IP Inbound Manipulation Table)
Call Setup Rule
(Call Setup Rules Table)
IP-to-IP Routing
(IP-to-IP Routing Routing Table)
Inbound SIP Message Manipulation
(Message Manipulations Table > IP Group Table)
IP-to-IP Outbound Number Manipulation
(IP to IP Outbound Manipulation Table)
Outbound SIP Message Manipulation
(Message Manipulations Table > IP Group Table)
Outgoing SBC Call

The SIP dialog-initiating process consists of the following stages:

1. **Determining Source and Destination URL:** The SIP protocol has more than one URL in a dialog-initiating requests that may represent the source and destination URLs. The device obtains the source and destination URLs from certain SIP headers. Once the URLs are determined, the user and host parts of the URLs can be used as matching rule characteristics for classification, message manipulation, and call routing.

   - **All SIP requests (e.g., INVITE) except REGISTER:**

     - Source URL: Obtained from the From header. If the From header contains the value 'Anonymous', the source URL is obtained from the P-Preferred-Identity header. If the P-Preferred-Identity header does not exist, the source URL is obtained from the P-Asserted-Identity header.

    - Destination URL: Obtained from the Request-URI.
- **REGISTER dialogs:**
    - Source URL: Obtained from the To header.
    - Destination URL: Obtained from the Request-URI.

> **Note:** You can specify the SIP header from where you want the device to obtain the source URL in the incoming dialog request. This is configured in the IP Group table using the 'Source URI Input' parameter (see "Configuring IP Groups" on page 340).

2. **Determining SIP Interface:** The device checks the SIP Interface on which the SIP dialog is received. The SIP Interface defines the local SIP "listening" port and IP network interface. For more information, see "Configuring SIP Interfaces" on page 333.
3. **Applying SIP Message Manipulation:** Depending on configuration, the device can apply a SIP message manipulation rule (assigned to the SIP Interface) on the incoming SIP message. A SIP Message Manipulation rule defines a matching characteristics (*condition*) of the incoming SIP message and the corresponding manipulation operation (e.g., remove the P-Asserted-Identity header), which can apply to almost any aspect of the message (add, remove or modify SIP headers and parameters). For more information, see "Configuring SIP Message Manipulation" on page 370.
4. **Classifying to an IP Group:** Classification identifies the incoming SIP dialog request as belonging to a specific IP Group (i.e., from where the SIP dialog request originated). The classification process is based on the SRD to which the dialog belongs (the SRD is determined according to the SIP Interface). For more information, see "Configuring Classification Rules" on page 569.
5. **Applying IP-to-IP Inbound Manipulation:** Depending on configuration, the device can apply an IP-to-IP Inbound Manipulation rule to the incoming dialog. This manipulates the user part of the SIP URI for source (e.g., in the SIP From header) and destination (e.g., in the Request-URI line). The manipulation rule is associated with the incoming dialog, by configuring the rule with incoming matching characteristics such as source IP Group and destination host name. The manipulation rules are also assigned an SBC Routing Policy, which in turn, is assigned to IP-to-IP routing rules. As most deployments require only one SBC Routing Policy, the default Routing Policy is automatically assigned to manipulation and routing rules. For more information, see "Configuring IP-to-IP Inbound Manipulations" on page 597.
6. **SBC IP-to-IP Routing:** The device searches the IP-to-IP Routing table for a routing rule that matches the characteristics of the incoming call. If found, the device routes the call to the configured destination which can be, for example, an IP Group, the Request-URI if the user is registered with the device, and a specified IP address. For more information, see "Configuring SBC IP-to-IP Routing Rules" on page 578.
7. **Applying Inbound SIP Message Manipulation:** Depending on configuration, the device can apply a SIP message manipulation rule (assigned to the IP Group) on the incoming dialog. For more information, see Stage 3.
8. **Applying IP-to-IP Outbound Manipulation:** Depending on configuration, the device can apply an IP-to-IP Outbound Manipulation rule to the outbound dialog. This manipulates the user part of the Request-URI for source (e.g., in the SIP From header) or destination (e.g., in the SIP To header) or calling name in the outbound SIP dialog. The manipulation rule is associated with the dialog, by configuring the rule with incoming matching characteristics such as source IP Group and destination host name. The manipulation rules are also assigned an SBC Routing Policy, which in turn, is assigned to IP-to-IP routing rules. As most deployments require only one SBC Routing Policy, the default Routing Policy is automatically assigned to manipulation rules and routing rules. For more information, see "Configuring IP-to-IP Outbound Manipulations" on page 601.

9. **Applying Outbound SIP Message Manipulation:** Depending on configuration, the device can apply a SIP message manipulation rule (assigned to the IP Group) on the outbound dialog. For more information, see Stage 3.
10. The call is sent to the configured destination.

27.4 User Registration

The device provides a registration database for registering users. Only users belonging to a User-type IP Group can register with the device. User-type IP Groups represent a group of SIP user agents that share the following characteristics:

- Perform registrations and share the same serving proxy\registrar
- Same SIP and media behavior
- Same IP Profile
- Same SIP handling configuration
- Same Call Admission Control (CAC)

Typically, the device is configured as the user's outbound proxy, routing requests (using the IP-to-IP Routing table) from the user's User-type IP Group to the serving proxy, and vice versa. Survivability can be achieved using the alternative routing feature.

The device forwards registration requests (REGISTER messages) from a Server-type IP Group, but does not save the registration binding in its' registration database.

27.4.1 Initial Registration Request Processing

A summary of the device's handling of registration requests (REGISTER messages) is as follows:

- The URL in the To header of the REGISTER message constitutes the primary Address of Record (AOR) for registration (according to standard). The device can save other AORs in its registration database as well. When the device searches for a user in its' registration database, any of the user’s AORs can result in a match.
- The device's Classification process for initial REGISTER messages is slightly different than for other SIP messages. Unlike other requests, initial REGISTER requests can’t be classified according to the registration database.
- If registration succeeds (replied with 200 OK by the destination server), the device adds a record to its' registration database, which identifies the specific contact of the specific user (AOR). The device uses this record to route subsequent SIP requests to the specific user (in normal or survivability modes).
- The records in the device's registration database include the Contact header. The device adds every REGISTER request to the registration database before manipulation, allowing correct user identification in the Classification process for the next received request.
- You can configure Call Admission Control (CAC) rules for incoming and outgoing REGISTER messages. For example, you can limit REGISTER requests from a specific IP Group or SRD. Note that this applies only to concurrent REGISTER dialogs and not concurrent registrations in the device's registration database.

The device provides a dynamic registration database that it updates according to registration requests traversing it. Each database entry for a user represents a binding between an AOR (obtained from the SIP To header), optional additional AORs, and one or more contacts (obtained from the SIP Contact headers). Database bindings are added upon successful registration responses from the proxy server (SIP 200 OK). The device removes database bindings in the following cases:

- Successful de-registration responses (REGISTER with Expires header that equals zero).

- Registration failure responses.
- Timeout of the Expires header value (in scenarios where the UA did not send a refresh registration request).

Notes:

- The same contact cannot belong to more than one AOR.
- Contacts with identical URIs and different ports and transport types are not supported (same key is created).
- Multiple contacts in a single REGISTER message is not supported.
- One database is shared between all User-type IP Groups.

27.4.2 Classification and Routing of Registered Users

The device can classify incoming SIP dialog requests (e.g., INVITE) from registered users to an IP Group, by searching for the sender's details in the registration database. The device uses the AOR from the From header and the URL in the Contact header of the request to locate a matching registration binding. The found registration binding contains information regarding the registered user, including the IP Group to which it belongs. (Upon initial registration, the Classification table is used to classify the user to a User-type IP Group and this information is then added with the user in the registration database.)

The destination of a dialog request can be a registered user and the device thus uses its registration database to route the call. This can be achieved by various ways such as configuring a rule in the IP-to-IP Routing table where the destination is a User-type IP Group or any matching user registered in the database ('Destination Type' is configured to **All Users**). The device searches the registration database for a user that matches the incoming Request-URI (listed in chronological order):

1. Unique Contact generated by the device and sent in the initial registration request to the serving proxy.
2. AOR. The AOR is originally obtained from the incoming REGISTER request and must either match both user part and host part of the Request-URI, or only user part.
3. Contact. The Contact is originally obtained from the incoming REGISTER request.

If registrations are destined to the database (using the above rules), the device does not attempt to find a database match, but instead replies with a SIP 200 OK (used for Survivability). Once a match is found, the request is routed either to the contact received in the initial registration or (if the device identifies that the user agent is behind a NAT) to the source IP address of the initial registration.

You can configure (using the SBCDBRoutingSearchMode parameter) for which part of the destination Request-URI in the INVITE message the device must search in the registration database:

- Only by entire Request-URI (user@host), for example, "4709@joe.company.com".
- By entire Request-URI, but if not found, by the user part of the Request-URI, for example, "4709".

When an incoming INVITE is received for routing to a user and the user is located in the registration database, the device sends the call to the user's corresponding contact address specified in the database.

Note: If the Request-URI contains the "tel:" URI or "user=phone" parameter, the device searches only for the user part.

27.4.3 General Registration Request Processing

The device's general handling of registration requests (REGISTER messages) for unregistered users is as follows:

- The device routes REGISTER requests according to the IP-to-IP Routing table. If the destination is a User-type IP Group, the device does not forward the registration; instead, it accepts (replies with a SIP 200 OK response) or rejects (replies with a SIP 4xx) the request according to the user's IP Group configuration.
- Alternative routing can be configured for REGISTER requests, in the IP-to-IP Routing table.
- By default, the Expires header has the same value in incoming and outgoing REGISTER messages. However, you can modify the Expires value using the following parameters: SBCUserRegistrationTime, SBCProxyRegistrationTime, SBCRandomizeExpires, and SBCSurvivabilityRegistrationTime. You can also modify the Expires value of REGISTER requests received from users located behind NAT, using the IP Profile parameters IpProfile_SBCUserBehindUdpNATRegistrationTime and IpProfile_SBCUserBehindTcpNATRegistrationTime.
- By default, the Contact header in outgoing REGISTER message is different than the Contact header in the incoming REGISTER. The user part of the Contact is populated with a unique contact generated by the device and associated with the specific registration. The IP address in the host part is changed to the address of the device. Alternatively, the original user can be retained in the Contact header and used in the outgoing REGISTER request (using the SBCKeepContactUserinRegister parameter).

27.4.4 Registration Refreshes

Registration refreshes are incoming REGISTER requests from users that are registered in the device's registration database. The device sends these refreshes to the serving proxy only if the serving proxy's Expires time is about to expire; otherwise, the device responds with a 200 OK to the user, without routing the REGISTER. Each such refreshes also refresh the internal timer set on the device for this specific registration.

The device automatically notifies SIP proxy / registrar servers of users that are registered in its registration database and whose registration timeout has expired. When a user's registration timer expires, the device removes the user's record from the database and sends an un-register notification (REGISTER message with the Expires header set to 0) to the proxy/registrar. This occurs only if a REGISTER message is sent to an IP Group destination type (in the IP-to-IP Routing table).

You can also apply a graceful period to unregistered requests, using the 'User Registration Grace Time' parameter (SBCUserRegistrationGraceTime):

- You can configure the device to add extra time (grace period) to the expiration timer of registered users in the database. If you configure this grace period, the device keeps the user in the database (and does not send an un-register to the registrar server), allowing the user to send a "late" re-registration to the device. The device removes the user from the database only when this additional time expires.
- The graceful period is also used before removing a user from the registration database when the device receives a successful unregister response (200 OK) from the registrar/proxy server. This is useful in scenarios, for example, in which users (SIP user agents) such as IP Phones erroneously send unregister requests. Instead of immediately removing the user from the registration database upon receipt of a successful unregister response, the device waits until it receives a successful unregister response from the registrar server, waits the user-defined graceful time and if no register refresh request is received from the user agent, removes the contact (or AOR) from the database.

The device keeps registered users in its' registration database even if connectivity with the proxy is lost (i.e., proxy does not respond to users' registration refresh requests). The device removes users from the database only when their registration expiry time is reached (with the additional grace period, if configured).

27.4.5 Registration Restriction Control

The device provides flexibility in controlling user registrations:

- **Limiting Number of Registrations:** You can limit the number of users that can register with the device per IP Group, SIP Interface, and/or SRD, in the IP Group, SIP Interface and SRD tables respectively. By default, no limitation exists.
- **Blocking Incoming Calls from Unregistered Users:** You can block incoming calls (INVITE requests) from unregistered users belonging to User-type IP Groups. By default, calls from unregistered users are not blocked. This is configured per SIP Interface or SRD. When the call is rejected, the device sends a SIP 500 (Server Internal Error) response to the remote end.

27.4.6 Deleting Registered Users

You can remove registered users from the device's registration database through CLI:

- To delete a specific registered user:

  ```
  # clear voip register db sbc user <AOR of user – user part or
  user@host>
  ```

  For example:

  ```
  # clear voip register db sbc user John@10.33.2.22
  # clear voip register db sbc user John
  ```

- To delete all registered users belonging to a specific IP Group:

  ```
  # clear voip register db sbc ip-group <ID or name>
  ```

27.5 Media Handling

Media behavior includes anything related to the establishment, management and termination of media sessions within the SIP protocol. Media sessions are created using the SIP offer-answer mechanism. If successful, the result is a bi-directional media (RTP) flow (e.g. audio, fax, modem, DTMF). Each offer-answer may create multiple media sessions of different types (e.g. audio and fax). In a SIP dialog, multiple offer-answer transactions may occur and each may change the media session characteristics (e.g. IP address, port, coders, media types, and RTP mode). The media capabilities exchanged in an offer-answer transaction include the following:

- Media types (e.g., audio, secure audio, video, fax, and text)
- IP addresses and ports of the media flow
- Media flow mode (send receive, receive only, send only, inactive)
- Media coders (coders and their characteristics used in each media flow)
- Other (standard or proprietary) media and session characteristics

Typically, the device does not change the negotiated media capabilities (mainly performed by the remote user agents). However, it does examine and may take an active role in the SDP offer-answer mechanism. This is done mainly to anchor the media to the device (default) and also to change the negotiated media type, if configured. Some of the media handling features, which are described later in this section, include the following:

- Media anchoring (default)
- Direct media
- Audio coders restrictions
- Early media and ringback tone handling

- Call hold translations and held tone generation
- NAT traversal
- RTP broken connections
- Media firewall
    - RTP pin holes - only RTP packets related to a successful offer-answer negotiation traverse the device: When the device initializes, there are no RTP pin holes opened. This means that each RTP\RTCP packets destined to the device are discarded. Once an offer-answer transaction ends successfully, an RTP pin hole is opened and RTP\RTCP flows between the two remote user agents. Once a pin hole is opened, the payload type and RTP header version is validated for each packet. RTP pin holes close if one of the associated SIP dialogs is closed (may also be due to broken connection).
    - Late rogue detection - once a dialog is disconnected, the related pin holes also disconnect.
    - Deep Packet inspection of the RTP that flows through the opened pin holes.

27.5.1 Media Anchoring

By default, the device anchors the media (RTP) traffic. In other words, the media between SIP endpoints traverses the device. You can change this default mode by enabling direct media between SIP endpoints. Media anchoring may be required, for example, to resolve NAT problems, enforce media security policies, perform media transcoding, and media monitoring.

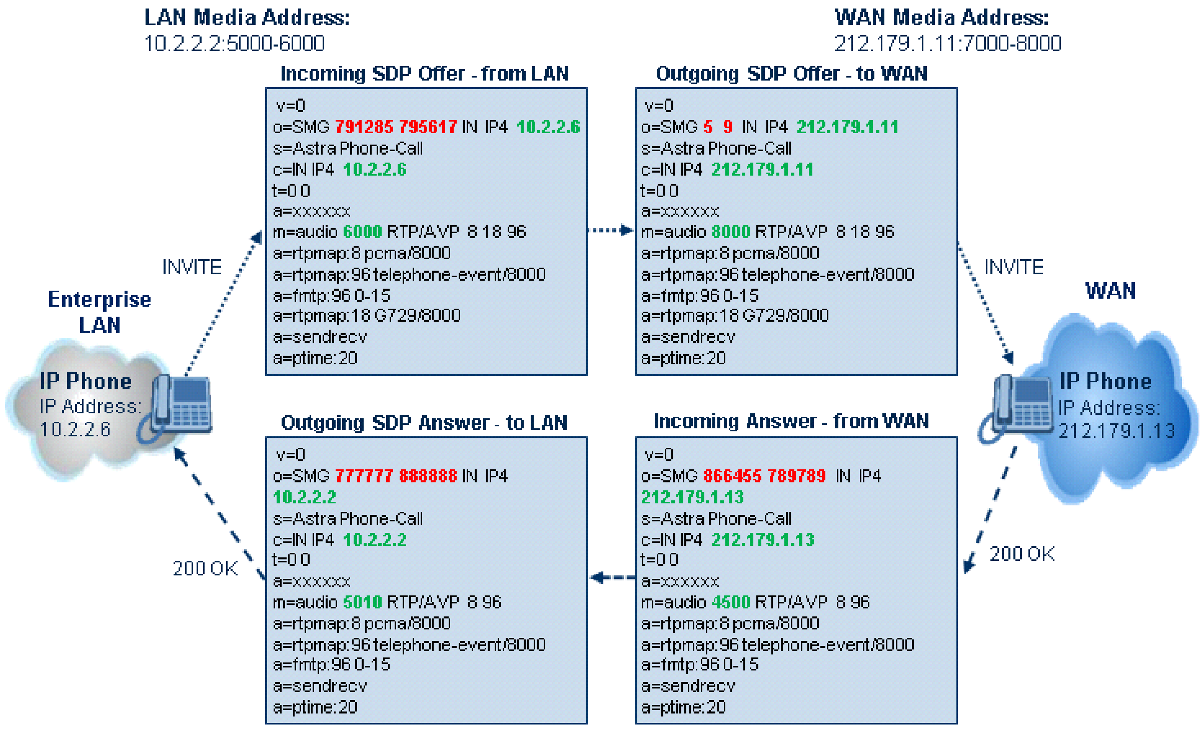
To enforce RTP traffic to flow through the device, the device modifies all IP address fields in the SDP:

- Origin: IP address, session and version id
- Session connection attribute ('c=' field)
- Media connection attribute ('c=' field)
- Media port number
- RTCP media attribute IP address and port

The device uses different local ports (e.g., for RTP, RTCP and fax) for each leg (inbound and outbound). The local ports are allocated from the Media Realm associated with each leg. The Media Realm assigned to the leg's IP Group (in the IP Group table) is used. If not assigned to the IP Group, the Media Realm assigned to the leg's SIP Interface (in the SIP

Interface table) is used. The following figure provides an example of SDP handling for a call between a LAN IP Phone 10.2.2.6 and a remote IP Phone 212.179.1.13 on the WAN.

Figure 27-3: SDP Offer/Answer Example



Incoming SDP Offer - from LAN

```
v=0
o=SMG 791285 795617 IN IP4 10.2.2.6
s=Astra Phone-Call
c=IN IP4 10.2.2.6
t=0 0
a=xxxxxx
m=audio 6000 RTP/AVP 8 18 96
a=rtpmap:8 pcma/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=rtpmap:18 G729/8000
a=sendrecv
a=ptime:20
```

Outgoing SDP Offer - to WAN

```
v=0
o=SMG 5 9 IN IP4 212.179.1.11
s=Astra Phone-Call
c=IN IP4 212.179.1.11
t=0 0
a=xxxxxx
m=audio 8000 RTP/AVP 8 18 96
a=rtpmap:8 pcma/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=rtpmap:18 G729/8000
a=sendrecv
a=ptime:20
```

Outgoing SDP Answer - to LAN

```
v=0
o=SMG 777777 888888 IN IP4
10.2.2.2
s=Astra Phone-Call
c=IN IP4 10.2.2.2
t=0 0
a=xxxxxx
m=audio 5010 RTP/AVP 8 96
a=rtpmap:8 pcma/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=sendrecv
a=ptime:20
```

Incoming Answer - from WAN

```
v=0
o=SMG 866455 789789 IN IP4
212.179.1.13
s=Astra Phone-Call
c=IN IP4 212.179.1.13
t=0 0
a=xxxxxx
m=audio 4500 RTP/AVP 8 96
a=rtpmap:8 pcma/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=sendrecv
a=ptime:20
```

27.5.2 Direct Media

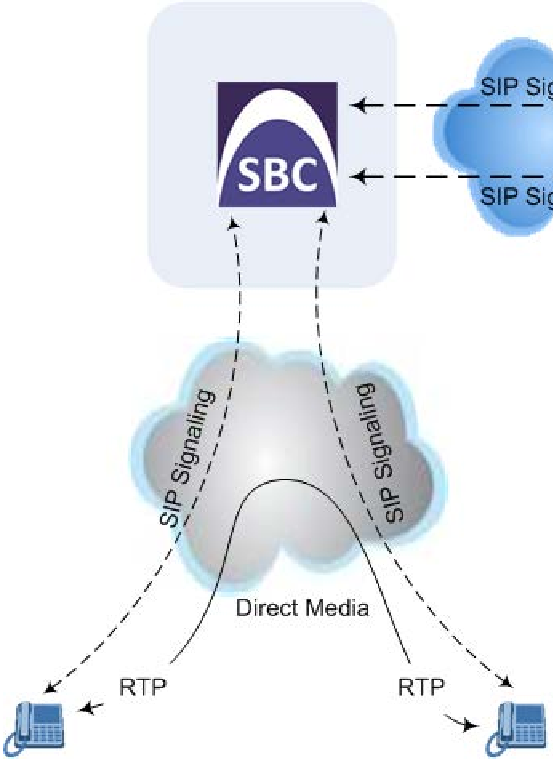
You can configure the device to allow the media (RTP/SRTP) session to flow directly between the SIP endpoints, without traversing the device. This is referred to as No Media Anchoring (also known as Anti-Tromboning or Direct Media). SIP signaling continues to traverse the device, with minimal intermediation and involvement, to enable certain SBC capabilities such as routing. By default, the device employs media anchoring, whereby the media session traverses the device, as described in "Media Anchoring" on page 537.

Direct media offers the following benefits:

- Saves network bandwidth
- Reduces the device's CPU usage (as there is no media handling)
- Avoids interference in SDP negotiation and header manipulation on RTP/SRTP

Direct media is typically implemented for calls between users located in the same LAN or domain, and where NAT traversal is not required and other media handling features such as media transcoding is not required. The following figure provides an example of direct media between LAN IP phones, while SIP signaling continues to traverse the device between LAN IP phones and the hosted WAN IP-PBX.

Figure 27-4: Direct Media Example



➢ To enable direct media:

- **For all calls:** Use the global parameter, SBCDirectMedia (overrides all other direct media configuration).
- **For specific calls:**
  - SIP Interface: You can enable direct media per SIP Interface (in the SIP Interface table), whereby calls (source and destination) associated with **this same** SIP Interface are handled as direct media calls. The SIP Interface can also enable direct media for users located behind the same NAT. For more information, see "Configuring SIP Interfaces" on page 333.
  - Direct Media Tag: You can enable direct media between users that are configured with the same Direct Media tag value. The tag is configured using the IP Profile table's IPProfile_SBCDirectMediaTag parameter (see "Configuring IP Profiles" on page 387).

The device employs direct media between endpoints under the following configuration conditions (listed in chronological order):

1. Direct media is enabled by the global parameter (SBCDirectMedia).
2. IP Groups of the endpoints are associated with IP Profiles whose 'Direct Media Tag' parameter has the same value (non-empty value).
3. IP Groups of the endpoints have the 'SBC Operation Mode' parameter set to Microsoft Server (direct media is required in the Lync environment). For more information, see "Configuring IP Groups" on page 340.
4. IP Groups of the endpoints use the same SIP Interface and the SIP Interface's 'SBC Direct Media' parameter is set to **Enable** (SIPInterface_SBCDirectMedia = 1).
5. IP Groups of the endpoints use the same SIP Interface and the SIP Interface's 'SBC Direct Media' parameter is set to Enable When Single NAT (SIPInterface_SBCDirectMedia = 2), and the endpoints are located behind the same

NAT.

Notes:

- If you enable direct media by the SBCDirectMedia parameter, direct media is applied to all calls even if direct media is disabled per SIP Interface.
- If you configure direct media for all calls (using the SBCDirectMedia parameter), the device does not open voice channels nor allocate media ports for the calls, as the media always bypasses the device. In contrast, if you configure direct media for specific calls, the device allocates ports for these calls. The reason is that the ports may be required for mid-call services (e.g., early media, call forwarding, call transfer, and playing on-hold tones) handled by the server (IP PBX), which traverse the device. Therefore, make sure that you have allocated sufficient media ports (Media Realm) for such calls.
- Direct media cannot operate with the following features:
  - Manipulation of SDP data (offer-answer transaction) such as ports, IP address, coders
  - Extension of RFC 2833 / out-of-band DTMF / in-band DTMF
  - Extension of SRTP/RTP
- All restriction features (Allowed Coders, restrict SRTP/RTP, restrict RFC 2833) can operate with direct media. Restricted coders are removed from the SDP offer message.
- For two users belonging to the same SIP Interface that is enabled for direct media and one of the users is defined as a foreign user (example, “follow me service”) located in the WAN while the other is located in the LAN: calls between these two users cannot be established until direct media is disabled for the SIP Interface. The reason for this is that the device does not interfere in the SIP signaling. In other words, parameters such as IP addresses are not manipulated for calls between LAN and WAN (although required).

27.5.3 Restricting Audio Coders

You can configure a list of permitted (allowed) voice coders that can be used for a specific SIP entity (leg). In other words, you can enforce the use of specific coders. If the SDP offer in the incoming SIP message does not contain any coder that is configured as an allowed coder, the device rejects the calls. If the SDP offer contains some coders that are configured as allowed coders, the device manipulates the SDP offer by removing the coders that are not configured as allowed coders, before routing the SIP message to its destination. The device also re-orders (prioritizes) the coder list in the SDP according to the listed order of configured allowed coders.

For example, assume the following:

- The SDP offer in the incoming SIP message contains the G.729, G.711, and G.723 coders.
- The allowed coders configured for the SIP entity include G.711 and G.729.

The device removes the G.723 coder from the SDP offer, re-orders the coder list so that G.711 is listed first, and sends the SIP message containing only the G.711 and G.729 coders in the SDP.

The allowed coders are configured in the Allowed Audio Coders Group table. For more information, see "Configuring Allowed Audio Coder Groups" on page 565.

27.5.4 Prioritizing Coder List in SDP Offer

In addition to restricting the use of coders using Allowed Coders Groups (see "Configuring Allowed Audio Coder Groups" on page 565), you can also prioritize the coders listed in the SDP offer. This feature is referred to as *Coder Preference* and applies to both SBC legs:

- **Incoming SDP offer:** The device arranges the coder list in the incoming SDP offer according to the order of appearance of the Allowed Coders Group that is associated with the incoming dialog. The coders listed higher up in the group take preference over ones listed lower down. To configure this, configure the 'Allowed Coders Mode' parameter (IpProfile_SBCAllowedCodersMode) in the associated IP Profile to **Preference** or **Restriction and Preference**. If you configure the parameter to to **Preference**, the coders in the SDP offer that also appear in the Allowed Coders Group are listed first in the SDP offer, and the coders in the SDP offer that do not appear in the Allowed Coders Group are listed after the Allowed coders in the SDP offer. Therefore, this setting does not restrict coder use to Allowed coders, but uses (prefers) the Allowed coders whenever possible.
- **Outgoing SDP offer:** If only Allowed coders are used, the device arranges the coders in the SDP offer as described above.

27.5.5 Multiple RTP Media Streams per Call Session

The device's SBC application supports multiple RTP media streams per SBC call session. Up to five different media types can be included in a session:

- Audio (m=audio)
- Video (m=video)
- Text (m=text)
- Fax (m=image)

27.5.6 Interworking Miscellaneous Media Handling

This section describes various interworking features relating to media handling.

27.5.6.1 Interworking RTP Redundancy

The device supports interworking of RTP redundancy (according to RFC 2198) between SIP entities. Employing IP Profiles, you can configure RTP redundancy handling per SIP entity:

- Generate RFC 2198 redundant packets (IpProfile_RTPRedundancyDepth parameter).
- Determine RTP redundancy support in the RTP redundancy negotiation in SDP offer/answer (IpProfile_SBCRTPRedundancyBehavior parameter). If not supported, the device discards RTP redundancy packets (if present) received from or sent to the SIP entity.

For more information, see the above parameters in "Configuring IP Profiles" on page 387.

27.5.6.2 Interworking RTP-RTCP Multiplexing

The device supports interworking of RTP-RTCP multiplexing onto a single, local UDP port (according to RFC 5761) between SIP entities. Employing IP Profiles, you can configure RTP multiplexing per SIP entity, using the IPProfile_SBCRTCPMux parameter (see "Configuring IP Profiles" on page 387).

27.5.6.3 Interworking RTCP Attribute in SDP

The device supports interworking the RTCP attribute 'a=rtcp' in the SDP between SIP entities. Employing IP Profiles, you can configure RTCP attribute handling (add, remove or transparent) per SIP entity, using the IpProfile_SBCSDPHandleRTCPAttribute parameter (see "Configuring IP Profiles" on page 387).

27.5.6.4 Interworking Crypto Lifetime Field

The device supports interworking the lifetime field in the 'a=crypto' attribute of the SDP, between SIP entities. Employing IP Profiles, you can configure the lifetime field handling (remove or retain) per SIP entity, using the IpProfile_SBCRemoveCryptoLifetimeInSDP parameter (see "Configuring IP Profiles" on page 387).

27.5.6.5 Interworking Media Security Protocols

The device supports interworking media security protocols for SRTP, between SIP entities. Employing IP Profiles, you can configure the security protocol (SDES or DTLS) per SIP entity, using the IPProfile_SBCMediaSecurityMethod parameter (see "Configuring IP Profiles" on page 387). For more information on SDES and DTLS, see "Configuring Media (SRTP) Security" on page 202.

27.5.6.6 Interworking ICE Lite for NAT Traversal

The device supports interworking ICE for NAT traversal, between SIP entities. Employing IP Profiles, you can enable ICE Lite per SIP entity, using the IPProfile_SBCIceMode parameter (see "Configuring IP Profiles" on page 387).

27.6 Limiting SBC Call Duration

You can define a maximum allowed duration (in minutes) for SBC calls. If an established call reaches this user-defined limit, the device terminates the call. This feature ensures calls are properly terminated, allowing available resources for new calls. This feature is configured using the MaxCallDuration parameter.

27.7 SBC Authentication

The device can authenticate SIP servers and SBC users (clients). The different authentication methods are described in the subsequent subsections.

27.7.1 SIP Authentication Server Functionality

The device can function as an Authentication server for authenticating received SIP message requests, based on HTTP authentication Digest with MD5. Alternatively, such requests can be authenticated by an external, third-party server.

When functioning as an Authentication server, the device can authenticate the following SIP entities:

- **SIP servers:** This is applicable to Server-type IP Groups. This provides protection from rogue SIP servers, preventing unauthorized usage of device resources and functionality. To authenticate remote servers, the device challenges the server with a user-defined username and password that is shared with the remote server. When the device receives an INVITE request from the remote server, it challenges the server by replying with a SIP 401 Unauthorized response containing the WWW-Authenticate header. The remote server then re-sends the INVITE containing an Authorization header with authentication information based on this username-password combination to confirm its identity. The device uses the username and password to authenticate the message prior to processing it.

- **SIP clients:** These are clients belonging to a User-type IP Group. This support prevents unauthorized usage of the device's resources by rogue SIP clients. When the device receives an INVITE or REGISTER request from a client (e.g., SIP phone) for SIP message authorization, the device processes the authorization as follows:
  1. The device challenges the received SIP message only if it is configured as a SIP method (e.g., INVITE) for authorization. This is configured in the IP Group table, using the 'Authentication Method List' parameter.
  2. If the message is received without a SIP Authorization header, the device "challenges" the client by sending a SIP 401 or 407 response. The client then resends the request with an Authorization header (containing the user name and password).
  3. The device validates the SIP message according to the AuthNonceDuration, AuthChallengeMethod and AuthQOP parameters.
     - If validation fails, the device rejects the message and sends a 403 (Forbidden) response to the client.
     - If validation succeeds, the device verifies client identification. It checks that the username and password received from the client is the same username and password in the device's User Information table / database (see "SBC User Information for SBC User Database" on page 664). If the client is not successfully authenticated after three attempts, the device sends a SIP 403 (Forbidden) response to the client. If the user is successfully identified, the device accepts the SIP message request.

The device's Authentication server functionality is configured per IP Group, using the 'Authentication Mode' parameter in the IP Group table (see "Configuring IP Groups" on page 340).

27.7.2 User Authentication based on RADIUS

The device can authenticate SIP clients (users) using a remote RADIUS server. The device supports the RADIUS extension for digest authentication of SIP clients, according to draft-sterman-aaa-sip-01. Based on this standard, the device generates the nonce (in contrast to RFC 5090, where it is done by the RADIUS server).

RADIUS based on draft-sterman-aaa-sip-01 operates as follows:

1. The device receives a SIP request without an Authorization header from the SIP client.
2. The device generates the nonce and sends it to the client in a SIP 407 (Proxy Authentication Required) response.
3. The SIP client sends the SIP request with the Authorization header to the device.
4. The device sends an Access-Request message to the RADIUS server.
5. The RADIUS server verifies the client's credentials and sends an Access-Accept (or Access-Reject) response to the device.
6. The device accepts the SIP client's request (sends a SIP 200 OK or forwards the authenticated request) or rejects it (sends another SIP 407 to the SIP client).

To configure this feature, set the SBCServerAuthMode ini file parameter to 2.

27.8 Interworking SIP Signaling

The device supports interworking of SIP signaling messages to ensure interoperability between communicating SIP UAs or entities. This is critical in network environments where the UAs on opposing SBC legs have different SIP signaling support. For example, some UAs may support different versions of a SIP method while others may not event support a specific SIP method. The configuration method for assigning specific SIP message handling modes to UAs, includes configuring an IP Profile with the required interworking mode, and then assigning the IP Profile to the relevant IP Group.

This section describes some of the device's support for handling SIP methods to ensure interoperability.

27.8.1 Interworking SIP 3xx Redirect Responses

The device supports interworking of SIP 3xx redirect responses. By default, the device's handling of SIP 3xx responses is to send the Contact header unchanged. However, some SIP UAs may support different versions of the SIP 3xx standard while others may not even support SIP 3xx.

The handling of SIP 3xx can be configured for all calls, using the global parameter SBC3xxBehavior. For configuring different SIP 3xx handling options for different UAs (i.e., per IP Group), use the IP Profile table parameter, 'SBC Remote 3xx Mode'.

27.8.1.1 Resultant INVITE Traversing Device

The device can handle SIP 3xx responses so that the new INVITE message sent as a result of the 3xx traverses the device. The reasons for enforcing resultant INVITEs to traverse the device may vary:

- The user that receives the 3xx is unable to route to the 3xx contact (i.e., the user is on the LAN and the new contact is on the WAN). In such a scenario, the device enables the user to reach the WAN contact and overcome NAT problems.
- Enforce certain SBC policies (e.g., call admission control, header manipulation) on the resultant INVITE.

The device enforces this by modifying each Contact in the 3xx response as follows:

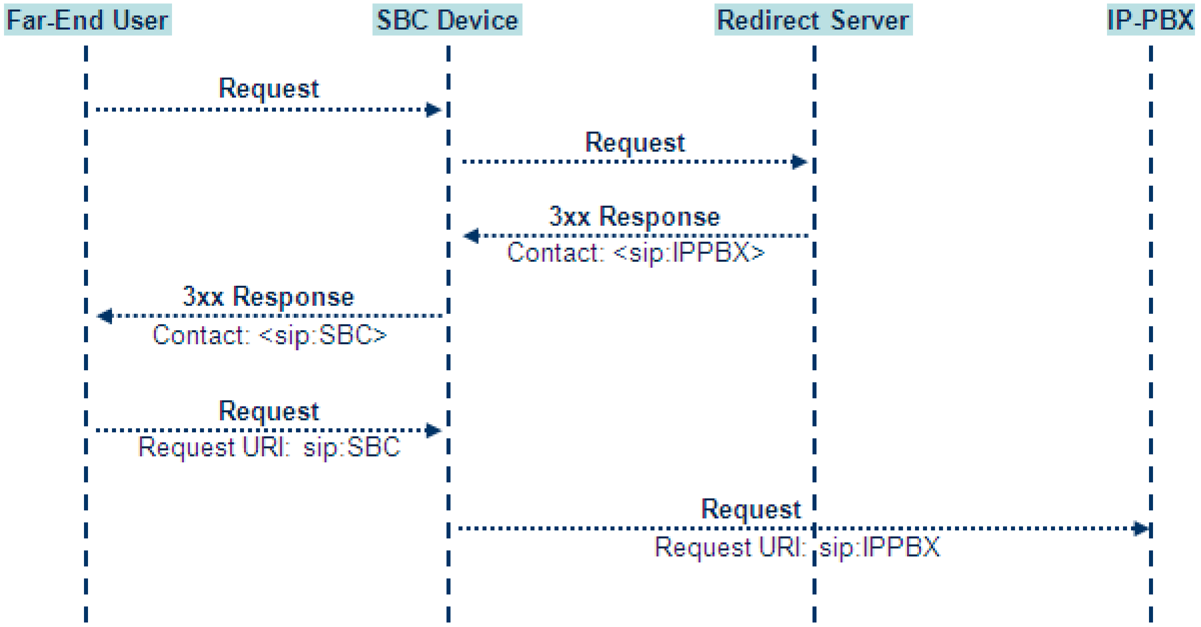
- Changes the host part to the device's IP address – this change causes the remote user agent to send the INVITE to the device.
- Adds a special prefix ("T~&R_") to the Contact user part – to identify the new INVITE as a 3xx resultant INVITE.

The SBC handling for the 3xx resultant INVITE is as follows:

1. The incoming INVITE is identified as a 3xx resultant INVITE according to the special prefix.
2. The device automatically replaces the SBC host part (in the Request-URI) with the host from the 3xx Contact.
3. The prefix ("T~&R_") remains in the user part for the classification, manipulation, and routing mechanisms.
4. The classification, manipulation, and routing processes are done exactly like any other INVITE handling. The special prefix can be used for specific routing rules for 3xx resultant INVITEs.

5. The prefix is removed before the resultant INVITE is sent to the destination.

Figure 27-5: SIP 3xx Response Handling



The process of this feature is described using an example:

1. The device receives the Redirect server's SIP 3xx response (e.g., Contact: <sip:User@IPPBX:5060;transport=tcp;param=a>;q=0.5).
2. The device replaces the Contact header value with the special prefix and database key value as user part, and with the device's URL as host part (e.g., Contact: <sip:Prefix_Key_User@SBC:5070;transport=udp>;q=0.5).
3. The device sends this manipulated SIP 3xx response to the Far-End User (FEU).
4. The FEU sends a new request with the Request-URI set to the value of the received 3xx response's Contact header (e.g., RequestURI: sip:Prefix_Key_User@SBC:5070;transport=udp).
5. Upon receipt of the new request from the FEU, the device replaces the Request-URI with the new destination address (e.g., RequestURI: sip:Prefix_User@IPPBX:5070;transport=tcp;param=a).
6. The device removes the user prefix from the Request-URI, and then sends this Request-URI to the new destination (e.g., RequestURI: sip:User@IPPBX:5070;transport=tcp;param=a).

27.8.1.2 Local Handling of SIP 3xx

The device can handle SIP 3xx responses on behalf of the dialog-initiating UA and retry the request (e.g., INVITE) using one or more alternative URIs included in the 3xx response. The new request includes SIP headers from the initial request such as Diversion, History-Info, P-Asserted-Id, and Priority. The source and destination URIs can be manipulated using the regular manipulation mechanism.

The device sends the new request to the alternative destination according to the IP-to-IP Routing table rules. (where the 'Call Trigger' field is set to **3xx**). It is also possible to specify the IP Group that sent the 3xx request as matching criteria for the re-routing rule in this table ('ReRoute IP Group ID' field).

27.8.2 Interworking SIP Diversion and History-Info Headers

This device can be configured to interwork between the SIP Diversion and History-Info headers. This is important, for example, to networks that support the Diversion header but not the History-Info header, or vice versa. Therefore, mapping between these headers is crucial for preserving the information in the SIP dialog regarding how and why (e.g., call redirection) the call arrived at a certain SIP UA. If the Diversion header is used, you can specify the URI type (e.g., "tel:") to use in the header, using the SBCDiversionUriType parameter.

This feature is configured in the IP Profile table (IPProfile parameter) using the following parameters:

- SBCDiversionMode - defines the device's handling of the Diversion header
- SBCHistoryInfoMode - defines the device’s handling of the History-Info header

The handling of the SIP Diversion and History-Info headers is described in the table below:

Table 27-1: Handling of SIP Diversion and History-Info Headers

Parameter Value	SIP Header Present in Received SIP Message		
	Diversion	History-Info	Diversion and History-Info
HistoryInfoMode = Add **DiversionMode = Remove**	Diversion converted to History-Info. Diversion removed.	Not present	Diversion removed.
HistoryInfoMode = Remove **DiversionMode = Add**	Not present.	History-Info converted to Diversion. History-Info removed.	History-Info added to Diversion. History-Info removed.
HistoryInfoMode = Disable **DiversionMode = Add**	Diversion converted to History-Info.	Not present.	Diversion added to History-Info.
HistoryInfoMode = Disable **DiversionMode = Add**	Not present.	History-Info converted to Diversion.	History-Info added to Diversion.
HistoryInfoMode = Add **DiversionMode = Add**	Diversion converted to History-Info.	History-Info converted to Diversion.	Headers are synced and sent.
HistoryInfoMode = Remove **DiversionMode = Remove**	Diversion removed.	History-Info removed.	Both removed.

27.8.3 Interworking SIP REFER Messages

The device supports interworking of SIP REFER messages. SIP UAs may support different versions of the REFER standard while others may not even support REFER.

This feature supports the following:

- Attended, unattended, and semi-attended call transfers
- Sending INVITE, REFER-notifications, BYE, PRACK and Session Timer on behalf of peer PBXs

- Advanced routing rules for the new, initiated INVITE
- Forwarding early media after REFER while attempting to avoid transcoding (by sending session update)
- Interoperate with environments were different SIP UAs lack basic SIP functionality such as re-INVITE, UPDATE, PRACK, Delayed Offer, re-INVITE without SDP
- Session updates after connect to avoid transcoding

The handling of REFER can be configured for all calls, using the global parameter SBCReferBehavior. For configuring different REFER handling options for different UAs (i.e., IP Groups), use the IP Profile table parameter, 'Remote REFER Mode'.

- Local handling of REFER: This option is used for UAs that do not support REFER. Upon receipt of a REFER request, instead of forwarding it to the IP Group, the device handles it locally. It generates a new INVITE to the alternative destination according to the rules in the IP-to-IP Routing table (where the 'Call Trigger' field is set to **REFER**). It is also possible to specify the IP Group that sent the REFER request, as matching criteria for the re-routing rule in this table ('ReRoute IP Group ID' field).
- Transparent handling: The device forwards the REFER with the Refer-To header unchanged.
- Re-routing through SBC: The device changes the Refer-To header so that the re-routed INVITE is sent through the SBC application.
- IP Group Name: The device sets the host part in the REFER message to the name configured for the IP Group in the IP Group table.

27.8.4 Interworking SIP PRACK Messages

The device supports interworking of SIP Provisional Response ACKnowledgement (PRACK) messages (18x). While some UAs may not support PRACK (RFC 3262) others may require it. The device can be configured to resolve this interoperable issue and enable sessions between such endpoints. SIP PRACK handling is configured using the IP Profile parameter, 'SBC Prack Mode':

- Optional: PRACK is optional for these UAs. If required, the device performs the PRACK process on behalf of the destination UA.
- Mandatory: PRACK is required for these UAs. Calls from UAs that do not support PRACK are rejected. Calls destined to these UAs are also required to support PRACK.
- Transparent (default): The device does not intervene with the PRACK process and forwards the request as is.

27.8.5 Interworking SIP Session Timer

The device supports interworking of the SIP signaling keep-alive mechanism. The SIP standard provides a signaling keep-alive mechanism using re-INVITE and UPDATE messages. In certain setups, keep-alive may be required by some SIP UAs while for others it may not be supported. The device can resolve this mismatch by performing the keep-alive process on behalf of SIP UAs that do not support it.

For configuring the handling of session expires, use the IP Profile parameter, 'SBC Session Expires Mode'.

27.8.6 Interworking SIP Early Media

The device supports early media. Early media is when the media flow starts before the SIP call is established (i.e., before the 200 OK response). This occurs when the first SDP offer-answer transaction completes. The offer-answer options can be included in the following SIP messages:

- Offer in first INVITE, answer on 180, and no or same answer in the 200 OK
- Offer in first INVITE, answer on 180, and a different answer in the 200 OK (not standard)
- INVITE without SDP, offer in 180, and answer in PRACK
- PRACK and UPDATE transactions can also be used for initiating subsequent offer-answer transactions before the INVITE 200 OK response.
- In a SIP dialog life time, media characteristics after originally determined by the first offer-answer transaction can be changed by using subsequent offer-answer transactions. These transactions may be carried either in UPDATE or re-INVITE transactions. The media handling is similar to the original offer-answer handling. If the offer is rejected by the remote party, no media changes occur (e.g., INVITE without SDP, then 200 OK and ACK, offer-answer within an offer-answer, and Hold re-INVITE with IP address of 0.0.0.0 - IP address is unchanged).

The device supports various interworking modes for early media between SIP UAs (i.e., IP Groups):

- **Early Media Enabling:** The device supports the interworking of early media between SIP UAs that support early media and those that do not support receipt of early media. Early media can arrive in provisional responses to an INVITE request. The device forwards the request of early media for IP Groups that support this capability; otherwise, the device terminates it. Provisional responses whose SDP are suppressed are changed to a SIP 180 response. This feature is also supported for delayed offers. This is configured using the IP Profile parameter, 'SBC Remote Early Media Support'. The device refers to the parameter also for features that require early media such as playing ringback tone.
- **Early Media Response Type:** The device supports the interworking of different SIP provisional response types between UAs for forwarding the early media to the caller. This can support all early media response types (default), SIP 180 only, or SIP 183 only, and is configured by the IP Profile parameter, 'SBC Remote Early Media Response Type'.
- **Multiple 18x:** The device supports the interworking of different support for multiple 18x responses (including 180 Ringing, 181 Call is Being Forwarded, 182 Call Queued, and 183 Session Progress) that are forwarded to the caller. The UA can be configured as supporting only receipt of the first 18x response (i.e., the device forwards only this response to the caller), or receipt of multiple 18x responses (default). This is configured by the IP Profile parameter, 'SBC Remote Multiple 18x Support'.
- **Early Media RTP:** The device supports the interworking with remote clients that send 18x responses with early media and whose subsequent RTP is delayed, and with remote clients that do not support this and require RTP to immediately follow the 18x response. Some clients do not support 18x with early media, while others require 18x with early media (i.e., they cannot play ringback tone locally). These various interworking capabilities are configured by the IP Profile parameters, 'Remote Early Media RTP Detection Mode', 'SBC Remote Supports RFC 3960', and 'SBC Remote Can Play Ringback'. See the flowcharts below for the device's handling of such

scenarios:

Figure 27-6: SBC Early Media RTP 18x without SDP

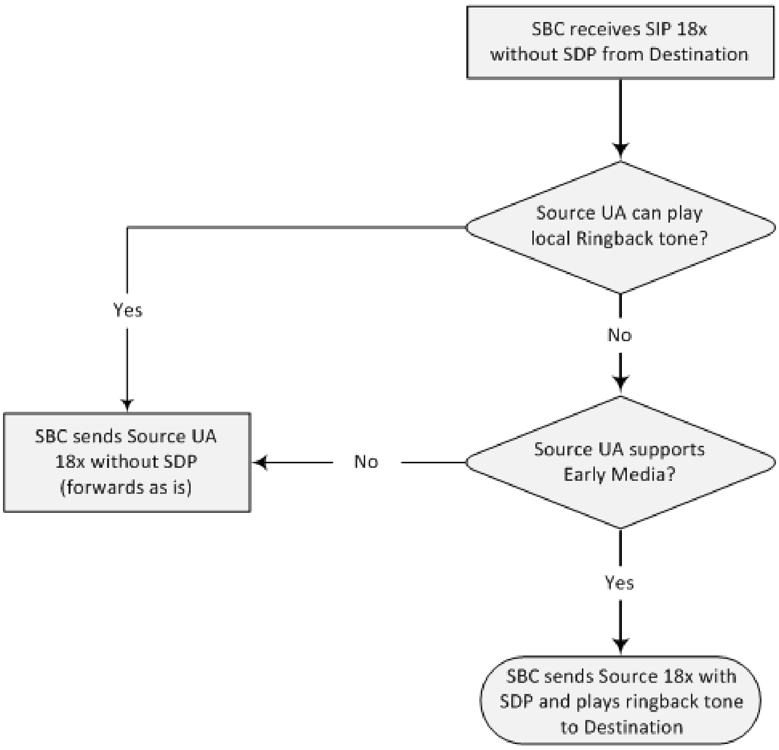
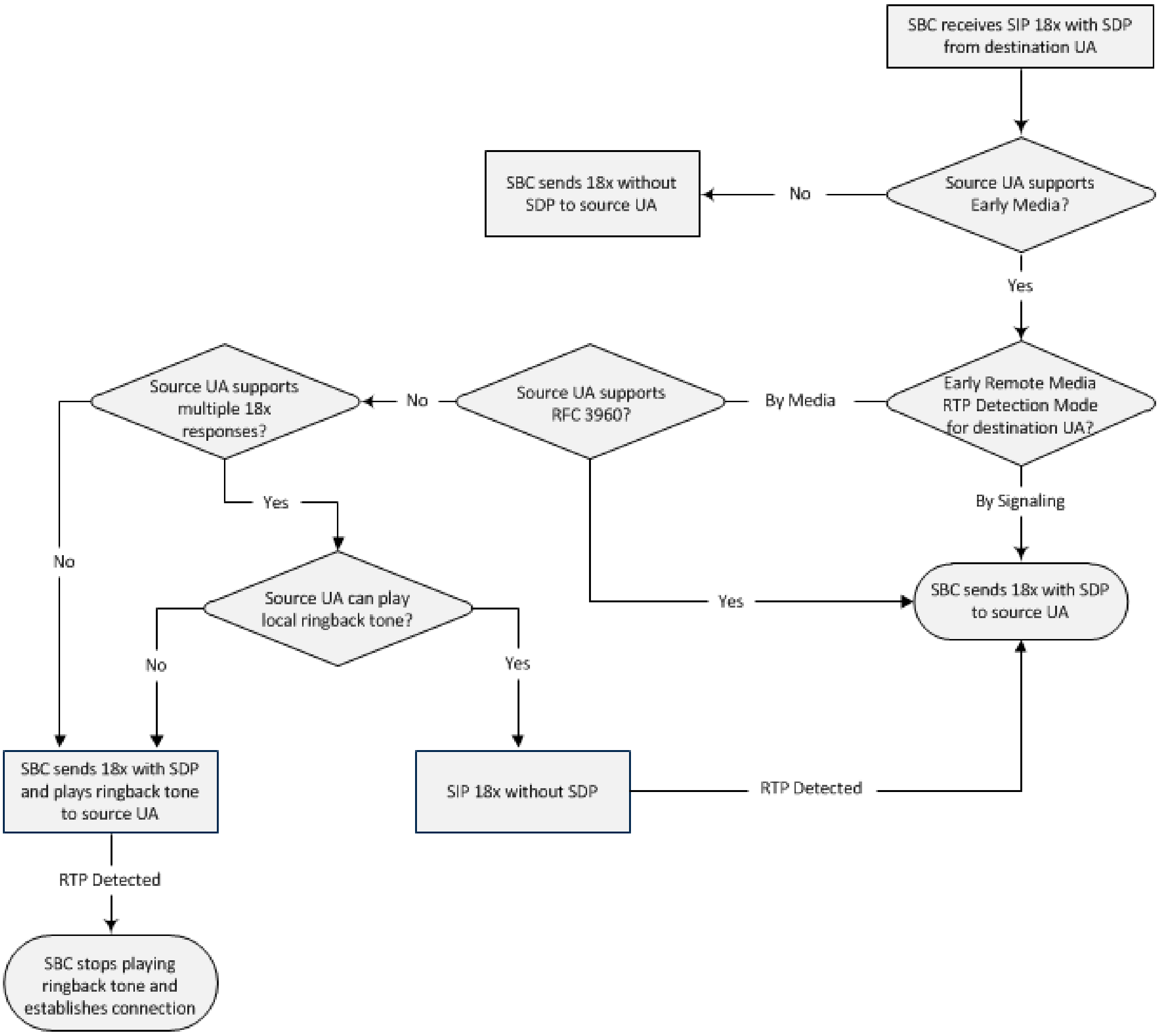

SBC receives SIP 18x without SDP from Destination
Source UA can play local Ringback tone?
Yes
No
SBC sends Source UA 18x without SDP (forwards as is)
No
Source UA supports Early Media?
Yes
SBC sends Source 18x with SDP and plays ringback tone to Destination

Figure 27-7: Early Media RTP - SIP 18x with SDP


SBC receives SIP 18x with SDP from destination UA
Source UA supports Early Media?
No
SBC sends 18x without SDP to source UA
Yes
Early Remote Media RTP Detection Mode for destination UA?
By Media
Source UA supports RFC 3960?
No
Source UA supports multiple 18x responses?
Yes
No
Source UA can play local ringback tone?
No
Yes
By Signaling
SBC sends 18x with SDP to source UA
Yes
SBC sends 18x with SDP and plays ringback tone to source UA
SIP 18x without SDP
RTP Detected
RTP Detected
SBC stops playing ringback tone and establishes connection

27.8.7 Interworking SIP re-INVITE Messages

The device supports interworking of SIP re-INVITE messages. This enables communication between endpoints that generate re-INVITE requests and those that do not support the receipt of re-INVITEs. The device does not forward re-INVITE requests to IP Groups that do not support it. Instead, it sends a SIP response to the re-INVITE request, which can either be a success or a failure, depending on whether the device can bridge the media between the endpoints. The device can also handle re-INVITEs with or without an SDP body, enabling communication between endpoints that do not support re-INVITE requests without SDP, and those that require SDP. The device generates an SDP offer and adds it to the incoming re-INVITE request if it does not contain an SDP and only then forwards it to the destination endpoint. This interworking support is configured by the IP Profile parameter, 'SBC Remote Reinvite Support'.

27.8.8 Interworking SIP UPDATE Messages

The device supports interworking of the SIP UPDATED message. This enables communication between UAs that generate UPDATE requests and those that do not support the receipt of UPDATE requests. The device does not forward UPDATE requests to IP Groups that do not support it. Instead, it sends a SIP response to the UPDATE request which can either be a success or a failure, depending on whether the device can bridge the media between the endpoints. The handling of UPDATE messages is configured by the IP Profile parameter 'SBC Remote Update Support'.

27.8.9 Interworking SIP re-INVITE to UPDATE

The device enables communication between endpoints (IP Groups) that do not support re-INVITE requests but support the UPDATE method, and vice versa. The device translates the re-INVITE request to the UPDATE request, and vice versa. Note that if a re-INVITE request arrives without SDP, the device generates the SDP and adds it to the outgoing UPDATE request. To enable this feature, each IP Group needs to be configured with its unique capabilities by associating it with a relevant IP Profile. For example, an IP Group that supports UPDATE requests but not re-INVITEs would be configured as follows:

- SBCRemoteUpdateSupport = 2 (Supported)
- SBCRemoteReinviteSupport = 0 (Not Supported)

If a re-INVITE request needs to be forwarded to this IP Group, it is translated to an UPDATE request.

27.8.10 Interworking Delayed Offer

The device supports interworking of INVITE messages with and without SDP between SIP entities. The device enables sessions between endpoints (IP Groups) that send INVITEs without SDP (i.e., delayed media) and those that do not support the receipt of INVITEs without SDP. The device creates an SDP and adds it to INVITEs that arrive without SDP. Delayed offer is also supported when early media is present.

Employing IP Profiles, you can configure this interworking feature per SIP entity, using the 'SBC Remote Delayed Offer Support' parameter (see "Configuring IP Profiles" on page 387).

27.8.11 Interworking Call Hold

The device supports the interworking of call hold / retrieve requests between SIP entities supporting different call hold capabilities:

- Interworking SDP call hold formats. This is configured by the IP Profile parameter, 'SBC Remote Hold Format'.
- Interworking the play of the held tone for IP entities that cannot play held tones locally. This is configured by the IP Profile parameter, 'SBC Play Held Tone'.
- Interworking generation of held tone where the device generates the tone to the held party instead of the call hold initiator. This is configured by the IP Profile parameter, 'SBC Reliable Held Tone Source'.

For configuring IP Profiles, see "Configuring IP Profiles" on page 387.

27.8.12 Interworking SIP Via Headers

The device supports the interworking of SIP Via headers between SIP entities. For the outgoing message sent to a SIP entity, the device can remove or retain all the Via headers received in the incoming SIP request from the other side. Employing IP Profiles, you can configure this interworking feature per SIP entity, using the IpProfile_SBCKeepVIAHeaders parameter (see "Configuring IP Profiles" on page 387).

27.8.13 Interworking SIP User-Agent Headers

The device supports the interworking of SIP User-Agent headers between SIP entities. For the outgoing message sent to a SIP entity, the device can remove or retain all the User-Agent headers received in the incoming SIP request/response from the other side. Employing IP Profiles, you can configure this interworking feature per SIP entity, using the IpProfile_SBCKeepUserAgentHeader parameter (see "Configuring IP Profiles" on page 387).

27.8.14 Interworking SIP Record-Route Headers

The device supports the interworking of SIP Record-Route headers between IP entities. For the outgoing message sent to a SIP entity, the device can remove or retain all the Record-Route headers received in the incoming SIP request/response from the other side. Employing IP Profiles, you can configure this interworking feature per SIP entity, using the IpProfile_SBCKeepRoutingHeaders parameter (see "Configuring IP Profiles" on page 387).

27.8.15 Interworking SIP To-Header Tags in Multiple SDP Answers

The device supports the interworking of SIP To-header tags in call forking responses (i.e., multiple SDP answers) between IP entities. The device can either use the same To-header tag value for all SDP answers sent to the SIP entity, or send each SDP answer with its original tag. Employing IP Profiles, you can configure this interworking feature per SIP entity, using the IpProfile_SBCRemoteMultipleEarlyDialogs parameter (see "Configuring IP Profiles" on page 387).

27.8.16 Interworking In-dialog SIP Contact and Record-Route Headers

The device supports the interworking of in-dialog, SIP Contact and Record-Route headers between SIP entities. Employing IP Profiles, you can configure this interworking feature per SIP entity, using the IpProfile_SBCRemoteRepresentationMode parameter (see "Configuring IP Profiles" on page 387).

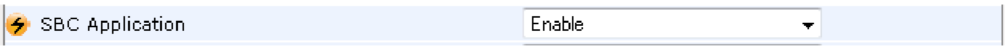
28 Enabling the SBC Application

Before you can start configuring the SBC, you must first enable the SBC application. Once enabled, the Web interface displays the menus and parameter fields relevant to the SBC application.

> **Note:** The SBC feature is available only if the device is installed with a Software License Key that includes this feature. For installing a Software License Key, see "Software License Key" on page 668.

➢ To enable the SBC application:

1. Open the Applications Enabling page (**Configuration** tab > **VoIP** menu > **Applications Enabling** > **Applications Enabling**).



SBC Application: Enable

2. From the 'SBC Application' drop-down list, select **Enable**.
3. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

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29 Utilizing Gateway Channel Resources for SBC

The device can utilize resources of non-configured Gateway channels (digital) for SBC sessions, regardless of whether the device is licensed for SBC functionality. This feature, in essence, allows "call" resources to be migrated from the Gateway application to the SBC application, allowing you to migrate your Gateway deployment to an all IP-based voice network with only a simple configuration change. One of the main advantages of the feature is that if you purchased the device for deploying it initially as a Gateway for PSTN calls, you can at any stage easily use the device for SBC calls without having to purchase an SBC license.

A Gateway channel is considered "not configured" if it is not associated with any Trunk Group (see 'Configuring Trunk Groups' on page 433). If all Gateway channels are configured, resources from these channels cannot be used for SBC sessions. If the resources of a currently active SBC call is obtained from a Gateway channel and you configure all Gateway channels during the call, the device maintains the SBC call until it is terminated by the call parties, but obtaining resources from Gateway channels for new SBC calls will not be made possible.

For every non-configured Gateway channel, one SBC session can be processed. For example, a License Key licensing 1 BRI can support up to 2 SBC sessions (2 channels for 1 BRI port) if all the Gateway channels are not configured. If the License Key also provides a license for 5 SBC sessions, up to 7 SBC sessions (2 channels for 1 BRI port + 5 for SBC) can be supported.

The number of SBC sessions that can be supported if Gateway channels are not configured is displayed in the Web interface's License Key page (see 'Viewing the License Key' on page 668), which displays the licensed features according to the installed License Key.

Note:

- To support the feature, the License Key installed on your device must include the "TDMtoSBC" feature key; otherwise, to purchase the feature, contact your AudioCodes sales representative to upgrade your License Key.
- The maximum number of SBC sessions that can be supported is according to the device's maximum SBC capacity (see 'Channel Capacity' on page 1033).

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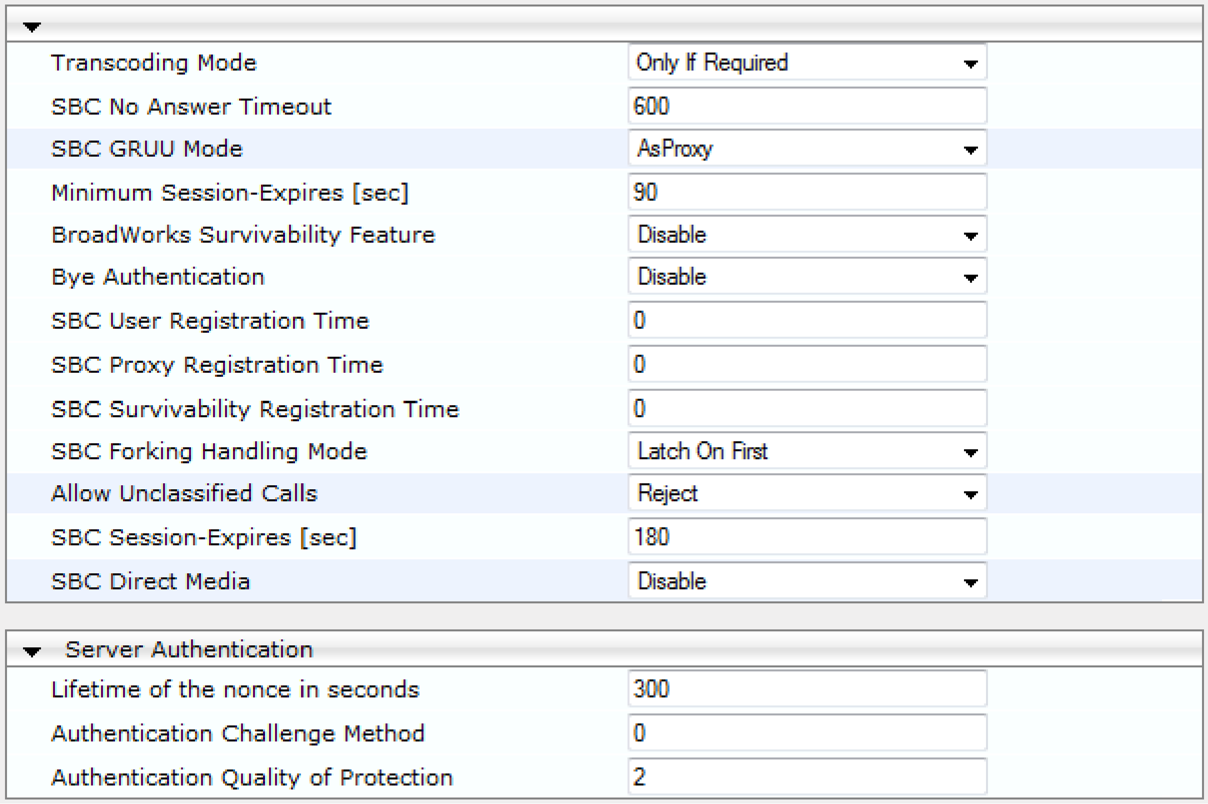
30 Configuring General SBC Settings

The General Settings page allows you to configure general SBC parameters. For a description of these parameters, see "SBC Parameters" on page 998.

➢ **To configure general parameters:**

1. Open the General Settings page (**Configuration** tab > **VoIP** menu > **SBC** > **General Settings**).

Figure 30-1: General Settings Page



Transcoding Mode	Only If Required
SBC No Answer Timeout	600
SBC GRUU Mode	AsProxy
Minimum Session-Expires [sec]	90
BroadWorks Survivability Feature	Disable
Bye Authentication	Disable
SBC User Registration Time	0
SBC Proxy Registration Time	0
SBC Survivability Registration Time	0
SBC Forking Handling Mode	Latch On First
Allow Unclassified Calls	Reject
SBC Session-Expires [sec]	180
SBC Direct Media	Disable

Server Authentication

Lifetime of the nonce in seconds	300
Authentication Challenge Method	0
Authentication Quality of Protection	2

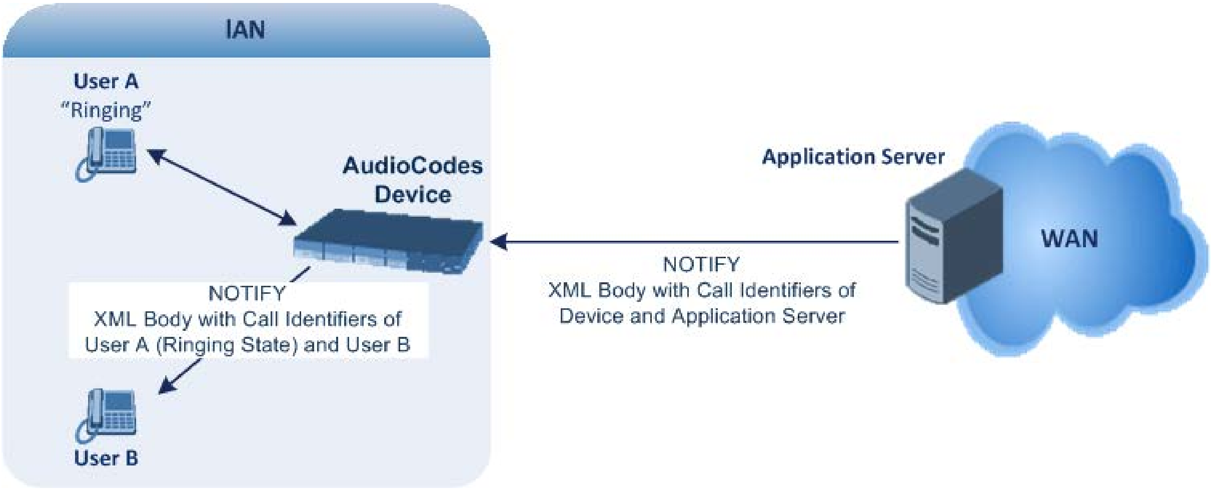
2. Configure the parameters as required.
3. Click **Submit**.
4. To save the changes to flash memory, see "Saving Configuration" on page 643.

30.1 Interworking Dialog Information in SIP NOTIFY Messages

You can enable the device to interwork dialog information (XML body) received in SIP NOTIFY messages from a remote (WAN) application server. The NOTIFY message is sent by application servers to notify a SIP client, subscribed to a service and located behind the device (LAN), of the status of another SIP client in the LAN. For example, user B can subscribe to an application server for call pick-up service, whereby if user A's phone rings, the application server notifies user B. User B can then press a pre-configured key sequence to answer the call.

The NOTIFY message contains the XML body with call identifiers (call-id and tags). However, as the application server is located in the external network WAN and the SIP clients behind the device, the call dialog information sent by the application server reflects only the dialog between the device and itself; not that of the involved SIP clients. This is due to, for example, the device's topology hiding (e.g., IP address) of its LAN elements. The device resolves this by replacing the call identifiers received from the application server with the correct call identifiers (e.g., user A and user B). Thus, users subscribed to the service can receive relevant NOTIFY messages from the device and use the service.

Figure 30-2: Interworking NOTIFY XML Body for Application Server


IAN
User A
"Ringing"
AudioCodes Device
NOTIFY
XML Body with Call Identifiers of User A (Ringing State) and User B
User B
Application Server
WAN
NOTIFY
XML Body with Call Identifiers of Device and Application Server

To enable this feature, set the 'SBC Dialog-Info Interworking' (EnableSBCDialogInfoInterworking) parameter to **Enable**. When this feature is disabled, the device forwards the NOTIFY message as is, without modifying its XML body.

Below is an example of an XML body where the call-id, tags, and URIs have been replaced by the device:

```
<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info"
version="10" state="partial"
entity="sip:alice@example.com">
<dialog id="zxcvbnm3" call-id="67402270@10.132.10.150"
local-tag="1c137249965"
remote-tag="CCDORRTDRKIKWFVBRWYM" direction="initiator">
<state event="replaced">terminated</state>
</dialog>
<dialog id="sfhjsjk12" call-id="67402270@10.132.10.150"
local-tag="1c137249965"
remote-tag="CCDORRTDRKIKWFVBRWYM" direction="receiver">
<state reason="replaced">confirmed</state>
<replaces
call-id="67402270@10.132.10.150"
local-tag="1c137249965"
remote-tag="CCDORRTDRKIKWFVBRWYM"/>
<referred-by>
sip:bob-is-not-here@vm.example.net
</referred-by>
<local>
<identity display="Jason Forster">
sip:jforsters@home.net
</identity>
<target uri="sip:alice@pc33.example.com">
<param pname="+sip.rendering" pval="yes"/>
</target>
</local>
<remote>
<identity display="Cathy Jones">
sip:cjones@example.net
</identity>
<target uri="sip:line3@host3.example.net">
```

```
<param pname="actor" pval="attendant"/>
<param pname="automaton" pval="false"/>
</target>
</remote>
</dialog>
</dialog-info>
```

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31 Configuring Admission Control

The Admission Control table lets you configure up to 102 Call Admission Control rules (CAC). CAC rules define the maximum number of concurrent calls (SIP dialogs) permitted per IP Group, SIP Interface or SRD, and per user (identified by its registered contact). CAC rules also define a guaranteed (*reserved*) number of concurrent calls. Thus, CAC rules can be useful for implementing Service Level Agreements (SLA) policies.

CAC rules can be applied per SIP request type and SIP dialog direction (inbound and/or outbound). These relate to requests that initiate SIP dialogs and not the subsequent requests that can be of different type and direction. The SIP dialog-initiating request types can include INVITE, REGISTER, and/or SUBSCRIBE messages, or it can be configured to include the total number of all dialogs.

This feature also provides support for SIP-dialog rate control, using the “token bucket” mechanism. The token bucket is a control mechanism that dictates the rate of SIP-dialog setups based on the presence of tokens in the bucket – a logical container that holds aggregate SIP dialogs to be accepted or transmitted. Tokens in the bucket are removed ("cashed in") for the ability to setup a dialog. Thus, a flow can setup dialogs up to its peak burst rate if there are adequate tokens in the bucket and if the burst threshold is configured appropriately:

- Every SIP dialog setup request must attempt to take a token from the bucket.
- If there are no tokens, the request is dropped.
- New tokens are added to the bucket at a user-defined rate (token rate).
- If the bucket contains the maximum number of tokens, tokens to be added at that moment are dropped.

Reserved capacity is especially useful when the device operates with multiple SIP entities such as in a contact center environment handling multiple customers. For example, if the total call capacity of the device is 200 call sessions, a scenario may arise where one SIP entity may reach the maximum configured call capacity of 200 and thereby, leaving no available call resources for the other SIP entities. Thus, reserved capacity guarantees a minimum capacity for each SIP entity. If the reserved call capacity of a SIP entity is threatened by a new call for a different SIP entity, the device rejects the call to safeguard the reserved capacity.

Reserved call capacity can be configured for an SRD and each of its associated IP Groups, by configuring multiple CAC rules. In such a setup, the SRD's reserved call capacity must be greater or equal to the summation of the reserved call capacity of all these IP Groups. In other words, the SRD serves as the "parent" reserved call capacity. If the SRD's reserved call capacity is greater, the extra call capacity can be used as a shared pool between the IP Groups for unreserved calls when they exceed their reserved capacity. For example, assume that the reserved capacities for an SRD and its associated IP Groups are as follows:

- SRD reserved call capacity: 40
- IP Group ID 1 reserved call capacity: 10
- IP Group ID 2 reserved call capacity: 20

In this setup, the SRD offers a shared pool for unreserved call capacity of 10 [i.e., 40 – (10 + 20)]. If IP Group ID 1 needs to handle 15 calls, it is guaranteed 10 calls and the remaining 5 is provided from the SRD's shared pool. If the SDR's shared pool is currently empty and resources for new calls are required, the quota is taken from the device's total capacity, if available. For example, if IP Group ID 1 needs to handle 21 calls, it's guaranteed 10, the SRD's shared pool provides another 10, and the last call is provided from the device's total call capacity support (e.g., of 200).

Requests that reach the user-defined call limit (maximum concurrent calls and/or call rate) are sent to an alternative route, if configured in the IP-to-IP Routing table. If no alternative routing rule is located, the device rejects the SIP request with a SIP 480 "Temporarily Unavailable" response.

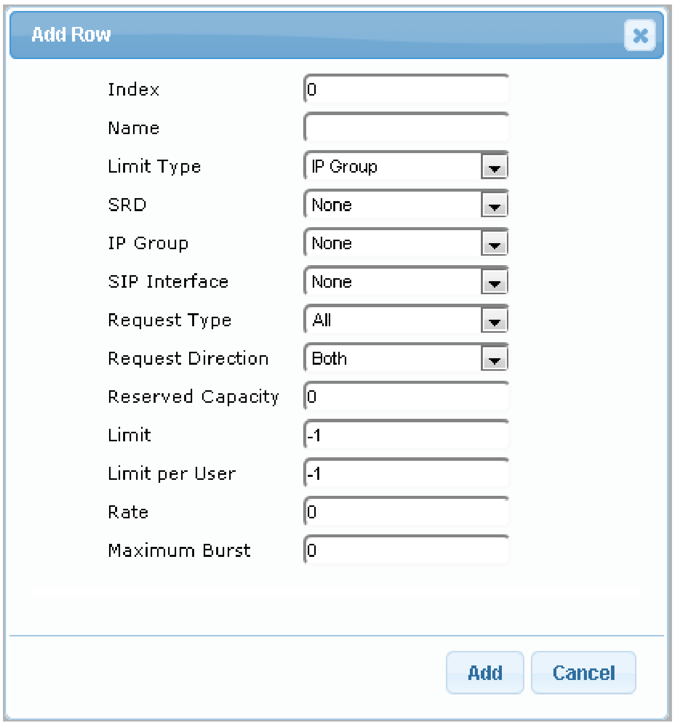
Note: The device applies the CAC rule for the incoming leg immediately after the Classification process. If the call/request is rejected at this stage, no routing is performed. The enforcement for the outgoing leg is performed within each alternative route iteration. This is accessed from two places: one during initial classification/routing, and another during alternative routing process.

The following procedure describes how to configure CAC rules through the Web interface. You can also configure it through ini file (SBCAdmissionControl) or CLI (configure voip > sbc sbc-admission-control).

➢ To configure a CAC rule:

1. Open the Admission Control table (**Configuration** tab > **VoIP** menu > **SBC** > **Admission Control**).
2. Click **Add**; the following dialog box appears:

Figure 31-1: Admission Control Table - Add Row Dialog Box



Add Row

Index	0
Name	
Limit Type	IP Group
SRD	None
IP Group	None
SIP Interface	None
Request Type	All
Request Direction	Both
Reserved Capacity	0
Limit	-1
Limit per User	-1
Rate	0
Maximum Burst	0

3. Configure an Admission Control rule according to the parameters described in the table below.
4. Click **Add**.

Table 31-1: Admission Control Table Parameter Description

Parameter	Description
Index [SBCAdmissionControl_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `admission-name` [SBCAdmissionControl_AdmissionControlName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. By default, no value is defined.
Limit Type	Defines the entity to which the rule applies.

Parameter	Description
`limit-type` [SBCAdmissionControl_LimitType]	- **[0]** IP Group (default) - **[1]** SRD - **[2]** SIP Interface
SRD `srd-name` [SBCAdmissionControl_SRDName]	Assigns an SRD to the rule. For all SRDs, configure the parameter to **Any**. By default, no value is defined (**None**). **Note:** The parameter is applicable only if 'Limit Type' is configured to **SRD**.
IP Group `ip-group-name` [SBCAdmissionControl_IPGroupName]	Assigns an IP Group to the rule. For all IP Groups, configure the parameter to **Any**. By default, no value is defined (**None**). **Note:** The parameter is applicable only if 'Limit Type' is configured to **IP Group**.
SIP Interface `sip-interface-name` [SBCAdmissionControl_SIPInterfaceName]	Assigns a SIP Interface to the rule. For all SIP Interfaces, configure the parameter to **Any**. By default, no value is defined (**None**). **Note:** The parameter is applicable only if 'Limit Type' is configured to **SIP Interface**.
Request Type `request-type` [SBCAdmissionControl_RequestType]	Defines the SIP dialog-initiating request type to which you want to apply the rule (not subsequent requests that can be of different type and direction). - **[0]** All (default) - **[1]** INVITE - **[2]** SUBSCRIBE - **[3]** Other
Request Direction `request-direction` [SBCAdmissionControl_RequestDirection]	Defines the direction of the SIP request to which the rule applies. - **[0]** Both = (Default) Rule applies to inbound and outbound SIP dialogs. - **[1]** Inbound = Rule applies only to inbound SIP dialogs. - **[2]** Outbound = Rule applies only to outbound SIP dialogs.
Reserved Capacity `reservation` [SBCAdmissionControl_Reservation]	Defines the guaranteed (minimum) call capacity. **Notes:** - Reserved call capacity is applicable only to IP Groups and SRDs (i.e., the 'Limit Type' parameter must be configured to IP Group or SRD). If you configure the 'Limit Type' parameter to SIP Interface, leave the 'Reserved Capacity' parameter at its' default (i.e., 0).. - Reserved call capacity is applicable only to INVITE and SUBSCRIBE messages. - Reserved call capacity must be less than the maximum capacity (limit) configured for the CAC rule. - The total reserved call capacity configured for all the CAC rules must be within the device's total call capacity support.
Limit `limit` [SBCAdmissionControl_Limit]	Defines the maximum number of concurrent SIP dialogs per IP Group, SIP Interface or SRD. You can also use the following special values: - **[0]** 0 = Block all these dialogs. - **[-1]** -1 = (Default) Unlimited.

Parameter	Description
Limit Per User `limit-per-user` [SBCAdmissionControl_LimitPerUser]	Defines the maximum number of concurrent SIP dialogs per user belonging to the specified IP Group, SIP Interface or SRD. You can also use the following special values: ▪ **[0]** 0 = Block all these dialogs. ▪ **[-1]** -1 = (Default) Unlimited.
Rate `rate` [SBCAdmissionControl_Rate]	Defines the maximum number of SIP dialogs per IP Group, SIP Interface or SRD that can be handled per second. The default is 0 (i.e., unlimited rate). **Notes:** You must first configure the Maximum Burst parameter (see below) before configuring the Rate parameter. The token bucket feature is per IP Group, SRD, SIP request type, and SIP request direction.
Maximum Burst `max-burst` [SBCAdmissionControl_MaxBurst]	Defines the maximum number of tokens (SIP dialogs) that the bucket can hold. The device only accepts a SIP dialog if a token exists in the bucket. Once the SIP dialog is accepted, a token is removed from the bucket. If a SIP dialog is received by the device and the token bucket is empty, then the device rejects the SIP dialog. Alternatively, if the bucket is full, for example, 100 tokens, and 101 SIP dialogs arrive (before another token is added to the bucket, i.e., faster than that defined in the Rate field), then the device accepts the first 100 SIP dialogs and rejects the last one. Dropped requests are replied with the SIP 480 “Temporarily Unavailable” response. Dropped requests are not counted in the bucket. The default is 0 (i.e., unlimited SIP dialogs). **Note:** The token bucket feature is per IP Group, SIP Interface, SRD, SIP request type, and SIP request direction.

32 Configuring Coder Groups

32.1 Configuring Allowed Audio Coder Groups

The Allowed Audio Coders Group table lets you configure up to 10 Allowed Audio Coders Groups. For each Allowed Audio Coders Group, you can configure up to 10 audio coders, which can include default coders and user-defined (string) coders for non-standard or unknown coders.

Allowed Audio Coders Groups restrict coders used for SIP entities. Only coders listed in the Allowed Audio Coders Group (i.e., allowed coders) that is associated with the SIP entity can be used. If the coders in the SDP offer ('a=rtpmap' field) of the incoming SIP message are not listed in the Allowed Audio Coders Group, the device rejects the calls. If the SDP offer contains some coders that are listed in the Allowed Audio Coders Group, the device manipulates the SDP offer by removing the coders that are not listed in the Allowed Audio Coders Group, before routing the SIP message to its destination. Thus, only coders that are common between the coders in the SDP offer and the coders in the Allowed Audio Coders Group are used. For more information on coder restriction, see "Restricting Audio Coders" on page 540.

For example, assume the following:

- The SDP offer in the incoming SIP message contains the G.729, G.711, and G.723 coders.
- The allowed coders configured for the SIP entity include G.711 and G.729.

The device removes the G.723 coder from the SDP offer, re-orders the coder list so that G.711 is listed first, and sends the SIP message containing only the G.711 and G.729 coders in the SDP.

To apply an Allowed Audio Coders Group for restricting coders to a SIP entity:

1. Configure an Allowed Audio Coders Group in the Allowed Audio Coders Group table (see description below).
2. In the IP Profile associated with the SIP entity (see "Configuring IP Profiles" on page 387):
   - Assign the Allowed Audio Coders Group (using the IpProfile_SBCAllowedCodersGroupID parameter).
   - Enable the use of Allowed Audio Coder Groups (by configuring the IpProfile_SBCAllowedCodersMode parameter to **Restriction** or **Restriction and Preference**).

The device also re-orders (prioritizes) the coder list in the SDP according to the order of appearance of the coders listed in the Allowed Audio Coders Group. The first listed coder has the highest priority and the last coder has the lowest priority. For more information, see "Prioritizing Coder List in SDP Offer" on page 541.

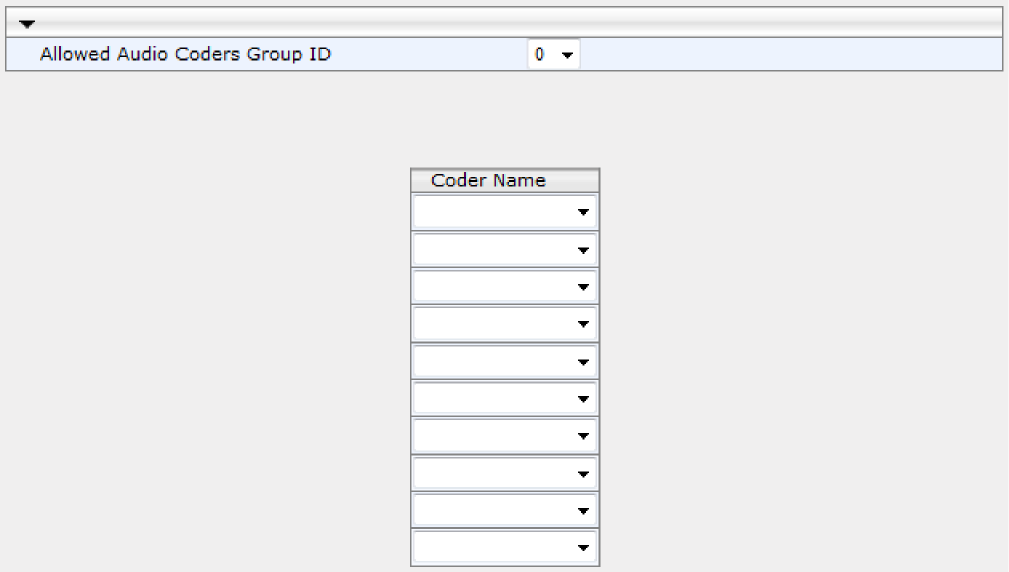
The following procedure describes how to configure Allowed Audio Coder Groups through the Web interface. You can also configure it through ini file (AllowedCodersGroup) or CLI (configure voip > sbc allowed-coders-group group-0).

➢ To configure an Allowed Coders Group:

1. Open the Allowed Audio Coders Group page (**Configuration** tab > **VoIP** menu > **SBC**

> **Allowed Audio Coders Group**).

Figure 32-1: Allowed Audio Coders Group Page



Allowed Audio Coders Group ID: 0

2. Configure an Allowed Audio Coders Group according to the parameters described in the table below.
3. Click **Submit**.

Table 32-1: Allowed Audio Coders Group Table Parameter Descriptions

Parameter	Description
Allowed Audio Coders Group ID [AllowedCodersGroupX]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Coder Name name [AllowedCodersGroupX_Name]	Defines the audio coder. This can be a pre-defined coder or a user-defined coder. The valid value for user-defined coders is a string of up to 25 characters (case-insensitive). For example, "HD.123" (without quotes). **Note:** Each coder type (e.g., G.729) can be configured only once per Allowed Coders Group.

32.2 Configuring Allowed Video Coder Groups

The Allowed Video Coders Group table lets you configure up to four Allowed Video Coders Groups. An Allowed Video Coders Group defines a list of video coders that can be used when forwarding video streams to a specific SIP entity. Each Allowed Video Coders Group can be configured with up to 20 coders. The coders can include default video coders and user-defined (string) video coders for non-standard or unknown coders. Allowed Video Coders Groups are assigned to SIP entities, using IP Profiles (see "Configuring IP Profiles" on page 387). The video coders appear in the SDP media type "video" ('m=video' line). Coders that are not listed in the Allowed Video Coders Group are removed from the SDP offer that is sent to the SIP entity. Only coders that are common between the coders in the SDP offer and the coders listed in the Allowed Video Coders Group are used. Thus, Allowed Video Coders Groups enable you to enforce the use of only specified coders. For more information, see "Restricting Audio Coders" on page 540.

The order of appearance of the coders listed in the Allowed Video Coders Group determines the priority (preference) of the coders in the SDP offer. The device arranges the SDP offer's coder list according to their order in the Allowed Video Coders Group. The priority is in descending order, whereby the first coder in the list is given the highest priority and the last coder, the lowest priority. For more information, see "Prioritizing Coder List in SDP Offer" on page 541.

Currently, the Allowed Video Coder Groups table can only be configured through ini file (AllowedVideoCodersGroup) or CLI (configure voip > sbc allowed-video-coders-group group-0). The table below describes the parameter.

Table 32-2: Allowed Video Coders Group Table Parameter Descriptions

Parameter	Description
Allowed Coders Group ID [AllowedVideoCodersGroupX]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Coder Name `name` [AllowedVideoCodersGroupX_Name]	Defines the video coder. This can be a default coder or a user-defined coder. The valid value for user-defined coders is a string of up to 25 characters (case-insensitive). For example, "WOW.789" (but without quotes). **Note:** Each coder type can be configured only once per Allowed Video Coders Group.

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33 Routing SBC

This section describes the configuration of the call routing entities for the SBC application.

33.1 Configuring Classification Rules

The Classification table lets you configure up to 102 Classification rules. A Classification rule classifies incoming SIP dialog-initiating requests (e.g., INVITE messages) to a "source" IP Group. The source IP Group is the SIP entity that sent the SIP dialog request. Once classified, the device uses the IP Group to process the call (manipulation and routing).

You can also use the Classification table for employing SIP-level access control for successfully classified calls, by configuring Classification rules with whitelist and blacklist settings. If a Classification rule is configured as a whitelist ("Allow"), the device accepts the SIP dialog and processes the call. If the Classification rule is configured as a blacklist ("Deny"), the device rejects the SIP dialog.

Configuration of Classification rules includes two areas:

- **Rule:** Defines the matching characteristics of the incoming IP call (e.g, source SIP Interface and IP address). Classification is primarily based on the SIP Interface (as the matching characteristics) on which the incoming dialog is received. As Classification rules must first be assigned with an SRD, the SIP Interface is one that belongs to the SRD. Therefore, Classification rules are configured per SRD, where multiple SIP Interfaces can be used as matching characteristics. However, as multiple SRDs are relevant only for multi-tenant deployments, for most deployments only a single SRD is required. As the device provides a default SRD ("Default_SRD"), when only one SRD is required, the device automatically assigns it to the Classification rule.
- **Action:** Defines the action that is done if the incoming call matches the characteristics of the rule (i.e., classifies the call to the specified IP Group).

The device searches the table from top to bottom for the first rule that matches the characteristics of the incoming call. If it finds a matching rule, it classifies the call to the IP Group configured for that rule. If it doesn't find a matching rule (i.e., classification fails), the device either rejects or allows the call depending on the setting of the 'Unclassified Calls' parameter (see Configuring General SBC Settingson page 557). If the parameter is set to Allow, the incoming SIP dialog is assigned to an IP Group as follows:

1. The device determines on which SIP listening port (e.g., 5061) the incoming SIP dialog request was received and the SIP Interface configured with this port (in the SIP Interface table).
2. The device determines the SRD associated with this SIP Interface (in the SIP Interface table) and then classifies the SIP dialog to the first IP Group in the IP Group table that is associated with the SRD. For example, if IP Groups 3 and 4 belong to the same SRD, the device classifies the call to IP Group 3.

Note: If classification of a SIP request fails and the device is configured to reject unclassified calls, the device can send a specific SIP response code per SIP Interface. This is configured by the 'Classification Failure Response Type' parameter in the SIP Interface table (see "Configuring SIP Interfaces" on page 333).

The Classification table is used to classify incoming SIP dialog requests **only if** the following classification stages fail:

1. **Classification Stage 1 - Based on User Registration Database:** The device searches its users registration database to check whether the incoming SIP dialog arrived from a registered user. The device searches the database for a user that matches the address-of-record (AOR) and Contact of the incoming SIP message:

- Compares the SIP Contact header to the contact value of the user in the database.
- Compares the URL in the SIP P-Asserted-Identity/From header to the registered address-of-record (AOR) in the database.

If the device finds a matching registered user, it classifies the user to the IP Group associated with the user in the database. If this classification stage fails, the device proceeds to classification based on Proxy Set.

2. **Classification Stage 2 - Based on Proxy Set:** If the database search fails, the device performs classification based on Proxy Set. This classification is applicable only to Server-type IP Groups and is done only if classification based on Proxy Set is enabled (see the 'Classify By Proxy Set' parameter in the IP Group table in ''Configuring IP Groups'' on page 340). The device checks whether the incoming INVITE's IP address (if host name, then according to the dynamically resolved IP address list) is configured for a Proxy Set (in the Proxy Set table). If such a Proxy Set exists, the device classifies the INVITE to the IP Group that is associated with the Proxy Set. The Proxy Set is assigned to the IP Group in the IP Group table.

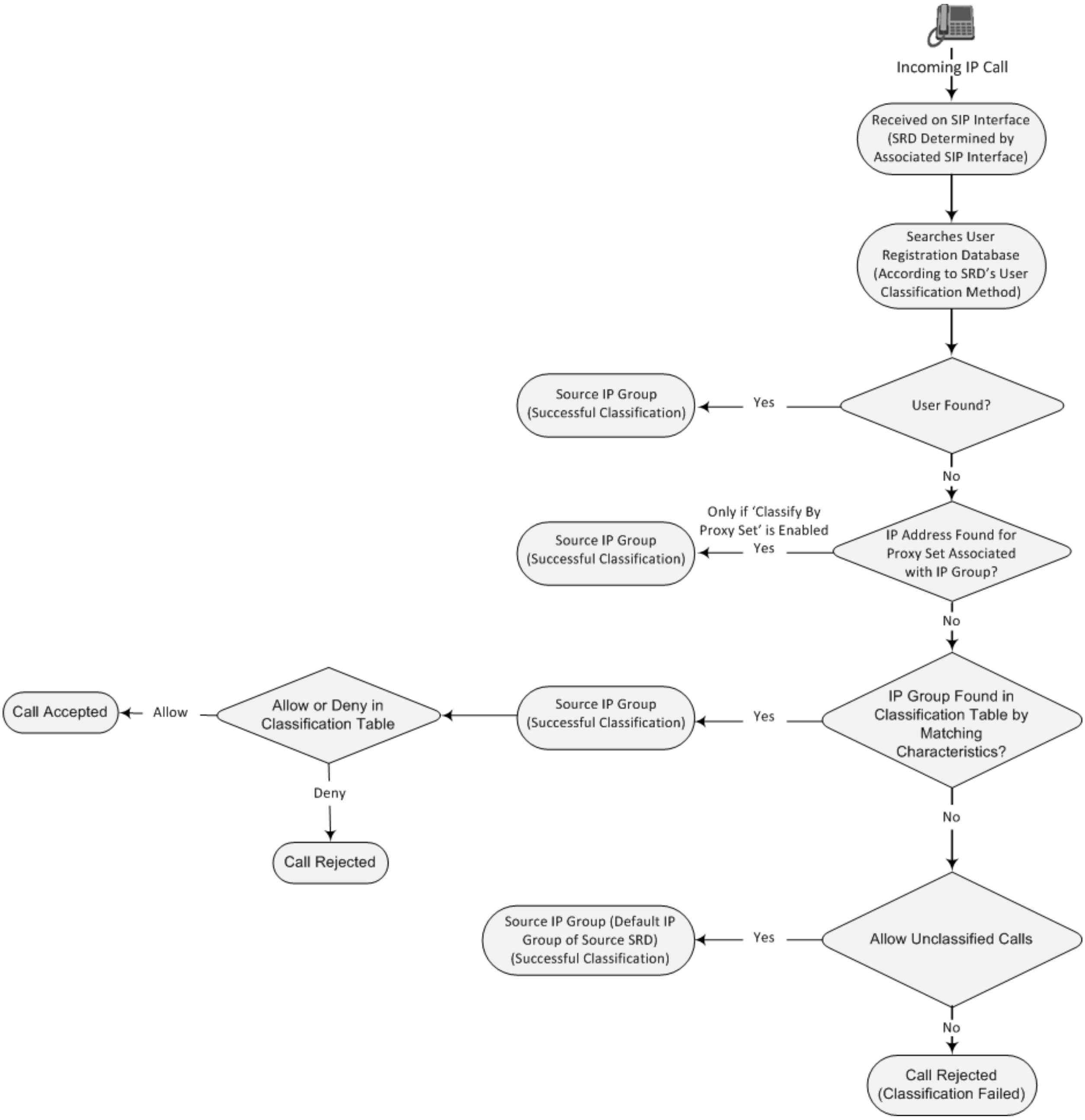
   If classification based on Proxy Set fails (or classification based on Proxy Set is disabled), the device proceeds to classification based on the Classification table.

Note:

- For security, it is recommended to classify SIP dialogs based on Proxy Set only if the IP address of the Server-type IP Group is unknown. In other words, if the Proxy Set associated with the IP Group is configured with an FQDN. In such cases, the device classifies incoming SIP dialogs to the IP Group based on the DNS-resolved IP address. If the IP address is known, it is recommended to use a Classification rule instead (and disable the Classify by Proxy Set feature), where the rule is configured with not only the IP address, but also with SIP message characteristics to increase the strictness of the classification process. The reason for preferring classification based on Proxy Set when the IP address is unknown is that IP address forgery (commonly known as IP spoofing) is more difficult than malicious SIP message tampering and therefore, using a Classification rule without an IP address offers a weaker form of security. When classification is based on Proxy Set, the Classification table for the specific IP Group is ignored.
- If multiple IP Groups are associated with the same Proxy Set, use Classification rules to classify the incoming dialogs to the IP Groups (do not use the Classify by Proxy Set feature).
- The device saves incoming SIP REGISTER messages in its registration database. If the REGISTER message is received from a User-type IP Group, the device sends the message to the configured destination.

The flowchart below illustrates the classification process:

Figure 33-1: Classification Process (Identifying IP Group or Rejecting Call)


Incoming IP Call
Received on SIP Interface (SRD Determined by Associated SIP Interface)
Searches User Registration Database (According to SRD's User Classification Method)
User Found?
Yes
Source IP Group (Successful Classification)
No
IP Address Found for Proxy Set Associated with IP Group?
Only if 'Classify By Proxy Set' is Enabled
Yes
Source IP Group (Successful Classification)
No
IP Group Found in Classification Table by Matching Characteristics?
Yes
Source IP Group (Successful Classification)
Allow or Deny in Classification Table
Allow
Call Accepted
Deny
Call Rejected
No
Allow Unclassified Calls
Yes
Source IP Group (Default IP Group of Source SRD) (Successful Classification)
No
Call Rejected (Classification Failed)

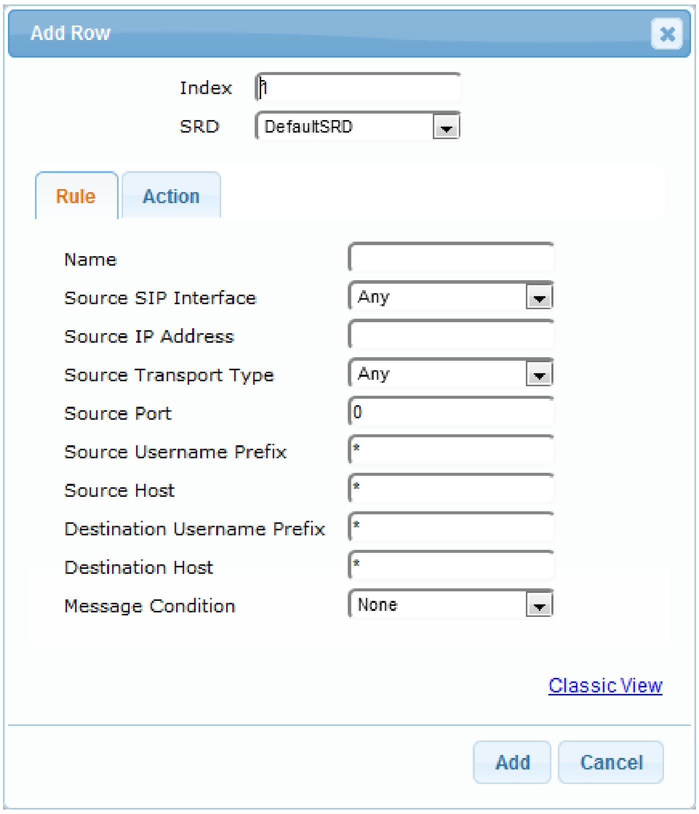
The following procedure describes how to configure Classification rules through the Web interface. You can also configure it through ini file (Classification) or CLI (configure voip > sbc routing classification).

➢ To configure a Classification rule:

1. Open the Classification table (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **Classification Table**).

2. Click **Add**; the following dialog box appears:

Figure 33-2: Classification Table - Add Row Dialog Box



Index: 1
SRD: DefaultSRD

Name:
Source SIP Interface: Any
Source IP Address:
Source Transport Type: Any
Source Port: 0
Source Username Prefix: *
Source Host: *
Destination Username Prefix: *
Destination Host: *
Message Condition: None

3. Configure the Classification rule according to the parameters described in the table below.
4. Click **Add**.

Table 33-1: Classification Table Parameter Descriptions

Parameter	Description
Index [Classification_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `classification-name` [Classification_ClassificationName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. By default, no name is defined. **Note:** Each row must be configured with a unique name.
Rule (Matching Characteristics)	
SRD `srd-name` [Classification_SRDName]	Assigns an SRD to the rule as a matching characteristics for the incoming SIP dialog. If only one SRD is configured in the SRD table, the SRD is assigned to the rule, by default. If multiple SRDs are configured in the SRD table, no value is assigned. For configuring SRDs, see "Configuring SRDs" on page 325. **Note:** The parameter is mandatory.
Source SIP Interface `src-sip-interface-name` [Classification_SrcSIPInterfaceName]	Assigns a SIP Interface to the rule as a matching characteristics for the incoming SIP dialog.

Parameter	Description
	The default is **Any** (i.e., all SIP Interfaces belonging to the SRD assigned to the rule). **Note:** The SIP Interface must belong to the SRD assigned to the rule (see the 'SRD' parameter in the table).
Source IP Address src-ip-address [Classification_SrcAddress]	Defines a source IP address as a matching characteristics for the incoming SIP dialog. The valid value is an IP address in dotted-decimal notation. In addition, the following wildcards can be used: ▪ "x" wildcard: represents single digits. For example, 10.8.8.xx represents all addresses between 10.8.8.10 and 10.8.8.99. ▪ Asterisk (*) wildcard: represents any number between 0 and 255. For example, 10.8.8.* represents all addresses between 10.8.8.0 and 10.8.8.255. By default, no value is defined (i.e., any source IP address is accepted). Note: ▪ The parameter is applicable only to Server-type IP Groups. ▪ If the IP address is unknown (i.e., configured for the associated Proxy Set as an FQDN), it is recommended to classify incoming dialogs based on Proxy Set (instead of using a Classification rule). For more information on classification by Proxy Set or by Classification rule, see the note bulletin in the beginning of this section.
Source Transport Type `src-transport-type` [Classification_SrcTransportType]	Defines the source transport type as a matching characteristics for the incoming SIP dialog. ▪ **[-1]** Any = (Default) All transport types ▪ **[0]** UDP ▪ **[1]** TCP ▪ **[2]** TLS
Source Port `src-port` [Classification_SrcPort]	Defines the source port number as a matching characteristics for the incoming SIP dialog.
Source Username Prefix `src-user-name-prefix` [Classification_SrcUsernamePrefix]	Defines the prefix of the source URI user part as a matching characteristics for the incoming SIP dialog. The URI is typically located in the SIP From header. However, you can configure the SIP header from where the device obtains the source URI, in the IP Group table ('Source URI Input' parameter). For more information on how the device obtains the URI, see "SIP Dialog Initiation Process" on page 531. The default is the asterisk (*) symbol, which represents any source username prefix. The prefix can be a single digit or a range of digits. For available notations, see "Dialing Plan Notation for Routing and Manipulation" on page 809. **Note:** For REGISTER requests, the source URI is obtained from the To header.

Parameter	Description
Source Host `src-host` [Classification_SrcHost]	Defines the prefix of the source URI host name as a matching characteristics for the incoming SIP dialog. The URI is typically located in the SIP From header. However, you can configure the SIP header from where the device obtains the source URI, in the IP Group table ('Source URI Input' parameter). For more information on how the device obtains this URI, see "Call Processing of SIP Dialog Requests" on page 531. The default is the asterisk (*) symbol, which represents any source host prefix. **Note:** For REGISTER requests, the source URI is obtained from the To header.
Destination Username Prefix `dst-user-name-prefix` [Classification_DestUsernamePrefix]	Defines the prefix of the destination Request-URI user part as a matching characteristics for the incoming SIP dialog. The default is the asterisk (*) symbol, which represents any destination username. The prefix can be a single digit or a range of digits. For available notations, see "Dialing Plan Notation for Routing and Manipulation" on page 809.
Destination Host `dst-host` [Classification_DestHost]	Defines the prefix of the destination Request-URI host name as a matching characteristics for the incoming SIP dialog. The default is the asterisk (*) symbol, which represents any destination host prefix.
Message Condition `message-condition-name` [Classification_MessageConditionName]	Assigns a Message Condition rule to the Classification rule as a matching characteristics for the incoming SIP dialog. By default, no value is defined (**None**). To configure Message Condition rules, see "Configuring Message Condition Rules" on page 576.
Action	
Action Type `action-type` [Classification_ActionType]	Defines a whitelist or blacklist for the matched incoming SIP dialog. ▪ **[0]** Deny = Blocks incoming SIP dialogs that match the characteristics of the rule (blacklist). ▪ **[1]** Allow = (Default) Allows incoming SIP dialogs that match the characteristics of the rule (whitelist) and assigns it to the associated IP Group.
Destination Routing Policy `dest-routing-policy` [Classification_DestRoutingPolicy]	Assigns an SBC Routing Policy to the matched incoming SIP dialog. The assigned SBC Routing Policy overrides the SBC Routing Policy assigned to the SRD (in the SRD table). The option to assign Routing Policies to Classification rules is useful in deployments requiring different routing and manipulation rules for specific calls pertaining to the **same** SRD. In such scenarios, you need to configure multiple Classification rules for the same SRD, where for some rules no Routing Policy is assigned (i.e., the SRD's assigned Routing Policy is used) while for others a

Parameter	Description
	different Routing Policy is specified to override the SRD's assigned Routing Policy. By default, no value is defined (**None**). For configuring SBC Routing Policies, see "Configuring SBC Routing Policy Rules" on page 590.
Source IP Group `src-ip-group-name` [Classification_SrcIPGroupName]	Assigns an IP Group to the matched incoming SIP dialog. By default, no value is defined (**None**). For configuring IP Groups, see "Configuring IP Groups" on page 340. **Note:** The IP Group must be associated with the assigned SRD (see the 'SRD' parameter in the table).
IP Profile `ip-profile-id` [Classification_IpProfileName]	Assigns an IP Profile to the matched incoming SIP dialog. The assigned IP Profile overrides the IP Profile assigned to the IP Group (in the IP Group table) to which the SIP dialog is classified. Therefore, assigning an IP Profile during classification allows you to assign different IP Profiles to specific users (calls) that belong to the same IP Group (User or Server type). For example, you can configure two Classification rules to classify incoming calls to the same IP Group. However, one Classification rule is a regular rule that doesn't specify any IP Profile (IP Profile assigned to IP Group is used), while the second rule is configured with an additional matching characteristic for the source hostname prefix (e.g., "abcd.com") and with an additional action that assigns a different IP Profile. By default, no value is defined (**None**). **Note:** For User-type IP Groups, if a user is already registered with the device (from a previous, initial classification process), the device classifies subsequent INVITE requests from the user according to the device's users database instead of the Classification table. In such a scenario, the same IP Profile that was previously assigned to the user by the Classification table is also used (in other words, the device's users database stores the associated IP Profile).

33.1.1 Classification Based on URI of Selected Header Example

The following example describes how to configure classification of incoming calls to IP Groups, based on source URI in a specific SIP header.

This example assumes the following incoming INVITE message:

```
INVITE sip:8000@10.33.4.226 SIP/2.0
Via: SIP/2.0/UDP 10.33.4.226;branch=z9hG4bKVEBTDAHSUYRTEXEDEGJY
From: <sip:100@10.33.4.226>;tag=YSQQKXXREVDPYPTNFMWG
To: <sip:8000@10.33.4.226>
Call-ID: FKPNOYRNKROIMEGBSSKS@10.33.4.226
CSeq: 1 INVITE
Contact: <sip:100@10.33.4.226>
```

```
Route: <sip:2000@10.10.10.10.10>,<sip:300@10.10.10.30>
Supported: em,100rel,timer,replaces
P-Called-Party-ID: <sip:1111@10.33.38.1>
User-Agent: Sip Message Generator V1.0.0.5
Content-Length: 0
```

1. In the Classification table, add the following classification rules:

Index	Source Username Prefix	Destination Username Prefix	Destination Host	Source IP Group
0	333	-	-	1
1	1111	2000	10.10.10.10	2

2. In the IP Group table, add the following IP Groups:

Index	Source URI Input	Destination URI Input
1	-	-
2	P-Called-Party-ID	Route

In this example, a match exists only for Classification Rule #1. This is because the source (1111) and destination (2000) username prefixes match those in the INVITE's P-Called-Party-ID header (i..e., "<sip:1111@10.33.38.1>") and Route header (i.e., "<sip:2000@10.10.10.10.10>"), respectively. These SIP headers were determined in IP Group 2.

33.2 Configuring Message Condition Rules

The Message Condition table lets you configure up to 82 Message Condition rules. A Message Condition defines special conditions (requisites) for incoming SIP messages. These rules can be used as additional matching criteria for the following:

- Classification rules in the Classification table (see "Configuring Classification Rules" on page 569)
- IP-to-IP routing rules in the IP-to-IP Routing table (see "Configuring SBC IP-to-IP Routing Rules" on page 578)
- IP-to-IP outbound manipulation rules in the IP to IP Outbound Manipulation table (see "Configuring IP-to-IP Outbound Manipulations" on page 601)

Message Condition rules are configured using the same syntax as that used for Conditions when configuring Message Manipulation rules in the Message Manipulations table (see "Configuring SIP Message Manipulation" on page 370). You can configure simple Message Condition rules, for example, "header.to.host contains company", meaning SIP messages whose To header has a host part containing the string "company". You can configure complex rules using the "AND" or "OR" Boolean operands and also use regular expressions (regex), for example:

- "body.sdp regex pcmu" can be used to enable routing based on the offered codec (G.711 Mu) in the incoming SDP message.
- "body.sdp regex (AVP[0-9||\s]*\s8[\s||\n])" can be used to enable routing based on payload type 8 in the incoming SDP message.

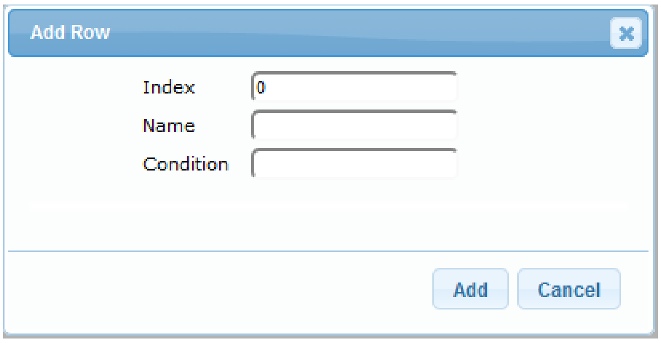
Note: For a description on SIP message manipulation syntax, refer to the *SIP Message Manipulations Quick Reference Guide.*

The following procedure describes how to configure Message Condition rules through the Web interface. You can also configure it through ini file (ConditionTable) or CLI (configure voip > sbc routing condition-table).

➢ To configure a Message Condition rule:

1. Open the Message Condition table (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **Message Condition Table**).
2. Click **Add**; the following dialog box appears:

Figure 33-3: Message Condition Table - Add Row Dialog Box

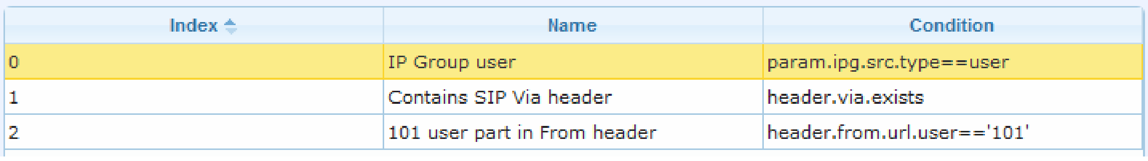


Index: 0
Name:
Condition:

3. Configure a Message Condition rule according to the parameters described in the table below.
4. Click **Add**.

An example of configured Message Condition rules is shown in the figure below:

Figure 33-4: Example of Configured SIP Message Conditions



Index	Name	Condition
0	IP Group user	param.ipg.src.type==user
1	Contains SIP Via header	header.via.exists
2	101 user part in From header	header.from.url.user=='101'

- **Index 0:** Incoming SIP dialog that is classified as belonging to a User-type IP Group.
- **Index 1:** Incoming SIP dialog that contains a SIP Via header.
- **Index 2:** Incoming SIP dialog with 101 as the user part in the SIP From header.

Table 33-2: Message Condition Table Parameter Descriptions

Parameter	Description
Index [ConditionTable_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `name` [ConditionTable_Name]	Defines a brief description of the Condition rule. The valid value is a string of up to 59 characters.
Condition `condition` [ConditionTable_Condition]	Defines the Condition rule of the SIP message. The valid value is a string. **Note:** User and host parts must be enclosed in single quotes.

33.3 Configuring SBC IP-to-IP Routing

The IP-to-IP Routing table lets you configure up to 615 SBC IP-to-IP routing rules.

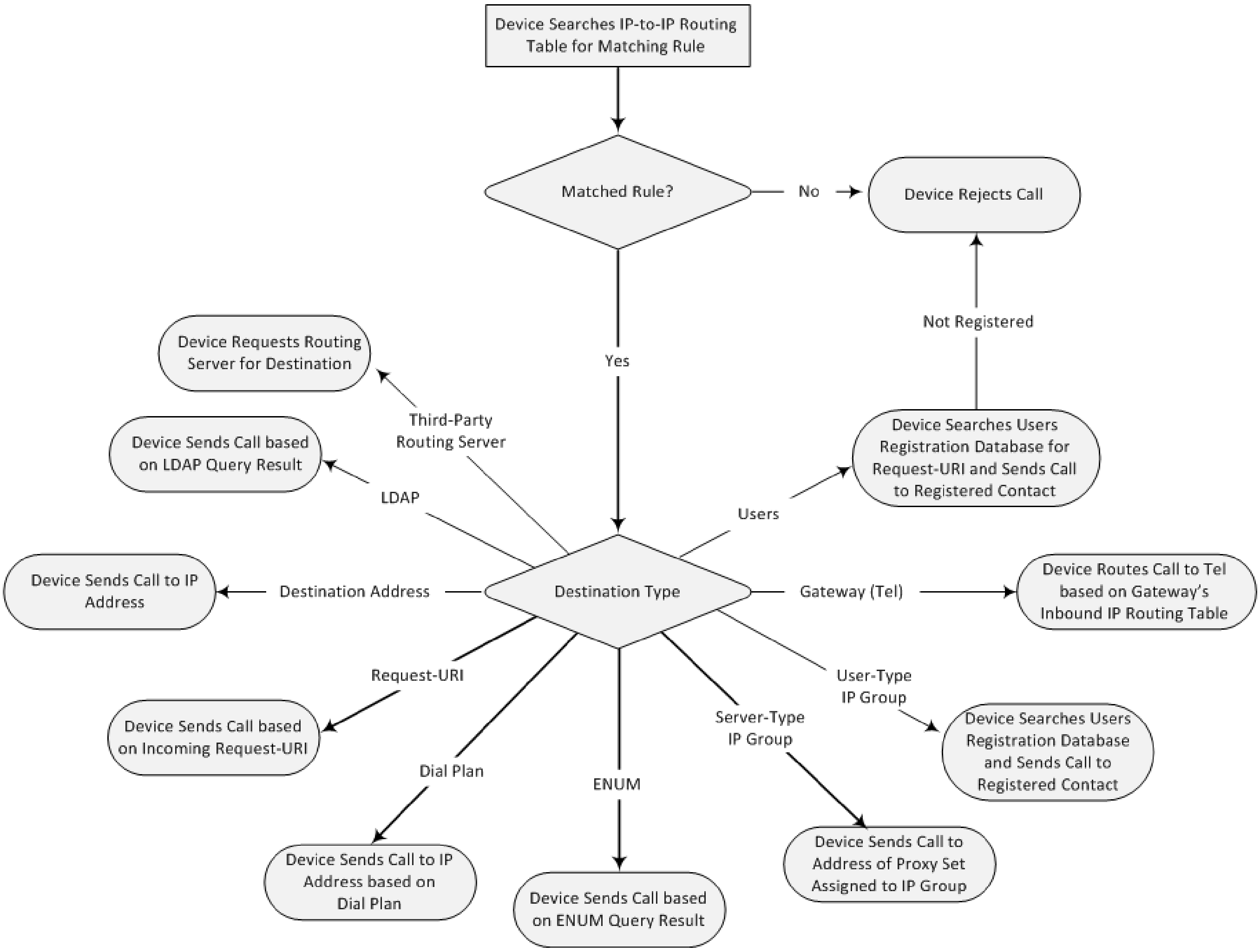
Configuration of IP-to-IP routing rules includes two areas:

- **Rule:** Defines the characteristics of the incoming SIP dialog message (e.g., IP Group from which the message is received).
- **Action:** Defines the action that is done if the incoming call matches the characteristics of the rule (i.e., routes the call to the specified destination).

The device searches the table from top to bottom for the first rule that matches the characteristics of the incoming call. If it finds a matching rule, it sends the call to the destination configured for that rule. If it doesn't find a matching rule, it rejects the call.

An IP-to-IP routing rule routes received SIP dialog messages (e.g., INVITE) to any of the following configurable IP destinations:

- According to registered user Contact listed in the device's registration database (only for User-type IP Groups).
- IP Group - the destination is the address configured for the Proxy Set associated with the IP Group.
- IP address in dotted-decimal notation or FQDN. Routing to a host name can be resolved using NAPTR/SRV/A-Record.
- Request-URI of incoming SIP dialog-initiating requests.
- Any registered user in the registration database. If the Request-URI of the incoming INVITE exists in the database, the call is sent to the corresponding contact address specified in the database.
- According to result of an ENUM query.
- Hunt Group - used for call survivability of call centers (see "Call Survivability for Call Centers" on page 623).
- According to result of LDAP query (for more information on LDAP-based routing, see "Routing Based on LDAP Active Directory Queries" on page 232).
- Third-party routing server, which determines the destination (next hop) of the call (IP Group). The IP Group represents the next device in the routing path to the final destination. For more information, see "Centralized Third-Party Routing Server or ARM" on page 272.
- Tel destination (i.e., Gateway call). The rule redirects the call to the Inbound IP Routing table where the device searches for a matching IP-to-Tel routing rule. This feature can also be done for alternative routing. If an IP-to-IP routing rule fails and it is configured with a "Gateway" routing rule as an alternative route, the device uses the Inbound IP Routing table to send the call to the Tel. The device identifies (internally) calls re-directed for alternative Gateway routing, by appending a user-defined string to the prefix destination Request-URI user part (by default, "acgateway-<prefix destination>", for example, acgateway-200). The device removes this prefix before sending it to the Tel side. To configure this prefix string, use the GWDirectRoutePrefix ini file parameter.


Device Searches IP-to-IP Routing Table for Matching Rule
Matched Rule?
No
Device Rejects Call
Not Registered
Yes
Device Requests Routing Server for Destination
Third-Party Routing Server
Device Sends Call based on LDAP Query Result
LDAP
Device Searches Users Registration Database for Request-URI and Sends Call to Registered Contact
Users
Device Sends Call to IP Address
Destination Address
Destination Type
Gateway (Tel)
Device Routes Call to Tel based on Gateway's Inbound IP Routing Table
Request-URI
User-Type IP Group
Server-Type IP Group
Device Searches Users Registration Database and Sends Call to Registered Contact
Device Sends Call based on Incoming Request-URI
Dial Plan
ENUM
Device Sends Call to IP Address based on Dial Plan
Device Sends Call based on ENUM Query Result
Device Sends Call to Address of Proxy Set Assigned to IP Group

To configure and apply an IP-to-IP Routing rule, the rule must be associated with a Routing Policy. The Routing Policy associates the routing rule with an SRD(s). Therefore, the Routing Policy lets you configure routing rules for calls belonging to specific SRD(s). However, as multiple Routing Policies are relevant only for multi-tenant deployments (if needed), for most deployments, only a single Routing Policy is required. As the device provides a default Routing Policy ("Default_SBCRoutingPolicy"), when only one Routing Policy is required, the device automatically assigns the default Routing Policy to the routing rule. If you are implementing LDAP-based routing (with or without Call Setup Rules) and/or Least Cost Routing (LCR), you need to configure these settings for the Routing Policy (regardless of the number of Routing Policies employed). For more information on Routing Policies, see "Configuring SBC Routing Policy Rules" on page 590.

The IP-to-IP Routing table also provides the following features:

- **Alternative routing or load balancing:** In addition to the alternative routing/load balancing provided by the Proxy Set associated with the destination IP Group, the table allows the configuration of alternative routes whereby if a route fails, the next adjacent (below) rule in the table that is configured as 'Alt Route Ignore/Consider Inputs' are used. The alternative routes rules can be set to enforce the input matching criteria or to ignore any matching criteria. Alternative routing occurs upon one of the following conditions:
  - A request sent by the device is responded with one of the following:
    - SIP response code (i.e., 4xx, 5xx, and 6xx SIP responses) configured in the SBC Alternative Routing Reasons table (see "Configuring SIP Response Codes for Alternative Routing Reasons" on page 588).
    - SIP 408 Timeout or no response (after timeout).

- The DNS resolution includes IP addresses that the device has yet to try (for the current call).

Messages are re-routed with the same SIP Call-ID and CSeq header fields (increased by 1).

> **Note:** If the Proxy Set (see Configuring Proxy Sets on page 353) associated with the destination of the call is configured with multiple IP addresses, the device first attempts to route the call to one of these IP addresses, starting with the first listed address. Only when the call cannot be routed to any of the Proxy Set's IP addresses does the device search the IP-to-IP Routing table for an alternative routing rule for the call.

- **Re-routing of SIP requests:** This table enables you to configure "re-routing" rules of requests (e.g., INVITEs) that the device sends upon receipt of SIP 3xx responses or REFER messages. These rules are configured for destinations that do not support receipt of 3xx or REFER and where the device handles the requests locally (instead of forwarding the 3xx or REFER to the destination).
- **Least cost routing (LCR):** If the LCR feature is enabled, the device searches the routing table for matching routing rules and then selects the one with the lowest call cost. The call cost of the routing rule is done by assigning it a Cost Group. For configuring Cost Groups, see "Least Cost Routing" on page 261. If two routing rules have identical costs, then the rule appearing higher up in the table (i.e., first-matched rule) is used. If a selected route is unavailable, the device uses the next least-cost routing rule. However, even if a matched rule is not assigned a Cost Group, the device can select it as the preferred route over other matched routing rules that are assigned Cost Groups, according to the default LCR settings configured for the assigned Routing Policy (see "Configuring SBC Routing Policy Rules" on page 590).
- **Call Forking:** The IP-to-IP Routing table can be configured to route an incoming IP call to multiple destinations (call forking). The incoming call can be routed to multiple destinations of any type such as an IP Group or IP address. The device forks the call by sending simultaneous INVITE messages to all the specified destinations. It handles the multiple SIP dialogs until one of the calls is answered and then terminates the other SIP dialogs.

  Call forking is configured by creating a Forking group. A Forking group consists of a main routing rule ('Alternative Route Options' set to **Route Row**) whose 'Group Policy' is set to **Forking**, and one or more associated routing rules ('Alternative Route Options' set to **Group Member Ignore Inputs** or **Group Member Consider Inputs**). The group members must be configured in contiguous table rows to the main routing rule. If an incoming call matches the input characteristics of the main routing rule, the device routes the call to its destination and all those of the group members.

  An alternative routing rule can also be configured for the Forking group. The alternative route is used if the call fails for the Forking group (i.e., main route and all its group members). The alternative routing rule must be configured in the table row immediately below the last member of the Forking group. The 'Alternative Route Options' of this alternative route must be set to **Alt Route Ignore Inputs** or **Alt Route Consider Inputs**. The alternative route can also be configured with its own forking group members, where if the device uses the alternative route, the call is also sent to its group members. In this case, instead of setting the alternative route's 'Group Policy' to **None**, you must set it to **Forking**. The group members of the alternative route must be configured in the rows immediately below it.

  The LCR feature can also be employed with call forking. The device calculates a maximum call cost for each Forking group and routes the call to the Forking group with the lowest cost. Thus, even if the call can successfully be routed to the main routing rule, a different routing rule can be chosen (even an alternative route, if configured) based on LCR. If routing to one Forking group fails, the device tries to route the call to the Forking group with the next lowest cost (main or alternative route), and so on. The prerequisite for this functionality is that the incoming call must successfully match the input characteristics of the main routing rule.

- **Dial Plan Tags for Representing Source / Destination Numbers:** If your deployment includes calls of many different called (source URI user name) and/or calling (destination URI user name) numbers that need to be routed to the same destination, you can employ user-defined tags to represent these numbers. Thus, instead of configuring many routing rules, you need to configure only one routing rule using the tag as the source and destination number matching characteristics, and a destination for the calls. For more information on prefix tags, see Configuring Dial Plans on page 607.

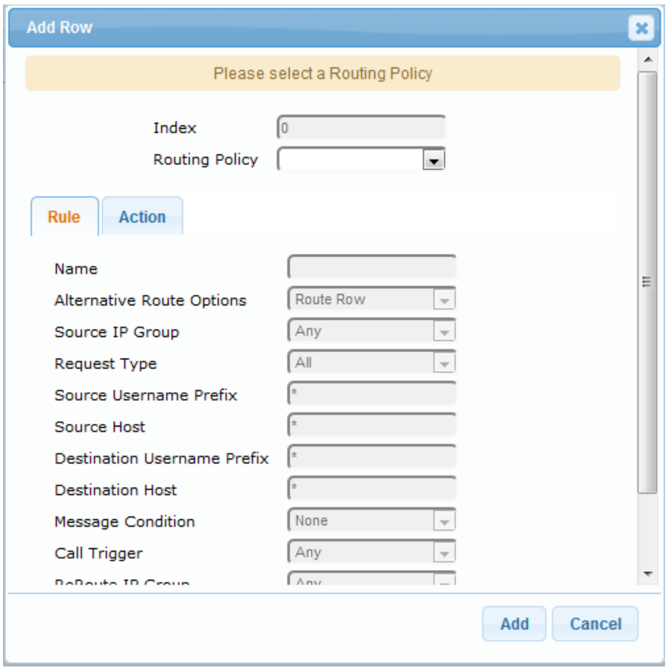
Note: Call forking is not applicable to LDAP-based IP-to-IP routing rules.

The following procedure describes how to configure IP-to-IP routing rules through the Web interface. You can also configure it through ini file (IP2IPRouting) or CLI (configure voip > sbc routing ip2ip-routing).

➢ To configure an IP-to-IP routing rule:

1. Open the IP-to-IP Routing table (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **IP-to-IP Routing Table**).
2. Click **Add**; the following dialog box appears:

Figure 33-5: IP-to-IP Routing Table - Add Row Dialog Box



Add Row

Please select a Routing Policy

Index	0
Routing Policy	

Rule | Action

Name	
Alternative Route Options	Route Row
Source IP Group	Any
Request Type	All
Source Username Prefix	*
Source Host	*
Destination Username Prefix	*
Destination Host	*
Message Condition	None
Call Trigger	Any
ReRoute IP Group	Any

3. Configure an IP-to-IP routing rule according to the parameters described in the table below.
4. Click **Add**.

Table 33-3: IP-to-IP Routing Table Parameter Descriptions

Parameter	Description
Index	Defines an index number for the new table row.

Parameter	Description
[IP2IPRouting_Index]	**Note:** Each row must be configured with a unique index.
Routing Policy `sbc-routing-policy-name` [IP2IPRouting_RoutingPolicyName]	Assigns an SBC Routing Policy to the rule. The SBC Routing Policy associates the rule with an SRD(s). The SBC Routing Policy also defines default LCR settings as well as the LDAP servers used if the routing rule is based on LDAP routing (and Call Setup Rules). If only one SBC Routing Policy is configured in the SBC Routing Policy table, the SBC Routing Policy is automatically assigned. If multiple SBC Routing Policies are configured, no value is assigned. For configuring SBC Routing Policies, see "Configuring SBC Routing Policy Rules" on page 590. **Note:** The parameter is mandatory.
Name `route-name` [IP2IPRouting_RouteName]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. By default, no value is defined.
Rule (Matching Characteristics)	
Alternative Route Options `alt-route-options` [IP2IPRouting_AltRouteOptions]	Determines whether this routing rule is the main routing rule or an alternative routing rule (to the rule defined directly above it in the table). ▪ [0] Route Row = (Default) Main routing rule - the device first attempts to route the call to this route if the incoming SIP dialog's input characteristics matches this rule. ▪ [1] Alternative Route Ignore Inputs = If the call cannot be routed to the main route (Route Row), the call is routed to this alternative route regardless of the incoming SIP dialog's input characteristics. ▪ [2] Alternative Route Consider Inputs = If the call cannot be routed to the main route (Route Row), the call is routed to this alternative route only if the incoming SIP dialog matches this routing rule's input characteristics. ▪ [3] Group Member Ignore Inputs = This routing rule is a member of the Forking routing rule. The incoming call is also forked to the destination of this routing rule. The matching input characteristics of the routing rule are ignored. ▪ [4] Group Member Consider Inputs = This routing rule is a member of the Forking routing rule. The incoming call is also forked to the destination of this routing rule only if the incoming call matches this rule's input characteristics. **Notes:** ▪ The alternative routing entry ([1] or [2]) must be defined in the next consecutive table entry index to the Route Row entry (i.e., directly below it). For example, if Index 4 is configured as a Route Row, Index 5 must be configured as the alternative route. ▪ The Forking Group members must be configured in a table row that is immediately below the main Forking routing rule, or below an alternative routing rule for the main rule, if configured. ▪ For IP-to-IP alternative routing, configure alternative routing reasons upon receipt of 4xx, 5xx, and 6xx SIP responses

Parameter	Description
	(see "Configuring SIP Response Codes for Alternative Routing Reasons" on page 588). However, if no response, ICMP, or a SIP 408 response is received, the device attempts to use the alternative route even if no entries are configured in the 'SBC Alternative Routing Reasons' table. ▪ Multiple alternative route entries can be configured (e.g., Index 1 is the main route - Route Row - and indices 2 through 4 are configured as alternative routes).
Source IP Group `src-ip-group-name` [IP2IPRouting_SrcIPGroupName]	Defines the IP Group from where the IP call was received. Typically, the IP Group of an incoming SIP dialog is determined (or classified) using the Classification table (see "Configuring Classification Rules" on page 569). The default is **Any** (i.e., any IP Group). **Note:** The selectable IP Group for the parameter depends on the assigned SBC Routing Policy (in the 'Routing Policy' parameter in this table). For more information, see "Configuring SBC Routing Policy Rules" on page 590.
Request Type `request-type` [IP2IPRouting_RequestType]	Defines the SIP dialog request type (SIP Method) of the incoming SIP dialog. ▪ [0] All (default) ▪ [1] INVITE ▪ [2] REGISTER ▪ [3] SUBSCRIBE ▪ [4] INVITE and REGISTER ▪ [5] INVITE and SUBSCRIBE ▪ [6] OPTIONS
Source Username Prefix `src-user-name-prefix` [IP2IPRouting_SrcUsernamePrefix]	Defines the prefix of the user part of the incoming SIP dialog's source URI (usually the From URI). You can use special notations for denoting the prefix. To denote calls without a user part in the URI, use the $ sign. For available notations, see "Dialing Plan Notation for Routing and Manipulation" on page 809. The default is the asterisk (*) symbol (i.e., any prefix). If this rule is not required, leave this field empty. **Note:** If you need to route calls of many different source URI user names to the same destination, you can use tags (see 'Source Tags' parameter below) instead of this parameter.
Source Host `src-host` [IP2IPRouting_SrcHost]	Defines the host part of the incoming SIP dialog's source URI (usually the From URI). The default is the asterisk (*) symbol (i.e., any host name). If this rule is not required, leave this field empty.
Source Tags `src-tags` [IP2IPRouting_SrcTags]	Assigns a tag to denote source URI user names corresponding to the tag configured in the associated Dial Plan. The valid value is a string of up to 20 characters. The tag is case insensitive. To configure tags, see Configuring Dial Plans on page 612. **Note:**

Parameter	Description
	▪ Make sure that you assign the Dial Plan in which you have configured the tag, to the related IP Group or SRD. ▪ Instead of using tags and configuring the parameter, you can use the 'Source Username Prefix' parameter to specify a specific URI source user or all source users.
Destination Username Prefix `dst-user-name-prefix` [IP2IPRouting_DestUsernamePrefix]	Defines the prefix of the incoming SIP dialog's destination URI (usually the Request URI) user part. You can use special notations for denoting the prefix. To denote calls without a user part in the URI, use the $ sign. For available notations, see "Dialing Plan Notation for Routing and Manipulation" on page 809. The default is the asterisk (*) symbol (i.e., any prefix). If this rule is not required, leave this field empty. **Note:** If you need to route calls of many different destination URI user names to the same destination, you can use tags (see 'Destination Tags' parameter below) instead of this parameter.
Destination Host `dst-host` [IP2IPRouting_DestHost]	Defines the host part of the incoming SIP dialog's destination URI (usually the Request-URI). The default is the asterisk (*) symbol (i.e., any destination host). If this rule is not required, leave this field empty.
Destination Tags dst-tags [IP2IPRouting_DestTags]	Assigns a tag to denote destination URI user names corresponding to the tag configured in the associated Dial Plan. The valid value is a string of up to 20 characters. The tag is case insensitive. To configure tags, see Configuring Dial Plans on page 612. **Note:** ▪ Make sure that you assign the Dial Plan in which you have configured the tag, to the related IP Group or SRD. ▪ Instead of using tags and configuring the parameter, you can use the 'Destination Username Prefix' parameter to specify a specific URI destination user or all destination users.
Message Condition `message-condition-name` [IP2IPRouting_MessageConditionName]	Assigns a SIP Message Condition rule to the IP-to-IP Routing rule. For configuring Message Condition rules, see "Configuring Message Condition Rules" on page 576.
Call Trigger `trigger` [IP2IPRouting_Trigger]	Defines the reason (i.e., trigger) for re-routing the SIP request: ▪ **[0]** Any = (Default) This routing rule is used for all scenarios (re-routes and non-re-routes). ▪ **[1]** 3xx = Re-routes the request if it was triggered as a result of a SIP 3xx response. ▪ **[2]** REFER = Re-routes the INVITE if it was triggered as a result of a REFER request. ▪ **[3]** 3xx or REFER = Applies to options [1] and [2]. ▪ **[4]** Initial only = This routing rule is used for regular requests that the device forwards to the destination. This rule is not used for re-routing of requests triggered by the receipt of REFER or 3xx. ▪ [5] Broken Connection = If the device detects a broken RTP connection during the call and the Broken RTP Connection feature is enabled (IpProfile_DisconnectOnBrokenConnection parameter is

Parameter	Description
	configured to [2]), you can use this option as an explicit matching characteristics to route the call to an alternative destination. Therefore, for alternative routing upon broken RTP detection, position the routing rule configured with this option above the regular routing rule associated with the call. Such a configuration setup ensures that the device uses this alternative routing rule only when RTP broken connection is detected.
ReRoute IP Group `re-route-ip-group-id` [IP2IPRouting_ReRouteIPGroupName]	Defines the IP Group that initiated (sent) the SIP redirect response (e.g., 3xx) or REFER message. This parameter is typically used for re-routing requests (e.g., INVITEs) when interworking is required for SIP 3xx redirect responses or REFER messages. For more information, see "Interworking SIP 3xx Redirect Responses" on page 544 and "Interworking SIP REFER Messages" on page 546, respectively. The parameter functions together with the 'Call Trigger' parameter (in the table). The default is **Any** (i.e., any IP Group). **Note:** The selectable IP Group for the parameter depends on the assigned SBC Routing Policy (in the 'Routing Policy' parameter in this table). For more information, see "Configuring SBC Routing Policy Rules" on page 590.
Action	
Destination Type `dst-type` [IP2IPRouting_DestType]	Determines the destination type to which the outgoing SIP dialog is sent. ▪ **[0]** IP Group = (Default) The SIP dialog is sent to the IP Group as defined in the 'Destination IP Group' (IP2IPRouting_DestIPGroupName) parameter. For more information on the actual address, see the 'Destination IP Group' parameter. ▪ **[1]** Dest Address = The SIP dialog is sent to the address configured in the following parameters: 'Destination Address', 'Destination Port' and 'Destination Transport Type'. ▪ **[2]** Request URI = The SIP dialog is sent to the address indicated in the incoming Request-URI. If the fields 'Destination Port' and 'Destination Transport Type' are configured, the incoming Request-URI parameters are overridden and these fields take precedence. ▪ **[3]** ENUM = An ENUM query is sent to include the destination address. If the fields 'Destination Port' and 'Destination Transport Type' are configured, the incoming Request-URI parameters are overridden and these fields take precedence. ▪ **[4]** Hunt Group = Used for call center survivability. For more information, see "Call Survivability for Call Centers" on page 623. ▪ **[5]** Dial Plan = The IP destination is determined by a Dial Plan index of the loaded Dial Plan file. The syntax of the Dial Plan index in the Dial Plan file is as follows: <destination / called prefix number>,0,<IP destination> Note that the second parameter "0" is ignored. An example of a configured Dial Plan (# 6) in the Dial Plan file is shown below:

Parameter	Description

	`[ PLAN6 ]`
	`200,0,10.33.8.52     ; called prefix 200 is routed to destination 10.33.8.52`
	`201,0,10.33.8.52`
	`300,0,itsp.com       ; called prefix 300 is routed to destination itsp.com`
	Once the Dial Plan is defined, you need to assign it (0 to 7) to the routing rule as the destination in the 'Destination Address' parameter, where "0" denotes [PLAN1], "1" denotes [PLAN2], and so on.
	▪ [7] LDAP = LDAP-based routing. Make sure that the Routing Policy assigned to the routing rule is configured with the LDAP Server Group for defining the LDAP server(s) to query.
	▪ [8] Gateway = The device routes the SBC call to the Tel side (Gateway call) using an IP-to-Tel routing rule in the Inbound IP Routing table (see Configuring IP to Trunk Group Routing Rules on page 476). The IP-to-Tel routing rule must be configured with the same call matching characteristics as this SBC IP-to-IP routing rule. This option is also used for alternative routing of an IP-to-IP route to the PSTN. In such a case, the IP-to-Tel routing rule must also be configured with the same call matching characteristics as this SBC IP-to-IP routing rule.
	▪ [9] Routing Server = Device sends a request to a third-party routing server for an appropriate destination (next hop) for the matching call.
	▪ [10] All Users = Device checks whether the Request-URI (i.e., destination user) in the incoming INVITE is registered in its' users' database, and if yes, it sends the INVITE to the address of the corresponding contact specified in the database. If the Request-URI is not registered, the call is rejected.

Destination IP Group `dst-ip-group-name` [IP2IPRouting_DestIPGroupName]	Defines the IP Group to where you want to route the call. The actual destination of the SIP dialog message depends on the IP Group type (as defined in the 'Type' parameter): ▪ Server-type IP Group: The SIP dialog is sent to the IP address configured for the Proxy Set that is associated with the IP Group. ▪ User-type IP Group: The device checks if the SIP dialog is from a registered user, by searching for a match between the Request-URI of the received SIP dialog and an AOR registration record in the device's database. If found, the device sends the SIP dialog to the IP address specified in the database for the registered contact. By default, no value is defined (**None**). **Notes:** ▪ The parameter is only relevant if the parameter 'Destination Type' is set to **IP Group**. ▪ The selectable IP Group for the parameter depends on the assigned SBC Routing Policy (in the 'Routing Policy' parameter in this table). For more information, see "Configuring SBC Routing Policy Rules" on page 590.
Destination SIP Interface	Defines the destination SIP Interface to where the call is sent.

Parameter	Description
`dst-srd-id` [IP2IPRouting_DestSIPInterfaceName]	By default, no value is defined (**None**). For configuring SIP Interfaces, see "Configuring SIP Interfaces" on page 333. **Notes:** ▪ The parameter is applicable only if the 'Destination Type' parameter is configured to any value other than **IP Group**. If the 'Destination Type' parameter is configured to **IP Group**, the following SIP Interface is used: ✓ Server-type IP Groups: SIP Interface that is assigned to the Proxy Set associated with the IP Group. ✓ User-type IP Groups: SIP Interface is determined during user registration with the device. ▪ For multi-tenancy, if the assigned SBC Routing Policy is not shared (i.e., the Routing Policy is associated with an Isolated SRD), the SIP Interface must be one that is associated with the Routing Policy or with a shared Routing Policy (i.e., the Routing Policy is associated with one or more Shared SRDs). If the Routing Policy is shared, the SIP Interface can be one that is associated with any SRD or Routing Policy (but it's recommended that it belong to the same SRD/Routing Policy or to shared SRD/Routing Policy to avoid "bleeding").
Destination Address `dst-address` [IP2IPRouting_DestAddress]	Defines the destination address to where the call is sent. The address can be an IP address or a domain name (e.g., domain.com). If ENUM-based routing is used (i.e., the 'Destination Type' parameter is set to **ENUM**) the parameter defines the IP address or domain name (FQDN) of the ENUM service, for example, e164.arpa, e164.customer.net or NRENum.net. The device sends the ENUM query containing the destination phone number to an external DNS server, configured in the Interface table. The ENUM reply includes a SIP URI (user@host) which is used as the destination Request-URI in this routing table. The valid value is a string of up to 50 characters (IP address or FQDN). By default, no value is defined. **Notes:** ▪ The parameter is applicable only if the 'Destination Type' parameter is set to **Dest Address** [1] or **ENUM** [3]. ▪ When using domain names, enter a DNS server IP address or alternatively, define these names in the Internal DNS table (see "Configuring the Internal SRV Table" on page 148). ▪ To terminate SIP OPTIONS messages at the device (i.e., to handle them locally), set the parameter to "internal".
Destination Port `dst-port` [IP2IPRouting_DestPort]	Defines the destination port to where the call is sent.
Destination Transport Type `dst-transport-type` [IP2IPRouting_DestTransportType]	Defines the transport layer type for sending the call: ▪ **[-1]** = (Default) Not configured - the transport type is determined by the SIPTransportType global parameter. ▪ **[0]** UDP ▪ **[1]** TCP ▪ **[2]** TLS

Parameter	Description
Call Setup Rules Set ID `call-setup-rules-set-id` [IP2IPRouting_CallSetupRulesSetId]	Assigns a Call Setup Rule Set ID to the routing rule. The device performs the Call Setup rules of this Set ID if the incoming call matches the characteristics of this routing rule. The device routes the call to the destination according to the routing rule's configured action, only after it has performed the Call Setup rules. For configuring Call Setup rules, see "Configuring Call Setup Rules" on page 283.
Group Policy `group-policy` [IP2IPRouting_GroupPolicy]	Defines whether the routing rule includes call forking. ▪ **[0]** None = (Default) Call uses only this route (even if Forking Group members are configured in the rows below it). ▪ **[1]** Forking = Call uses this route and the routes of Forking Group members, if configured (in the rows below it). **Note:** Each Forking Group can contain up to 20 members. In other words, up to 20 routing rules can be configured for the same Forking Group.
Cost Group `cost-group` [IP2IPRouting_CostGroup]	Assigns a Cost Group to the routing rule for determining the cost of the call. By default, no value is defined (**None**). For configuring Cost Groups, see "Configuring Cost Groups" on page 263. **Notes:** ▪ To implement LCR and its Cost Groups, you must enable LCR for the Routing Policy assigned to the routing rule (see "Configuring SBC Routing Policy Rules" on page 590). If LCR is disabled, the device ignores the parameter. ▪ The Routing Policy also determines whether matched routing rules that are **not** assigned Cost Groups are considered as a higher or lower cost route compared to matching routing rules that are assigned Cost Groups. For example, if the 'Default Call Cost' parameter in the Routing Policy is configured to **Lowest Cost**, even if the device locates matching routing rules that are assigned Cost Groups, the first-matched routing rule without an assigned Cost Group is considered as the lowest cost route and thus, chosen as the preferred route.

33.4 Configuring SIP Response Codes for Alternative Routing Reasons

The SBC Alternative Routing Reasons table lets you configure up to 20 SIP response codes for call release (termination) reasons. If a call (outgoing SIP dialog-initiating methods, e.g., INVITE, OPTIONS, and SUBSCRIBE messages) is released as a result of a configured SIP code (provided in SIP 4xx, 5xx, and 6xx), the device does alternative routing as follows: If the destination Proxy Set is configured with multiple IP addresses (see Configuring Proxy Sets on page 353), the device first attempts to route the call to one of these IP addresses, starting with the first listed address. If unsuccessful, the device then searches for an alternative routing rule in the IP-to-IP Routing table (see 'Configuring SBC IP-to-IP Routing Rules' on page 578).

Typically, the device performs alternative routing when there is no response at all to an INVITE message. This is done after a user-defined number of INVITE re-transmissions, configured by the SIPMaxRtx parameter. In such a scenario, the device issues itself the SIP response code 408 (Request Timeout). Alternative routing is only done if you have configured this response code in the SBC Alternative Routing Reasons table.

You can also configure alternative routing for the following proprietary response codes, if configured in the table, that are issued by the device itself:

- **805 IP Profile Call Limit:** The device generates this response code when Call Admission Control (CAC) limits (such as maximum concurrent calls) are exceeded for an IP Group (or SRD). The CAC rules are configured in the Admission Control table (see "Configuring Admission Control" on page 561). When this occurs, the device sends a SIP 480 (Temporarily Unavailable) response to the SIP entity. In such a scenario, an alternative route configured in the IP-to-IP Routing table can be used.
- **806 Media Limits Exceeded:** The device generates this response code when the call is terminated due to crossed thresholds of QoE metrics such as MOS, packet delay, and packet loss (configured in the Quality of Experience Profile table) and/or media bandwidth (configured in the Bandwidth profile table). When this occurs, the device sends a SIP 480 (Temporarily Unavailable) response to the SIP entity. This is configured by 1) assigning an IP Group a QoE and/or Bandwidth profile that rejects calls if the threshold is crossed, 2) configuring 806 in the SBC Alternative Routing Reasons table and 3) configuring an alternative routing rule.

Notes:

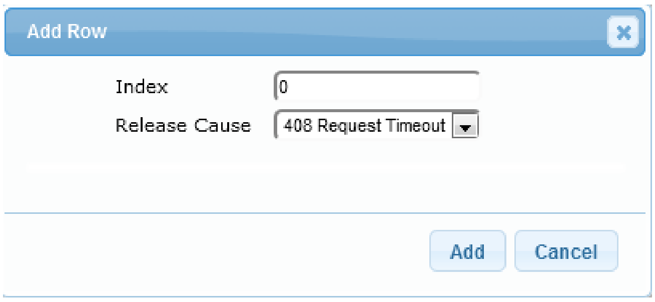
- If the device receives a SIP 408 response, an ICMP message, or no response, alternative routing is still performed even if the SBC Alternative Routing Reasons table is not configured.
- SIP requests belonging to an SRD or IP Group that have reached the call limit (maximum concurrent calls and/or call rate) as configured in the Call Admission table are sent to an alternative route if configured in the IP-to-IP Routing table for the SRD or IP Group. If no alternative routing rule is located, the device automatically rejects the SIP request with a SIP 480 (Temporarily Unavailable) response.

The following procedure describes how to configure the SBC Alternative Routing Reasons table through the Web interface. You can also configure it through ini file (SBCAlternativeRoutingReasons) or CLI (configure voip > sbc routing sbc-alt-routing-reasons).

➢ **To configure SIP reason codes for alternative IP routing:**

1. Open the SBC Alternative Routing Reasons table (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **Alternative Routing Reasons**).
2. Click **Add**; the following dialog box appears:

Figure 33-6: SBC Alternative Routing Reasons Table - Add Row Dialog Box



Index: 0

Release Cause: 408 Request Timeout

3. Configure a SIP response code for alternative routing according to the parameters described in the table below.
4. Click **Add**.

Table 33-4: SBC Alternative Routing Reasons Table Parameter Descriptions

Parameter	Description
Index [SBCAlternativeRoutingReasons_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Release Cause `rel-cause` [SBCAlternativeRoutingReasons_ReleaseCause]	Defines a SIP response code for triggering the device's alternative routing mechanism.

33.5 Configuring SBC Routing Policy Rules

The SBC Routing Policy table lets you configure up to 41 SBC Routing Policy rules. A Routing Policy determines the routing and manipulation (inbound and outbound) rules per SRD in a multiple SRD configuration topology. The Routing Policy also configures the following:

- Enables Least Cost Routing (LCR), and configures default call cost (highest or lowest) and average call duration for routing rules that are not assigned LCR Cost Groups. The default call cost determines whether matched routing rules that are not assigned Cost Groups are considered as a higher or lower cost route compared to other matching routing rules that are assigned Cost Groups. If you disable LCR, the device ignores the Cost Groups assigned to the routing rules in the IP-to-IP Routing table.
- Assigns LDAP servers (LDAP Server Group) for LDAP-based routing. IP-to-IP routing rules configured for LDAP or CSR (Call Setup Rules) queries use the LDAP server(s) that is assigned to the routing rule's associated Routing Policy. You can configure a Routing Policy per SRD or alternatively, configure a single Routing Policy that is shared between all SRDs.

The implementation of Routing Policies is intended for the following deployments **only**:

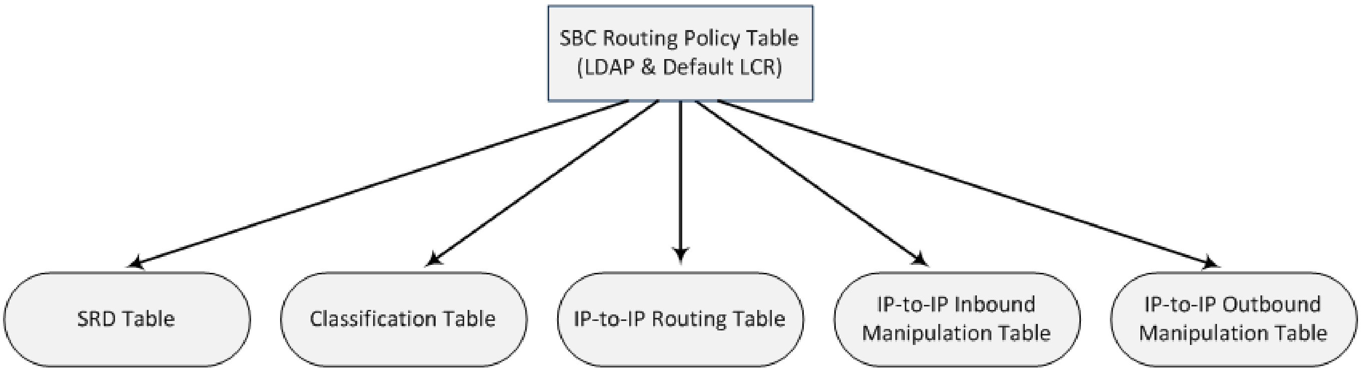
- Deployments requiring LCR and/or LDAP-based routing.
- Multi-tenant deployments that require multiple, logical routing tables where each tenant has its own dedicated ("separated") routing (and manipulation) table. In such scenarios, each SRD (tenant) is configured as an Isolated SRD and assigned its own unique Routing Policy, implementing an almost isolated, non-bleeding routing configuration topology.

For all other deployment scenarios, the Routing Policy is irrelevant and the handling of the configuration entity is not required as a default Routing Policy ("Default_SBCRoutingPolicy" at Index 0) is provided. When only one Routing Policy is required, the device automatically associates the default Routing Policy with newly added configuration entities that can be associated with the Routing Policy (as mentioned later in this section, except for Classification rules). This facilitates configuration, eliminating the need to handle the Routing Policy configuration entity (except if you need to enable LCR and/or assign an LDAP server to the Routing Policy). In such a setup, where only one Routing Policy is used, single routing and manipulation tables are employed for all SRDs.

Note: If possible, it is recommended to use only **one** Routing Policy for all SRDs (tenants), unless deployment requires otherwise (i.e., a dedicated Routing Policy per SRD).

Once configured, you need to associate the Routing Policy with an SRD(s) in the SRD table. To determine the routing and manipulation rules for the SRD, you need to assign the Routing

Policy to routing and manipulation rules. The figure below shows the configuration entities to which Routing Policies can be assigned:



Typically, assigning a Routing Policy to a Classification rule is not required, as when an incoming call is classified it uses the Routing Policy associated with the SRD to which it belongs. However, if a Routing Policy is assigned to a Classification rule, it overrides the Routing Policy assigned to the SRD. The option to assign Routing Policies to Classification rules is useful in deployments requiring different routing and manipulation rules for specific calls pertaining to the **same** SRD. In such scenarios, you need to configure multiple Classification rules for the same SRD, where for some rules no Routing Policy is assigned (i.e., the SRD's assigned Routing Policy is used) while for others a different Routing Policy is specified to override the SRD's assigned Routing Policy.

In multi-tenant environments employing multiple SRDs and Routing Policies, the IP Groups that can be used in routing rules (in the IP-to-IP Routing table) are as follows:

- If the Routing Policy is assigned to only one SRD and the SRD is an Isolated SRD, the routing rules of the Routing Policy can be configured with IP Groups belonging to the Isolated SRD and IP Groups belonging to all Shared SRDs.
- If the Routing Policy is assigned to a Shared SRD, the routing rules of the Routing Policy can be configured with any IP Group (i.e., belonging to Shared and Isolated SRDs). In effect, the Routing Policy can include routing rules for call routing between Isolated SRDs.
- If the Routing Policy is assigned to multiple SRDs (Shared and/or Isolated), the routing rules of the Routing Policy can be configured with IP Groups belonging to all Shared SRDs as well as IP Groups belonging to Isolated SRDs that are assigned the Routing Policy.

To facilitate the configuration of routing rules in the IP-to-IP Routing table through the Web interface, only the permitted IP Groups (according to the above) are displayed as optional values.

The general flow for processing the call for multi-tenant deployments and Routing Policies is as follows:

1. Using the Classification table, the device classifies the incoming call to an IP Group, based on the SIP Interface on which the call is received. Based on the SIP Interface, the device associates the call to the SRD that is assigned to the SIP Interface.
2. Once the call has been successfully classified to an IP Group, the Routing Policy assigned to the associated SRD is used. However, if a Routing Policy is configured in the Classification table, it overrides the Routing Policy assigned to the SRD.
3. The regular manipulation (inbound and outbound) and routing processes are done according to the associated Routing Policy.

Notes:

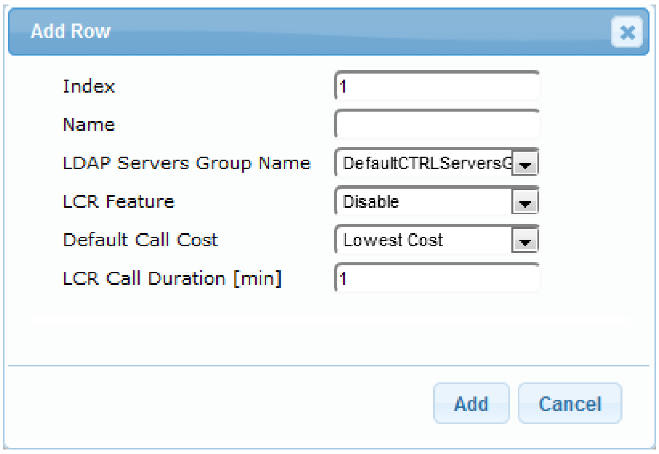
- The Classification table is used only if classification by registered user in the device's users registration database or by Proxy Set fails.
- If the device receives incoming calls (e.g., INVITE) from users that have already been classified and registered in the device's registration database, the device ignores the Classification table and uses the Routing Policy that was determined for the user during the initial classification process.

The following procedure describes how to configure SBC Routing Policy rules through the Web interface. You can also configure it through ini file (SBCRoutingPolicy) or CLI (configure voip > sbc routing sbc-routing-policy).

➢ To configure an SBC Routing Policy rule:

1. Open the SBC Routing Policy table (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **SBC Routing Policy**).
2. Click **Add**; the following dialog box appears:

Figure 33-7: SBC Routing Policy Table - Add Row Dialog Box



Index: 1
Name:
LDAP Servers Group Name: DefaultCTRLServersG
LCR Feature: Disable
Default Call Cost: Lowest Cost
LCR Call Duration [min]: 1

3. Configure the SBC Routing Policy rule according to the parameters described in the table below.
4. Click **Add**.

Table 33-5: SBC Routing Policy Table Parameter Descriptions

Parameter	Description
Index	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name name [SBCRoutingPolicy_Name]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 40 characters. By default, no name is defined. If you don't configure a name, the device automatically assigns a name in the following format: "SBCRoutingPolicy_<Index>", for example, "SBCRoutingPolicy_2". **Note:** Each row must be configured with a unique name.

Parameter	Description

LDAP Servers Group Name
`ldap-srv-group-name`
`[SBCRoutingPolicy_LdapServersGroupName]`
	Assigns an LDAP Server Group to the Routing Policy. Routing rules in the IP-to-IP Routing table that are associated with the Routing Policy and that are configured with LDAP and/or Call Setup Rules, use the LDAP server(s) configured for this LDAP Server Group.
	By default, no value is defined (**None**).
	For more information on LDAP Server Groups, see "Configuring LDAP Server Groups" on page 234.
	Note: The default SBC Routing Policy is assigned the default LDAP Server Group ("DefaultCTRLServersGroup").

LCR Feature `lcr-enable` [SBCRoutingPolicy_LCREnable]	Enables the Least Cost Routing (LCR) feature for the Routing Policy. ▪ [0] Disable (default) ▪ [1] Enable For more information on LCR, see "Least Cost Routing" on page 261.
Default Call Cost `lcr-default-cost` [SBCRoutingPolicy_LCRDefaultCost]	Defines whether routing rules in the IP-to-IP Routing table that are not assigned a Cost Group are considered a higher cost or lower cost route compared to other matched routing rules that are assigned Cost Groups. ▪ [0] Lowest Cost = (Default) The device considers a matched routing rule (belonging to the Routing Policy) that is not assigned a Cost Group as the lowest cost route. Therefore, it uses the routing rule. ▪ [1] Highest Cost = The device considers a matched routing rule (belonging to the Routing Policy) that is not assigned a Cost Group as the highest cost route. Therefore, it is only used if the other matched routing rules that are assigned Cost Groups are unavailable. **Note:** If multiple matched routing rules without an assigned Cost Group exist, the device selects the first matched rule in the table.
LCR Call Duration `lcr-call-length` [SBCRoutingPolicy_LCRAverageCallLength]	Defines the average call duration (in minutes) and is used to calculate the variable portion of the call cost. This is useful, for example, when the average call duration spans over multiple time bands. The LCR is calculated as follows: cost = call connect cost + (minute cost * average call duration). The valid value is 0-65533. The default is 1. For example, assume the following Cost Groups: ▪ "Weekend A": call connection cost is 1 and charge per minute is 6. Therefore, a call of 1 minute cost 7 units. ▪ "Weekend B": call connection cost is 6 and charge per minute is 1. Therefore, a call of 1 minute cost 7 units. Therefore, for calls under one minute, "Weekend A" carries the lower cost. However, if the average call duration is more than one minute, "Weekend B" carries the lower cost.

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34 SBC Manipulations

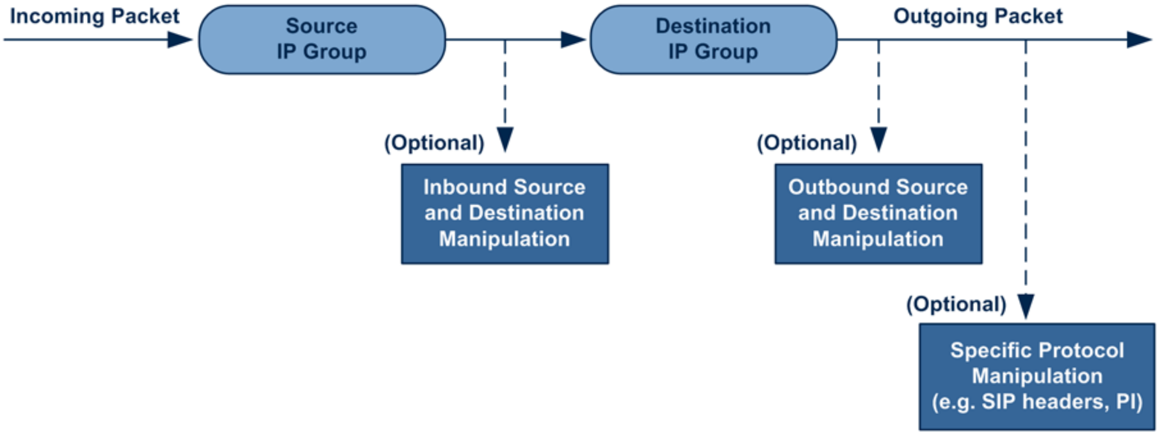
This section describes the configuration of the manipulation rules for the SBC application.

Note: For additional manipulation features, see the following:

- "Configuring SIP Message Policy Rules".
- "Configuring SIP Message Manipulation" on page 370.

The device supports SIP URI user part (source and destination) manipulations for inbound and outbound routing. These manipulations can be applied to a source IP group, source and destination host and user prefixes, and/or user-defined SIP request (e.g., INVITE, OPTIONS, SUBSCRIBE, and/or REGISTER). Since outbound manipulations are performed after routing, the outbound manipulation rule matching can also be done by destination IP Group. Manipulated destination user and host are performed on the following SIP headers: Request-URI, To, and Remote-Party-ID (if exists). Manipulated source user and host are performed on the following SIP headers: From, P-Asserted (if exists), P-Preferred (if exists), and Remote-Party-ID (if exists).

Figure 34-1: SIP URI Manipulation in IP-to-IP Routing


Incoming Packet
Source IP Group
Destination IP Group
Outgoing Packet
(Optional)
(Optional)
Inbound Source and Destination Manipulation
Outbound Source and Destination Manipulation
(Optional)
Specific Protocol Manipulation (e.g. SIP headers, PI)

You can also restrict source user identity in outgoing SIP dialogs in the Outbound Manipulation table (using the column PrivacyRestrictionMode). The device identifies an incoming user as restricted if one of the following exists:

- From header user is 'anonymous'.
- P-Asserted-Identity and Privacy headers contain the value 'id'.

All restriction logic is done after the user number has been manipulated.

Host name (source and destination) manipulations are simply host name substitutions with the names defined for the source and destination IP Groups respectively (if any, in the IP Group table).

Below is an example of a call flow and consequent SIP URI manipulations:

- **Incoming INVITE from LAN:**

```
INVITE sip:1000@10.2.2.3;user=phone;x=y;z=a SIP/2.0
Via: SIP/2.0/UDP 10.2.2.6;branch=z9hGLLLLLan
From:<sip:7000@10.2.2.6;user=phone;x=y;z=a>;tag=OlLAN;paramer1
=abe
To: <sip:1000@10.2.2.3;user=phone>
Call-ID: USELLLAN@10.2.2.3
CSeq: 1 INVITE
Contact: <sip:7000@10.2.2.3>
```

```
Supported: em,100rel,timer,replaces
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK
User-Agent: Sip Message Generator V1.0.0.5
Content-Type: application/sdp
Content-Length: 155
v=0
o=SMG 791285 795617 IN IP4 10.2.2.6
s=Phone-Call
c=IN IP4 10.2.2.6
t=0 0
m=audio 6000 RTP/AVP 8
a=rtpmap:8 pcma/8000
a=sendrecv
a=ptime:20
```

- **Outgoing INVITE to WAN:**

```
INVITE sip: 9721000@ITSP;user=phone;x=y;z=a SIP/2.0
Via: SIP/2.0/UDP 212.179.1.12;branch=z9hGWwan
From:
<sip:97000@IP_PBX;user=phone;x=y;z=a>;tag=OWan;paramer1=abe
To: <sip: 9721000@ ITSP;user=phone>
Call-ID: USEVWWAN@212.179.1.12
CSeq: 38  INVITE
Contact: <sip:7000@212.179.1.12>
Supported: em,100rel,timer,replaces
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER
User-Agent: Sip Message Generator V1.0.0.5
Content-Type: application/sdp
Content-Length: 155
v=0
o=SMG 5  9  IN IP4 212.179.1.11
s=Phone-Call
c=IN IP4 212.179.1.11
t=0 0
m=audio 8000 RTP/AVP 8
a=rtpmap:8 pcma/8000
a=sendrecv
a=ptime:20
```

The SIP message manipulations in the example above (contributing to typical topology hiding) are as follows:

- Inbound source SIP URI user name from "7000" to "97000":

```
From:<sip:7000@10.2.2.6;user=phone;x=y;z=a>;tag=OlLAN;paramer1
=abe
```

to

```
From:
<sip:97000@IP_PBX;user=phone;x=y;z=a>;tag=OWan;paramer1=abe
```

- Source IP Group name (i.e., SIP URI host name) from "10.2.2.6" to "IP_PBX":

```
From:<sip:7000@10.2.2.6;user=phone;x=y;z=a>;tag=OlLAN;paramer1
=abe
```

to

```
From:
<sip:97000@IP_PBX;user=phone;x=y;z=a>;tag=OWan;paramer1=abe
```

- Inbound destination SIP URI user name from "1000" to 9721000":

```
INVITE sip:1000@10.2.2.3;user=phone;x=y;z=a SIP/2.0
To: <sip:1000@10.2.2.3;user=phone>
```

to

```
INVITE sip:9721000@ITSP;user=phone;x=y;z=a SIP/2.0
To: <sip:9721000@ITSP;user=phone>
```

- Destination IP Group name (SIP URI host name) from "10.2.2.3" to "ITSP":

```
INVITE sip:1000@10.2.2.3;user=phone;x=y;z=a SIP/2.0
To: <sip:1000@10.2.2.3;user=phone>
```

to

```
INVITE sip:9721000@ITSP;user=phone;x=y;z=a SIP/2.0
To: <sip:9721000@ITSP;user=phone>
```

34.1 Configuring IP-to-IP Inbound Manipulations

The IP to IP Inbound Manipulation table lets you configure up to 205 IP-to-IP Inbound Manipulation rules. An IP-to-IP Inbound Manipulation rule defines a manipulation sequence for the source or destination SIP URI user part of inbound SIP dialog requests. You can apply these manipulations to different SIP dialog message types (e.g., INVITE or REGISTER) and SIP headers as follows:

- Manipulated destination URI user part are done on the following SIP headers: Request-URI, To, and Remote-Party-ID (if exists)
- Manipulated source URI user part are done on the following SIP headers: From, P-Asserted-Identity (if exists), P-Preferred-Identity (if exists), and Remote-Party-ID (if exists)

The configuration of an IP-to-IP Inbound Manipulation rule includes two areas:

- **Rule:** Defines the matching characteristics of an incoming SIP dialog (e.g., source host name).
- **Action:** Defines the operation that must be done if the incoming call matches the characteristics of the rule. In other words, the device manipulates the source or destination SIP URI user part of the SIP dialog (e.g., removes a user-defined number of characters from the left of the SIP URI user part).

To configure and apply an IP-to-IP Inbound Manipulation rule, the rule must be associated with a Routing Policy. The Routing Policy associates the rule with an SRD(s). Therefore, the Routing Policy lets you configure manipulation rules for calls belonging to specific SRD(s). However, as multiple Routing Policies are relevant only for multi-tenant deployments (if needed), for most deployments, only a single Routing Policy is required. As the device provides a default Routing Policy ("Default_SBCRoutingPolicy"), when only one Routing Policy is required, the device automatically assigns the default Routing Policy to the routing rule. If you are implementing LDAP-based routing (with or without Call Setup Rules) and/or Least Cost Routing (LCR), you need to configure these settings for the Routing Policy (regardless of the number of Routing Policies employed). For more information on Routing Policies, see "Configuring SBC Routing Policy Rules" on page 590.

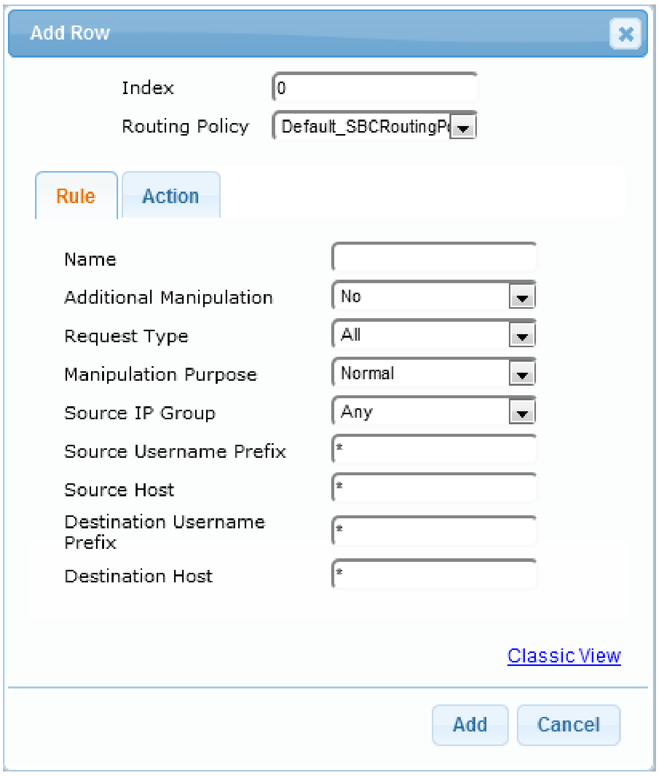
Note: The IP Group table can be used to configure a host name that overwrites the received host name. This manipulation can be done for source and destination IP Groups (see "Configuring IP Groups" on page 340).

The following procedure describes how to configure IP-to-IP Inbound Manipulation rules through the Web interface. You can also configure it through ini file (IPInboundManipulation) or CLI (configure voip > sbc manipulations ip-inbound-manipulation).

➢ To configure an IP-to-IP Inbound Manipulation rule:

1. Open the IP to IP Inbound Manipulation table (**Configuration** tab > **VoIP** menu > **SBC** > **Manipulations SBC** > **IP-to-IP Inbound**).
2. Click **Add**; the following dialog box appears:

Figure 34-2: IP to IP Inbound Manipulation Table - Add Row Dialog Box



Add Row

Index: 0
Routing Policy: Default_SBCRoutingP(

Name:
Additional Manipulation: No
Request Type: All
Manipulation Purpose: Normal
Source IP Group: Any
Source Username Prefix: *
Source Host: *
Destination Username Prefix: *
Destination Host: *

3. Configure the IP-to-IP inbound manipulation rule according to the parameters described in the table below.
4. Click **Add**.

Table 34-1: IP to IP Inbound Manipulation Table Parameter Descriptions

Parameter	Description
Index [IPInboundManipulation_Index]	Defines an index number for the new table record. **Note:** Each table row must be configured with a unique index.
Routing Policy `routing-policy-name` [IPInboundManipulation_RoutingPolicyName]	Assigns an SBC Routing Policy to the rule. The SBC Routing Policy associates the rule with an SRD(s). The SBC Routing Policy also defines default LCR settings as well as the LDAP servers if the routing rule is based on LDAP routing (and Call Setup Rules). If only one SBC Routing Policy is configured in the SBC Routing Policy table, the SBC Routing Policy is automatically assigned. If multiple SBC Routing Policies are configured, no value is assigned. For configuring SBC Routing Policies, see "Configuring SBC Routing Policy Rules" on page 590. **Note:** The parameter is mandatory.
Manipulation Name `manipulation-name`	Defines an arbitrary name to easily identify the manipulation rule.

Parameter	Description
[IPInboundManipulation_ManipulationName]	The valid value is a string of up to 40 characters. By default, no value is defined.
Matching Characteristics - Rule	
Additional Manipulation CLI: is-additional-manipulation **[IPInboundManipulation_IsAdditionalManipulation]**	Determines whether additional SIP URI user part manipulation is done for the table entry rule listed directly above it. ▪ **[0]** No = (Default) Regular manipulation rule (not done in addition to the rule above it). ▪ **[1]** Yes = If the above row entry rule matched the call, consider this row entry as a match as well and perform the manipulation specified by this rule. **Note:** Additional manipulation can only be done on a different SIP URI, source or destination, to the rule configured in the row above as configured by the 'Manipulated URI' parameter (see below).
Manipulation Purpose CLI: purpose **[IPInboundManipulation_ManipulationPurpose]**	Defines the purpose of the manipulation: ▪ **[0]** Normal = (Default) Inbound manipulations affect the routing input and source and/or destination number. ▪ **[1]** Routing input only = Inbound manipulations affect the routing input only, retaining the original source and destination number. ▪ **[2]** Shared Line = Used for the Shared-Line Appearance feature. This manipulation is for registration requests to change the destination number of the secondary extension numbers to the primary extension. For more information, see "BroadSoft's Shared Phone Line Call Appearance for SBC Survivability" on page 621.
Source IP Group CLI: src-ip-group-name **[IPInboundManipulation_SrcIpGroupName]**	Defines the IP Group from where the incoming INVITE is received. The default is **Any** (i.e., any IP Group).
Source Username Prefix CLI: src-user-name-prefix **[IPInboundManipulation_SrcUsernamePrefix]**	Defines the prefix of the source SIP URI user name (usually in the From header). The default is the asterisk (*) symbol (i.e., any source username prefix). **Note:** The prefix can be a single digit or a range of digits. For available notations, see "Dialing Plan Notation for Routing and Manipulation" on page 809.
Source Host CLI: src-host **[IPInboundManipulation_SrcHost]**	Defines the source SIP URI host name - full name (usually in the From header). The default is the asterisk (*) symbol (i.e., any host name).
Destination Username Prefix CLI: dst-user-name-prefix **[IPInboundManipulation_DestUsernamePrefix]**	Defines the prefix of the destination SIP URI user name, typically located in the Request-URI and To headers. The default is the asterisk (*) symbol (i.e., any destination username prefix). **Note:** The prefix can be a single digit or a range of digits. For available notations, see "Dialing Plan Notation for Routing and Manipulation" on page 809.

Parameter	Description
Destination Host CLI: dst-host **[IPInboundManipulation_DestHost]**	Defines the destination SIP URI host name - full name, typically located in the Request URI and To headers. The default is the asterisk (*) symbol (i.e., any destination host name).
Request Type CLI: request-type **[IPInboundManipulation_RequestType]**	Defines the SIP request type to which the manipulation rule is applied. ▪ **[0]** All = (Default) All SIP messages. ▪ **[1]** INVITE = All SIP messages except REGISTER and SUBSCRIBE. ▪ **[2]** REGISTER = Only REGISTER messages. ▪ **[3]** SUBSCRIBE = Only SUBSCRIBE messages. ▪ **[4]** INVITE and REGISTER = All SIP messages except SUBSCRIBE. ▪ **[5]** INVITE and SUBSCRIBE = All SIP messages except REGISTER.
Manipulated URI CLI: manipulated-uri **[IPInboundManipulation_ManipulatedURI]**	Determines whether the source or destination SIP URI user part is manipulated. ▪ **[0]** Source = (Default) Manipulation is done on the source SIP URI user part. ▪ **[1]** Destination = Manipulation is done on the destination SIP URI user part.
Operation Rule - Action	
Remove From Left CLI: remove-from-left **[IPInboundManipulation_RemoveFromLeft]**	Defines the number of digits to remove from the left of the user name prefix. For example, if you enter 3 and the user name is "john", the new user name is "n".
Remove From Right CLI: remove-from-right **[IPInboundManipulation_RemoveFromRight]**	Defines the number of digits to remove from the right of the user name prefix. For example, if you enter 3 and the user name is "john", the new user name is "j". **Note:** If both 'Remove From Right' and 'Leave From Right' parameters are configured, the 'Remove From Right' setting is applied first.
Leave From Right CLI: leave-from-right **[IPInboundManipulation_LeaveFromRight]**	Defines the number of characters that you want retained from the right of the user name. **Note:** If both 'Remove From Right' and 'Leave From Right' parameters are configured, the 'Remove From Right' setting is applied first.
Prefix to Add CLI: prefix-to-add **[IPInboundManipulation_Prefix2Add]**	Defines the number or string that you want added to the front of the user name. For example, if you enter 'user' and the user name is "john", the new user name is "userjohn".
Suffix to Add CLI: suffix-to-add **[IPInboundManipulation_Suffix2Add]**	Defines the number or string that you want added to the end of the user name. For example, if you enter '01' and the user name is "john", the new user name is "john01".

34.2 Configuring IP-to-IP Outbound Manipulations

The IP to IP Outbound Manipulation table lets you configure up to 205 IP-to-IP Outbound Manipulation rules. An IP-to-IP Outbound Manipulation rule defines a manipulation action for the SIP Request-URI user part (source or destination) or calling name of outbound SIP dialog requests. The IP-to-IP Outbound Manipulation rules can be applied to any SIP request type (e.g., INVITE). Manipulated destination URI user part are done on the SIP headers - Request URI, To, and Remote-Party-ID (if exists). Manipulated source URI user part are done on the SIP headers - From, P-Asserted (if exists), P-Preferred (if exists), and Remote-Party-ID (if exists).

The configuration of an IP-to-IP Outbound Manipulation rule includes two areas:

- **Rule:** Defines the matching characteristics of an incoming SIP dialog (e.g., source host name). As the device performs outbound manipulations only after the routing process, destination IP Groups can also be used as matching characteristics.
- **Action:** Defines the operation that must be done if the incoming call matches the characteristics of the rule. In other words, the device manipulates the source or destination SIP URI user part or calling name of the SIP dialog (e.g., removes a user-defined number of characters from the left of the SIP URI user part).

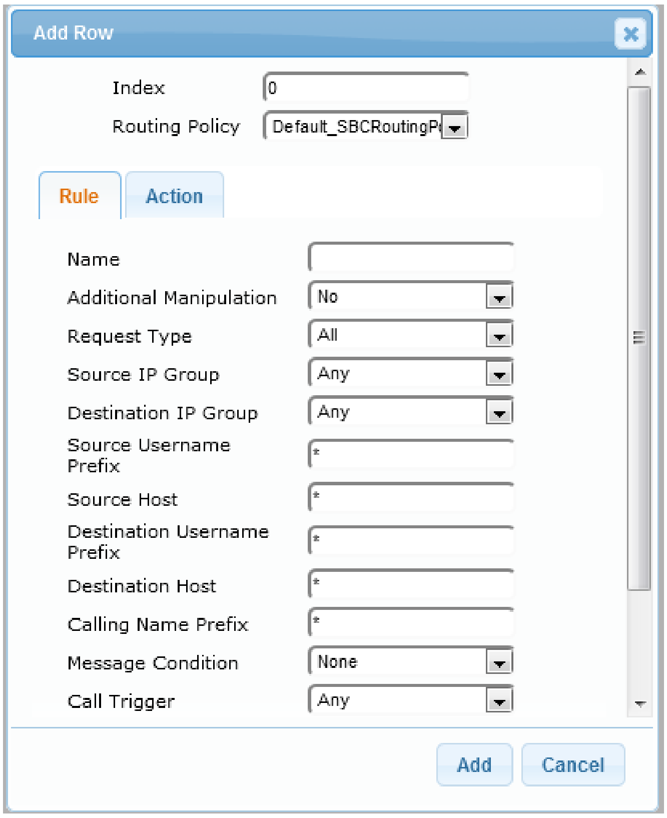
Note: SIP URI host name (source and destination) manipulations can also be configured in the IP Group table. These manipulations are simply host name substitutions with the names configured for the source and destination IP Groups, respectively.

The following procedure describes how to configure IP-to-IP Outbound Manipulation rules through the Web interface. You can also configure it through ini file (IPOutboundManipulation) or CLI (configure voip > sbc manipulations ip-outbound-manipulation).

➢ **To configure IP-to-IP outbound manipulation rules:**

1. Open the IP to IP Outbound Manipulation table (**Configuration** tab > **VoIP** menu > **SBC** > **Manipulations SBC** > **IP-to-IP Outbound**).
2. Click **Add**; the following dialog box appears:

Figure 34-3: IP to IP Outbound Manipulation Table- Add Row Dialog Box



Index: 0
Routing Policy: Default_SBCRoutingP
Name:
Additional Manipulation: No
Request Type: All
Source IP Group: Any
Destination IP Group: Any
Source Username Prefix: *
Source Host: *
Destination Username Prefix: *
Destination Host: *
Calling Name Prefix: *
Message Condition: None
Call Trigger: Any

3. Configure an IP-to-IP outbound manipulation rule according to the parameters described in the table below.
4. Click **Add**.

Table 34-2: IP to IP Outbound Manipulation Table Parameter Description

Parameter	Description
Index [IPOutboundManipulation_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Routing Policy `routing-policy-name` [IPOutboundManipulation_RoutingPolicyName]	Assigns an SBC Routing Policy to the rule. The SBC Routing Policy associates the rule with an SRD(s). The SBC Routing Policy also defines default LCR settings as well as the LDAP servers if the routing rule is based on LDAP routing (and Call Setup Rules). If only one SBC Routing Policy is configured in the SBC Routing Policy table, the SBC Routing Policy is automatically assigned. If multiple SBC Routing Policies are configured, no value is assigned. For configuring SBC Routing Policies, see "Configuring SBC Routing Policy Rules" on page 590. **Note:** The parameter is mandatory.
Manipulation Name `manipulation-name`	Defines an arbitrary name to easily identify the row.

Parameter	Description
[IPOutboundManipulation_ManipulationName]	The valid value is a string of up to 40 characters. By default, no value is defined.
Rule (Matching Characteristics)	
Additional Manipulation `is-additional-manipulation` [IPOutboundManipulation_IsAdditionalManipulation]	Determines whether additional manipulation is done for the table entry rule listed directly above it. ▪ **[0]** No = (Default) Regular manipulation rule - not done in addition to the rule above it. ▪ **[1]** Yes = If the previous table row entry rule matched the call, consider this row entry as a match as well and perform the manipulation specified by this rule. **Note:** Additional manipulation can only be done on a different item (source URI, destination URI, or calling name) to the rule configured in the row above (configured by the 'Manipulated URI' parameter).
Source IP Group `src-ip-group-name` [IPOutboundManipulation_SrcIPGroupName]	Defines the IP Group from where the INVITE is received. The default value is **Any** (i.e., any IP Group).
Destination IP Group `dst-ip-group-name` [IPOutboundManipulation_DestIPGroupName]	Defines the IP Group to where the INVITE is to be sent. The default value is **Any** (i.e., any IP Group).
Source Username Prefix `src-user-name-prefix` [IPOutboundManipulation_SrcUsernamePrefix]	Defines the prefix of the source SIP URI user name, typically used in the SIP From header. The default value is the asterisk (*) symbol (i.e., any source username prefix). The prefix can be a single digit or a range of digits. For available notations, see "Dialing Plan Notation for Routing and Manipulation" on page 809. **Note:** If you need to manipulate calls of many different source URI user names, you can use tags (see 'Source Tags' parameter below) instead of this parameter.
Source Host `src-host` [IPOutboundManipulation_SrcHost]	Defines the source SIP URI host name - full name, typically in the From header. The default value is the asterisk (*) symbol (i.e., any source host name).
Source Tags src-tags [IPOutboundManipulation_SrcTags]	Assigns a prefix tag to denote source URI user names corresponding to the tag configured in the associated Dial Plan. The valid value is a string of up to 20 characters. The tag is case insensitive. To configure prefix tags, see Configuring Dial Plans on page 607. **Note:** ▪ Make sure that you assign the Dial Plan in which you have configured the prefix tag, to the related IP Group or SRD. ▪ Instead of using tags and configuring the parameter, you can use the 'Source Username Prefix' parameter to specify a specific URI source user or all source users.

Parameter	Description
Destination Username Prefix `dst-user-name-prefix` [IPOutboundManipulation_DestUsernamePrefix]	Defines the prefix of the destination SIP URI user name, typically located in the Request-URI and To headers. The default value is the asterisk (*) symbol (i.e., any destination username prefix). The prefix can be a single digit or a range of digits. For available notations, see "Dialing Plan Notation for Routing and Manipulation" on page 809. **Note:** If you need to manipulate calls of many different destination URI user names, you can use tags (see 'Destination Tags' parameter below) instead of this parameter.
Destination Host `dst-host` [IPOutboundManipulation_DestHost]	Defines the destination SIP URI host name - full name, typically located in the Request-URI and To headers. The default value is the asterisk (*) symbol (i.e., any destination host name).
Destination Tags `dest-tags` [IPOutboundManipulation_DestTags]	Assigns a prefix tag to denote destination URI user names corresponding to the tag configured in the associated Dial Plan. The valid value is a string of up to 20 characters. The tag is case insensitive. To configure prefix tags, see Configuring Dial Plans on page 607. **Note:** ▪ Make sure that you assign the Dial Plan in which you have configured the prefix tag, to the related IP Group or SRD. ▪ Instead of using tags and configuring the parameter, you can use the 'Destination Username Prefix' parameter to specify a specific URI destination user or all destinations users.
Calling Name Prefix `calling-name-prefix` [IPOutboundManipulation_CallingNamePrefix]	Defines the prefix of the calling name (caller ID). The calling name appears in the SIP From header. The valid value is a string of up to 37 characters. By default, no prefix is defined.
Message Condition `message-condition-name` [IPOutboundManipulation_MessageConditionName]	Assigns a Message Condition rule as a matching characteristic. Message Condition rules define required SIP message formats. For configuring Message Condition rules, see "Configuring Message Condition Rules" on page 576.
Request Type `request-type` [IPOutboundManipulation_RequestType]	Defines the SIP request type to which the manipulation rule is applied. ▪ **[0]** All = (Default) all SIP messages. ▪ **[1]** INVITE = All SIP messages except REGISTER and SUBSCRIBE. ▪ **[2]** REGISTER = Only SIP REGISTER messages. ▪ **[3]** SUBSCRIBE = Only SIP SUBSCRIBE messages. ▪ **[4]** INVITE and REGISTER = All SIP messages except SUBSCRIBE. ▪ **[5]** INVITE and SUBSCRIBE = All SIP messages except REGISTER.
ReRoute IP Group `re-route-ip-group-name`	Defines the IP Group that initiated (sent) the SIP redirect response (e.g., 3xx) or REFER message. The parameter is typically used for re-routing requests (e.g., INVITEs) when

Parameter	Description
[IPOutboundManipulation_ReRouteIPGroupName]	interworking is required for SIP 3xx redirect responses or REFER messages. The default is **Any** (i.e., any IP Group). **Notes:** ▪ The parameter functions together with the 'Call Trigger' parameter (see below). ▪ For more information on interworking of SIP 3xx redirect responses or REFER messages, see "Interworking SIP 3xx Redirect Responses" on page 544 and "Interworking SIP REFER Messages" on page 546, respectively.
Call Trigger `trigger` [IPOutboundManipulation_Trigger]	Defines the reason (i.e., trigger) for the re-routing of the SIP request: ▪ **[0]** Any = (Default) Re-routed for all scenarios (re-routes and non-re-routes). ▪ **[1]** 3xx = Re-routed if it triggered as a result of a SIP 3xx response. ▪ **[2]** REFER = Re-routed if it triggered as a result of a REFER request. ▪ **[3]** 3xx or REFER = Applies to options [1] and [2]. ▪ **[4]** Initial only = Regular requests that the device forwards to a destination. In other words, re-routing of requests triggered by the receipt of REFER or 3xx does not apply.
Action	
Manipulated Item `manipulated-uri` [IPOutboundManipulation_IsAdditionalManipulation]	Defines the element in the SIP message that you want manipulated. ▪ **[0]** Source URI = (Default) Manipulates the source SIP Request-URI user part. ▪ **[1]** Destination URI = Manipulates the destination SIP Request-URI user part. ▪ **[2]** Calling Name = Manipulates the calling name in the SIP message.
Remove From Left `remove-from-left` [IPOutboundManipulation_RemoveFromLeft]	Defines the number of digits to remove from the left of the manipulated item prefix. For example, if you enter 3 and the user name is "john", the new user name is "n".
Remove From Right `remove-from-right` [IPOutboundManipulation_RemoveFromRight]	Defines the number of digits to remove from the right of the manipulated item prefix. For example, if you enter 3 and the user name is "john", the new user name is "j".
Leave From Right `leave-from-right` [IPOutboundManipulation_LeaveFromRight]	Defines the number of digits to keep from the right of the manipulated item.

Parameter	Description
Prefix to Add `prefix-to-add` [IPOutboundManipulation_Prefix2Add]	Defines the number or string to add in the front of the manipulated item. For example, if you enter 'user' and the user name is "john", the new user name is "userjohn". If you set the 'Manipulated Item' parameter to **Source URI** or **Destination URI**, you can configure the parameter to a string of up 49 characters. If you set the 'Manipulated Item' parameter to **Calling Name**, you can configure the parameter to a string of up 36 characters.
Suffix to Add `suffix-to-add` [IPOutboundManipulation_Suffix2Add]	Defines the number or string to add at the end of the manipulated item. For example, if you enter '01' and the user name is "john", the new user name is "john01". If you set the 'Manipulated Item' parameter to **Source URI** or **Destination URI**, you can configure the parameter to a string of up 49 characters. If you set the 'Manipulated Item' parameter to **Calling Name**, you can configure the parameter to a string of up 36 characters.
Privacy Restriction Mode `privacy-restriction-mode` [IPOutboundManipulation_PrivacyRestrictionMode]	Defines user privacy handling (i.e., restricting source user identity in outgoing SIP dialogs). ▪ **[0]** Transparent = (Default) No intervention in SIP privacy. ▪ **[1]** Don't change privacy = The user identity in the outgoing SIP dialog remains the same as in the incoming SIP dialog. If a restricted number exists, the restricted presentation is normalized as follows: ✓ From URL header: "anonymous@anonymous.invalid" ✓ If a P-Asserted-Identity header exists (either in the incoming SIP dialog or added by the device), a Privacy header is added with the value "id". ▪ **[2]** Restrict = The user identity is restricted. The restriction presentation is as follows: ✓ From URL header: "anonymous@anonymous.invalid" ✓ If a P-Asserted-Identity header exists (either in the incoming SIP dialog or added by the device), a Privacy header is added with the value "id". ▪ **[3]** Remove Restriction = The device attempts to reveal the user identity by setting user values in the From header and removing the privacy "id" value if the Privacy header exists. If the From header user is anonymous, the value is taken from the P-Preferred-Identity, P-Asserted-Identity, or Remote-Party-ID header (if exists). **Note:** ▪ Restriction is done only after user number manipulation (if any). ▪ The device identifies an incoming user as restricted if one of the following exists: ✓ From header user is "anonymous". ✓ P-Asserted-Identity and Privacy headers contain the value "id".

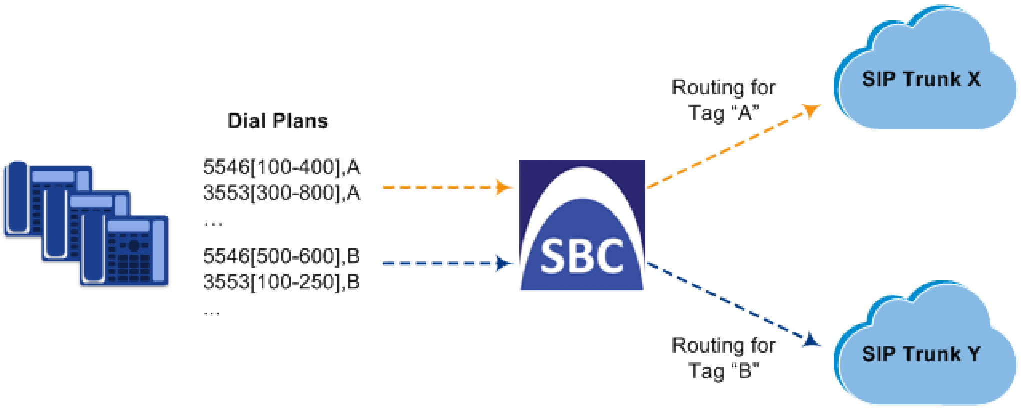
35 Configuring Dial Plans

Dial Plans let you categorize users (source and/or destination) according to source and/or destination numbers of the incoming SIP dialog-initiating requests. The device categorizes users by searching in the Dial Plan for rules that match these numbers according to prefix, suffix, and/or whole number. The categorization result in the Dial Plan is a *tag* corresponding to the matched rules. You can then use the tags to represent these users (source and/or destination users) as matching characteristics (source and/or destination tags) for the following:

- IP-to-IP Routing rules (see 'Using Dial Plan Tags for IP-to-IP Routing' on page 657)
- IP-to-IP Outbound Manipulation rules ('Using Dial Plan Tags for Outbound Manipulation' on page 616)

The figure below shows a conceptual example of routing based on tags, where users categorized as tag "A" are routed to SIP Trunk "X" and those categorized as tag "B" are routed to SIP Trunk "Y":

Figure 35-1: Routing based on Tags



Note:

- User categorization by Dial Plan is done only after the device's Classification and Inbound Manipulation processes, and before the routing process.
- Once the device successfully categorizes an incoming call by Dial Plan, it not only uses the resultant tag in the immediate routing or manipulation process, but also in subsequent routing and manipulation processes that may occur, for example, due to alternative routing or local handling of call transfer and call forwarding (SIP 3xx\REFER).
- For manipulation, tags are applicable only to outbound manipulation.

You can assign a Dial Plan to an IP Group or SRD. After Classification and Inbound Manipulation, the device checks if a Dial Plan is associated with the incoming call. It first checks the source IP Group and if no Dial Plan is assigned, it checks the SRD. If a Dial Plan is assigned to the IP Group or SRD, the device first searches the Dial Plan for a dial plan rule that matches the source number and then it searches the Dial Plan for a rule that matches the destination number. If matching dial plan rules are found, the tags configured for these rules are used in the routing and/or manipulation processes as source and/or destination tags.

The Dial Plan itself is a set of dial plan rules having the following attributes:

- **Prefix:** The prefix is matched against the source and/or destination number of the incoming SIP dialog-initiating request.
- **Tag:** The tag corresponds to the matched prefix of the source and/or destination number and is the categorization result.

You can use various syntax notations for configuring the prefix numbers in dial plan rules. You can configure the prefix as a complete number (all digits) or as a partial number using some digits and various syntax notations (patterns) to allow the device to match a dial pan rule for similar source and/or destination numbers. For more information, see the description of the 'Prefix' parameter (DialPlanRule_Prefix) described later in this section.

The device employs a "best-match" method instead of a "first-match" method to match the source/destination numbers to prefixes configured in the dial plan. The matching order is done digit-by-digit and from left to right. The numbers are first matched to the rule configured with the most constrained (specific) character set. Most constrained implies that the dial plan pattern that has the fewest possible matches for a digit is matched first. For example, if one rule contains the "x" wildcard character, which has ten possible matches (i.e., 0-9) and another rule a specific digit (e.g., 4), the rule with the specific digit is selected as the matching rule.

The best match priority is listed below in chronological order:

- Specific character (prefix)
- Number range
- "x" wildcard, which denotes any digit (0-9)
- Suffix, where the longest digits is first matched. For example, ([001-999]) takes precedence over ([01-99]) which takes precedence over ([1-9]).
- . (dot), which denotes any character

The following examples show how the best-matching method is done. Each example has two dial plan rules which are shown listed in chronological order as they would be configured in the table.

- For incoming calls with prefix number "5234", the rule with tag B is chosen (more specific for digit "4"):
```
523x,A
5234,B
```
- For incoming calls with prefix number "5234", the rule with tag B is chosen (more specific for digit "4"):
```
523x,A
523[1-9],B
```
- For incoming calls with prefix number "53211111", the rule with tag B is chosen (more specific for fourth digit):
```
532[1-9]1111,A
5321,B
```
- For incoming calls with prefix number "53124", the rule with tag B is chosen (more specific for digit "1"):
```
53([2-4]),A
531(4),B
```
- For incoming calls with prefix number "321444", the rule with tag A is chosen and for incoming calls with prefix number "32144", the rule with tag B is chosen:
```
321xxx,A
321,B
```
- For incoming calls with prefix number "53124", the rule with tag C is chosen (longest suffix - C has three digits, B two digits and A one digit):
```
53([2-4]),A
53([01-99]),B
53([001-999]),C
```

- For incoming calls with prefix number "5324", the rule with tag B is chosen (prefix is more specific for digit "4"):

  ```
  532[1-9],A
  532[2-4],B
  ```

- For incoming calls with prefix number "53124", the rule with tag B is chosen (suffix is more specific for digit "4"):

  ```
  53([2-4]),A
  53(4),B
  ```

Dial Plans are configured using two tables with parent-child type relationship:

- Parent table: Dial Plan table, which defines the name of the Dial Plan. You can configure up to five Dial Plans.
- Child table: Dial Plan Rule table, which defines the actual dial plans (rules) per Dial Plan. You can configure up to 2,000 dial plan rules in total (where all can be configured for one Dial Plan or configured between different Dial Plans).

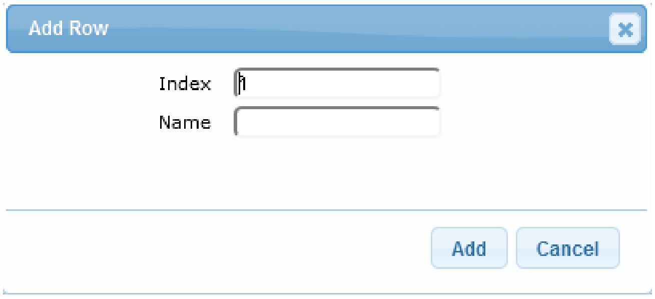
The following procedure describes how to configure Dial Plans through the Web interface. You can also configure it through other management platforms:

- **Dial Plan table:** *ini* file (DialPlans) or CLI (configure voip > sbc dial-plan)
- **Dial Plan Rule table:** *ini* file (DialPlanRule) or CLI (configure voip > sbc dial-plan-rule)

➢ To configure Dial Plans:

1. Open the Dial Plan table (**Configuration** tab > **VoIP** menu > **SBC** > **Dial Plan**).
2. Click **New**; the following dialog box appears:

Figure 35-2: Dial Plan Table - Add Row Dialog Box



Index 1

Name

3. Configure a Dial Plan name according to the parameters described in the table below.
4. Click **Apply**.

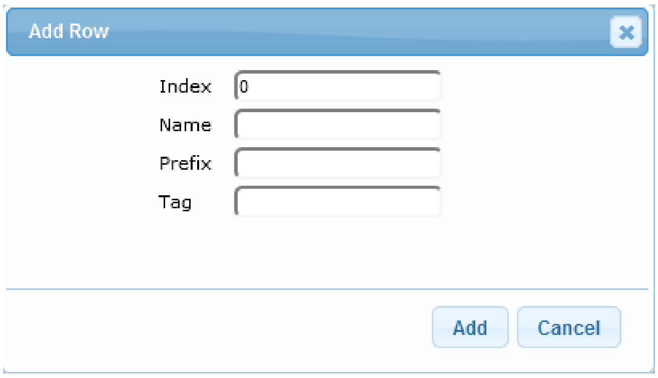
Dial Plan Table Parameter Descriptions

Parameter	Description
Index [DialPlans_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name name [DialPlans_Name]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 15 characters. **Note:** Each row must be configured with a unique name.

5. In the Dial Plan table, select the row for which you want to configure dial plan rules, and then click the **Dial Plan Rule** link located below the table; the Dial Plan Rule table appears.

6. Click **New**; the following dialog box appears:

Figure 35-3: Dial Plan Rule Table



Index 0

Name

Prefix

Tag

7. Configure a dial plan rule according to the parameters described in the table below.
8. Click **New**, and then save ("burn") your settings to flash memory.

Dial Plan Rule Table Parameter Descriptions

Parameter	Description
Index `index` [DialPlanRule_RuleIndex]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Name `name` [DialPlanRule_Name]	Defines an arbitrary name to easily identify the row. The valid value is a string of up to 15 characters.
Prefix `prefix` [DialPlanRule_Prefix]	Defines the prefix number of the source or destination number. The valid value is up to 50 characters. The following syntax can be used: ▪ **0-9:** Specific digit. ▪ **x:** Wildcard denoting any digit from 0 through 9. ▪ **z:** Denotes a number from 1 through 9. ▪ **n:** Denotes a number from 2 through 9. ▪ **a-z:** Lower-case letter. ▪ **A-Z:** Upper-case letter. ▪ ***:** (Asterisk symbol) If it is the only character in the rule, it denotes any number. To denote the asterisk "*" symbol itself, precede it with the escape "\\" character (see below). ▪ **\\:** (Backslash escape character) When it prefixes a wildcard character (*, z, n, and x), the character itself is used and not the meta-meaning. For example, "\\x" denotes the character "x", while "x" is the wildcard denoting any digits from 0-9. ▪ **#:** (Pound or hash symbol) When used at the end of the prefix it denotes the end of the number. For example, "54324#" represents a 5-digit number that starts with the digits 54324. ▪ **.:** (Period) Denotes any letter or digit. ▪ [n-m], (n-m), or ([n1-m1,n2-m2,a,b,c,n3-m3]): Represents a mixed notation of single numbers and multiple ranges. To represent the prefix, the notation is enclosed by square brackets [...]; to represent the suffix, the notation is enclosed by square brackets which are enclosed by parenthesis ([...]). For example, to denote numbers 123 through 130, 455, 766, and 780 through 790:

Parameter	Description
	✓ Prefix: [123-130,455,766,780-790] ✓ Suffix: ([123-130,455,766,780-790]) **Note:** The ranges and the single numbers in the syntax must have the same amount of digits. For example, each number range and single number in the example above consists of three digits.
Tag `tag` [DialPlanRule_Tag]	Defines a tag. The valid value is up to 16 characters. The tag is case insensitive.

35.1 Importing and Exporting Dial Plans

You can import/export Dial Plans from/to a remote server in comma-separated value (CSV) file format through the CLI:

- **Export:**
  - To export a specific Dial Plan from the device to a file:

    ```
    (config-voip)# sbc dial-plan-rule export-csv-to <Dial Plan
    name or index> <URL to CSV file>
    ```

    Example:

    ```
    # sbc dial-plan-rule export-csv-to 0
    http://10.8.8.20/upload/index_0_Dial_Plans.csv
    ```

  - To export all Dial Plans from the device to a file:

    ```
    (config-voip)# sbc dial-plan-rule export-csv-to all <URL to
    CSV file>
    ```

- **Import:**
  - To import dial plan rules from a file to a specific Dial Plan on the device:

    ```
    (config-voip)# sbc dial-plan-rule import-csv-from <Dial
    Plan name or index> <URL path to CSV file>
    ```

    The rules of the imported file replace all existing rules of the corresponding Dial Plan on the device. The Dial Plan name (or index) must exist on the device; otherwise, the Dial Plan is not imported.

    Example:

    ```
    # sbc dial-plan-rule import-csv-from 0
    http://10.8.8.20/upload/Dial_Plan_1_Rules.csv
    ```

  - To overwrite all Dial Plans on the device by importing all dial plan rules from a file:

    ```
    (config-voip)# sbc dial-plan-rule import-csv-from all <URL
    to CSV file>
    ```

    The rules of the Dial Plans in the imported file replace all existing rules of the corresponding Dial Plans on the device. For Dial Plans on the device that are not listed in the imported file, the device deletes all of their rules. For example, if the imported file contains Dial Plan 1 and the device is currently configured with Dial Plans 1 and 2, the rules of Dial Plan 1 in the imported file replace the rules of Dial Plan 1 on the device, and the rules of Dial Plan 2 on the device are deleted (the Dial Plan itself remains). The Dial Plan names in the imported file must be identical to the existing Dial Plan names on the device; otherwise, the specific Dial Plan is not imported.

For creating Dial Plans in a CSV file for import, see 'Creating Dial Plan Files for Import' on page 612.

35.2 Creating Dial Plan Files

You can configure Dial Plans in an external file (*.csv) and then import them into the device, as described in 'Importing and Exporting Dial Plans' on page 611. You can create the file using any text-based editor such as Notepad or Microsoft Excel. The file must be saved with the *.csv file name extension.

To configure Dial Plans in a file, use the following syntax:

```
<Dial Plan>,<Rule>,<Prefix>,<Tag>
```

Where:

- *Dial Plan* is the name of the Dial Plan.
- *Rule* is the name of the dial plan rule.
- *Prefix* is the source or destination number prefix
- *Tag* is the result of the user categorization and can be used as matching characteristics for routing and outbound manipulation

For example:

```
DialPlanName,Name,Prefix,Tag
PLAN1,rule_100,5511361xx,A
PLAN1,rule_101,551136184[4000-9999]#,B
MyDialPlan,My_rule_200,5511361840000#,itsp_1
MyDialPlan,My_rule_201,66666#,itsp_2
```

35.3 Using Dial Plan Tags for IP-to-IP Routing

For deployments requiring hundreds of routing rules (which may exceed the maximum number of rules that can be configured in the IP-to-IP Routing table), you can employ tags to represent the many different calling (source URI user names) and called (destination URI user names) prefix numbers in your routing rules. Tags are typically implemented when you have users of many different called and/or calling numbers that need to be routed to the same destination (e.g., IP Group or IP address). In such a scenario, instead of configuring many routing rules to match all the required prefix numbers, you need only to configure a single routing rule using the tag to represent all the possible prefix numbers.

An example scenario where employing tags could be useful is in deployments where the device needs to service calls in a geographical area that consists of hundreds of local area codes, where each area code is serviced by one of two SIP Trunks in the network. In such a deployment, instead of configuring hundreds of routing rules to represent each local area code, you can simply configure two routing rules where each is assigned a unique tag representing a group of local area codes and the destination IP Group associated with the SIP Trunk servicing them.

Note:

- Source and destination tags can be used in the same routing rule.
- The same tag can be used for source and destination tags in the same routing rule.

The following procedure describes how to configure IP-to-IP routing based on tags.

➢ To configure IP-to-IP routing based on tags:

1. In the Dial Plan table configure a Dial Plan (see 'Configuring Dial Plans' on page 607).
2. In the IP Group or SRD associated with the calls for which you want to use tag-based routing, assign the Dial Plan that you configured in Step 1.
   - IP Group table: 'Dial Plan' parameter (IPGroup_SBCDialPlanName) - see Configuring IP Groups

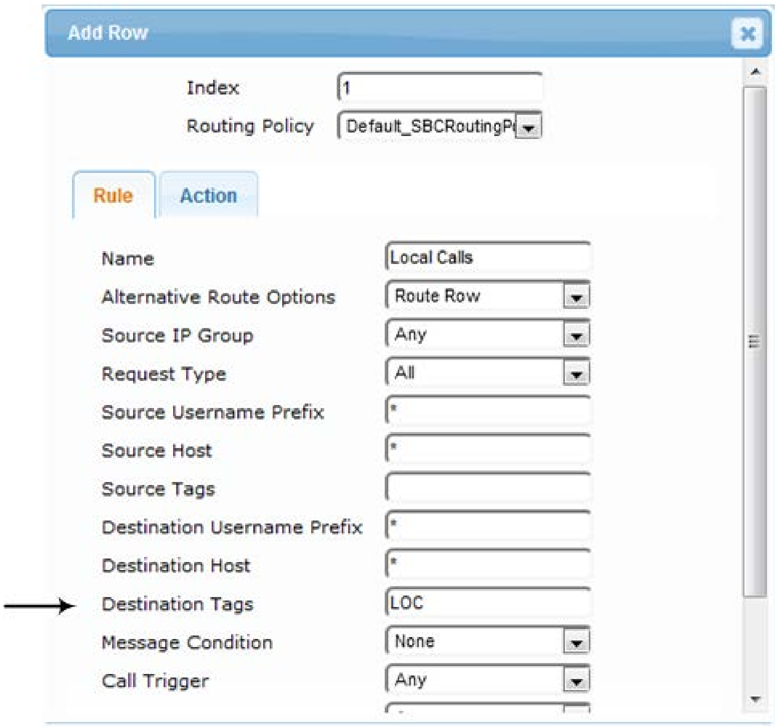
- SRD table: 'Dial Plan' parameter (SRD_SBCDialPlanName) - see Configuring SRDs

3. In the IP-to-IP Routing table (see Configuring SBC IP-to-IP Routing Rules), configure a routing rule with the required destination and whose matching characteristics include the tag(s) that you configured in your Dial Plan in Step 1. The tags are assigned under the **Rule** tab using the following parameters:

   - 'Source Tags' parameter (IP2IPRouting_SrcTags): tag denoting the calling user
   - 'Destination Tags' parameter (IP2IPRouting_DestTags): tag denoting the called user

   An example of a routing rule using a destination tag "LOC" is shown below:

Figure 35-4: Assigning Tag to Routing Rule



Index	1
Routing Policy	Default_SBCRoutingP(
Name	Local Calls
Alternative Route Options	Route Row
Source IP Group	Any
Request Type	All
Source Username Prefix	*
Source Host	*
Source Tags	
Destination Username Prefix	*
Destination Host	*
Destination Tags	LOC
Message Condition	None
Call Trigger	Any

35.3.1 Dial Plan Backward Compatibility

Note: This section is for backward compatibility **only**. It is recommended to migrate your Dial Plan configuration to the latest Dial Plan feature (see 'Using Dial Plan Tags for IP-to-IP Routing' on page 657).

Configure prefix tags in the Dial Plan file using the following syntax:

```
[ PLAN<index> ]
<prefix number>,0,<prefix tag>
```

where:

- *Index* is the Dial Plan index
- *prefix number* is the called or calling number prefix (ranges can be defined in brackets)
- *prefix tag* is the user-defined prefix tag of up to nine characters, representing the prefix number

Each prefix tag type - called or calling - must be configured in a dedicated Dial Plan index number. For example, Dial Plan 1 can be for called prefix tags and Dial Plan 2 for calling prefix tags.

The example Dial Plan file below defines the prefix tags "LOCL"and "INTL" to represent different called number prefixes for local and long distance calls:

```
[ PLAN1 ]
42520[3-5],0,LOCL
425207,0,LOCL
42529,0,LOCL
425200,0,INTL
425100,0,INTL
....
```

Note:

- Called and calling prefix tags can be used in the same routing rule.
- When using prefix tags, you need to configure manipulation rules to remove the tags before the device sends the calls to their destinations.

The following procedure describes how to configure IP-to-IP routing using prefix tags.

➢ **To configure IP-to-IP routing using prefix tags:**

1. Configure a Dial Plan file with prefix tags, and then load the file to the device.
2. Add the prefix tags to the numbers of specific incoming calls using Inbound IP-to-IP Manipulation rules:
   a. Open the IP to IP Inbound Manipulation table (**Configuration** tab > **VoIP** menu > **SBC** > **Manipulations SBC** > **IP-to-IP Inbound**), and then click **New**.
   b. Click the **Rule** tab, and then configure matching characteristics for the incoming call (e.g., set 'Source IP Group' to "1").
   c. From the 'Manipulated URI' drop-down list, select **Source** to add the tag to the calling URI user part, or **Destination** to add the tag to the called URI user part.
   d. Click the **Action** tab, and then enter the Dial Plan index for which you configured your prefix tag, in the 'Prefix to Add' or 'Suffix to Add' fields, using the following syntax: $DialPlan<x>, where *x* is the Dial Plan index (0 to 7). For example, if the called number is 4252000555, the device manipulates it to LOCL4252000555.

3. Add an SBC IP-to-IP routing rule using the prefix tag to represent the different source or destination URI user parts:

   a. Open the IP-to-IP Routing table (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **IP-to-IP Routing Table**), and then click **New**.

   b. Click the **Rule** tab, and then enter the prefix tag in the 'Source Username Prefix' or 'Destination Username Prefix' fields (e.g., "LOCL", without the quotes).

   c. Continue configuring the rule as required.

4. Configure a manipulation rule to remove the prefix tags before the device sends the message to the destination:

   a. Open the IP to IP Outbound Manipulation table (**Configuration** tab > **VoIP** menu > **SBC** > **Manipulations SBC** > **IP-to-IP Outbound**), and then click **New**.

   b. Click the **Rule** tab, and then configure matching characteristics for the incoming call (e.g., set 'Source IP Group' to "1"), including calls with the prefix tag (in the 'Source Username Prefix' or 'Destination Username Prefix' fields, enter the prefix tag to remove).

   c. Click the **Action** tab, and then in the 'Remove from Left' or 'Remove from Right' fields (depending on whether you added the tag at the beginning or end of the URI user part, respectively), enter the number of characters making up the tag.

35.4 Using Dial Plan Tags for Outbound Manipulation

You can use Dial Plan tags to denote source and/or destination URI user names in Outbound Manipulation rules in the IP-to-IP Outbound Manipulation table.

The following procedure describes how to configure Outbound Manipulation based on tags.

➢ **To configure Outbound Manipulation based on tags:**

1. In the Dial Plan table configure a Dial Plan (see 'Configuring Dial Plans' on page 607).
2. In the IP Group or SRD associated with the calls for which you want to use tag-based routing, assign the Dial Plan that you configured in Step 1.
    - IP Group table: 'Dial Plan' parameter (IPGroup_SBCDialPlanName) - see Configuring IP Groups
    - SRD table: 'Dial Plan' parameter (SRD_SBCDialPlanName) - see Configuring SRDs
3. In the Outbound Manipulations table (see Configuring IP-to-IP Outbound Manipulations), configure a rule with the required manipulation and whose matching characteristics include the tag(s) that you configured in your Dial Plan in Step 1. The tags are assigned using the following parameters:
    - 'Source Tags' parameter (IPOutboundManipulation_SrcTags): tag denoting the calling users
    - 'Destination Tags' parameter (IPOutboundManipulation_DestTags): tag denoting the called users

36 Advanced SBC Features

36.1 Configuring Call Preemption for SBC Emergency Calls

The device supports emergency call preemption for SBC calls, by prioritizing emergency calls over regular calls. If the device receives an incoming emergency call when there are unavailable resources to process the call, the device preempts one of the active regular calls to free up resources for sending the emergency call to its' destination (i.e., emergency service provider), and not reject it. The device may preempt more than one active call in order to provide sufficient resources for processing the emergency call. Available resources depends on the number of INVITE messages currently being processed by the device.

If the device preempts a call, it disconnects the call as follows:

- If the call is being setup (not yet established), it sends a SIP 488 response to the incoming leg and a SIP CANCEL message to the outgoing leg.
- If the call is already established, it sends a SIP BYE message to each leg. The device includes in the SIP BYE message, the Reason header describing the cause as "preemption".

Once the device terminates the regular call, it immediately sends the INVITE message of the emergency call to its' destination without waiting for any response from the remote sides (e.g., 200 OK after BYE). If the device is unable to preempt a call for the emergency call, it rejects the emergency call with a SIP 503 "Emergency Call Failed" (instead of "Service Unavailable") response.

For the device to identify incoming calls as emergency calls, you need to configure a Message Condition rule. Below are examples of Message Condition rules, configured in the Message Condition table, for identifying emergency calls:

- Indices 0 and 1: SIP Resource-Priority header contains a string indicating an emergency call.
- Indices 2 to 4: Destination user-part contains the emergency provider's address.

Figu

Table 36-1: Examples of Message Condition Rules for Emergency Calls

Index	Name	Condition
0	Emergency1 - RP header	header.resource-priority contains 'emergency'
1	Emergency2 - RP header	header.resource-priority contains 'esnet'
2	Emergency1 - user with providers address	header.to.url.user=='911'
3	Emergency2 - user with providers address	header.to.url.user=='100' ||header.to.url.user=='101'||header.to.url.user=='102'
4	Emergency3 - user with providers address	header.request.uri contains 'urn:service:sos'

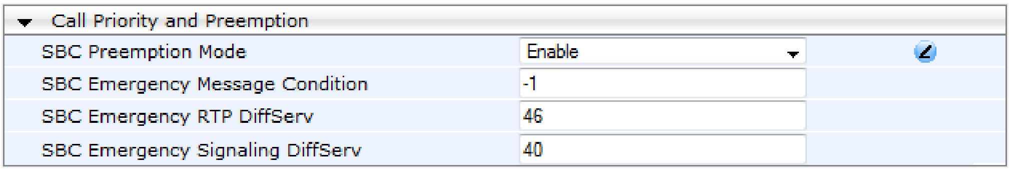
The device applies the Message Condition rule only after call classification (but, before inbound manipulation).

➢ **To configure SBC emergency call preemption:**

1. In the Message Condition table, configure a Message Condition rule to identify incoming emergency calls. See above for examples. For more information on Message Conditions, see "Configuring Message Condition Rules" on page 576.

2. Open the SBC General Settings page (**Configuration** tab > **VoIP** > **SBC** > **SBC General Settings**), and then scroll down the page to the Call Priority and Preemption group:

Figure 36-1: Configuring Emergency SBC Call Preemption



Call Priority and Preemption

SBC Preemption Mode	Enable
SBC Emergency Message Condition	-1
SBC Emergency RTP DiffServ	46
SBC Emergency Signaling DiffServ	40

3. From the 'SBC Preemption Mode' drop-down list (SBCPreemptionMode), select **Enable** to enable the SBC call preemption feature.
4. In the 'SBC Emergency Message Condition' field, enter the row index of the Message Condition rule that you configured in Step 1 for identifying incoming emergency calls.
5. (Optional) Assign DiffServ levels (markings) to packets belonging to emergency calls:
   a. In the 'SBC Emergency RTP DiffServ' field (SBCEmergencyRTPDiffServ), enter the QoS level for RTP packets.
   b. In the 'SBC Emergency Signaling DiffServ' field (SBCEmergencySignalingDiffServ), enter the QoS level for SIP signaling packets.
6. Click **Submit**.

- **Note:** The device does not preempt already established emergency calls.

36.2 Emergency Call Routing using LDAP to Obtain ELIN

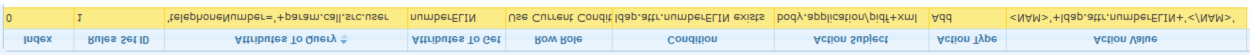
The device can route emergency calls (e.g., 911) for INVITE messages that are received without an ELIN number. This is in contrast to when the device is deployed in a Microsoft Lync environment, whereby INVITE messages received from Lync contain ELIN numbers. (For a detailed explanation on ELIN numbers and handling of emergency calls by emergency server providers, see "Enhanced 9-1-1 Support for Lync Server" on page 289.)

To obtain an ELIN number for emergency calls received without ELINs, you can configure the device to query an LDAP server for the 911 caller's ELIN number. The device adds the resultant ELIN number and a Content-Type header for the PIDF XML message body to the outgoing INVITE message, for example:

```
Content-Type: application/pidf+xml
            <NAM>1234567890</NAM>
```

To enable this functionality, you need to configure a Call Setup rule in the Call Setup Rules table (see "Configuring Call Setup Rules" on page 283). The following example shows a Call Setup rule that queries an Active Directory (AD) server for the attribute "telephoneNumber" whose value is the E9-1-1 caller's number, and then retrieves the user's ELIN number from the attribute, "numberELIN":

Figure 36-2: Example of Call Setup Rule for LDAP Query of ELIN



Index	Rules Set ID	Attributes To Query	Attributes To Get	Row Role	Condition	Action Subject	Action Type	Action Value
0	1	'telephoneNumber='+param.call.src.user	numberELIN	Use Current Condit	ldap.attr.numberELIN exists	body.application/pidf+xml	Add	'<NAM>'+ldap.attr.numberELIN+'</NAM>'

The rest of the process is similar to emergency call routing in a Lync environment.

Configuration includes the following:

- Enabling E9-1-1 by configuring the 'PSAP Mode' parameter to **PSAP Server** in the IP Group table for the IP Group of the PSAP server (see "Enabling the E9-1-1 Feature" on page 299).
- Configuring routing rules in the IP-to-IP Routing table for routing between the emergency callers' IP Group and the PSAP server's IP Group. The only special configuration is required on the rule for routing from emergency callers to the PSAP server:
  - Configure the emergency number (e.g., 911) in the 'Destination Username Prefix' field.
  - Assign the Call Setup rule, which you configured for obtaining the ELIN number from the AD, in the 'Call Setup Rules Set ID' field (see "Confiiguring SBC IP-to-IP Routing Rule for E9-1-1" on page 300).

36.3 Call Forking

This section describes various Call Forking features supported by the device.

36.3.1 Initiating SIP Call Forking

The SBC device supports call forking of an incoming call to multiple SBC users (destinations). Call forking is supported by the device's capability of registering multiple SIP client user phone contacts (mobile and fixed-line extensions) under the same Address of Record (AOR) in its registration database. This feature can be implemented in the following example scenarios:

- An enterprise Help Desk, where incoming customer calls are simultaneously sent to multiple customer service agent extensions.
- An employee's phone devices, where the incoming call is simultaneously sent to multiple devices (e.g., to the employee's office phone and mobile SIP phone).
- An enterprise reception desk, where an incoming call is simultaneously sent to multiple receptionists.

The device supports various modes of call forking. For example, in Parallel call forking mode, the device sends the INVITE message simultaneously to all the users registered under the same AOR, resulting in the ringing of all extensions; the first extension to pick up the call receives the call, and all other extensions stop ringing. The Call Forking feature is configured by creating a User-type IP Group and configuring the IP Group table's parameter, 'SBC Client Forking Mode' (see "Configuring IP Groups" on page 340).

The device can also fork INVITE messages received for a Request-URI of a specific contact (user), belonging to the destination IP Group User-type, registered in the database to all other users located under the same AOR as the specific contact. This is configured using the SBCSendInviteToAllContacts parameter.

36.3.2 SIP Forking Initiated by SIP Proxy Server

The device can handle the receipt of multiple SIP 18x responses as a result of SIP forking initiated by a proxy server. This occurs when the device sends an INVITE, received from a user agent (UA), to a proxy server and the proxy server then forks the INVITE request to multiple UAs. Several UAs may answer and the device may therefore, receive several replies (responses) for the single INVITE request. Each response has a different 'tag' value in the SIP To header.

During call setup, forked SIP responses may result in a single SDP offer with two or more SDP answers. The device "hides" all the forked responses from the INVITE-initiating UA, except the first received response ("active" UA) and it forwards only subsequent requests and responses from this active UA to the INVITE-initiating UA. All requests/responses from the other UAs are handled by the device; SDP offers from these UAs are answered with an "inactive" media.

The device supports two forking modes, configured by the SBCForkingHandlingMode parameter:

- **Latch On First:** The device forwards only the first received 18x response to the INVITE-initiating UA and disregards subsequently received 18x forking responses (with or without SDP).
- **Sequential:** The device forwards all 18x responses to the INVITE-initiating UA, sequentially (one after another). If 18x arrives with an offer only, only the first offer is forwarded to the INVITE-initiating UA.

The device also supports media synchronization for call forking. If the active UA is the first one to send the final response (e.g., 200 OK), the call is established and all other final responses are acknowledged and a BYE is sent if needed. If another UA sends the first final response, it is possible that the SDP answer that was forwarded to the INVITE-initiating UA is irrelevant and thus, media synchronization is needed between the two UAs. Media synchronization is done by sending a re-INVITE request immediately after the call is established. The re-INVITE is sent without an SDP offer to the INVITE-initiating UA. This causes the INVITE-initiating UA to send an offer which the device forwards to the UA that confirmed the call. Media synchronization is enabled by the EnableSBCMediaSync parameter.

36.3.3 Call Forking-based IP-to-IP Routing Rules

You can configure call forking routing rules in the IP-to-IP Routing table. This is done by configuring multiple routing rules under a forking group. These rules send an incoming IP call to multiple destinations of any type (e.g., IP Group or IP address). The device forks the call by sending simultaneous INVITE messages to all the specified destinations. It handles the multiple SIP dialogs until one of the calls is answered and then terminates the other SIP dialogs. For more information, see "Configuring SBC IP-to-IP Routing Rules" on page 578.

36.4 Call Survivability

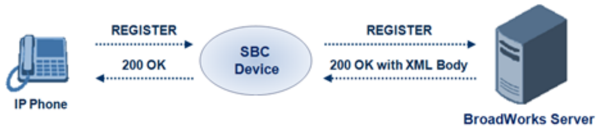
This section describes various call survivability features supported by the SBC device.

36.4.1 Auto-Provisioning of Subscriber-Specific Information for BroadWorks Server for Survivability

This feature enables SBC user registration for interoperability with BroadSoft BroadWorks server to provide call survivability in case of connectivity failure with the BroadWorks server, for example, due to a WAN failure. This feature enables local users to dial a local extension (or any other configured alias) that identifies another local user, in survivability mode. This feature is enabled using the SBCExtensionsProvisioningMode parameter.

In normal operation, when subscribers (such as IP phones) register to the BroadWorks server through the device, the device includes the SIP Allow-Events header in the sent REGISTER message. In response, the BroadWorks server sends the device a SIP 200 OK containing an XML body with subscriber information such as extension number, phone number, and URIs (aliases). The device forwards the 200 OK to the subscriber (without the XML body).

Figure 36-3: Interoperability with BroadWorks Registration Process



The device saves the users in its registration database with their phone numbers and extensions, enabling future routing to these destinations during survivability mode. When in survivability mode, the device routes the call to the Contact associated with the dialed phone number or extension number in the registration database.

Below is an example of an XML body received from the BroadWorks server:

```
<?xml version="1.0" encoding="utf-8"?>
  <BroadsoftDocument version="1.0" content="subscriberData">
    <phoneNumbers>
      <phoneNumber>2403645317</phoneNumber>
      <phoneNumber>4482541321</phoneNumber>
    </phoneNumbers>
    <aliases>
      <alias>sip:bob@broadsoft.com</alias>
      <alias>sip:rhughes@broadsoft.com</alias>
    </aliases>
    <extensions>
      <extension>5317</extension>
      <extension>1321</extension>
    </extensions>
  </BroadSoftDocument>
```

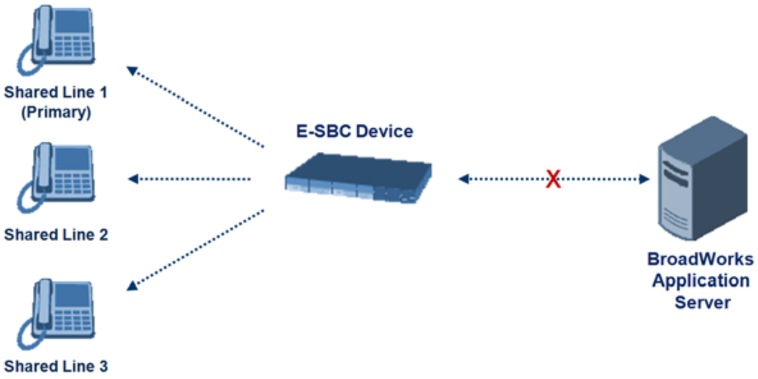
36.4.2 BroadSoft's Shared Phone Line Call Appearance for SBC Survivability

The device can provide redundancy for BroadSoft's Shared Call Appearance feature. When the BroadSoft application server switch (AS) fails or does not respond, or when the network connection between the device and the BroadSoft AS is down, the device manages the Shared Call Appearance feature for the SIP clients.

This feature is supported by configuring a primary extension and associating it with secondary extensions (i.e., *shared lines*) so that incoming calls to the primary extension also ring at the secondary extensions. The call is established with the first extension to answer the call and consequently, the ringing at the other extensions stop. For example, assume primary extension number 600 is shared with secondary extensions 601 and 602. In the case of an incoming call to 600, all three phone extensions ring simultaneously, using the device's call forking feature as described in "SIP Forking Initiated by SIP Proxy Server" on page 619.

Note that incoming calls specific to extensions 601 or 602 ring only at these specific extensions.

Figure 36-4: Call Survivability for BroadSoft's Shared Line Appearance



To configure this capability, you need to configure a shared-line, inbound manipulation rule for registration requests to change the destination number of the secondary extension numbers (e.g. 601 and 602) to the primary extension (e.g., 600). Call forking must also be enabled. The following procedure describes the main configuration required.

Notes:

- The device enables outgoing calls from all equipment that share the same line simultaneously (usually only one simultaneous call is allowed per a specific shared line).
- You can configure whether REGISTER messages from secondary lines are terminated on the device or forwarded transparently (as is), using the SBCSharedLineRegMode parameter.
- The LED indicator of a shared line may display the wrong current state.

➢ To configure the Shared Line feature:

1. In the IP Group table (see "Configuring IP Groups" on page 340), add a Server-type IP Group for the BroadWorks server.
2. In the IP Group table, add a User-type IP Group for the IP phone users and set the 'SBC Client Forking Mode' parameter to **Parallel** so that the device forks incoming calls to all contacts under the same AOR registered in the device's registration database.
3. In the IP-to-IP Routing table (see "Configuring SBC IP-to-IP Routing Rules" on page 578), add a rule for routing calls between the above configured IP Groups.
4. In the IP to IP Inbound Manipulation table (see "Configuring IP-to-IP Inbound Manipulations" on page 597), add a manipulation rule for the secondary extensions (e.g., 601 and 602) so that they also register to the device's database under the primary extension contact (e.g., 600):
   - Set the 'Manipulation Purpose' field to **Shared Line**.
   - Set the 'Source IP Group' field to the IP Group that you created for the users (e.g., 2).
   - Set the 'Source Username Prefix' field to represent the secondary extensions (e.g., 601 and 602).
   - Set the 'Manipulated URI' field to **Source** to manipulate the source URI.
   - Set the 'Remove From Right' field to "1" to remove the last digit of the extensions (e.g., 601 is changed to 60).
   - Set the 'Suffix to Add' field to "0" to add 0 to the end of the manipulated number (e.g., 60 is changed to 600).

36.4.3 Call Survivability for Call Centers

The device supports call survivability for call centers. When a communication failure (e.g., in the network) occurs with the remote voice application server responsible for handling the call center application (such as IVR), the device routes the incoming calls received from the customer (i.e., from the TDM gateway) to the call center agents.

In normal operation, the device registers the agents in its users registration database. Calls received from the TDM gateway are forwarded by the device to the application server, which processes the calls and sends them to specific call center agents, through the device. Upon a failure with the application server, the device routes the calls from the TDM Gateway to the agents. The device routes the call to the first available user it founds. If the call is not answered by the user, the device routes it to the next available user. The SBC can handle a sequence of up to five users, after which the session is timed out and the call is dropped.

Figure 36-5: Normal Operation in Call Center Application

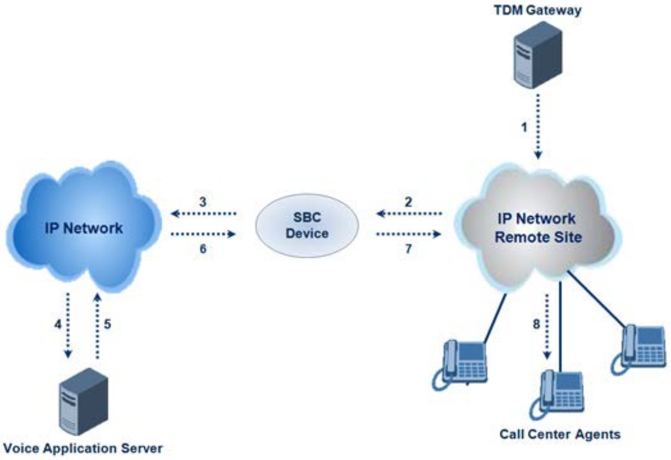
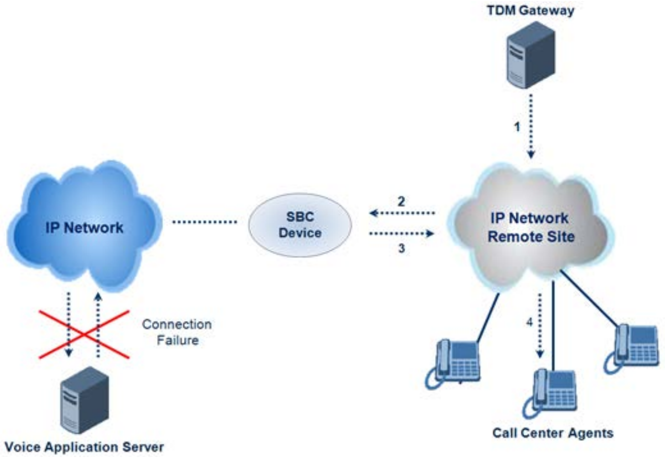


Figure 36-6: Call Survivability for Call Center



➢ **To configure call survivability for a call center application:**

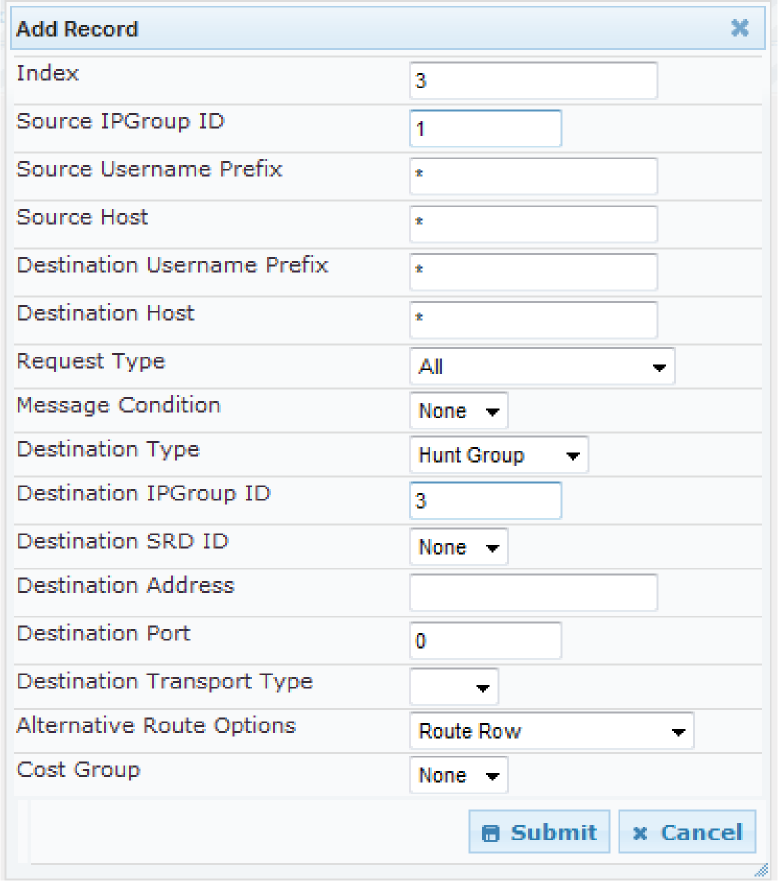
1. In the IP Group table (see "Configuring IP Groups" on page 340), add IP Groups for the following entities:
   - TDM Gateway (Server-type IP Group). This entity forwards the customer calls through the device to the Application server.
   - Application server (Server-type IP Group). This entity processes the call and sends the call through the device to the specific call center agent located on a different network (remote).

   - Call center agents (User-type IP Group). You can configure multiple IP Groups to represent different groups of call center agents, for example, agents and managers.

2. In the Classification table (see "Configuring Classification Rules" on page 569), add rules to classify incoming calls that are received from the entities listed in Step 1, to IP Groups.
3. In the SBC IP-to-IP Routing table (see "Configuring SBC IP-to-IP Routing Rules" on page 578), add the following IP-to-IP routing rules:
   - For normal operation:
     - Routing from TDM Gateway to Application server.
     - Routing from Application server to call center agents.
   - For call survivability mode: Routing from TDM Gateway to call center agents. This configuration is unique due to the following settings:
     - The 'Source IP Group' field is set to the IP Group of the TDM Gateway.
     - The 'Destination Type' field is set to **Hunt Group**, which is specifically used for call center survivability.
     - The 'Destination IP Group' field is set to the IP Group of the call center agents.

   The figure below displays a routing rule example, assuming IP Group "1" represents the TDM Gateway and IP Group "3" represents the call center agents:

Figure 36-7: Routing Rule Example for Call Center Survivability



Index	3
Source IPGroup ID	1
Source Username Prefix	*
Source Host	*
Destination Username Prefix	*
Destination Host	*
Request Type	All
Message Condition	None
Destination Type	Hunt Group
Destination IPGroup ID	3
Destination SRD ID	None
Destination Address	
Destination Port	0
Destination Transport Type	
Alternative Route Options	Route Row
Cost Group	None

36.4.4 Survivability Mode Display on Aastra IP Phones

If the SBC device is deployed in an Enterprise network with Aastra IP phones and connectivity with the WAN fails, the device provides call survivability by enabling communication between IP phone users within the LAN enterprise. In such a scenario, the device can be configured to notify the IP phones that it is currently operating in Survivability mode. When this occurs, the Aastra IP phones display the message, "StandAlone Mode" on their LCD screens. This feature is enabled by setting the SBCEnableSurvivabilityNotice parameter to 1.

When this feature is enabled and the SBC device is in Survivability mode, it responds to SIP REGISTER messages from the IP phones with a SIP 200 OK containing the following XML body:

```
Content-Type: application/xml
<?xml version="1.0" encoding="utf-8"?>
<LMIDocument version="1.0">
<LocalModeStatus>
   <LocalModeActive>true</LocalModeActive>
   <LocalModeDisplay>StandAlone Mode</LocalModeDisplay>
</LocalModeStatus>
</LMIDocument>
```

36.5 Alternative Routing on Detection of Failed SIP Response

The device can detect failure of a sent SIP response (e.g., TCP timeout, and UDP ICMP). In such a scenario, the device re-sends the response to an alternative destination. This support is in addition to alternative routing if the device detects failed SIP requests.

For example, assume the device sends a SIP 200 OK in response to a received INVITE request. If the device does not receive a SIP ACK in response to this, it sends a new 200 OK to the next alternative destination. This new destination can be the next given IP address resolved from a DNS from the Contact or Record-Route header in the request related to the response.

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Part VII

Cloud Resilience Package

37 CRP Overview

The device's Cloud Resilience Package (CRP) application enhances cloud-based or hosted communications environments by ensuring survivability, high voice quality and security at enterprise branch offices and cloud service customer premises. CRP is designed to be deployed at customer sites and branches of:

- Cloud-based and hosted communications
- Cloud-based or hosted contact-center services
- Distributed PBX or unified communications deployments

The CRP application is based on the functionality of the SBC application, providing branch offices with call routing and survivability support similar to AudioCodes' Stand-Alone Survivability (SAS) application. CRP is implemented in a network topology where the device is located at the branch office, routing calls between the branch users, and/or between the branch users and other users located elsewhere (at headquarters or other branch offices), through a hosted server (IP PBX) located at the Enterprise's headquarters. The device maintains call continuity even if a failure occurs in communication with the hosted IP PBX. It does this by using its Call Survivability feature, enabling the branch users to call one another or make external calls through the device's PSTN gateway interface (if configured).

Notes:

- The CRP application is available only if the device is installed with a Software License Key that includes this feature. For installing a Software License Key, see "Software License Key" on page 668.
- For the maximum number of supported CRP sessions and CRP users than can be registered in the device's registration database, see "Technical Specifications" on page 1035.
- The CRP application supersedes the SAS application and is the recommended application to use. However, SAS is still supported by the device. For a detailed description on SAS, refer to the *SAS Application Configuration Guide*.

For cloud providers, CRP ensures uninterrupted communications in the event of lost connection with the cloud providers' control systems. For distributed enterprises and contact centers, CRP is an essential solution for enterprises deploying geographically distributed communications solutions or distributed call centers with many branch offices. CRP ensures the delivery of internal and external calls even when the connection with the centralized control servers is lost.

Table 37-1: Key Features

Survivability	Quality of Experience/Service	Security
▪ PSTN fallback ▪ WAN redundancy ▪ Local mode ▪ High availability ▪ Emergency calling (E911) ▪ Basic call routing between registering users and device, or any other route to responding server ▪ Short number dialog (short numbers are learned	▪ QoE monitoring ▪ Call Admission Control ▪ SLA fulfillment ▪ SIP mediation ▪ Media transcoding ▪ Test call agent	▪ Layer 3 to 7 protection ▪ Media encryption ▪ Call control encryption ▪ NAT traversal ▪ Topology hiding

Survivability	Quality of Experience/Service	Security
dynamically in the registration process) ▪ Survivability indication to IP phone ▪ Call hold and retrieve ▪ Call transfer (if IP phone initiates REFER) ▪ Basic Shared Line Appearance (excluding correct busy line indications) ▪ Call waiting (if supported by IP phone)		

One of the main advantages of CRP is that it enables quick-and-easy configuration. This is accomplished by its pre-configured routing entities, whereby only minimal configuration is required. For example, defining IP addresses to get the device up and running and deployed in the network.

38 CRP Configuration

This section describes configuration specific to the CRP application. As CRP has similar functionality to the SBC application, for configuration that is common to the SBC, which is not covered in this section, see the following SBC sections:

- "Configuring General SBC Settings" on page 557
- "Configuring Admission Control" on page 561
- "Configuring Allowed Audio Coder Groups" on page 565
- "Configuring Classification Rules" on page 569
- "Configuring Message Condition Rules" on page 576
- "Configuring SBC IP-to-IP Routing Rules" on page 578
- "Configuring SIP Response Codes for Alternative Routing Reasons" on page 588
- "Configuring IP-to-IP Inbound Manipulations" on page 597
- "Configuring IP-to-IP Outbound Manipulations" on page 601

Note: The main difference in the common configuration between the CRP and SBC applications is the navigation menu paths to opening these Web configuration pages. Wherever "SBC" appears in the menu path, for the CRP application it appears as "CRP".

38.1 Enabling the CRP Application

Before you can start configuring the CRP, you must first enable the CRP application. Once enabled, the Web interface displays the menus and parameter fields relevant to the CRP application.

Note: The CRP feature is available only if the device is installed with a Software License Key that includes this feature. For installing a Software License Key, see "Software License Key" on page 668.

➢ **To enable the CRP application:**

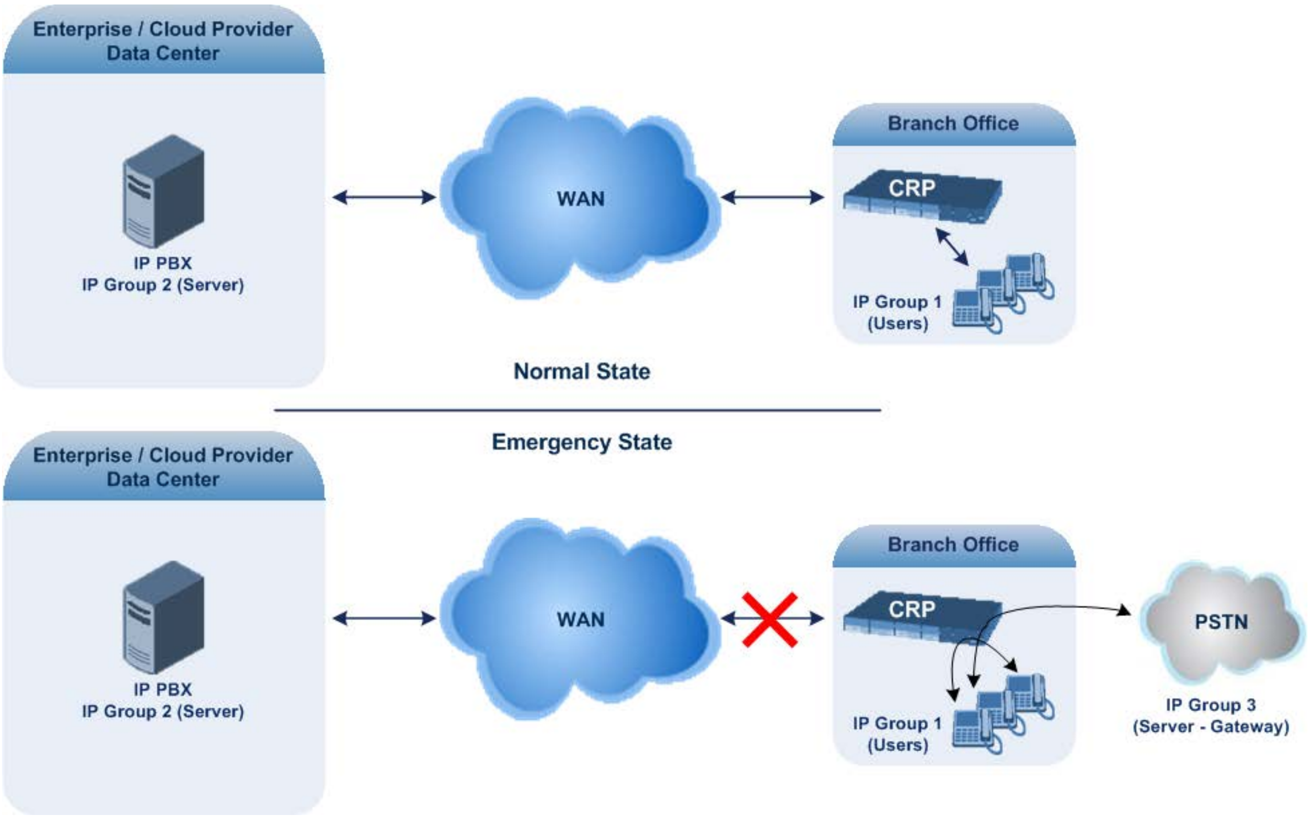
1. Open the Applications Enabling page (**Configuration** tab > **VoIP** menu > **Applications Enabling** > **Applications Enabling**).
2. From the 'CRP Application' drop-down list, select **Enable**.
3. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

38.2 Configuring Call Survivability Mode

The CRP can be configured to operate in one of the following call survivability modes:

- **Normal (Default):** The CRP interworks between the branch users and the IP PBX located at headquarters. The CRP forwards all requests (such as for registration) from the branch users to the IP PBX, and routes the calls based on the IP-to-IP routing rules. If communication with the IP PBX fails (i.e., Emergency mode), it still allows calls between the branch users themselves. If this fails, it routes the calls to the PSTN (if employed).

Figure 38-1: CRP in Normal & Auto Answer to Registrations Modes

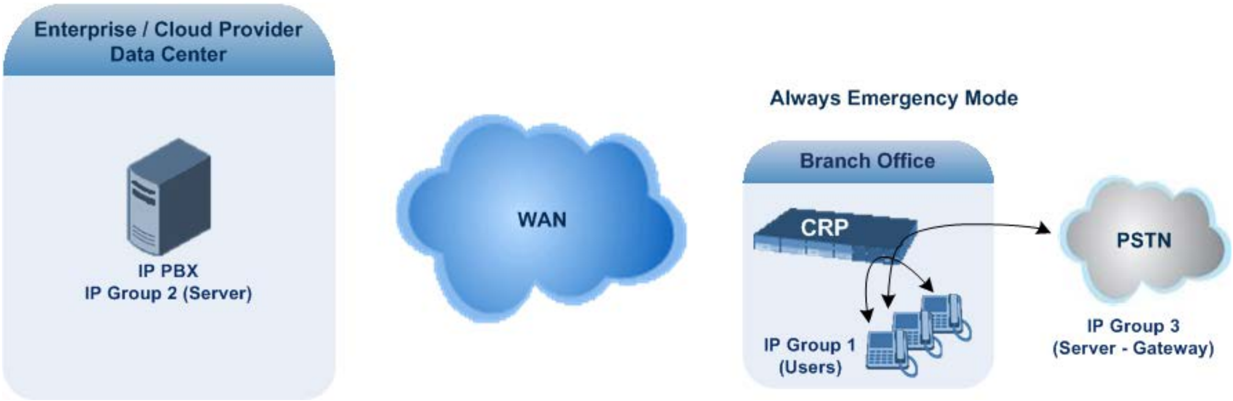


- **Auto Answer to Registrations:** This mode is the same as the Normal mode, except that the CRP registers the branch users in its registration database instead of forwarding them to the IP PBX.

Note: SIP REGISTER and OPTIONS requests are terminated at the CRP.

- **Always Emergency:** The CRP routes the calls between the branch users themselves as if connectivity failure has occurred with the IP PBX. The CRP also registers the branch users in its registration database.

Figure 38-2: CRP in Always Emergency Mode



➢ **To configure the Call Survivability mode:**

1. Open the General Settings page (**Configuration** tab > **VoIP** menu > **CRP** > **General Settings**).
2. From the 'CRP Survivability Mode' drop-down list, select the required mode.
3. Click **Submit**.

38.3 Pre-Configured IP Groups

For CRP, the device is pre-configured with the following IP Groups in the IP Group table:

Table 38-1: Pre-configured IP Groups in the IP Group Table

Index	Name	Type	Description
1	"CRP Users"	User	LAN users (e.g., IP phones) at the branch office
2	"CRP Proxy"	Server	Server (e.g., hosted IP PBX at the Enterprise's headquarters)
3	"CRP Gateway"	Server	Device's interface with the PSTN

These IP Groups are used in the IP-to-IP routing rules to indicate the source and destination of the call (see "Pre-Configured IP-to-IP Routing Rules" on page 633).

Notes:

- These IP Groups cannot be deleted and additional IP Groups cannot be configured. The IP Groups can be edited, except for the fields listed above, which are read-only.
- For accessing the IP Group table and for a description of its parameters, see "Configuring IP Groups" on page 340.

38.4 Pre-Configured IP-to-IP Routing Rules

For the CRP application, the IP-to-IP Routing table is pre-configured with IP-to-IP routing rules. These rules depend on the configured Call Survivability mode, as described in "Configuring Call Survivability Mode" on page 631.

Notes:

- The IP-to-IP Routing table is read-only.
- For accessing the IP-to-IP Routing table and for a description of its parameters, see "Configuring SBC IP-to-IP Routing Rules" on page 578.

38.4.1 Normal Mode

The pre-configured IP-to-IP routing rules for the Normal CRP call survivability mode are shown in the table below:

Table 38-2: Pre-Configured IP-to-IP Routing Rules for CRP Normal Mode

Index	Source IP Group / Emergency	Request Type	Destination Type	Destination IP Group	Destination Address	Alternative Route Options
1	Any	OPTIONS	Dest Address	-	Internal	Route Row
3	#1 [CRP Users]	All	IP Group	#2 [CRP Proxy]	-	Route Row
4	#1 [CRP Users]	All	IP Group	#1 [CRP Users]	-	Alternative Route Ignore Inputs
5	#1 [CRP Users]	All	IP Group	#3 [CRP Gateway]	-	Alternative Route Ignore Inputs
6	#2 [CRP Proxy]	All	IP Group	#1 [CRP Users]	-	Route Row
7	#2 [CRP Proxy]	All	IP Group	#3 [CRP Gateway]	-	Route Row
8	#3 [CRP Gateway]	All	IP Group	#2 [CRP Proxy]	-	Route Row
9	#3 [CRP Gateway]	All	IP Group	#1 [CRP Users]	-	Alternative Route Ignore Inputs

Note: Index 7 appears only if the CRPGatewayFallback parameter is enabled (see 'Configuring PSTN Fallback' on page 637). This routing rule is used if the device can't find a matching destination user for IP Group 1 (User-type IP Group) in its registration database. If the CRPGatewayFallback parameter is disabled and no matching user is find, the device rejects the call.

38.4.2 Emergency Mode

The pre-configured IP-to-IP routing rules for the Emergency CRP call survivability mode are shown in the table below:

Table 38-3: Pre-Configured IP-to-IP Routing Rules for Emergency Mode

Mode	Index	Source IP Group / Emergency	Request Type	Destination Type	Destination IP Group ID	Destination Address	Alternative Route Options
Always Emergency	1	Any	OPTIONS	Dest Address	-	Internal	Route Row
	4	#1 [CRP Users]	All	IP Group	#1 [CRP Users]	-	Route Row
	5	#1 [CRP Users]	All	IP Group	#3 [CRP Gateway]	-	Route Row
	9	#3 [CRP Gateway]	All	IP Group	#1 [CRP Users]	-	Route Row

Note: The routing rule at Index 5 appears only if the CRPGatewayFallback parameter is enabled (1).

38.4.3 Auto Answer to Registrations

The pre-configured IP-to-IP routing rules for the Auto Answer to Registrations CRP call survivability mode are shown in the table below:

Table 38-4: Pre-Configured IP-to-IP Routing Rule for Auto Answer to Registrations Mode

Mode	Index	Source IP Group	Request Type	Destination Type	Destination IP Group	Destination Address	Alternative Route Options
Auto Answer to Registrations	1	Any	OPTIONS	Dest Address	-	Internal	Route Row
	2	Any	REGISTER	IP Group	Any	-	Route Row
	3	#1 [CRP Users]	All	IP Group	#2 [CRP Proxy]	-	Route Row
	4	#1 [CRP Users]	All	IP Group	#1 [CRP Users]	-	Alternative Route Ignore Inputs
	5	#1 [CRP Users]	All	IP Group	#3 [CRP Gateway]	-	Alternative Route Ignore Inputs
	6	#2 [CRP Proxy]	All	IP Group	#1 [CRP Users]	-	Route Row
	7	#2 [CRP Proxy]	All	IP Group	#3 [CRP Gateway]	-	Route Row
	8	#3 [CRP Gateway]	All	IP Group	#2 [CRP Proxy]	-	Route Row
	9	#3 [CRP Gateway]	All	IP Group	#1 [CRP Users]	-	Alternative Route Ignore Inputs

Note:

- The destination for the routing rule at Index 2 is the source IP Group (i.e., from where the REGISTER message is received).
- Routing rule at Index 7 appears only if the CRPGatewayFallback parameter is enabled (see Configuring PSTN Fallback on page 637).

38.5 Configuring PSTN Fallback

You can enable the CRP to route emergency calls (or PSTN-intended calls) such as "911" from the Proxy server (IP Group 2) to the PSTN (IP Group 3). In addition, for calls from the Proxy server to Users (IP Group 1), the device searches for a matching user in its Users Registration database and if not not located, it sends the call to the PSTN (IP Group 3), as an alternative route.

To enable this feature, set the ini file parameter CRPGatewayFallback to 1. When enabled, the alternative routing rule appears immediately below the IP Group 2 to IP Group 1 rule in the IP-to-IP Routing table.

Notes:

- Enabling this feature (this routing rule) may expose the device to a security "hole", allowing calls from the WAN to be routed to the Gateway. Thus, configure this feature with caution and only if necessary.
- This PSTN routing rule is not an alternative routing rule. In other words, if a match for a user is located in the database, this PSTN rule will never be used regardless of the state of the user endpoint (e.g., busy).

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Part VIII

Maintenance

39 Basic Maintenance

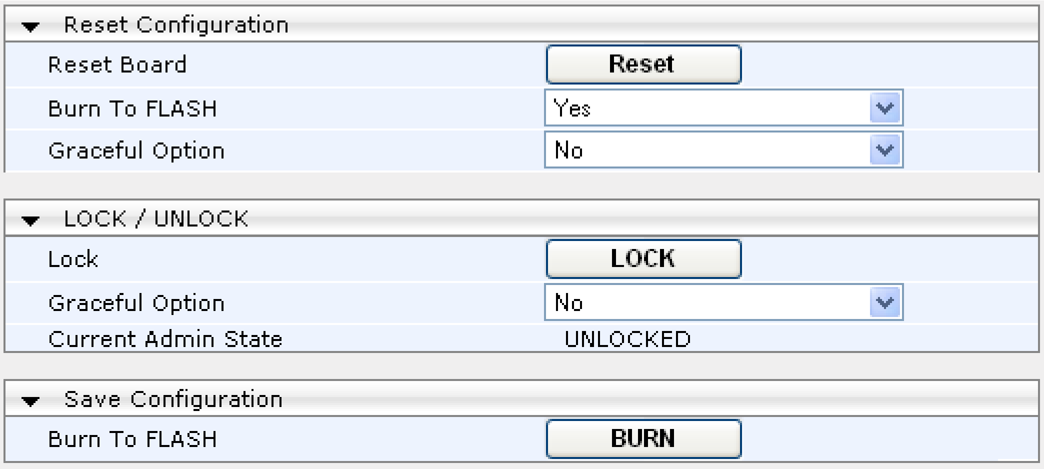
The Maintenance Actions page allows you to perform the following:

- Reset the device - see "Resetting the Device" on page 641
- Lock and unlock the device - see "Locking and Unlocking the Device" on page 642
- Save configuration to the device's flash memory - see "Saving Configuration" on page 643

➢ **To access the Maintenance Actions page, do one of the following:**

- On the toolbar, click the **Device Actions** button, and then from the drop-down menu, choose **Reset**.
- On the Navigation bar, click the **Maintenance** tab, and then in the Navigation tree, select the **Maintenance** menu and choose **Maintenance Actions**.

Figure 39-1: Maintenance Actions Page



Reset Configuration

Reset Board	Reset
Burn To FLASH	Yes
Graceful Option	No

LOCK / UNLOCK

Lock	LOCK
Graceful Option	No
Current Admin State	UNLOCKED

Save Configuration

Burn To FLASH	BURN

39.1 Remotely Resetting Device using SIP NOTIFY

The device can be remotely reset upon the receipt of a SIP NOTIFY that includes an Event header set to 'check-sync;reboot=true', as shown in the example below:

```
NOTIFY sip:<user>@<dsthost> SIP/2.0
To: sip:<user>@<dsthost>
From: sip:sipsak@<srchost>
CSeq: 10 NOTIFY
Call-ID: 1234@<srchost>
Event: check-sync;reboot=true
```

➢ **To enable remote reset upon receipt of SIP NOTIFY:**

1. Open the Advanced Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Advanced Parameters**).
2. Under the Misc Parameters group, set the 'SIP Remote Rest' parameter to **Enable**.
3. Click **Submit**.

Note: This SIP Event header value is proprietary to AudioCodes.

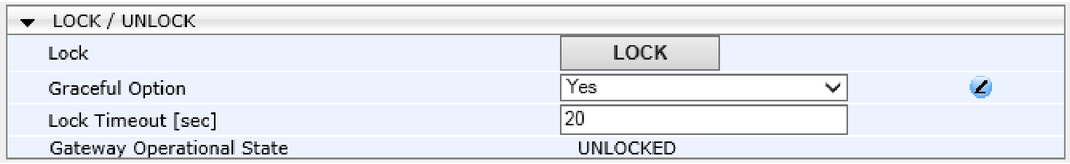
39.2 Locking and Unlocking the Device

The Lock and Unlock option allows you to lock the device so that it doesn't accept any new calls and maintains only the current calls. This is useful when, for example, you are uploading new software files to the device and you don't want any traffic to interfere with the process.

➢ **To lock the device:**

1. Open the Maintenance Actions page (see "Basic Maintenance" on page 641).
2. Scroll down to the 'LOCK / UNLOCK' group:

Figure 39-2: Locking the Device



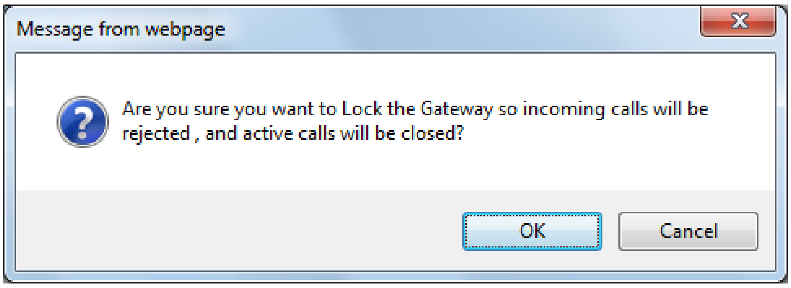
▾ LOCK / UNLOCK

Lock	LOCK
Graceful Option	Yes
Lock Timeout [sec]	20
Gateway Operational State	UNLOCKED

3. From the 'Graceful Option' drop-down list, select one of the following options:
   - **Yes:** The device is locked only after the user-defined time in the 'Lock Timeout' field (see Step 4) expires or no more active traffic exists (the earliest thereof). In addition, no new traffic is accepted.
   - **No:** The device is locked regardless of traffic. Any existing traffic is terminated immediately.

   Note: These options are only available if the current status of the device is in UNLOCKED state.
4. If you set 'Graceful Option' to **Yes** (in the previous step), then in the 'Lock Timeout' field, enter the time (in seconds) after which the device locks. If no traffic exists and the time has not yet expired, the device locks immediately.
5. Click the **LOCK** button; a confirmation message box appears requesting you to confirm device lock:

Figure 39-3: Device Lock Confirmation Message Box



Message from webpage

Are you sure you want to Lock the Gateway so incoming calls will be rejected , and active calls will be closed?

6. Click **OK** to confirm device lock; if you set 'Graceful Option' to **Yes**, a lock icon is delayed

and a window appears displaying the number of remaining calls and time. If you set 'Graceful Option' to **No**, the lock process begins immediately. The 'Gateway Operational State' field displays "LOCKED".

➢ **To unlock the device:**

- Under the 'LOCK / UNLOCK' group, click the **UNLOCK** button. Unlock starts immediately and the device accepts new incoming calls. The 'Gateway Operational State' field displays "UNLOCKED".

> **Note:** The Home page's General Information pane displays whether the device is locked or unlocked (see "Viewing the Home Page" on page 68).

39.3 Saving Configuration

The Maintenance Actions page allows you to save (*burn*) the current parameter configuration (including loaded Auxiliary files) to the device's *non-volatile* memory (i.e., flash). The parameter modifications that you make throughout the Web interface's pages are temporarily saved (to the *volatile* memory - RAM) when you click the **Submit** or **Add** buttons on these pages. Parameter settings that are saved only to the device's RAM revert to their previous settings after a hardware/software reset (or power failure). Therefore, to ensure that your configuration changes are retained, you must save them to the device's flash memory using the burn option described below.

➢ **To save the changes to the non-volatile flash memory:**

1. Open the Maintenance Actions page (see "Basic Maintenance" on page 641).
2. Under the 'Save Configuration' group, click the **BURN** button; a confirmation message appears when the configuration successfully saves.

> **Notes:**
>
> - Saving configuration to the *non-volatile* memory may disrupt current traffic on the device. To avoid this, disable all new traffic before saving, by performing a graceful lock (see "Locking and Unlocking the Device" on page 642).
> - Throughout the Web interface, parameters displayed with the lightning ⚡ symbol are not applied on-the-fly and require that you reset the device for them to take effect (see "Resetting the Device" on page 641).
> - The Home page's General Information pane displays whether the device is currently "burning" the configuration (see "Viewing the Home Page" on page 68).

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40 Channel Maintenance

This chapter describes various channel-related maintenance procedures.

40.1 Disconnecting Active Calls

You can forcibly disconnect all active (established) calls or disconnect specific calls based on their Session ID. This is done in the CLI using the following commands (from basic command mode):

- Disconnects all active calls:

```
# clear voip calls
```

- Disconnects active calls belonging to a specified Session ID:

```
# clear voip calls <Session ID>
```

40.2 Restarting a B-Channel

You can restart a specific B-channel belonging to an ISDN trunk, using the SNMP MIB variable, acTrunkISDNCommonRestartBChannel. This may be useful, for example, for troubleshooting specific voice channels.

Notes:

- If a voice call is currently in progress on the B-channel, it is disconnected when the B-channel is restarted.
- B-channel restart can only be done if the D-channel of the trunk to which it belongs is synchronized (Layer 2).
- B-channel restart does not affect the B-channel's configuration.

40.3 Locking and Unlocking Trunk Groups

You can lock a Trunk Group to take its trunks (and their channels) out of service. When you initiate the lock process, the device rejects all new incoming calls for the Trunk Group and immediately terminates active calls (busy channels), eventually taking the entire Trunk Group out of service. You can also lock a Trunk Group “gracefully”, whereby the device also rejects new incoming calls, but terminates busy channels only after a user-defined graceful period if the channel is still busy by the end of the period. The graceful period is configured by the GracefulBusyOutTimeout parameter. When configured to 0, graceful lock is disabled. When you lock a Trunk Group, the method for taking trunks/channels out-of-service is determined by the DigitalOOSBehaviorForTrunk parameter for per trunk or DigitalOOSBehavior parameter for all trunks.

If you have configured registration for the Trunk Group (see the 'Registration Mode' parameter in the Trunk Group Settings table) and you lock the Trunk Group, it stops performing registration requests with the Serving IP Group with which you have configured it to register. When you unlock such a Trunk Group, it starts performing registration requests with the Serving IP Group once its trunks return to service.

➢ To lock or unlock a Trunk Group:

1. Open the Trunk Group Settings table (**Configuration** tab > **VoIP** menu > **Gateway** > **Trunk Group** > **Trunk Group Settings**).
2. Select the table row of a Trunk Group that you want to lock or unlock.

3. From the **Action** drop-down list located on the table's toolbar, choose one of the following commands:
    - **Lock:** Locks the Trunk Group.
    - **Unlock**: Unlocks a locked Trunk Group.

The Trunk Group Settings table provides the following read-only fields related to locking and unlocking of a Trunk Group:

- 'Admin State': Displays the administrators state - "Locked" or "Unlocked"
- 'Status': Displays the current status of the channels in the Trunk Group:
    - "In Service": Indicates that all channels in the Trunk Group are in service, for example, when the Trunk Group is unlocked or Busy Out state cleared (see the EnableBusyOut parameter for more information).
    - "Going Out Of Service": Appears as soon as you choose the **Lock** button and indicates that the device is starting to lock the Trunk Group and take channels out of service.
    - "Going Out Of Service (<duration remaining of graceful period> sec / <number of calls still active> calls)": Appears when the device is locking the Trunk Group and indicates the number of buys channels and the time remaining until the graceful period ends, after which the device locks the channels regardless of whether the call has ended or not.
    - "Out Of Service": All fully configured trunks in the Trunk Group are out of service, for example, when the Trunk Group is locked or in Busy Out state (see the EnableBusyOut parameter).

Note: If the device is reset, a locked Trunk Group remains locked. If the device is reset while graceful lock is in progress, the Trunk Group is forced to lock immediately after the device finishes its reset.

41 Software Upgrade

This chapter describes various software update procedures.

41.1 Auxiliary Files

You can install various Auxiliary files on the device. Auxiliary files provide the device with additional configuration settings. The table below lists the different types of Auxiliary files.

Table 41-1: Auxiliary Files

File	Description
INI	Configures the device. The Web interface enables practically full device provisioning. However, some features may only be configured by ini file or you may wish to configure your device using the ini file. For more information on the ini file, see "INI File-Based Management" on page 99.
Call Progress Tones	Region-specific, telephone exchange-dependent file that contains the Call Progress Tones (CPT) levels and frequencies for the device. The default CPT file is U.S.A. For more information, see "Call Progress Tones File" on page 650.
Prerecorded Tones	The Prerecorded Tones (PRT) file enhances the device's capabilities of playing a wide range of telephone exchange tones that cannot be defined in the CPT file. For more information, see "Prerecorded Tones File" on page 652.
Dial Plan	Provides dialing plans, for example, to know when to stop collecting dialed digits and start forwarding them or for obtaining the destination IP address for outbound IP routing. For more information, see "Dial Plan File" on page 653.
User Info	The User Information file maps PBX extensions to IP numbers. This file can be used to represent PBX extensions as IP phones in the global 'IP world'. For more information, see "User Information File" on page 659.
AMD Sensitivity	Answer Machine Detector (AMD) Sensitivity file containing the AMD Sensitivity suites. For more information, see AMD Sensitivity File on page 667.

41.1.1 Loading Auxiliary Files

You can load Auxiliary files to the device using one of the following methods:

- Web interface - see "Loading Auxiliary Files through Web Interface" on page 648
- CLI - see Loading Auxiliary Files through CLI on page 649
- TFTP - see "Loading Auxiliary Files through ini File using TFTP" on page 649

Notes:

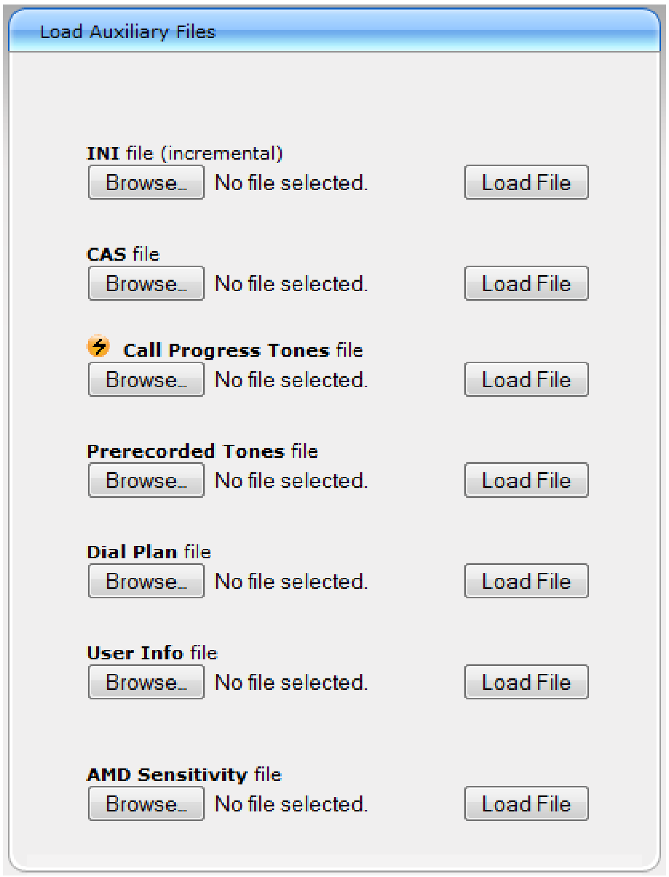
- You can schedule automatic loading of updated Auxiliary files using HTTP/HTTPS. For more information, see Automatic Update Mechanism.
- Saving Auxiliary files to flash memory may disrupt traffic on the device. To avoid this, disable all traffic on the device by performing a graceful lock as described in "Locking and Unlocking the Device" on page 642.
- To delete installed Auxiliary files, see "Viewing Device Information" on page 705.

41.1.1.1 Loading Auxiliary Files through Web Interface

The following procedure describes how to load Auxiliary files through the Web interface.

➢ **To load Auxiliary files through the Web interface:**

1. Open the Load Auxiliary Files page (**Maintenance** tab > **Software Update** menu > **Load Auxiliary Files**).



Load Auxiliary Files

INI file (incremental)
Browse... No file selected. Load File

CAS file
Browse... No file selected. Load File

Call Progress Tones file
Browse... No file selected. Load File

Prerecorded Tones file
Browse... No file selected. Load File

Dial Plan file
Browse... No file selected. Load File

User Info file
Browse... No file selected. Load File

AMD Sensitivity file
Browse... No file selected. Load File

> **Note:** The appearance of certain file load fields depends on the installed Software License Key.

2. Click the **Browse** button corresponding to the Auxiliary file type that you want to load, navigate to the folder in which the file is located, and then click **Open**; the name of the file appears in the field next to the **Browse** button.
3. Click the **Load File** button corresponding to the file you want to load.
4. Repeat steps 2 through 3 for each file you want to load.
5. Reset the device with a burn-to-flash for your settings to take effect (if you have loaded a Call Progress Tones file).

> **Note:** When loading an *ini* file using the Web interface, Auxiliary files that are already installed on the device are maintained if the ini file does not contain these Auxiliary files.

41.1.1.2 Loading Auxiliary Files through CLI

You can load Auxiliary files from user-defined URLs, using the following CLI commands:

- **Single Auxiliary file:**

  ```
  # copy <file> from <URL>
  ```

  For example:

  ```
  # copy call_progress_tones from
  http://192.169.11.11:80/cpt_us.dat
  ```

- **Multiple (batch) Auxiliary files:** The Auxiliary files must be contained in a TAR (Tape ARchive) file (.tar). The TAR file can contain any number and type of Auxiliary files (e.g., Dial Plan file and CPT file).

  ```
  # copy aux-package from | to <URL with TAR file name>
  ```

  For example:

  ```
  # copy aux-package from http://192.169.11.11:80/aux_files.tar
  ```

For more information on CLI commands, refer to the *CLI Reference Guide*.

41.1.1.3 Loading Auxiliary Files through ini File using TFTP

You can load Auxiliary files to the device through the ini file, using a TFTP server. For more information on Auxiliary ini file parameters, see "Auxiliary and Configuration File Name Parameters" on page 825.

➢ **To load Auxiliary files through ini file:**

1. Create an ini file that includes the names of the Auxiliary files that you want loaded, for example:

   ```
   CallProgressTonesFilename = 'usa_tones_13.dat'
   DialPlanFileName = 'dial-plan-us.dat'
   ```

2. Save the ini file and the Auxiliary files in the same folder on your TFTP server.
3. Reset the device (you can power off and then power on the device); the device loads the ini file and then the Auxiliary files as defined in the ini file, through TFTP.

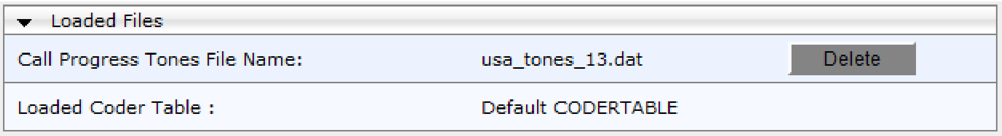
41.1.2 Deleting Auxiliary Files

You can delete loaded Auxiliary files through the Web interface, as described below.

➢ **To delete a loaded file:**

1. Open the Device Information page (**Status & Diagnostics** tab > **System Status** menu > **Device Information**); loaded files are listed under the Loaded Files group, as shown in the example below:

Figure 41-1: Loaded Files Listed on Device Information Page



▼ Loaded Files

Call Progress Tones File Name:	usa_tones_13.dat
Loaded Coder Table :	Default CODERTABLE

2. Click the **Delete** button corresponding to the file that you want to delete; a confirmation message box appears.
3. Click **OK** to confirm deletion.
4. Reset the device with a burn-to-flash for your settings to take effect.

41.1.3 Call Progress Tones File

The Call Progress Tones (CPT) Auxiliary file includes the definitions of the CPT (levels and frequencies) that are detected / generated by the device.

You can use one of the supplied Auxiliary files (.dat file format) or create your own file. To create your own file, it's recommended to modify the supplied *usa_tone.ini* file (in any standard text editor) to suit your specific requirements and then convert the modified *ini* file into binary format, using AudioCodes DConvert utility. For a description on converting a CPT *ini* file into a binary *dat* file, refer to the *DConvert Utility User's Guide*.

> **Note:** Only the *dat* file format can be loaded to the device.

You can create up to 32 different Call Progress Tones, each with frequency and format attributes. The frequency attribute can be single or dual-frequency (in the range of 300 to 1980 Hz) or an Amplitude Modulated (AM). Up to 64 different frequencies are supported. Only eight AM tones, in the range of 1 to 128 kHz, can be configured (the detection range is limited to 1 to 50 kHz). Note that when a tone is composed of a single frequency, the second frequency field must be set to zero.

The format attribute can be one of the following:

- **Continuous:** A steady non-interrupted sound (e.g., a dial tone). Only the 'First Signal On time' should be specified. All other on and off periods must be set to zero. In this case, the parameter specifies the detection period. For example, if it equals 300, the tone is detected after 3 seconds (300 x 10 msec). The minimum detection time is 100 msec.
- **Cadence:** A repeating sequence of on and off sounds. Up to four different sets of on/off periods can be specified.
- **Burst:** A single sound followed by silence. Only the 'First Signal On time' and 'First Signal Off time' should be specified. All other on and off periods must be set to zero. The burst tone is detected after the off time is completed.

You can specify several tones of the same type. These additional tones are used only for tone detection. Generation of a specific tone conforms to the first definition of the specific tone. For example, you can define an additional dial tone by appending the second dial tone's definition lines to the first tone definition in the *ini* file. The device reports dial tone detection if either of the two tones is detected.

The Call Progress Tones section of the *ini* file comprises the following segments:

- **[NUMBER OF CALL PROGRESS TONES]:** Contains the following key:
  'Number of Call Progress Tones' defining the number of Call Progress Tones that are defined in the file.
- **[CALL PROGRESS TONE #X]:** containing the Xth tone definition, starting from 0 and not exceeding the number of Call Progress Tones less 1 defined in the first section (e.g., if 10 tones, then it is 0 to 9), using the following keys:
  - **Tone Type:** Call Progress Tone types:
    - **[1]** Dial Tone
    - **[2]** Ringback Tone
    - **[3]** Busy Tone
    - **[4]** Congestion Tone
    - **[6]** Warning Tone
    - **[7]** Reorder Tone
    - **[17]** Call Waiting Ringback Tone - heard by the calling party
    - **[18]** Comfort Tone

  - **[23]** Hold Tone
  - **[46]** Beep Tone
- **Tone Modulation Type:** Amplitude Modulated (1) or regular (0)
- **Tone Form:** The tone's format can be one of the following:
  - Continuous (1)
  - Cadence (2)
  - Burst (3)
- **Low Freq [Hz]:** Frequency (in Hz) of the lower tone component in case of dual frequency tone, or the frequency of the tone in case of single tone. This is not relevant to AM tones.
- **High Freq [Hz:** Frequency (in Hz) of the higher tone component in case of dual frequency tone, or zero (0) in case of single tone (not relevant to AM tones).
- **Low Freq Level [-dBm]:** Generation level 0 dBm to -31 dBm in dBm (not relevant to AM tones).
- **High Freq Level:** Generation level of 0 to -31 dBm. The value should be set to 32 in the case of a single tone (not relevant to AM tones).
- **First Signal On Time [10 msec]:** 'Signal On' period (in 10 msec units) for the first cadence on-off cycle. For continuous tones, the parameter defines the detection period. For burst tones, it defines the tone's duration.
- **First Signal Off Time [10 msec]:** 'Signal Off' period (in 10 msec units) for the first cadence on-off cycle (for cadence tones). For burst tones, the parameter defines the off time required after the burst tone ends and the tone detection is reported. For continuous tones, the parameter is ignored.
- **Second Signal On Time [10 msec]:** 'Signal On' period (in 10 msec units) for the second cadence on-off cycle. Can be omitted if there isn't a second cadence.
- **Second Signal Off Time [10 msec]:** 'Signal Off' period (in 10 msec units) for the second cadence on-off cycle. Can be omitted if there isn't a second cadence.
- **Third Signal On Time [10 msec]:** 'Signal On' period (in 10 msec units) for the third cadence on-off cycle. Can be omitted if there isn't a third cadence.
- **Third Signal Off Time [10 msec]:** 'Signal Off' period (in 10 msec units) for the third cadence on-off cycle. Can be omitted if there isn't a third cadence.
- **Fourth Signal On Time [10 msec]:** 'Signal On' period (in 10 msec units) for the fourth cadence on-off cycle. Can be omitted if there isn't a fourth cadence.
- **Fourth Signal Off Time [10 msec]:** 'Signal Off' period (in 10 msec units) for the fourth cadence on-off cycle. Can be omitted if there isn't a fourth cadence.
- **Carrier Freq [Hz]:** Frequency of the carrier signal for AM tones.
- **Modulation Freq [Hz]:** Frequency of the modulated signal for AM tones (valid range from 1 to 128 Hz).
- **Signal Level [-dBm]:** Level of the tone for AM tones.
- **AM Factor [steps of 0.02]:** Amplitude modulation factor (valid range from 1 to 50). Recommended values from 10 to 25.

Notes:

- When the same frequency is used for a continuous tone and a cadence tone, the 'Signal On Time' parameter of the continuous tone must have a value that is greater than the 'Signal On Time' parameter of the cadence tone. Otherwise, the continuous tone is detected instead of the cadence tone.
- The tones frequency must differ by at least 40 Hz between defined tones.

For example, to configure the dial tone to 440 Hz only, enter the following text:

```
[NUMBER OF CALL PROGRESS TONES]
Number of Call Progress Tones=1
#Dial Tone
[CALL PROGRESS TONE #0]
Tone Type=1
Tone Form =1 (continuous)
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=10 (-10 dBm)
High Freq Level [-dBm]=32 (use 32 only if a single tone is
required)
First Signal On Time [10msec]=300; the dial tone is detected after
3 sec
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=0
Second Signal Off Time [10msec]=0
```

41.1.4 Prerecorded Tones File

The Prerecorded Tone (PRT) is a .dat file containing a set of prerecorded tones that can be played by the device. For example, it can be used to play music on hold (MoH) to a call party that has been put on hold. Up to 40 tones (totaling approximately 10 minutes) can be stored in a single PRT file on the device's flash memory. Play of tones from a PRT file is supported for Gateway and SBC calls.

The PRT file overcomes the limitations of the CPT file such as limited number of predefined tones and limited number of frequency integrations in one tone. If a specific prerecorded tone exists in the PRT file, it overrides the same tone that exists in the CPT file, and is played instead.

You can define a PRT file for SBC calls with multiple tones of the same tone type but with different coders. If one of the tones is defined with the same coder as used in the current call, the device always selects it in order to eliminate the need for using DSP resources. If the coder of the tone is the same as that of the call, DSPs are not required; if they are different, DSPs are required.

Notes:

- The PRT file only generates (plays) tones; detection of tones is according to the CPT file.
- The device does not require DSPs for playing tones from a PRT file if the coder defined for the tone is the same as that used by the current call. If the coders are different, the device uses DSPs.
- The device requires DSPs for local generation of tones.
- For SBC calls, the PRT file supports only the ringback tone and hold tone.

The prerecorded tones can be created using standard third-party, recording utilities such as Adobe Audition, and then combined into a single file (PRT file) using AudioCodes DConvert utility (refer to the document, *DConvert Utility User's Guide* for more information).

The raw data files must be recorded with the following characteristics:

- Coders: G.711 A-law or G.711 µ-law (and other coders)
- Rate: 8 kHz
- Resolution: 8-bit
- Channels: mono

The prerecorded tones are played repeatedly. This allows you to record only part of the tone and then play the tone for the full duration. For example, if a tone has a cadence of 2 seconds on and 4 seconds off, the recorded file should contain only these 6 seconds. The device repeatedly plays this cadence for the configured duration. Similarly, a continuous tone can be played by repeating only part of it.

Once created, you need to install the PRT file on the device. This can be done using the Web interface (see "Loading Auxiliary Files" on page 647).

41.1.5 Dial Plan File

The Dial Plan file can be used for various digit mapping features, as described in this section.

41.1.5.1 Creating a Dial Plan File

The Dial Plan file is a text-based file that can contain up to 8 Dial Plans (Dial Plan indices) and up to 8,000 rules (lines). The general syntax rules for the Dial Plan file are as follows (syntax specific to the feature is described in the respective section):

- Each Dial Plan index must begin with a Dial Plan name enclosed in square brackets "[...]" on a new line.
- Each line under the Dial Plan index defines a rule.
- Empty lines are ignored.
- Lines beginning with a semicolon ";" are ignored. The semicolon can be used for comments.

Creating a Dial Plan file is similar for all Dial Plan features. The main difference is the syntax used in the Dial Plan file and the method for selecting the Dial Plan index.

➢ **To create a Dial Plan file:**

1. Create a new file using a text-based editor (such as Notepad) and configure your Dial Plans as required.
2. Save the file with the *ini* file extension name (e.g., mydialplanfile.ini).
3. Convert the *ini* file to a *dat* binary file, using AudioCodes DConvert utility. For more information, refer to *DConvert Utility User's Guide*.
4. Load the converted file to the device, as described in "Loading Auxiliary Files" on page 647.
5. Select the Dial Plan index that you want to use. This depends on the feature and is described in the respective section.

41.1.5.2 Dialing Plans for Digit Collection

The device enables you to configure multiple dialing plans in an external Dial Plan file, which can be installed on the device. If a Dial Plan file is implemented, the device first attempts to locate a matching digit pattern in a specified Dial Plan index listed in the file and if not found, attempts to locate a matching digit pattern in the Digit Map. The Digit Map is configured by the 'Digit Mapping Rules' parameter, located in the DTMF & Dialing page (**Configuration** tab > **VoIP** menu > **Gateway** > **DTMF and Supplementary** > **DTMF & Dialing**).

The Dial Plan is used for the following:

- ISDN Overlap Dialing (Tel-to-IP calls): The file allows the device to know when digit collection ends, after which it starts sending all the collected (or dialed) digits in the outgoing INVITE message. This also provides enhanced digit mapping.

The Dial Plan file can contain up to 8 Dial Plans (Dial Plan indices), with a total of up to 8,000 dialing rules (lines) of distinct prefixes (e.g. area codes, international telephone number patterns) for the PSTN to which the device is connected.

The Dial Plan file is created in a textual *ini* file with the following syntax:

```
<called number prefix>,<total digits to wait before sending>
```

- Each new Dial Plan index begins with a Dial Plan name enclosed in square brackets "[...]" on a new line.
- Each line under the Dial Plan index defines a dialing prefix and the number of digits expected to follow that prefix. The prefix is separated by a comma "," from the number of additional digits.
- The prefix can include numerical ranges in the format [x-y], as well as multiple numerical ranges [n-m][x-y] (no comma between them).
- The prefix can include the asterisk "*" and number "#" signs.
- The number of additional digits can include a numerical range in the format x-y.
- Empty lines are ignored.
- Lines beginning with a semicolon ";" are ignored. The semicolon can be used for comments.

Below shows an example of a Dial Plan file (in *ini*-file format), containing two dial plans:

```
; Example of dial-plan configuration.
; This file contains two dial plans:
[ PLAN1 ]
; Destination cellular area codes 052, 054, and 050 with 8 digits.

052,8
054,8
050,8
; Defines International prefixes 00, 012, 014.
; The number following these prefixes may
; be 7 to 14 digits in length.
00,7-14
012,7-14
014,7-14
; Defines emergency number 911. No additional digits are expected.
911,0
[ PLAN2 ]
; Defines area codes 02, 03, 04.
; In these area codes, phone numbers have 7 digits.
0[2-4],7
; Operator services starting with a star: *41, *42, *43.
; No additional digits are expected.
*4[1-3],0
```

The following procedure provides a summary on how to create a Dial Plan file and select the required Dial Plan index.

➢ **To create a Dial Plan file:**

1. Create a new file using a text-based editor (such as Notepad) and configure your Dial Plans, as required.
2. Save the file with the *ini* file extension name (e.g., mydialplans.ini).
3. Convert the *ini* file to a *dat* binary file, using AudioCodes DConvert utility. For more information, refer to *DConvert Utility User's Guide*.
4. Install the converted file on the device, as described in "Loading Auxiliary Files" on page 647.
5. The required Dial Plan is selected using the 'Dial Plan Index' parameter. The parameter

can be set to **0** through **7**, where **0** denotes PLAN1, **1** denotes PLAN2, and so on.

Notes:

- The Dial Plan file must not contain overlapping prefixes. Attempting to process an overlapping configuration by the DConvert utility results in an error message specifying the problematic line.
- The Dial Plan index can be selected globally for all calls (as described in the previous procedure), or per specific calls using Tel Profiles.
- It may be useful to configure both Dial Plan file and Digit Maps. For example, the Digit Map can be used for complex digit patterns (which are not supported by the Dial Plan file) and the Dial Plan can be used for long lists of relatively simple digit patterns. In addition, as timeout between digits is not supported by the Dial Plan, the Digit Map can be used to configure digit patterns that are shorter than those defined in the Dial Plan or left at default (MaxDigits parameter). For example, the “xx.T” digit map instructs the device to use the Dial Plan and if no matching digit pattern is found, it waits for two more digits and then after a timeout (TimeBetweenDigits parameter), it sends the collected digits. Therefore, this ensures that calls are not rejected as a result of their digit pattern not been completed in the Dial Plan.
- By default, if no matching digit pattern is found in both the Dial Plan and Digit Map, the device rejects the call. However, if you set the DisableStrictDialPlan parameter to 1, the device attempts to complete the call using the MaxDigits and TimeBetweenDigits parameters. In such a setup, it collects the number of digits configured by the MaxDigits parameters. If more digits are received, it ignores the settings of the parameter and collects the digits until the inter-digit timeout configured by the TimeBetweenDigits parameter is exceeded.

41.1.5.3 Dial Plan Prefix Tags for Routing

41.1.5.3.1Dial Plan Prefix Tags for IP-to-Tel Routing

For deployments requiring many IP-to-Tel routing rules that exceed the maximum number of rules that can be configured in the IP to Trunk Group Routing table, you can employ user-defined string labels (tags) to represent the many different prefix calling (source) and called (destination) numbers. The prefix tags are used in the IP to Trunk Group Routing table (see "Configuring IP-to-Trunk Group Routing Rules" on page 476) as source and destination number matching characteristics for the routing rule. Prefix tags are typically implemented when you have calls of many different called or calling numbers that need to be routed to the same destination. Thus, instead of configuring a routing rule for each prefix number, you need to configure only one routing rule using the prefix tag.

For example, this feature is useful in deployments that need to handle hundreds of call routing scenarios such as for a large geographical area (a state in the US). Such an area could consist of hundreds of local area codes as well as codes for international calls. The local calls and international calls would need to be routed to different SIP trunks. Thus, instead of configuring many routing rules for each call destination type, you can simply configure two routing rules, one with a unique prefix tag representing the different local area codes and the other with a prefix tag representing international calls.

Note: When using prefix tags, you need to configure manipulation rules to remove the tags before the device sends the calls to their destinations.

You configure prefix tags in the Dial Plan file, using the following syntax:

```
[ PLAN<index> ]
<prefix number>,0,<prefix tag>
```

where:

- *Index* is the Dial Plan index
- *prefix number* is the called or calling number prefix (ranges can be defined in brackets)
- *prefix tag* is the user-defined prefix tag of up to nine characters, representing the prefix number

Each prefix tag type - called or calling - must be configured in a dedicated Dial Plan index number. For example, Dial Plan 1 can be for called prefix tags and Dial Plan 2 for calling prefix tags.

The example Dial Plan file below defines the prefix tags "LOCL"and "LONG" to represent different called number prefixes for local and long distance calls:

```
[ PLAN1 ]
42520[3-5],0,LOCL
425207,0,LOCL
42529,0,LOCL
425200,0,LONG
425100,0,LONG
....
```

> **Note:** Called and calling prefix tags can be used in the same routing rule.

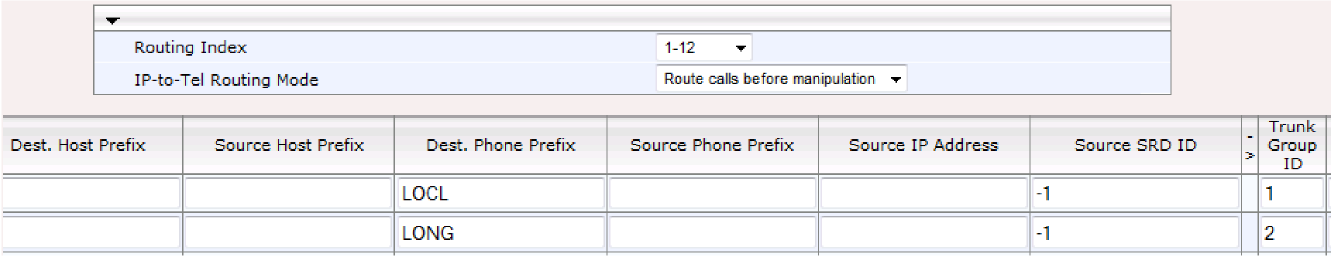
The following procedure describes how to configure IP-to-Tel routing using prefix tags.

➢ To configure IP-to-Tel routing using prefix tags:

1. Configure a Dial Plan file with prefix tags, and then load the file to the device.
2. On the Routing General Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**), specify the Dial Plan indices (e.g., 1) where you configured the prefix tags:
   - Prefix tags for called number prefixes: 'IP-to-Tel Tagging Destination Dial Plan Index' parameter
   - Prefix tags for calling number prefixes: 'IP-to-Tel Tagging Source Dial Plan Index' parameter
3. Open the IP to Trunk Group Routing table (**Configuration** tab > **VoIP** menu > **Gateway** > **Routing** > **IP to Trunk Group Routing**).
   a. From the 'IP-to-Tel Routing Mode' drop-down list, select **Route calls before manipulation** so that the device performs the routing processing before manipulation.
   b. Configure routing rules using the prefix tags as matching characteristics for destination or source number prefixes:
      - ♦ Prefix tags for called number prefixes: 'Destination Phone Prefix'. For example, configure two routing rules:
        - ✓ Set this field to "LOCL" and the 'Trunk Group ID' field to 1 (local Trunk Group).
        - ✓ Set this field to "LONG" and the 'Trunk Group ID' field to 2 (long distance Trunk Group).

- ♦ Prefix tags for calling number prefixes: 'Source Phone Prefix'.

Figure 41-2: Configuring Dial Plan File Label for IP-to-Tel Routing



Routing Index	1-12
IP-to-Tel Routing Mode	Route calls before manipulation

Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	Source SRD ID	->	Trunk Group ID
		LOCL			-1		1
		LONG			-1		2

4. Configure manipulation rules to remove the prefix called tags:
   a. Open the Destination Phone Number Manipulation Table for IP-to-Tel Calls table (**Configuration** tab > **VoIP** menu > **Gateway** > **Manipulations** > **Dest Number IP->Tel**).
   b. In the 'Destination Prefix' field, enter the prefix called tag (e.g., "LOCL").
   c. In the 'Stripped Digits From Left' field, enter the number of characters in the prefix called tag (e.g., "4").
5. Configure manipulation rules to remove the prefix calling tags:
   a. Open the Source Phone Number Manipulation Table for IP-to-Tel Calls table (**Configuration** tab > **VoIP** menu > **Gateway** > **Manipulations** > **Source Number IP->Tel**).
   b. In the 'Source Prefix' field, enter the prefix calling tag.
   c. In the 'Stripped Digits From Left' field, enter the number of characters in the prefix calling tag.

41.1.5.4 Obtaining IP Destination from Dial Plan File

You can use a Dial Plan index listed in a loaded Dial Plan file for determining the IP destination of Tel-to-IP and SBC calls. This enables the mapping of called numbers to IP addresses (in dotted-decimal notation) or FQDNs (up to 15 characters).

> **Note:** For the SBC application, the method described in this section for obtaining an IP address using the Dial Plan file is for backward compatibility purposes only. For the new method, see Configuring Dial Plans on page 607.

➢ **To configure routing to an IP destination based on Dial Plan:**

1. Create the Dial Plan file. The syntax of the Dial Plan index for this feature is as follows:

```
<destination / called prefix number>,0,<IP destination>
```

Note: The second parameter "0" is not used and ignored.

An example of a configured Dial Plan (# 6) in the Dial Plan file is shown below:

```
[ PLAN6 ]
200,0,10.33.8.52    ; called prefix 200 is routed to
10.33.8.52
201,0,10.33.8.52
300,0,itsp.com      ; called prefix 300 is routed to itsp.com
```

2. Convert the file to a loadable file and then load it to the device (see "Creating a Dial Plan File" on page 653).
3. Assign the Dial Plan index to the required routing rule:
   - SBC Calls: In the SBC IP-to-IP Routing table, do the following:
     a. Set the 'Destination Type' field to Dial Plan.

b. In the 'Destination Address' field, enter the required Dial Plan index, where "0" denotes [PLAN1] in the Dial Plan file, "1" denotes [PLAN2], and so on.

- Tel-to-IP Calls (Gateway application): In the Tel-to-IP Routing table, do the following:

   a. In the 'Destination Address' field, enter the required Dial Plan index using the following syntax:

      DialPlan<index>

      Where "DialPlan0" denotes [PLAN1] in the Dial Plan file, "DialPlan1" denotes [PLAN2], and so on.

Note: The "DialPlan" string is case-sensitive.

41.1.5.5 Modifying ISDN-to-IP Calling Party Number

The device can use the Dial Plan file to change the Calling Party Number value (source number) of the incoming ISDN call when sending to IP. For this feature, the Dial Plan file supports the following syntax:

<ISDN Calling Party Number>,0,<new calling number>

- The first number contains the calling party number (or its prefix) received in the ISDN call SETUP message. The source number can also be a range, using the syntax [x-y] in the Dial Plan file. This number is used as the display name in the From header of the outgoing INVITE.
- The second number must always be set to "0".
- The third number is a string of up to 12 characters containing the mapped number that is used as the URI user part in the From and Contact headers of the outgoing INVITE.

The Dial Plan index used in the Dial Plan file for this feature is defined by the Tel2IPSourceNumberMappingDialPlanIndex parameter.

An example of such a configuration in the Dial Plan file is shown below:

```
[ PLAN1 ]
; specific received number changed to 04343434181.
0567811181,0,04343434181
; number range that changes to 04343434181.
056788118[2-4],0,04343434181
```

If we take the first Dial Plan rule in the example above (i.e., "0567811181,0,04343434181"), the received Calling Number Party of 0567811181 is changed to 04343434181 and sent to the IP with a SIP INVITE as follows:

```
Via: SIP/2.0/UDP 211.192.160.214:5060;branch=z9hG4bK3157667347
From: <sip:04343434181@kt.co.kr:5060>;tag=de0004b1
To: sip:01066557573@kt.co.kr:5060
Call-ID: 585e60ec@211.192.160.214
CSeq: 1 INVITE
Contact:<sip:04343434181@211.192.160.214:5060;transport=udp>
```

The initial Dial Plan text file must be converted to *.dat file format using the DConvert utility. This is done by clicking the DConvert's **Process Dial Plan File** button. For more information, refer to *DConvert Utility User's Guide*.

You can load this *.dat file to the device using the Web interface (see "Loading Auxiliary Files" on page 647), AcBootP utility, or using the Auto-update mechanism from an external HTTP server.

Notes:

- Tel-to-IP routing is performed on the original source number if the parameter 'Tel to IP Routing Mode' is set to 'Route calls before manipulation'.
- Tel-to-IP routing is performed on the modified source number as defined in the Dial Plan file, if the parameter 'Tel To IP Routing Mode' is set to 'Route calls after manipulation'.
- Source number Tel-to-IP manipulation is performed on the modified source number as defined in the Dial Plan file.

41.1.5.6 Viewing Information of Installed Dial Plan File

You can view information about the Dial Plan file currently installed on the device, through the device's CLI:

- **Viewing Dial Plan file information:** You can view the file name of the installed Dial Plan file and the names of the Dial Plans defined in the Dial Plan file, by entering the following CLI command (in Enable mode):

```
# debug auxilary-files dial-plan info
```

  For example, the following shows the file name of the installed Dial Plan file and lists its Dial Plans:

```
# debug auxilary-files dial-plan info
  File Name: MyDialPlan.txt
  Plans:
  Plan #0 = PLAN1
  Plan #1 = PLAN2
```

  Note that the index number of the first Dial Plan is 0.

- **Searching a prefix number:** You can check whether a specific prefix number is defined in a specific Dial Plan (and view the corresponding tag if the Dial Plan implements tags), by entering the following CLI command (in Enable mode):

```
# debug auxilary-files dial-plan match-number <Dial Plan
number> <prefix number>
```

  For example, the following checks whether the called prefix number 2000 is defined in Dial Plan 1, which is used for obtaining the destination IP address (tag):

```
# debug auxilary-files dial-plan match-number PLAN1 2000
  Match found for 4 digits
  Matched prefix: 2000
  Tag: 10.33.45.92
```

41.1.6 User Information File

This section describes the User Info table and how to configure the table.

41.1.6.1 Enabling the User Info Table

Before you can use the User Info table, you need to enable the User Info functionality as described in the following procedure.

➢ **To enable the User Info table:**

1. Open the Advanced Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Advanced Parameters**).
2. Set the 'Enable User-Information Usage' parameter (EnableUserInfoUsage) to **Enable**.
3. Save this setting to the device with a reset for the setting to take effect.

41.1.6.2 Gateway User Information for PBX Extensions and "Global" Numbers

The GW User Info table contains user information that can be used for the following Gateway-related features:

- **Mapping (Manipulating) PBX Extension Numbers with Global Phone Numbers:** maps PBX extension number, connected to the device, with any "global" phone number (alphanumerical) for the IP side. In this context, the "global" phone number serves as a routing identifier for calls in the "IP world" and the PBX extension uses this mapping to emulate the behavior of an IP phone. This feature is especially useful in scenarios where unique or non-consecutive number translation per PBX is needed. This number manipulation feature supports the following call directions:
    - IP-to-Tel Calls: Maps the called "global" number (in the Request-URI user part) to the PBX extension number. For example, if the device receives an IP call destined for "global" number 638002, it changes this called number to the PBX extension number 402, and then sends the call to the PBX extension on the Tel side.

> **Note:** If you have configured regular IP-to-Tel manipulation rules (see "Configuring Source/Destination Number Manipulation" on page 441), the device applies these rules before applying the mapping rules of the User Info table.

- Tel-to-IP Calls: Maps the calling (source) PBX extension to the "global" number. For example, if the device receives a Tel call from PBX extension 402, it changes this calling number to 638002, and then sends call to the IP side with this calling number. In addition to the "global" phone number, the display name (caller ID) configured for the PBX user in the User Info table is used in the SIP From header.

> **Note:** If you have configured regular Tel-to-IP manipulation rules (see "Configuring Source/Destination Number Manipulation" on page 441), the device applies these rules before applying the mapping rules of the User Info table.

- **Registering Users:** The device can register each PBX user configured in the User Info table. For each user, the device sends a SIP REGISTER to an external IP-based Registrar server, using the "global" number in the From/To headers. If authentication is necessary for registration, the device sends the user's username and password, configured in the User Info table, in the SIP MD5 Authorization header.

You can configure up to 500 mapping rules in the GW User Info table. These rules can be configured using any of the following methods:

- Web interface - see "Configuring GW User Info Table through Web Interface" on page 661
- CLI - see Configuring GW User Info Table through CLI on page 662
- Loadable User Info file - see "Configuring GW User Info Table in Loadable Text File" on page 663

Notes:

- To enable user registration, set the following parameters on the Proxy & Registration page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Proxy & Registration**) as shown:
  - √ 'Enable Registration': **Enable** (IsRegisterNeeded is set to 1).
  - √ 'Registration Mode': **Per Endpoint** (AuthenticationMode is set to 0).

41.1.6.2.1 Configuring GW User Info Table through Web Interface

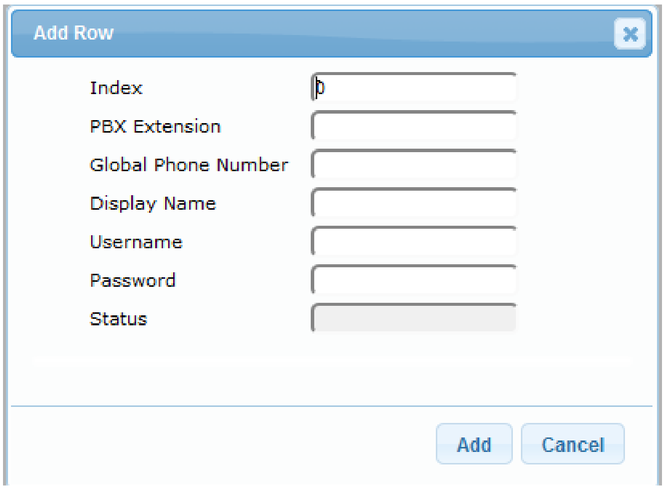
The following procedure describes how to configure and register users in the GW User Info table through the Web interface.

Note: If a User Info file is loaded to the device (as described in "Configuring GW User Info Table in Loadable Text File" on page 663), all previously configured entries are removed from the table in the Web interface and replaced with the entries from the loaded User Info file.

➢ **To configure the GW User Info table through the Web interface:**

1. Open the GW User Info table (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **User Information** > **GW User Info Table**).
2. Click **Add**; the following dialog box appears:

Figure 41-3: GW User Info Table - Add Row Dialog Box



Add Row

Index	0
PBX Extension	
Global Phone Number	
Display Name	
Username	
Password	
Status	

3. Configure the GW User Info table parameters according to the table below.
4. Click **Add**.
5. To save the changes to flash memory, see "Saving Configuration" **on page 643.**

To register a user, select the user's table entry, and then from the **Action** button's drop-down list , choose **Register**. To un-register a user, select the user, and then from the **Action** button's drop-down list , choose **Un-Register**.

Table 41-2: GW User Info Table Parameter Descriptions

Parameter	Description
Index [GWUserInfoTable_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
PBX Extension [GWUserInfoTable_PBXExtension]	Defines the PBX extension number. The valid value is a string of up to 10 characters. **Note:** The parameter is mandatory.
Global Phone Number [GWUserInfoTable_GlobalPhoneNumber]	Defines the "global" phone number for the IP side. The valid value is a string of up to 20 characters. **Note:** The parameter is mandatory.
Display Name [GWUserInfoTable_DisplayName]	Defines the Caller ID of the PBX extension. The valid value is a string of up to 30 characters.
Username [GWUserInfoTable_Username]	Defines the username for registering the user when authentication is necessary. The valid value is a string of up to 40 characters.
Password [GWUserInfoTable_Password]	Defines the password for registering the user when authentication is necessary. The valid value is a string of up to 20 characters.
Status	(Read-only field) Displays the status of the user - "Registered" or "Not Registered".

41.1.6.2.2Configuring GW User Info Table through CLI

The GW User Info table can be configured in the CLI using the following commands:

- To add and/or modify a user (example):

```
# configure voip
(config-voip)# sip-definition proxy-and-registration
(sip-def-proxy-and-reg)# user-info gw-user-info <index, e.g.,
1>
(gw-user-info-1)# username JohnDee
(gw-user-info-1)# <activate | exit>
```

- To delete a specific user, use the no command:

```
(sip-def-proxy-and-reg)# no user-info gw-user-info <index,
e.g., 1>
```

- To view all table entries:

```
(sip-def-proxy-and-reg)# user-info gw-user-info display
---- gw-user-info-0 ----
  pbx-ext (405)
  global-phone-num (405)
  display-name (Ext405)
  username (user405)
  password (0aGzoKfh5uI=)
  status (not-resgistered)
```

- To view a specific entry (example):

```
(sip-def-proxy-and-reg)# user-info gw-user-info <index, e.g.,
0>
(gw-user-info-0)# display
  pbx-ext (405)
  global-phone-num (405)
  display-name (Ext405)
  username (user405)
  password (0aGzoKfh5uI=)
  status (not-resgistered)
```

- To search a user by pbx-ext:

```
(sip-def-proxy-and-reg)# user-info find <pbx-ext e.g., 405>
405: Found at index 0 in GW user info table, not registered
```

41.1.6.2.3Configuring GW User Info Table in Loadable Text File

The GW User Info table can be configured as a User Info file using a text-based file (*.txt). This file can be created using any text-based program such as Notepad. You can load the User Info file using any of the following methods:

- Web interface - see "Loading Auxiliary Files" on page 647
- *ini* file, using the UserInfoFileName parameter - see "Auxiliary and Configuration File Name Parameters" on page 825
- Automatic Update mechanism, using the UserInfoFileURL parameter - see Automatic Update Mechanism

To add mapping rules to the User Info file, use the following syntax:

```
[ GW ]
FORMAT
PBXExtensionNum,GlobalPhoneNum,DisplayName,UserName,Password
```

Where:

- *[GW]* indicates that this part of the file is the GW User Info table
- *PBXExtensionNum* is the PBX extension number (up to 10 characters)
- *GlobalPhoneNum* is the "global" phone number (up to 20 characters) for the IP side
- *DisplayName* is the Caller ID (string of up to 30 characters) of the PBX extension
- *UserName* is the username (string of up to 40 characters) for registering the user when authentication is necessary
- *Password* is the password (string of up to 20 characters) for registering the user when authentication is necessary

Each line in the file represents a mapping rule of a single PBX extension user.

Notes:

- Make sure that there are no spaces between the values.
- Make sure that the last line in the User Info file ends with a carriage return (i.e., by pressing the <Enter> key).
- To modify the GW User Info table using a User Info file, you need to load to the device a new User Info file containing your modifications.

Below is an example of a configured User Info file:

```
[ GW ]
FORMAT
```

```
PBXExtensionNum,GlobalPhoneNum,DisplayName,UserName,Password
401,638001,Mike,miked,1234
402,638002,Lee,leem,4321
403,638003,Sue,suer,8790
404,638004,John,johnd,7694
405,638005,Pam,pame,3928
406,638006,Steve,steveg,1119
407,638007,Fred,frede,8142
408,638008,Maggie,maggiea,9807
```

41.1.6.3 User Information File for SBC User Database

You can use the SBC User Info table for the following:

- Registering each user to an external registrar server.
- Authenticating (for any SIP request and as a client) each user if challenged by an external server.
- Authenticating as a server incoming user requests (for SBC security).

If the device registers on behalf of users and the users do not perform registration, any SIP request destined to the user is routed to the Proxy Set associated with the user's IP Group.

You can configure up to 800 users (table rows) in the SBC User Info table. The SBC User Info table can be configured using any of the following methods:

- Web interface - see "Configuring SBC User Info Table through Web Interface" on page 664
- CLI - see Configuring SBC User Info Table through CLI on page 665
- Loadable User Info file - see "Configuring SBC User Info Table in Loadable Text File" on page 666

41.1.6.3.1Configuring SBC User Info Table through Web Interface

The following procedure describes how to configure the SBC User Info table through the Web interface.

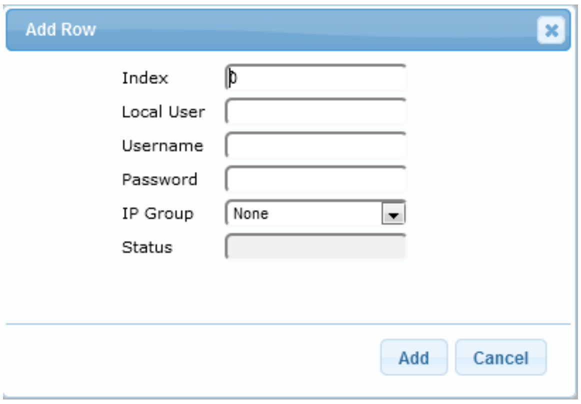
Note: If you load any User Info file to the device, all previously configured entries are removed from the table in the Web interface and replaced with the entries from the loaded User Info file.

➢ **To configure the SBC User Info table through the Web interface:**

1. Open the SBC User Info table (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **User Information** > **SBC User Info Table**).

2. Click **Add**; the following dialog box appears:

Figure 41-4: SBC User Info Table - Add Row Dialog Box



3. Configure the SBC User Info table parameters according to the table below.
4. Click **Add**.
5. To save the changes to flash memory, see "Saving Configuration" **on page 643.**

To register a user, select the user's table entry, and then from the **Action** button's drop-down list, choose **Register**. To un-register a user, select the user, and then from the **Action** button's drop-down list, choose **Un-Register**.

Table 41-3: SBC User Info Table Parameter Descriptions

Parameter	Description
Index [SBCUserInfoTable_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Local User [SBCUserInfoTable_LocalUser]	Defines the user and is used as the Request-URI user part for the AOR in the database. The valid value is a string of up to 10 characters.
Username [SBCUserInfoTable_Username]	Defines the username for registering the user when authentication is necessary. The valid value is a string of up to 40 characters.
Password [SBCUserInfoTable_Password]	Defines the password for registering the user when authentication is necessary. The valid value is a string of up to 20 characters.
IP Group [SBCUserInfoTable_IPGroupName]	Assigns an IP Group to the user and is used as the Request-URI source host part for the AOR in the database. For configuring IP Groups, see "Configuring IP Groups" on page 340.
Status [SBCUserInfoTable_Status]	(Read-only field) Displays the status of the user - "Registered" or "Not Registered".

41.1.6.3.2Configuring SBC User Info Table through CLI

The SBC User Info table can be configured in the CLI using the following commands:

- To add and/or modify a user (example):

```
# configure voip
(config-voip)# sip-definition proxy-and-registration
(sip-def-proxy-and-reg)# user-info sbc-user-info <index, e.g.,
1>
(sbc-user-info-1)# username JohnDee
(sbc-user-info-1)# <activate | exit>
```

- To delete a specific user, use the no command:

```
(sip-def-proxy-and-reg)# no user-info sbc-user-info <index,
e.g., 1>
```

- To view all table entries:

```
(sip-def-proxy-and-reg)# user-info sbc-user-info display
---- sbc-user-info-0 ----
  local-user (JohnDee)
  username (userJohn)
  password (s3fn+fn=)
  ip-group-id (1)
  status (not-resgistered)
---- sbc-user-info-1 ----
 local-user (SuePark)
  username (userSue)
  password (t6sn+un=)
  ip-group-id (1)
  status (not-resgistered)
```

- To view a specific entry (example):

```
(sip-def-proxy-and-reg)# user-info sbc-user-info <index, e.g.,
0>
(sbc-user-info-0)# display
  local-user (JohnDee)
  username (userJohn)
  password (s3fn+fn=)
  ip-group-id (1)
  status (not-resgistered)
```

- To search a user by local-user:

```
(sip-def-proxy-and-reg)# user-info find <local-user, e.g.,
JohnDoe>
JohnDee: Found at index 0 in SBC user info table, not
registered
```

41.1.6.3.3Configuring SBC User Info Table in Loadable Text File

The SBC User Info table can be configured as a User Info file using a text-based file (*.txt). This file can be created using any text-based program such as Notepad. This User Info file is the same file used for the GW User Info table. Thus, this file can include both Gateway and SBC user information.

You can load the User Info file using any of the following methods:

- Web interface - see "Loading Auxiliary Files" on page 647
- *ini* file, using the UserInfoFileName parameter - see "Auxiliary and Configuration File Name Parameters" on page 825
- Automatic Update mechanism, using the UserInfoFileURL parameter - see Automatic Update Mechanism

To add SBC users to the SBC User Info file, use the following syntax:

```
[ SBC ]
FORMAT LocalUser,UserName,Password,IPGroupID
```

where:

- *[SBC]* indicates that this part of the file is the SBC User Info table
- *LocalUser* is the user and is used as the Request-URI user part for the AOR in the database

- *UserName* is the user's authentication username
- *Password* is the user's authentication password
- *IPGroupID* is the IP Group ID to which the user belongs and is used as the Request-URI source host part for the AOR in the database

Note:

- Make sure that there are no spaces between the values.
- To modify the SBC User Info table using a User Info file, you need to load to the device a new User Info file containing your modifications.

Below is an example of a configured User Info file:

```
[ SBC ]
FORMAT LocalUser,UserName,Password,IPGroupID
john,john_user,john_pass,2
sue,sue_user,sue_pass,1
```

41.1.6.4 Viewing the Installed User Info File Name

You can view the name of the User Info file currently installed on the device, through the device's CLI (in Enable mode):

```
# debug auxilary-files user-info info
```

For example:

```
# debug auxilary-files user-info info
  User Info File Name MyUsers.txt
```

41.1.7 AMD Sensitivity File

The device is shipped with a default, pre-installed *AMD Sensitivity* file for its Answering Machine Detection (AMD) feature. This file includes the detection algorithms for detecting whether a human or answering machine has answered the call, and is based on North American English. In most cases, the detection algorithms in this file suffice even when your deployment is in a region where a language other than English is spoken. However, if you wish to replace the default file with a different AMD Sensitivity file containing customized detection algorithms, please contact your AudioCodes sales representative for more information.

The AMD Sensitivity file is created in .xml format and then converted to a binary .dat file that can be installed on the device. The XML-to-binary format conversion can be done using AudioCodes DConvert utility. For more information on using this utility, refer to *DConvert Utility User's Guide*. Only one AMD Sensitivity file can be installed on the device. To install a new AMD Sensitivity file, use any of the following methods:

- Web interface: On the Load Auxiliary Files page - see "Loading Auxiliary Files" on page 647.
- TFTP during initialization: You need to configure the *ini* file parameter, AMDSensitivityFileName, and then copy the AMD Sensitivity file to the TFTP directory.
- Automatic Update feature: For more information, see Automatic Update Mechanism. For this method, the AMDSensitivityFileUrl parameter must be set through SNMP or *ini* file.

For more information on the AMD feature, see "Answering Machine Detection (AMD)" on page 198.

41.2 Software License Key

The License Key determines the device's supported features and call capacity, as ordered from your AudioCodes sales representative. You can upgrade or change your device's supported features and capacity, by purchasing and installing a new License Key that match your requirements.

> **Note:** The availability of certain Web pages depends on the installed License Key.

41.2.1 Viewing the License Key

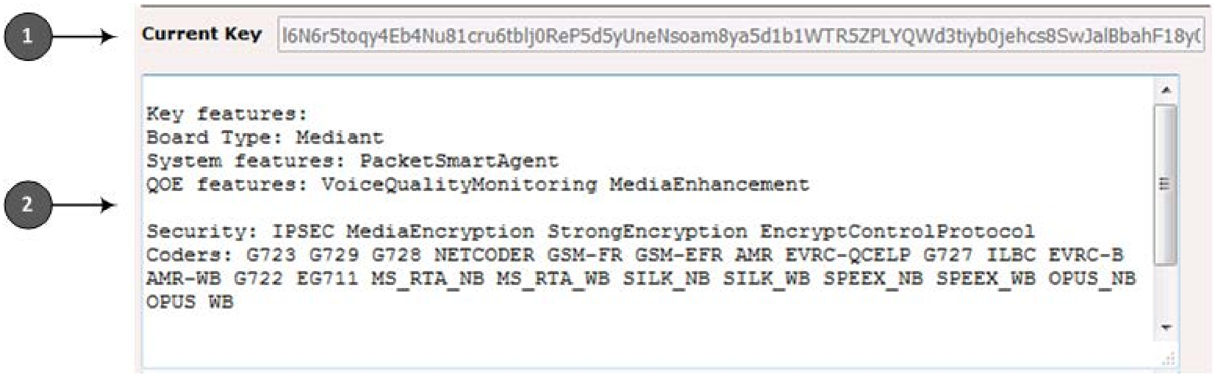
The following procedure describes how to view the device's License Key.

➢ **To view the License Key:**

- Open the Software Upgrade Key Status page (**Maintenance** tab > **Software Update** folder > **Software Upgrade Key**).

The License Key is displayed in encrypted-string format in the 'Current Key' field (1) and the main features provided by the License Key are displayed in the pane (2) below it, as shown in the example below:

Figure 41-5: Viewing License Key (Example)



Current Key l6N6r5toqy4Eb4Nu81cru6tblj0ReP5d5yUneNsoam8ya5d1b1WTR5ZPLYQWd3tiyb0jehcs8SwJalBbahF18y(

```
Key features:
Board Type: Mediant
System features: PacketSmartAgent
QOE features: VoiceQualityMonitoring MediaEnhancement

Security: IPSEC MediaEncryption StrongEncryption EncryptControlProtocol
Coders: G723 G729 G728 NETCODER GSM-FR GSM-EFR AMR EVRC-QCELP G727 ILBC EVRC-B
AMR-WB G722 EG711 MS_RTA_NB MS_RTA_WB SILK_NB SILK_WB SPEEX_NB SPEEX_WB OPUS_NB
OPUS WB
```

41.2.2 Installing a New Software License Key

This section describes how to install a new License Key on the device.

> **Note:** When you install a new License Key, it overwrites the previously installed License Key. Any license-based features that were included in the old License Key, but not included in the new License Key, will no longer be available.

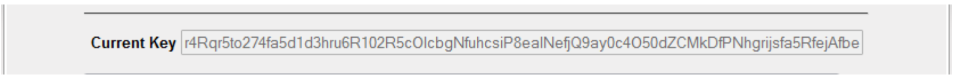
41.2.2.1 Installing Software License Key through Web Interface

The following procedure describes how to install the Software License Key through the Web interface.

➢ **To install the Software License Key through the Web interface:**

1. Open the Software Upgrade Key Status page (**Maintenance** tab > **Software Update** menu > **Software Upgrade Key**).
2. Back up the Software License Key currently installed on the device, as a precaution. If the new Software License Key does not comply with your requirements, you can re-load this backup to restore the device's original capabilities.
   a. In the 'Current Key' field, select the entire text string and copy it to any standard text file (e.g., Notepad):

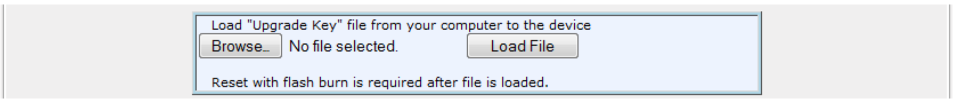
Figure 41-6: Current Key Field



Current Key r4Rqr5to274fa5d1d3hru6R102R5cOlcbgNfuhcsiP8ealNefjQ9ay0c4O50dZCMkDfPNhgrijsfa5RfejAfbe

   b. Save the text file with any file name and file extension (e.g., key.txt) to a folder on your computer.
3. If the device is operating in High-Availability (HA) mode, load the License Key as follows (otherwise, skip this step):
   a. Under the 'Load Upgrade Key file... text, click the **Browse** button, and then navigate to and select the License Key file on your computer:

Figure 41-7: Load File Button



Load "Upgrade Key" file from your computer to the device

No file selected.

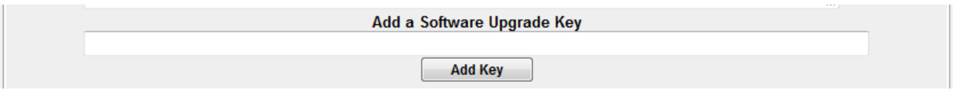
Reset with flash burn is required after file is loaded.

   b. Click **Load File**; the new License Key is installed on the device and saved to flash memory. The License Key is displayed in the 'Current Key' field.

Note: The License Key file for HA includes two License Keys - one for the active device and one for the redundant device. Each License Key has a different serial number ("S/N").

4. (For a non-HA standalone device only) Load the License Key as follows:
   a. Open the License Key file using a text-based program such as Notepad.
   b. Copy-and-paste the contents of the file into the 'Add a Software Upgrade Key' field.
   c. Click **Add Key**.

Figure 41-8: Add Key Button



Add a Software Upgrade Key

Verify that the Software License Key was successfully installed, by doing one of the following:

- In the Software Upgrade Key Status page, check that the listed features and capabilities activated by the installed Software License Key match those that were ordered.
- Access the Syslog server and ensure that the following message appears in the Syslog server:

  "S/N___ Key Was Updated. The Board Needs to be Reloaded with ini file\n"

5. Reset the device; the new capabilities and resources enabled by the Software License Key are active.

> **Note:** If the Syslog server indicates that the Software License Key was unsuccessfully loaded (i.e., the "SN_" line is blank), do the following preliminary troubleshooting procedures:
>
> 1. Open the Software License Key file and check that the "S/N" line appears. If it does not appear, contact AudioCodes.
> 2. Verify that you have loaded the correct file. Open the file and ensure that the first line displays "[LicenseKeys]".
> 3. Verify that the content of the file has not been altered.

41.2.2.2 Installing Software License Key through CLI

To install the Software License Key through CLI, use the following commands:

- To install the Software License Key:

```
(config-system)# feature-key <"string enclosed in double
quotation marks">
```

- To view the Software License Key:

```
show system feature-key
```

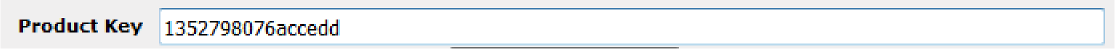
41.2.3 Viewing the Device's Product Key

The Product Key identifies a specific purchase of your device installation for the purpose of subsequent communication with AudioCodes (e.g., for support and software upgrades). The Product Key is your chassis' serial number--"S/N(Product Key)"--which also appears on the product label affixed to the chassis.

The Product Key is included in the License Key. Once the License Key is installed, you can view the Product Key in the following Web pages:

- Software Upgrade Key Status page (**Maintenance** tab > **Software Update** folder > **Software Upgrade Key**). The Product Key is displayed in the read-only 'Product Key' field, as shown in the example below:

Figure 41-9: Viewing Product Key



Product Key 1352798076accedd

- Device Information page (see Viewing Device Information on page 705).

41.3 Upgrading SBC Capacity Licenses by License Pool Manager Server

The device can receive SBC capacity licenses from a centralized pool of SBC resources managed by the License Pool Manager Server running on AudioCodes EMS. The License Pool Manager Server can dynamically allocate and de-allocate SBC capacity licenses from the pool to devices in the network to meet capacity demands of each device whenever required. The License Pool Manager Server holds a pool of customer-ordered SBC capacity (resource) licenses, which can include any of the following license types:

- SBC sessions (media and signaling)
- SBC signaling sessions
- SBC transcoding sessions
- SBC registrations (number of SIP endpoints that can register with the SBC)

Therefore, the device can be upgraded by the License Pool Manager Server with any of the above SBC license types.

Communication between the device and License Pool Manager Server is through HTTPS (port 443) and SNMP. If a firewall exists in the network, ensure that ports for these applications are opened. The device periodically checks with the License Pool Manager Server for SBC capacity licenses. The License Pool Manager Server identifies the device by serial number. If it has an SBC license for the device, it sends it to the device. If the device's installed Software Feature Key already includes SBC capacity figures, the SBC license allocated from the pool is simply added to it (but up to the device's maximum supported capacity capabilities). A device reset is required for the allocated SBC license to take effect.

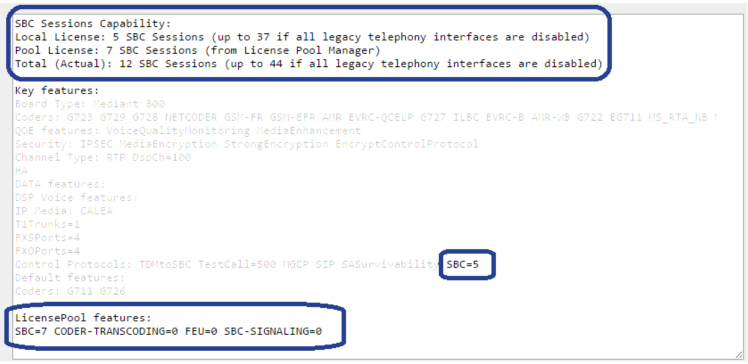
The Web interface's Software Upgrade Key Status page (**Maintenance** tab > **Software Update** menu > **Software Upgrade Key**) indicates the SBC license allocated by the License Pool Manager Server:

- "Local License": Number of SBC sessions according to the installed Software Feature Key file. The actual license is indicated on the page in the "SBC=" field (e.g., SBC=5, as shown in the example figure below).
- "Pool License": Number of SBC sessions allocated by the License Pool Manager Server.
- "Total (Actual)": Total number of SBC sessions permitted on the device based on the installed Software Feature Key file and the SBC sessions allocated by the License Pool Manager Server.
- "LicensePool features":
  - "SBC": Number of SBC sessions (media and signaling) allocated by the License Pool Manager Server.
  - "CODER-TRANSCODING": Number of SBC transcoding sessions allocated by the License Pool Manager Server.
  - "FEU": Number of SBC registrations allocated by the License Pool Manager Server.
  - "SBC-SIGNALING": Number of SBC signaling sessions allocated by the License Pool Manager Server.

The Software Upgrade Key Status page also displays the number of SBC sessions if all legacy telephony interfaces are disabled.

The following displays an example of the indication of SBC licenses allocated by the License Pool Manager Server in the Software Upgrade Key Status page:

Figure 41-10: Software Upgrade Key Status Page Displaying Licenses from License Pool



```
SBC Sessions Capability:
Local License: 5 SBC Sessions (up to 37 if all legacy telephony interfaces are disabled)
Pool License: 7 SBC Sessions (from License Pool Manager)
Total (Actual): 12 SBC Sessions (up to 44 if all legacy telephony interfaces are disabled)
Key features:
Board Type: Mediant 800
Coders: G723 G729 G728 NETCODER GSM-FR GSM-EFR AMR EVRC-QCELP G727 ILBC EVRC-B AMR-WB G722 EG711 MS_RTA_NB
QOE features: VoiceQualityMonitoring MediaEnhancement
Security: IPSEC MediaEncryption StrongEncryption EncryptControlProtocol
Channel Type: RTP DspCh=100
HA
DATA features:
DSP Voice features:
IP Media: CALEA
T1Trunks=1
FXSPorts=4
FXOPorts=4
Control Protocols: TDMtoSBC TestCall=500 MGCP SIP SASurvivability SBC=5
Default features:
Coders: G711 G726

LicensePool features:
SBC=7 CODER-TRANSCODING=0 FEU=0 SBC-SIGNALING=0
```

If communication with the License Pool Manager Server is lost for a long duration, the device discards the allocated SBC license (i.e., expires) and resets with its initial, "local" SBC license. This mechanism prevents misuse of SBC licenses allocated by the License Pool Manager Server.

The following SNMP alarms relate to the allocation/de-allocation of SBC licenses by the License Pool Manager Server:

- acLicensePoolInfraAlarm (1.3.6.1.4.1.5003.9.10.1.21.2.0.106):
  - Sent when the device receives a new SBC license from the License Pool Manager Server and a device reset is required.
  - Sent when the device is unable to access the License Pool Manager Server.
  - Sent when the SBC license allocated by the License Pool Manager Server is about to expire (e.g., when communication with the License Pool Manager Server is lost)
- acLicensePoolApplicationAlarm (1.3.6.1.4.1.5003.9.10.1.21.2.0.107):
  - Sent when the device receives an SBC license from the License Pool Manager Server that exceeds the maximum SBC session capacity that can be supported by the device.
  - Sent when the device resets with an SBC license allocated by the License Pool Manager Server that exceeds the maximum SBC session capacity that can be supported by the device. The device sets the capacity to its maximum (and values beyond the device's capability are not applied)

Notes:

- No configuration is required on the device; the License Pool Manager Server controls the allocation/de-allocation of its resource pool to the managed devices. For more information on the License Pool Manager Server, refer to the *EMS User's Manual*.
- The allocation/de-allocation of SBC licenses to the device by the License Pool Manager Server is service affecting and requires a device reset.
- If the device is restored to factory defaults, the SBC license allocated by the License Pool Manager Server is deleted.
- If the device is allocated an SBC license by the License Pool Manager Server that exceeds the maximum number of sessions that it can support, the device sets the number of sessions to its maximum supported

41.4 Software Upgrade Wizard

The Web interface's Software Upgrade Wizard lets you easily upgrade the device's software version (.cmp file). The wizard also provides you the option to load other files such as an *ini* file and Auxiliary files (e.g., Call Progress Tone / CPT file). However, loading a .cmp file is mandatory through the wizard and before you can load any other type of file, the .cmp file must be loaded.

Notes:

- You can obtain the latest software files from AudioCodes Web site at http://www.audiocodes.com/downloads.
- When you start the wizard, the rest of the Web interface is unavailable. After the files are successfully installed with a device reset, access to the full Web interface is restored.
- If you upgraded your firmware (.cmp file) and the "SW version mismatch" message appears in the Syslog or Web interface, your Software License Key does not support the new .cmp file version. If this occurs, contact AudioCodes support for assistance.
- If the device disconnects from the power source (e.g., power outage or disconnection of the power cable) during the upgrade process, the upgrade process fails and when the device is powered up again, it runs with the previously installed software version.
- Instead of manually upgrading the device, you can use the device's Automatic Update feature for automatic provisioning (see ''Automatic Provisioning'' on page 679).
- You can also upgrade the device's firmware by loading a .cmp file from an external USB hard drive connected to the device's USB port. For more information, see USB Storage Capabilities on page 701.

The following procedure describes how to load files using the Web interface's Software Upgrade Wizard. Alternatively, you can load files using the CLI:

- cmp file:

```
copy firmware from <URL>
```

- ini or Auxiliary file:

```
copy <ini file or auxiliary file> from <URL>
```

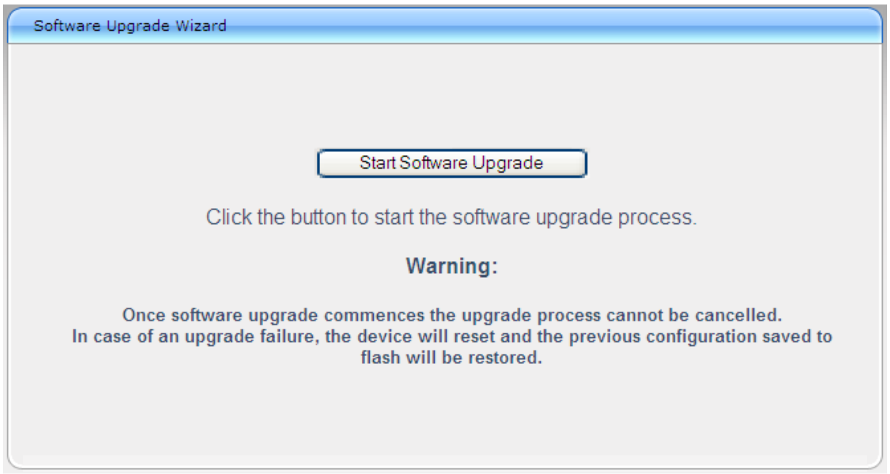
- CLI script file:

```
copy cli-script from <URL>
```

➢ To upgrade the device using the Software Upgrade Wizard:

1. Make sure that you have installed a new Software License Key (see "Software License Key" on page 668) that is compatible with the software version to be installed.
2. It is recommended to enable the Graceful Lock feature (see "Locking and Unlocking the Device" on page 642). The wizard resets the device at the end of the upgrade process, thereby causing current calls to be untimely terminated. To minimize this traffic disruption, the Graceful Lock feature prevents the establishment of new calls.
3. It is recommended to save a copy of the device's configuration to your computer. If an upgrade failure occurs, you can restore your configuration settings by uploading the backup file to the device. For saving and restoring configuration, see "Backing Up and Loading Configuration File" on page 677.
4. Open the Software Upgrade wizard, by performing one of the following:
   - Select the **Maintenance** tab, click the **Software Update** menu, and then click **Software Upgrade Wizard**.
   - On the toolbar, click **Device Actions**, and then choose **Software Upgrade Wizard**.

Figure 41-11: Start Software Upgrade Wizard Screen


Software Upgrade Wizard

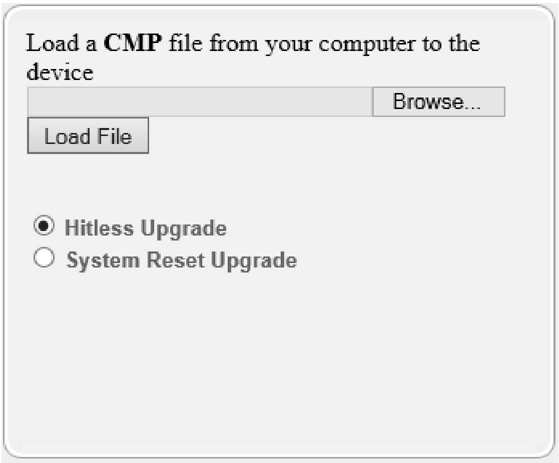
Start Software Upgrade

Click the button to start the software upgrade process.

Warning:

Once software upgrade commences the upgrade process cannot be cancelled.
In case of an upgrade failure, the device will reset and the previous configuration saved to flash will be restored.

5. Click **Start Software Upgrade**; the wizard starts, prompting you to load a .cmp file:


Load a CMP file from your computer to the device

Browse...

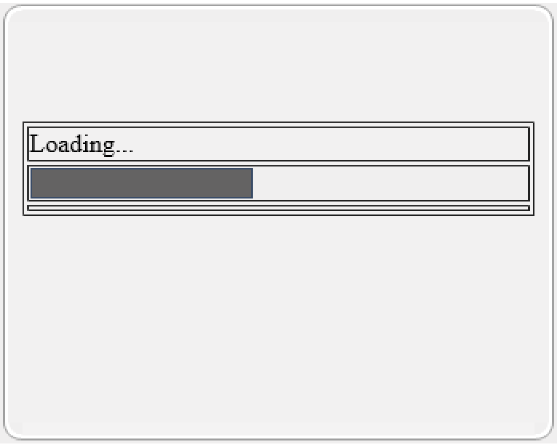
Load File

● Hitless Upgrade
○ System Reset Upgrade

Note: At this stage, you can quit the Software Upgrade Wizard without having to reset the device, by clicking **Cancel** ☒. However, if you continue with the wizard and start loading the cmp file, the upgrade process must be completed with a device reset.

6. Click **Browse**, and then navigate to where the .cmp file is located on your computer. Select the file, and then click **Open**.

7. Click **Load File**; the device begins to install the .cmp file. A progress bar displays the status of the loading process and a message informs you when file load successfully completes:


Loading...

8. If you want to load additional files, skip this step and continue with the next step. If you **only** want to load a .cmp file, click **Reset** ⟳ ; the device burns the .cmp file to its flash memory and then resets. The device uses the existing configuration (*ini*) and Auxiliary files.

⚠ **Note:** Device reset may take a few minutes (even up to 30 minutes), depending on cmp file version.

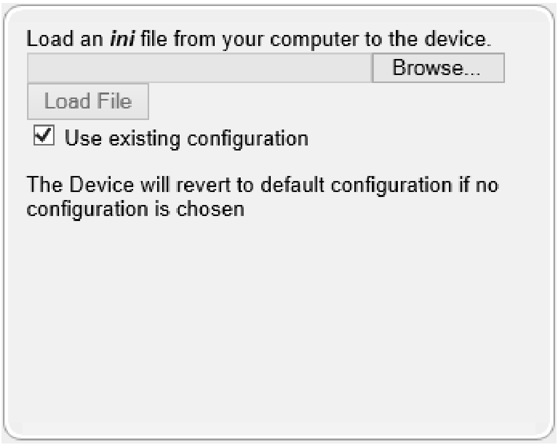
9. To load additional files, use the **Next** ▶ and **Back** ◀ buttons to navigate through the wizard to the desired file-load wizard page. Alternatively, you can navigate to the relevant file-load wizard page by clicking the respective file-name buttons listed in the left pane of the wizard pages.

10. The wizard page for loading an *ini* file provides you with the following options:

    - **Load a new ini file:** In the 'Load an ini file...' field, click **Browse**, and then navigate to where the ini file is located on your computer. Select the file, and then click **Load File**; the device loads the *ini* file.
    - **Retain the existing configuration (default):** Select the 'Use existing configuration' check box to use the current configuration (and do not select an ini file).

- **Restore configuration to factory defaults:** Clear the 'Use existing configuration' check box (and do not select an ini file).

Figure 41-12: Software Upgrade Wizard - Load INI File



Load an ***ini*** file from your computer to the device.

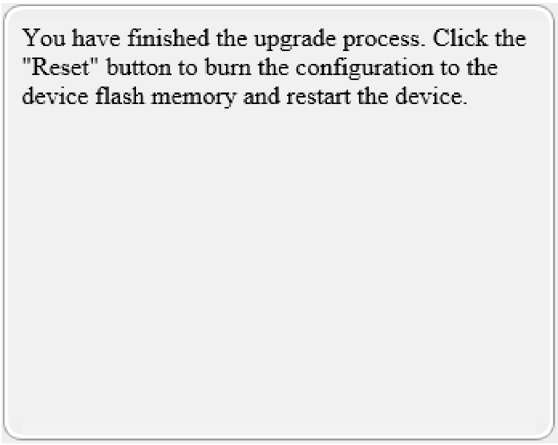
☑ Use existing configuration

The Device will revert to default configuration if no configuration is chosen

Note: If you use the wizard to load an *ini* file, parameters excluded from the *ini* file are assigned default values (according to the .cmp file running on the device) and thereby, overwrite values previously configured for these parameters.

11. When you have completed loading all the desired files, click **Next** ▶ until the last wizard page appears (the **FINISH** button is highlighted in the left pane):

Figure 41-13: Software Upgrade Wizard - Files Loaded



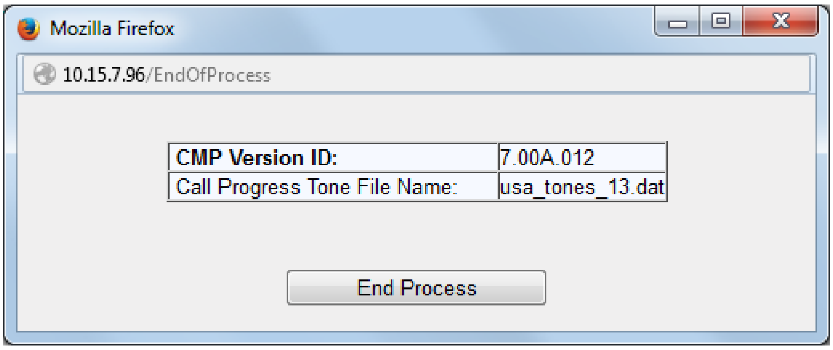
You have finished the upgrade process. Click the "Reset" button to burn the configuration to the device flash memory and restart the device.

12. Click **Reset** 🔄 to burn the files to the device's flash memory; the "Burn and reset in progress" message is displayed and the device 'burns' the newly loaded files to flash memory and then resets.

Note: Device reset may take a few minutes (even up to 30 minutes), depending on .cmp file version.

When the device finishes the installation process and resets, the following wizard page is displayed, showing the installed software version and other files (ini file and Auxiliary files) that you may also have installed:

Figure 41-14: Software Upgrade Process Completed Successfully (Example)



CMP Version ID:	7.00A.012
Call Progress Tone File Name:	usa_tones_13.dat

13. Click **End Process** to close the wizard; the Web Login dialog box appears.
14. Enter your login username and password, and then click **Login**; a message box appears informing you of the new .cmp file version.
15. Click **OK**; the Web interface becomes active, reflecting the upgraded device.

41.5 Backing Up and Loading Configuration File

You can save a copy/backup of the device's current configuration settings as an *ini* file to a folder on your computer, using the Configuration File page. The saved file includes only parameters that were modified and parameters with other than default values. The Configuration File page also allows you to load an *ini* file to the device. If the device has "lost" its configuration, you can restore the device's configuration by loading the previously saved *ini* file or by simply loading a newly created *ini* file.

You can also save the current configuration to a remote server or USB and update configuration from an external USB hard drive connected to the device's USB port. For more information, see USB Storage Capabilities on page 701.

```
# copy cli-script to <URL of TFTP/HTTP/HTTPS server or USB>
```

For example:

- Remote server:

  ```
  # copy cli-script to tftp://192.168.0.3/config-device1.txt
  ```

- USB:

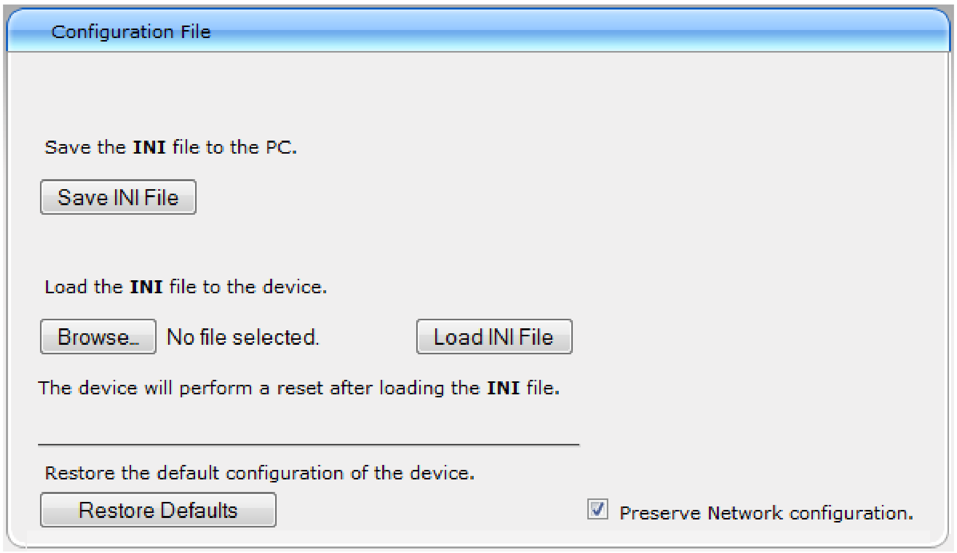
  ```
  # copy cli-script to usb://config-device1.txt
  ```

Note: When loading an *ini* file using the Configuration File page, parameters not included in the *ini* file are reset to default settings.

➢ To save or load an ini file:

1. Open the Configuration File page by doing one of the following:
   - From the Navigation tree, click the **Maintenance** tab, click the **Software Update** menu, and then click **Configuration File**.
   - On the toolbar, click **Device Actions**, and then from the drop-down menu, choose **Load Configuration File** or **Save Configuration File**.

Figure 41-15: Configuration File Page



Configuration File

Save the **INI** file to the PC.

Load the **INI** file to the device.

The device will perform a reset after loading the **INI** file.

Restore the default configuration of the device.

☑ Preserve Network configuration.

2. To save the *ini* file to a folder on your computer:
   a. Click the **Save INI File** button; the File Download dialog box appears.
   b. Click the **Save** button, navigate to the folder where you want to save the file, and then click **Save**.
3. To load the *ini* file to the device:
   a. Click the **Browse** button, navigate to the folder where the file is located, select the file, and then click **Open**; the name and path of the file appear in the field beside the **Browse** button.
   b. Click the **Load INI File** button, and then at the prompt, click **OK**; the device uploads the file and then resets. Once complete, the Web Login screen appears, requesting you to enter your user name and password.

42 Automatic Provisioning

This chapter describes the device's automatic provisioning mechanisms.

42.1 Automatic Configuration Methods

The table below summarizes the automatic provisioning methods supported by the device:

Table 42-1: Automatic Provisioning Methods

BootP / TFTP	DHCP		Automatic Update Methods				SNMP (EMS)
	67	66	HTTP/S	TFTP	FTP	NFS	
No	Yes	Yes	Yes	Yes	Yes	No	Yes

42.1.1 DHCP-based Provisioning

A third-party DHCP server can be configured to automatically provide each device, acting as a DHCP client, with a temporary IP address so that individual MAC addresses are not required. The DHCP server can provide additional networking parameters such as subnet mask, default gateway, primary and secondary DNS server, and two SIP server addresses. These network parameters have a time limit, after which the device must 'renew' its lease from the DHCP server.

The device can use a host name in the DHCP request. The host name is set to acl_nnnnn, where *nnnnn* denotes the device's serial number. The serial number is the last six digits of the MAC address converted to decimal representation. In networks that support this feature and if the DHCP server registers this host name to a DNS server, you can access the device (through a Web browser) using the URL, http://acl_<serial number> (instead of using the device's IP address). For example, if the device's MAC address is 00908f010280, the DNS name is acl_66176.

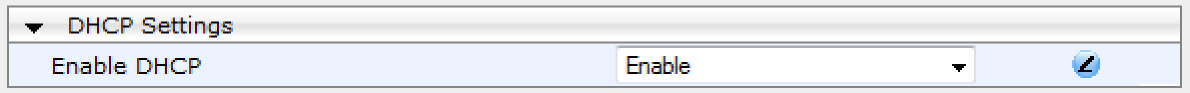
Notes:

- When using DHCP to acquire an IP address, the Interface table, VLANs and other advanced configuration options are disabled.
- For additional DHCP parameters, see "DHCP Parameters" on page 836.

➢ **To enable the device as a DHCP client:**

1. Open the Application Settings page (**Configuration** tab > **System** menu > **Application Settings**).

Figure 42-1: Enabling DHCP - Application Settings Page



DHCP Settings

Enable DHCP: Enable

2. From the 'Enable DHCP" drop-down list, select **Enable**.
3. Click **Submit**.
4. To activate the DHCP process, reset the device.

The following shows an example of a configuration file for a Linux DHCP server (dhcpd.conf). The devices are allocated temporary IP addresses in the range 10.31.4.53 to 10.31.4.75.

TFTP is assumed to be on the same computer as the DHCP server (alternatively, the "next-server" directive may be used).

```
ddns-update-style ad-hoc;
default-lease-time 60;
max-lease-time 60;
class "gateways" {
    match if(substring(hardware, 1, 3) = 00:90:8f);
}
subnet 10.31.0.0 netmask 255.255.0.0 {
    pool {
          allow members of "audiocodes";
          range 10.31.4.53 10.31.4.75;
          filename "SIP_F6.60A.217.003.cmp –fb;device.ini";
          option routers                  10.31.0.1;
          option subnet-mask              255.255.0.0;
    }
}
```

Notes:

- If the DHCP server denies the use of the device's current IP address and specifies a different IP address (according to RFC 1541), the device must change its networking parameters. If this occurs while calls are in progress, they are not automatically rerouted to the new network address. Therefore, administrators are advised to configure DHCP servers to allow renewal of IP addresses.
- If the device's network cable is disconnected and then reconnected, a DHCP renewal is performed (to verify that the device is still connected to the same network). The device also includes its product name in the DHCP Option 60 Vendor Class Identifier. The DHCP server can use this product name to assign an IP address accordingly.
- After power-up, the device performs two distinct DHCP sequences. Only in the second sequence is DHCP Option 60 included. If the device is software reset (e.g., from the Web interface or SNMP), only a single DHCP sequence containing Option 60 is sent.

42.1.1.1 Provisioning from HTTP Server using DHCP Option 67

Most DHCP servers support the configuration of individual DHCP option values for different devices on the network. The DHCP configuration should be modified so that the device receives a URL to the configuration file in Option 67, along with IP addressing and DNS server information. The DHCP response is processed by the device upon startup and the device automatically downloads the configuration file from the HTTP server specified in the DHCP response. This method is NAT-safe.

Below is an example of a Linux DHCP configuration file (dhcpd.conf) showing the required format of Option 67:

```
ddns-update-style ad-hoc;
default-lease-time 3600;
max-lease-time 3600;
class "audiocodes" {
    match if(substring(hardware, 1, 3) = 00:90:8f);
}
subnet 10.31.0.0 netmask 255.255.0.0 {
    pool {
          allow members of "audiocodes";
          range 10.31.4.53 10.31.4.75;
          option routers                    10.31.0.1;
          option subnet-mask                255.255.0.0;
          option domain-name-servers        10.1.0.11;
          option bootfile-name
"INI=http://www.corp.com/master.ini";
          option dhcp-parameter-request-list 1,3,6,51,67;
    }
}
```

42.1.1.2 Provisioning from TFTP Server using DHCP Option 66

This method is suitable when the network in which the device is deployed contains a provisioning TFTP server for all network equipment, without being able to distinguish between AudioCodes and non-AudioCodes devices.

Upon startup, the device searches for Option 66 in the DHCP response from the DHCP server. If Option 66 contains a valid IP address, the device attempts to download, through TFTP, a file that has a filename containing the device's MAC address (e.g., 00908f0130aa.ini). This method requires a provisioning server at the customer premises.

This method loads the configuration file to the device as a one-time action. The download is only repeated if the device is manually restored to factory defaults (by pressing the hardware reset button while the Ethernet cable is not connected) and DHCP is enabled (see note below).

Notes:

- For TFTP configuration using DHCP Option 66, enable DHCP on your device: DHCPEnable = 1 and DHCPRequestTFTPParams = 1.
- Access to the core network using TFTP is not NAT-safe.
- The TFTP data block size (packets) when downloading a file from a TFTP server for the Automatic Update mechanism can be configured using the AUPDTftpBlockSize parameter.

42.1.2 HTTP-based Provisioning

An HTTP or HTTPS server can be located in the network in which the device is deployed, storing configuration and software files for the device to download. This does not require additional servers and is NAT-safe.

For example, assume the core network HTTPS server is https://www.corp.com. A master configuration ini file can be stored on the server, e.g., https://www.corp.com/gateways/master.ini. This file could point to additional ini files, Auxiliary files (e.g., call progress tones), and software files (cmp), all on the same HTTP server or different HTTP servers in the network.

The main advantage of this method is that the device can be configured to periodically check the HTTP server for file updates. HTTP(S) is not sensitive to NAT devices, enabling configuration whenever needed without on-site intervention. For additional security, the URL may contain a different port, and username and password.

The only configuration required is to preconfigure the device(s) with the URL of the initial (master) ini file. This can be done using one of the following methods:

- DHCP as described in "DHCP-based Provisioning" on page 679 or via TFTP at a staging warehouse. The URL is configured using the IniFileURL parameter.
- Private labeling (preconfigured during the manufacturing process).

- Using DHCP Option 67 (see Provisioning from HTTP Server using DHCP Option 67 on page 680).
- Manually on-site, using the RS-232 port or Web interface.

When the device is deployed at the customer site, local DHCP server provides the devices with IP addressing and DNS server information. From the URL provided in the DHCP response, the device can then contact the HTTP server at the core network and automatically download its configuration. The URL can be a simple file name or contain the device's MAC or IP address, e.g.:

- *http://corp.com/config-<MAC>.ini* - which becomes, for example, http://corp.com/config-00908f030012.ini
- *http://corp.com/<IP>/config.ini* - which becomes, for example, http://corp.com/192.168.0.7/config.ini

For more information on HTTP-based provisioning, see "HTTP/S-Based Provisioning using the Automatic Update Feature" on page 682.

42.1.3 FTP- based Provisioning

Some networks block access to HTTP(S). The Automatic Update feature provides limited support for FTP/FTPS connectivity. Periodic polling for updates is not possible since these protocols do not support conditional fetching, i.e., updating files only if it is changed on the server.

The only difference between this method and those described in "HTTP-based Provisioning" on page 681 and Provisioning from HTTP Server using DHCP Option 67 on page 680 is that the protocol in the URL is "ftp" (instead of "http").

42.1.4 Provisioning using AudioCodes EMS

AudioCodes EMS server functions as a core-network provisioning server. The device's SNMP Manager should be configured with the IP address of the EMS server, using one of the methods detailed in the previous sections. As soon as a registered device contacts the EMS server through SNMP, the EMS server handles all required configuration automatically, upgrading software as needed. This alternative method doesn't require additional servers at the customer premises, and is NAT-safe.

42.2 HTTP/S-Based Provisioning using the Automatic Update Feature

The Automatic Update feature can be used for automatic provisioning of the device through HTTP/S. Automatic provisioning is useful for large-scale deployment of devices. In some cases, the devices are shipped to the end customer directly from the manufacturer. In other cases, they may pass through a staging warehouse. Configuration may occur at the staging warehouse or at the end-customer premises.

The device may be preconfigured during the manufacturing process (commonly known as private labeling). Typically, a two-stage configuration process is implemented whereby initial configuration includes only basic configuration, while the final configuration is done only when the device is deployed in the live network.

Warning: If you use the IniFileURL parameter for the Automatic Update feature, do not use the Web interface to configure the device. If you do configure the device through the Web interface and save (burn) the new settings to the device's flash memory, the IniFileURL parameter is automatically set to 0 and Automatic Updates is consequently disabled. To enable Automatic Updates again, you need to re-load the ini file (using the Web interface or BootP) with the correct IniFileURL settings. As a safeguard to an unintended burn-to-flash when resetting the device, if the device is configured for Automatic Updates, the 'Burn To FLASH' field under the Reset Configuration group in the Web interface's Maintenance Actions page is automatically set to No by default.

Notes:

- For a description of all the Automatic Update parameters, see "Automatic Update Parameters" on page 827 or refer to the CLI Reference Guide.
- For additional security, use HTTPS or FTPS. The device supports HTTPS (RFC 2818) and FTPS using the AUTH TLS method <draft-murray-auth-ftp-ssl-16>.

42.2.1 Files Provisioned by Automatic Update

You can use the Automatic Update feature to update the device with any of the following files:

- Software file (*cmp)*
- Auxiliary files (e.g., Call Progress Tones, SSL Certificates, SSL Private Key)
- Configuration file:
  - ini File: Contains only ini file parameters and configures all the device's functionalities
  - CLI Script File: Contains only CLI commands and configures all the device's functionalities (except commands such as show, debug or copy). The file updates the device's configuration only according to the configuration settings in the file. The device's existing configuration settings (not included in the file) are retained. The device does not undergo a reset and therefore, this file typically contains configuration settings that do not require a device reset. If a reset is required, for example, to apply certain settings, you must include the following CLI command (root level) at the end of the file:

```
# reload if-needed
```

42.2.2 File Location for Automatic Update

The files for updating the device can be stored on any standard Web (HTTP/S), TFTP or FTP NFS server. The files can be loaded periodically to the device using HTTP, HTTPS, FTP, or TFTP. This mechanism can be used even when the device is installed behind NAT and firewalls.

The Automatic Update feature is done per file and configured by specifying the file name and URL address of the provisioning server where the file is located. For a description of the parameters used to configure URLs per file, see "Automatic Update Parameters" on page 827. Below are examples for configuring the file names and their URLs for Automatic Update:

- ini File:

```
IniFileURL = 'http://www.corp.com/configuration.ini'
CptFileURL = 'http://www.corp.com/call_progress.dat'
FeatureKeyURL = 'https://www.company.com/License_Key.txt'
AutoCmpFileUrl = 'http://www.corp.com/SIP_F7.00A.008.cmp
```

- CLI:

```
# configure system
(config-system)# automatic update
(automatic-update)# cli-script https://company.com/cli/<MAC>
(automatic-update)# voice-configuration http://www.company.com/configuration.ini
( automatic-update)# feature-key http://www.company.com/License_Key.txt
(automatic-update)# call-progress-tones http://www.company.com/call_progress.dat
(automatic-update)# auto-firmware http://www.company.com/SIP_F7.00A.008.cmp
```

Note: For configuration files (ini), the file name in the URL can automatically contain the device's MAC address for enabling the device to download a file unique to the device. For more information, see "MAC Address Placeholder in Configuration File Name" on page 689.

42.2.3 Triggers for Automatic Update

The Automatic Update feature can be triggered by the following:

- Upon device startup (reset or power up). To disable this trigger, run the following CLI command:

```
(config-system)# automatic-update
(automatic-update)# run-on-reboot off
```

- Periodically:
  - Specified time of day (e.g., 18:00), configured by the ini file parameter AutoUpdatePredefinedTime or CLI command configure system > automatic-update > predefined-time.
  - Interval between Automatic Updates (e.g., every 60 minutes), configured by the ini file parameter AutoUpdateFrequency or CLI command configure system > automatic-update > update-frequency.
- Centralized provisioning server request:
  - Upon receipt of an SNMP request from the provisioning server.
  - Upon receipt of a special SIP NOTIFY message from the provisioning server. The NOTIFY message includes an Event header with the AudioCodes proprietary value, "check-sync;reboot=false", as shown in the example below:

```
NOTIFY sip:<user>@<dsthost> SIP/2.0
To: sip:<user>@<dsthost>
From: sip:sipsak@<srchost>
CSeq: 10 NOTIFY
Call-ID: 1234@<srchost>
Event: check-sync;reboot=false
```

To enable this feature through the Web interface:

a. Open the Advanced Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Advanced Parameters**).

b. Under the **Misc Parameters** group, set the 'SIP Remote Reset' parameter to **Enable**.

c. Click **Submit**.

To enable through CLI: configure voip > sip-definition advanced-settings > sip-remote-reset.

42.2.4 Access Authentication with HTTP Server

You can configure the device to authenticate itself with the HTTP/S server. The device authenticates itself by providing the HTTP/S server with its authentication username and password. You can configure one of the following HTTP authentication schemes:

- **Basic Access Authentication:** The device provides its username and password to the HTTP server. The username and password is configured in the URL that you define for downloading the file:
  - ini file:

    ```
    AutoCmpFileUrl = 'https://<username>:<password>@<IP address
    or domain name>/<file name>'
    ```

  - CLI:

    ```
    # configure system
    (config-system)# automatic update
    (automatic-update)# auto-firmware https://<username>:<password>@<IP
    address or domain name>/<file name>
    ```

- **Digest Access Authentication:** The authentication username and password is negotiated between the device and HTTP/S server, using digest MD5 cryptographic hashing. This method is safer than basic access authentication. The digest authentication username and password are configured using the AUPDDigestUsername and AUPDDigestPassword parameters, respectively.

42.2.5 Querying Provisioning Server for Updated Files

Each time the Automatic Update feature is triggered, for each file and its configured URL the device does the following:

1. If you have configured the device to authenticate itself to the HTTP/S server for secure access, the device sends the access authentication username and password to the HTTP/S server (for more information, see "Access Authentication with HTTP Server" on page 685). If authentication succeeds, Step 2 occurs.
2. The device establishes an HTTP/S connection with the URL host (provisioning server). If the connection is HTTPS, the device verifies the certificate of the provisioning server, and presents its own certificate if requested by the server.
3. The device queries the provisioning server for the requested file by sending an HTTP Get request. This request contains the HTTP User-Agent Header, which identifies the device to the provisioning server. By default, the header includes the device's model name, MAC address, and currently installed software and configuration versions. Based on its own dynamic applications for logic decision making, the provisioning server uses this information to check if it has relevant files available for the device and determines which files must be downloaded (working in conjunction with the HTTP If-Modified-Since header, described further on in this section).

   You can configure the information sent in the User-Agent header, using the AupdHttpUserAgent parameter or CLI command, configure system > http-user-agent. The information can include any user-defined string or the following supported string variable tags (case-sensitive):

   - **<NAME>:** product name, according to the installed Software License Key
   - **<MAC>:** device's MAC address
   - **<VER>:** software version currently installed on the device, e.g., "7.00.200.001"
   - **<CONF>:** configuration version, as configured by the ini file parameter, INIFileVersion or CLI command, configuration-version

   The device automatically populates these tag variables with actual values in the sent header. By default, the device sends the following in the User-Agent header:

   ```
   User-Agent: Mozilla/4.0 (compatible; AudioCodes;
   <NAME>;<VER>;<MAC>;<CONF>)
   ```

For example, if you set AupdHttpUserAgent = MyWorld-<NAME>;<VER>(<MAC>), the device sends the following User-Agent header:

```
User-Agent: MyWorld-Mediant;7.00.200.001(00908F1DD0D3)
```

Note: If you configure the AupdHttpUserAgent parameter with the <CONF> variable tag, you must reset the device with a burn-to-flash for your settings to take effect.

4. If the provisioning server has relevant files available for the device, the following occurs, depending on file type and configuration:

   - **File Download upon each Automatic Update process:** This is applicable to software (.cmp), ini files. In the sent HTTP Get request, the device uses the HTTP If-Modified-Since header to determine whether to download these files. The header contains the date and time (timestamp) of when the device last downloaded the file from the specific URL. This date and time is regardless of whether the file was installed or not on the device. An example of an If-Modified-Since header is shown below:

     ```
     If-Modified-Since: Mon, 1 January 2014 19:43:31 GMT
     ```

     If the file on the provisioning server was unchanged (modified) since the date and time specified in the header, the server replies with an HTTP 304 response and the file is not downloaded. If the file was modified, the provisioning server sends an HTTP 200 OK response with the file in the body of the HTTP response. The device downloads the file and compares the version of the file with the currently installed version on its flash memory. If the downloaded file is of a later version, the device installs it after the device resets (which is only done after the device completes all file downloads); otherwise, the device does not reset and does not install the file.

     To enable the automatic software (.cmp) file download method based on this timestamp method, use the ini file parameter, AutoCmpFileUrl or CLI command, configure system > automatic-update > auto-firmware <URL>. The device uses the same configured URL to download the .cmp file for each subsequent Automatic Update process.

     You can also enable the device to run a CRC on the downloaded configuration file (ini) to determine whether the file has changed in comparison to the previously downloaded file. Depending on the CRC result, the device can install or discard the downloaded file. For more information, see ''Cyclic Redundancy Check on Downloaded Configuration Files'' on page 688.

Notes:

- When this method is used, there is typically no need for the provisioning server to check the device's current firmware version using the HTTP-User-Agent header.
- The Automatic Update feature assumes that the Web server conforms to the HTTP standard. If the Web server ignores the If-Modified-Since header or doesn't provide the current date and time during the HTTP 200 OK response, the device may reset itself repeatedly. To overcome this problem, modify the update frequency, using the ini file parameter AutoUpdateFrequency or CLI command configure system > automatic update > update-frequency.

- **One-time File Download:** This is applicable to software (.cmp) and Auxiliary (e.g., License Key, CPT and Dial Plan) files. The device downloads these files only **once**, regardless of how many times the device may repeat the Automatic Update process. Once they are downloaded, the device discards their configured URLs. To update these files again, you need to configure their URL addresses and filenames again. Below is an example of how to configure URLs for some of these files:

  Auxiliary Files:

  - ini:

    ```
    CptFileURL =
    'https://www.company.com/call_progress.dat'
    FeatureKeyURL =
    'https://www.company.com/License_Key.txt'
    ```

  - CLI:

    ```
    (config-system)# automatic-update
    (automatic-update)# call-progress-tones
    http://www.company.com/call_progress.dat
    (automatic-update)# tls-root-cert https://company.com/root.pem
    ```

  Software (.cmp) File:

  - ini:

    ```
    CmpFileUrl =
    'https://www.company.com/device/v.6.80A.227.005.cmp'
    ```

  - CLI:

    ```
    (config-system)# automatic-update
    (automatic-update)# firmware
    https://www.company.com/device/v.6.80A.227.005.cmp
    ```

Notes:

- For one-time file download, the HTTP Get request sent by the device does not include the If-Modified-Since header. Instead, the HTTP-User-Agent header can be used in the HTTP Get request to determine whether firmware update is required.
- When downloading SSL certificates (Auxiliary file), it is recommended to use HTTPS with mutual authentication for secure transfer of the SSL Private Key.
- After the device downloads the License Key file, it checks that the serial number in the file (“S/N <serial number>") is the same as that of the device. If the serial number is the same and the license key is different to the one currently installed on the device, it applies the new License Key.

5. If the device receives an HTTP 301/302/303 redirect response from the provisioning server, it establishes a connection with the new server at the redirect URL and re-sends the HTTP Get request.

42.2.6 File Download Sequence

Whenever the Automatic Update feature is triggered (see "Triggers for Automatic Update" on page 684), the device attempts to download each file from the configured URLs, in the following order:

1. ini file
2. CLI Script file
3. Periodic software file (.cmp) download
4. One-time software file (.cmp) download

5. Auxiliary file(s)

The following files automatically instruct the device to reset:

- Periodic software file (.cmp)
- One-time software file (.cmp)

When multiple files requiring a reset are downloaded, the device resets only **after** it has downloaded and installed **all** the files. However, you can explicitly instruct the device to immediately reset for the following files:

- ini file: Use the ResetNow in file parameter
- CLI Script file: Use the reload if-needed CLI command

Warning: If you use the ResetNow parameter in an ini file for periodic automatic provisioning with non-HTTP (e.g., TFTP) and without CRC, the device resets after every file download. Therefore, use the parameter with caution and only if necessary for your deployment requirements.

Notes:

- For ini file downloads, by default, parameters not included in the file are set to defaults. To retain the current settings of these parameters, set the SetDefaultOnINIFileProcess parameter to 0.
- If you have configured one-time software file (.cmp) download (configured by the ini file parameter CmpFileURL or CLI command configure system > automatic-update > firmware), the device will only apply the file if one-time software updates are enabled. This is disabled by default to prevent unintentional software upgrades. To enable one-time software upgrades, set the ini file parameter AutoUpdateCmpFile to 1 or CLI command, configure system > automatic-update > update-firmware on.
- If you need to update the device's software and configuration, it is recommended to first update the software. This is because the current ("old") software (before the upgrade) may not be compatible with the new configuration. However, if both files are available for download on the provisioning server(s), the device first downloads and applies the new configuration, and only then does it download and install the new software. Therefore, this is a very important issue to take into consideration.
- If more than one file needs to be updated - CLI Script and cmp: The device downloads and applies the CLI Script file on the currently ("old") installed software version. It then downloads and installs the cmp file with a reset. Therefore, the CLI Script file MUST have configuration compatible with the "old" software version.

42.2.7 Cyclic Redundancy Check on Downloaded Configuration Files

You can enable the device to perform cyclic redundancy checks (CRC) on downloaded configuration files (ini) during the Automatic Update process. The CRC checks whether the content (raw data) of the downloaded file is different to the content of the previously downloaded file from the previous Automatic Update process. The device compares the CRC check value (code) result with the check value of the previously downloaded file. If the check values are identical, it indicates that the file has no new configuration settings, and the device discards the file. If the check values are different, it indicates that the downloaded file is different (i.e., includes updates), and the device installs the downloaded file and applies the new configuration settings.

CRC is useful, for example, when the service provider replaces a file, on the provisioning server, with another file whose contents are the same. When the device sends an HTTP Get request during the Automatic Update process, the provisioning server sends the new file to the device. This occurs as the timestamp between the previously downloaded file and this new file is different (determined by the HTTP If-Modified-Since header in the Get request). Therefore, the CRC feature can be used to prevent the device from installing such files.

For enabling CRC, use the ini file parameter AUPDCheckIfIniChanged or CLI command, configure system > automatic-update > crc-check regular. By default, CRC is disabled. For more information on the parameter, see ''Automatic Update Parameters'' on page 827.

42.2.8 MAC Address Placeholder in Configuration File Name

You can configure the file name of the configuration file (ini) in the URL to automatically include the MAC address of the device. As described in ''File Location for Automatic Update'' on page 683, the file name is included in the configured URL of the provisioning server where the file is located.

Including the MAC address in the file name is useful if you want the device to download a file that is unique to the device. This feature is typically implemented in mass provisioning of devices where each device downloads a specific configuration file. In such a setup, the provisioning server stores configuration files per device, where each file includes the MAC address of a specific device in its file name.

To support this feature, you need to include the MAC address placeholder string, "<MAC>" anywhere in the configured file name of the URL, for example:

```
IniFileURL = 'https://www.company.com/config_<MAC>.ini'
(automatic-update)# cli-script
https://company.com/files/cli_script_<MAC>.txt
```

The device automatically replaces the string with its hardware MAC address, resulting in a file name request that contains the device's MAC address, for example, config_00908F033512.ini. Therefore, you can configure all the devices with the same URL and file name.

Note: If you write the MAC address placeholder string in lower case (i.e., "<mac>"), the device adds the MAC address in lower case to the file name (e.g., config_<mac>.ini results in config_00908f053736e); if in upper case (i.e., "<MAC>"), the device adds the MAC address in upper case to the file name (e.g., config_<MAC>.ini results in config_00908F053736E).

42.2.9 File Template for Automatic Provisioning

To facilitate automatic provisioning setup, you can use a single template to define the files to download during automatic provisioning. The template uses special keywords to denote the different file types to download and in the URL address of the provisioning server it uses a placeholder for the file names which is replaced by hardcoded file names and extensions according to file type, as described in more detail below.

Note:

- Unlike the parameters that define specific URLs for Auxiliary files (e.g., CptFileURL), the file template feature always retains the URLs after each automatic update process. Therefore, with the file template the device always attempts to download the files upon each automatic update process.
- If you configure a parameter that defines a URL for a specific file (e.g., CptFileURL), the settings of the file template (TemplateUrl parameter) is ignored for the specific file type (e.g., CPT file).
- Additional placeholders can be used in the file name in the URL, for example, <MAC> for MAC address (see MAC Address Placeholder in Configuration File Name on page 689).

➢ **To use a file template for automatic provisioning:**

1. Define the file **types** to download by the file template, using the AupdFilesList parameter. Use the keywords listed in the table below to specify each file type. For example, to specify ini, Feature Key, and CPT files:
   - ini File:
     ```
     AupdFilesList = 'ini', 'fk', 'cpt'
     ```
   - CLI:
     ```
     # configure system
     (config-system)# automatic update
     (automatic-update)# template-files-list ini,fk,cpt
     ```
2. Define the URL address of the provisioning server on which the files (specified in Step 1) are located, using the TemplateUrl parameter. When you configure the URL, you must include the file type placeholder, "<FILE>", which represents the file name. For each file type specified in Step 1, the device sends an HTTP request to the server, where the placeholder in the URL is replaced with the filename and extension, as listed in the below table. For example, if you configure the AupdFilesList parameter as in Step 1 and the TemplateUrl parameter to:
   - ini File:
     ```
     TemplateUrl = 'http://10.8.8.20/Site1_<FILE>'
     ```
   - CLI:
     ```
     # configure system
     (config-system)# automatic update
     (automatic-update)# template-url http://10.8.8.20/Site1_<FILE>
     ```

   The device sends HTTP requests to the following URLs:
   - http://10.8.8.20/Site1_**device.ini**
   - http://10.8.8.20/Site1_**fk.ini**
   - http://10.8.8.20/Site1_**cpt.data**

3. Place the files to download on the provisioning server. Make sure that their file names and extensions are based on the hardcoded string values specific to the file type for the <FILE> placeholder (e.g., "Site1_device.ini" for the ini file), as shown in the table below.

File Template Keywords and Placeholder Values per File Type

File Type	Keywords for Template File	Value Replacing <FILE> Placeholder
ini file	ini	device.ini
CLI Script file	cli	cliScript.txt
CMP file based on timestamp	acmp	autoFirmware.cmp
User Info file	usrinf	userInfo.txt
CMP file	cmp	firmware.cmp
Feature Key file	fk	fk.ini
Call Progress Tone (CPT) file	cpt	cpt.dat
Prerecorded Tones (PRT) file	prt	prt.dat
Dial Plan file	dpln	dialPlan.dat
Answering Machine Detection (AMD) file	amd	amd.dat
SSL/TLS Private Key file	sslp	pkey.pem pkey<ID>.pem (for multi-certificate system)
SSL/TLS Root Certificate file	sslr	root.pem root<ID>.pem (for multi-certificate system)
SSL/TLS Certificate file	sslc	cert.pem cert<ID>.pem (for multi-certificate system)

42.2.10 Automatic Update Configuration Examples

This section provides a few examples on configuring the Automatic Update feature.

42.2.10.1 Automatic Update for Single Device

This simple example describes how to configure the Automatic Update feature for updating a single device. In this example, the device queries the provisioning server for software, configuration and Auxiliary files every 24 hours.

➢ **To set up Automatic Provisioning for single device (example):**

1. Set up an HTTP Web server (e.g., http://www.company.com) and place all the required configuration files on this server.
2. Configure the device with the IP address of the DNS server for resolving the domain name (e.g., http://www.company.com) that is used in the URL of the provisioning server. You configure this in the Interface table:
   - ini File:

```
[ InterfaceTable ]
FORMAT InterfaceTable_Index =
InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_VlanID, InterfaceTable_InterfaceName,
InterfaceTable_PrimaryDNSServerIPAddress,
InterfaceTable_SecondaryDNSServerIPAddress,
InterfaceTable_UnderlyingDevice;
InterfaceTable 0 = 6, 10, 10.15.7.95, 16, 10.15.0.1, 1,
"Voice", 80.179.52.100, 0.0.0.0, "vlan 1";
[ \InterfaceTable ]
```

- CLI:

```
# configure voip
(config-voip)# interface network-if 0
(network-if-0)# primary-dns 80.179.52.100
```

3. Configure the device with the following Automatic Update settings:

   a. Automatic Update is done every 24 hours (1440 minutes):

   - ini File:

   ```
   AutoUpdateFrequency = 1440
   ```

   - CLI:

   ```
   # configure system
   (config-system)# automatic update
   (automatic-update)# update-frequency 1440
   ```

   b. Automatic Update of software file (.cmp):

   - ini File:

   ```
   AutoCmpFileUrl =  'https://www.company.com/sw.cmp'
   ```

   - CLI:

   ```
   # configure system
   (config-system)# automatic update
   (automatic-update)# auto-firmware 'http://www.company.com/sw.cmp'
   ```

   c. Automatic Update of Call Progress Tone file:

   - ini File:

   ```
   CptFileURL =
   'https://www.company.com/call_progress.dat'
   ```

   - CLI:

   ```
   # configure system
   (config-system)# automatic update
   (automatic-update)# call-progress-tones
   'http://www.company.com/call_progress.dat'
   ```

   d. Automatic Update of ini configuration file:

   - ini File:

   ```
   IniFileURL = 'https://www.company.com/config.ini'
   ```

   - CLI:

   ```
   # configure system
   (config-system)# automatic update
   (automatic-update)# voice-configuration
   'http://www.company.com/config.ini'
   ```

   e. Enable Cyclical Redundancy Check (CRC) on downloaded ini file:

   - ini File:

   ```
   AUPDCheckIfIniChanged = 1
   ```

   - CLI:

   ```
   # configure system
   (config-system)# automatic update
   (automatic-update)# crc-check regular
   ```

4. Power down and then power up the device.

42.2.10.2 Automatic Update from Remote Servers

This example describes how to configure the Automatic Update feature where files are stored and downloaded from different file server types. The example scenario includes the following:

- FTPS server at ftpserver.corp.com for storing the Voice Prompts (VP) file. The login credentials to the server are username "root" and password "wheel".
- HTTP server at www.company.com for storing the configuration file (ini).
- DNS server at 80.179.52.100 for resolving the domain names of the provisioning servers (FTPS and HTTP).

➢ **To set up Automatic Provisioning for files stored on different server types (example):**

1. **VP file:**
   a. Set up an FTPS server and copy the VP file to the server.
   b. Configure the device with the URL path of the VP file:
      - ini File:
        ```
        VPFileUrl =
        'ftps://root:wheel@ftpserver.corp.com/vp.dat'
        ```
      - CLI:
        ```
        # configure system
        (config-system)# automatic update
        (automatic-update)# voice-prompts
        'ftps://root:wheel@ftpserver.corp.com/vp.dat'
        ```
2. **Software (.cmp) and ini files:**
   a. Set up an HTTP Web server and copy the .cmp and configuration files to the server.
   b. Configure the device with the URL paths of the .cmp and ini files:
      - ini File:
        ```
        AutoCmpFileUrl =
        'http://www.company.com/device/sw.cmp'
        IniFileURL = 'http://www.company.com/device/inifile.ini'
        ```
      - CLI:
        ```
        # configure system
        (config-system)# automatic update
        (automatic-update)# auto-firmware 'http://www.company.com/sw.cmp'
        ```
3. Configure the device with the IP address of the DNS server for resolving the domain names of the FTPS and HTTP servers:
   ```
   [ InterfaceTable ]
   FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
   InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
   InterfaceTable_PrefixLength, InterfaceTable_Gateway,
   InterfaceTable_VlanID, InterfaceTable_InterfaceName,
   InterfaceTable_PrimaryDNSServerIPAddress,
   InterfaceTable_SecondaryDNSServerIPAddress,
   InterfaceTable_UnderlyingDevice;
   InterfaceTable 0 = 6, 10, 10.15.7.95, 16, 10.15.0.1, 1,
   "Voice", 80.179.52.100, 0.0.0.0, "vlan 1";
   [ \InterfaceTable ]
   ```
4. Configure the device to perform the Automatic Update process daily at 03:00 (3 a.m):

- ini File:

  ```
  AutoUpdatePredefinedTime = '03:00'
  ```

- CLI:

  ```
  # configure system
  (config-system)# automatic update
  (automatic-update)# predefined-time 03:00
  ```

42.2.10.3 Automatic Update for Mass Deployment

This example describes how to configure the Automatic Update feature for updating multiple devices (i.e., mass deployment) using an HTTP provisioning server. In this example, all the devices are configured to download the same "master" configuration file. This file serves as the configuration template and instructs the devices which files to download and how often to perform the Automatic Update process. In addition, the master file also instructs each device to download an ini configuration file whose file name contains the MAC address of the device.

The example scenario is as follows:

- All devices download a "master" configuration file that contains the following:
  - Common configuration shared by all device's.
  - Specific configuration that instructs each device to download a specific configuration file based on the device's MAC address, using the special string "<MAC>" in the URL, as described in "MAC Address Placeholder in Configuration File Name" on page 689.
- Device queries the provisioning server daily at 24:00 (midnight) for software, configuration and Auxiliary files.
- HTTP-based provisioning server at www.company.com for storing the files.
- DNS server at 80.179.52.100 for resolving the domain name of the provisioning server.

➢ **To set up automatic provisioning for mass provisioning (example):**

1. Create a "master" configuration file template named "master_configuration.ini" with the following settings:
   - Common configuration for all devices:
     - ini file:

       ```
       AutoUpdatePredefinedTime = '24:00'
       CptFileURL = 'https://www.company.com/call_progress.dat'
       AutoCmpFileUrl = 'https://www.company.com/sw.cmp'
       ```

     - CLI:

       ```
       # configure system
       (config-system)# automatic update
       (automatic-update)# update-frequency 24:00
       (automatic-update)# call-progress-tones
       https://www.company.com/call_progress.dat
       (automatic-update)# auto-firmware https://www.company.com/sw.cmp
       ```

   - Configuration per device based on MAC address:
     - ini file:

       ```
       IniFileURL = 'http://www.company.com/config_<MAC>.ini'
       ```

     - CLI:

       ```
       # configure system
       (config-system)# automatic update
       (automatic-update)# cli-script
       https://company.com/files/cli_script_<MAC>.txt
       ```

```
(automatic-update)# voice-configuration
http://www.company.com/config_<MAC>.ini
```

2. Copy the master configuration file that you created in Step 1 as well as the CPT and .cmp files to the HTTP-based provisioning server.
3. Configure **each** device with the following:
   a. URL of the master configuration file:
      - ini File:

        ```
        IniFileURL =
        'http://www.company.com/master_configuration.ini'
        ```

      - CLI:

        ```
        # configure system
        (config-system)# automatic update
        (automatic-update)# voice-configuration
        http://www.company.com/master_configuration.ini
        (automatic-update)# cli-script
        https://company.com/files/master_startup.txt
        ```

   b. Configure the device with the IP address of the DNS server for resolving the domain name (e.g., http://www.company.com) that is used in the URL for the provisioning server. This is done in the Interface table:
      - ini File:

        ```
        [ InterfaceTable ]
        FORMAT InterfaceTable_Index =
        InterfaceTable_ApplicationTypes,
        InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
        InterfaceTable_PrefixLength, InterfaceTable_Gateway,
        InterfaceTable_VlanID, InterfaceTable_InterfaceName,
        InterfaceTable_PrimaryDNSServerIPAddress,
        InterfaceTable_SecondaryDNSServerIPAddress,
        InterfaceTable_UnderlyingDevice;
        InterfaceTable 0 = 6, 10, 10.15.7.95, 16, 10.15.0.1, 1,
        "Voice", 80.179.52.100, 0.0.0.0, "vlan 1";
        [ \InterfaceTable ]
        ```

      - CLI:

        ```
        # configure voip
        (config-voip)# interface network-if 0
        (network-if-0)# primary-dns 80.179.52.100
        ```

4. Power down and then power up the device.

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43 Restoring Factory Defaults

This section describes the different ways that you can restore the device's configuration to factory defaults.

43.1 Restoring Factory Defaults through CLI

You can restore the device to factory defaults through CLI, as described in the following procedure.

➢ **To restore factory defaults through CLI:**

1. Access the CLI:
   a. Connect the RS-232 serial port of the device to the communication port on your computer. For serial cabling, refer to the Hardware Installation Manual.
   b. Establish serial communication with the device using a serial communication program (such as HyperTerminalTM) with the following communication port settings:
      - Baud Rate: 115,200 bps
      - Data Bits: 8
      - Parity: None
      - Stop Bits: 1
      - Flow Control: None
2. At the CLI prompt, type the username (default is "Admin" - case sensitive), and then press Enter:
   ```
   # Username: Admin
   ```
3. At the prompt, type the password (default is "Admin" - case sensitive), and then press Enter:
   ```
   # Password: Admin
   ```
4. At the prompt, type the following, and then press Enter:
   ```
   # enable
   ```
5. At the prompt, type the password again, and then press Enter:
   ```
   # Password: Admin
   ```
6. At the prompt, type the following to reset the device to default settings, and then press Enter:
   ```
   # write factory
   ```

43.2 Restoring Factory Defaults through Web Interface

You can restore the device to factory defaults through the Web interface.

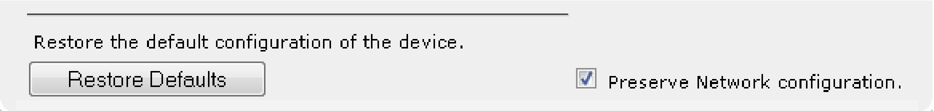
> **Note:** When restoring to factory defaults, you can preserve your IP network settings that are configured in the Interface table (see "Configuring IP Network Interfaces" on page 133), as described in the procedure below. This may be important, for example, to maintain connectivity with the device (through the OAMP interface) after factory defaults have been applied.

➢ **To restore factory defaults through Web interface:**

1. Open the Configuration File page:

- Toolbar: From the **Device Actions** drop-down list, choose **Restore Defaults**
- Navigation Tree: **Maintenance** tab > **Software Update** > **Configuration File**

Figure 43-1: Restoring Factory Defaults through Web



Restore the default configuration of the device.

2. To keep your current IP network settings, select the **Preserve Network Configuration** check box. To overwrite all your IP network settings with the default IP network interface, clear the **Preserve Network Configuration** check box.
3. Click the **Restore Defaults** button; a message appears requesting you to confirm.
4. Click **OK** to confirm or **Cancel** to return to the page.
5. Once the device is restored to factory defaults, reset the device for the settings to take effect.

43.3 Restoring Defaults through ini File

You can restore the device to factory defaults by loading an empty *ini* file to the device. This is done using the Web interface's Configuration File page (see "Backing Up and Loading Configuration File" on page 677). If the *ini* file does include content (e.g., parameters), ensure that they are on lines beginning with comment signs (i.e., semicolons ";") so that the device ignores them.

> **Note:** The only settings that are not restored to default are the management (OAMP) LAN IP address and the Web interface's login user name and password.

44 Automatic Archiving of Configuration File

You can configure the device to automatically save a configuration file each time you modify the device's configuration. The archived file can be saved to a user-defined URL of a remote server (TFTP or HTTP/S), or to a USB storage device attached to the device. The device first saves the configuration to its flash memory and then sends the file to the remote server or USB. The configuration in the archived file is based only on CLI commands. Archiving configuration can be useful, for example, when you need to revert to a previously backed-up configuration (for whatever reason).

To configure configuration-file archiving, use the following CLI command (root level):

- **Archiving to a remote server:**

  ```
  # write-and-backup to <URL path with file name>
  ```

- Archiving to a USB:

  ```
  # write-and-backup to usb:///<file name>
  ```

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45 USB Storage Capabilities

The device supports USB storage using an external USB hard drive or flash disk (disk on key) connected to its USB port. The storage capabilities are configured using the CLI and include the following:

- To save network captures to the USB:

  ```
  # debug capture data physical stop usb
  ```

- To update the device's firmware from the USB:

  ```
  # copy firmware from usb:///<cmp file name>
  ```

- To update the device's configuration from the USB:

  ```
  # copy voice-configuration from usb:///<ini configuration file
  name>
  ```

- To save the current configuration to the USB:

  ```
  # copy voice-configuration to usb:///<ini configuration file
  name>
  ```

Note: Only a single USB storage (formatted to FAT/FAT32) operation is supported at any given time.

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Part IX

Status, Performance Monitoring and Reporting

46 System Status

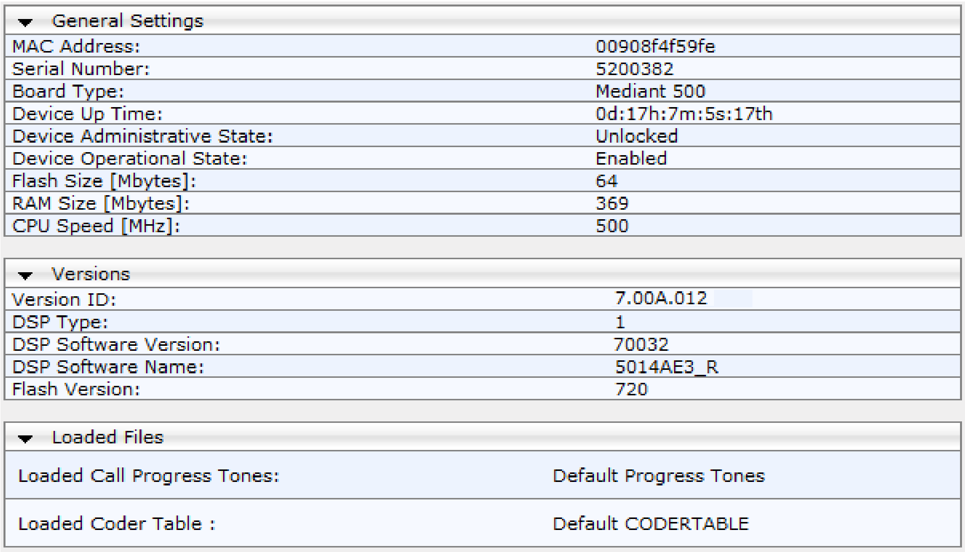
This section describes how to view various system statuses.

46.1 Viewing Device Information

The Device Information page displays hardware and software information about the device. The page also lists Auxiliary files that have been installed on the device and allows you to remove them (see "Deleting Auxiliary Files" on page 649).

➢ **To access the Device Information page:**

- Open the Device Information page (**Status & Diagnostics** tab > **System Status** menu > **Device Information**).



▼ General Settings	
MAC Address:	00908f4f59fe
Serial Number:	5200382
Board Type:	Mediant 500
Device Up Time:	0d:17h:7m:5s:17th
Device Administrative State:	Unlocked
Device Operational State:	Enabled
Flash Size [Mbytes]:	64
RAM Size [Mbytes]:	369
CPU Speed [MHz]:	500

▼ Versions	
Version ID:	7.00A.012
DSP Type:	1
DSP Software Version:	70032
DSP Software Name:	5014AE3_R
Flash Version:	720

▼ Loaded Files	
Loaded Call Progress Tones:	Default Progress Tones
Loaded Coder Table :	Default CODERTABLE

Device Information Description

Parameter	Description
General Settings	
MAC Address	Media access control (MAC) address.
Serial Number	Serial number of the CPU. This serial number also appears on the product label that is affixed to the chassis, as "CPU S/N".
Product Key	Product Key, which identifies the specific device purchase. The Product Key also appears on the product label that is affixed to the chassis, as "S/N(Product Key)". For more information, see Viewing the Device's Product Key on page 670.
Board Type	Product name of the device.
Device Up Time	Duration that the device has been up and running since the last reset. The duration is displayed in the following format: *dd:hh:mm:ss:100th of a second*
Device Administrative State	Administrative status ("Unlocked" or "Locked"), as performed in Locking and Unlocking the Device on page 642.

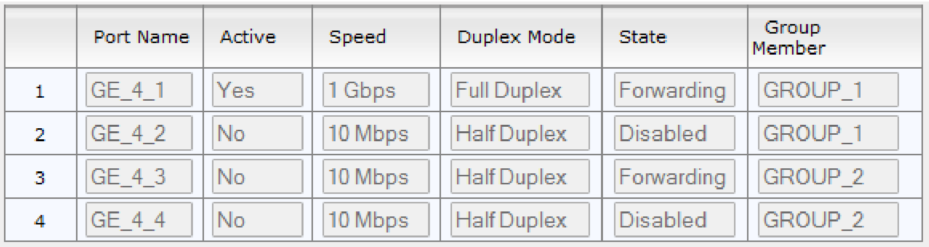
Parameter	Description
Device Operational State	Operational status: ▪ "Disabled" ▪ "Enabled" ▪ "Error" ▪ "Unknown"
Flash Size [Mbytes]	Size of the non-volatile storage memory (flash), measured in megabytes.
RAM Size [Mbytes]	Size of the random access memory (RAM), measured in megabytes.
CPU Speed [MHz]	Clock speed of the CPU, measured in megahertz (MHz).
Versions	
Version ID	Software version number.
DSP Type	Type of DSP.
DSP Software Version	DSP software version.
DSP Software Name	DSP software name.
Flash Version	Flash memory version number.
Loaded Files: Displays installed Auxiliary files. You can also delete a file, by clicking the corresponding **Delete** button, as described in Deleting Auxiliary Files on page 649.	

46.2 Viewing Ethernet Port Information

The Ethernet Port Information page displays read-only information about the Ethernet Port Group connections.

➢ To view Ethernet port information:

- Open the Ethernet Port Information page:
  - Navigation menu tree: **Status & Diagnostics** tab > **System Status** menu > **Ethernet Port Info**
  - On the Home page, click any Ethernet port on the graphical display of the device (see "Viewing the Home Page" on page 68)



	Port Name	Active	Speed	Duplex Mode	State	Group Member
1	GE_4_1	Yes	1 Gbps	Full Duplex	Forwarding	GROUP_1
2	GE_4_2	No	10 Mbps	Half Duplex	Disabled	GROUP_1
3	GE_4_3	No	10 Mbps	Half Duplex	Forwarding	GROUP_2
4	GE_4_4	No	10 Mbps	Half Duplex	Disabled	GROUP_2

Table 46-1: Ethernet Port Information Parameters

Parameter	Description
Port Name	Displays the name of the port.
Active	Displays whether the port is active ("Yes") or not ("No").

Parameter	Description
Speed	Displays the speed (in Mbps) of the Ethernet port.
Duplex Mode	Displays whether the port is half- or full-duplex.
State	Displays the state of the port: ▪ "Forwarding": Active port (data is being received and sent) ▪ "Disabled": Redundancy port
Group Member	Displays the port-pair group ID to which the port belongs.

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47 Carrier-Grade Alarms

This section describes how to view SNMP alarms raised by the device.

47.1 Viewing Active Alarms

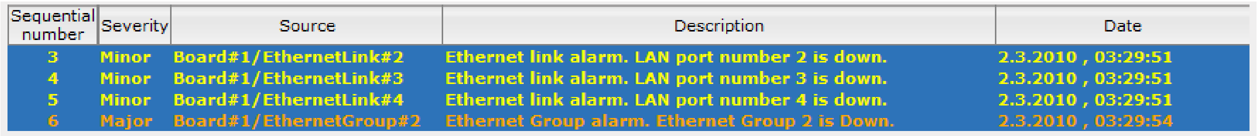
The Active Alarms table displays a list of currently active alarms that have been raised by the device. Once an alarm has been resolved (cleared), the device moves it into the History Alarms table (see "Viewing History Alarms" on page 709). For detailed information on SNMP alarms, refer to the *SNMP Reference Guide* document.

Note:

- The alarms in the table are deleted upon a device reset.
- To configure the maximum number of active alarms that can be displayed in the table, see the ini file parameter, ActiveAlarmTableMaxSize.

➢ **To view active alarms:**

- Open the Active Alarms table (**Status & Diagnostics** tab > **System Status** menu > **Carrier-Grade Alarms** > **Active Alarms**). You can also access the table from the Home page (see "Viewing the Home Page" on page 68).



Sequential number	Severity	Source	Description	Date
3	Minor	Board#1/EthernetLink#2	Ethernet link alarm. LAN port number 2 is down.	2.3.2010 , 03:29:51
4	Minor	Board#1/EthernetLink#3	Ethernet link alarm. LAN port number 3 is down.	2.3.2010 , 03:29:51
5	Minor	Board#1/EthernetLink#4	Ethernet link alarm. LAN port number 4 is down.	2.3.2010 , 03:29:51
6	Major	Board#1/EthernetGroup#2	Ethernet Group alarm. Ethernet Group 2 is Down.	2.3.2010 , 03:29:54

For each alarm, the following information is provided:

- **Sequential Number:** number of the alarm (sequential numbering of each alarm)
- **Severity:** severity level of the alarm:
  - Critical (red)
  - Major (orange)
  - Minor (yellow)
- **Source:** device component from which the alarm was raised
- **Description:** brief explanation of the reason of the alarm
- **Date:** date and time that the alarm was generated

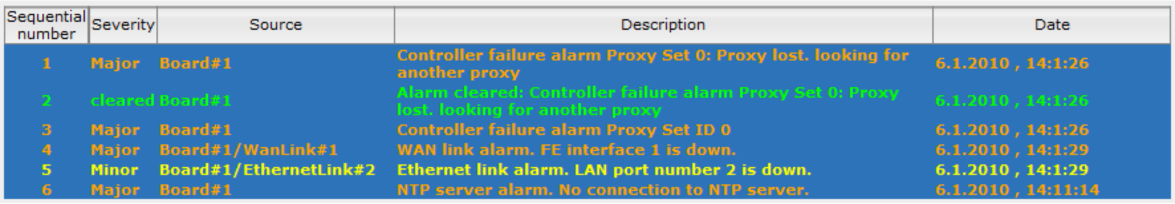
You can view the next 20 alarms (if exist), by clicking the **Go to page** button.

47.2 Viewing History Alarms

The Alarms History table displays a list of alarms that have been cleared (resolved). You can configure the maximum number of alarms displayed in the table, using the AlarmHistoryTableMaxSize ini file parameter. If the maximum is reached and a new alarm is added to the table, the oldest alarm is removed from the table to accommodate the new alarm.

➢ **To view history alarms:**

- Open the Alarms History table (**Status & Diagnostics** tab > **System Status** menu > **Carrier-Grade Alarms** > **Alarms History**).



Sequential number	Severity	Source	Description	Date
1	Major	Board#1	Controller failure alarm Proxy Set 0: Proxy lost. looking for another proxy	6.1.2010 , 14:1:26
2	cleared	Board#1	Alarm cleared: Controller failure alarm Proxy Set 0: Proxy lost. looking for another proxy	6.1.2010 , 14:1:26
3	Major	Board#1	Controller failure alarm Proxy Set ID 0	6.1.2010 , 14:1:26
4	Major	Board#1/WanLink#1	WAN link alarm. FE interface 1 is down.	6.1.2010 , 14:1:29
5	Minor	Board#1/EthernetLink#2	Ethernet link alarm. LAN port number 2 is down.	6.1.2010 , 14:1:29
6	Major	Board#1	NTP server alarm. No connection to NTP server.	6.1.2010 , 14:11:14

For each alarm, the following information is provided:

- **Severity:** severity level of the alarm:
  - Critical (red)
  - Major (range)
  - Minor (yellow)
  - Cleared (green)
- **Source:** unit from which the alarm was raised
- **Description:** brief explanation of the alarm
- **Date:** date and time that the alarm was generated

To view the next 20 alarms (if exist), click the **Go to page** button.

➢ **To delete all the alarms in the table:**

1. Click the **Delete History Table** button; a confirmation message box appears.
2. Click **OK** to confirm.

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48 Performance Monitoring

This section describes how to view performance monitoring in the device's Web interface.

48.1 Viewing MOS per Media Realm

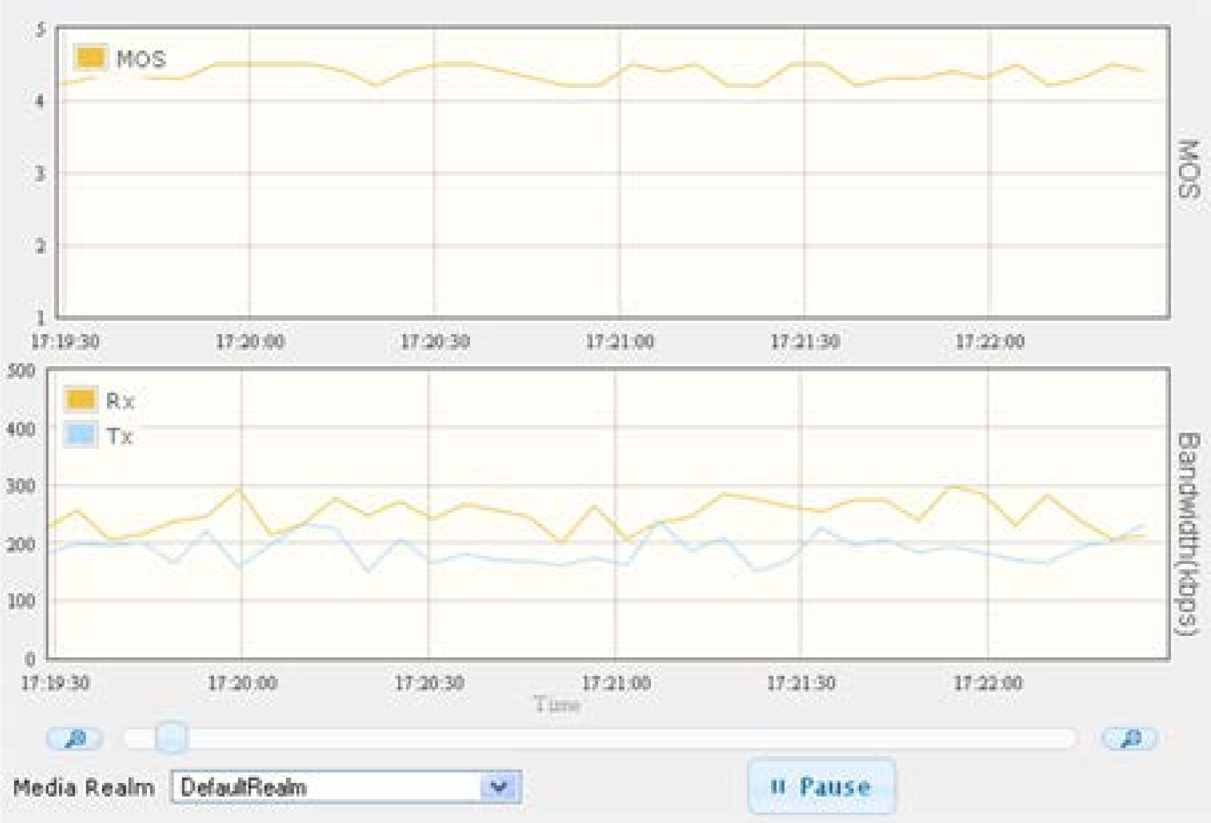
The MOS Per Media Realm page displays statistics on Media Realms (configured in "Configuring Media Realms" on page 317). This page provides two graphs:

- Upper graph: displays the Mean Opinion Score (MOS) quality in RTCP data per selected Media Realm.
- Lower graph: displays the bandwidth of transmitted media (in Kbps) in RTCP data per Media Realm.

➢ To view the MOS per Media Realm graph:

1. Open the MOS Per Media Realm page (**Status & Diagnostics** tab > **Performance Monitoring** menu > **MOS Per Media Realm**).

Figure 48-1: MOS Per Media Realm Graph



Media Realm: DefaultRealm

2. From the 'Media Realm' drop-down list, select the Media Realm for which you want to view.

Use the **Zoom In** 🔍 button to increase the displayed time resolution or the **Zoom Out** 🔍 button to decrease it. Instead of using these zoom buttons, you can use the slide ruler. As you increase the resolution, more data is displayed on the graph. The minimum resolution is about 30 seconds; the maximum resolution is about an hour.

To pause the graph, click the **Pause** button; click **Play** to resume.

48.2 Viewing Trunk Utilization

The Trunk Utilization page provides an X-Y graph that displays the number of active channels per trunk over time. The x-axis indicates the time; the y-axis indicates the number of active trunk channels.

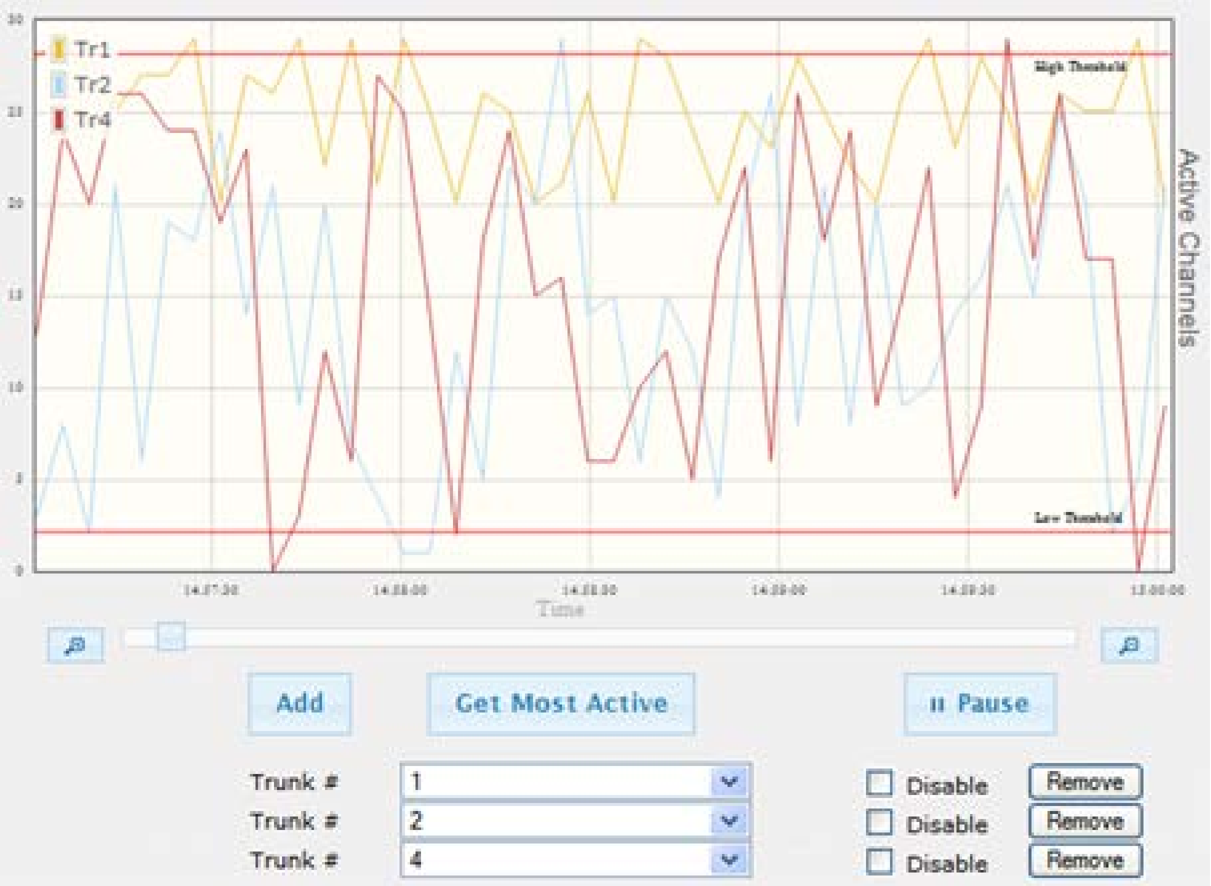
Notes:

- The Trunk Utilization page is available only if your device is equipped with have trunks and the SBC application is disabled.
- If you navigate to a different page, the data displayed in the graph and all its settings are cleared.

➢ To view the number of active trunk channels

1. Open the Trunk Utilization page (**Status & Diagnostics** tab > **Performance Monitoring** menu > **Trunk Utilization**).

Figure 48-2: Trunk Utilization Page



Trunk # 1
Trunk # 2
Trunk # 4

2. From the 'Trunk' drop-down list, select the trunk for which you want to view active channels.

For more graph functionality, see the following table:

Table 48-1: Additional Graph Functionality for Trunk Utilization

Button	Description
Add button	Displays additional trunks in the graph. Up to five trunks can be displayed simultaneously in the graph. To view another trunk, click this button and then from the new 'Trunk' drop-down list, select the required trunk. Each trunk is displayed in a different color, according to the legend shown in the top-left corner of the graph.
Remove button	Removes the selected trunk display from the graph.
Disable check box	Hides or shows an already selected trunk. Select this check box to temporarily hide the trunk display; clear this check box to show the trunk. This is useful if you do not want to remove the trunk entirely (using the **Remove** button).
Get Most Active button	Displays only the trunk with the most active channels (i.e., trunk with the most calls).
Pause button	Pauses the display in the graph.
Play button	Resumes the display in the graph.
Zoom slide ruler and buttons	Increases or reduces the trunk utilization display resolution concerning time. The **Zoom In** 🔍 button increases the time resolution; the **Zoom Out** 🔍 button decreases it. Instead of using the buttons, you can use the slide ruler. As you increase the resolution, more data is displayed on the graph. The minimum resolution is about 30 seconds; the maximum resolution is about an hour.

48.3 Viewing Quality of Experience

The Quality Of Experience page provides statistical information on calls per SRD or IP Group. The statistics can be further filtered to display incoming and/or outgoing call direction, and type of SIP dialog (INVITE, SUBSCRIBE, or all).

Note: The Quality Of Experience page is applicable only to SBC calls.

This page provides three pie charts:

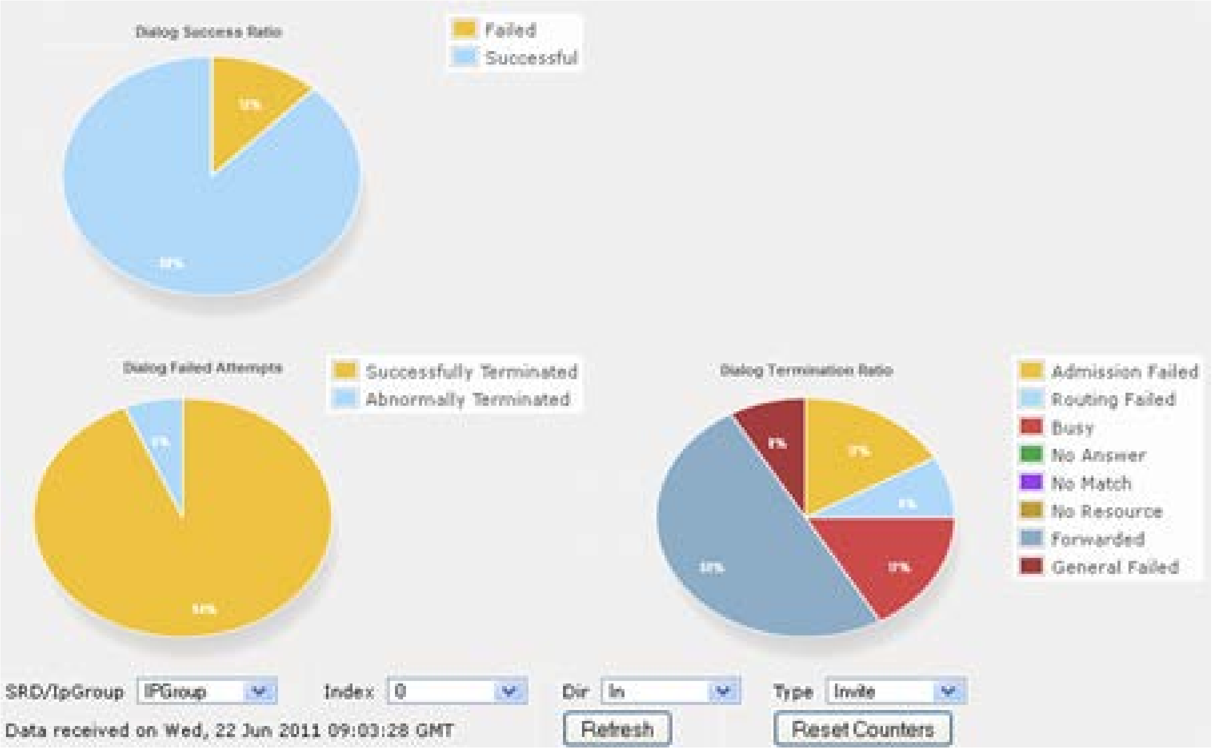
- Dialog Success Ratio: displays the SIP call and subscribe (SUBSCRIBE) dialog success-failed ratio.
- Dialog Failed Attempts: displays the failed call attempts. This includes the number of calls and subscribes which were successfully and abnormally terminated.
- Dialog Termination Ratio: displays call termination by reason (e.g., due to no answer).

➢ **To view Quality of Experience:**

1. Open the Quality Of Experience page (**Status & Diagnostics** tab > **Performance**

Monitoring menu > **Quality Of Experience**).

Figure 48-3: Quality Of Experience Graph



SRD/IpGroup: IPGroup | Index: 0 | Dir: In | Type: Invite

Data received on Wed, 22 Jun 2011 09:03:28 GMT

2. From the 'SRD/IpGroup' drop-down list, select whether you want to view QoE for an SRD or IP Group.
3. From the 'Index' drop-down list, select the SRD or IP Group index.
4. From the 'Dir' drop-down list, select the call direction:
   - **In** - incoming calls
   - **Out** - outgoing calls
   - **Both** - incoming and outgoing calls
5. From the 'Type' drop-down list, select the SIP message type:
   - **Invite** - INVITE
   - **Subscribe** - SUBSCRIBE
   - **Other** - all SIP messages

To refresh the charts, click **Refresh**. To reset the counters, click **Reset Counters**.

48.4 Viewing Average Call Duration

The Average Call Duration page displays information about a specific SRD or IP Group. This page includes two graphs:

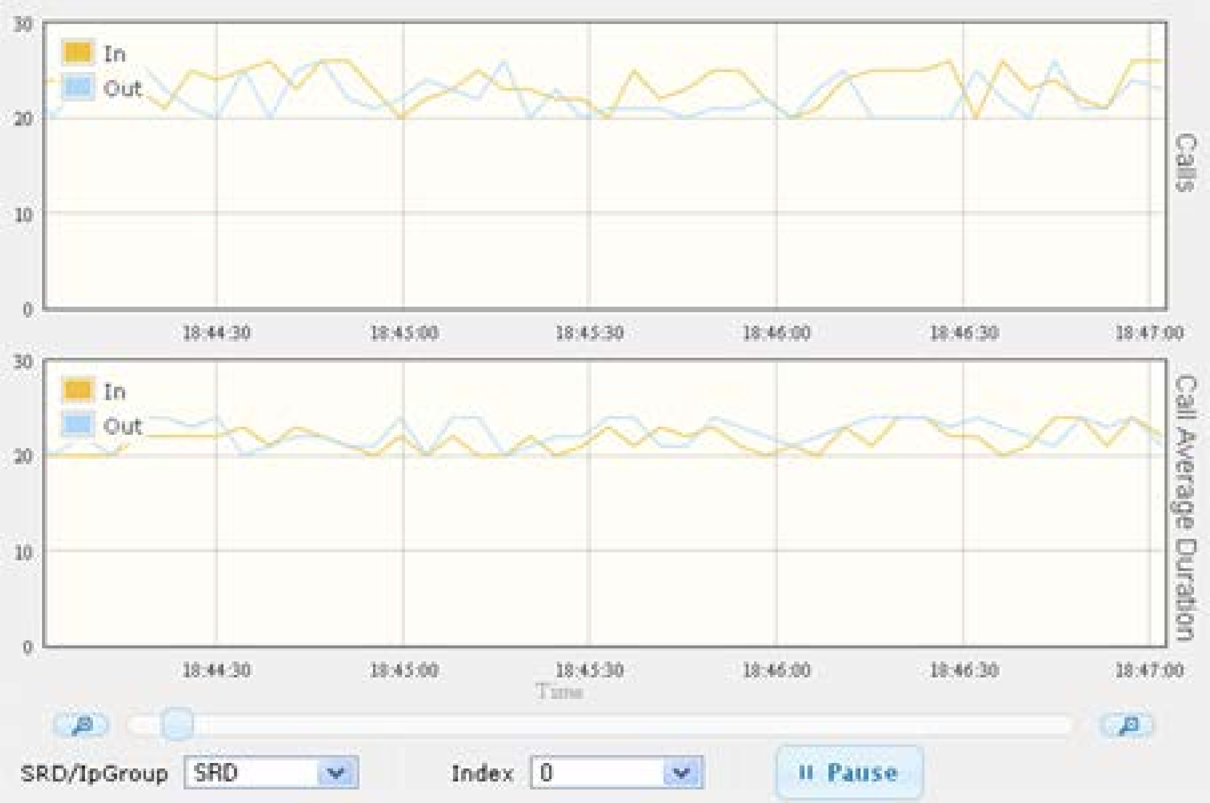
- Upper graph: displays the number of calls (INVITEs).
- Lower graph: displays the average call duration.

Note: The Quality Of Experience page is applicable only to SBC calls.

➢ **To view average call duration:**

1. Open the Average Call Duration page (**Status & Diagnostics** tab > **Performance Monitoring** menu > **Average Call Duration**).

Figure 48-4: Average Call Duration Graph



SRD/IpGroup SRD

Index 0

2. From the 'SRD/IpGroup' drop-down list, select whether you want to view information for an SRD or IP Group.
3. From the 'Index' drop-down list, select the SRD or IP Group index.

Use the **Zoom In** 🔍 button to increase the displayed time resolution or the **Zoom Out** 🔍 button to decrease it. Instead of using these zoom buttons, you can use the slide ruler. As you increase the resolution, more data is displayed on the graph. The minimum resolution is about 30 seconds; the maximum resolution is about an hour.

To pause the graph, click the **Pause** button; click **Play** to resume.

48.5 Configuring PacketSmart for Network Monitoring

You can configure the device to send voice traffic data to BroadSoft's BroadCloud™ PacketSmart™ solution for monitoring and assessing the network in which the device is deployed. The support is offered by the PacketSmart management agent embedded in the device. The PacketSmart embedded agent allows network operators and service providers to remotely measure and manage network performance at the point of demarcation and simplify the deployment of VoIP networks. By providing real-time monitoring of live traffic, PacketSmart can identify any network issues as they arise that may impact VoIP quality, enabling service providers to address issues prior to customer complaints.

Note:

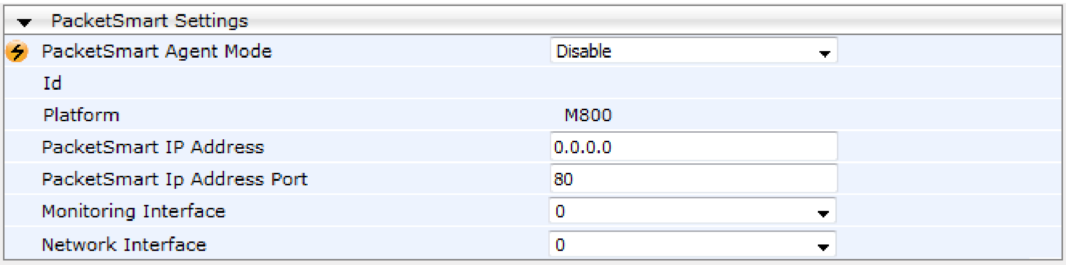
- The PacketSmart feature is a license-dependent feature and is available only if it is included in the Software License Key installed on the device. For ordering the feature, please contact your AudioCodes sales representative.
- Before configuring the PacketSmart agent, configure the following:
  - √ Correct data and time of the device. It is recommended to use an NTP server to obtain the date and time (see Configuring Automatic Date and Time using SNTP on page 121).
  - √ IP network interface for communicating with the PacketSmart server. Typically, the OAMP interface is used. For configuring IP network interfaces, see Configuring IP Network Interfaces on page 133).
  - √ IP network interface for the VoIP traffic that you want monitored by PacketSmart.
- For detailed information on setting up the PacketSmart solution, refer to the document, *Mediant Gateways and SBCs with BroadCloud PacketSmart Configuration Note*.

The following procedure describes how to configure PacketSmart through the Web interface. You can also configure it through ini file or CLI (configure system > packetsmart).

➢ **To configure the PacketSmart agent:**

1. Open the Application Settings page (Configuration > System > Application Settings).

Figure 48-5: Configuring PacketSmart Agent



▼ PacketSmart Settings

PacketSmart Agent Mode	Disable
Id	
Platform	M800
PacketSmart IP Address	0.0.0.0
PacketSmart Ip Address Port	80
Monitoring Interface	0
Network Interface	0

2. From the 'PacketSmart Agent Mode' drop-down list, select **Enable** to enable the feature.
3. Configure the remaining parameters, as required. For parameter descriptions, see 'PacketSmart Parameters' on page 848.
4. Click **Submit**, and then reset the device with a burn-to-flash for your settings to take effect.

This page is intentionally left blank.

49 VoIP Status

This section describes how to view VoIP status and statistics.

49.1 Viewing Trunks & Channels Status

The Trunks & Channels Status page displays the status of the device's trunks and corresponding channels. It also enables you to view trunk configuration and channel information.

➢ **To view the status of the device's trunks and channels:**

1. Open the Home page.
2. On the graphical display of the device, click the required trunk, and then from the shortcut menu, choose Port Settings; the Trunks & Channels Status page appears.

> **Note:** The number of displayed trunks and channels depends on configuration.

The status of the trunks is depicted by color-coded icons, as described in the table below:

Table 49-1: Description of Color-Coded Icons for Trunk Status

Icon	Color	Trunk
		Label
▨	Gray	**Disabled**
🟩	Green	**Active - OK**
🟨	Yellow	**RAI Alarm**
🟥	Red	**LOS / LOF Alarm**
🟦	Blue	**AIS Alarm**
🟧	Light Orange	**D-Channel Alarm**
🟧	Dark Orange	NFAS Alarm
🟪	Purple	**Lower Layer Down (DS3 physical layer is disabled)**

The status of the channels is depicted by color-coded icons, as described in the table below:

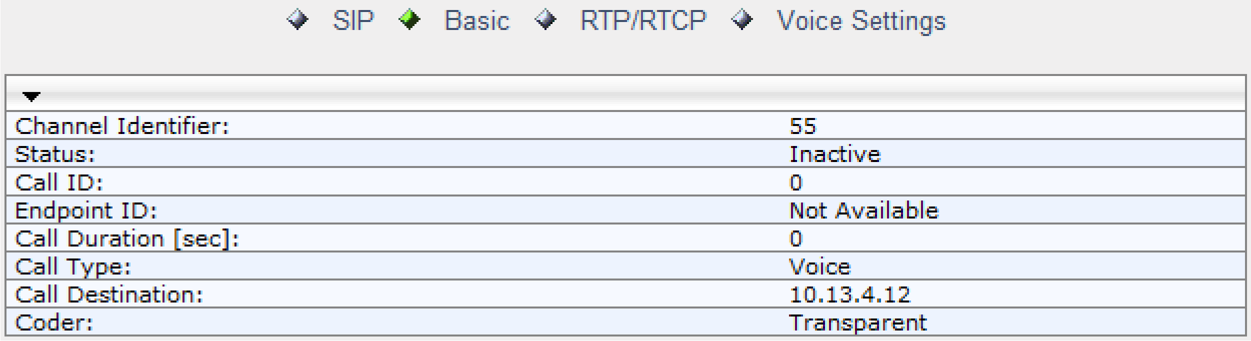
Table 49-2: Description of Color-Coded Icons for Channel Status

Icon	Color	Label	Description
🟦	Light blue	**Inactive**	Channel is configured, but currently has no calls
🟩	Green	**Active**	Call in progress (RTP traffic) and no alarms
🟪	Purple	**SS7**	Channel is configured for SS7 **Note:** Currently, SS7 is not supported.

Icon	Color	Label	Description
▼	Gray	**Non Voice**	Channel is not configured
▼	Blue	**ISDN Signaling**	Channel is configured as a D-channel
▼	Dark Orange	**Maintenance**	B-channel has been intentionally taken out of service due to maintenance
▼	Red	**Out Of Service**	B-channel is out of service

3. To view detailed information on a specific trunk's channel, click the required channel icon; the Basic Channel Information page appears, displaying information under the **Basic** tab (displayed in green):

Figure 49-1: Basic Channel Information Page



Channel Identifier:	55
Status:	Inactive
Call ID:	0
Endpoint ID:	Not Available
Call Duration [sec]:	0
Call Type:	Voice
Call Destination:	10.13.4.12
Coder:	Transparent

To view additional channel information, click the required tab (**SIP**, **RTP/RTCP**, and **Voice Settings**).

4. To view the settings of a specific trunk, click the required trunk icon, and then from the shortcut menu, choose **Port Settings**; the Trunk Settings page opens, displaying the trunk's settings. If needed, you can modify the settings (see Configuring Trunk Settings).

49.2 Viewing Active IP Interfaces

The IP Interface Status page displays the device's active IP interfaces that are listed in the Interface table (see "Configuring IP Network Interfaces" on page 133).

➢ **To view active IP network interfaces:**

- Open the IP Interface Status page (**Status & Diagnostics** tab > **VoIP Status** menu > **IP Interface Status**).

49.3 Viewing Ethernet Device Status

The Ethernet Device Status page displays the configured Ethernet Devices that have been successfully applied to the device. For configuring Ethernet Devices, see "Configuring Underlying Ethernet Devices" on page 131.

➢ **To view the configured and applied Ethernet Devices:**

- Open the Ethernet Device Status page (**Status & Diagnostics** tab > **VoIP Status** menu >**Ethernet Device Status Table**).

49.4 Viewing Static Routes Status

The IP Routing Status table displays the status of the static routes. These are routes configured in the Static Route table (see "Configuring Static IP Routing" on page 141) and routes through the Default Gateway.

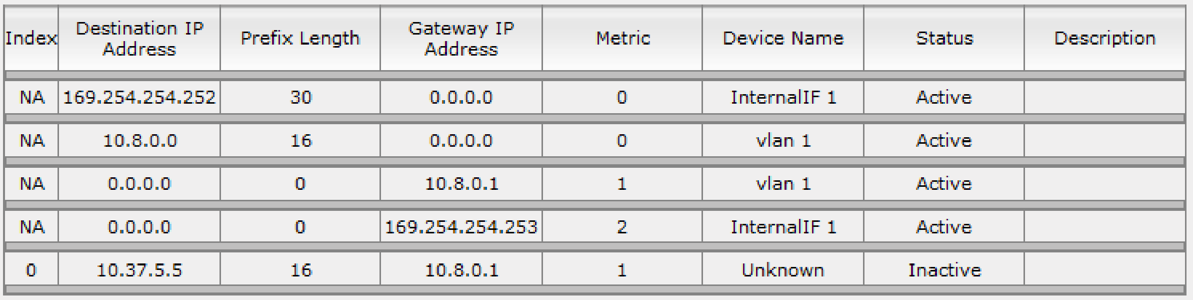
The status of the static routes can be one of the following:

- "Active": Static route is used by the device.
- "Inactive": Static route is not used. When the destination IP address is not on the same segment with the next hop, or the interface does not exist, the route state changes to "Inactive".

➢ To view the status of static IP routing:

- Open the IP Routing Status table (**Status & Diagnostics** tab > **VoIP Status** menu >**Static Route Status**).

Figure 49-2: IP Routing Status Table Page



Index	Destination IP Address	Prefix Length	Gateway IP Address	Metric	Device Name	Status	Description
NA	169.254.254.252	30	0.0.0.0	0	InternalIF 1	Active	
NA	10.8.0.0	16	0.0.0.0	0	vlan 1	Active	
NA	0.0.0.0	0	10.8.0.1	1	vlan 1	Active	
NA	0.0.0.0	0	169.254.254.253	2	InternalIF 1	Active	
0	10.37.5.5	16	10.8.0.1	1	Unknown	Inactive	

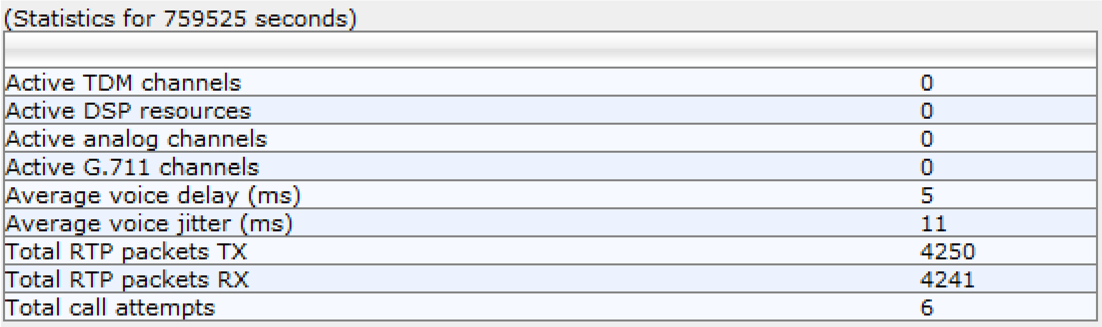
49.5 Viewing Performance Statistics

The Basic Statistics page provides read-only, device performance statistics. This page is refreshed every 60 seconds. The duration that the currently displayed statistics has been collected is displayed above the statistics table.

➢ To view performance statistics:

- Open the Basic Statistics page (**Status & Diagnostics** tab > **VoIP Status** menu > **Performance Statistics**).

Figure 49-3: Basic Statistics Page



(Statistics for 759525 seconds)

Active TDM channels	0
Active DSP resources	0
Active analog channels	0
Active G.711 channels	0
Average voice delay (ms)	5
Average voice jitter (ms)	11
Total RTP packets TX	4250
Total RTP packets RX	4241
Total call attempts	6

The duration that the displayed statistics were collected is displayed in seconds above the table. To reset the performance statistics to zero, click the **Reset Statistics** button.

49.6 Viewing CDR History

The CDR History table displays historical Call Detail Record (CDR) information of Gateway calls. CDR history information is stored on the device’s memory. The CDR History table can contain up to 4,096 CDRs. When a new CDR is generated, the device adds it to the top of the table and all previous entries are shifted one down in the table. If the table has reached maximum capacity of entries and a new CDR is added, the last CDR entry is removed from the table.

> **Note:** If the device is reset, all CDR history information is deleted from memory and subsequently, the CDR History table appears empty.

The following procedure describes how to view CDR history in the Web interface. You can also view CDR history using the following CLI commands:

- All CDR history:

```
# show voip calls history
```

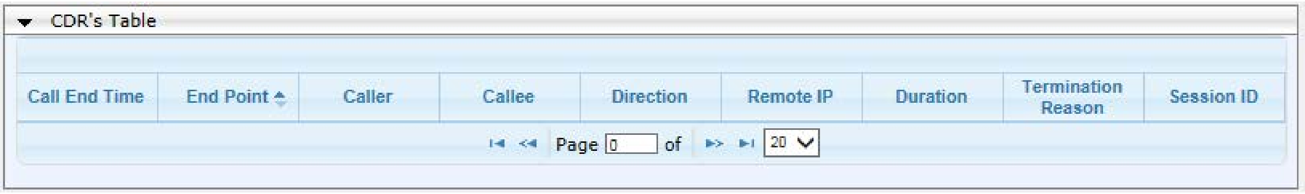
- CDR history for a specific SIP session ID:

```
# show voip calls history <session ID>
```

➢ **To view CDR history:**

- Open the CDR History page (**Status & Diagnostics** tab > **VoIP Status** menu > **CDR History**).

Figure 49-4: CDR History Table



Call End Time	End Point	Caller	Callee	Direction	Remote IP	Duration	Termination Reason	Session ID

Table 49-3: CDR History Table

Field	Description
Call End Time	Displays the time at which the call ended. The time is displayed in the format, hh:mm:ss, where *hh* is the hour, *mm* the minutes and *ss* the seconds (e.g., 15:06:36).
End Point	Displays the device's endpoint involved in the call, displayed in the format: ▪ Digital: <interface>-<module>/<Trunk ID>/<B-channel>. For example, "ISDN-1/2/3" denotes ISDN module 1, Trunk ID 2, B-channel 3.
Caller	Displays the phone number (source number) of the party who made the call.
Callee	Displays the phone number (destination number) of the party to whom the call was made.
Direction	Displays the direction of the call with regards to IP and Tel sides: ▪ "Incoming": IP-to-Tel call ▪ "Outgoing": Tel-to-IP call

Field	Description
Remote IP	Displays the IP address of the call party. For an "Incoming" call, this is the source IP address; for an "Outgoing" call, this is the destination IP address.
Duration	Displays the duration of the call, displayed in the format hh:mm:ss, where *hh* is hours, *mm* minutes and *ss* seconds. For example, 00:01:20 denotes 1 minute and 20 seconds.
Termination Reason	Displays the reason for the call being released (ended). For example, "NORMAL_CALL_CLEAR" indicates a normal off-hook (hang up) of the call party.
Session ID	Displays the SIP session ID of the call.

49.7 Viewing Call Counters

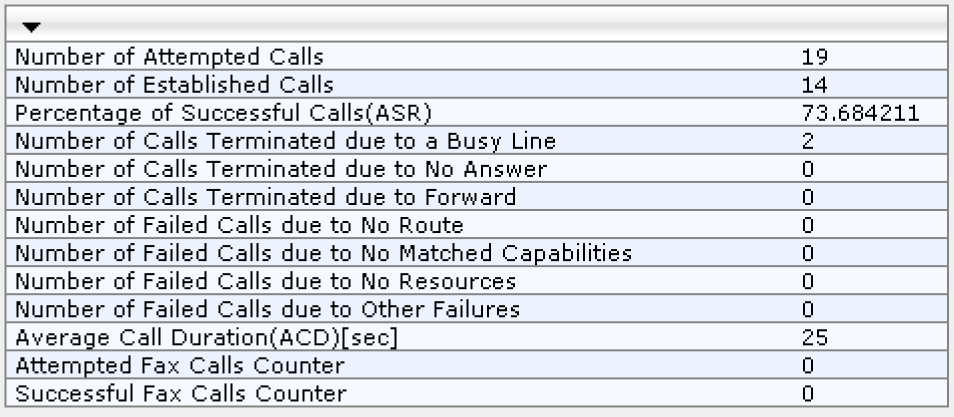
The IP to Tel Calls Count page and Tel to IP Calls Count page provide you with statistical information on incoming (IP-to-Tel) and outgoing (Tel-to-IP) calls. The statistical information is updated according to the release reason that is received after a call is terminated (during the same time as the end-of-call Call Detail Record or CDR message is sent). The release reason can be viewed in the 'Termination Reason' field in the CDR message.

You can reset the statistical data displayed on the page (i.e., refresh the display), by clicking the **Reset Counters** button located below the table.

➢ **To view IP-to-Tel and Tel-to-IP call counters:**

- Open the Call Counters page that you want to view (**Status & Diagnostics** tab > **VoIP Status** menu > **IP to Tel Calls Count** or **Tel to IP Calls Count**); the figure below shows the IP to Tel Calls Count page.

Figure 49-5: Calls Count Page



Number of Attempted Calls	19
Number of Established Calls	14
Percentage of Successful Calls(ASR)	73.684211
Number of Calls Terminated due to a Busy Line	2
Number of Calls Terminated due to No Answer	0
Number of Calls Terminated due to Forward	0
Number of Failed Calls due to No Route	0
Number of Failed Calls due to No Matched Capabilities	0
Number of Failed Calls due to No Resources	0
Number of Failed Calls due to Other Failures	0
Average Call Duration(ACD)[sec]	25
Attempted Fax Calls Counter	0
Successful Fax Calls Counter	0

The fields in this page are described in the following table:

Table 49-4: Call Counters Description

Counter	Description
Number of Attempted Calls	Indicates the number of attempted calls. It is composed of established and failed calls. The number of established calls is represented by the 'Number of Established Calls' counter. The number of failed calls is represented by the failed-call counters. Only one of the established / failed call counters is incremented every time.
Number of Established Calls	Indicates the number of established calls. It is incremented as a result of one of the following release reasons if the duration of the call is greater than zero: ▪ GWAPP_REASON_NOT_RELEVANT (0) ▪ GWAPP_NORMAL_CALL_CLEAR (16) ▪ GWAPP_NORMAL_UNSPECIFIED (31) And the internal reasons: ▪ RELEASE_BECAUSE_UNKNOWN_REASON ▪ RELEASE_BECAUSE_REMOTE_CANCEL_CALL ▪ RELEASE_BECAUSE_MANUAL_DISC ▪ RELEASE_BECAUSE_SILENCE_DISC ▪ RELEASE_BECAUSE_DISCONNECT_CODE **Note:** When the duration of the call is zero, the release reason GWAPP_NORMAL_CALL_CLEAR increments the 'Number of Failed Calls due to No Answer' counter. The rest of the release reasons increment the 'Number of Failed Calls due to Other Failures' counter.
Percentage of Successful Calls (ASR)	The percentage of established calls from attempted calls.
Number of Calls Terminated due to a Busy Line	Indicates the number of calls that failed as a result of a busy line. It is incremented as a result of the following release reason: GWAPP_USER_BUSY (17)
Number of Calls Terminated due to No Answer	Indicates the number of calls that weren't answered. It's incremented as a result of one of the following release reasons: ▪ GWAPP_NO_USER_RESPONDING (18) ▪ GWAPP_NO_ANSWER_FROM_USER_ALERTED (19) ▪ GWAPP_NORMAL_CALL_CLEAR (16) (when the call duration is zero)
Number of Calls Terminated due to Forward	Indicates the number of calls that were terminated due to a call forward. The counter is incremented as a result of the following release reason: RELEASE_BECAUSE_FORWARD
Number of Failed Calls due to No Route	Indicates the number of calls whose destinations weren't found. It is incremented as a result of one of the following release reasons: ▪ GWAPP_UNASSIGNED_NUMBER (1) ▪ GWAPP_NO_ROUTE_TO_DESTINATION (3)
Number of Failed Calls due to No Matched Capabilities	Indicates the number of calls that failed due to mismatched device capabilities. It is incremented as a result of an internal identification of capability mismatch. This mismatch is reflected to CDR via the value of the parameter DefaultReleaseReason (default is GWAPP_NO_ROUTE_TO_DESTINATION (3)) or by the GWAPP_SERVICE_NOT_IMPLEMENTED_UNSPECIFIED (79) reason.

Counter	Description
Number of Failed Calls due to No Resources	Indicates the number of calls that failed due to unavailable resources or a device lock. The counter is incremented as a result of one of the following release reasons: ▪ GWAPP_RESOURCE_UNAVAILABLE_UNSPECIFIED ▪ RELEASE_BECAUSE_GW_LOCKED
Number of Failed Calls due to Other Failures	This counter is incremented as a result of calls that failed due to reasons not covered by the other counters.
Average Call Duration (ACD) [sec]	The average call duration (ACD) in seconds of established calls. The ACD value is refreshed every 15 minutes and therefore, this value reflects the average duration of all established calls made within a 15 minute period.
Attempted Fax Calls Counter	Indicates the number of attempted fax calls.
Successful Fax Calls Counter	Indicates the number of successful fax calls.

49.8 Viewing Registered Users

You can view SAS and SBC users listed in the device's Users Registration database. The list shows each Address of Record (AOR) and its corresponding contact. The contact's registration status is also shown:

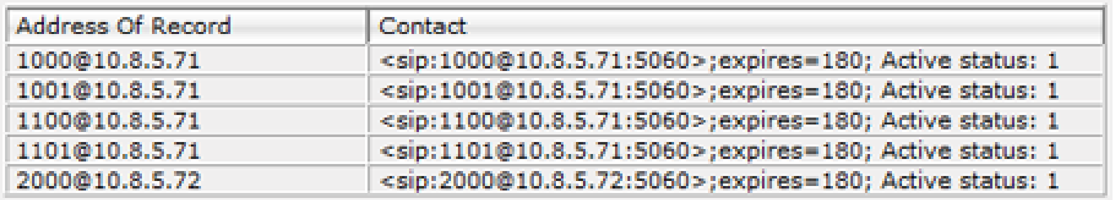
- "Active status:1" indicates that the contact has been successfully registered and thus, calls can be routed to it.
- "Active status:0" indicates that the device has recently received a REGISTER request from the contact, but the contact has yet to be registered. The device removes the contact from the database if no response is received within 10 seconds from the proxy/registrar server.

An AOR is a SIP or SIPS URI that points to a domain with a location service that can map the URI to another URI (contact) where the user might be available. A contact is a SIP URI that can be used to contact that specific instance of the user agent for subsequent requests.

➢ To view registered SAS/SBC users in the Users Registration database:

- Web: SAS/SBC Registered Users page (**Status & Diagnostics** tab > **VoIP Status** menu > **SAS/SBC Registered Users**).

Figure 49-6: SAS/SBC Registered Users Page



Address Of Record	Contact
1000@10.8.5.71	<sip:1000@10.8.5.71:5060>;expires=180; Active status: 1
1001@10.8.5.71	<sip:1001@10.8.5.71:5060>;expires=180; Active status: 1
1100@10.8.5.71	<sip:1100@10.8.5.71:5060>;expires=180; Active status: 1
1101@10.8.5.71	<sip:1101@10.8.5.71:5060>;expires=180; Active status: 1
2000@10.8.5.72	<sip:2000@10.8.5.72:5060>;expires=180; Active status: 1

- CLI:
  - SBC users:
    ```
    # show voip register db sbc list
    ```
  - SBC contacts of a specified AOR:
    ```
    # show voip register db sbc user <Address Of Record>
    ```
  -

SAS users:

```
# show voip register db sas list
```

49.9 Viewing Registration Status

The Registration Status page displays the registration status of the device's endpoints and SIP Accounts, which are configured in the Accounts table (see "Configuring Registration Accounts" on page 363).

➢ To view registration status:

- Open the Registration Status page (**Status & Diagnostics** tab > **VoIP Status** menu > **Registration Status**).
  - Registered Per Gateway (applicable only to the Gateway application): Registration of device as one entity - "YES" or "NO"
  - **Accounts Registration Status:**
    - **Group Type:** served Trunk Group or IP Group
    - **Group Name:** name of served Trunk Group or IP Group, if applicable
    - **Status:** "Registered" or "Unregistered"
  - BRI Phone Number Status:
    - Phone Number: phone number of BRI endpoint
    - Module/Port: module/port number of BRI endpoint
    - Status: "Registered" or "Unregistered"

> Note: The registration mode (i.e., per device, endpoint, account. or no registration) is configured in the Trunk Group Settings table (see Configuring Trunk Group Settings on page 435) or using the TrunkGroupSettings ini file parameter.

49.10 Viewing Proxy Set Status

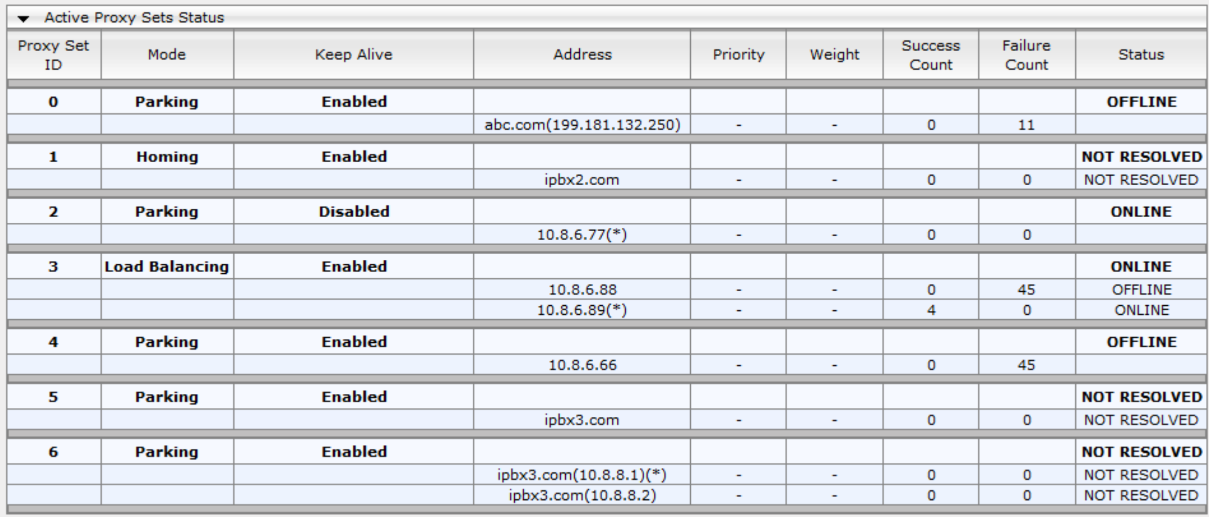
You can view the status of Proxy Sets that are used in your call routing topology. Proxy Sets that are not associated with any routing rule are not displayed.

To configure proxy Sets, see Configuring Proxy Sets on page 352.

➢ To view Proxy Set status:

- Open the Active Proxy Set Status page (**Monitor** menu > **Monitor** tab > **VoIP Status** folder > **Proxy Sets Status**).

Figure 49-7: Viewing Proxy Sets Status



Active Proxy Sets Status

Proxy Set ID	Mode	Keep Alive	Address	Priority	Weight	Success Count	Failure Count	Status
0	**Parking**	**Enabled**						**OFFLINE**
			abc.com(199.181.132.250)	-	-	0	11	
1	**Homing**	**Enabled**						**NOT RESOLVED**
			ipbx2.com	-	-	0	0	NOT RESOLVED
2	**Parking**	**Disabled**						**ONLINE**
			10.8.6.77(*)	-	-	0	0	
3	**Load Balancing**	**Enabled**						**ONLINE**
			10.8.6.88	-	-	0	45	OFFLINE
			10.8.6.89(*)	-	-	4	0	ONLINE
4	**Parking**	**Enabled**						**OFFLINE**
			10.8.6.66	-	-	0	45	
5	**Parking**	**Enabled**						**NOT RESOLVED**
			ipbx3.com	-	-	0	0	NOT RESOLVED
6	**Parking**	**Enabled**						**NOT RESOLVED**
			ipbx3.com(10.8.8.1)(*)	-	-	0	0	NOT RESOLVED
			ipbx3.com(10.8.8.2)	-	-	0	0	NOT RESOLVED

Table 49-5: Proxy Sets Status Table Description

Parameter	Description
Proxy Set ID	Displays the Proxy Set ID.
Mode	Displays the Proxy Sets' operational mode: ▪ "Parking" or "Homing": Redundancy mode, as configured by the ProxySet_ProxyRedundancyMode parameter. ▪ "Load Balancing: Proxy load balancing mode, as configured by the ProxySet_ProxyRedundancyMode parameter. For more information, see Configuring Proxy Sets.
Keep Alive	Displays whether the Proxy Keep-Alive feature is enabled ("Enabled") or disabled ("Disabled"), as configured by the ProxySet_EnableProxyKeepAlive parameter (see Configuring Proxy Sets).
Address	Displays the IP address of the proxy server. This can be the IP address as configured in dotted-decimal notation for the Proxy Set, or the resolved IP address of a DNS query if an FQDN is configured for the Proxy Set. IP addresses resolved from FQDNs are displayed as "<FQDN name>(<resolved IP address>)", for example, "abc.com(10.8.6.80)". The IP address that is currently used for routing is indicated with an asterisk, for example, "10.8.6.89(*)". If the FQDN failed to be resolved, only the FQDN name is displayed (e.g., "abc.com").
Priority	Displays the priority of IP addresses resolved from FQDNs. **Note:** The field is applicable only to Proxy Sets configured with FQDNs.
Weight	Displays the weight of IP addresses resolved from FQDNs. **Note:** The field is applicable only to Proxy Sets configured with FQDNs.
Success Count	Displays the total number of successful keep-alive messages (by SIP OPTIONS) sent by the device to the proxy.

Parameter	Description
Failure Count	Displays the total number of failed keep-alive messages (by SIP OPTIONS) sent by the device to the proxy.
Status	Displays the status of the Proxy Set and its' proxy servers. ▪ "ONLINE": ✓ Proxy Set ID row: At least one proxy is online as determined by the device's keep-alive feature. The status is also "ONLINE" for IP addresses resolved from DNS queries even if keep-alive is disabled. ✓ Proxy server rows (if multiple addresses): The proxy server is online as determined by the device's keep-alive feature. ▪ "OFFLINE": The proxy is offline as determined by the device's keep-alive feature and the Proxy Set is configured for Homing ('Redundancy Mode' parameter) or enabled for load balancing ('Proxy Load Balancing Method' parameter): ✓ Homing: The proxy is the main proxy, but the keep-alive has failed. ✓ Load balancing: The keep-alive for the proxy has failed. ▪ "NOT RESOLVED": Proxy address is configured as an FQDN, but the DNS resolution has failed. ▪ Empty field: Keep-alive for the proxy is disabled or the device has yet to send a keep-alive to the proxy.

49.11 Viewing IP Connectivity

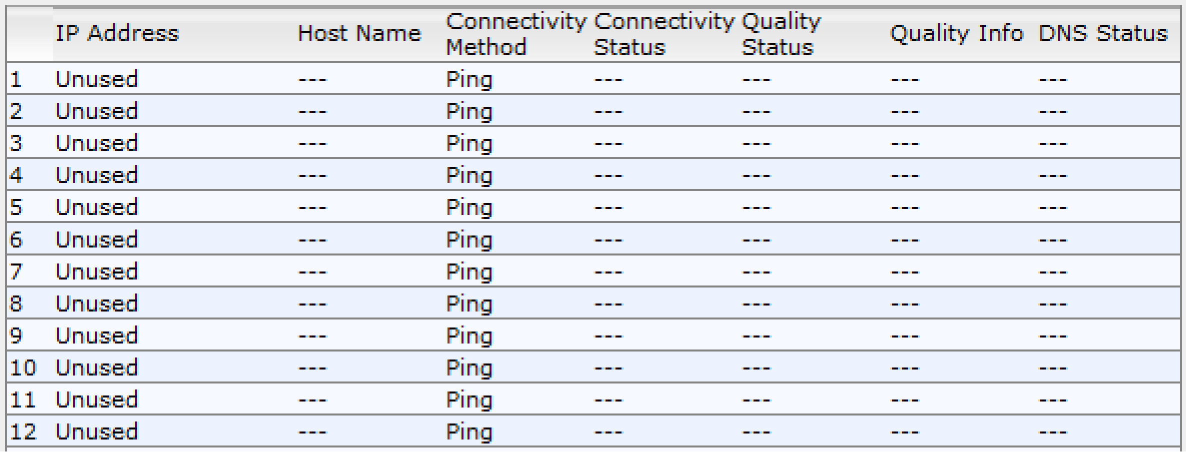
The IP Connectivity page displays on-line, read-only network diagnostic connectivity information on all destination IP addresses configured in the Tel-to-IP Routing table (see "Configuring Tel-to-IP Routing Rules" on page 467).

> **Note:** he information in columns 'Quality Status' and 'Quality Info' (per IP address) is reset if two minutes elapse without a call to that destination.

➢ To view IP connectivity information:

1. In the Routing General Parameters page, set the 'Enable Alt Routing Tel to IP' parameter (AltRoutingTel2IPMode) to **Enable** or **Status Only** (see "Configuring General Routing Parameters" on page 467).
2. Open the IP Connectivity page (**Status & Diagnostics** tab > **VoIP Status** menu > **IP Connectivity**).

Figure 49-8: IP Connectivity Page



	IP Address	Host Name	Connectivity Method	Connectivity Status	Quality Status	Quality Info	DNS Status
1	Unused	---	Ping	---	---	---	---
2	Unused	---	Ping	---	---	---	---
3	Unused	---	Ping	---	---	---	---
4	Unused	---	Ping	---	---	---	---
5	Unused	---	Ping	---	---	---	---
6	Unused	---	Ping	---	---	---	---
7	Unused	---	Ping	---	---	---	---
8	Unused	---	Ping	---	---	---	---
9	Unused	---	Ping	---	---	---	---
10	Unused	---	Ping	---	---	---	---
11	Unused	---	Ping	---	---	---	---
12	Unused	---	Ping	---	---	---	---

Table 49-6: IP Connectivity Parameters

Column Name	Description
IP Address	The IP address can be one of the following: ▪ IP address defined as the destination IP address in the Tel-to-IP Routing table. IP address resolved from the host name defined as the destination IP address in the Tel-to-IP Routing table.
Host Name	Host name (or IP address) as configured in the Tel-to-IP Routing table.
Connectivity Method	The method according to which the destination IP address is queried periodically (SIP OPTIONS request).
Connectivity Status	The status of the IP address' connectivity according to the method in the 'Connectivity Method' field. ▪ OK = Remote side responds to periodic connectivity queries. ▪ Lost = Remote side didn't respond for a short period. ▪ Fail = Remote side doesn't respond. ▪ Init = Connectivity queries not started (e.g., IP address not resolved).

Column Name	Description
	▪ Disable = The connectivity option is disabled, i.e., parameter 'Alt Routing Tel to IP Mode' (AltRoutingTel2IPMode *ini*) is set to 'None' or 'QoS'.
Quality Status	Determines the QoS (according to packet loss and delay) of the IP address. ▪ Unknown = Recent quality information isn't available. ▪ OK ▪ Poor **Notes:** ▪ The parameter is applicable only if the parameter 'Alt Routing Tel to IP Mode' is set to 'QoS' or 'Both' (AltRoutingTel2IPMode = 2 or 3). ▪ The parameter is reset if no QoS information is received for 2 minutes.
Quality Info.	Displays QoS information: delay and packet loss, calculated according to previous calls. **Notes:** ▪ The parameter is applicable only if the parameter 'Alt Routing Tel to IP Mode' is set to 'QoS' or 'Both' (AltRoutingTel2IPMode = 2 or 3). ▪ The parameter is reset if no QoS information is received for 2 minutes.
DNS Status	DNS status can be one of the following: ▪ DNS Disable ▪ DNS Resolved ▪ DNS Unresolved

50 Reporting Information to External Party

This section describes features for reporting various information to an external party.

50.1 Configuring RTCP XR

RTP Control Protocol Extended Reports (RTCP XR) is a VoIP management control that defines a set of metrics containing information for assessing VoIP call quality and for diagnosing problems. RTCP XR (RFC 3611) extends the RTCP reports defined in RFC 3550 by providing additional VoIP metrics (Quality of Experience). RTCP XR information publishing is implemented in the device according to RFC 6035. This draft defines how a SIP User Agent (UA) publishes the detailed information to a defined collector. RTCP XR measures VoIP call quality such as packet loss, delay, signal / noise / echo levels, estimated R-factor, and mean opinion score (MOS). RTCP XR measures these parameters using metrics as listed in the table below.

Notes:

- The RTCP XR feature is available only if the device is installed with a Software License Key that includes this feature. For installing a Software License Key, see "Software License Key" on page 668.
- If the RTCP XR feature is unavailable (not licensed or disabled), the R-factor VoIP metrics are not provided in CDRs (CDR fields, Local R Factor and Remote R Factor) generated by the device. Instead, these CDR fields are sent with the value 127, meaning that information is unavailable.

RTCP XR messages containing key call-quality-related metrics are exchanged periodically (user-defined) between the device and the SIP UA. This allows an analyzer to monitor these metrics midstream, or a device to retrieve them using SNMP.

You can configure the device to send RTCP XR to an IP Group.

The device sends RTCP XR in SIP PUBLISH messages. The PUBLISH message contains the following RTCP XR related header values:

- From and To: Telephone extension number of the user.
- Request-URI: IP Group name to where RTCP XR is sent
- Event: "vq-rtcpxr"
- Content-Type: "application/vq-rtcpxr"

You can configure the stage of the call at which you want the device to send RTCP XR:

- End of the call.
- Periodically, according to a user-defined interval between consecutive reports.
- (Gateway Application Only) End of a media segment. A media segment is a change in media, for example, when the coder is changed or when the caller toggles between two called parties (using call hold/retrieve). The RTCP XR sent at the end of a media segment contains information only of that segment. For call hold, the device sends RTCP XR each time the call is placed on hold and each time it is retrieved. In addition, the Start timestamp in the RTCP XR indicates the start of the media segment; the End timestamp indicates the time of the last sent periodic RTCP XR (typically, up to 5 seconds before reported segment ends).

Table 50-1: RTCP XR Published VoIP Metrics

Group	Metric Name
General	Start Timestamp
	Stop Timestamp
	Call-ID
	Local Address (IP, Port & SSRC)
	Remote Address (IP, Port & SSRC)
Session Description	Payload Type
	Payload Description
	Sample Rate
	Frame Duration
	Frame Octets
	Frames per Packets
	Packet Loss Concealment
	Silence Suppression State
Jitter Buffer	Jitter Buffer Adaptive
	Jitter Buffer Rate
	Jitter Buffer Nominal
	Jitter Buffer Max
	Jitter Buffer Abs Max
Packet Loss	Network Packet Loss Rate
	Jitter Buffer Discard Rate
Burst Gap Loss	Burst Loss Density
	Burst Duration
	Gap Loss Density
	Gap Duration
	Minimum Gap Threshold
Delay	Round Trip Delay
	End System Delay
	One Way Delay
	Interarrival Jitter
	Min Absolute Jitter
	Signal
	Signal Level
	Noise Level
	Residual Echo Return Noise
Quality Estimates	Listening Quality R

Group	Metric Name
	RLQ Est. Algorithm
	Conversational Quality R
	RCQ Est. Algorithm
	External R In
	Ext. R In Est. Algorithm
	External R Out
	Ext. R Out Est. Algorithm
	MOS-LQ
	MOS-LQ Est. Algorithm
	MOS-CQ
	MOS-CQ Est. Algorithm
	QoE Est. Algorithm

Below shows an example of a SIP PUBLISH message sent with RTCP XR and QoE information:

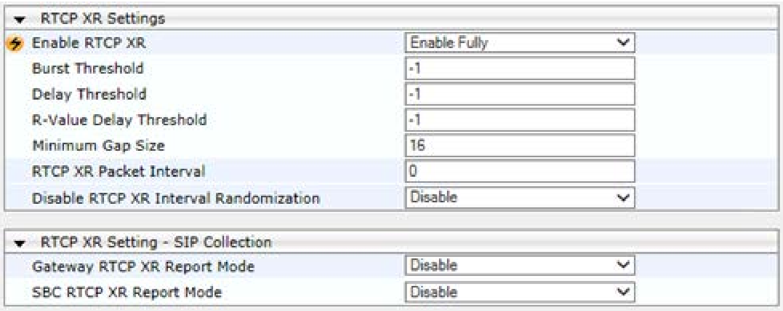
```
PUBLISH sip:172.17.116.201 SIP/2.0
Via: SIP/2.0/UDP 172.17.116.201:5060;branch=z9hG4bKac2055925925
Max-Forwards: 70
From: <sip:172.17.116.201>;tag=1c2055916574
To: <sip:172.17.116.201>
Call-ID: 20559160721612201520952@172.17.116.201
CSeq: 1 PUBLISH
Contact: <sip:172.17.116.201:5060>
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUB
SCRIBE,UPDATE
Event: vq-rtcpxr
Expires: 3600
User-Agent: device/<swver>
Content-Type: application/vq-rtcpxr
Content-Length: 1066
VQSessionReport
CallID=20328634741612201520943@172.17.116.201
LocalID: <sip:1000@172.17.116.201>
RemoteID: <sip:2000@172.17.116.202;user=phone>
OrigID: <sip:1000@172.17.116.201>
LocalAddr: IP=172.17.116.201 Port=6000 SSRC=0x54c62a13
RemoteAddr: IP=172.17.116.202 Port=6000 SSRC=0x243220dd
LocalGroup:
RemoteGroup:
LocalMAC: 00:90:8f:57:d9:71
LocalMetrics:
Timestamps: START=2015-12-16T20:09:45Z STOP=2015-12-16T20:09:52Z
SessionDesc: PT=8 PD=PCMA SR=8000 FD=20 PLC=3 SSUP=Off
JitterBuffer: JBA=3 JBR=0 JBN=7 JBM=10 JBX=300
```

```
PacketLoss: NLR=0.00 JDR=0.00
BurstGapLoss: BLD=0.00 BD=0 GLD=0.00 GD=6325 GMIN=16
Delay: RTD=0 ESD=11
Signal: SL=-34 NL=-67 RERL=17
QualityEst: RLQ=93 MOSLQ=4.1
MOSCQ=4.10
RemoteMetrics:
Timestamps: START=2015-12-16T20:09:45Z STOP=2015-12-16T20:09:52Z
JitterBuffer: JBA=3 JBR=0 JBN=0 JBM=0 JBX=300
PacketLoss: NLR=0.00 JDR=0.00
BurstGapLoss: BLD=0.00 BD=0 GLD=0.00 GD=0 GMIN=16
Delay: RTD=65535 ESD=0
QualityEst:
DialogID: 20328634741612201520943@172.17.116.201;to-
tag=1c1690611502;from-tag=1c2032864069
```

➢ To configure RTCP XR:

1. Open the RTP/RTCP Settings page (**Configuration** tab > **VoIP** menu > **Media** > **RTP/RTCP Settings**). The RTCP XR parameters are listed under the RTCP XR Settings group:

Figure 50-1: RTCP XR Parameters in RTP/RTCP Settings Page



RTCP XR Settings

Enable RTCP XR	Enable Fully
Burst Threshold	-1
Delay Threshold	-1
R-Value Delay Threshold	-1
Minimum Gap Size	16
RTCP XR Packet Interval	0
Disable RTCP XR Interval Randomization	Disable

RTCP XR Setting - SIP Collection

Gateway RTCP XR Report Mode	Disable
SBC RTCP XR Report Mode	Disable

2. Under the RTCP XR Settings group, configure the following:
   - 'Enable RTCP XR' (*VQMonEnable*) - enables voice quality monitoring and RTCP XR.
   - 'Burst Threshold' (*VQMonBurstHR*) - defines the voice quality monitoring excessive burst alert threshold.
   - 'Delay Threshold' (*VQMonDelayTHR*) - defines the voice quality monitoring excessive delay alert threshold.
   - 'R-Value Delay Threshold' (*VQMonEOCRValTHR*) - defines the voice quality monitoring end of call low quality alert threshold.
   - 'Minimum Gap Size' (*VQMonGMin*) - defines the voice quality monitoring minimum gap size (number of frames).
   - 'RTCP XR Packet Interval' (*RTCPInterval*) - defines the time interval between adjacent RTCP reports.
   - 'Disable RTCP XR Interval Randomization' (*DisableRTCPRandomize*) - determines whether RTCP report intervals are randomized or whether each report interval accords exactly to the parameter RTCPInterval.
3. Under the RTCP XR Setting - SIP Collection group, configure the following:
   - (Gateway Application Only) 'Gateway RTCP XR Report Mode' (RTCPXRReportMode) - enables RTCP XR reports and defines the interval at which they are sent.

- (SBC Application Only) 'SBC RTCP XR Report Mode' (SBCRtcpXrReportMode) - enables the sending of RTCP XR reports of QoE metrics at the end of each call session (i.e., after a SIP BYE).

4. Using the PublicationIPGroupID ini file parameter, define the IP Group to where you want to send the RTCP XR.
5. Click **Submit**, and then reset the device with a save ("burn") for your settings to take effect.

50.2 Generating Call Detail Records

The Call Detail Record (CDR) contains vital statistic information on calls made from the device. The device can be configured to generate and report CDRs for various stages of the call, including SIP messages and/or media. You can configure when CDRs for a call are generated, for example, only at the end of the call or only at the start and end of the call.

Once generated, the device can send the CDRs to any of the following:

- Syslog server. The CDR Syslog message complies with RFC 3164 and is identified by Facility 17 (local1) and Severity 6 (Informational).
- RADIUS server. For CDR in RADIUS format, see "Configuring RADIUS Accounting" on page 760. For configuring RADIUS servers for CDR reporting, see "Configuring RADIUS Servers" on page 224.

Note: You can view the latest CDRs, which are stored on the device's memory, in the CDR History table. For more information, see Viewing CDR History on page 723.

50.2.1 CDR Field Description

This section describes the default CDR fields that are generated by the device.

Note: You can customize the default CDR fields if desired. For customizing Gateway-related CDRs, see Customizing CDRs for Gateway Calls on page 751. For customizing SBC-related CDRs, see Customizing CDRs for SBC Calls on page 754.

50.2.1.1 CDR Fields for SBC Signaling

The default CDR fields for SBC signaling are listed in the table below.

A typical SBC session consists of two SBC legs. Each leg generates its own signaling CDRs. Each leg generates three CDR types: at call start (SBCReportType=CALL_START), connect time (SBCReportType=CALL_CONNECT) and when the call ends (SBCReportType=CALL_END). CDRs belonging to the same SBC session (both legs) have the same Session ID (SessionId CDR field). CDRs belonging to the same SBC leg have the same SIP Call ID (SIPCallId CDR field).

For billing applications, the CDR that is sent when the call ends (CALL_END) is usually sufficient. Billing may be based on the following:

- Call ID (SIPCallId CDR field)
- Source URI (SrcURI CDR field)
- Destination URI (DstURI CDR field)
- Call originator (Orig CDR field) - indicates the call direction (caller)

- Call duration (Durat CDR field) - call duration (elapsed time) from call connect
- Call time is based on SetupTime, ConnectTime and ReleaseTime CDR fields

Table 50-2: Default CDR Fields for SBC Signaling

CDR Field Name	Description	Format
SBCReportType	Report type: ▪ "CALL_START" ▪ "CALL_CONNECT" ▪ "CALL_END" ▪ "DIALOG_START" ▪ "DIALOG_END"	String
EPTyp	Endpoint type: ▪ "SBC"	String
SIPMethod	SIP message type	String of up to 10 characters
SIPCallId	Unique ID of call	String of up to 50 characters
SessionId	Unique Session ID	String of up to 10 characters
Orig	Call originator: ▪ "LCL" - local ▪ "RMT" - remote	String
SourceIp	Source IP address	String of up to 20 characters
SourcePort	Source UDP port	String of up to 10 characters
DestIp	Destination IP address	String of up to 20 characters
DestPort	Destination UDP port	String of up to 10 characters
TransportType	Transport type: ▪ "UDP" ▪ "TCP" ▪ "TLS"	String
SrcURI	Source URI	String of up to 41 characters
SrcURIBeforeMap	Source URI before manipulation	String of up to 41 characters
DstURI	Destination URI	String of up to 41 characters
DstURIBeforeMap	Destination URI before manipulation	String of up to 41 characters
Durat	Call duration (in seconds)	String of up to 5 characters
TrmSd	Termination side: ▪ "LCL" - local ▪ "RMT" - remote	String
TrmReason	Termination reason	String of up to 40 characters
TrmReasonCategory	Termination reason category: **Calls with duration 0 (i.e., not connected):** ▪ NO_ANSWER: ✓ "GWAPP_NORMAL_CALL_CLEAR" ✓ "GWAPP_NO_USER_RESPONDING"	String of up to 17 characters

CDR Field Name	Description	Format
	✓ "GWAPP_NO_ANSWER_FROM_USER_ALERTED" ▪ BUSY: ✓ "GWAPP_USER_BUSY" ▪ NO_RESOURCES: ✓ "GWAPP_RESOUUCE_UNAVAILABLE_UNSPECIFIED" ✓ "RELEASE_BECAUSE_NO_CONFERENCE_RESOURCES_LEFT" ✓ "RESOURCE_BECAUSE_NO_TRANSCODING_RESOURCES_LEFT" ✓ "RELEASE_BECAUSE_GW_LOCKED" ▪ NO_MATCH: ✓ "RELEASE_BECAUSE_UNMATCHED_CAPABILITIES" ▪ FORWARDED: ✓ "RELEASE_BECAUSE_FORWARD" ▪ GENERAL_FAILED: Any other reason **Calls with duration:** ▪ NORMAL_CALL_CLEAR: ✓ "GWAPP_NORMAL_CALL_CLEAR" ✓ ABNORMALLY_TERMINATED: Anything else **N/A** - Reasons not belonging to above categories.	
SetupTime	Call setup time **Note:** To configure the time zone string (e.g., "UTC" - default, "GMT+1", and "EST"), use the TimeZoneFormat parameter.	String of up to 35 characters
ConnectTime	Call connect time **Note:** To configure the time zone string (e.g., "UTC" - default, "GMT+1", and "EST"), use the TimeZoneFormat parameter.	String of up to 35 characters
ReleaseTime	Call release time **Note:** To configure the time zone string (e.g., "UTC" - default, "GMT+1", and "EST"), use the TimeZoneFormat parameter.	String of up to 35 characters
RedirectReason	Redirect reason	String of up to 15 characters
RedirectURINum	Redirection URI	String of up to 41 characters
RedirectURINumBeforeMap	Redirect URI number before manipulation	String of up to 41 characters

CDR Field Name	Description	Format
TxSigIPDiffServ	Signaling IP DiffServ	String of up to 15 characters
IPGroup	IP Group ID and name	String of up to 40 characters
SrdId	SRD ID and name	String of up to 29 characters
SIPInterfaceId	SIP Interface ID	String of up to 15 characters
ProxySetId	Proxy Set ID	String of up to 15 characters
IpProfileId	IP Profile ID and name	String of up to 34 characters
MediaRealmId	Media Realm ID and name	String of up to 55 characters
DirectMedia	Direct media or traversing SBC: ▪ "yes" ▪ "no"	String
SIPTrmReason	SIP call termination reason (BYE, CANCEL, or SIP error codes, e.g., 404)	String of up to 12 characters
SipTermDesc	Description of SIP termination reason: ▪ SIP Reason header, if exists, for example: SIP ;cause=200 ;text="Call completed elsewhere". ▪ If no SIP Reason header exists, the description is taken from the reason text, if exists, of the SIP response code, for example: "417 Unknown Resource-Priority". ▪ If no reason text exists in the SIP response code, the description is taken from an internal SIP response mapping mechanism. For example, if the device receives a SIP response "422", it sends in the CDR "422 Session Interval Too Small method" as the description.	String of up to 26 characters
Caller	Name of caller	String of up to 36 characters
Callee	Name of called party	String of up to 36 characters

Below shows an example of an SBC signaling CDR sent at the end of a call (call was terminated normally):

```
[S=40] |SBCReportType |EPTyp |SIPCallId |SessionId |Orig |SourceIp
|SourcePort |DestIp |DestPort |TransportType |SrcURI
|SrcURIBeforeMap |DstURI |DstURIBeforeMap |Durat |TrmSd |TrmReason
|TrmReasonCategory |SetupTime |ConnectTime |ReleaseTime
|RedirectReason |RedirectURINum |RedirectURINumBeforeMap
|TxSigIPDiffServ|IPGroup (description) |SrdId (name)
|SIPInterfaceId |ProxySetId |IpProfileId (name) |MediaRealmId
(name) |DirectMedia |SIPTrmReason |SIPTermDesc |Caller |Callee
[S=41] |CALL_END |SBC |20767593291410201017029@10.33.45.80
|1871197419|LCL |10.33.45.80 |5060 |10.33.45.72 |5060 |UDP
|9001@10.8.8.10 |9001@10.8.8.10 |6001@10.33.45.80
|6001@10.33.45.80 |15 |LCL |GWAPP_NORMAL_CALL_CLEAR
|NORMAL_CALL_CLEAR |17:00:29.954  UTC Thu Oct 14 2014
```

```
|17:00:49.052  UTC Thu Oct 14 2014 |17:01:04.953  UTC Thu Oct 14
2014 |-1 | | |40 |1 |0 (SRD_GW) |1 |1 |1 () |0 (MR_1) |no |BYE
|Q.850 ;cause=16 ;text="loc |user 9928019 |
```

50.2.1.2 CDR Fields for SBC Media

The default CDR fields for SBC media are listed in the table below. The media CDRs are published for each active media stream, thereby allowing multiple media CDRs, where each media CDR has a unique call ID corresponding to the signaling CDR.

Table 50-3: Default CDR Fields for SBC Media

CDR Field Name	Description
MediaReportType	Report type (media start, update, or end)
SIPCallId	Unique call ID
Cid	Channel CID
MediaType	Media type (audio, video, or text)
Coder	Coder name
PacketInterval	Coder packet interval
LocalRtpIp	Local RTP IP address
LocalRtpPort	Local RTP port
RemoteRtpIp	Remote RTP IP address
RemoteRtpPort	Remote RTP port
InPackets	Number of received packets
OutPackets	Number of sent packets
LocalPackLoss	Local packet loss
RemotePackLoss	Remote packet loss
RTPdelay	RTP delay
RTPjitter	RTP jitter
TxRTPssrc	Tx RTP SSRC
RxRTPssrc	Local RTP SSRC
LocalRFactor	Local conversation quality **Note:** If the RTCP XR feature is unavailable (not licensed or disabled), this R-factor VoIP metric is not provided. Instead, the device sends the CDR field with the value 127, meaning that information is unavailable.
RemoteRFactor	Remote conversation quality **Note:** If the RTCP XR feature is unavailable (not licensed or disabled), this R-factor VoIP metric is not provided. Instead, the device sends the CDR field with the value 127, meaning that information is unavailable.
LocalMosCQ	Local MOS for conversation
RemoteMosCQ	Remote MOS for conversation
TxRTPIPDiffServ	Media IP DiffServ
LatchedRtpIp	Remote IP address of the incoming RTP stream that the device "latched" on to as a result of the RTP latching mechanism for NAT traversal.

CDR Field Name	Description
LatchedRtpPort	Remote RTP port of the incoming RTP stream that the device "latched" on to as a result of the RTP latching mechanism for NAT traversal.
LatchedT38Ip	Latching of a new T.38 stream - new IP address
LatchedT38Port	Latching of a new T.38 stream - new port

50.2.1.3 CDR Fields for Gateway Application

The default CDR fields for the Gateway calls are listed in the table below.

Table 50-4: Default CDR Fields for Gateway Calls

Field Name	Description
GWReportType	Report type: ▪ CALL_START ▪ CALL_CONNECT ▪ CALL_END
Cid	Port number
SessionId	SIP session identifier
Trunk	Physical trunk number
BChan	Selected B-channel
ConId	SIP conference ID
TG	Trunk Group ID
EPTyp	Endpoint type: ▪ FXO ▪ FXS ▪ EANDM ▪ ISDN ▪ CAS ▪ DAA ▪ IPMEDIA ▪ NETANN ▪ STREAMING ▪ TRANSPARENT ▪ MSCML ▪ VXML
Orig	Call originator: ▪ LCL (Tel side) ▪ RMT (IP side)
SourceIp	Source IP address
DestIp	Destination IP address
TON	Source phone number type
NPI	Source phone number plan

Field Name	Description
SrcPhoneNum	Source phone number
SrcNumBeforeMap	Source number before manipulation
TON	Destination phone number type
NPI	Destination phone number plan
DstPhoneNum	Destination phone number
DstNumBeforeMap	Destination number before manipulation
Durat	Call duration
Coder	Selected coder
Intrv	Packet interval
RtpIp	RTP IP address
Port	Remote RTP port
TrmSd	Initiator of call release (IP, Tel, or Unknown)
TrmReason	SIP call termination reason (see "Release Reasons in CDR for Gateway Application" on page 746)
Fax	Fax transaction during call
InPackets	Number of incoming packets
OutPackets	Number of outgoing packets
PackLoss	Local packet loss
RemotePackLoss	Number of outgoing lost packets
SIPCallId	Unique SIP call ID
SetupTime	Call setup time **Note:** To configure the time zone string (e.g., "UTC" - default, "GMT+1", and "EST"), use the TimeZoneFormat parameter.
ConnectTime	Call connect time **Note:** To configure the time zone string (e.g., "UTC" - default, "GMT+1", and "EST"), use the TimeZoneFormat parameter.
ReleaseTime	Call release time **Note:** To configure the time zone string (e.g., "UTC" - default, "GMT+1", and "EST"), use the TimeZoneFormat parameter.
RTPdelay	RTP delay
RTPjitter	RTP jitter
RTPssrc	Local RTP SSRC
RemoteRTPssrc	Remote RTP SSRC
RedirectReason	Redirect reason
TON	Redirection phone number type
NPI	Redirection phone number plan
RedirectPhonNum	Redirection phone number
MeteringPulses	Number of generated metering pulses

Field Name	Description
SrcHost	Source host name
SrcHostBeforeMap	Source host name before manipulation
DstHost	Destination host name
DstHostBeforeMap	Destination host name before manipulation
IPG	IP Group description
LocalRtpIp	Remote RTP IP address
LocalRtpPort	Local RTP port
Amount	0-999999 Data is stored per call and sent in the syslog as follows: ▪ currency-type: amount multiplier for currency charge (euro or usd) ▪ recorded-units: for unit charge (1-999999)
Mult	0,001-1000 (in steps of 10) (See explanation above.)
TrmReasonCategory	Termination reason category: ▪ Calls with duration 0 (i.e., not connected): ✓ **NO_ANSWER -** GWAPP_NORMAL_CALL_CLEAR, GWAPP_NO_USER_RESPONDING, GWAPP_NO_ANSWER_FROM_USER_ALERTED ✓ **BUSY -** GWAPP_USER_BUSY ✓ **NO_RESOURCES -** GWAPP_RESOUUCE_UNAVAILABLE_UNSPECIFIED, RELEASE_BECAUSE_NO_CONFERENCE_RESOURCES_LEFT, RESOURCE_BECAUSE_NO_TRANSCODING_RESOURCES_LEFT, RELEASE_BECAUSE_GW_LOCKED ✓ **NO_MATCH -** RELEASE_BECAUSE_UNMATCHED_CAPABILITIES ✓ **FORWARDED -** RELEASE_BECAUSE_FORWARD ✓ **GENERAL_FAILED -** any other reason ▪ Calls with duration: ✓ **NORMAL_CALL_CLEAR -** GWAPP_NORMAL_CALL_CLEAR ✓ **ABNORMALLY_TERMINATED -** Anything else ▪ **N/A -** Reasons not belonging to above categories
RedirectNumBeforeMap	Redirect number before manipulation
SrdId	SRD ID name
SIPInterfaceId	SIP interface ID
ProxySetId	Proxy Set ID
IpProfileId	IP Profile name
MediaRealmId	Media Realm name
SigTransportType	SIP signaling transport type (UDP, TCP, or TLS)
TxRTPIPDiffServ	Media IP DiffServ
TxSigIPDiffServ	Signaling IP DiffServ

Field Name	Description
LocalRFactor	Local R-factor **Note:** If the RTCP XR feature is unavailable (not licensed or disabled), this R-factor VoIP metric is not provided. Instead, the device sends the CDR field with the value 127, meaning that information is unavailable.
RemoteRFactor	Remote R-factor **Note:** If the RTCP XR feature is unavailable (not licensed or disabled), this R-factor VoIP metric is not provided. Instead, the device sends the CDR field with the value 127, meaning that information is unavailable.
LocalMosCQ	Local MOS for conversation quality
RemoteMosCQ	Remote MOS for conversation quality
SigSourcePort	SIP source port
SigDestPort	SIP destination port
MediaType	Media type - audio, video, or text
AMD	Information relating to the Automatic Machine Detection (AMD) feature: ▪ V - voice ▪ A - answer machine ▪ S - silence ▪ U - unknown
%	Information relating to AMD that shows the success that the answering type (probability) was correctly detected
SIPTrmReason	SIP call termination reason (BYE, CANCEL, or SIP error codes, e.g., 404)
SipTermDesc	Description of SIP termination reason: ▪ SIP Reason header, if exists, for example: SIP ;cause=200 ;text="Call completed elsewhere". ▪ If no SIP Reason header exists, the description is taken from the reason text, if exists, of the SIP response code, for example: "417 Unknown Resource-Priority". ▪ If no reason text exists in the SIP response code, the description is taken from an internal SIP response mapping mechanism. For example, if the device receives a SIP response "422", it sends in the CDR "422 Session Interval Too Small method" as the description.
PstnTermReason	Q.850 protocol termination reason (0-127).
LatchedRtpIp	Remote IP address of the incoming RTP stream that the device "latched" on to as a result of the RTP latching mechanism for NAT traversal.
LatchedRtpPort	Remote RTP port of the incoming RTP stream that the device "latched" on to as a result of the RTP latching mechanism for NAT traversal.
LatchedT38Ip	Latching of a new T.38 stream - new IP address
LatchedT38Port	Latching of a new T.38 stream - new port

50.2.1.3.1Release Reasons in CDR for Gateway Application

The possible reasons for call termination for the Gateway application which is represented in the CDR field **TrmReason** are listed below:

- "REASON N/A"

- "RELEASE_BECAUSE_NORMAL_CALL_DROP"
- "RELEASE_BECAUSE_DESTINATION_UNREACHABLE"
- "RELEASE_BECAUSE_DESTINATION_BUSY"
- "RELEASE_BECAUSE_NOANSWER"
- "RELEASE_BECAUSE_UNKNOWN_REASON"
- "RELEASE_BECAUSE_REMOTE_CANCEL_CALL"
- "RELEASE_BECAUSE_UNMATCHED_CAPABILITIES"
- "RELEASE_BECAUSE_UNMATCHED_CREDENTIALS"
- "RELEASE_BECAUSE_UNABLE_TO_HANDLE_REMOTE_REQUEST"
- "RELEASE_BECAUSE_NO_CONFERENCE_RESOURCES_LEFT"
- "RELEASE_BECAUSE_CONFERENCE_FULL"
- "RELEASE_BECAUSE_VOICE_PROMPT_PLAY_ENDED"
- "RELEASE_BECAUSE_VOICE_PROMPT_NOT_FOUND"
- "RELEASE_BECAUSE_TRUNK_DISCONNECTED"
- "RELEASE_BECAUSE_RSRC_PROBLEM"
- "RELEASE_BECAUSE_MANUAL_DISC"
- "RELEASE_BECAUSE_SILENCE_DISC"
- "RELEASE_BECAUSE_RTP_CONN_BROKEN"
- "RELEASE_BECAUSE_DISCONNECT_CODE"
- "RELEASE_BECAUSE_GW_LOCKED"
- "RELEASE_BECAUSE_NORTEL_XFER_SUCCESS"
- "RELEASE_BECAUSE_FAIL"
- "RELEASE_BECAUSE_FORWARD"
- "RELEASE_BECAUSE_ANONYMOUS_SOURCE"
- "RELEASE_BECAUSE_IP_PROFILE_CALL_LIMIT"
- "GWAPP_UNASSIGNED_NUMBER"
- "GWAPP_NO_ROUTE_TO_TRANSIT_NET"
- "GWAPP_NO_ROUTE_TO_DESTINATION"
- "GWAPP_CHANNEL_UNACCEPTABLE"
- "GWAPP_CALL_AWARDED_AND "
- "GWAPP_PREEMPTION"
- "PREEMPTION_CIRCUIT_RESERVED_FOR_REUSE"
- "GWAPP_NORMAL_CALL_CLEAR"
- "GWAPP_USER_BUSY"
- "GWAPP_NO_USER_RESPONDING"
- "GWAPP_NO_ANSWER_FROM_USER_ALERTED"
- "MFCR2_ACCEPT_CALL"
- "GWAPP_CALL_REJECTED"
- "GWAPP_NUMBER_CHANGED"
- "GWAPP_NON_SELECTED_USER_CLEARING"
- "GWAPP_INVALID_NUMBER_FORMAT"
- "GWAPP_FACILITY_REJECT"
- "GWAPP_RESPONSE_TO_STATUS_ENQUIRY"
- "GWAPP_NORMAL_UNSPECIFIED"

- "GWAPP_CIRCUIT_CONGESTION"
- "GWAPP_USER_CONGESTION"
- "GWAPP_NO_CIRCUIT_AVAILABLE"
- "GWAPP_NETWORK_OUT_OF_ORDER"
- "GWAPP_NETWORK_TEMPORARY_FAILURE"
- "GWAPP_NETWORK_CONGESTION"
- "GWAPP_ACCESS_INFORMATION_DISCARDED"
- "GWAPP_REQUESTED_CIRCUIT_NOT_AVAILABLE"
- "GWAPP_RESOURCE_UNAVAILABLE_UNSPECIFIED"
- "GWAPP_PERM_FR_MODE_CONN_OUT_OF_S"
- "GWAPP_PERM_FR_MODE_CONN_OPERATIONAL"
- "GWAPP_PRECEDENCE_CALL_BLOCKED"
    - "RELEASE_BECAUSE_PREEMPTION_ANALOG_CIRCUIT_RESERVED_FOR_REUSE"
    - "RELEASE_BECAUSE_PRECEDENCE_CALL_BLOCKED"
- "GWAPP_QUALITY_OF_SERVICE_UNAVAILABLE"
- "GWAPP_REQUESTED_FAC_NOT_SUBSCRIBED"
- "GWAPP_BC_NOT_AUTHORIZED"
- "GWAPP_BC_NOT_PRESENTLY_AVAILABLE"
- "GWAPP_SERVICE_NOT_AVAILABLE"
- "GWAPP_CUG_OUT_CALLS_BARRED"
- "GWAPP_CUG_INC_CALLS_BARRED"
- "GWAPP_ACCES_INFO_SUBS_CLASS_INCONS"
- "GWAPP_BC_NOT_IMPLEMENTED"
- "GWAPP_CHANNEL_TYPE_NOT_IMPLEMENTED"
- "GWAPP_REQUESTED_FAC_NOT_IMPLEMENTED"
- "GWAPP_ONLY_RESTRICTED_INFO_BEARER"
- "GWAPP_SERVICE_NOT_IMPLEMENTED_UNSPECIFIED"
- "GWAPP_INVALID_CALL_REF"
- "GWAPP_IDENTIFIED_CHANNEL_NOT_EXIST"
- "GWAPP_SUSPENDED_CALL_BUT_CALL_ID_NOT_EXIST"
- "GWAPP_CALL_ID_IN_USE"
- "GWAPP_NO_CALL_SUSPENDED"
- "GWAPP_CALL_HAVING_CALL_ID_CLEARED"
- "GWAPP_INCOMPATIBLE_DESTINATION"
- "GWAPP_INVALID_TRANSIT_NETWORK_SELECTION"
- "GWAPP_INVALID_MESSAGE_UNSPECIFIED"
- "GWAPP_NOT_CUG_MEMBER"
- "GWAPP_CUG_NON_EXISTENT"
- "GWAPP_MANDATORY_IE_MISSING"
- "GWAPP_MESSAGE_TYPE_NON_EXISTENT"
- "GWAPP_MESSAGE_STATE_INCONSISTENCY"
- "GWAPP_NON_EXISTENT_IE"
- "GWAPP_INVALID_IE_CONTENT"
- "GWAPP_MESSAGE_NOT_COMPATIBLE"

- "GWAPP_RECOVERY_ON_TIMER_EXPIRY"
- "GWAPP_PROTOCOL_ERROR_UNSPECIFIED"
- "GWAPP_INTERWORKING_UNSPECIFIED"
- "GWAPP_UKNOWN_ERROR"
- "RELEASE_BECAUSE_HELD_TIMEOUT"

50.2.1.4 CDR Fields for Locally Stored SBC

The CDR fields for SBC calls that are stored locally (history) on the device are listed in the table below. For storing CDRs locally, see "Storing CDRs on the Device" on page 758.

Table 50-5: Default CDR Fields for Locally Stored (History) CDRs

CDR Field	Title
Report Type	SBCReportType
Endpoint Type	EPTyp
Call Id	SIPCallId
Session ID	SessionId
Call Orig	Orig
Source IP	SourceIp
Source Port	SourcePort
Destination IP	DestIp
Destination Port	DestPort
Transport Type	TransportType
Source URI	SrcURI
Source URI Before Manipulation	SrcURIBeforeMap
Destination URI	DstURI
Destination URI Before Manipulation	DstURIBeforeMap
Call Duration	Durat
Termination Side	TrmSd
Termination Reason	TrmReason
Termination Reason Category	TrmReasonCategory
Setup Time	SetupTime **Note:** To configure the time zone string (e.g., "UTC" - default, "GMT+1", and "EST"), use the TimeZoneFormat parameter.
Connect Time	ConnectTime **Note:** To configure the time zone string (e.g., "UTC" - default, "GMT+1", and "EST"), use the TimeZoneFormat parameter.
Release Time	ReleaseTime **Note:** To configure the time zone string (e.g., "UTC" - default, "GMT+1", and "EST"), use the TimeZoneFormat parameter.
Redirect Reason	RedirectReason
Redirect URI	RedirectURINum
Redirect URI Before Manipulation	RedirectURINumBeforeMap
Signaling IP DiffServ	TxSigIPDiffServ
IP Group Description	IPGroup (description)

CDR Field	Title
SRD Name	SrdId (name)
SIP Interface ID	SIPInterfaceId
Proxy Set ID	ProxySetId
IP Profile ID	IpProfileId (name)
Media Realm Name	MediaRealmId (name)
Direct Media	DirectMedia
SIP Termination Reason	SIPTrmReason
SIP Termination Description	SIPTermDesc
Caller Display ID	Caller
Callee Display ID	Callee

50.2.2 Customizing CDRs for Gateway Calls

The Gateway CDR Format table lets you configure CDR customization rules for Gateway-related CDRs that are sent in Syslog messages and/or RADIUS accounting request messages. The table lets you configure up to 128 CDR customization rules for Syslog CDRs and up to 40 rules for RADIUS-accounting CDRs. If you do not configure a CDR customization rule for a specific CDR, the device generates the CDR in a predefined default CDR format (see CDR Field Description on page 737).

For RADIUS accounting, you can customize CDRs for standard RADIUS Attributes and vendor-specific RADIUS Attributes (VSA). You can customize the RADIUS Attribute's prefix name (*Column Type*) and ID. For example, instead of the default VSA name, "h323-connect-time" with ID 28, you can change the name to "Call-Connect-Time" with ID 29.

Notes:

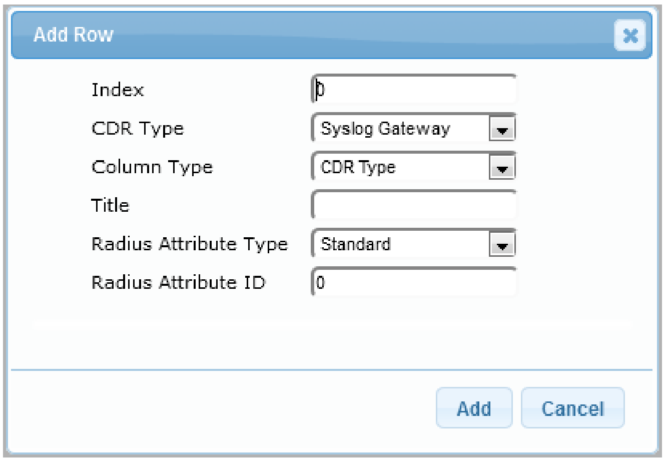
- The following standard RADIUS Attributes cannot be customized: 1 through 6, 18 through 20, 22, 23, 27 through 29, 32, 34 through 39, 41, 44, 52, 53, 55, 60 through 85, 88, 90, and 91.
- If the RTCP XR feature is unavailable (not licensed or disabled), the R-factor VoIP metrics are not provided in CDRs (CDR fields, Local R Factor and Remote R Factor) generated by the device. Instead, these CDR fields are sent with the value 127, meaning that information is unavailable.

The following procedure describes how to customize Gateway CDRs through the Web interface. You can also configure it through ini file (GWCDRFormat) or CLI (configure voip > services cdr > cdr-format gw-cdr-format).

➢ **To customize Gateway CDRs:**

1. Open the Gateway CDR Format table (**Configuration** tab > **System** menu > **Call Detail Record** > **Gateway CDR Format**).
2. Click **Add**; the following dialog box appears:

Figure 50-2: Gateway CDR Format Table - Add Row Dialog Box



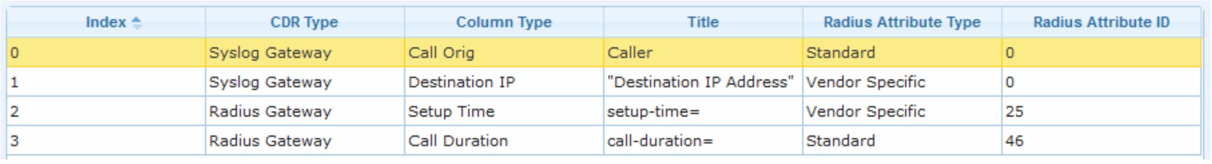
Add Row

Index	0
CDR Type	Syslog Gateway
Column Type	CDR Type
Title	
Radius Attribute Type	Standard
Radius Attribute ID	0

3. Configure the CDR according to the parameters described in the table below.
4. Click **Add**.

An example of CDR customization rules configured in the table is shown below:

Figure 50-3: Example of CDR Customization Rules for Gateway Calls



Index	CDR Type	Column Type	Title	Radius Attribute Type	Radius Attribute ID
0	Syslog Gateway	Call Orig	Caller	Standard	0
1	Syslog Gateway	Destination IP	"Destination IP Address"	Vendor Specific	0
2	Radius Gateway	Setup Time	setup-time=	Vendor Specific	25
3	Radius Gateway	Call Duration	call-duration=	Standard	46

Gateway CDR Format Table Parameter Descriptions

Parameter	Description
Index [GWCDRFormat_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
CDR Type `cdr-type` [GWCDRFormat_CDRType]	Defines the application type for which you want to customize CDRs. ▪ [0] Syslog Gateway = (Default) Customizes CDR field names for CDRs sent in Syslog messages. ▪ [1] RADIUS Gateway = Customizes CDR field names (RADIUS Attribute prefix names) for CDRs sent in RADIUS accounting requests.
Column Type `col-type` [GWCDRFormat_ColumnType]	Defines the CDR field (column) that you want to customize. [0] CDR Type (**default**); [1] Call ID; [2] Session ID; [3] Report Type; [4] Media Type; [5] Accounting Status Type; [6] H323 ID; [7] Radius Call ID; [8] Blank; [10] Endpoint Type; [11] Call Orig; [12] Source IP; [13] Destination IP; [14] Remote IP; [15] Source Port; [16] Dest Port; [17] Remote Port; [18] Call Duration; [19] Termination Side; [20] Termination Reason; [21] Setup Time; [22] Connect Time; [23] Release Time; [24] Redirect Reason; [25] Was Call Started; [26] IP Group ID; [27] IP Group Name; [28] SRD ID; [29] SRD Name; [30] SIP Interface ID; [31] Transport Type; [32] Signaling IP DiffServ; [33] Termination Reason Category; [34] Proxy Set ID; [35] IP Profile ID; [36] IP Profile Name; [37] Media Realm ID; [38] Media Realm Name; [39] SIP Termination Reason; [40] SIP Termination Description; [41] Caller Display ID; [42] Callee Display ID; [43] SIPInterface Name; [44] Call Orig Radius; [45] Termination Side Radius; [46] Termination Side Yes No; [47] Termination Reason Value; [48] ProxySet Name; [100] Trunk ID; [101] B-Channel; [102] Conn ID; [103] Trunk Group ID; [104] Metering Pulses Generated; [105] Fax On Call; [106] Source Number Before Manipulation; [107]

Parameter	Description
	Source Number; [108] Source Number Type; [109] Source Number Plan; [110] Destination Number Before Manipulation; [111] Destination Number; [112] Destination Number Type; [113] Destination Number Plan; [114] Redirect Number Before Manipulation; [115] Redirect Number; [116] Source Host Name Before Manipulation; [117] Source Host Name; [118] Destination Host Name Before Manipulation; [119] Destination Host Name; [120] PSTN Termination Reason; [121] Module And Port; [122] AOC Currency; [123] AOC Amount; [124] AOC Multiplier; [125] ISDN Line Type; [150] Channel ID; [151] Coder Type;[152] Packet Interval; [153] Payload Type; [154] Local Input Packets; [155] Local Output Packets; [156] Local Input Octets; [157] Local Output Octets; [158] Local Packet Loss; [159] Local Round Trip Delay; [160] Local Jitter; [161] Local SSRC Sender; [162] Remote Input Packets; [163] Remote Output Packets; [164] Remote Input Octets; [165] Remote Output Octets; [166] Remote Packet Loss; [167] Remote Round Trip Delay; [168] Remote Jitter; [169] Remote SSRC Sender; [170] Local RTP IP; [171] Local RTP Port; [172] Remote RTP IP; [173] Remote RTP Port; [174] RTP IP DiffServ; [175] Local R Factor; [176] Remote R Factor; [177] Local MOS CQ; [178] Remote MOS CQ; [179] AMD Decision; [180] AMD Decision Probability; [181] Latched RTP IP; [182]Latched RTP Port; [183] Latched T38 IP; [184] Latched T38 Port.
Title `title` [GWCDRFormat_Title]	Defines a new name for the CDR field (for Syslog) or for the RADIUS Attribute prefix name (for RADIUS accounting) that you selected in the 'Column Type' parameter. The valid value is a string of up to 31 characters. You can configure the name to be enclosed by apostrophes (single or double). For example, if you want the CDR field name to appear as 'Phone Duration', you must configure the parameter to 'Phone Duration'. You can also configure the CDR field name with an equals (=) sign, for example "call-connect-time=". **Notes:** ▪ For RADIUS Attributes that do not require a prefix name, leave the parameter undefined. ▪ The parameter's value is case-sensitive. For example, if you want the CDR field name to be Phone-Duration, you must configure the parameter to "Phone-Duration" (i.e., upper case "P" and "D").
RADIUS Attribute Type `radius-type` [GWCDRFormat_RadiusType]	Defines whether the RADIUS Attribute of the CDR field is a standard or vendor-specific attribute. ▪ [0] Standard = (Default) For standard RADIUS Attributes. ▪ [1] Vendor Specific = For vendor-specific RADIUS Attributes (VSA). **Note:** The parameter is applicable only for RADIUS accounting (i.e., 'CDR Type' parameter configured to **RADIUS Gateway**).
RADIUS Attribute ID `radius-id` [GWCDRFormat_RadiusID]	Defines an ID for the RADIUS Attribute. For vendor-specific Attributes, this represents the VSA ID; for standard attributes, this represents the Attribute ID (first byte of the Attribute). The valid value is 0 to 255 (one byte). The default is 0. **Notes:** ▪ The parameter is applicable only for RADIUS accounting (i.e., 'CDR Type' parameter configured to **RADIUS Gateway**). ▪ For VSA's (i.e., 'RADIUS Attribute Type' parameter configured to **Vendor Specific**), the parameter must be configured to any value other than 0.

Parameter	Description
	▪ For standard RADIUS Attributes (i.e., 'RADIUS Attribute Type' parameter configured to **Standard**), the value **must** be a "known" RADIUS ID (per RFC for RADIUS). However, if you configure the ID to 0 (default) for any of the RADIUS Attributes (configured in the 'Column Type' parameter) listed below and then apply your rule (click **Add**), the device automatically replaces the value with the RADIUS Attribute's ID according to the RFC: ✓ **Destination Number:** 30 ✓ **Source Number:** 31 ✓ **Accounting Status Type:** 40 ✓ **Local Input Octets:** 42 ✓ **Local Output Octets:** 43 ✓ **Call Duration:** 46 ✓ **Local Input Packets:** 47 ✓ **Local Output Packets:** 48 If you configure the value to 0 and the RADIUS Attribute is not any of the ones listed above, the configuration is invalid.

50.2.3 Customizing CDRs for SBC Calls

The SBC CDR Format table lets you customize SBC-related CDRs that are generated by the device for the following:

- CDRs (media and SIP signaling) sent in Syslog messages. For CDRs sent in Syslog messages, you can customize the name of the CDR field. The table lets you configure up to 128 Syslog CDR customization rules.
- CDRs related to RADIUS accounting and sent in RADIUS accounting request messages. For RADIUS accounting CDRs, you can customize the RADIUS Attribute's prefix name and RADIUS Attribute's ID, for standard RADIUS Attributes and vendor-specific RADIUS Attributes (VSA). For example, instead of the default VSA name, "h323-connect-time" with RADIUS Attribute ID 28, you can change the name to "Call-Connect-Time" with ID 29. The table lets you configure up to 40 RADIUS-accounting CDR customization rules. For more information on RADIUS accounting, see Configuring RADIUS Accounting on page 760.
- CDRs stored locally on the device. For local storage of CDRs, you can customize the name of the CDR field. The table lets you configure up to 64 locally-stored CDR customization rules. For more information on storing CDRs on the device, see Storing CDRs on the Device on page 758.

If you do not configure a CDR customization rule for a specific CDR, the device generates the CDR in a predefined default CDR format (see CDR Field Description on page 737).

Notes:

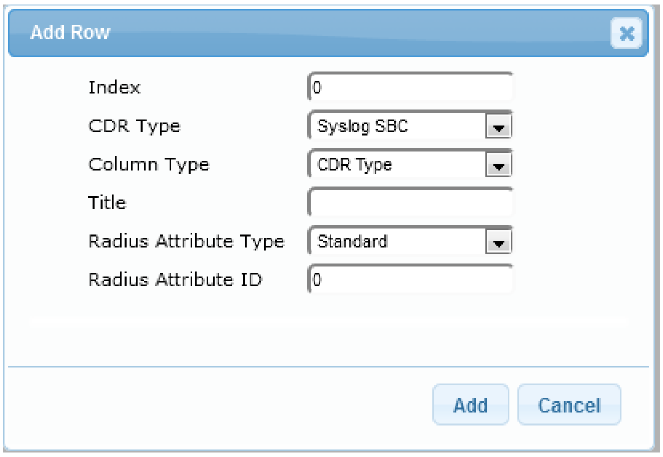
- The following standard RADIUS Attributes cannot be customized: 1 through 6, 18 through 20, 22, 23, 27 through 29, 32, 34 through 39, 41, 44, 52, 53, 55, 60 through 85, 88, 90, and 91.
- If the RTCP XR feature is unavailable (not licensed or disabled), the R-factor VoIP metrics are not provided in CDRs (CDR fields, Local R Factor and Remote R Factor) generated by the device. Instead, these CDR fields are sent with the value 127, meaning that information is unavailable.

The following procedure describes how to customize SBC-related CDRs through the Web interface. You can also configure it through ini file (SBCCDRFormat) or CLI (configure voip > services cdr > cdr-format sbc-cdr-format).

➢ To customize SBC-related CDRs:

1. Open the SBC CDR Format table (**Configuration** tab > **System** menu > **Call Detail Record** > **SBC CDR Format**).
2. Click **Add**; the following dialog box appears:

Figure 50-4: SBC CDR Format Table - Add Row Dialog Box

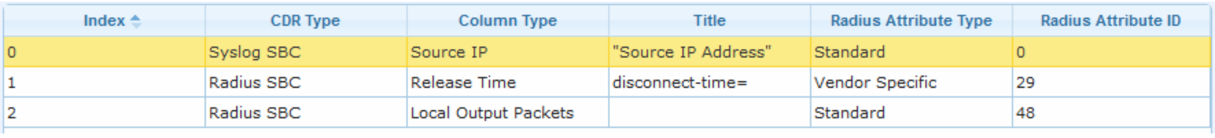


Index	0
CDR Type	Syslog SBC
Column Type	CDR Type
Title	
Radius Attribute Type	Standard
Radius Attribute ID	0

3. Configure the CDR according to the parameters described in the table below.
4. Click **Add**.

An example of CDR customization rules configured in the table is shown below:

Figure 50-5: Example of CDR Customization Rules for SBC Calls



Index	CDR Type	Column Type	Title	Radius Attribute Type	Radius Attribute ID
0	Syslog SBC	Source IP	"Source IP Address"	Standard	0
1	Radius SBC	Release Time	disconnect-time=	Vendor Specific	29
2	Radius SBC	Local Output Packets		Standard	48

SBC CDR Format Table Parameter Descriptions

Parameter	Description
Index [SBCCDRFormat_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
CDR Type `cdr-type` [SBCCDRFormat_CDRType]	Defines the application type for which you want to customize CDRs. ▪ [1] Syslog SBC = (Default) Customizes CDR fields for SIP signaling-related CDRs sent in Syslog messages. ▪ [3] Syslog Media = Customizes CDR fields for media-related CDRs sent in Syslog messages. ▪ [5] History SBC = Customizes CDR fields that are stored locally on the device. ▪ [7] RADIUS SBC = Customizes CDR fields (i.e., RADIUS Attributes) for CDRs sent in RADIUS accounting request messages.
Column Type `col-type` [SBCCDRFormat_ColumnType]	Defines the CDR field (column) that you want to customize. The applicable CDR field depends on the settings of the 'CDR Type' parameter: ▪ **For all types:** [0] CDR Type (**default**); [1] Call ID; [2] Session ID; [3] Report Type; [4] Media Type; [5] Accounting Status Type; [6] H323 ID; [7] Radius Call ID; [8] Blank.

Parameter	Description
	- **Syslog SBC,** History SBC, **and RADIUS SBC:** [10] Endpoint Type; [11] Call Orig; [12] Source IP; [13] Destination IP; [14] Remote IP; [15] Source Port; [16] Dest Port; [17] Remote Port; [18] Call Duration; [19] Termination Side; [20] Termination Reason; [21] Setup Time; [22] Connect Time; [23] Release Time; [24] Redirect Reason; [25] Was Call Started; [26] IP Group ID; [27] IP Group Name; [28] SRD ID; [29] SRD Name; [30] SIP Interface ID; [31] Transport Type; [32] Signaling IP DiffServ; [33] Termination Reason Category; [34] Proxy Set ID; [35] IP Profile ID; [36] IP Profile Name; [37] Media Realm ID; [38] Media Realm Name; [39] SIP Termination Reason; [40] SIP Termination Description; [41] Caller Display ID; [42] Callee Display ID; [43] SIPInterface Name; [44] Call Orig Radius; [45] Termination Side Radius; [46] Termination Side Yes No; [47] Termination Reason Value; [48] ProxySet Name. - **Syslog Media and RADIUS SBC:** [150] Channel ID; [151] Coder Type;[152] Packet Interval; [153] Payload Type; [154] Local Input Packets; [155] Local Output Packets; [156] Local Input Octets; [157] Local Output Octets; [158] Local Packet Loss; [159] Local Round Trip Delay; [160] Local Jitter; [161] Local SSRC Sender; [162] Remote Input Packets; [163] Remote Output Packets; [164] Remote Input Octets; [165] Remote Output Octets; [166] Remote Packet Loss; [167] Remote Round Trip Delay; [168] Remote Jitter; [169] Remote SSRC Sender; [170] Local RTP IP; [171] Local RTP Port; [172] Remote RTP IP; [173] Remote RTP Port; [174] RTP IP DiffServ; [175] Local R Factor; [176] Remote R Factor; [177] Local MOS CQ; [178] Remote MOS CQ; [179] AMD Decision; [180] AMD Decision Probability; [181] Latched RTP IP; [182]Latched RTP Port; [183] Latched T38 IP; [184] Latched T38 Port. - **Syslog SBC,** History SBC, **and RADIUS SBC:** [200] Source URI; [201] Destination URI; [202] Source URI Before Manipulation; [203] Destination URI Before Manipulation; [204] Redirect URI; [205] Redirect URI Before Manipulation; [206] SIP Method; [207] Direct Media; [208] Source Username; [209] Destination Username; [210] Source Username Before Manipulation; [211] Destination Username Before Manipulation; [212] Source Host; [213] Destination Host; [214] Source Host Before Manipulation; [215] Destination Host Before Manipulation.
Title `title` [SBCCDRFormat_Title]	Defines a new name for the CDR field (for Syslog or local storage) or for the RADIUS Attribute prefix name (for RADIUS accounting) that you selected in the 'Column Type' parameter. You can configure the name to be enclosed by apostrophes (single or double). For example, if you want the CDR field name to appear as 'Phone Duration', you must configure the parameter to 'Phone Duration'. You can also configure the CDR field name with an equals (=) sign, for example "call-connect-time=". **Notes:** - For RADIUS Attributes that do not require a prefix name, leave the parameter undefined. - The parameter's value is case-sensitive. For example, if you want the CDR field name to be Phone-Duration, you must configure the parameter to "Phone-Duration" (i.e., upper case "P" and "D").
RADIUS Attribute Type `radius-type`	Defines whether the RADIUS Attribute of the CDR field is a standard or vendor-specific attribute. - [0] Standard = (Default) For standard RADIUS Attributes.

Parameter	Description
[SBCCDRFormat_RadiusType]	▪ [1] Vendor Specific = For vendor-specific RADIUS Attributes (VSA). **Note:** The parameter is applicable only for RADIUS accounting (i.e., 'CDR Type' parameter configured to **RADIUS SBC**).
RADIUS Attribute ID `radius-id` [SBCCDRFormat_RadiusID]	Defines an ID for the RADIUS Attribute. For VSAs, this represents the VSA ID; for standard Attributes, this represents the Attribute ID (first byte of the Attribute). The valid value is 0 to 255 (one byte). The default is 0. **Notes:** ▪ The parameter is applicable only for RADIUS accounting (i.e., 'CDR Type' parameter configured to **RADIUS SBC**). ▪ For VSA's (i.e., 'RADIUS Attribute Type' parameter configured to **Vendor Specific**), the parameter must be configured to any value other than 0. ▪ For standard RADIUS Attributes (i.e., 'RADIUS Attribute Type' parameter configured to **Standard**), the value **must** be a "known" RADIUS ID (per RFC for RADIUS). However, if you configure the ID to 0 (default) for any of the RADIUS Attributes (configured in the 'Column Type' parameter) listed below and then apply your rule (click **Add**), the device automatically replaces the value with the RADIUS Attribute's ID according to the RFC: ✓ **Destination Username:** 30 ✓ **Source Username:** 31 ✓ **Accounting Status Type:** 40 ✓ **Local Input Octets:** 42 ✓ **Local Output Octets:** 43 ✓ **Call Duration:** 46 ✓ **Local Input Packets:** 47 ✓ **Local Output Packets:** 48 If you configure the value to 0 and the RADIUS Attribute is not any of the ones listed above, the configuration is invalid.

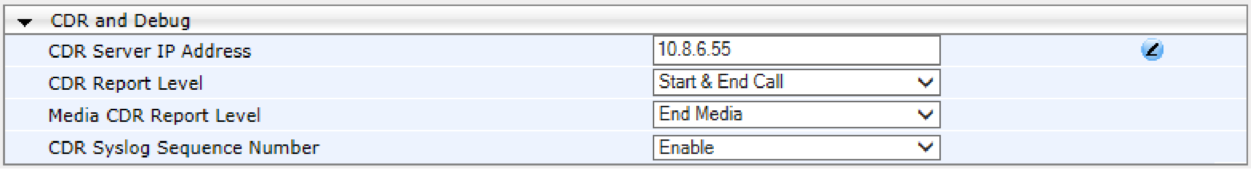
50.2.4 Configuring CDR Reporting

The following procedure describes how to configure CDR reporting.

➢ **To configure CDR reporting:**

1. Enable the Syslog feature for sending log messages generated by the device to a collecting log message server. For more information, see "Enabling Syslog" on page 783.
2. Open the Call Detail Record Settings page (**Configuration** tab > **System** menu > **Call Detail Record** > **Call Detail Record Settings**).

Figure 50-6: CDR Parameters in Call Detail Record Settings Page



CDR and Debug

CDR Server IP Address	10.8.6.55
CDR Report Level	Start & End Call
Media CDR Report Level	End Media
CDR Syslog Sequence Number	Enable

3. Configure the parameters as required. For a description of the parameters, see "Syslog, CDR and Debug Parameters" on page 841.
4. Click **Submit**.

Note:

- If you do not configure an IP address for a CDR server, the device sends CDRs to the Syslog server, as configured in 'Enabling Syslog' on page 783.
- The device sends CDRs only for dialog-initiating INVITE messages (call start), 200 OK responses (call connect) and BYE messages (call end). For SBC calls only: If you want to enable the generation of CDRs for non-call SIP dialogs (such as SUBSCRIBE, OPTIONS, and REGISTER), use the EnableNonCallCdr parameter.
- To configure the time zone string (e.g., GMT+1) that is displayed with the timestamp in CDRs ("Connect Time", "Release Time", and "Setup Time" CDR fields), use the TimeZoneFormat parameter.

50.2.5 Storing CDRs on the Device

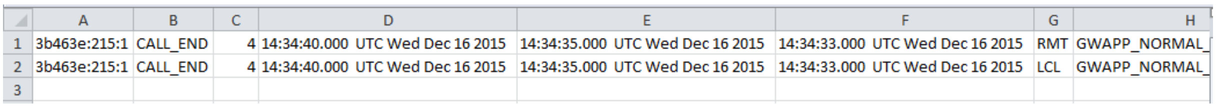
The CDRs of of Gateway and SBC calls generated by the device can also be stored locally on the device (RAM).

Notes When the device is reset or powered off, stored CDRs are deleted.

You can specify the calls (configuration entities) for which you wish to create and store CDRs locally. This is done using Logging Filter rules in the Logging Filters table. For example, you can configure a rule to create CDRs for traffic belonging only to IP Group 2 and store the CDRs locally.

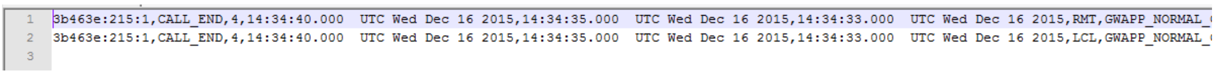
The CDRs are saved in a comma-separated values file (*.csv), where each CDR is shown on a dedicated row. An example of a CSV file with two CDRs are shown below:

- CSV file viewed in Excel:



	A	B	C	D	E	F	G	H
1	3b463e:215:1	CALL_END	4	14:34:40.000 UTC Wed Dec 16 2015	14:34:35.000 UTC Wed Dec 16 2015	14:34:33.000 UTC Wed Dec 16 2015	RMT	GWAPP_NORMAL_
2	3b463e:215:1	CALL_END	4	14:34:40.000 UTC Wed Dec 16 2015	14:34:35.000 UTC Wed Dec 16 2015	14:34:33.000 UTC Wed Dec 16 2015	LCL	GWAPP_NORMAL_
3								

- CSV file viewed in a text editor (Notepad):



```
3b463e:215:1,CALL_END,4,14:34:40.000  UTC Wed Dec 16 2015,14:34:35.000  UTC Wed Dec 16 2015,14:34:33.000  UTC Wed Dec 16 2015,RMT,GWAPP_NORMAL_
3b463e:215:1,CALL_END,4,14:34:40.000  UTC Wed Dec 16 2015,14:34:35.000  UTC Wed Dec 16 2015,14:34:33.000  UTC Wed Dec 16 2015,LCL,GWAPP_NORMAL_
```

To view the CDR column headers corresponding to the CDR data in the CSV file, run the following CLI command:

- SBC CDRs:

```
(config-system)# cdr
(cdr)# cdr-format show-title local-storage-sbc
session id,report type,call duration, call end time, call
connect time,call start time, call originator, termination
reason, call id, srce uri, dest uri
```

- Gateway CDRs:

```
(config-system)# cdr
(cdr)# cdr-format show-title local-storage-gw
```

Note: Each CDR can contain up to 1023 characters. If it contains more than this, the device removes the extra characters.

You can do the following with locally saved CDR files (*.csv), through the CLI (root menu):

- View stored CDR files:
  - View all stored CDR files:
    ```
    show storage-history
    ```
  - View all stored, unused CDR files:
    ```
    show storage-history unused
    ```
- Delete stored CDR files:
  - Delete all stored files:
    ```
    clear storage-history cdr-storage-history all
    ```
  - Delete all stored, unused CDR files:
    ```
    clear storage-history cdr-storage-history unused
    ```
- Save stored CDR files to an external destination:
  ```
  copy storage-history cdr-storage-history <filename> to
  <protocol://destination>
  ```
  Where:
  - *filename:* name you want to assign the file. Any file extension name can be used, but as the file content is in CSV format, it is recommended to use the .csv file extension.
  - *protocol:* protocol over which the file is sent (tftp, http, or https).

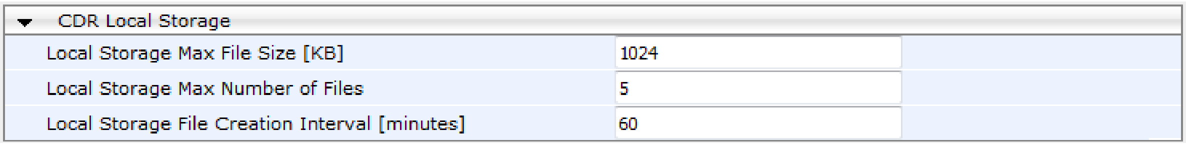
  For example:
  ```
  copy storage-history cdr-storage-history my_cdrs.csv to
  tftp://company.com/cdrs
  ```

The following procedure describes how to configure local CDR storage through the Web interface.

➢ To configure local CDR storage:

1. Open the Logging Settings page (**Configuration** tab > **System** menu > **Logging** > **Logging Settings**), and then scroll down to the Local Storage group:

Figure 50-7: CDR Local Storage on Logging Settings Page



▾ CDR Local Storage

Local Storage Max File Size [KB]	1024
Local Storage Max Number of Files	5
Local Storage File Creation Interval [minutes]	60

2. Configure the following parameters:
   - 'Local Storage Max File Size' (CDRLocalMaxFileSize): Enter the maximum size (in kilobytes) of the CDR file. Once the file size is reached, the device creates a new file for subsequent CDRs, and so on.
   - 'Local Storage Max Number Of Files' (CDRLocalMaxNomOfFiles): Enter the maximum number of CDR files. Once the maximum is reached, a subsequent CDR file replaces the oldest created file.
   - 'Local Storage File Creation Interval' (CDRLocalInterval): Enter the time in minutes for how often the device creates a new CDR file. For example, if configured to 60, it creates a new file every hour even if the maximum file size has not been reached.

   For a detailed description of each parameter, see Syslog, CDR and Debug Parameters on page 841.

3. Open the Logging Filters table (**Configuration** tab > **System** menu > **Logging** > **Logging Filters Table**), and then configure a log filtering rule with the following parameter settings:
   - 'Filter Type' and 'Value': (as desired)
   - 'Log Destination': Local Storage
   - 'Log Type': CDR Only
   - 'Mode': Enable

   For more information on the Logging Filters table, see Configuring Log Filter Rules on page 769.

Notes:

- If you have enabled the CDR storage feature and you later decide to change the maximum number of files (CDRLocalMaxNomOfFiles) to a lower value (e.g., from 50 to 10), the device stores the remaining files (e.g., 40) in its memory (i.e., unused files).
- For customizing CDR fields, see "Customizing CDR Fields for SBC Calls" on page 754.

50.3 Configuring RADIUS Accounting

The device can send accounting data of SIP calls as call detail records (CDR) to a RADIUS Accounting server. CDR-based accounting messages can be sent upon call release, call connection and release, or call setup and release. For a list of the CDR attributes for RADIUS accounting, see the table following the procedure below.

RADIUS CDR attributes have the following format:

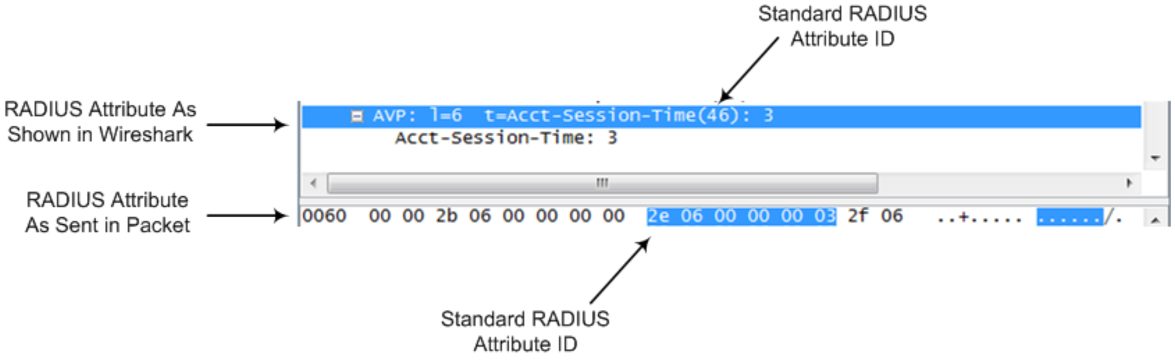
- **Standard RADIUS attributes (per RFC):** A typical standard RADIUS attribute is shown below. The RADIUS attribute ID depends on the attribute.

Figure 50-8: Typical Standard RADIUS Attribute

```
2e 06 00 00 00 03 --- Data
|  |
|  Length (including header)
RADIUS ID
```

The following figure shows a standard RADIUS attribute collected by Wireshark. The bottom pane shows the RADIUS attribute information as sent in the packet; the upper pane is Wireshark's interpretation of the RADIUS information in a more readable format. The example shows the attribute in numeric format (32-bit number in 4 bytes).

Figure 50-9: Example of Standard RADIUS Attribute Collected by Wireshark


Standard RADIUS Attribute ID
RADIUS Attribute As Shown in Wireshark
RADIUS Attribute As Sent in Packet
Standard RADIUS Attribute ID

```
⊟ AVP: l=6  t=Acct-Session-Time(46): 3
      Acct-Session-Time: 3

0060  00 00 2b 06 00 00 00 00  2e 06 00 00 00 03 2f 06   ..+..... ....../.
```

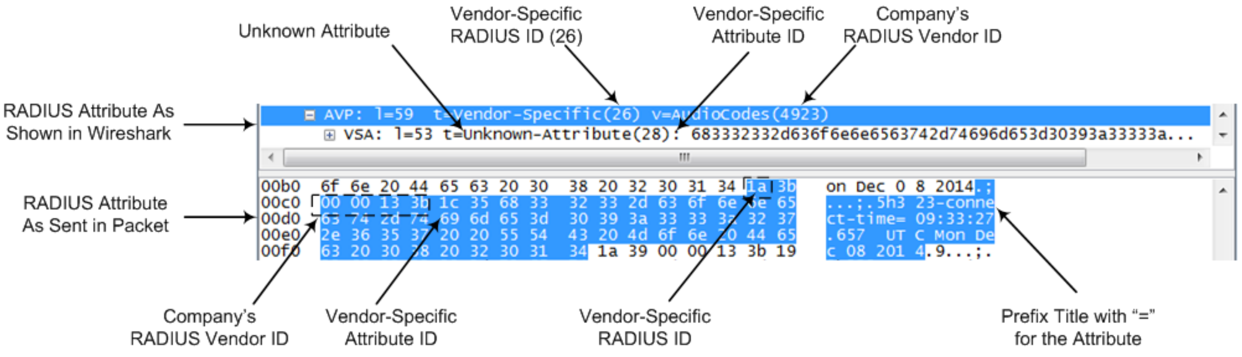
- **Vendor-specific RADIUS attributes:** RADIUS attributes that are specific to the device (company) are referred to as Vendor-specific attributes (VSA). The CDR of VSAs are sent with a general RADIUS ID of 26 to indicate that they are vendor-specific (non-standard). In addition, the company's registered vendor ID (as registered with the Internet Assigned Numbers Authority or IANA) is also included in the packet. The device's default vendor ID is 5003, which can be changed by using the RadiusVSAVendorID parameter. The VSA ID is also included in the packet.

Figure 50-10: Example of a Vendor-Specific Attribute

```
1a 13 00 00 13 8b 21 0d 68 33 32 33 2d 67 77 2d 69 64 3d --- Data
|  |  |  |  |  |  |  |
|  |  Vendor ID    |  Vendor part length
|  |  (5003)       Vendor-Specific Attribute (VSA) ID
|  Length (including header)
RADIUS ID indicating vendor-specific (26)
```

The following figure shows a vendor-specific RADIUS attribute collected by Wireshark. The bottom pane shows the RADIUS attribute information as sent in the packet; the upper pane is Wireshark's interpretation of the RADIUS information in a more readable format. The example shows the attribute in string-of-characters format.

Figure 50-11: Example of Vendor-Specific RADIUS Attribute Collected by Wireshark


Unknown Attribute
Vendor-Specific RADIUS ID (26)
Vendor-Specific Attribute ID
Company's RADIUS Vendor ID
RADIUS Attribute As Shown in Wireshark
RADIUS Attribute As Sent in Packet
Company's RADIUS Vendor ID
Vendor-Specific Attribute ID
Vendor-Specific RADIUS ID
Prefix Title with "=" for the Attribute

```
⊟ AVP: l=59  t=Vendor-Specific(26) v=AudioCodes(4923)
   ⊞ VSA: l=53 t=Unknown-Attribute(28): 683332332d636f6e6e6563742d74696d653d30393a33333a...

00b0  6f 6e 20 44 65 63 20 30  38 20 32 30 31 34 1a 3b   on Dec 0 8 2014.;
00c0  00 00 13 3b 1c 35 68 33  32 33 2d 63 6f 6e 6e 65   ...;.5h3 23-conne
00d0  63 74 2d 74 69 6d 65 3d  30 39 3a 33 33 3a 32 37   ct-time= 09:33:27
00e0  2e 36 35 37 20 20 55 54  43 20 4d 6f 6e 20 44 65   .657  UT C Mon De
00f0  63 20 30 38 20 32 30 31  34 1a 39 00 00 13 3b 19   c 08 201 4.9...;.
```

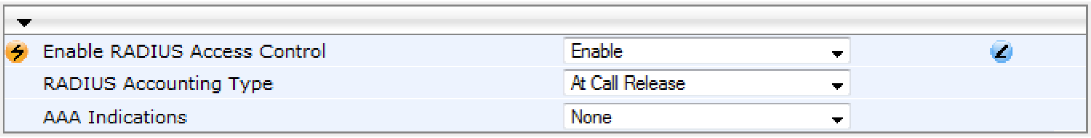
Notes:

- You can customize the prefix title of the RADIUS attribute name and the ID. For more information, see Customizing CDRs for Gateway Calls on page 751 and Customizing CDRs for SBC Calls on page 754.
- To configure the address of the RADIUS Accounting server, see "Configuring RADIUS Servers" on page 224.

➢ To configure RADIUS accounting:

1. Open the Call Detail Record Settings page (**Configuration** tab > **System** menu > **Call Detail Record** > **Call Detail Record Settings**).

Figure 50-12: Call Detail Record Settings Page



Enable RADIUS Access Control	Enable
RADIUS Accounting Type	At Call Release
AAA Indications	None

2. Set the 'Enable RADIUS Access Control' parameter to **Enable**.
3. Configure the remaining parameters as required. For a description of these parameters, see "RADIUS Parameters" on page 1024.
4. Click **Submit**.

5. For your settings to take effect, reset the device with a flash burn.

The table below lists the RADIUS Accounting CDR attributes included in the communication packets transmitted between the device and a RADIUS server.

Table 50-6: Supported RADIUS Accounting CDR Attributes

Attribute ID	Attribute Name	Vendor-Specific Attribute (VSA) ID	Description	Value Format	Example	AAA
Request Attributes						
1	user-name	(Standard)	Account number or calling party number or blank	String up to 15 digits long	5421385747	Start Acc Stop Acc
4	nas-ip-address	(Standard)	IP address of the requesting device	Numeric	192.168.14.43	Start Acc Stop Acc
6	service-type	(Standard)	Type of service requested	Numeric	1: login	Start Acc Stop Acc
26	h323-incoming-conf-id	1	SIP call identifier	Up to 32 octets	h323-incoming-conf-id=38393530	Start Acc Stop Acc
26	h323-remote-address	23	IP address of the remote gateway	Numeric	-	Stop Acc
26	h323-conf-id	24	H.323/SIP call identifier	Up to 32 octets		Start Acc Stop Acc
26	h323-setup-time	25	Setup time in NTP format 1	String	h323-setup-time=09:33:26.621 Mon Dec 2014	Start Acc Stop Acc
26	h323-call-origin	26	Originator of call: ▪ "answer": Call originated from the IP side (Gateway) or incoming leg (SBC) ▪ "originate": Call originated from the Tel side (Gateway) or outgoing leg (SBC)	String	h323-call-origin=answer	Start Acc Stop Acc

Attribute ID	Attribute Name	Vendor-Specific Attribute (VSA) ID	Description	Value Format	Example	AAA
26	h323-call-type	27	Protocol type or family used on this leg of the call	String	h323-call-type=VOIP	Start Acc Stop Acc
26	h323-connect-time	28	Connect time in NTP format	String	h323-connect-time=09:33:37.657 UTC Mon Dec 08 2015	Stop Acc
26	h323-disconnect-time	29	Disconnect time in NTP format	String	-	Stop Acc
26	h323-disconnect-cause	30	Disconnect cause code (Q.850)	Numeric	h323-disconnect-cause=16	Stop Acc
26	h323-gw-id	33	Name of the gateway	String	h323-gw-id=<SIP ID string>	Start Acc Stop Acc
26	sip-call-id	34	SIP Call ID	String	sip-call-id=abcde@ac.com	Start Acc Stop Acc
26	call-terminator	35	Terminator of the call: ▪ "yes": Call terminated by the Tel side (Gateway) or outgoing leg (SBC) ▪ "no": Call terminated by the IP side (Gateway) or incoming leg (SBC)	String	call-terminator=yes	Stop Acc
26	terminator	37	Terminator of the call: ▪ "answer": Call originated from the IP side (Gateway) or incoming leg (SBC) ▪ "originate": Call originated from the Tel side (Gateway) or	String	terminator=originate	Stop Acc

Attribute ID	Attribute Name	Vendor-Specific Attribute (VSA) ID	Description	Value Format	Example	AAA
			outgoing leg (SBC)			
30	called-station-id	(Standard)	Destination phone number (Gateway call) or Destination URI (SBC call)	String	8004567145	Start Acc
31	calling-station-id	(Standard)	Calling Party Number (ANI) (Gateway call) or Source URI (SBC call)	String	5135672127	Start Acc Stop Acc
40	acct-status-type	(Standard)	Account Request Type - start (1) or stop (2) **Note:** 'start' isn't supported on the Calling Card application.	Numeric	1	Start Acc Stop Acc
41	acct-delay-time	(Standard)	No. of seconds tried in sending a particular record	Numeric	5	Start Acc Stop Acc
42	acct-input-octets	(Standard)	Number of octets received for that call duration (for SBC calls, applicable only if media anchoring)	Numeric	-	Stop Acc
43	acct-output-octets	(Standard)	Number of octets sent for that call duration (for SBC calls, applicable only if media anchoring)	Numeric	-	Stop Acc
44	acct-session-id	(Standard)	A unique accounting identifier - match start & stop	String	34832	Start Acc Stop Acc
46	acct-session-time	(Standard)	For how many seconds the user received the service	Numeric	-	Stop Acc
47	acct-input-packets	(Standard)	Number of packets received during the call	Numeric	-	Stop Acc
48	acct-oputput-packets	(Standard)	Number of packets sent during the call	Numeric	-	Stop Acc

Attribute ID	Attribute Name	Vendor-Specific Attribute (VSA) ID	Description	Value Format	Example	AAA
61	nas-port-type	(Standard)	Physical port type of device on which the call is active	String	0: Asynchronous	Start Acc Stop Acc
Response Attributes						
26	h323-return-code	103	The reason for failing authentication (0 = ok, other number failed)	Numeric	0 Request accepted	Stop Acc
44	acct-session-id	(Standard)	A unique accounting identifier – match start & stop	String	-	Stop Acc

Below is an example of RADIUS Accounting, where non-standard parameters are preceded with brackets:

```
Accounting-Request (4)
user-name = 111
acct-session-id = 1
nas-ip-address = 212.179.22.213
nas-port-type = 0
acct-status-type = 2
acct-input-octets = 4841
acct-output-octets = 8800
acct-session-time = 1
acct-input-packets = 122
acct-output-packets = 220
called-station-id = 201
calling-station-id = 202
// Accounting non-standard parameters:
(4923 33) h323-gw-id =
(4923 23) h323-remote-address = 212.179.22.214
(4923 1) h323-ivr-out = h323-incoming-conf-id:02102944 600a1899
3fd61009 0e2f3cc5
(4923 30) h323-disconnect-cause = 22 (0x16)
(4923 27) h323-call-type = VOIP
(4923 26) h323-call-origin = Originate
(4923 24) h323-conf-id = 02102944 600a1899 3fd61009 0e2f3cc5
```

50.4 Querying Device Channel Resources using SIP OPTIONS

The device reports its maximum and available channel resources in SIP 200 OK responses upon receipt of SIP OPTIONS messages. The device sends this information in the SIP X-Resources header with the following parameters:

- **telchs:** Specifies the total telephone channels and the number of free (available) telephone channels.
- **mediachs:** Not applicable.

Below is an example of the X-Resources:

```
X-Resources: telchs= /;mediachs=0/0
```

In the example above, "telchs" specifies the number of available channels and the number of occupied channels (channels are occupied and channels are available).

Note: This feature is applicable only to the Gateway application.

Part X

Diagnostics

51 Syslog and Debug Recording

For debugging and troubleshooting, you can use the device's Syslog and/or Debug Recording capabilities:

- **Syslog:** Syslog is an event notification protocol that enables a device to send event notification messages across IP networks to event message collectors, also known as Syslog servers. The device contains an embedded Syslog client, which sends error reports / events that it generates to a remote Syslog server using the IP / UDP protocol. This information is a collection of error, warning, and system messages that records every internal operation of the device.
- **Debug Recording:** The device can send debug recording packets to a debug capturing server. When the debug recording is activated, the device duplicates all messages that are sent and/or received by it and then sends them to an external server defined by IP address. The debug recording can be done for different types of traffic such as RTP/RTCP, T.38, ISDN, and SIP. Debug recording is used for advanced debugging when you need to analyze internal messages and signals. Debug recording is also useful for recording network traffic in environments where hub or port mirroring is unavailable and for recording internal traffic between two endpoints on the same device.

Note: You can include Syslog messages in debug recording (see "Configuring Log Filter Rules" on page 769).

51.1 Configuring Log Filter Rules

The Logging Filters table lets you configure up to 60 rules for filtering debug recording (DR) packets, Syslog messages, and Call Detail Records (CDR). The log filter determines the calls for which you want to generate DR packets, Syslog messages or CDRs. For example, you can add a rule to generate Syslog messages only for calls belonging to IP Groups 2 and 4, or for calls belonging to all IP Groups except for IP Group 3. You can also configure log filters for generating CDRs only and saving them on the device (local storage). DR log filters can include signaling information such as SIP messages, Syslog messages, PSTN traces (ISDN), CDRs, media (RTP, RTCP, and T.38), and pulse-code modulation (PCM) of voice signals from and to the TDM.

If you don't configure any rules in the Logging Filters table and you have enabled DR, Syslog, and/or CDR generation (done by simply configuring an IP address for the relevant servers - see Note below), logs are generated for all calls. Thus, the benefit of log filtering is that it allows you to create logs per specific calls, eliminating the need for additional device resources (CPU consumption), otherwise required when logs are generated for all calls.

You can enable and disable configured Logging Filter rules. Enabling a rule activates the rule, whereby the device starts generating the DR packets, Syslog messages, or CDRs. Disabling a rule is useful, for example, if you no longer require the rule, but may need it in the future. Thus, instead of deleting the rule entirely, you can simply disable it.

Notes:

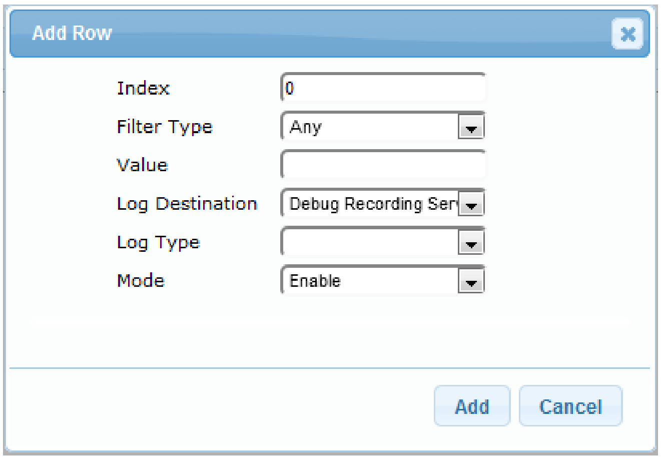
- If you want to configure a Logging Filter rule that logs Syslog messages to a Syslog server (i.e., not to a Debug Recording server), you must enable Syslog functionality, using the 'Enable Syslog' (EnableSyslog) parameter (see "Enabling Syslog" on page 783). Enabling Syslog functionality is not required for rules that include Syslog messages in the DR sent to a Debug Recording server.
- To configure the Syslog server's address, see "Configuring Address of Syslog Server" on page 782. To configure additional, global Syslog settings, see Configuring Syslog on page 774.
- To configure the Debug Recording server's address, see "Configuring Address of Debug Recording Server" on page 786.
- To configure additional, global CDR settings such as at what stage of the call the CDR is generated (e.g., start and end of call), see Configuring CDR Reporting on page 757.

The following procedure describes how to configure Logging Filter rules through the Web interface. You can also configure it through ini file (LoggingFilters) or CLI (configure system > logging > logging-filters).

➢ **To configure a logging filtering rule:**

1. Open the Logging Filters table (**Configuration** tab > **System** menu > **Logging** > **Logging Filters Table**).
2. Click **Add**; the following dialog box appears:

Figure 51-1: Logging Filters Table - Add Row Dialog Box



Add Row

Index: 0

Filter Type: Any

Value:

Log Destination: Debug Recording Ser

Log Type:

Mode: Enable

3. Configure a logging filter according to the parameters described in the table below.
4. Click **Add**.

Logging Filters Table Parameter Descriptions

Parameter	Description
Index [LoggingFilters_Index]	Defines an index number for the new table row. **Note:** Each row must be configured with a unique index.
Filter Type filter-type	Defines the filter type criteria. ▪ [1] Any (default)

Parameter	Description
[LoggingFilters_FilterType]	▪ [2] Trunk ID = Filters log according to Trunk ID. Note: Applicable only to the Gateway application. ▪ [3] Trunk Group ID = Filters log according to Trunk Group ID. For configuring Trunk Groups, see Configuring Trunk Groups on page 433. Note: Applicable only to the Gateway application. ▪ [4] Trunk & B-channel = Filters log according to Trunk and B-channel. Note: Applicable only to the Gateway application. ▪ [6] Tel-to-IP = Filters log according to a Tel-to-IP routing rule. For configuring Tel-to-IP routing rules, see Configuring Tel-to-IP Routing Rules on page 467. Note: Applicable only to the Gateway application. ▪ [7] IP-to-Tel = Filters log according to an IP-to-Tel routing rule. For configuring IP-to-Tel routing rules, see Configuring IP-to-Trunk Group Routing Rules on page 476. Note: Applicable only to the Gateway application. ▪ [8] IP Group = Filters log according to an IP Group. For configuring IP Groups, see "Configuring IP Groups" on page 340. ▪ [9] SRD = Filters log according to an SRD. For configuring SRDs, see Configuring SRDs on page 325. ▪ [10] Classification = Filters log according to a Classification rule. For configuring Classification rules, see Configuring Classification Rules on page 569. Note: Applicable only to the SBC application. ▪ [11] IP-to-IP Routing = Filters log according to an IP-to-IP routing rule. For configuring IP-to-IP routing rules, see Configuring SBC IP-to-IP Routing Rules on page 578. Note: Applicable only to the SBC application. ▪ [12] User = Filters log according to a user. The user is defined by username or username@hostname in the Request-URI of the SIP Request-Line. For example, "2222@10.33.45.201", which represents the following INVITE: `INVITE sip:2222@10.33.45.201;user=phone SIP/2.0` ▪ [13] IP Trace = Filters log according to an IP network trace, Wireshark-like expression. For more information on configuring IP traces, see "Filtering IP Network Traces" on page 773. ▪ [14] SIP Interface = Filters log according to SIP Interface. For configuring SIP Interfaces, see Configuring SIP Interfaces on page 333.
Value value [LoggingFilters_Value]	Defines the value for the selected filtering type in the 'Filter Type' parameter. The value can include the following: ▪ A single value. ▪ A range, using a hyphen "-" between the two values. For example, to specify IP Groups 1, 2 and 3, configure the parameter to "1-3" (without apostrophes). ▪ Multiple, non-contiguous values, using commas "," between each value. For example, to specify IP Groups 1, 3 and 9, configure the parameter to "1,3,9" (without apostrophes). ▪ Trunks pertaining to a module, using the syntax module number/port or port, for example: ✓ "1/2" (without apostrophes), means module 1, port 2 ✓ "1/[2-4]" (without apostrophes), means module 1, ports 2 through 4 ▪ The exclamation (**!**) wildcard character can be used for excluding a specific configuration entity from the filter. For example, to include all IP Groups in the filter except IP Group ID 2, configure the 'Filter Type' parameter to **IP Group** and the 'Value' parameter to "!2" (without apostrophes). Note that for SBC calls, a Logging Filter rule applies to the entire session, which is both legs (i.e., not per leg). For example, a

Parameter	Description
	call between IP Groups 1 and 2 are logged for both legs even if the 'Value' parameter is configured to "!2". ▪ **Any** to indicate all. **Notes:** ▪ You can use the index number or string name to specify the configuration entity for the following 'Filter Types': **Tel-to-IP**, **IP-to-Tel**, **IP Group**, **SRD**, **Classification**, **IP-to-IP Routing**, or **SIP Interface**. For example, to specify IP Group at Index 2 with the name "SIP Trunk", configure the parameter to either "2" or "SIP Trunk" (without apostrophes). ▪ For IP trace expressions, see "Filtering IP Network Traces" on page 773.
Log Destination log-dest [LoggingFilters_LogDestination]	Defines where the device sends the log file. ▪ [0] Syslog Server = The device generates Syslog messages based on the configured log filter and sends them to a user-defined Syslog server. The Syslog messages can contain one of the following types of information, depending on the settings of the 'Log Type' parameter (described later in the table): ✓ Not configured (default): The Syslog messages contain the regular syslog information. ✓ CDR Only: The Syslog messages contain only CDRs (no system information and alerts). ▪ [1] Debug Recording Server = (Default) The device generates DR packets based on the configured log filter and sends them to a user-defined Debug Recording server. ▪ [2] Local Storage = The device generates CDRs based on the configured log filter and stores them locally on the device. For more information on local CDR storage, see Storing CDRs on the Device on page 758. **Notes:** ▪ If the 'Filter Type' parameter is configured to **IP Trace**, you must configure the parameter to **Debug Recording Server**. ▪ If you configure the parameter to **Local Storage**, you must configure the 'Log Type' parameter to **CDR Only**. ▪ If you configure the parameter to **Syslog Server** and the debug level (GwDebugLevel) is configured to **No Debug** (see "Configuring Syslog Debug Level" on page 781), the Syslog messages include only system Warnings and Errors.
Log Type log-type [LoggingFilters_CaptureType]	Defines the type of messages to include in the log file. ▪ [0] = (Default) Not configured. The option is applicable only for sending Syslog messages to a Syslog server (i.e., 'Log Destination' parameter is configured to **Syslog Server**). ▪ [1] Signaling = The option is applicable only for DR (i.e., 'Log Destination' parameter is configured to **Debug Recording Server**). The DR includes signaling information such as SIP signaling messages, Syslog messages, CDRs, and the device's internal processing messages. ▪ [2] Signaling & Media = The option is applicable only for DR (i.e., 'Log Destination' parameter is configured to **Debug Recording Server**). The DR includes signaling, Syslog messages, and media (RTP/RTCP/T.38). ▪ [3] Signaling & Media & PCM = The option is applicable only for DR (i.e., 'Log Destination' parameter is configured to D**ebug Recording**

Parameter	Description
	Server). The DR includes signaling, Syslog messages, media, and PCM (voice signals from and to TDM). ▪ [4] PSTN Trace = The option is applicable only for DR (i.e., 'Log Destination' parameter is configured to **Debug Recording Server**) and if the 'Filter Type' parameter is configured to **Trunk ID**. The DR includes ISDN traces. ▪ [5] CDR Only = Only CDRs are generated. The option is applicable only if the 'Log Destination' parameter is configured to **Syslog Server** or **Local Storage**. When configured to **Syslog Server**, only CDRs are included in the Syslog messages (excluding all system logs and alerts) sent to the Syslog server. **Notes:** ▪ If you configure the 'Log Destination' parameter to **Local Storage**, the 'Log Type' parameter must be configured to **CDR Only**. ▪ The parameter is not applicable when the 'Filter Type' parameter is configured to **IP Trace**. ▪ To include Syslog messages in DR, it is unnecessary to enable Syslog functionality**..**
Mode `mode` [LoggingFilters_Mode]	Enables and disables the rule. ▪ [0] Disable ▪ [1] Enable (default)

51.1.1 Filtering IP Network Traces

You can filter Syslog and debug recording messages for IP network traces, by setting the 'Filter Type' parameter to **IP Trace** in the Logging Filters table. IP traces are used to record any IP stream, according to destination and/or source IP address, or port and Layer-4 protocol (UDP, TCP or any other IP type as defined by http://www.iana.com). Network traces are typically used to record HTTP.

When the **IP Trace** option is selected, only the ‘Value’ parameter is applicable; the 'Syslog' and 'Capture Type' parameters are not relevant. The 'Value' parameter configures the Wireshark-like filtering expressions for your IP trace. The following Wireshark-like expressions are supported:

Table 51-1: Supported Wireshark-like Expressions for 'Value' Parameter

Expression	Description
ip.src, ip.dst	Source and destination IP address
ip.addr	IP address - up to two IP addresses can be entered
ip.proto	IP protocol type (PDU) entered as an enumeration value (e.g., 1 is ICMP, 6 is TCP, 17 is UDP)
udp, tcp, icmp, sip, ldap, http, https	Single expressions for protocol type
udp.port, tcp.port	Transport layer
udp.srcport, tcp.srcport	Transport layer for source port
udp.dstport, tcp.dstport	Transport layer for destination port
and, &&, ==, <, >	Between expressions

Below are examples of configured expressions for the 'Value' parameter:

- udp && ip.addr==10.8.6.55
- ip.src==10.8.6.55 && udp.port>=5000 and udp.port<6000
- ip.dst==10.8.0.1/16
- ip.addr==10.8.6.40

For conditions requiring the "or" / "||" expression, add multiple table rows. For example, the Wireshark condition "(ip.src == 1.1.1.1 or ip.src == 2.2.2.2) and ip.dst == 3.3.3.3" can be configured using the following two table row entries:

1. ip.src == 1.1.1.1 and ip.dst == 3.3.3.3
2. ip.src == 2.2.2.2 and ip.dst == 3.3.3.3

Notes:

- If the 'Value' field is undefined, the device records all IP traffic types.
- You cannot use ip.addr or udp/tcp.port together with ip.src/dst or udp/tcp.srcport/dstport. For example, "ip.addr==1.1.1.1 and ip.src==2.2.2.2" is an invalid configuration value.

51.2 Configuring Syslog

This section describes the Syslog message format, how to configure and enable Syslog, and how to view the generated Syslog messages. For filtering Syslog messages for specific calls, see "Configuring Log Filter Rules" on page 769.

51.2.1 Syslog Message Format

The Syslog message is sent from the device to a Syslog server as an ASCII (American Standard Code for Information Interchange) message. Syslog uses UDP as its underlying transport layer mechanism. By default, UDP port 514 is assigned to Syslog, but this can be changed (see "Enabling Syslog" on page 783).

Syslog includes two types of log messages:

- SIP call session logs: Logs relating to call sessions (e.g., call established). These logs are identified by a session ID ("SID"), described in detail in the table below. The following is an example of a SIP-session related Syslog message:

  ```
  13:10:57.811 : 10.13.4.12 : NOTICE  : [S=235][SID:2ed1c8:96:5]
  (lgr_flow)(63)  UdpTransportObject#0- Adding socket event for
  address 10.33.2.42:5060 [Time: 04-19-2012@18:29:39]
  ```

- Board logs: Logs relating to the operation of the device (infrastructure) that are non-call session related (e.g., device reset or Web login). These logs are identified by a board ID ("BID"), described in detail in the table below. The following is an example of a board Syslog message:

  ```
  10:21:28.037 : 10.15.7.95 : NOTICE  : [S=872] [BID=3aad56:32]
  Activity Log: WEB: Successful login at 10.15.7.95:80. User:
  Admin. Session: HTTP (10.13.22.54)
  ```

The format of the Syslog message is described in the following table below:

Table 51-2: Syslog Message Format Description

Message Item	Description
Message Types	Syslog generates the following types of messages:

Message Item	Description
	- ERROR: Indicates that a problem has been identified that requires immediate handling. - WARNING: Indicates an error that might occur if measures are not taken to prevent it. - NOTICE: Indicates that an unusual event has occurred. - INFO: Indicates an operational message. - DEBUG: Messages used for debugging. **Notes:** - The INFO and DEBUG messages are required only for advanced debugging and by default, they are not sent by the device. - When viewing Syslog messages in the Web interface, these message types are color coded.
Message Sequence Number [S=<number>]	By default, Syslog messages are sequentially numbered in the format [S=<number>], for example, "[S=643]". A skip in the number sequence of messages indicates a loss of message packets. For example, in the below Syslog, messages 238 through 300 were not received. In other words, 63 Syslog messages were lost (the sequential numbers are indicated below in bold font): 18:38:14. 52 : 10.33.45.72 : NOTICE: [**S=235**][SID:1034099026] (lgr_psbrdex)(619) recv <-- DIGIT(0) Ch:0 OnTime:0 InterTime:100 Direction:0 System:1 [File: Line:-1] 18:38:14. 83 : 10.33.45.72 : NOTICE: [**S=236**][SID:2ed1c8:96:5] (lgr_flow)(620) #0:DIGIT_EV [File: Line:-1] 18:38:14. 83 : 10.33.45.72 : NOTICE: [**S=237**][SID:2ed1c8:96:5] (lgr_flow)(621) | #0:DIGIT_EV [File: Line:-1] 18:38:14.958 : 10.33.45.72 : NOTICE: [**S=301**][SID:2ed1c8:96:5] (lgr_flow)(625) | #0:DIGIT_EV [File: Line:-1] You can disable the inclusion of the message sequence number in Syslog messages, by setting the 'CDR Syslog Sequence Number' parameter to **Disable** (see "Configuring Syslog" on page 783).
Log Number (lgr)(number)	Ignore this number; it has been replaced by the Message Sequence Number (described previously).
Session ID (SID)	Unique SIP call session and device identifier. The device identifier facilitates debugging by clearly identifying the specific device that sent the log message, especially useful in deployments consisting of multiple devices. In addition, the benefit of unique numbering is that it enables you to filter the information (such as SIP, Syslog, and media) according to device or session ID. The syntax of the session and device identifiers are as follows: [SID=<last 6 characters of device's MAC address>:<number of times device has reset>:<unique SID counter indicating the call session; increments consecutively for each new session; resets to 1 after a device reset>]

Message Item	Description
	For example: `14:32:52.028: 10.33.8.70: NOTICE: [S=9369] [SID=2ed1c8:96:5] (lgr_psbrdex)(274) recv <-- OFF_HOOK Ch:4` Where: ▪ *2ed1c8* is the device's MAC address. ▪ *96* is the number of times the device has reset. ▪ *5* is a unique SID session number (in other words, this is the fifth call session since the last device reset). ✓ Gateway application: A call session is considered either as a Tel-to-IP leg or an IP-to-Tel leg, where each leg is assigned a unique session number. ✓ SBC application: A session includes both the outgoing and incoming legs, where both legs share the same session number. ✓ Forked legs and alternative legs share the same session number. **Note:** You can configure the device to maintain the same SID value for calls traversing multiple AudioCodes' devices. For more information, see "Maintaining Same Syslog SID/BID over Multiple Devices" on page 778.
Board ID (BID)	Unique non-SIP session related (e.g., device reset or a Trunk alarm) and device identifier. The device identifier facilitates debugging by clearly identifying the specific device that sent the log message, especially useful in deployments consisting of multiple devices. In addition, the benefit of unique numbering is that it enables you to filter the information according to device. The syntax of the BID is as follows: [BID=<last 6 characters in MAC>:<number of times device has reset>] For example: `14:32:52.062: 10.33.8.70: WARNING: [S=9399] [BID=2ed1c8:96] invalid Physical index` Where: ▪ *2ed1c8* is the device's MAC address. ▪ *96* is the number of times the device has reset. ▪ **Note:** You can configure the device to maintain the same BID value for calls traversing multiple AudioCodes' devices. For more information, see "Maintaining Same Syslog SID over Multiple Devices" on page 778.
Message Body	Describes the message.
Timestamp	When the Network Time Protocol (NTP) is enabled, a timestamp string [hour:minutes:seconds] is added to all Syslog messages.

51.2.1.1 Event Representation in Syslog Messages

The Syslog message events that the device sends are denoted by unique abbreviations. The following example shows an abbreviated event in a Syslog message indicating packet loss (PL):

```
Apr  4 12:00:12 172.30.1.14 PL:5  [Code:3a002] [CID:3294] [Time:
20:17:00]
```

The table below lists these unique event abbreviations:

Table 51-3: Syslog Error Name Descriptions

Error Abbreviation	Error Name Description
AA	Invalid Accumulated Packets Counter
AC	Invalid Channel ID
AL	Invalid Header Length
AO	Invalid Codec Type
AP	Unknown Aggregation Payload Type
AR	Invalid Routing Flag Received
AT	Simple Aggregation Packets Lost
CC	Command Checksum Error
CE	Invalid Cell Coder Code
CS	Command Sequence Error
ES	8 sec Timeout Before Disconnect
HO	Host Received Overrun
IA	Invalid AMR Payload
IC	Invalid CID Error
IG	Invalid G723 Code
IP	Invalid payload length
IR	Invalid RTCP Packet
IS	Invalid SID Length
LC	Transmitter Received Illegal Command
LF	Lost Fax Frames In High Speed Mode
LM	Lost Modem Frames In High Speed Mode
MI	Misalignment Error
MR	Modem Relay Is Not Supported
OR	DSP JB Overrun
PD	RTP Packet Duplicated
PH	Packet Header Error
PL	RTP Packet Loss
RB	Counts the number of BFI Frames Received From The Host
RD	No Available Release Descriptor

Error Abbreviation	Error Name Description
RO	RTP Reorder
RP	Unknown RTP Payload Type
RS	RTP SSRC Error
UF	Unrecognized Fax Relay Command

51.2.1.2 Identifying AudioCodes Syslog Messages using Facility Levels

The device's Syslog messages can easily be identified and distinguished from Syslog messages from other equipment, by setting its Facility level. The Facility levels of the device's Syslog messages are numerically coded with decimal values. Facility level may use any of the "local use" facilities (0 through 7), according to RFC 3164. Implementing Facility levels is useful, for example, if you collect the device's as well as other equipments' Syslog messages on the same server. Therefore, in addition to filtering Syslog messages according to IP address, the messages can be filtered according to Facility level.

The Facility level is configured using the SyslogFacility ini file parameter, which provides the following options:

Table 51-4: Syslog Facility Levels

Numerical Value	Facility Level
16 (default)	local use 0 (local0)
17	local use 1 (local1)
18	local use 2 (local2)
19	local use 3 (local3)
20	local use 4 (local4)
21	local use 5 (local5)
22	local use 6 (local6)
23	local use 7 (local7)

Syslog messages begin with a less-than ("**<**") character, followed by a number, which is followed by a greater-than ("**>**") character. This is optionally followed by a single ASCII space. The number is known as the *Priority* and represents both the Facility level and the Severity level. A Syslog message with Facility level 16 is shown below:

```
Facility: LOCAL0 - reserved for local use (16)
```

51.2.1.3 Syslog Fields for Answering Machine Detection (AMD)

The Syslog message can include information relating to the Answering Machine Detection (AMD) feature. AMD is used to detect whether a human (including a fax machine), an answering machine, silence, or answering machine beeps have answered the call on the remote side.

- AMDSignal – this field can acquire one of the following values:
  - voice (V)
  - answer machine (A)
  - silence (S)
  - unknown (U)
- AMDDecisionProbability – probability (in %) success that correctly detects answering type

Below is an example of such a Syslog message with AMD information:

```
CallMachine:EVENT_DETECTED_EV - AMDSignal = <type – V/A/S/U>,
AMDDecisionProbability = <percentage> %
```

If there is no AMD detection, the AMDSignal field is shown empty (i.e. AMDSignal =).

For more information on the AMD feature, see "Answering Machine Detection (AMD)" on page 198.

51.2.1.4 SNMP Alarms in Syslog Messages

SNMP alerts are sent to the Syslog server using the following formats:

- **Raised Alarms:** RAISE-ALARM: <Alarm Name>; Textual Description: <Textual Description>; Severity <Alarm Severity>; Source <Alarm Source>; Unique ID: <Alarm Unique ID >.

  If additional information exists in the alarm, then these are also added: Additional Info1:/ Additional Info2:/ Additional Info3

  The Messages' Severity is as follows:

Table 51-5: Syslog Message Severity

ITU Perceived Severity (SNMP Alarm's Severity)	AudioCodes' Syslog Severity
Critical	RecoverableMsg
Major	RecoverableMsg
Minor	RecoverableMsg
Warning	Notice
Indeterminate	Notice
Cleared	Notice

- **Cleared Alarms:** CLEAR-ALARM: <Alarm Name>; Textual Description: <Textual Description>; Severity <Alarm Severity>; Source <Alarm Source>; Unique ID: <Alarm Unique ID >; If exists Additional Info1:/ Additional Info2:/ Additional Info3:

51.2.2 Configuring Web User Activities to Report to Syslog

The device can report operations (activities) performed in the Web interface by management users, by including them in Syslog messages. The Syslog message indicates these logs with the string, "Activity Log". Each logged user activity includes the following information:

- Username (e.g., "Admin") of the user that performed the action
- IP address of the client PC from where the Web user accessed the management interface
- Protocol used for the session (e.g., SSH or HTTP)

The following example shows a Web-user activity log (indicating a login action) with the above-mentioned information:

```
14:07:46.300 : 10.15.7.95 : Local 0   :NOTICE  : [S=3149]
[BID=3aad56:32]  Activity Log: WEB: Successful login at
10.15.7.95:80. User: Admin. Session: HTTP (10.13.22.54)
```

The device can report the following Web user activities:

- Modifications of individual parameters, for example:

  ```
  14:33:00.162 : 10.15.7.95 : Local 0   :NOTICE  : [S=3403]
  [BID=3aad56:32]  Activity Log: Max Login Attempts was changed
  from '3' to '2'. User: Admin. Session: HTTP (10.13.22.54)
  ```

- Modifications of table fields, and addition and deletion of table rows, for example:

  ```
  14:42:48.334 : 10.15.7.95 : NOTICE  : [S=3546] [BID=3aad56:32]
  Activity Log: Classification - remove line 2. User: Admin.
  Session: HTTP (10.13.22.54)
  ```

- Entered CLI commands (modifications of security-sensitive commands are logged without the entered value).
- Configuration file load (reported without per-parameter notifications).
- Auxiliary file load and software update.
- Device reset and burn to flash memory.
- Access to unauthorized Web pages according to the Web user's access level.
- Modifications of "sensitive" parameters.
- Login and logout.
- Actions that are not related to parameter changes (for example, file uploads, file delete, lock-unlock maintenance actions, LDAP clear cache, register-unregister, and start-stop trunk. In the Web, these actions are typically done by clicking a button (e.g., the LOCK button).

For more information on each of the above listed options, see "Syslog, CDR and Debug Parameters" on page 841.

You can also configure the device to send an SNMP trap each time a user performs an activity. To enable trap notification, use the parameter, EnableActivityTrap (see "Configuring SNMP Community Strings" on page 91).

Notes:

- You can also view logged user activities in the Web interface (see "Viewing Web User Activity Logs" on page 785).
- Logging of CLI commands can only be configured through CLI or ini file.

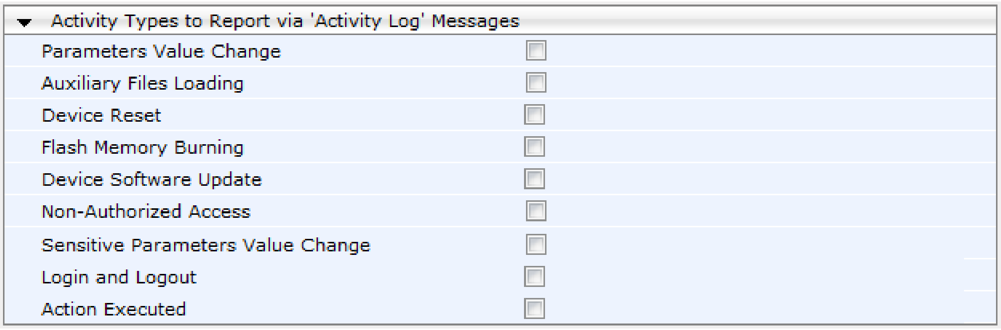
The following procedure describes how to configure Web user activity logging through the Web interface. You can also configure it through ini file (ActivityListToLog) or CLI (config-system > logging > activity-log).

➢ To configure Web user activities to report to Syslog server:

1. Open the Syslog Settings page (**Configuration** tab > **System** menu > **Syslog Settings**).
2. Under the Activity Types to Report via Activity Log Messages group, select the Web

actions to report to the Syslog server.

Figure 51-2: Web Activities to Report to Syslog



Activity Types to Report via 'Activity Log' Messages

Parameters Value Change	□
Auxiliary Files Loading	□
Device Reset	□
Flash Memory Burning	□
Device Software Update	□
Non-Authorized Access	□
Sensitive Parameters Value Change	□
Login and Logout	□
Action Executed	□

3. Click **Submit**.

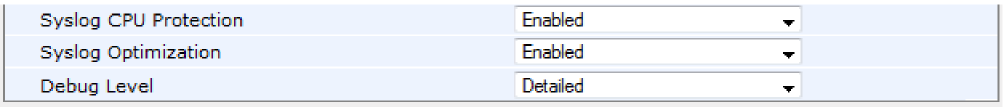
51.2.3 Configuring Syslog Debug Level

You can configure the amount of information (debug level) to include in Syslog messages. In addition, you can enable the device to send multiple Syslog messages bundled into a single packet as well as enable a protection mechanism that automatically lowers the debug level when the device's CPU resources become low, ensuring sufficient CPU resources are available for processing voice traffic.

➢ **To configure the Syslog debug level:**

1. Open the Syslog Settings page (**Configuration** tab > **System** menu > **Syslog Settings**).

Figure 51-3: Configuring Syslog Debug Level



Syslog CPU Protection	Enabled
Syslog Optimization	Enabled
Debug Level	Detailed

2. From the 'Debug Level' (GwDebugLevel) drop-down list, select the desired debug level of the Syslog messages:
   - **No Debug:** Disables Syslog and no Syslog messages are sent.
   - **Basic:** Sends debug logs of incoming and outgoing SIP messages.
   - **Detailed:** Sends debug logs of incoming and outgoing SIP message as well as many other logged processes.
3. From the 'Syslog Optimization' (SyslogOptimization) drop-down list, select whether you want the device to accumulate and bundle multiple debug messages into a single UDP packet before sending it to a Syslog server. The benefit of this feature is that it reduces the number of UDP Syslog packets, thereby improving (optimizing) CPU utilization. The size of the bundled message is configured by the MaxBundleSyslogLength parameter.
4. From the 'Syslog CPU Protection' (SyslogCpuProtection) drop-down list, select whether you want to enable the protection feature for the device's CPU resources during debug reporting, ensuring voice traffic is unaffected. If CPU resources drop (i.e., high CPU usage) to a critical level (user-defined threshold), the device automatically lowers the debug level to free up CPU resources that were required for the previous debug-level functionality. When CPU resources become available again, the device increases the debug level to its' previous setting. For example, if you set the 'Debug Level' to **Detailed** and CPU resources decrease to the defined threshold, the device automatically changes the level to **Basic**, and if that is not enough, it changes the level to **No Debug**. Once CPU resources are returned to normal, the device automatically changes the debug level back to its' original setting (i.e., **Detailed**). The threshold is configured by the DebugLevelHighThreshold parameter.

5. Click **Submit**.

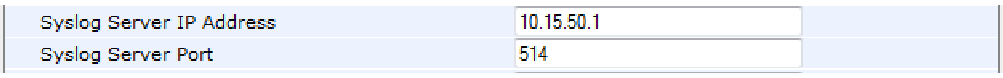
51.2.4 Configuring Address of Syslog Server

The following procedure describes how to configure the Syslog server's address to where the device sends the Syslog messages.

➢ **To configure the address of the Syslog server:**

1. Open the Syslog Settings page (**Configuration** tab > **System** menu > **Syslog Settings**).

Figure 51-4: Configuring the Syslog Address



Syslog Server IP Address	10.15.50.1
Syslog Server Port	514

2. In the 'Syslog Server IP Address' field, define the IP address of the Syslog server.
3. In the 'Syslog Server Port' field, define the port of the Syslog server.
4. Click **Submit**.

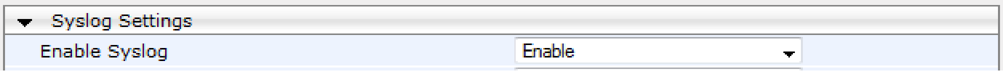
51.2.5 Enabling Syslog

The following procedure describes how to enable Syslog.

➢ **To enable Syslog:**

1. Open the Syslog Settings page (**Configuration** tab > **System** menu > **Syslog Settings**).

Figure 51-5: Syslog Settings Page



▾ Syslog Settings

Enable Syslog: Enable ▾

2. From the 'Enable Syslog' drop-down list, select **Enable**.
3. Click **Submit**.

51.2.6 Viewing Syslog Messages

You can receive and view Syslog messages generated by the device using any of the following Syslog server types:

- **Wireshark -** third-party network protocol analyzer (http://www.wireshark.org).

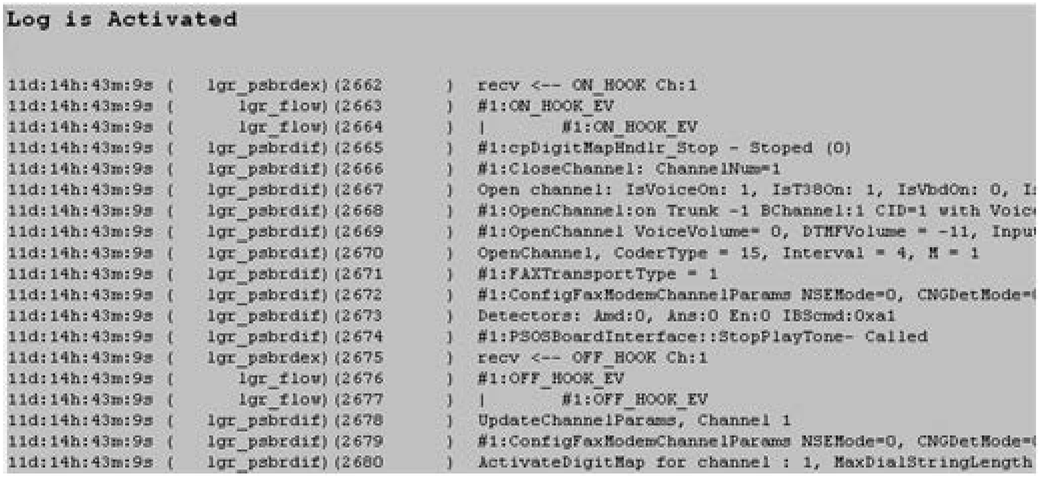
> **Note:** When debug recording is enabled and Syslog messages are also included in the debug recording, to view Syslog messages using Wireshark, you must install AudioCodes' Wireshark plug-in (acsyslog.dll). Once the plug-in is installed, the Syslog messages are decoded as "AC SYSLOG" and displayed using the "acsyslog" filter (instead of the regular "syslog" filter). For more information on debug recording, see "Debug Recording" on page 786.

- **Third-party, Syslog Server:** Any third-party Syslog server program that enables filtering of messages according to parameters such as priority, IP sender address, time, and date.
- Device's CLI Console: The device sends the error messages (e.g. Syslog messages) to the CLI console as well as to the configured destination. Use the following commands:

```
debug log              ; Starts the debug
no debug log           ; Stops the debug
no debug log all       ; Stops all debug process
```

- **Device's Web Interface:** The device provides an embedded Syslog server, which is accessed through the Web interface (**Status & Diagnostics** tab > **System Status** menu > **Message Log**). This provides limited Syslog server functionality.

Figure 51-6: Message Log Page



```
Log is Activated

11d:14h:43m:9s (   lgr_psbrdex)(2662      )  recv <-- ON_HOOK Ch:1
11d:14h:43m:9s (      lgr_flow)(2663      )  #1:ON_HOOK_EV
11d:14h:43m:9s (      lgr_flow)(2664      )  |        #1:ON_HOOK_EV
11d:14h:43m:9s (   lgr_psbrdif)(2665      )  #1:cpDigitMapHndlr_Stop - Stoped (0)
11d:14h:43m:9s (   lgr_psbrdif)(2666      )  #1:CloseChannel: ChannelNum=1
11d:14h:43m:9s (   lgr_psbrdif)(2667      )  Open channel: IsVoiceOn: 1, IsT38On: 1, IsVbdOn: 0, I
11d:14h:43m:9s (   lgr_psbrdif)(2668      )  #1:OpenChannel:on Trunk -1 BChannel:1 CID=1 with Voic
11d:14h:43m:9s (   lgr_psbrdif)(2669      )  #1:OpenChannel VoiceVolume= 0, DTMFVolume = -11, Inpu
11d:14h:43m:9s (   lgr_psbrdif)(2670      )  OpenChannel, CoderType = 15, Interval = 4, M = 1
11d:14h:43m:9s (   lgr_psbrdif)(2671      )  #1:FAXTransportType = 1
11d:14h:43m:9s (   lgr_psbrdif)(2672      )  #1:ConfigFaxModemChannelParams NSEMode=0, CNGDetMode=
11d:14h:43m:9s (   lgr_psbrdif)(2673      )  Detectors: Amd:0, Ans:0 En:0 IBScmd:0xa1
11d:14h:43m:9s (   lgr_psbrdif)(2674      )  #1:PSOSBoardInterface::StopPlayTone- Called
11d:14h:43m:9s (   lgr_psbrdex)(2675      )  recv <-- OFF_HOOK Ch:1
11d:14h:43m:9s (      lgr_flow)(2676      )  #1:OFF_HOOK_EV
11d:14h:43m:9s (      lgr_flow)(2677      )  |        #1:OFF_HOOK_EV
11d:14h:43m:9s (   lgr_psbrdif)(2678      )  UpdateChannelParams, Channel 1
11d:14h:43m:9s (   lgr_psbrdif)(2679      )  #1:ConfigFaxModemChannelParams NSEMode=0, CNGDetMode=
11d:14h:43m:9s (   lgr_psbrdif)(2680      )  ActivateDigitMap for channel : 1, MaxDialStringLength
```

The displayed logged messages are color-coded as follows:

- Yellow - fatal error message
- Blue - recoverable error message (i.e., non-fatal error)
- Black - notice message

To stop and clear the Message Log, close the Message Log page by accessing any another page in the Web interface.

Notes:

- It's not recommended to keep a Message Log session open for a prolonged period. This may cause the device to overload. For prolonged (and detailed) debugging, use an external Syslog server.
- You can select the Syslog messages in this page, and copy and paste them into a text editor such as Notepad. This text file (*txt*) can then be sent to AudioCodes Technical Support for diagnosis and troubleshooting.

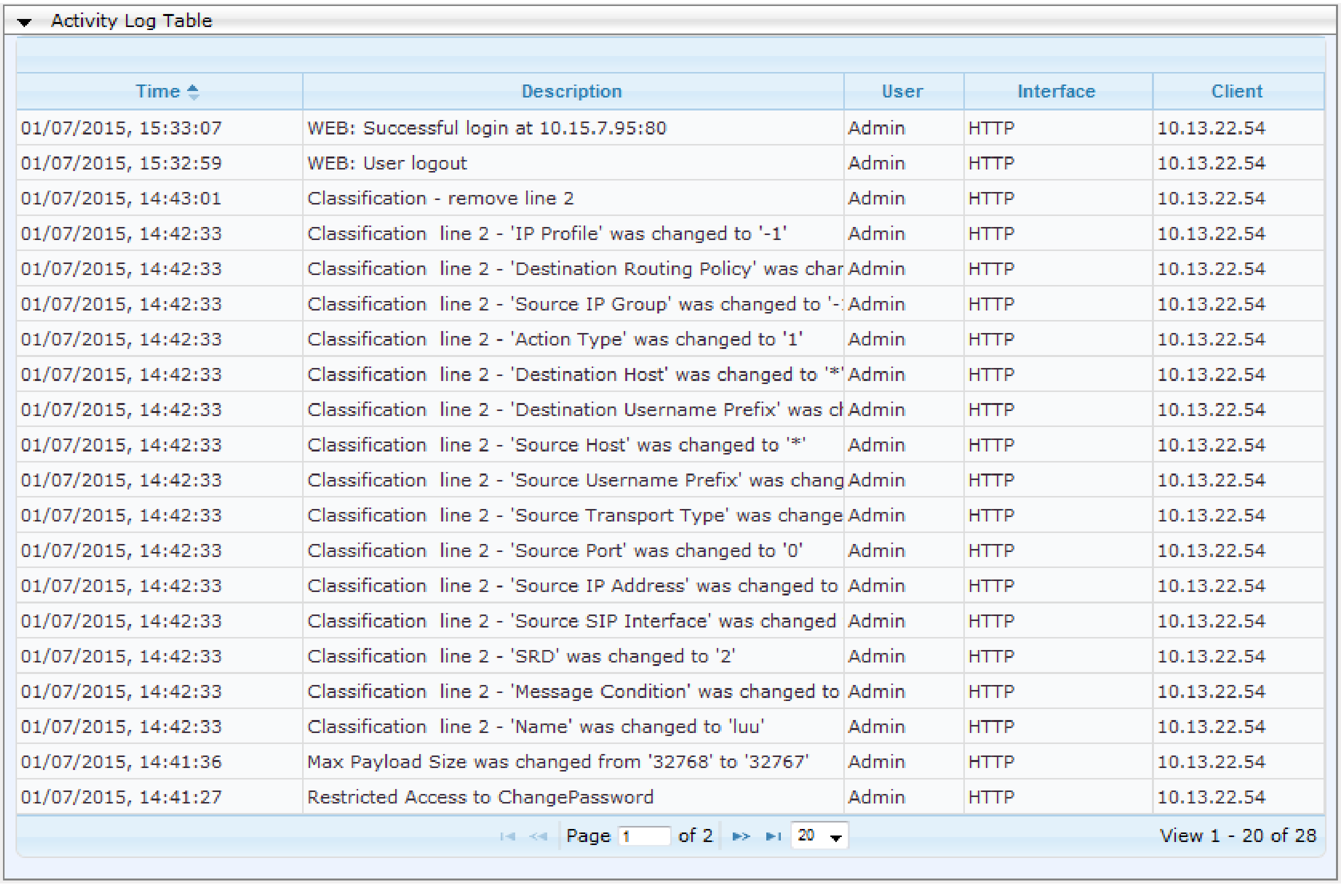
51.2.7 Viewing Web User Activity Logs

If you have enabled the reporting of Web user activities, you can view logged activities in the Web interface's Activity Log table (read-only). For enabling the logging of Web user activities, see "Configuring Web User Activities to Report to Syslog" on page 779.

➢ **To view Web user activity logs:**

- Open the Activity Log table (**Status & Diagnostics** tab > **System Status** menu > **Activity Log**).

Figure 51-7: Activity Log Table



Activity Log Table

Time	Description	User	Interface	Client
01/07/2015, 15:33:07	WEB: Successful login at 10.15.7.95:80	Admin	HTTP	10.13.22.54
01/07/2015, 15:32:59	WEB: User logout	Admin	HTTP	10.13.22.54
01/07/2015, 14:43:01	Classification - remove line 2	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'IP Profile' was changed to '-1'	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'Destination Routing Policy' was char	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'Source IP Group' was changed to '-	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'Action Type' was changed to '1'	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'Destination Host' was changed to '*'	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'Destination Username Prefix' was cl	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'Source Host' was changed to '*'	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'Source Username Prefix' was chang	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'Source Transport Type' was change	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'Source Port' was changed to '0'	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'Source IP Address' was changed to	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'Source SIP Interface' was changed	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'SRD' was changed to '2'	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'Message Condition' was changed to	Admin	HTTP	10.13.22.54
01/07/2015, 14:42:33	Classification line 2 - 'Name' was changed to 'luu'	Admin	HTTP	10.13.22.54
01/07/2015, 14:41:36	Max Payload Size was changed from '32768' to '32767'	Admin	HTTP	10.13.22.54
01/07/2015, 14:41:27	Restricted Access to ChangePassword	Admin	HTTP	10.13.22.54

The table includes the following information:

Table 51-6: Activity Log Table Description

Parameter	Description
Time	Date and time that the user activity was performed.
Description	Description of the user activity.
User	Username of the user that performed the activity.
Interface	Protocol used for the connection to the management interface (e.g., SSH or HTTP).
Client	IP address of the client PC from where the user accessed the Web interface.

51.3 Configuring Debug Recording

This section describes how to configure and activate debug recording, and how to collect debug recording packets. For filtering debug recording packets for specific calls, see "Configuring Log Filter Rules" on page 769.

> **Notes:**
>
> - Debug recording is collected only on the device's OAMP interface.
> - For a detailed description of the debug recording parameters, see "Syslog, CDR and Debug Parameters" on page 841.

51.3.1 Configuring Address of Debug Recording Server

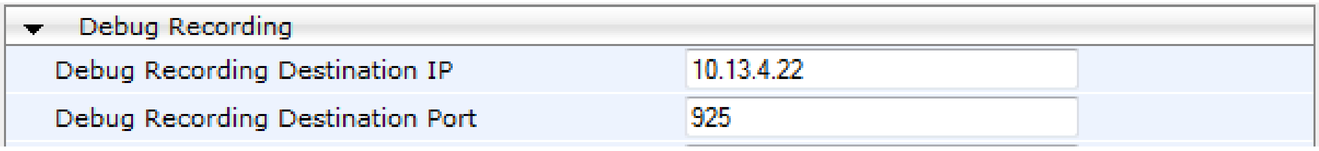
The procedure below describes how to configure the address of the debug recording (capturing) server to where the device sends the captured traffic. Once you configure an address, the device generates DR packets for all calls. However, you can configure the device to generate DR packets for specific calls, using Logging Filter rules in the Logging Filters table (see "Configuring Log Filter Rules" on page 769).

> Note: You can also save debug recordings to an external USB hard drive that is connected to the device's USB port. For more information, see USB Storage Capabilities on page 701.

➢ **To configure the debug recording server's address:**

1. Open the Logging Settings page (**Configuration** tab > **System** menu > **Logging** > **Logging Settings**).

Figure 51-8: Logging Settings Page



▼ Debug Recording

Debug Recording Destination IP: 10.13.4.22

Debug Recording Destination Port: 925

2. In the 'Debug Recording Destination IP' field, configure the IP address of the debug capturing server.
3. In the 'Debug Recording Destination Port' field, configure the port of the debug capturing server.
4. Click **Submit**.

51.3.2 Collecting Debug Recording Messages

To collect debug recording packets, use the open source packet capturing program, Wireshark. AudioCodes proprietary plug-in files for Wireshark are required.

Notes:

- The default debug recording port is 925. You can change the port in Wireshark (**Edit** menu > **Preferences** > **Protocols** > **AC DR**).
- The plug-in files are per major software release of Wireshark. For more information, contact your AudioCodes sales representative.
- The plug-in files are applicable only to Wireshark 32-bit for Windows.

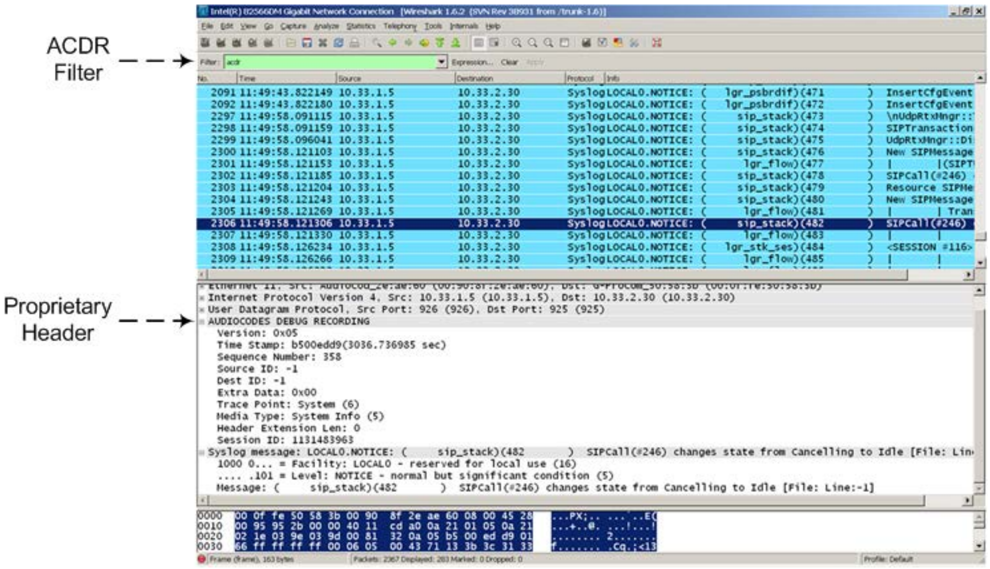
➢ **To install Wireshark and the plug-ins for debug recording:**

1. Install Wireshark on your computer. The Wireshark program can be downloaded from http://www.wireshark.org.
2. Download the proprietary plug-in files from www.audiocodes.com/downloads.
3. Copy the plug-in files to the directory in which you installed Wireshark, as follows:

Copy this file	To this folder on your PC
...\dtds\cdr.dtd	Wireshark\dtds\
...\plugins\<Wireshark ver.>*.dll	Wireshark\plugins\<Wireshark ver.>
...\tpncp\tpncp.dat	Wireshark\tpncp

4. Start Wireshark.
5. In the Filter field, type "acdr" (see the figure below) to view the debug recording messages. Note that the source IP address of the messages is always the OAMP IP address of the device.

The device adds the header "AUDIOCODES DEBUG RECORDING" to each debug recording message, as shown below:


ACDR Filter
Proprietary Header
Intel(R) 82566DM Gigabit Network Connection [Wireshark 1.6.2 (SVN Rev 38931 from /trunk-1.6)]
File Edit View Go Capture Analyze Statistics Telephony Tools Internals Help
Filter: acdr
Expression... Clear Apply
No. Time Source Destination Protocol Info

```
2091 11:49:43.822149 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (   lgr_psbrdif)(471     )  InsertCfgEvent
2092 11:49:43.822180 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (   lgr_psbrdif)(472     )  InsertCfgEvent
2297 11:49:58.091115 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (     sip_stack)(473     )  \nUdpRtxMngr::
2298 11:49:58.091159 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (     sip_stack)(474     )  SIPTransaction
2299 11:49:58.096041 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (     sip_stack)(475     )  UdpRtxMngr::Di
2300 11:49:58.121103 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (     sip_stack)(476     )  New SIPMessage
2301 11:49:58.121153 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (      lgr_flow)(477     )  |          |(SIPT
2302 11:49:58.121185 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (     sip_stack)(478     )  SIPCall(#246)
2303 11:49:58.121204 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (     sip_stack)(479     )  Resource SIPMe
2304 11:49:58.121243 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (     sip_stack)(480     )  New SIPMessage
2305 11:49:58.121269 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (      lgr_flow)(481     )  |          | Tran
2306 11:49:58.121306 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (     sip_stack)(482     )  SIPCall(#246)
2307 11:49:58.121330 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (      lgr_flow)(483     )  |          |
2308 11:49:58.126234 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (   lgr_stk_ses)(484     )  <SESSION #116>
2309 11:49:58.126266 10.33.1.5    10.33.2.30   SyslogLOCAL0.NOTICE: (      lgr_flow)(485     )  |          |

Ethernet II, Src: AudioCod_2e:ae:60 (00:90:8f:2e:ae:60), Dst: G-Procom_50:58:3b (00:0f:fe:50:58:3b)
Internet Protocol Version 4, Src: 10.33.1.5 (10.33.1.5), Dst: 10.33.2.30 (10.33.2.30)
User Datagram Protocol, Src Port: 926 (926), Dst Port: 925 (925)
AUDIOCODES DEBUG RECORDING
  Version: 0x05
  Time Stamp: b500edd9(3036.736985 sec)
  Sequence Number: 358
  Source ID: -1
  Dest ID: -1
  Extra Data: 0x00
  Trace Point: System (6)
  Media Type: System Info (5)
  Header Extension Len: 0
  Session ID: 1131483963
Syslog message: LOCAL0.NOTICE: (     sip_stack)(482       )  SIPCall(#246) changes state from Cancelling to Idle [File: Lin
  1000 0... = Facility: LOCAL0 - reserved for local use (16)
  .... .101 = Level: NOTICE - normal but significant condition (5)
  Message: (     sip_stack)(482       )  SIPCall(#246) changes state from Cancelling to Idle [File: Line:-1]

0000  00 0f fe 50 58 3b 00 90  8f 2e ae 60 08 00 45 28   ...PX;.. ...`..E(
0010  00 95 95 2b 00 00 40 11  cd a0 0a 21 01 05 0a 21   ...+..@. ...!...!
0020  02 1e 03 9e 03 9d 00 81  32 0a 05 b5 00 ed d9 01   ........ 2.......
0030  66 ff ff ff ff 00 06 05  00 43 71 13 3b 3c 31 33   f....... .Cq.;<13
```

51.3.3 Debug Capturing on Physical VoIP Interfaces

You can capture traffic on the device's physical (Ethernet LAN) VoIP interfaces (Layer-2 VLAN tagged packets). The captured traffic can be saved in a PCAP-format file (suitable for Wireshark) to a TFTP (default) or an FTP server. The generated PCAP file is in the Extensible Record Format (ERF). The capture can also be saved to a USB device. The maximum file size of debug captures that can be saved to the device is 20 MB.

To capture traffic on physical VoIP interfaces, use the following CLI commands:

- Starts physical VoIP debug capture:

  ```
  # debug capture voip physical eth-lan
  # debug capture voip physical start
  ```

- Captures packets continuously in a cyclical buffer (packets always captured until stop command):

  ```
  # debug capture VoIP physical cyclic buffer
  ```

- Retrieves latest capture (PCAP file) saved on a specified server:

  ```
  # debug capture VoIP physical get_last_capture <TFTP/FTP
  server IP address>
  ```

  The file is saved to the device's memory (not flash) and erased after a device reset.

- Marks the captured file (useful for troubleshooting process):

  ```
  # debug capture VoIP physical insert-pad
  ```

  Before running this command, the debug capture must be started.

- Displays debug status and configured rules:

  ```
  # debug capture VoIP physical show
  ```

- Specifies the destination (FTP, TFTP, or USB) where you want the PCAP file sent:

  ```
  # debug capture VoIP physical target <ftp|tftp|usb>
  ```

- Stops the debug capture, creates a file named debug-capture-voip-<timestamp>.pcap, and sends it to the TFTP or FTP server:

  ```
  # debug capture voip physical stop <TFTP/FTP server IP
  address>
  ```

  If no IP address is defined, the capture is saved on the device for later retrieval.

52 Self-Testing

The device features the following self-testing modes to identify faulty hardware components:

- **Detailed Test (Configurable):** This test verifies the correct functioning of the different hardware components on the device. This test is done when the device is taken out of service (i.e., not in regular service for processing calls). The test is performed on startup when initialization of the device completes.

  To enable this test, set the ini file parameter, EnableDiagnostics to 1 or 2, and then reset the device. Upon completion of the test and if the test fails, the device sends information on the test results of each hardware component to the Syslog server.

Notes:

- To return the device to regular operation and service, disable the test by setting the ini file parameter, EnableDiagnostics to 0, and then reset the device.
- While the test is enabled, ignore errors sent to the Syslog server.

- **Startup Test (automatic):** This hardware test has minor impact in real-time. While this test is executed, the regular operation of the device is disabled. If an error is detected, an error message is sent to the Syslog.

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53 Creating Core Dump and Debug Files upon Device Crash

For debugging purposes, you can create a core dump file and/or debug file. The files may assist you in identifying the cause of the crash. The core dump can either be included in or excluded from the debug file, or alternatively, sent separately to a TFTP server. You can then provide the files to AudioCodes support team for troubleshooting.

- **Core Dump File:** You can enable the device to send a core dump file to a remote destination upon a device crash. The core dump is a copy of the memory image at the time of the crash. It provides a powerful tool for determining the root cause of the crash. When enabled, the core dump file is sent to a user-defined TFTP server (IP address). If no address is configured, the core dump file is saved to the device's flash memory (if it has sufficient memory). The core dump file is saved as a binary file in the following name format: "**core**_<device name>_ver_<firmware version>_mac_<MAC address>_<date>_<time>", for example, *core_acMediant_ver_700-8-4_mac_00908F099096_1-02-2015_3-29-29*.
- **Debug File:** You can manually retrieve the debug file from the device and save it to a folder on your local PC. The debug file contains the following information:
  - Exception information, indicating the specific point in the code where the crash occurred and a list of up to 50 of the most recent SNMP alarms that were raised by the device before it crashed.
  - Latest log messages that were recorded prior to the crash.
  - Core dump (**only** if enabled, no IP address has been defined, and the device has sufficient memory on its flash).
  - May include additional application-proprietary debug information.

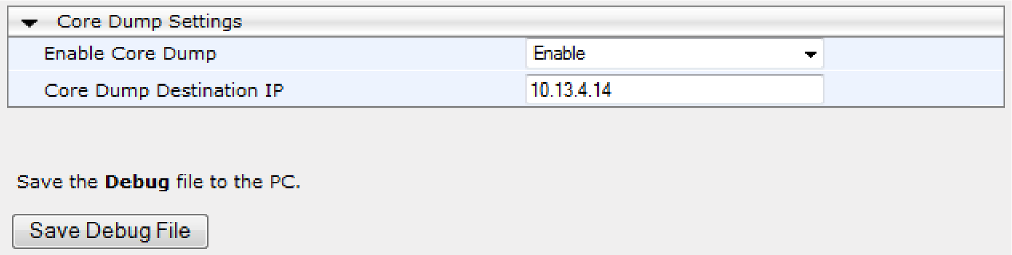
  The debug file is saved as a zipped file in the following name format: "**debug**_<device name>_ver_<firmware version>_mac_<MAC address>_<date>_<time>", for example, *debug_acMediant_ver_700-8-4_mac_00908F099096_1-03-2015_3-29-29*.

The following procedure describes how to configure core dump file creation through the Web interface.

➢ To enable core dump file generation:

1. Set up a TFTP server to where you want to send the core dump file.
2. Open the Debug Utilities page (**Maintenance** tab > **Maintenance** menu > **Debug Utilities**).

Figure 53-1: Debug Utilities Page



Enable Core Dump: Enable
Core Dump Destination IP: 10.13.4.14

Save the **Debug** file to the PC.

3. From the 'Enable Core Dump' drop-down list, select **Enable**.
4. In the 'Core Dump Destination IP' field, enter an IP address of the remote server to where you want the file to be sent (optional).
5. Click **Submit**, and then reset the device with a save-to-flash for your settings to take effect.

The following procedure describes how to retrieve the debug file from the device through the Web interface.

➢ **To save the debug file from the device:**

- In the Debug Utilities page, click the **Save Debug File** button.

54 Testing SIP Signaling Calls

A simulated endpoint can be configured on the device to test SIP signaling of calls between it and a remote destination. This feature is useful in that it can remotely verify SIP message flow without involving the remote end side in the debug process. The SIP test call simulates the SIP signaling process - call setup, SIP 1xx responses, through to completing the SIP transaction with a 200 OK.

The test call sends Syslog messages to a Syslog server, showing the SIP message flow, tone signals (e.g., DTMF), termination reasons, as well as voice quality statistics and thresholds (e.g., MOS).

54.1 Configuring Test Call Endpoints

The Test Call table lets you test the SIP signaling (setup and registration) and media (DTMF signals) of calls between a simulated phone on the device and a remote endpoint. These tests involve both incoming and outgoing calls, where the test endpoint can be configured as the caller or called party. Test calls can be dialed automatically at a user-defined interval and/or manually when required. The simulated phone and remote endpoints are defined as SIP URIs (user@host) and the remote destination can be defined as an IP Group, IP address, or according to a Tel-to-IP routing rule. You can also enable automatic registration of the endpoint.

When a SIP test call is initiated, the device generates a SIP INVITE towards the remote endpoint (e.g., a SIP proxy server or softswitch). It simulates the SIP call setup process, managing SIP 1xx responses and completing the SIP transaction with a 200 OK.

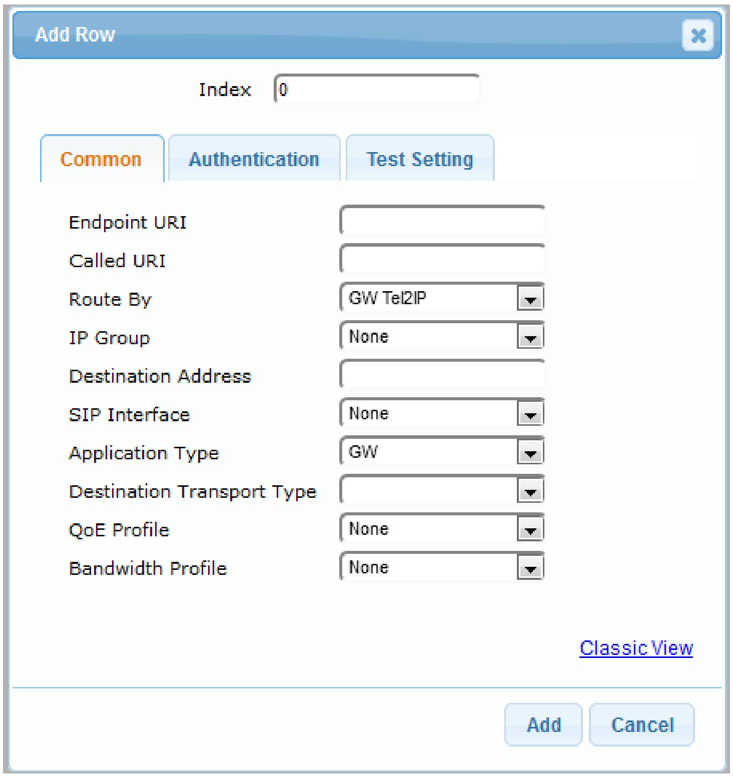
Note: By default, you can configure up to five test calls. However, this number can be increased by installing the relevant Software License Key. For more information, contact your AudioCodes sales representative.

The following procedure describes how to configure test calls through the Web interface. You can also configure it through ini file (Test_Call) or CLI (configure system > test-call > test-call-table).

➢ **To configure a test call:**

1. Open the Test Call table (**Configuration** tab > **System** menu > **Test Call** > **Test Call Table**).
2. Click **Add**; the following dialog box appears:

Figure 54-1: Test Call Table - Add Row Dialog Box



Index: 0

Endpoint URI	
Called URI	
Route By	GW Tel2IP
IP Group	None
Destination Address	
SIP Interface	None
Application Type	GW
Destination Transport Type	
QoE Profile	None
Bandwidth Profile	None

3. Configure a test call according to the parameters described in the table below.
4. Click **Add**, and then save ("burn") your settings to flash memory.

Table 54-1: Test Call Table Parameter Descriptions

Parameter	Description
Common Tab	
Endpoint URI `endpoint-uri` [Test_Call_EndpointURI]	Defines the endpoint's URI. This can be defined as a user or user@host. The device identifies this endpoint only by the URI's user part. The URI's host part is used in the SIP From header in REGISTER requests. The valid value is a string of up to 150 characters. By default, the parameter is not configured. **Note:** The parameter is mandatory.
Called URI `called-uri` [Test_Call_CalledURI]	Defines the destination (called) URI (user@host). The valid value is a string of up to 150 characters. By default, the parameter is not configured.

Parameter	Description
Route By `route-by` [Test_Call_RouteBy]	Defines the type of routing method. This applies to incoming and outgoing calls. ▪ **[0]** GW Tel2IP = (Default) Calls are matched by (or routed to) an SRD and Application type (defined in the SRD and Application Type parameters below). ▪ **[1]** IP Group = Calls are matched by (or routed to) an IP Group. To specify the IP Group, see the 'IP Group' parameter in the table. ▪ **[2]** Dest Address = Calls are matched by (or routed to) an SRD and application type. To specify the address, see the 'Destination Address' parameter in the table. **Notes:** ▪ If configured to **GW Tel2IP** or **Dest Address**, you must assign a SIP Interface (see the 'SIP Interface' parameter in the table). ▪ For REGISTER messages: ✓ The **GW Tel2IP** option cannot be used as the routing method. ✓ If configured to **IP Group,** only Server-type IP Groups can be used.
IP Group `ip-group-id` [Test_Call_IPGroupName]	Assigns an IP Group to the rule, which is the IP Group that the test call is sent to or received from. By default, no value is defined (**None**). **Notes:** ▪ The parameter is applicable only if the 'Route By' parameter is configured to **IP Group** [1]. ▪ The IP Group is used for incoming and outgoing calls.
Destination Address `dst-address` [Test_Call_DestAddress]	Defines the destination host. This can be defined as an IP address[:port] or DNS name[:port]. **Note:** The parameter is applicable only if the 'Route By' parameter is configured to **Dest Address** [2].
SIP Interface `sip-interface-name` [Test_Call_SIPInterfaceName]	Assigns a SIP Interface to the rule, which is the SIP Interface to which the test call is sent and received from. By default, no value is defined (**None**). **Note:** The parameter is applicable only if the 'Route By' parameter is configured to GW Tel2IP or Dest Address.
Application Type `application-type` [Test_Call_ApplicationType]	Defines the application type for the endpoint. This associates the IP Group and SRD to a specific SIP interface. For example, assume two SIP Interfaces are configured in the SIP Interface table where one is set to "GW" and one to "SBC" for the 'Application Type'. If the parameter is set to "SBC", the device uses the SIP Interface set to "SBC". ▪ [0] GW (default) = Gateway application ▪ [2] SBC = SBC application

Parameter	Description
Destination Transport Type `dst-transport` [Test_Call_DestTransportType]	Defines the transport type for outgoing calls. - **[-1]** = Not configured (default) - **[0]** UDP - **[1]** TCP - **[2]** TLS **Note:** The parameter is applicable only if the 'Route By' parameter is set to [2] (Dest Address).
QoE Profile `qoe-profile` [Test_Call_QOEProfile]	Assigns a QoE Profile to the test call. By default, no value is defined (**None**). To configure QoE Profiles, see "Configuring Quality of Experience Profiles" on page 307.
Bandwidth Profile `bandwidth-profile` [Test_Call_BWProfile]	Assigns a Bandwidth Profile to the test call. By default, no value is defined (**None**). To configure Bandwidth Profiles, see "Configuring Bandwidth Profiles" on page 311.
Authentication Tab **Note:** These parameters are applicable only if the Call Party parameter is set to **Caller.**	
Auto Register `auto-register` [Test_Call_AutoRegister]	Enables automatic registration of the endpoint. The endpoint can register to the device itself or to the 'Destination Address' or 'IP Group' parameter settings (see above). - **[0]** Disable (default) - **[1]** Enable
Username `user-name` [Test_Call_UserName]	Defines the authentication username. By default, no username is defined.
Password `password` [Test_Call_Password]	Defines the authentication password. By default, no password is defined.
Test Setting Tab	
Call Party `call-party` [Test_Call_CallParty]	Defines whether the test endpoint is the initiator or receiving side of the test call. - **[0]** Caller (default) - **[1]** Called
Maximum Channels for Session `max-channels` [Test_Call_MaxChannels]	Defines the maximum number of concurrent channels for the test session. For example, if you have configured an endpoint "101" and you set the parameter to "3", the device automatically creates three simulated endpoints - "101", "102" and "103" (i.e., consecutive endpoint URIs are assigned). The default is 1.
Call Duration `call-duration` [Test_Call_CallDuration]	Defines the call duration (in seconds). The valid value is -1 to 100000. The default is 20. A value of 0 means infinite. A value of -1 means that the parameter value is automatically calculated according to the values of the 'Calls per Second' and 'Maximum Channels for Session' parameters. **Note:** The parameter is applicable only if 'Call Party' is set to **Caller**.

Parameter	Description
Calls per Second `calls-per-second` [Test_Call_CallsPerSecond]	Defines the number of calls per second. **Note:** The parameter is applicable only if 'Call Party' is set to **Caller**.
Test Mode `test-mode` [Test_Call_TestMode]	Defines the test session mode. ▪ **[0]** Once = (Default) The test runs until the lowest value between the following is reached: ✓ Maximum channels is reached for the test session, configured by 'Maximum Channels for Session'. ✓ Call duration ('Call Duration') multiplied by calls per second ('Calls per Second'). ✓ Test duration expires, configured by 'Test Duration'. ▪ **[1]** Continuous = The test runs until the configured test duration is reached. If it reaches the maximum channels configured for the test session (in the 'Maximum Channels for Session'), it waits until the configured call duration of a currently established tested call expires before making the next test call. In this way, the test session stays within the configured maximum channels. **Note:** The parameter is applicable only if 'Call Party' is set to **Caller**.
Test Duration `test-duration` [Test_Call_TestDuration]	Defines the test duration (in minutes). The valid value is 0 to 100000. The default is 0 (i.e., unlimited). **Note:** The parameter is applicable only if 'Call Party' is set to **Caller**.
Play `play` [Test_Call_Play]	Enables and defines the playing of a tone to the answered side of the call. ▪ **[0]** Disable ▪ **[1]** DTMF = (Default) Plays a user-defined DTMF string, configured in "Configuring DTMF Tones for Test Calls" on page 800. ▪ **[2]** PRT = Plays a non-DTMF tone from the PRT file (Test Call Tone). For this option, a PRT file must be loaded to the device (see "Prerecorded Tones File" on page 652). **Note:** To configure the DTMF signaling type (e.g., out-of-band or in-band) use the 'DTMF Transport Type' parameter (see "Configuring DTMF Transport Types" on page 189).
Schedule Interval `schedule-interval` [Test_Call_ScheduleInterval]	Defines the interval (in minutes) between automatic outgoing test calls. The valid value range is 0 to 100000. The default is 0 (i.e., scheduling is disabled). **Note:** The parameter is applicable only if 'Call Party' is set to **Caller**.

54.2 Starting and Stopping Test Calls

The following procedure describes how to start, stop, and restart test calls.

➢ **To start, stop, and restart a test call:**

1. In the Test Call table, select the required test call entry; the **Actions** button appears above the table.
2. From the **Actions** drop-down list, choose the required command:
   - **Dial:** starts the test call (this action is applicable only if the test call party is the caller).
   - **Drop Call:** stops the test call.
   - **Restart:** ends all established calls and then starts the test call session again.

The status of the test call is displayed in the 'Test Status' field of the Test Call table:

- "Idle": test call is not active.
- "Scheduled": test call is planned to run (according to 'Schedule Interval' parameter settings)
- "Running": test call has been started (i.e., the **Dial** command was clicked)
- "Receiving": test call has been automatically activated by calls received for the test call endpoint from the remote endpoint (when all these calls end, the status returns to "Idle")
- "Terminating": test call is in the process of terminating the currently established calls (this occurs if the **Drop Call** command is clicked to stop the test)
- "Done": test call has been successfully completed (or was prematurely stopped by clicking the **Drop Call** command)

A more detailed description of this field is displayed below the table when you click the **Show/Hide** button (see "Viewing Test Call Statistics" on page 798).

54.3 Viewing Test Call Statistics

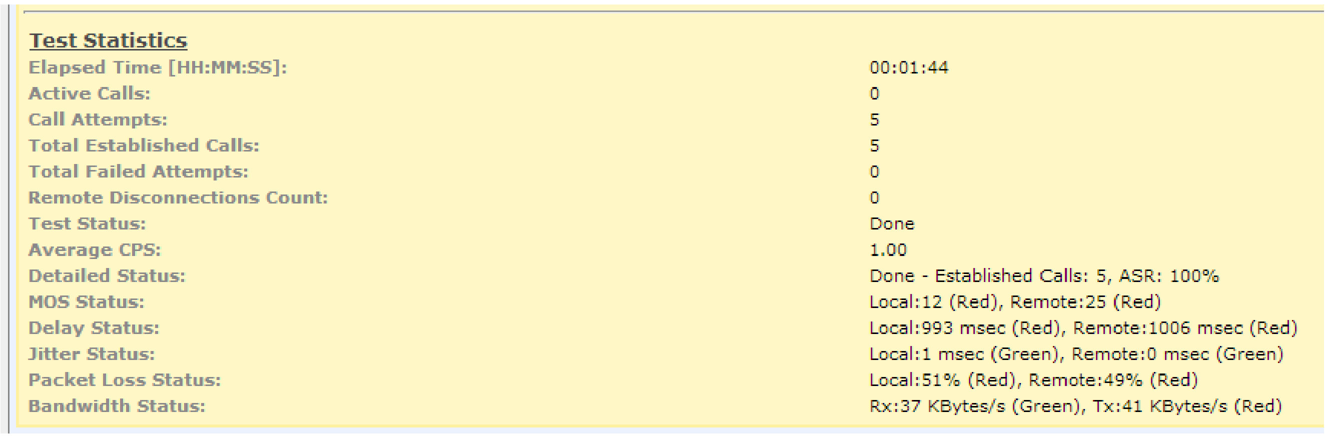
In addition to viewing a brief status description of the test call in the 'Test Status' field (as described in "Starting, Stopping and Restarting Test Calls" on page 798), you can also view a more detailed status description which includes test call statistics.

➢ **To view statistics of a test call:**

1. Open the Test Call table (**Configuration** tab > **System** menu > **Test Call** > **Test Call Table**).
2. Select the test call table entry whose call statistics you want to view.
3. Click the **Show/Hide** button; the call statistics are displayed in the **Test Statistics** pane

located below the table, as shown below:

Figure 54-2: Viewing Test Call Statistics



Test Statistics

Elapsed Time [HH:MM:SS]:	00:01:44
Active Calls:	0
Call Attempts:	5
Total Established Calls:	5
Total Failed Attempts:	0
Remote Disconnections Count:	0
Test Status:	Done
Average CPS:	1.00
Detailed Status:	Done - Established Calls: 5, ASR: 100%
MOS Status:	Local:12 (Red), Remote:25 (Red)
Delay Status:	Local:993 msec (Red), Remote:1006 msec (Red)
Jitter Status:	Local:1 msec (Green), Remote:0 msec (Green)
Packet Loss Status:	Local:51% (Red), Remote:49% (Red)
Bandwidth Status:	Rx:37 KBytes/s (Green), Tx:41 KBytes/s (Red)

The 'Test Statistics' pane displays the following test session information:

- **Elapsed Time:** Duration of the test call since it was started (or restarted).
- **Active Calls:** Number of currently established test calls.
- **Call Attempts:** Number of calls that were attempted.
- **Total Established Calls:** Total number of calls that were successfully established.
- **Total Failed Attempts:** Total number of call attempts that failed.
- **Remote Disconnections Count:** Number of calls that were disconnected by the remote side.
- **Average CPS:** Average calls per second.
- **Test Status:** Displays the status (brief description) as displayed in the 'Test Status' field (see "Starting, Stopping and Restarting Test Calls" on page 798).
- **Average CPS:** Average calls per second.
- **Detailed Status:** Displays a detailed description of the test call status:
  - "Idle": test call is currently not active.
  - "Scheduled - Established Calls: <number of established calls>, ASR: <%>": test call is planned to run (according to 'Schedule Interval' parameter settings) and also shows the following summary of completed test calls:
    - Total number of test calls that were established.
    - Number of successfully answered calls out of the total number of calls attempted (ASR).
  - "Running (Calls: <number of active calls>, ASR: <%>)": test call has been started (i.e., the **Dial** command was clicked) and shows the following:
    - Number of currently active test calls.
    - Number of successfully answered calls out of the total number of calls attempted (Answer Seizure Ratio or ASR).
  - "Receiving (<number of active calls>)": test call has been automatically activated by calls received for this configured test call endpoint from the configured remote endpoint. When all these calls terminate, the status returns to "Idle".
  - "Terminating (<number of active calls>)": the **Drop Call** command has been clicked to stop the test call and the test call is in the process of terminating the currently active test calls.
  - "Done - Established Calls: <number of established calls>, ASR: <%>": test call has been successfully completed (or was prematurely stopped by clicking the **Drop Call** command) and shows the following:
    - Total number of test calls that were established.

- Number of successfully answered calls out of the total number of calls attempted (ASR).

- **MOS Status:** MOS count and color threshold status of local and remote sides according to the assigned QoE Profile.
- **Delay Status:** Packet delay count and color-threshold status of local and remote sides according to the assigned QoE Profile.
- **Jitter Status:** Jitter count and color-threshold status of local and remote sides according to the assigned QoE Profile.
- **Packet Loss Status:** Packet loss count and color-threshold status of local and remote sides according to the assigned QoE Profile.
- **Bandwidth Status:** Tx/Rx bandwidth and color-threshold status according to the assigned Bandwidth Profile.

Note: On the receiving side, when the first call is accepted in "Idle" state, statistics are reset.

54.4 Configuring DTMF Tones for Test Calls

By default, the device plays the DTMF signal tone "3212333" to remote tested endpoints for answered calls (incoming and outgoing). For basic test calls (as described in Configuring Basic Test Calls on page 800), the device can play only the configured DTMF tones (or none, if not configured). For test call endpoints that are configured in the Test Call Rules table, you can configure the device to play either DTMF tones or a tone from an installed PRT file (Test Call Tone). For more information, see Configuring Test Call Endpoints on page 793.

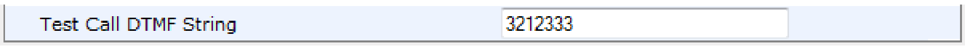
Notes:

- The DTMF signaling type (e.g., out-of-band or in-band) can be configured using the 'DTMF Transport Type' parameter. For more information, see "Configuring DTMF Transport Types" on page 189.
- To generate DTMF tones, the device's DSP resources are required.

➢ **To configure the played DTMF signal to answered test call:**

1. Open the Test Call Settings page (**Configuration** tab > **System** menu > **Test Call** > **Test Call Settings**).

Figure 54-3: DTMF in Test Call Settings Page



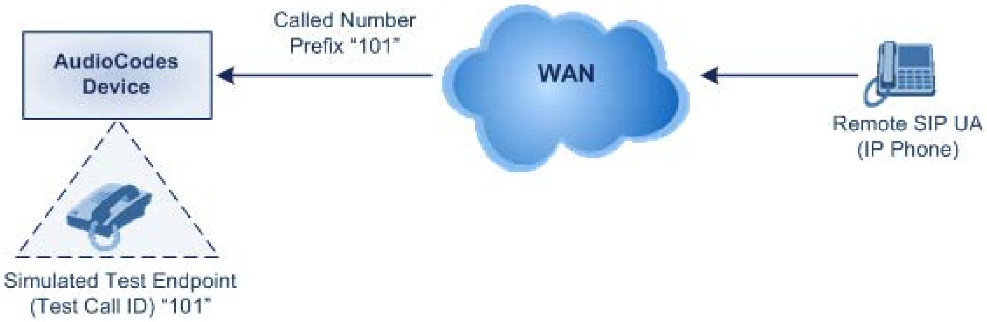
Test Call DTMF String: 3212333

2. In the 'Test Call DTMF String' field, enter the DTMF string (up to 15 digits).
3. Click **Submit**.

54.5 Configuring Basic Test Call

The Basic Test Call feature tests incoming calls from remote SIP (IP) endpoints to a simulated test endpoint on the device. The only required configuration is to assign a prefix number (test call ID) to the simulated endpoint. Incoming calls with this called (destination) prefix number are identified by the device as test calls and sent to the simulated endpoint. The figure below displays a basic test call example:

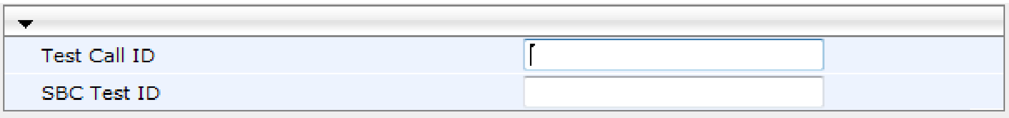
Figure 54-4: Incoming Test Call Example



➢ **To configure basic call testing:**

1. Open the Test Call Settings page (**Configuration** tab > **System** menu > **Test Call** > **Test Call Settings**).

Figure 54-5: Test Call Settings Page



Test Call ID	
SBC Test ID	

2. In the 'Test Call ID' field, enter a prefix for the simulated endpoint.
3. Click **Submit**.

Notes:

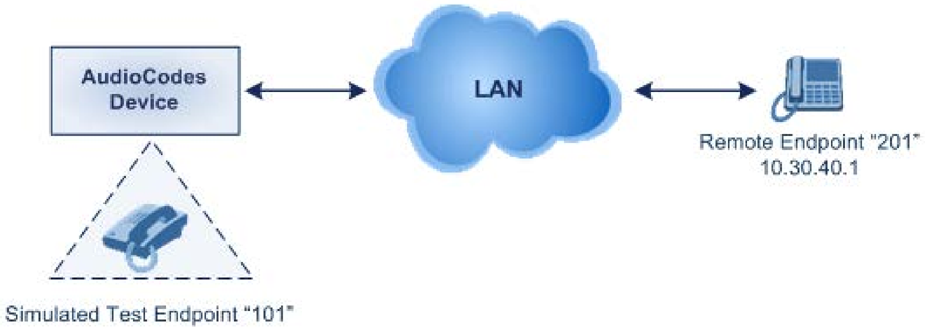
- The device can play DTMF tones to the remote endpoint. For more information, see Configuring DTMF Tones for Test Calls on page 800.
- Test calls are done on all SIP Interfaces.

54.6 Test Call Configuration Examples

Below are a few examples of test call configurations.

- **Single Test Call Scenario:** This example describes the configuration of a simple test call scenario that includes a single test call between a simulated test endpoint on the device and a remote endpoint.

Figure 54-6: Single Test Call Example



- Test Call table configuration:
  - Endpoint URI: "101"

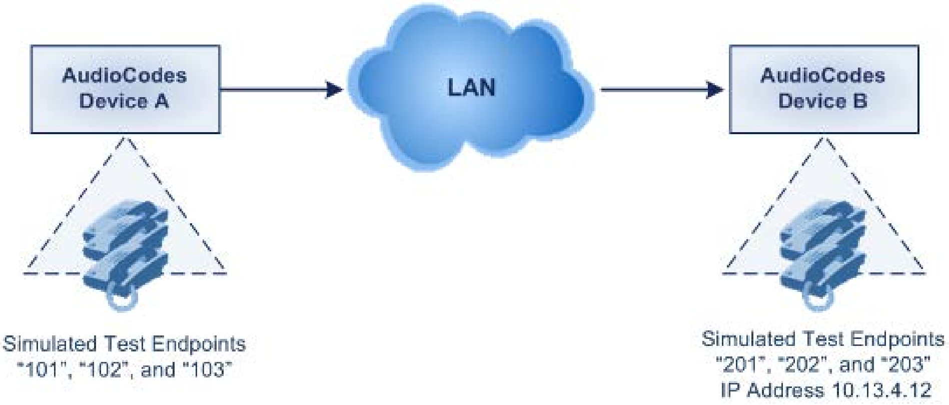
- Called URI: "201"
- Route By: **Dest Address**
- Destination Address: "10.30.40.01"
- SIP Interface: SIPInterface_0
- Call Party: **Caller**
- Test Mode: **Once**

Alternatively, if you want to route the test call using the Tel-to-IP Routing table for the Gateway application, configure the following:

- Test Call table configuration:
  - Endpoint URI: 101@10.0.0.1
  - Route By: GW Tel2IP
  - SIP Interface: SIPInterface_0
  - Called URI: 201@10.30.40.1
  - Call Party: Caller
- Tel-to-IP Routing table configuration:
  - Destination Phone Prefix: 201 (i.e., the Called URI user-part)
  - Source Phone Prefix: 101 (i.e., the Endpoint URI user-part)
  - Destination IP Address: 10.30.40.1

- **Batch Test Call Scenario:** This example describes the configuration of a batch test call setup for scheduled and continuous call testing of multiple endpoints. The test call is done between two AudioCodes devices - Device A and Device B - with simulated test endpoints. This eliminates the need for phone users, who would otherwise need to answer and end calls many times for batch testing. The calls are initiated from Device A, where Device B serves as the remote answering endpoint.

Figure 54-7: Batch Test Call Example


AudioCodes Device A
LAN
AudioCodes Device B
Simulated Test Endpoints "101", "102", and "103"
Simulated Test Endpoints "201", "202", and "203" IP Address 10.13.4.12

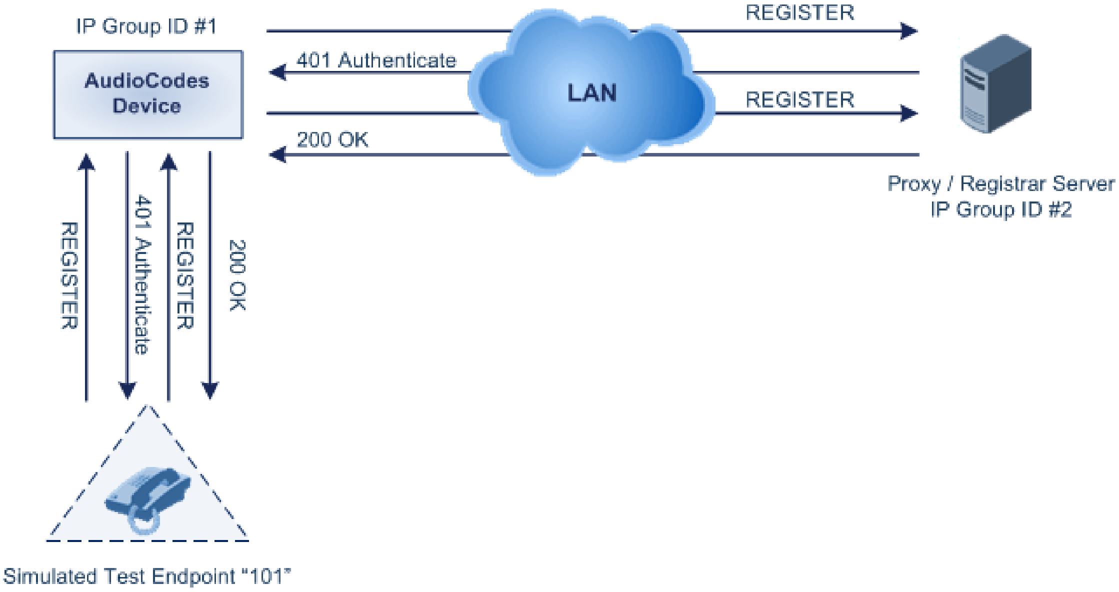
- Test Call table configuration at Device A:
  - Endpoint URI: "101"
  - Called URI: "201"
  - Route By: **Dest Address**
  - Destination Address: "10.13.4.12"
  - SIP Interface: SIPInterface_0
  - Call Party: **Caller**
  - Maximum Channels for Session: "3" (configures three endpoints - "101", "102" and "103")
  - Call Duration: "5" (seconds)
  - Calls per Sec: "1"
  - Test Mode: **Continuous**
  - Test Duration: "3" (minutes)

    - Schedule Interval: "180" (minutes)

- Test Call table configuration at Device B:
    - Endpoint URI: "201"
    - Maximum Channels for Session: "3" (configures three endpoints - "201", "202" and "203)

- **Registration Test Call Scenario:** This example describes the configuration for testing the registration and authentication (i.e., username and pas,sword) process of a simulated test endpoint on the device with an external proxy/registrar server. This is useful, for example, for verifying that endpoints located in the LAN can register with an external proxy and subsequently, communicate with one another.

Figure 54-8: Test Call Registration Example



This example assumes that you have configured your device for communication between LAN phone users such as IP Groups to represent the device (10.13.4.12) and the proxy server, and IP-to-IP routing rules to route calls between these IP Groups.

- Test Call table configuration:
    - Endpoint URI: "101"
    - Called URI: "itsp"
    - Route By: **Dest Address**
    - Destination Address: "10.13.4.12" (this is the IP address of the device itself)
    - SIP Interface: SIPInterface_0
    - Auto Register: **Enable**
    - User Name: "testuser"
    - Password: "12345"
    - Call Party: **Caller**

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55 Pinging a Remote Host or IP Address

You can verify the network connectivity with a remote host or IP address by pinging the network entity.

- IPv4: The ping to an IPv4 address can be done from any of the device's VoIP interfaces that is configured with an IPv4 address. The ping is done using the following CLI command:

  ```
  # ping <IPv4 ip address or host name> source [voip|data]
  interface
  ```

For a complete description of the ping command, refer to the *CLI Reference Guide*.

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Part XI

Appendix

56 Dialing Plan Notation for Routing and Manipulation

The device supports flexible dialing plan notations for denoting the prefix and/or suffix source and/or destination numbers and SIP URI user names in the routing and manipulation tables.

Note: When configuring phone numbers or prefixes in the Web interface, enter them only as digits without any other characters. For example, if you wish to enter the phone number 555-1212, it must be entered as 5551212 without the hyphen (-). If the hyphen is entered, the entry is invalid.

Table 56-1: Dialing Plan Notations for Prefixes and Suffixes

<table>
<tr><th>Notation</th><th>Description</th></tr>
<tr><td><b>x</b> (letter "x")</td><td>Wildcard that denotes any single digit or character.</td></tr>
<tr><td># (pound symbol)</td><td>▪ When used at the end of a prefix, it denotes the end of a number. For example, <b>54324#</b> represents a 5-digit number that starts with the digits 54324.
▪ When used anywhere else in the number (not at the end), it is part of the number (pound key). For example, <b>3#45</b> represents the prefix number 3#45.
▪ To denote the pound key when it appears at the end of the number, the pound key must be enclosed in square brackets. For example, 134[#] represents any number that starts with 134#.</td></tr>
<tr><td><b>*</b> (asterisk symbol)</td><td>▪ When used on its own, it denotes any number or string.
▪ When used as part of a number, it denotes the asterisk key. For example, *345 represents a number that starts with *345.</td></tr>
<tr><td><b>$</b> (dollar sign)</td><td>Denotes an empty prefix for incoming IP calls that do not have a user part in the Request-URI, or for incoming Tel calls that do not have a called or calling number. This is used for the following matching criteria:
▪ Source and Destination Phone Prefix
▪ Source and Destination Username
▪ Source and Destination Calling Name Prefix</td></tr>
<tr><td colspan="2"><b>Range of Digits</b>
<b>Notes:</b>
▪ Dial plans denoting a prefix that is a range must be enclosed in square brackets, e.g., <b>[4-8]</b> or <b>23xx[456]</b>.
▪ Dial plans denoting a prefix that is not a range is not enclosed, e.g., <b>12345#</b>.
▪ Dial plans denoting a suffix must be enclosed in parenthesis, e.g., <b>(4)</b> and <b>(4-8)</b>.
▪ Dial plans denoting a suffix that include multiple ranges, the range must be enclosed in square brackets, e.g., <b>(23xx[4,5,6])</b>.
▪ An example for entering a combined prefix and suffix dial plan - assume you want to match a rule whose destination phone prefix is 4 to 8, and suffix is 234, 235, or 236. The entered value would be the following: <b>[4-8](23[4,5,6])</b>.</td></tr>
<tr><td><b>[n-m]</b> or <b>(n-m)</b></td><td>Represents a range of numbers.
Examples:
▪ To depict prefix numbers from 5551200 to 5551300:
✓ <b>[5551200-5551300]#</b></td></tr>
</table>

Notation	Description
	▪ To depict prefix numbers from 123100 to 123200: ✓ **123[100-200]#** ▪ To depict prefix and suffix numbers together: ✓ 03(100): for any number that starts with 03 and ends with 100. ✓ [100-199](100,101,105): for a number that starts with 100 to 199 and ends with 100, 101 or 105. ✓ 03(abc): for any number that starts with 03 and ends with abc. ✓ 03(5xx): for any number that starts with 03 and ends with 5xx. ✓ 03(400,401,405): for any number that starts with 03 and ends with 400 or 401 or 405. **Notes:** ▪ The value *n* must be less than the value *m*. ▪ Only numerical ranges are supported (not alphabetical letters). ▪ For suffix ranges, the starting (*n*) and ending (*m*) numbers in the range must include the same number of digits. For example, (23-34) is correct, but (3-12) is not.
[n,m,...] or **(n,m,...)**	Represents multiple numbers. The value can include digits or characters. Examples: ▪ To depict a one-digit number starting with 2, 3, 4, 5, or 6: **[2,3,4,5,6]** ▪ To depict a one-digit number ending with 7, 8, or 9: **(7,8,9)** ▪ Prefix with Suffix: **[2,3,4,5,6](7,8,9)** - prefix is denoted in square brackets; suffix in parenthesis For **prefix only**, the notations *d[n,m]e* and *d[n-m]e* can also be used: ▪ To depict a five-digit number that starts with 11, 22, or 33: **[11,22,33]xxx#** ▪ To depict a six-digit number that starts with 111 or 222: **[111,222]xxx#**
[n1-m1,n2-m2,a,b,c,n3-m3] or **(n1-m1,n2-m2,a,b,c,n3-m3)**	Represents a mixed notation of single numbers and multiple ranges. For example, to depict numbers 123 to 130, 455, 766, and 780 to 790: ▪ Prefix: **[123-130,455,766,780-790]** ▪ Suffix: **(123-130,455,766,780-790)** **Note:** The ranges and the single numbers used in the dial plan must have the same number of digits. For example, each number range and single number in the dialing plan example above consists of three digits.

Notation	Description
Special ASCII Characters	The device does not support the use of ASCII characters in manipulation rules and therefore, for LDAP-based queries, the device can use the hexadecimal (HEX) format of the ASCII characters for phone numbers instead. The HEX value must be preceded by a backslash “\”. For example, you can configure a manipulation rule that changes the received number **+49 (7303) 165-xxxxx** to **+49 \287303\29 165-xxxxx**, where *\28* is the ASCII HEX value for “(“ and *\29* is the ASCII HEX value for “)”. The manipulation rule in this example would denote the parenthesis in the destination number prefix using "x" wildcards (e.g., xx165xxxxx#); the prefix to add to the number would include the HEX values (e.g., +49 \287303\29 165-). Below is a list of common ASCII characters and their corresponding HEX values: **ASCII Character** **HEX Value** * \2a (\28) \29 \ \5c / \2f

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57 Configuration Parameters Reference

The device's configuration parameters, default values, and their descriptions are documented in this section.

Note: Parameters and values enclosed in square brackets [...] represent the *ini* file parameters and their enumeration values.

57.1 Management Parameters

This section describes the device's management-related parameters.

57.1.1 General Parameters

The general management parameters are described in the table below.

Table 57-1: General Management Parameters

Parameter	Description
[WebLoginBlockAutoComplete]	Disables autocompletion when entering the management login username in the 'Username' field of the device's Web interface. Disabling autocompletion may be useful for security purposes by hiding previously entered usernames and thereby, preventing unauthorized access to the device's management interface. ▪ [0] Disable = (Default) Autocompletion is enabled and the 'Username' field automatically offers previously logged in usernames. ▪ [1] Enable = Autocompletion is disabled.
[EnforcePasswordComplexity]	Enables the enforcement of management login-password complexity requirements to ensure strong passwords. ▪ [0] Disable (default) ▪ [1] Enable For more information on password complexity requirements, see the 'Password' parameter in Configuring Management User Accounts on page 73.
[CustomerSN]	Defines a serial number (S/N) for the device. **Note:** The device's original S/N is automatically added at the end of the configured S/N. For example, if the original S/N is 8906721 and the configured S/N is "abc123", the resultant S/N is "abc1238906721".

Parameter	Description
Web and Telnet Access List Table [WebAccessList_x]	This table configures up to ten IP addresses that are permitted to access the device's Web interface and Telnet interfaces. Access from an undefined IP address is denied. When no IP addresses are defined in this table, this security feature is inactive (i.e., the device can be accessed from any IP address). The default is 0.0.0.0 (i.e., the device can be accessed from any IP address). For example: WebAccessList_0 = 10.13.2.66 WebAccessList_1 = 10.13.77.7 For a description of the parameter, see "Configuring Web and Telnet Access List" on page 79.

57.1.2 Web Parameters

The Web parameters are described in the table below.

Table 57-2: Web Parameters

Parameter	Description
Enable web access from all interfaces `web-access-from-all-interfaces` [EnableWebAccessFromAllInterfaces]	Enables Web access from any of the device's IP network interfaces. This feature applies to HTTP and HTTPS protocols. ▪ [0] = (Default) Disable – Web access is only through the OAMP interface. ▪ [1] = Enable - Web access is through any network interface. **Note:** For the parameter to take effect, a device reset is required.
Password Change Interval [WebUserPassChangeInterval]	Defines the duration (in minutes) of the validity of Web login passwords. When this duration expires, the password of the Web user must be changed. The valid value is 0 to 100000, where 0 means that the password is always valid. The default is 1140. **Note:** The parameter is applicable only when using the Web Users table, where the default value of the 'Password Age' parameter in the Web Users table inherits the parameter's value.
User Inactivity Timer [UserInactivityTimer]	Defines the duration (in days) for which a user has not logged in to the Web interface, after which the status of the user becomes inactive and can no longer access the Web interface. These users can only log in to the Web interface if their status is changed (to New or Valid) by a System Administrator or Master user. The valid value is 0 to 10000, where 0 means inactive. The default is 90. **Note:** The parameter is applicable only when using the Web Users table.
Session Timeout [WebSessionTimeout]	Defines the duration (in minutes) of inactivity of a logged-in user in the Web interface, after which the user is automatically logged off the Web session. In other words,

Parameter	Description
	the session expires when the user has not performed any operations (activities) in the Web interface for the configured duration. The valid value is 0-100000, where 0 means no timeout. The default is 15. **Note:** You can also configure the functionality per user in the Web Users table (see Advanced User Accounts Configuration on page 73), which overrides this global setting.
Deny Access On Fail Count [DenyAccessOnFailCount]	Defines the maximum number of failed login attempts, after which the requesting IP address is blocked. The valid value range is 0 to 10. The values 0 and 1 mean immediate block. The default is 3.
Deny Authentication Timer [DenyAuthenticationTimer]	Defines the duration (in seconds) for which login to the Web interface is denied from a specific IP address (for all users) when the number of failed login attempts has exceeded the maximum. This maximum is defined by the DenyAccessOnFailCount parameter. Only after this time expires can users attempt to login from this same IP address. The valid value is 0 to 100000, where 0 means that login is not denied regardless of number of failed login attempts. The default is 60.
Display Login Information [DisplayLoginInformation]	Enables display of user's login information on each successful login attempt. ▪ **[0]** = Disable (default) ▪ **[1]** = Enable
[EnableMgmtTwoFactorAuthentication]	Enables Web login authentication using a third-party, smart card. ▪ **[0]** = Disable (default) ▪ **[1]** = Enable When enabled, the device retrieves the Web user's login username from the smart card, which is automatically displayed (read-only) in the Web Login screen; the user is then required to provide only the login password. Typically, a TLS connection is established between the smart card and the device's Web interface, and a RADIUS server is implemented to authenticate the password with the username. Thus, this feature implements a two-factor authentication - what the user has (the physical card) and what the user knows (i.e., the login password).
`http-port` **[HTTPport]**	Defines the LAN HTTP port for Web management (default is 80). To enable Web management from the LAN, configure the desired port. **Note:** For the parameter to take effect, a device reset is required.
[DisableWebConfig]	Determines whether the entire Web interface is read-only. ▪ **[0]** = (Default) Enables modifications of parameters. ▪ **[1]** = Web interface is read-only.

Parameter	Description
	When in read-only mode, parameters can't be modified. In addition, the following pages can't be accessed: Web User Accounts, TLS Contexts, Time and Date, Maintenance Actions, Load Auxiliary Files, Software Upgrade Wizard, and Configuration File. **Note:** For the parameter to take effect, a device reset is required.
[ResetWebPassword]	Enables the device to restore the default management users: ▪ Security Administrator user (username "Admin"; password "Admin") ▪ Monitor user (username "User"; password "User") In addition, all other users that may have been configured (in the Web Users table) are deleted. ▪ [0] = (Default) Disabled. Currently configured users (usernames and passwords) are retained. ▪ [1] = Enabled. Default users are restored (see description above) and all other configured users are deleted. **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ In addition to the ini file (see above), you can also restore the default user accounts through the following management platforms: ✓ SNMP (restores default users and retains other configured users: 1) Set acSysGenericINILine to WEBPasswordControlViaSNMP = 1, and reset the device with a flash burn (set acSysActionSetResetControl to 1 and acSysActionSetReset to 1). 2) Change the username and password in the acSysWEBAccessEntry table. Use the following format: Username acSysWEBAccessUserName: old/pass/new Password acSysWEBAccessUserCode: username/old/new
Customizing Web GUI	

Parameter	Description
[WelcomeMessage] `configure system > welcome-msg`	Defines a welcome message displayed on the Web interface's Web Login page. The format of the ini file table parameter is: [WelcomeMessage] FORMAT WelcomeMessage_Index = WelcomeMessage_Text [\WelcomeMessage] For Example: FORMAT WelcomeMessage_Index = WelcomeMessage_Text WelcomeMessage 1 = "***********************************" ; WelcomeMessage 2 = "********* This is a Welcome message ***" ; WelcomeMessage 3 = "***********************************" ; For more information, see Creating a Login Welcome Message on page 67. **Note:** ▪ Each index row represents a line of text. Up to 20 lines (or rows) of text can be defined. ▪ The configured text message must be enclosed in double quotation marks (i.e., "..."). ▪ If the parameter is not configured, no Welcome message is displayed.
[UseProductName]	Enables the option to customize the name of the device (product) that appears in the management interfaces. ▪ [0] = Disabled (default). ▪ [1] = Enables the display of a user-defined name, which is configured by the UserProductName parameter. For more information, see Customizing the Product Name on page 66.
[UserProductName]	Defines a name for the device instead of the default name. The value can be a string of up to 29 characters. For more information, see Customizing the Product Name on page 66. **Note:** To enable customization of the device name, see the UseProductName parameter.
[UseWebLogo]	Defines whether the Web interface displays a logo image or text. ▪ [0] = (Default) The Web interface displays a logo image, configured by the LogoFileName parameter. ▪ [1] = The Web interface displays text, configured by the WebLogoText parameter. For more information, see Replacing the Corporate Logo on page 64.

Parameter	Description
[WebLogoText]	Defines the text that is displayed instead of the logo in the Web interface. The valid value is a string of up to 15 characters. For more information, see Replacing the Corporate Logo with Text on page 65. **Note:** The parameter is applicable only when the UseWebLogo parameter is configured to 1.
[LogoWidth]	Defines the width (in pixels) of the logo image that you want displayed in the Web interface instead of the default logo. The valid value is 0 to 199. The default is 145. For more information, see Replacing the Corporate Logo with an Image on page 65. **Notes:** ▪ The optimal setting depends on your screen resolution. ▪ If the width of the loaded image is greater than the maximum value, the device automatically resizes the image to the default width size. ▪ The height is limited to 24 pixels. ▪ The parameter is applicable only when the UseWebLogo parameter is configured to 0. ▪ To define the image file, see the LogoFileName parameter.
[LogoFileName]	Defines the name of the image file that you want loaded to the device. This image is displayed as the logo in the Web interface (instead of AudioCodes logo). The file name can be up to 47 characters. For more information, see Replacing the Corporate Logo with an Image on page 65. **Notes:** ▪ The image file type can be one of the following: GIF, PNG, JPG, or JPEG. ▪ The size of the image file can be up to 64 Kbytes. ▪ The parameter is applicable only when the UseWebLogo parameter is configured to 0.

57.1.3 Telnet Parameters

The Telnet parameters are described in the table below.

Table 57-3: Telnet Parameters

Parameter	Description
Embedded Telnet Server `telnet` [TelnetServerEnable]	Enables the device's embedded Telnet server**[0]** Disable ▪ **[1]** Enable Unsecured (default) ▪ **[2]** Enable Secured **Note:** Only management users with Security Administrator level, Administrator level, or Master level can access the device through Telnet (see "Configuring Web User Accounts" on page 70).

Parameter	Description
Telnet Server TCP Port `telnet-port` [TelnetServerPort]	Defines the port number for the embedded Telnet server. The valid range is all valid port numbers. The default port is 23.
Telnet Server Idle Timeout `idle-timeout` [TelnetServerIdleDisconnect]	Defines the duration of an idle CLI (Telnet or SSH) session after which the session is automatically disconnected. The valid range is any value. The default is 5. When configured to 0, idle sessions are not disconnected. **Note:** If you change the parameter's value when there are current Telnet/SSH sessions, the parameter's previous setting is still applied to these current sessions and the parameter's new setting is applied only to new sessions.
Maximum Telnet Sessions `telnet-max-sessions` [TelnetMaxSessions]	Defines the maximum number of permitted, concurrent Telnet/SSH sessions. The valid range is 1 to 5 sessions. The default is 2. **Note:** Before changing the value, make sure that not more than this number of sessions are currently active; otherwise, the new setting will not take effect.
[CLIPrivPass]	Defines the password to access the Enable configuration mode in the CLI. The valid value is a string of up to 50 characters. The default is "Admin". **Note:** The password is case-sensitive.

57.1.4 ini File Parameters

The parameters relating to ini-file management are described in the table below.

Table 57-4: ini File Parameters

Parameter	Description
[INIPasswordsDisplayType]	Defines how passwords are displayed in the ini file. ▪ **[0]** Disable (default) = Passwords are obscured ("encoded"). The passwords are displayed in the following syntax: $1$<obscured password> (e.g., $1$S3p+fno=). ▪ **[1]** Enable = All passwords are hidden and replaced by an asterisk (*).

57.1.5 SNMP Parameters

The SNMP parameters are described in the table below.

Table 57-5: SNMP Parameters

Parameter	Description
Disable SNMP `disable` [DisableSNMP]	Enables and disables SNMP.. ▪ **[0]** No = (Default) SNMP is enabled. ▪ **[1]** Yes = SNMP is disabled.

Parameter	Description
	Note: For the parameter to take effect, a device reset is required.
`port` [SNMPPort]	Defines the device's local (LAN) UDP port used for SNMP Get/Set commands. The range is 100 to 3999. The default port is 161. **Note:** For the parameter to take effect, a device reset is required.
[ChassisPhysicalAlias]	Defines the 'alias' name object for the physical entity as specified by a network manager, and provides a non-volatile 'handle' for the physical entity. The valid range is a string of up to 255 characters.
[ChassisPhysicalAssetID]	Defines the user-assigned asset tracking identifier object for the device's chassis as specified by an EMS, and provides non-volatile storage of this information. The valid range is a string of up to 255 characters.
[ifAlias]	Defines the textual name of the interface. The value is equal to the ifAlias SNMP MIB object. The valid range is a string of up to 64 characters.
`auto-send-keep-alive` [SendKeepAliveTrap]	Enables the device to send NAT keep-alive traps to the port of the SNMP network management station (e.g., AudioCodes EMS). This is used for NAT traversal, and allows SNMP communication with AudioCodes EMS management platform, located in the WAN, when the device is located behind NAT. It is needed to keep the NAT pinhole open for the SNMP messages sent from EMS to the device. The device sends the trap periodically - every 9/10 of the time configured by the NATBindingDefaultTimeout parameter. The trap that is sent is acKeepAlive. For more information on the SNMP trap, refer to the *SNMP Reference Guide*. ▪ **[0]** = (Default) Disable ▪ **[1]** = Enable For configuring the port number, use the KeepAliveTrapPort parameter. **Note:** For the parameter to take effect, a device reset is required.
[KeepAliveTrapPort]	Defines the port of the SNMP network management station to which the device sends keep-alive traps. The valid range is 0 - 65534. The default is port 1161. To enable NAT keep-alive traps, use the SendKeepAliveTrap parameter.
[PM_EnableThresholdAlarms]	Enables the sending of the SNMP trap event, acPerformanceMonitoringThresholdCrossing which is sent every time the threshold (high and low) of a Performance Monitored object (e.g., acPMMediaRealmAttributesMediaRealmBytesTxHighThreshold) is crossed. ▪ [0] = (Default) Disable ▪ [1] = Enable

Parameter	Description
`sys-oid` [SNMPSysOid]	Defines the base product system OID. The default is eSNMP_AC_PRODUCT_BASE_OID_D. **Note:** For the parameter to take effect, a device reset is required.
[SNMPTrapEnterpriseOid]	Defines the Trap Enterprise OID. The default is eSNMP_AC_ENTERPRISE_OID. The inner shift of the trap in the AcTrap subtree is added to the end of the OID in the parameter. **Note:** For the parameter to take effect, a device reset is required.
[acUserInputAlarmDescription]	Defines the description of the input alarm.
[acUserInputAlarmSeverity]	Defines the severity of the input alarm.
[AlarmHistoryTableMaxSize]	Defines the maximum number of rows in the Alarm History table. The parameter can be controlled by the Config Global Entry Limit MIB (located in the Notification Log MIB). The valid range is 50 to 1000. The default is 500. **Note:** For the parameter to take effect, a device reset is required.
[ActiveAlarmTableMaxSize]	Defines the maximum number of currently active alarms that can be displayed in the Active Alarms table. When the table reaches this user-defined maximum capacity (i.e., full), the device sends the SNMP trap event, acActiveAlarmTableOverflow. If the table is full and a new alarm is raised by the <device>, the new alarm is not displayed in the table. The valid range is 50 to 300. The default is 120. For more information on the Active Alarms table, see Viewing Active Alarms on page 709. **Note:** ▪ For the parameter to take effect, a <device> reset is required. ▪ To clear the acActiveAlarmTableOverflow trap, you must reset the device. The reset also deletes all the alarms in the Active Alarms table.
`no-alarm-for-disabled-port` [NoAlarmForDisabledPort]	Enables the device to not send the SNMP trap acBoardControllerFailureAlarm, which indicates a "disabled" (non-configured) telephony port. A disabled port is one that is not configured at all or that is configured but without a Trunk Group ID (i.e., Trunk Group ID is 0), in the Trunk Group table. ▪ [0] Disable = (Default) The device sends the SNMP trap for non-configured ports. ▪ [1] Enable = The device does not send the SNMP trap for non-configured ports. **Note:** ▪ The parameter is applicable to all telephony (analog and digital) port types. ▪ The parameter is applicable only to the Gateway application.

Parameter	Description
`engine-id` [SNMPEngineIDString]	Defines the SNMP engine ID for SNMPv2/SNMPv3 agents. This is used for authenticating a user attempting to access the SNMP agent on the device. The ID can be a string of up to 36 characters. The default is 00:00:00:00:00:00:00:00:00:00:00:00 (12 Hex octets characters). The provided key must be set with 12 Hex values delimited by a colon (":") in the format xx:xx:...:xx. For example, 00:11:22:33:44:55:66:77:88:99:aa:bb **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ Before setting the parameter, all SNMPv3 users must be deleted; otherwise, the parameter setting is ignored. ▪ If the supplied key does not pass validation of the 12 Hex values input or it is set with the default value, the engine ID is generated according to RFC 3411.
SNMP Trap Destination Parameters (configure system/snmp trap destination) **Note:** Up to five SNMP trap managers can be defined.	
SNMP Manager [SNMPManagerIsUsed_x]	Determines the validity of the parameters (IP address and port number) of the corresponding SNMP Manager used to receive SNMP traps. ▪ **[0]** (Check box cleared) = Disabled (default) ▪ **[1]** (Check box selected) = Enabled
IP Address `ip-address` [SNMPManagerTableIP_x]	Defines the IP address of the remote host used as an SNMP Manager. The device sends SNMP traps to this IP address. Enter the IP address in dotted-decimal notation, e.g., 108.10.1.255.
Trap Port `port` [SNMPManagerTrapPort_x]	Defines the port number of the remote SNMP Manager. The device sends SNMP traps to this port. The valid SNMP trap port range is 100 to 4000. The default port is 162.
Trap Enable `send-trap` [SNMPManagerTrapSendingEnable_x]	Enables the sending of traps to the corresponding SNMP manager. ▪ **[0]** Disable = Sending is disabled. ▪ **[1]** Enable = (Default) Sending is enabled.
Trap User `trap-user` [SNMPManagerTrapUser_x]	Defines the SNMPv3 USM user or SNMPv2 user to associate with the trap destination. This determines the trap format, authentication level, and encryption level. By default, it is associated with the SNMPv2 user (SNMP trap community string). The valid value is a string.
Trap Manager Host Name `manager-host-name` [SNMPTrapManagerHostName]	Defines an FQDN of the remote host used as an SNMP manager. The resolved IP address replaces the last entry in the Trap Manager table (defined by the SNMPManagerTableIP parameter) and the last trap manager entry of snmpTargetAddrTable in the snmpTargetMIB. For example: 'mngr.corp.mycompany.com'. The valid range is a string of up to 99 characters.

Parameter	Description
SNMP Community String Parameters	
Community String - Read Only `configure system > snmp > ro-community-string` [SNMPReadOnlyCommunityString_x]	Defines a read-only SNMP community string. Up to five read-only community strings can be configured. The valid value is a string of up to 19 characters that can include only the following: ▪ Upper- and lower-case letters (a to z, and A to Z) ▪ Numbers (0 to 9) ▪ Hyphen (-) ▪ Underline (_) For example, "Public-comm_string1". The default is "public".
Community String - Read / Write `configure system > snmp > rw-community-string` [SNMPReadWriteCommunityString_x]	Defines a read-write SNMP community string. Up to five read-write community strings can be configured. The valid value is a string of up to 19 characters that can include only the following: ▪ Upper- and lower-case letters (a to z, and A to Z) ▪ Numbers (0 to 9) ▪ Hyphen (-) ▪ Underline (_) For example, "Private-comm_string1". The default is "private".
Trap Community String `configure system > snmp trap > community-string` [SNMPTrapCommunityString]	Defines the community string for SNMP traps. The valid value is a string of up to 19 characters that can include only the following: ▪ Upper- and lower-case letters (a to z, and A to Z) ▪ Numbers (0 to 9) ▪ Hyphen (-) ▪ Underline (_) For example, "Trap-comm_string1". The default is "trapuser".
SNMP Trusted Managers Table	
SNMP Trusted Managers `configure system > snmp > trusted-managers` [SNMPTrustedMgr_x]	Defines up to five IP addresses of remote trusted SNMP managers from which the SNMP agent accepts and processes SNMP Get and Set requests. **Notes:** ▪ By default, the SNMP agent accepts SNMP Get and Set requests from any IP address, as long as the correct community string is used in the request. Security can be enhanced by using Trusted Managers, which is an IP address from which the SNMP agent accepts and processes SNMP requests. ▪ If no values are assigned to these parameters any manager can access the device. ▪ Trusted managers can work with all community strings.
SNMP V3 Users Table	
SNMP V3 Users	The p*aramet*er table defines SNMP v3 users. The format of the ini file table parameter is:

Parameter	Description
`configure system > snmp v3-users` [SNMPUsers]	[SNMPUsers] FORMAT SNMPUsers_Index = SNMPUsers_Username, SNMPUsers_AuthProtocol, SNMPUsers_PrivProtocol, SNMPUsers_AuthKey, SNMPUsers_PrivKey, SNMPUsers_Group; [\SNMPUsers] For example: SNMPUsers 1 = v3admin1, 1, 0, myauthkey, -, 1; The example above configures user 'v3admin1' with security level authNoPriv(2), authentication protocol MD5, authentication text password 'myauthkey', and ReadWriteGroup2. For a description of the table, see "Configuring SNMP V3 Users" on page 95.

57.1.6 Serial Parameters

The serial interface parameters are described in the table below.

Table 57-6: Serial Parameters

Parameter	Description
[DisableRS232]	Enables the device's RS-232 (serial) port. ▪ [0] = Enabled ▪ [1] = (Default) Disabled The RS-232 serial port can be used to change the networking parameters and view error/notification messages. For how to establish a serial communication with the device, refer to the Installation Manual. **Note:** For the parameter to take effect, a device reset is required.
[SerialBaudRate]	Defines the serial communication baud rate. The valid values include the following: 1200, 2400, 9600, 14400, 19200, 38400, 57600, or 115200 (default). **Note:** For the parameter to take effect, a device reset is required.
[SerialData]	Defines the serial communication data bit. ▪ **[7]** = 7-bit ▪ **[8]** = (Default) 8-bit **Note:** For the parameter to take effect, a device reset is required.
[SerialParity]	Defines the serial communication polarity. ▪ **[0]** = (Default) None ▪ **[1]** = Odd ▪ **[2]** = Even **Note:** For the parameter to take effect, a device reset is required.
[SerialStop]	Defines the serial communication stop bit. ▪ **[1]** = (Default) 1-bit (default) ▪ **[2]** = 2-bit **Note:** For the parameter to take effect, a device reset is required.
[SerialFlowControl]	Defines the serial communication flow control. ▪ **[0]** = (Default) None ▪ **[1]** = Hardware **Note:** For the parameter to take effect, a device reset is required.

57.1.7 Auxiliary and Configuration File Name Parameters

The configuration files (i.e., Auxiliary files) can be loaded to the device using the Web interface. For loading these files using the *ini* file, you need to configure these files in the *ini* file and configured whether they must be stored in the non-volatile memory. The table below lists the *ini* file parameters associated with these Auxiliary files. For more information on Auxiliary files, see "Loading Auxiliary Files" on page 647.

Table 57-7: Auxiliary and Configuration File Parameters

Parameter	Description
General Parameters	
[SetDefaultOnIniFileProcess]	Determines if all the device's parameters are set to their defaults before processing the updated *ini* file. ▪ **[0]** = Disable - parameters not included in the downloaded *ini* file are not returned to default settings (i.e., retain their current settings). ▪ **[1]** = Enable (default). **Note:** The parameter is applicable only for automatic HTTP update or Web *ini* file upload (not applicable if the *ini* file is loaded using BootP).
[SaveConfiguration]	Determines if the device's configuration (parameters and files) is saved to flash (non-volatile memory). ▪ **[0]** = Configuration isn't saved to flash memory. ▪ **[1]** = (Default) Configuration is saved to flash memory.
Auxiliary and Configuration File Name Parameters	
Call Progress Tones File [CallProgressTonesFilename]	Defines the name of the file containing the Call Progress Tones definitions. For the ini file, the name must be enclosed by single apostrophes, for example, 'cpt_us.dat'. For more information on how to create and load this file, refer to *DConvert Utility User's Guide*. **Note:** For the parameter to take effect, a device reset is required.
Prerecorded Tones File [PrerecordedTonesFileName]	Defines the name of the file containing the Prerecorded Tones. **Note:** For the parameter to take effect, a device reset is required.
Dial Plan [CasTrunkDialPlanName_x]	Defines the Dial Plan name (up to 11-character strings) per trunk. For the ini file, the name must be enclosed by single apostrophes, for example, 'dial_plan_2.dat'. **Note:** The x in the ini file parameter name denotes the trunk number, where 0 is Trunk 1.
Dial Plan File [DialPlanFileName]	Defines the name of the Dial Plan file. This file should be created using AudioCodes DConvert utility (refer to *DConvert Utility User's Guide*). For the ini file, the name must be enclosed by single apostrophes, for example, 'dial_plan.dat'.
[UserInfoFileName]	Defines the name of the file containing the User Information data. For the ini file, the name must be enclosed by single apostrophes, for example, 'userinfo_us.dat'.

57.1.8 Automatic Update Parameters

The automatic update of software and configuration files parameters are described in the table below.

Table 57-8: Automatic Update of Software and Configuration Files Parameters

Parameter	Description
General Automatic Update Parameters	
`configure system/automatic-update/update-firmware` [AutoUpdateCmpFile]	Enables the Automatic Update mechanism for the cmp file. ▪ **[0]** = (Default) The Automatic Update mechanism doesn't apply to the cmp file. ▪ **[1]** = The Automatic Update mechanism includes the cmp file. **Note:** For the parameter to take effect, a device reset is required.
`configure system > automatic-update > update-frequency` [AutoUpdateFrequency]	Defines the interval (in minutes) that the device waits between consecutive automatic updates. The default is 0 (i.e., the update at fixed intervals mechanism is disabled). **Note:** For the parameter to take effect, a device reset is required.
`configure system > automatic-update > predefined-time` [AutoUpdatePredefinedTime]	Defines schedules (time of day) for performing automatic updates. The format syntax of the parameter is 'hh:mm', where *hh* denotes the hour and *mm* the minutes. The value must be enclosed in single apostrophes. For example, '20:18'. **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ The actual update time is randomized by five minutes to reduce the load on the Web servers.

`automatic-update > http-user-agent`
[AupdHttpUserAgent]
	Defines the information sent in the HTTP User-Agent header in the HTTP Get requests sent by the device to the provisioning server for the Automatic Update mechanism.
	The valid value is a string of up to 511 characters. The information can include any user-defined string or the following string variable tags (case-sensitive):
	▪ <NAME>: product name, according to the installed Software License Key
	▪ <MAC>: device's MAC address
	▪ <VER>: software version currently installed on the device, e.g., "7.00.200.001"
	▪ <CONF>: configuration version, as configured by the ini file parameter, INIFileVersion or CLI command, configuration-version
	The device automatically populates these tag variables with actual values in the sent header. By default, the device sends the following in the User-Agent header:
	`User-Agent: Mozilla/4.0 (compatible; AudioCodes; <NAME>;<VER>;<MAC>;<CONF>)`
	For example, if you set AupdHttpUserAgent = MyWorld-<NAME>;<VER>(<MAC>), the device sends the following User-Agent header:
	`User-Agent: MyWorld-Mediant;7.00.200.001(00908F1DD0D3)`
	Notes:
	▪ The variable tags are case-sensitive.

Parameter	Description
	▪ If you configure the parameter with the <CONF> variable tag, you must reset the device with a burn-to-flash for your settings to take effect. ▪ The tags can be defined in any order. ▪ The tags must be defined adjacent to one another (i.e., no spaces).
`automatic-update > auto-firmware` [AutoCmpFileUrl]	Defines the filename and path (URL) to the provisioning server from where the software file (.cmp) can be downloaded, based on timestamp for the Automatic Updated mechanism. The valid value is an IP address in dotted-decimal notation or an FQDN.
`system > tls > aupd-verify-cert` [AUPDVerifyCertificates]	Determines whether the Automatic Update mechanism verifies server certificates when using HTTPS. ▪ **[0]** = Disable (default) ▪ **[1]** = Enable
[AUPDDigestUsername]	Defines the username for digest (MD5 cryptographic hashing) access authentication with the HTTP server used for the Automatic Update feature. The valid value is a string of up to 50 characters. By default, no value is defined.
[AUPDDigestPassword]	Defines the password for digest (MD5 cryptographic hashing) access authentication with the HTTP server used for the Automatic Update feature. The valid value is a string of up to 50 characters. By default, no value is defined.
`configure system > automatic-update > crc-check regular` [AUPDCheckIfIniChanged]	Enables the device to perform cyclic redundancy checks (CRC) on downloaded configuration files (ini) during the Automatic Update process. The CRC checks whether the content (raw data) of the downloaded file is different to the content of the previously downloaded file from the previous Automatic Update process. The device compares the CRC check value (code) result with the check value of the previously downloaded file. If the check values are identical, it indicates that the file has no new configuration settings, and the device discards the file. If the check values are different, the device installs the downloaded file and applies the new configuration settings. ▪ **[0]** = (Default) Disable - the device does not perform CRC and installs the downloaded file regardless. ▪ **[1]** = Enable CRC for the entire file, including line order (i.e., same text must be on the same lines). If there are differences between the files, the device installs the downloaded file. If there are no differences, the device discards the newly downloaded file. ▪ **[2]** = Enable CRC for individual lines only. Same as option [1], except that the CRC ignores the order of lines (i.e., same text can be on different lines).

Parameter	Description
`config-system > automatic-update tftp-block-size` [AUPDTftpBlockSize]	Defines the size of the TFTP data blocks (packets) when downloading a file from a TFTP server for the Automatic Update mechanism. This is in accordance to RFC 2348. TFTP block size is the physical packet size (in bytes) that a network can transmit. When configured to a value higher than the default (512 bytes), but lower than the client network's Maximum Transmission Unit (MTU), the file download speed can be significantly increased. The valid value is 512 to 8192. The default is 512. **Notes:** ▪ A higher value does not necessarily mean better performance. ▪ The block size should be small enough to avoid IP fragmentation in the client network (i.e., below MTU). ▪ This feature is applicable only to TFTP servers that support this option.
[ResetNow]	Invokes an immediate device reset. This option can be used to activate offline (i.e., not on-the-fly) parameters that are loaded using the parameter IniFileUrl. ▪ **[0]** = (Default) The immediate restart mechanism is disabled. ▪ **[1]** = The device immediately resets after an *ini* file with the parameter set to 1 is loaded. **Note:** If you use the parameter in an ini file for periodic automatic provisioning with non-HTTP (e.g., TFTP) and without CRC, the device resets upon every file download.
Software/Configuration File URL Path for Automatic Update Parameters	
`automatic-update > firmware` [CmpFileURL]	Defines the name of the *cmp* file and the path to the server (IP address or FQDN) from where the device can load the *cmp* file and update itself. The *cmp* file can be loaded using HTTP/HTTPS. For example, http://192.168.0.1/filename. **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ When the parameter is configured, the device always loads the *cmp* file after it is reset. ▪ The *cmp* file is validated before it's burned to flash. The checksum of the *cmp* file is also compared to the previously burnt checksum to avoid unnecessary resets. ▪ The maximum length of the URL address is 255 characters.

Parameter	Description
`voice-configuration` [IniFileURL]	Defines the name of the *ini* file and the path to the server (IP address or FQDN) on which it is located. The *ini* file can be loaded using HTTP/HTTPS. For example: http://192.168.0.1/filename http://192.8.77.13/config_<MAC>.ini https://<username>:<password>@<IP address>/<file name> **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ When using HTTP or HTTPS, the date and time of the *ini* file are validated. Only more recently dated *ini* files are loaded. ▪ The case-sensitive string, "<MAC>" can be used in the file name for instructing the device to replace it with the device's MAC address. For more information, see "MAC Address Placeholder in Configuration File Name" on page 689. This option allows the loading of specific configurations for specific devices. ▪ The maximum length of the URL address is 99 characters.
cli-script <URL> `[AUPDCliScriptURL]`	Defines the URL of the server where the CLI Script file containing the device's configuration is located. This file is used for automatic provisioning. **Note:** The case-sensitive string, "<MAC>" can be used in the file name for instructing the device to replace it with the device's MAC address. For more information, see MAC Address Placeholder in Configuration File Name on page 689.
`prerecorded-tones` [PrtFileURL]	Defines the name of the Prerecorded Tones (PRT) file and the path to the server (IP address or FQDN) on which it is located. For example: http://server_name/file, https://server_name/file. **Note:** The maximum length of the URL address is 99 characters.
`call-progress-tones` [CptFileURL]	Defines the name of the CPT file and the path to the server (IP address or FQDN) on which it is located. For example: http://server_name/file, https://server_name/file. **Note:** The maximum length of the URL address is 99 characters.
`tls-root-cert` [TLSRootFileUrl]	Defines the name of the TLS trusted root certificate file and the URL from where it can be downloaded. **Note:** For the parameter to take effect, a device reset is required.
`tls-cert` [TLSCertFileUrl]	Defines the name of the TLS certificate file and the URL from where it can be downloaded. **Note:** For the parameter to take effect, a device reset is required.
`tls-private-key` [TLSPkeyFileUrl]	Defines the URL for downloading a TLS private key file using the Automatic Update facility.
`user-info` [UserInfoFileURL]	Defines the name of the User Information file and the path to the server (IP address or FQDN) on which it is located. For example: http://server_name/file, https://server_name/file **Note:** The maximum length of the URL address is 99 characters.
`configure system > automatic-update > feature-key` [FeatureKeyURL]	Defines the name of the License Key file and the URL address of the server on which the file is located.

Parameter	Description

`configure system > automatic-update > template-url`
`[TemplateUrl]`
	Defines the URL address in the File Template for automatic updates, of the provisioning server on which the files to download are located.
	For more information, see File Template for Automatic Provisioning on page 690.

`configure system > automatic-update > template-files-list`
`[AupdFilesList]`
	Defines the list of file types in the File Template for automatic updates, to download from the provisioning server.
	For more information, see File Template for Automatic Provisioning on page 690.

`web-favicon` [WebFaviconFileUrl]	Defines the name of the favicon image file and the URL address of the server on which the file is located. This is used for the Automatic Update feature. For more information, see Customizing the Favicon on page 66.

57.2 Networking Parameters

This subsection describes the device's networking parameters.

57.2.1 Ethernet Parameters

The Ethernet parameters are described in the table below.

Table 57-9: Ethernet Parameters

Parameter	Description
Physical Ports Settings Table	
Physical Ports Settings `configure voip/physical-port` [PhysicalPortsTable]	The table configures the physical Ethernet ports. The format of the ini file table parameter is as follows: [PhysicalPortsTable] FORMAT PhysicalPortsTable_Index = PhysicalPortsTable_Port, PhysicalPortsTable_Mode, PhysicalPortsTable_SpeedDuplex, PhysicalPortsTable_PortDescription, PhysicalPortsTable_GroupMember, PhysicalPortsTable_GroupStatus; [\PhysicalPortsTable] For a detailed description of the table, see Configuring Physical Ethernet Ports on page 127.
Ethernet Group Settings Table	
Ethernet Group Settings `configure voip/ether-group` [EtherGroupTable]	Defines the transmit (Tx) and receive (Rx) settings for the Ethernet port groups. The format of the ini file table parameter is: [EtherGroupTable] FORMAT EtherGroupTable_Index = EtherGroupTable_Group, EtherGroupTable_Mode, EtherGroupTable_Member1, EtherGroupTable_Member2; [\EtherGroupTable] For a detailed description of the table, see Configuring Ethernet Port Groups on page 129. **Note:** For the parameter to take effect, a device reset is required.
Ethernet Device Table	

Parameter	Description
Ethernet Device Table [DeviceTable]	Defines Ethernet Devices (VLANs). The format of the ini file table parameter is as follows: [DeviceTable] FORMAT DeviceTable_Index = DeviceTable_VlanID, DeviceTable_UnderlyingInterface, DeviceTable_DeviceName, DeviceTable_Tagging; [\DeviceTable] For a detailed description of the table, see Configuring Underlying Ethernet Devices on page 131.

57.2.2 Multiple VoIP Network Interfaces and VLAN Parameters

The IP network interfaces and VLAN parameters are described in the table below.

Table 57-10: IP Network Interfaces and VLAN Parameters

Parameter	Description
Interface Table	
Interface Table `configure voip > interface network-if display` [InterfaceTable]	The table configures the Interface table. The format of the ini file table parameter is as follows: [InterfaceTable] FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes, InterfaceTable_InterfaceMode, InterfaceTable_IPAddress, InterfaceTable_PrefixLength, InterfaceTable_Gateway, InterfaceTable_VlanID, InterfaceTable_InterfaceName, InterfaceTable_PrimaryDNSServerIPAddress, InterfaceTable_SecondaryDNSServerIPAddress, InterfaceTable_UnderlyingDevice; [\InterfaceTable] For a detailed description of the table, see "Configuring IP Network Interfaces" on page 133.
[EnableNTPasOAM]	Defines the application type for Network Time Protocol (NTP) services. ▪ **[1]** = OAMP (default) ▪ **[0]** = Control **Note:** For the parameter to take effect, a device reset is required.

57.2.3 Routing Parameters

The IP network routing parameters are described in the table below.

Table 57-11: IP Network Routing Parameters

Parameter	Description
Send ICMP Unreachable Messages	Enables sending of ICMP Unreachable messages. ▪ [0] Enable = (Default) Device sends these messages.

Parameter	Description
[DisableICMPUnreachable]	▪ [1] Disable = Device does not send these messages.
Send and Receive ICMP Redirect Messages [DisableICMPRedirects]	Enables sending and receiving of ICMP Redirect messages. ▪ [0] Enable = (Default) Device sends and accepts these messages. ▪ [1] Disable = Device rejects these messages and also does not send them.
Static Route Table	
Static Route Table `configure voip > static` [StaticRouteTable]	Defines up to 30 static IP routes for the device. The format of the ini file table parameter is as follows: [StaticRouteTable] FORMAT StaticRouteTable_Index = StaticRouteTable_DeviceName, StaticRouteTable_Destination, StaticRouteTable_PrefixLength, StaticRouteTable_Gateway, StaticRouteTable_Description; [\StaticRouteTable] For a description of the parameter, see "Configuring Static IP Routes" on page 141.

57.2.4 Quality of Service Parameters

The Quality of Service (QoS) parameters are described in the table below.

Table 57-12: QoS Parameters

Parameter	Description
Layer-2 Class Of Service (CoS) Parameters (VLAN Tag Priority Field)	
DiffServ Table `configure voip > vlan-mapping` [DiffServToVlanPriority]	The table configures DiffServ-to-VLAN Priority mapping. For each packet sent to the LAN, the VLAN Priority of the packet is set according to the DiffServ value in the IP header of the packet. The format of this ini file is as follows: [DiffServToVlanPriority] FORMAT DiffServToVlanPriority_Index = DiffServToVlanPriority_DiffServ, DiffServToVlanPriority_VlanPriority; [\DiffServToVlanPriority] For example: DiffServToVlanPriority 0 = 46, 6; DiffServToVlanPriority 1 = 40, 6; DiffServToVlanPriority 2 = 26, 4; DiffServToVlanPriority 3 = 10, 2; For a description of the table, see Configuring Quality of Service on page 144. **Note:** For the parameter to take effect, a device reset is required.
Layer-3 Class of Service (TOS/DiffServ) Parameters	
Media Premium QoS `media-qos` [PremiumServiceClassMediaDiffServ]	Global parameter that defines the DiffServ value for Premium Media CoS content. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_IPDiffServ). For a detailed description of the

Parameter	Description
	parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
Control Premium QoS `control-qos` [PremiumServiceClassControlDiffServ]	Global parameter that defines the DiffServ value for Premium Control CoS content (Call Control applications). You can also configure this functionality per specific calls, using IP Profiles (IpProfile_SigIPDiffServ). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
Gold QoS `gold-qos` [GoldServiceClassDiffServ]	Defines the DiffServ value for the Gold CoS content (Streaming applications). The valid range is 0 to 63. The default is 26.
Bronze QoS `bronze-qos` [BronzeServiceClassDiffServ]	Defines the DiffServ value for the Bronze CoS content (OAMP applications). The valid range is 0 to 63. The default is 10.

57.2.5 NAT and STUN Parameters

The Network Address Translation (NAT) parameters are described in the table below.

Table 57-13: NAT Parameters

Parameter	Description
NAT Parameters	
NAT Mode `disable-NAT-traversal` [NATMode]	Enables the NAT feature for media when the device communicates with UAs located behind NAT. ▪ [0] Enable NAT Option = NAT traversal is performed only if the UA is located behind NAT: ✓ UA behind NAT: The device sends the media packets to the IP address:port obtained from the source address of the first media packet received from the UA. ✓ UA not behind NAT: The device sends the packets to the IP address:port specified in the SDP 'c=' line (Connection) of the first received SIP message. Note: If the SIP session is established (ACK) and the device (not the UA) sends the first packet, it sends it to the address obtained from the SIP message and only after the device receives the first packet from the UA does it determine whether the UA is behind NAT. ▪ [1] Disable NAT = (Default) The device considers the UA as not located behind NAT and sends media packets to the UA using the IP address:port specified in the SDP 'c=' line (Connection) of the first received SIP message.

Parameter	Description
	- [2] Force NAT = The device always considers the UA as behind NAT and sends the media packets to the IP address:port obtained from the source address of the first media packet received from the UA. The device only sends packets to the UA after it receives the first packet from the UA (to obtain the IP address). - [3] NAT By Signaling = The device identifies whether or not the UA is located behind NAT based on the SIP signaling. The device assumes that if signaling is behind NAT that the media is also behind NAT, and vice versa. If located behind NAT, the device sends media as described in option [2] Force NAT; if not behind NAT, the device sends media as described in option [1] Disable NAT. This option is applicable only to SBC calls. If the parameter is configured to this option, Gateway calls use option [0] Enable NAT Option, by default. For more information on handling calls from UAs behind NAT, see "First Incoming Packet Mechanism" on page 154.
NAT IP Address `nat-ip-addr` [StaticNatIP]	Defines the global (public) IP address of the device to enable static NAT between the device and the Internet. **Note:** For the parameter to take effect, a device reset is required.
[NATBindingDefaultTimeout]	The device sends SNMP keep-alive traps periodically - every 9/10 of the time configured by the parameter (in seconds). Therefore, the parameter is applicable only if the SendKeepAliveTrap parameter is set to 1. The parameter is used to allow SNMP communication with AudioCodes EMS management platform, located in the WAN, when the device is located behind NAT. It is needed to keep the NAT pinhole open for the SNMP messages sent from EMS to the device. The valid range is 0 to 2,592,000. The default is 30. **Note:** For the parameter to take effect, a device reset is required.
SIP NAT Detection `configure voip/sip-definition advanced-settings/sip-nat-detect` [SIPNatDetection]	Enables the device to detect whether the incoming INVITE message is sent from an endpoint located behind NAT. - **[0]** Disable = Disables the device's NAT Detection mechanism. Incoming SIP messages are processed as received from endpoints that are not located behind NAT and sent according to the SIP standard. - **[1]** Enable (default) = Enables the device's NAT Detection mechanism.

57.2.6 DNS Parameters

The Domain name System (DNS) parameters are described in the table below.

Table 57-14: DNS Parameters

Parameter	Description
Internal DNS Table	

Parameter	Description
Internal DNS Table `configure voip > voip-network dns Dns2Ip` [DNS2IP]	The table defines the internal DNS table for resolving host names into IP addresses. The format of the ini file table parameter is: [Dns2Ip] FORMAT Dns2Ip_Index = Dns2Ip_DomainName, Dns2Ip_FirstIpAddress, Dns2Ip_SecondIpAddress, Dns2Ip_ThirdIpAddress, Dns2Ip_FourthIpAddress; [\Dns2Ip] For example: Dns2Ip 0 = DnsName, 1.1.1.1, 2.2.2.2, 3.3.3.3, ; For a detailed description of the table, see "Configuring the Internal DNS Table" on page 147.
Internal SRV Table	
Internal SRV Table `configure voip > voip-network dns Srv2Ip` [SRV2IP]	The table defines the internal SRV table for resolving host names into DNS A-Records. Three different A-Records can be assigned to a host name. Each A-Record contains the host name, priority, weight, and port. The format of the ini file table parameter is: [SRV2IP] FORMAT SRV2IP_Index = SRV2IP_InternalDomain, SRV2IP_TransportType, SRV2IP_Dns1, SRV2IP_Priority1, SRV2IP_Weight1, SRV2IP_Port1, SRV2IP_Dns2, SRV2IP_Priority2, SRV2IP_Weight2, SRV2IP_Port2, SRV2IP_Dns3, SRV2IP_Priority3, SRV2IP_Weight3, SRV2IP_Port3; [\SRV2IP] For example: SRV2IP 0 = SrvDomain,0,Dnsname1,1,1,500,Dnsname2,2,2,501,$$,0,0,0; For a detailed description of the table, see "Configuring the Internal SRV Table" on page 148.

57.2.7 DHCP Parameters

The Dynamic Host Control Protocol (DHCP) parameters are described in the table below.

Table 57-15: DHCP Parameters

Parameter	Description
Enable DHCP [DHCPEnable]	Enables DHCP client functionality. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ For a detailed description of DHCP, see "DHCP-based Provisioning" on page 679. ▪ The parameter is a "hidden" parameter. Once defined and saved to flash memory, its value doesn't revert to default even if the parameter doesn't appear in the *ini* file.
[DHCPSpeedFactor]	Defines the device's DHCP renewal speed for a leased IP address from a DHCP server. ▪ **[0]** = Disable

Parameter	Description
	▪ **[1]** = (Default) Normal ▪ **[2]** to **[10]** = Fast When set to 0, the DHCP lease renewal is disabled. Otherwise, the renewal time is divided by this factor. Some DHCP-enabled routers perform better when set to 4. **Note:** For the parameter to take effect, a device reset is required.
DHCP Servers Table	
DHCP Servers Table `configure voip > dhcp server <index>` [DhcpServer]	Defines the device's embedded DHCP server. The format of the ini file table parameter is as follows: [DhcpServer] FORMAT DhcpServer_Index = DhcpServer_InterfaceName, DhcpServer_StartIPAddress, DhcpServer_EndIPAddress, DhcpServer_SubnetMask, DhcpServer_LeaseTime, DhcpServer_DNSServer1, DhcpServer_DNSServer2, DhcpServer_NetbiosNameServer, DhcpServer_NetbiosNodeType, DhcpServer_NTPServer1, DhcpServer_NTPServer2, DhcpServer_TimeOffset, DhcpServer_TftpServer, DhcpServer_BootFileName, DhcpServer_ExpandBootfileName, DhcpServer_OverrideRouter, DhcpServer_SipServer, DhcpServer_SipServerType; [\DhcpServer] For a detailed description of the table, see Configuring the Device's DHCP Server.
DHCP Vendor Class Table	
DHCP Vendor Class table `configure voip > dhcp vendor-class` [DhcpVendorClass]	Defines Vendor Class Identifier (VCI) names (DHCP Option 60) for the device's DHCP server. Only if the DHCPDiscover request message, received from the DHCP client, contains this value does the device provide DHCP services. The format of the ini file table parameter is as follows: [DhcpVendorClass] FORMAT DhcpVendorClass_Index = DhcpVendorClass_DhcpServerIndex, DhcpVendorClass_VendorClassId; [\DhcpVendorClass] For a detailed description of the table, see Configuring the Vendor Class Identifier on page 212.
DHCP Option Table	
DHCP Option table `configure voip > dhcp option` [DhcpOption]	Defines additional DHCP Options that the device's DHCP server can use to service its DHCP clients. The format of the ini file table parameter is as follows: [DhcpOption] FORMAT DhcpOption_Index = DhcpOption_DhcpServerIndex, DhcpOption_Option, DhcpOption_Type, DhcpOption_Value, DhcpOption_ExpandValue; [\DhcpOption] For a detailed description of the table, see Configuring Additional DHCP Options on page 213.
DHCP Static IP Table	

Parameter	Description
DHCP Static IP table `configure voip > dhcp static-ip <index>` [DhcpStaticIP]	Defines static "reserved" IP addresses that the device's DHCP server allocates to specific DHCP clients defined by MAC address. The format of the ini file table parameter is as follows: [DhcpStaticIP] FORMAT DhcpStaticIP_Index = DhcpStaticIP_DhcpServerIndex, DhcpStaticIP_IPAddress, DhcpStaticIP_MACAddress; [\DhcpStaticIP] For a detailed description of the table, see Configuring Static IP Addresses for DHCP Clients on page 215.

57.2.8 NTP and Daylight Saving Time Parameters

The Network Time Protocol (NTP) and daylight saving time parameters are described in the table below.

Table 57-16: NTP and Daylight Saving Time Parameters

Parameter	Description
NTP Parameters **Note:** For more information on Network Time Protocol (NTP), see "Simple Network Time Protocol Support" on page 121.	
Primary NTP Server Address `primary-server` [NTPServerIP]	Defines the IP address (in dotted-decimal notation or as an FQDN) of the NTP server. The advantage of using an FQDN is that multiple IP addresses can be resolved from the DNS server, providing NTP server redundancy. The default IP address is 0.0.0.0 (i.e., internal NTP client is disabled).
Secondary NTP Server Address [NTPSecondaryServerIP]	Defines a second NTP server's address as an FQDN or an IP address (in dotted-decimal notation). This NTP is used for redundancy; if the primary NTP server fails, then this NTP server is used. The default IP address is 0.0.0.0.
NTP Update Interval `update-interval` [NTPUpdateInterval]	Defines the time interval (in seconds) that the NTP client requests for a time update. The default interval is 86400 (i.e., 24 hours). The range is 0 to 214783647. **Note:** It is not recommend to set the parameter to beyond one month (i.e., 2592000 seconds).
NTP Authentication Key Identifier `configure system > ntp > auth-key-id` [NtpAuthKeyId]	Defines the NTP authentication key identifier for authenticating NTP messages. The identifier must match the value configured on the NTP server. The NTP server may have several keys configured for different clients; this number identifies which key is used. The valid value is 1 to 65535. The default is 0 (i.e., no authentication is done).
NTP Authentication Secret Key `configure system > ntp > auth-key-md5` [ntpAuthMd5Key]	Defines the secret authentication key shared between the device (client) and the NTP server, for authenticating NTP messages. The valid value is a string of up to 32 characters. By default, no key is defined.

Parameter	Description
Regional Clock and Daylight Saving Time Parameters	
UTC Offset `utc-offset` [NTPServerUTCOffset]	Defines the Universal Time Coordinate (UTC) offset (in seconds) from the local time. The valid range is -43200 to 43200. The default is 0. **Note:** The offset setting is applied only on the hour. For example, if you configure the parameter at 15:42, the device applies the setting only at 16:00.
Daylight Saving Time `summer-time` [DayLightSavingTimeEnable]	Enables daylight saving time (DST). ▪ **[0]** Disable (default) ▪ **[1]** Enable
Start Time / Day of Month Start `start` [DayLightSavingTimeStart]	Defines the date and time when DST begins. This value can be configured using any of the following formats: ▪ Day of year - *mm:dd:hh:mm*, where: ✓ *mm* denotes month ✓ *dd* denotes date of the month ✓ *hh* denotes hour ✓ *mm* denotes minutes For example, "05:01:08:00" denotes daylight saving starting from May 1 at 8 A.M. ▪ Day of month - *mm:day/wk:hh:mm*, where: ✓ *mm* denotes month (e.g., 04) ✓ *day* denotes day of week (e.g., FRI) ✓ *wk* denotes week of the month (e.g., 03) ✓ *hh* denotes hour (e.g., 23) ✓ *mm* denotes minutes (e.g., 10) For example, "04:FRI/03:23:00" denotes Friday, the third week of April, at 11 P.M. The week field can be 1-5, where 5 denotes the last occurrence of the specified day in the specified month. For example, "04:FRI/05:23:00" denotes the last Friday of April, at 11 P.M.
End Time / Day of Month End `end` [DayLightSavingTimeEnd]	Defines the date and time when DST ends. For a description of the format of this value, see the DayLightSavingTimeStart parameter.
Offset `offset` [DayLightSavingTimeOffset]	Defines the DST offset (in minutes). The valid range is 0 to 120. The default is 60. **Note:** The offset setting is applied only on the hour. For example, if you configure the parameter at 15:42, the device applies the setting only at 16:00.

57.3 Debugging and Diagnostics Parameters

This subsection describes the device's debugging and diagnostic parameters.

57.3.1 General Parameters

The general debugging and diagnostic parameters are described in the table below.

Table 57-17: General Debugging and Diagnostic Parameters

Parameter	Description
[EnableDiagnostics]	Determines the method for verifying correct functioning of the different hardware components on the device. On completion of the check and if the test fails, the device sends information on the test results of each hardware component to the Syslog server. ▪ **[0]** = (Default) Rapid and Enhanced self-test mode. ▪ **[1]** = Detailed self-test mode (full test of DSPs, PCM, Switch, LAN, PHY and Flash). ▪ **[2]** = A quicker version of the Detailed self-test mode (full test of DSPs, PCM, Switch, LAN, PHY, but partial test of Flash). **Note:** For the parameter to take effect, a device reset is required.
Delay After Reset [sec] `delay-after-reset` **[GWAppDelayTime]**	Defines the time interval (in seconds) that the device's operation is delayed after a reset. The valid range is 0 to 45. The default is 7 seconds. **Note:** This feature helps overcome connection problems caused by some LAN routers or IP configuration parameters' modifications by a DHCP server.
[EnableAutoRAITransmitBER]	Enables the device to send a remote alarm indication (RAI) when the bit error rate (BER) is greater than 0.001. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Ignore BRI LOS Alarm `ignore-bri-los-alarm` [IgnoreBRILOSAlarm]	Enables the device to ignore LOS alarms received from the BRI user-side trunk and attempts to make a call (relevant for IP-to-Tel calls). ▪ [0] Disable ▪ [1] Enable (default) **Note:** The parameter is applicable only to BRI interfaces.

57.3.2 SIP Test Call Parameters

The SIP Signaling Test Call parameters are described in the table below.

Table 57-18: SIP Test Call Parameters

Parameter	Description
Test Call DTMF String `testcall-dtmf-string` [TestCallDtmfString]	Defines the DTMF tone that is played for answered test calls (incoming and outgoing). The DTMF string can be up to 15 strings. The default is "3212333". If no string is defined (empty), DTMF is not played.
Test Call ID `testcall-id` [TestCallID]	Defines the test call prefix number (*ID*) of the simulated phone on the device. Incoming calls received with this called prefix number are identified as test calls. This can be any string of up to 15 characters. By default, no number is defined. **Notes:** ▪ The parameter is only for testing incoming calls destined to this prefix number. ▪ This feature is applicable to all applications (Gateway and SBC).

Parameter	Description
Test Call Table	
Test Call Table `configure system > test-call > test-call-table` [Test_Call]	Defines Test Call rules. [Test_Call] FORMAT Test_Call_Index = Test_Call_EndpointURI, Test_Call_CalledURI, Test_Call_RouteBy, Test_Call_IPGroupName, Test_Call_DestAddress, Test_Call_DestTransportType, Test_Call_SIPInterfaceName, Test_Call_ApplicationType, Test_Call_AutoRegister, Test_Call_UserName, Test_Call_Password, Test_Call_CallParty, Test_Call_MaxChannels, Test_Call_CallDuration, Test_Call_CallsPerSecond, Test_Call_TestMode, Test_Call_TestDuration, Test_Call_Play, Test_Call_ScheduleInterval, Test_Call_QOEProfile, Test_Call_BWProfile; [\Test_Call] For a description of the table, see "Configuring Test Call Endpoints" on page 793.

57.3.3 Syslog, CDR and Debug Parameters

The Syslog, CDR and debug parameters are described in the table below.

Table 57-19: Syslog, CDR and Debug Parameters

Parameter	Description
Enable Syslog `syslog` [EnableSyslog]	Determines whether the device sends logs and error messages (e.g., CDRs) generated by the device to a Syslog server. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Notes:** ▪ If you enable Syslog, you must enter an IP address of the Syslog server (using the SyslogServerIP parameter). ▪ Syslog messages may increase the network traffic. ▪ To configure Syslog SIP message logging levels, use the GwDebugLevel parameter.
Syslog Server IP Address `syslog-ip` [SyslogServerIP]	Defines the IP address (in dotted-decimal notation) of the computer on which the Syslog server is running. The Syslog server is an application designed to collect the logs and error messages generated by the device. The default IP address is 0.0.0.0.
Syslog Server Port `syslog-port` [SyslogServerPort]	Defines the UDP port of the Syslog server. The valid range is 0 to 65,535. The default port is 514.
CDR Server IP Address `cdr-srvr-ip-adrr` [CDRSyslogServerIP]	Defines the destination IP address to where CDR logs are sent. The default value is a null string, which causes CDR messages to be sent with all Syslog messages to the Syslog server. **Notes:** ▪ The CDR messages are sent to UDP port 514 (default Syslog port). ▪ This mechanism is active only when Syslog is enabled (i.e., the parameter EnableSyslog is set to 1).

Parameter	Description
CDR Report Level `cdr-report-level` [CDRReportLevel]	Enables media and signaling-related CDRs to be sent to a Syslog server and determines the call stage at which they are sent. ▪ **[0]** None = (Default) CDRs are not used. ▪ **[1]** End Call = CDR is sent to the Syslog server at the end of each call. ▪ **[2]** Start & End Call = CDR report is sent to Syslog at the start and end of each call. ▪ **[3]** Connect & End Call = CDR report is sent to Syslog at connection and at the end of each call. ▪ **[4]** Start & End & Connect Call = CDR report is sent to Syslog at the start, at connection, and at the end of each call. **Notes:** ▪ For the SBC application, the parameter enables only signaling-related CDRs. To enable media-related CDRs for SBC calls, use the MediaCDRReportLevel parameter. ▪ The CDR Syslog message complies with RFC 3164 and is identified by: Facility = 17 (local1) and Severity = 6 (Informational). ▪ This mechanism is active only when Syslog is enabled (i.e., the parameter EnableSyslog is set to 1).
Media CDR Report Level [MediaCDRReportLevel]	Enables media-related CDRs of SBC calls to be sent to a Syslog server and determines the call stage at which they are sent. ▪ [0] None = (Default) No media-related CDR is sent. ▪ [1] End Media = Sends a CDR only at the end of the call. ▪ [2] Start & End Media = Sends a CDR once the media starts. In some calls it may only be after the call is established, but in other calls the media may start at ringback tone. A CDR is also sent upon termination (end) of the media in the call. ▪ [3] Update & End Media = Sends a CDR when an update occurs in the media of the call. For example, a call starts and a ringback tone occurs, a re-INVITE is sent for a fax call and as a result, a CDR with the MediaReportType field set to "Update" is sent, as the media was changed from voice to T.38. A CDR is also sent upon termination (end) of the media in the call. ▪ [4] Start & End & Update Media = Sends a CDR at the start of the media, upon an update in the media (if occurs), and at the end of the media. **Note:** To enable CDR generation as well as enable signaling-related CDRs, use the CDRReportLevel parameter.
`configure voip > sip-definition settings > time-zone-format` [TimeZoneFormat]	Defines the time zone that is displayed with the timestamp in CDRs. The timestamp appears in the CDR fields "Setup Time", "Connect Time", and "Release Time". The valid value is a string of up to six characters. The default is UTC. For example, if you configure the parameter TimeZoneFormat = GMT+11, the timestamp in CDRs are generated with the following time zone display: `17:47:45.411  `**`GMT+11`**` Sun Jan 03 2018` **Note:** The time zone is only for display purposes; it does not configure the actual time zone.

Parameter	Description
Local Storage Max File Size `configure voip > services cdr > cdr-local-max-file-size` [CDRLocalMaxFileSize]	Defines the size (in kilobytes) of each stored CDR file. Once the file size is reached, the device creates a new file for subsequent CDRs, and so on. The valid value is 100 to 10000. The default is 1024.
Local Storage Max Number Of Files `configure voip > services cdr > cdr-local-max-files` [CDRLocalMaxNomOfFiles]	Defines the maximum number of stored CDR files. If the maximum number is reached, the device replaces (overwrites) the oldest created file with a subsequent new file, and so on. The valid value is 2 to 4096. The default is 5.
Local Storage File Creation Interval `configure voip > services cdr > cdr-local-interval` [CDRLocalInterval]	Defines how often (in minutes) the device creates a new CDR file. For example, if configured to 60, it creates a new file every hour. This occurs even if the maximum configured file size has not been reached (see the CDRLocalMaxFileSize parameter). However, if the maximum configured file size has been reached and the interval configured by the parameter has not been reached, a new CDR file is created. The valid value is 2 to 1440. The default is 60.
configure system > cdr > non-call-cdr-rprt [EnableNonCallCdr]	Enables creation of CDR messages for non-call SIP dialogs (such as SUBSCRIBE, OPTIONS, and REGISTER). ▪ [0] = (Default) Disable ▪ [1] = Enable **Note:** The parameter is applicable only to the SBC application.
Debug Level `configure system/logging/debug-level` [GwDebugLevel]	Enables Syslog debug reporting and logging level. ▪ **[0]** No Debug = (Default) Debug is disabled and Syslog messages are not sent. ▪ **[1]** Basic = Sends debug logs of incoming and outgoing SIP messages. ▪ **[5]** Detailed = Sends debug logs of incoming and outgoing SIP message as well as many other logged processes.
Syslog Optimization `configure system/logging/syslog-optimization` [SyslogOptimization]	Enables the device to accumulate and bundle multiple debug messages into a single UDP packet and then send it to a Syslog server. The benefit of this feature is that it reduces the number of UDP Syslog packets, thereby improving (optimizing) CPU utilization. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** The size of the bundled message is configured by the MaxBundleSyslogLength parameter.
`mx-syslog-lgth` [MaxBundleSyslogLength]	Defines the maximum size (in bytes) threshold of logged Syslog messages bundled into a single UDP packet, after which they are sent to a Syslog server. The valid value range is 0 to 1220 (where 0 indicates that no bundling occurs). The default is 1220. **Note:** The parameter is applicable only if the GWDebugLevel parameter is enabled.

Parameter	Description
Syslog CPU Protection `configure system/logging/syslog-cpu-protection` [SyslogCpuProtection]	Enables the protection of the device's CPU resources during debug reporting, ensuring voice traffic is unaffected. If CPU resources drop (i.e., high CPU usage) to a critical level (threshold), the device automatically lowers the debug level to free up CPU resources that were required for the previous debug-level functionality. When sufficient CPU resources become available again, the device increases the debug level. The threshold is configured by the 'Debug Level High Threshold' parameter (see below). ▪ **[0]** Disable ▪ **[1]** Enable (default)
Debug Level High Threshold `debug-level-high-threshold` [DebugLevelHighThreshold]	Defines the threshold (in percentage) for automatically switching to a different debug level, depending on CPU usage. The parameter is applicable only if the 'Syslog CPU Protection' parameter is enabled. The valid value is 0 to 100. The default is 90. The debug level is changed upon the following scenarios: ▪ CPU usage equals threshold: Debug level is reduced one level. ▪ CPU usage is at least 5% greater than threshold: Debug level is reduced another level. ▪ CPU usage is 5 to 19% less than threshold: Debug level is increased by one level. ▪ CPU usage is at least 20% less than threshold: Debug level is increased by another level. For example, assume that the threshold is set to 70% and the Debug Level to Detailed (5). When CPU usage reaches 70%, the debug level is reduced to Basic (1). When CPU usage increases by 5% or more than the threshold (i.e., greater than 75%), the debug level is disabled - No Debug (0). When the CPU usage decreases to 5% less than the threshold (e.g., 65%), the debug level is increased to Basic (1). When the CPU usage decreases to 20% less than the threshold (e.g., 50%), the debug level changes to Detailed (5). **Note:** The device does not increase the debug level to a level that is higher than what you configured for the 'Debug Level' parameter.
Syslog Facility Number [SyslogFacility]	Defines the Facility level (0 through 7) of the device’s Syslog messages, according to RFC 3164. This allows you to identify Syslog messages generated by the device. This is useful, for example, if you collect the device’s and other equipments’ Syslog messages, at one single server. The device’s Syslog messages can easily be identified and distinguished from other Syslog messages by its Facility level. Therefore, in addition to filtering Syslog messages according to IP address, the messages can be filtered according to Facility level. ▪ **[16]** = (Default) local use 0 (local0) ▪ **[17]** = local use 1 (local1) ▪ **[18]** = local use 2 (local2) ▪ **[19]** = local use 3 (local3) ▪ **[20]** = local use 4 (local4) ▪ **[21]** = local use 5 (local5) ▪ **[22]** = local use 6 (local6) ▪ **[23]** = local use 7 (local7)

Parameter	Description
CDR Syslog Sequence Number `cdr-seq-num` [CDRSyslogSeqNum]	Enables or disables the inclusion of the sequence number (S=) in CDR Syslog messages. ▪ **[0]** Disable ▪ **[1]** Enable (default)
Activity Types to Report via Activity Log Messages `config-system > logging > activity-log` [ActivityListToLog]	Defines the operations (activities) performed in the Web interface that are reported to a Syslog server. ▪ **[pvc]** Parameters Value Change = Changes made on-the-fly to parameters and tables, and Configuration file load. Note that the *ini* file parameter, EnableParametersMonitoring can also be used to set this option. ▪ **[afl]** Auxiliary Files Loading = Loading of Auxiliary files. ▪ **[dr]** Device Reset = Resetting of the device through the Maintenance Actions page. **Note:** For this option to take effect, a device reset is required. ▪ **[fb]** Flash Memory Burning = Saving configuration with burn to flash (in the Maintenance Actions page). ▪ **[swu]** Device Software Update = Software updates (i.e., loading of cmp file) through the Software Upgrade Wizard. ▪ **[ard]** Access to Restricted Domains = Access to restricted Web pages: ✓ (1) ini parameters (AdminPage) ✓ (2) General Security Settings ✓ (3) Configuration File ✓ ✓ (5) Software Upgrade Key Status ✓ ✓ (7) Web & Telnet Access List ✓ (8) Web User Accounts ▪ **[naa]** Non-Authorized Access = Attempts to log in to the Web interface with a false or empty username or password. ▪ **[spc]** Sensitive Parameters Value Change = Changes made to "sensitive" parameters: ✓ (1) IP Address ✓ (2) Subnet Mask ✓ (3) Default Gateway IP Address ✓ (4) ActivityListToLog ▪ **[ll]** Login and Logout = Web login and logout attempts. ▪ **[cli]** = CLI commands entered by the user. ▪ **[ae]** Action Executed = Logs user actions that are not related to parameter changes. The actions can include, for example, file uploads, file delete, lock-unlock maintenance actions, LDAP clear cache, register-unregister, and start-stop trunk. In the Web, these actions are typically done by clicking a button (e.g., the **LOCK** button). **Note:** For the *ini* file parameter, enclose values in single quotation marks, for example: ActivityListToLog = 'pvc', 'afl', 'dr', 'fb', 'swu', 'ard', 'naa', 'spc'.
Activity Trap `activity-trap` [EnableActivityTrap]	Enables the device to send an SNMP trap to notify of Web user activities in the Web interface. The activities to report are configured by the ActivityListToLog parameter. ▪ [0] Disable (default) ▪ [1] Enable

Parameter	Description
[EnableParametersMonitoring]	Enables the monitoring, through Syslog messages, of parameters that are modified on-the-fly. ▪ **[0]** = (Default) Disable ▪ **[1]** = Enable
`isdn-facility-trace` [FacilityTrace]	Enables ISDN traces of Facility Information Elements (IE) for ISDN call diagnostics. This allows you to trace all the parameters contained in the Facility IE and view them in the Syslog. ▪ [0] Disable (default) ▪ [1] Enable **Note:** For this feature to be functional, the GWDebugLevel parameter must be enabled (i.e., set to at least level 1).
Debug Recording Destination IP `configure system > logging > dbg-rec-dest-ip` [DebugRecordingDestIP]	Defines the IP address of the server for capturing debug recording.
Debug Recording Destination Port `configure system > logging > dbg-rec-dest-port` [DebugRecordingDestPort]	Defines the UDP port of the server for capturing debug recording. The default is 925.
Enable Core Dump [EnableCoreDump]	Enables the automatic generation of a Core Dump file upon a device crash. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** For the parameter to take effect, a device reset is required.
Core Dump Destination IP [CoreDumpDestIP]	Defines the IP address of the remote server where you want the device to send the Core Dump file. By default, no IP address is defined.
Logging Filters Table	
Logging Filters Table `configure system > logging > logging-filters` [LoggingFilters]	The table defines logging filtering rules for Syslog messages and debug recordings. The format of the ini file table parameter is: [LoggingFilters] FORMAT LoggingFilters_Index = LoggingFilters_FilterType, LoggingFilters_Value, LoggingFilters_LogDestination, LoggingFilters_CaptureType, LoggingFilters_Mode; [\LoggingFilters] For a detailed description of the table, see "Configuring Log Filter Rules" on page 769.
Gateway CDR Format Table	
Gateway CDR Format Table `configure voip > services cdr > cdr-format gw-cdr-format`	The table defines CDR customization rules for Gateway calls. . The format of the ini file table parameter is: [GWCDRFormat]

Parameter	Description
[GWCDRFormat]	FORMAT GWCDRFormat_Index = GWCDRFormat_CDRType, GWCDRFormat_ColumnType, GWCDRFormat_Title, GWCDRFormat_RadiusType, GWCDRFormat_RadiusID; [\GWCDRFormat] For a detailed description of the table, see Customizing CDRs for Gateway Calls on page 751.
SBC CDR Format Table	
SBC CDR Format Table `configure voip > services cdr > cdr-format sbc-cdr-format` [SBCCDRFormat]	The table defines CDR customization rules for SBC calls. The format of the ini file table parameter is: [SBCCDRFormat] FORMAT SBCCDRFormat_Index = SBCCDRFormat_CDRType, SBCCDRFormat_ColumnType, SBCCDRFormat_Title, SBCCDRFormat_RadiusType, SBCCDRFormat_RadiusID; [\SBCCDRFormat] For a detailed description of the table, see Customizing CDRs for SBC Calls on page 754.

57.3.4 Resource Allocation Indication Parameters

The Resource Allocation Indication (RAI) parameters are described in the table below.

Table 57-20: RAI Parameters

Parameter	Description
[EnableRAI]	Enables Resource Available Indication (RAI) alarm generation if the device's busy endpoints exceed a user-defined threshold, configured by the RAIHighThreshold parameter. When enabled and the threshold is crossed, the device sends the SNMP trap, acBoardCallResourcesAlarm. ▪ **[0]** = (Default) Disable ▪ **[1]** = Enable **Note:** For the parameter to take effect, a device reset is required.
[RAIHighThreshold]	Defines the high threshold percentage of total calls that are active (busy endpoints). When the percentage of the device's busy endpoints exceeds this high threshold, the device sends the SNMP acBoardCallResourcesAlarm alarm trap with a 'major' alarm status. The range is 0 to 100. The default is 90. **Note:** The percentage of busy endpoints is calculated by dividing the number of busy endpoints by the total number of “enabled” endpoints (trunks are physically connected and synchronized with no alarms and endpoints are defined in the Trunk Group table).
[RAILowThreshold]	Defines the low threshold percentage of total calls that are active (busy endpoints). When the percentage of the device's busy endpoints falls below this low threshold, the device sends an SNMP acBoardCallResourcesAlarm alarm trap with a 'cleared' alarm status. The range is 0 to 100%. The default is 90%.

Parameter	Description
[RAILoopTime]	Defines the time interval (in seconds) that the device periodically checks call resource availability. The valid range is 1 to 200. The default is 10.

57.3.5 PacketSmart Parameters

The PacketSmart parameters are described in the table below. For more information on PacketSmart, see 'Configuring PacketSmart for Network Monitoring' on page 719.

PacketSmart Parameters

Parameter	Description
PacketSmart Agent Mode `configure system > packetsmart enable` [PacketSmartAgentMode]	Enables the embedded PacketSmart agent. ▪ [0] Disable (Default) ▪ [1] Enable **Note:** For the parameter to take effect, a device reset is required.
PacketSmart IP Address `configure system > packetsmart server address` [PacketSmartIpAddress]	Defines the IP address of the PacketSmart server with which the PacketSmart agent communicates. The default is 0.0.0.0.
PacketSmart IP Address Port `configure system > packetsmart server address port` [PacketSmartIpAddressPort]	Defines the TCP port of the PacketSmart server to which the PacketSmart agent connects. The default is 80.
Monitoring Interface `configure system > packetsmart monitor voip interface-if` [PacketSmartMonitorInterface]	Assigns an IP network interface (configured in the IP Interface table) that handles the voice traffic. **Note:** For the parameter to take effect, a device reset is required.
Network Interface `configure system > packetsmart network voip interface-if` [PacketSmartNetworkInterface]	Assigns an IP network interface (configured in the IP Interface table) for communicating with the PacketSmart server. This is typically the OAMP interface. **Note:** For the parameter to take effect, a device reset is required.

57.4 Security Parameters

This subsection describes the device's security parameters.

57.4.1 General Security Parameters

The general security parameters are described in the table below.

Table 57-21: General Security Parameters

Parameter	Description
Firewall Table	
Internal Firewall Parameters `configure voip > access-list` [AccessList]	The table defines the device's access list (firewall), which defines network traffic filtering rules. The format of the ini file table parameter is: [AccessList] FORMAT AccessList_Index = AccessList_Source_IP, AccessList_Source_Port, AccessList_PrefixLen, AccessList_Source_Port, AccessList_Start_Port, AccessList_End_Port, AccessList_Protocol, AccessList_Use_Specific_Interface, AccessList_Interface_ID, AccessList_Packet_Size, AccessList_Byte_Rate, AccessList_Byte_Burst, AccessList_Allow_Type; [\AccessList] For example: AccessList 10 = mgmt.customer.com, , , 32, 0, 80, tcp, 1, OAMP, 0, 0, 0, allow; AccessList 22 = 10.4.0.0, , , 16, 4000, 9000, any, 0, , 0, 0, 0, block; In the example above, Rule #10 allows traffic from the host ‘mgmt.customer.com’ destined to TCP ports 0 to 80 on interface OAMP (OAMP). Rule #22 blocks traffic from the subnet 10.4.xxx.yyy destined to ports 4000 to 9000. For a detailed description of the table, see "Configuring Firewall Settings" on page 159.
Media Latching	
Inbound Media Latch Mode `inbound-media-latch-mode` [InboundMediaLatchMode]	Enables the Media Latching feature. ▪ [0] Strict = Device latches onto the first original stream (IP address:port). It does not latch onto any other stream during the session. ▪ [1] Dynamic = (Default) Device latches onto the first stream. If it receives at least a minimum number of consecutive packets (configured by New<media type>StreamPackets) from a different source(s) and the device has not received packets from the current stream for a user-defined period (TimeoutToRelatch<media type>Msec), it latches onto the next packet received from any other stream. If other packets of a different media type are received from the new stream, based on IP address and SSRC for RTCP/RTP and based on IP address only for T.38, the packet is accepted immediately. Note: If a packet from the original (first latched onto) IP address:port is received at any time, the device latches onto this stream. ▪ [2] Dynamic-Strict = Device latches onto the first stream. If it receives at least a minimum number of consecutive packets (configured by New<media type>StreamPackets) all from the same source which is different to the first stream and the device has not received packets from the current stream for a user-defined period (TimeoutToRelatch<media type>Msec), it latches onto the next packet received from any other stream.

Parameter	Description
	If other packets of different media type are received from the new stream based on IP address and SSRC for RTCP and based on IP address only for T.38, the packet is accepted immediately. Note: If a packet from the original (first latched onto) IP address:port is received at any time, the device latches onto this stream. ▪ [3] Strict-On-First = Typically used for NAT, where the correct IP address:port is initially unknown. The device latches onto the stream received in the first packet. The device does not change this stream unless a packet is later received from the original source.
New RTP Stream Packets [NewRtpStreamPackets]	Defines the minimum number of continuous RTP packets received by the device's channel to allow latching onto the new incoming stream. The valid range is 0 to 20. The default is 3. If set to 0, the device is left exposed to attacks against multiple packet streams.
New RTCP Stream Packets [NewRtcpStreamPackets]	Defines the minimum number of continuous RTCP packets received by the device's channel to allow latching onto the new incoming stream. The valid range is 0 to 20. The default is 3. If set to 0, the device is left exposed to attacks against multiple packet streams.
New SRTP Stream Packets [NewSRTPStreamPackets]	Defines the minimum number of continuous SRTP packets received by the device's channel to allow latching onto the new incoming stream. The valid range is 0 to 20. The default is 3. If set to 0, the device is left exposed to attacks against multiple packet streams.
New SRTCP Stream Packets [NewSRTCPStreamPackets]	Defines the minimum number of continuous SRTCP packets received by the device's channel to allow latching onto the new incoming stream. The valid range is 0 to 20. The default is 3. If set to 0, the device is left exposed to attacks against multiple packet streams.
Timeout To Relatch RTP [TimeoutToRelatchRTPMsec]	Defines a period (msec) during which if no packets are received from the current RTP session, the channel can re-latch onto another stream. The valid range is any value from 0. The default is 200.
Timeout To Relatch SRTP [TimeoutToRelatchSRTPMsec]	Defines a period (msec) during which if no packets are received from the current SRTP session, the channel can re-latch onto another stream. The valid range is any value from 0. The default is 200.
Timeout To Relatch Silence [TimeoutToRelatchSilenceMsec]	Defines a period (msec) during which if no packets are received from the current RTP/SRTP session and the channel is in silence mode, the channel can re-latch onto another stream. The valid range is any value from 0. The default is 200.
Timeout To Relatch RTCP [TimeoutToRelatchRTCPMsec]	Defines a period (msec) during which if no packets are received from the current RTCP session, the channel can re-latch onto another RTCP stream. The valid range is any value from 0. The default is 10,000.
Fax Relay Rx/Tx Timeout [FaxRelayTimeoutSec]	Defines a period (sec) during which if no T.38 packets are received or sent from the current T.38 fax relay session, the channel can re-latch onto another stream.

Parameter	Description
	The valid range is 0 to 255. The default is 10.

57.4.2 HTTPS Parameters

The Secure Hypertext Transport Protocol (HTTPS) parameters are described in the table below.

Table 57-22: HTTPS Parameters

Parameter	Description
Secured Web Connection (HTTPS) `secured-connection` [HTTPSOnly]	Determines the protocol used to access the Web interface. ▪ **[0]** HTTP and HTTPS (default). ▪ **[1]** HTTPs Only = Unencrypted HTTP packets are blocked. **Note:** For the parameter to take effect, a device reset is required.
`https-port` [HTTPSPort]	Defines the local Secured HTTPS port of the device. The parameter allows secure remote device Web management from the LAN. To enable secure Web management from the LAN, configure the desired port. The valid range is 1 to 65535 (other restrictions may apply within this range). The default port is 443. **Note:** For the parameter to take effect, a device reset is required.
HTTPS Cipher String `https-cipher-string` [HTTPSCipherString]	Defines the Cipher string for HTTPS (in OpenSSL cipher list format). For the valid range values, refer to URL http://www.openssl.org/docs/apps/ciphers.html. The default is 'RC4:EXP' (Export encryption algorithms). For example, use 'ALL' for all ciphers suites (e.g., for ARIA encryption for TLS). The only ciphers available are RC4 and DES, and the cipher bit strength is limited to 56 bits. **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ If the installed Software License Key includes the Strong Encryption feature, the default of the parameter is changed to 'RC4:EXP', enabling RC-128bit encryption. ▪ The value 'ALL' can be configured only if the installed Software License Key includes the Strong Encryption feature.
Requires Client Certificates for HTTPS connection `req-client-cert` [HTTPSRequireClientCertificate]	Enables the requirement of client certificates for HTTPS connection. ▪ **[0]** Disable = (Default) Client certificates are not required. ▪ **[1]** Enable = Client certificates are required. The client certificate must be preloaded to the device and its matching private key must be installed on the managing PC. Time and date must be correctly set on the device for the client certificate to be verified. **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ For a description on implementing client certificates, see "TLS for Remote Device Management" on page 118.

57.4.3 SRTP Parameters

The Secure Real-Time Transport Protocol (SRTP) parameters are described in the table below.

Table 57-23: SRTP Parameters

Parameter	Description
Media Security `media-security-enable` [EnableMediaSecurity]	Enables Secure Real-Time Transport Protocol (SRTP). ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** For the parameter to take effect, a device reset is required.
Media Security Behavior `media-sec-bhvior` [MediaSecurityBehaviour]	Global parameter that defines the handling of SRTP (when the EnableMediaSecurity parameter is set to 1). You can also configure this functionality per specific calls, using IP Profiles (IpProfile_MediaSecurityBehaviour). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see Configuring IP Profiles on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile. **Note:** The parameter is applicable only to the Gateway application.
Master Key Identifier (MKI) Size `SRTP-tx-packet-MKI-size` [SRTPTxPacketMKISize]	Global parameter that defines the size (in bytes) of the Master Key Identifier (MKI) in SRTP Tx packets. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_MKISize). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
Symmetric MKI Negotiation `symmetric-mki` [EnableSymmetricMKI]	Global parameter that enables symmetric MKI negotiation. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_EnableSymmetricMKI). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
Offered SRTP Cipher Suites `offer-srtp-cipher` [SRTPofferedSuites]	Defines the offered crypto suites (cipher encryption algorithms) for SRTP. ▪ **[0]** All = (Default) All available crypto suites.

Parameter	Description
	▪ **[1]** AES-CM-128-HMAC-SHA1-80 = device uses AES-CM encryption with a 128-bit key and HMAC-SHA1 message authentication with a 80-bit tag. ▪ **[2]** AES-CM-128-HMAC-SHA1-32 = device uses AES-CM encryption with a 128-bit key and HMAC-SHA1 message authentication with a 32-bit tag. ▪ [4] ARIA-CM-128-HMAC-SHA1-80 = device uses ARIA encryption algorithm with a 128-bit key and HMAC-SHA1 message authentication with a 32-bit tag. ▪ [8] ARIA-CM-192-HMAC-SHA1-80 = device uses ARIA encryption algorithm with a 192-bit key and HMAC-SHA1 message authentication with a 32-bit tag. **Note**s: ▪ For enabling ARIA encryption, use the AriaProtocolSupport parameter. ▪ The parameter also affects the selection of the crypto in the device's answer. For example, if the device receives an offer with two crypto lines containing HMAC_SHA1_80 and HMAC_SHA_32, it uses the HMAC_SHA_32 key in its SIP 200 OK response if the parameter is set to 2.
`configure voip > sbc general-setting > sbc-dtls-mtu` [SbcDtlsMtu]	Defines the maximum transmission unit (MTU) size for the DTLS handshake. The device does not attempt to send handshake packets that are larger than the configured value. Adjusting the MTU is useful when there are network constraints on the size of packets that can be sent. The valid value range is 228 to 1500. The default is 1500. **Note:** The parameter is applicable only to the SBC application.

Parameter	Description
Aria Protocol Support `ARIA-protocol-support` [AriaProtocolSupport]	Enables ARIA algorithm cipher encryption for SRTP. This is an alternative option to the existing support for the AES algorithm. ARIA is a symmetric key block cipher algorithm standard developed by the Korean National Security Research Institute. ▪ [0] Disable (default) ▪ [1] Enable **Notes:** ▪ To configure the ARIA bit-key encryption size (128 or 192 bit) with HMAC SHA-1 cryptographic hash function, use the SRTPofferedSuites parameter. ▪ The ARIA feature is available only if the device is installed with a Software License Key that includes this feature. For installing a Software License Key, see Software License Key on page 668.
Authentication On Transmitted RTP Packets `RTP-authentication-disable-tx` [RTPAuthenticationDisableTx]	Enables authentication on transmitted RTP packets in a secured RTP session. ▪ **[0]** Enable (default) ▪ **[1]** Disable
Encryption On Transmitted RTP Packets `RTP-encryption-disable-tx` [RTPEncryptionDisableTx]	Enables encryption on transmitted RTP packets in a secured RTP session. ▪ **[0]** Enable (default) ▪ **[1]** Disable
Encryption On Transmitted RTCP Packets `RTCP-encryption-disable-tx` [RTCPEncryptionDisableTx]	Enables encryption on transmitted RTCP packets in a secured RTP session. ▪ **[0]** Enable (default) ▪ **[1]** Disable
SRTP Tunneling Authentication for RTP `configure voip > media security > srtp-tnl-vld-rtp-auth` [SRTPTunnelingValidateRTPRxAuthentication]	Enables validation of SRTP tunneling authentication for RTP. ▪ [0] Disable = (Default) The device does not perform any validation and forwards the packets as is. ▪ [1] Enable = The device validates the packets (e.g., sequence number) and if successful, forwards the packets. If validation fails, it drops the packets. **Note:** ▪ The parameter is applicable only to SRTP-to-SRTP calls and when both endpoints use the same authentication keys. ▪ The parameter is applicable only to the SBC application.
SRTP Tunneling Authentication for RTCP `configure voip > media security > srtp-tnl-vld-rtcp-auth` [SRTPTunnelingValidateRTCPRxAuthentication]	Enables validation of SRTP tunneling authentication for RTCP. ▪ [0] Disable = (Default) The device does not perform any validation and forwards the packets as is.

Parameter	Description
	▪ [1] Enable = The device validates the packets (e.g., sequence number) and if successful, forwards the packets. If validation fails, it drops the packets. **Note:** ▪ The parameter is applicable only to SRTP-to-SRTP calls and when both endpoints use the same authentication keys. ▪ The parameter is applicable only to the SBC application.
`srtp-state-behavior-mode` [ResetSRTPStateUponRekey]	Global parameter that enables synchronization of the SRTP state between the device and a server when a new SRTP key is generated upon a SIP session expire. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_ResetSRTPStateUponRekey). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.

57.4.4 TLS Parameters

The Transport Layer Security (TLS) parameters are described in the table below.

Table 57-24: TLS Parameters

Parameter	Description
TLS Contexts Table	
TLS Contexts Table `configure system > tls #` [TLSContexts]	Defines SSL/TLS certificates. The format of the ini file table parameter is as follows: [TLSContexts] FORMAT TLSContexts_Index = TLSContexts_Name, TLSContexts_TLSVersion, TLSContexts_ServerCipherString, TLSContexts_ClientCipherString, TLSContexts_OcspEnable, TLSContexts_OcspServerPrimary, TLSContexts_OcspServerSecondary, TLSContexts_OcspServerPort, TLSContexts_OcspDefaultResponse; [\TLSContexts] For a detailed description of the table, see "Configuring TLS Certificate Contexts" on page 107.
TLS Client Re-Handshake Interval `tls-re-hndshk-int` [TLSReHandshakeInterval]	Defines the time interval (in minutes) between TLS Re-Handshakes initiated by the device. The interval range is 0 to 1,500 minutes. The default is 0 (i.e., no TLS Re-Handshake).
TLS Mutual Authentication	Defines the device's mode of operation regarding mutual authentication and certificate verification for TLS connections.

Parameter	Description
[SIPSRequireClientCertificate]	▪ **[0]** Disable = (Default) ✓ Device acts as a client: Verification of the server's certificate depends on the VerifyServerCertificate parameter. ✓ Device acts as a server: The device does not request the client certificate. ▪ **[1]** Enable = ✓ Device acts as a client: Verification of the server certificate is required to establish the TLS connection. ✓ Device acts as a server: The device requires the receipt and verification of the client certificate to establish the TLS connection. **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ This feature can be configured per SIP Interface (see "Configuring SIP Interfaces" on page 333). ▪ The SIPS certificate files can be changed using the parameters HTTPSCertFileName and HTTPSRootFileName.
Peer Host Name Verification Mode [PeerHostNameVerificationMode]	Determines whether the device verifies the Subject Name of a remote certificate when establishing TLS connections. ▪ **[0]** Disable (default). ▪ **[1]** Server Only = Verify Subject Name only when acting as a client for the TLS connection. ▪ **[2]** Server & Client = Verify Subject Name when acting as a server or client for the TLS connection. When the device receives a remote certificate and the parameter is not disabled, the IP address from which the certificate is received is compared with the addresses defined for the Proxy Sets. If no Proxy Set with the source address is found, the connection is refused. Otherwise, the value of SubjectAltName field in the certificate is compared with the addresses\ DNS Names of the classified Proxy Set. If a match is found for any of the configured Proxies, the TLS connection is established. The comparison is performed if the SubjectAltName is either a DNS name (DNSName) or an IP address. If no match is found and the SubjectAltName is marked as 'critical', the TLS connection is not established. If DNSName is used, the certificate can also use wildcards ('*') to replace parts of the domain name. If the SubjectAltName is not marked as 'critical' and there is no match, the CN value of the SubjectName field is compared with the parameter TLSRemoteSubjectName. If a match is found, the connection is established; otherwise, the connection is terminated. **Note:** If you set the parameter to [2] (Server & Client), for this functionality to operate you also need to set the SIPSRequireClientCertificate parameter to [1] (Enable).
TLS Client Verify Server Certificate `tls-vrfy-srvr-cert` [VerifyServerCertificate]	Determines whether the device, when acting as a client for TLS connections, verifies the Server certificate. The certificate is verified with the Root CA information. ▪ **[0]** Disable (default) ▪ **[1]** Enable

Parameter	Description
	Note: If Subject Name verification is necessary, the parameter PeerHostNameVerificationMode must be used as well.
Strict Certificate Extension Validation `require-strict-cert` [RequireStrictCert]	Enables the validation of the extensions (keyUsage and extentedKeyUsage) of peer certificates. This validation ensures that the signing CA is authorized to sign certificates and that the end-entity certificate is authorized to negotiate a secure TLS connection. ▪ **[0]** Disable (default) ▪ **[1]** Enable
TLS Remote Subject Name `tls-rmt-subs-name` [TLSRemoteSubjectName]	Defines the Subject Name that is compared with the name defined in the remote side certificate when establishing TLS connections. If the SubjectAltName of the received certificate is not equal to any of the defined Proxies Host names/IP addresses and is not marked as 'critical', the Common Name (CN) of the Subject field is compared with this value. If not equal, the TLS connection is not established. If the CN uses a domain name, the certificate can also use wildcards ('*') to replace parts of the domain name. The valid range is a string of up to 49 characters. **Note:** The parameter is applicable only if the parameter PeerHostNameVerificationMode is set to 1 or 2.
TLS Expiry Check Start `expiry-check-start` [TLSExpiryCheckStart]	Defines the number of days before the installed TLS server certificate is to expire at which the device must send a trap (acCertificateExpiryNotification) to notify of this. The valid value is 0 to 3650. The default is 60.
TLS Expiry Check Period `expiry-check-period` [TLSExpiryCheckPeriod]	Defines the periodical interval (in days) for checking the TLS server certificate expiry date. The valid value is 1 to 3650. The default is 7.

57.4.5 SSH Parameters

Secure Shell (SSH) parameters are described in the table below.

Table 57-25: SSH Parameters

Parameter	Description
Enable SSH Server `ssh` [SSHServerEnable]	Enables the device's embedded SSH server. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Server Port `ssh-port` [SSHServerPort]	Defines the port number for the embedded SSH server. Range is any valid port number. The default port is 22.
SSH Admin Key `ssh-admin-key` [SSHAdminKey]	Defines the RSA public key for strong authentication for logging in to the SSH interface (if enabled). The value should be a base64-encoded string. The value can be a maximum length of 511 characters.
Require Public Key `ssh-require-public-key` [SSHRequirePublicKey]	Enables RSA public keys for SSH. ▪ **[0]** = (Default) RSA public keys are optional if a value is configured for the parameter SSHAdminKey. ▪ **[1]** = RSA public keys are mandatory. **Note:** To define the key size, use the TLSPkeySize parameter.
Max Payload Size `ssh-max-payload-size` [SSHMaxPayloadSize]	Defines the maximum uncompressed payload size (in bytes) for SSH packets. The valid value is 550 to 32768. The default is 32768.
Max Binary Packet Size `ssh-max-binary-packet-size` [SSHMaxBinaryPacketSize]	Defines the maximum packet size (in bytes) for SSH packets. The valid value is 582 to 35000. The default is 35000.
Maximum SSH Sessions `ssh-max-sessions` [SSHMaxSessions]	Defines the maximum number of simultaneous SSH sessions. The valid range is 1 to 5. The default is 5 sessions.
Enable Last Login Message `ssh-last-login-message` [SSHEnableLastLoginMessage]	Enables message display in SSH sessions of the time and date of the last SSH login. The SSH login message displays the number of unsuccessful login attempts since the last successful login. ▪ **[0]** Disable ▪ **[1]** Enable (default) **Note:** The last SSH login information is cleared when the device is reset.
Max Login Attempts `ssh-max-login-attempts` **[SSHMaxLoginAttempts]**	Defines the maximum SSH login attempts allowed for entering an incorrect password by an administrator before the SSH session is rejected. The valid range is 1 to 5. the default is 3. **Note:** The new setting takes effect only for new subsequent SSH connections.

57.4.6 IDS Parameters

The Intrusion Detection System (IDS) parameters are described in the table below.

Table 57-26: IDS Parameters

Parameter	Description
Intrusion Detection System (IDS) `enable-ids` [EnableIDS]	Enables the IDS feature. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** For the parameter to take effect, a device reset is required.
`ids-clear-period` [IDSAlarmClearPeriod]	Defines the interval (in seconds) after which an IDS alarm is cleared from the Active Alarms table if no thresholds are crossed during this time. However, this "quiet" period must be at least twice the Threshold Window value. For example, if IDSAlarmClearPeriod is set to 20 sec and the Threshold Window is set to 15 sec, the IDSAlarmClearPeriod parameter is ignored and the alarm is cleared only after 30 seconds (2 x 15 sec). The valid value is 0 to 86400. The default is 300.
IDS Policy Table	
IDS Policy Table [IDSPolicy]	Defines IDS Policies. The format of the ini file parameter is: [IDSPolicy] FORMAT IDSPolicy_Index = IDSPolicy_Name, IDSPolicy_Description; [\IDSPolicy] For a detailed description of the table, see "Configuring IDS Policies" on page 164.
IDS Rule Table	
IDS Rule Table [IDSRule]	Defines rules for IDS Policies. The format of the ini file parameter is: [IDSRule] FORMAT IDSRule_Index = IDSRule_Policy, IDSRule_RuleID, IDSRule_Reason, IDSRule_ThresholdScope, IDSRule_ThresholdWindow, IDSRule_MinorAlarmThreshold, IDSRule_MajorAlarmThreshold, IDSRule_CriticalAlarmThreshold, IDSRule_DenyThreshold, IDSRule_DenyPeriod; [\IDSRule] For a detailed description of the table, see "Configuring IDS Policies" on page 164.
IDS Match Table	
IDS Match Table [IDSMatch]	Defines target rules per IDS Policy. The format of the ini file parameter is: [IDSMatch] FORMAT IDSMatch_Index = IDSMatch_SIPInterface, IDSMatch_ProxySet, IDSMatch_Subnet, IDSMatch_Policy; [\IDSMatch] For a detailed description of the table, see "Assigning IDS Policies" on page 168.

57.4.7 OCSP Parameters

The Online Certificate Status Protocol (OCSP) parameters are described in the table below.

Table 57-27: OCSP Parameters

Parameter	Description
Enable OCSP Server `enable` [OCSPEnable]	Enables or disables certificate checking using OCSP. ▪ **[0]** Disable (default) ▪ **[1]** Enable For a description of OCSP, see Configuring Certificate Revocation Checking (OCSP).
Primary Server IP `server-ip` [OCSPServerIP]	Defines the IP address of the OCSP server. The default IP address is 0.0.0.0.
Secondary Server IP `secondary-server-ip` [OCSPSecondaryServerIP]	Defines the IP address (in dotted-decimal notation) of the secondary OCSP server (optional). The default IP address is 0.0.0.0.
Server Port `server-port` [OCSPServerPort]	Defines the OCSP server's TCP port number. The default port number is 2560.
Default Response When Server Unreachable `default-response` [OCSPDefaultResponse]	Determines whether the device allows or rejects peer certificates when the OCSP server cannot be contacted. ▪ **[0]** Reject (default) ▪ **[1]** Allow

57.5 Quality of Experience Parameters

The Quality of Experience (QoE) parameters are described in the table below.

Table 57-28: Quality of Experience Parameters

Parameter	Description
SEM Parameters	
Server IP `configure voip/qoe configuration/server-ip` [QOEServerIP]	Defines the IP address of the primary Session Experience Manager (SEM) server to where the quality experience reports are sent. **Note:** For the parameter to take effect, a device reset is required.
Redundant Server IP `configure voip > qoe configuration > set secondary-server-ip` [QOESecondaryServerIp]	Defines the IP address of the secondary SEM server to where the quality experience reports are sent. This is applicable when the SEM/EMS server is in Geographical Redundancy HA mode. **Note:** For the parameter to take effect, a device reset is required.

Parameter	Description
Interface Name `configure voip/qoe configuration/interface-name` [QOEInterfaceName]	Defines the IP network interface on which the quality experience reports are sent. The default is the OAMP interface. **Note:** For the parameter to take effect, a device reset is required.
QoE Connection by TLS `configure voip > qoe configuration > tls-enable` [QOEEnableTLS]	Enables a TLS connection with the SEM server. ▪ [0] Disable (default) ▪ [1] Enable **Note:** For the parameter to take effect, a device reset is required.
QOE TLS Context Name `configure voip/qoe configuration/tls-context-name` [QoETLSContextName]	Selects a TLS Context (configured in the TLS Contexts table) for the TLS connection with the SEM server. The valid value is a string representing the name of the TLS Context as configured in the 'Name' field of the TLS Contexts table. The default is the default TLS Context (ID 0).
QoE Report Mode `report-mode` [QoeReportMode]	Defines at what stage of the call the device sends the QoE data of the call to the SEM server. ▪ [0] Report QoE During Call (default) ▪ [1] Report QoE At End Call **Note:** If a QoE traffic overflow between SEM and the device occurs, the device sends the QoE data only at the end of the call, regardless of the settings of the parameter.
Quality of Experience Profile Table	
Quality of Experience Profile `configure voip/qoe qoe-profile` [QOEProfile]	The table defines Quality of Experience Profiles. The format of the ini file table parameter is as follows: [QOEProfile] FORMAT QOEProfile_Index = QOEProfile_Name, QOEProfile_SensitivityLevel; [\QOEProfile] For a detailed description of the table, see "Configuring Quality of Experience Profiles" on page 307.
Quality of Experience Color Rules Table	
Quality of Experience Color Rules `configure voip/qoe qoe-profile qoe-color-rules` [QOEColorRules]	The table defines Quality of Experience Color Rules. The format of the ini file table parameter is as follows: [QOEColorRules] FORMAT QOEColorRules_Index = QOEColorRules_QoeProfile, QOEColorRules_ColorRuleIndex, QOEColorRules_monitoredParam, QOEColorRules_direction, QOEColorRules_profile, QOEColorRules_GreenYellowThreshold, QOEColorRules_GreenYellowHysteresis, QOEColorRules_YellowRedThreshold, QOEColorRules_YellowRedHysteresis; [\QOEColorRules] For a detailed description of the table, see "Configuring Quality of Experience Profiles" on page 307.
Bandwidth Profile Table	

Parameter	Description
Bandwidth Profile `configure voip/qoe bw-profile` [BWProfile]	The table defines Bandwidth Profiles. The format of the ini file table parameter is as follows: [BWProfile] FORMAT BWProfile_Index = BWProfile_Name, BWProfile_EgressAudioBandwidth, BWProfile_IngressAudioBandwidth, BWProfile_EgressVideoBandwidth, BWProfile_IngressVideoBandwidth, BWProfile_TotalEgressBandwidth, BWProfile_TotalIngressBandwidth, BWProfile_WarningThreshold, BWProfile_hysteresis, BWProfile_GenerateAlarms; [\BWProfile] For a detailed description of the table, see "Configuring Bandwidth Profiles" on page 311. **Note:** For the parameter to take effect, a device reset is required.
Media Enhancement Profile Table	
Media Enhancement Profile `configure voip/qoe media-enhancement` [MediaEnhancementProfile]	The table defines Media Enhancement Profiles. The format of the ini file table parameter is as follows: [MediaEnhancementProfile] FORMAT MediaEnhancementProfile_Index = MediaEnhancementProfile_ProfileName; [\MediaEnhancementProfile] For a detailed description of the table, see "Configuring Media Enhancement Profiles" on page 313.
Media Enhancement Rules Table	
Media Enhancement Rules `configure voip/qoe media-enhancement-rules` [MediaEnhancementRules]	The table defines Media Enhancement Rules. The format of the ini file table parameter is as follows: [MediaEnhancementRules] FORMAT MediaEnhancementRules_Index = MediaEnhancementRules_MediaEnhancementProfile, MediaEnhancementRules_RuleIndex, MediaEnhancementRules_Trigger, MediaEnhancementRules_Color, MediaEnhancementRules_ActionRule, MediaEnhancementRules_ActionValue; [\MediaEnhancementRules] For a detailed description of the table, see "Configuring Media Enhancement Profiles" on page 313.

57.6 Control Network Parameters

57.6.1 IP Group, Proxy, Registration and Authentication Parameters

The proxy server, registration and authentication SIP parameters are described in the table below.

Table 57-29: Proxy, Registration and Authentication SIP Parameters

Parameter	Description
IP Group Table	
IP Group Table `configure voip > voip-network ip-group` [IPGroup]	This table configures IP Groups. The format of the ini file table parameter is: [IPGroup] FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Name, IPGroup_ProxySetName, IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_SipReRoutingMode, IPGroup_AlwaysUseRouteTable, IPGroup_SRDName, IPGroup_MediaRealm, IPGroup_ClassifyByProxySet, IPGroup_ProfileName, IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet, IPGroup_OutboundManSet, IPGroup_RegistrationMode, IPGroup_AuthenticationMode, IPGroup_MethodList, IPGroup_EnableSBCClientForking, IPGroup_SourceUriInput, IPGroup_DestUriInput, IPGroup_ContactName, IPGroup_Username, IPGroup_Password, IPGroup_UUIFormat, IPGroup_QOEProfile, IPGroup_BWProfile, IPGroup_MediaEnhancementProfile, IPGroup_AlwaysUseSourceAddr, IPGroup_MsgManUserDef1, IPGroup_MsgManUserDef2, IPGroup_SIPConnect, IPGroup_SBCPSAPMode, IPGroup_DTLSContext, IPGroup_CreatedByRoutingServer, IPGroup_UsedByRoutingServer, IPGroup_SBCOperationMode, IPGroup_SBCRouteUsingRequestURIPort; [/IPGroup] For a description of the table, see "Configuring IP Groups" on page 340. **Note:** For the parameter to take effect, a device reset is required.
Account Table	
Account Table `configure voip > sip-definition account` [Account]	Defines user accounts for registering and/or authenticating (digest) Trunk Groups or IP Groups (e.g., an IP-PBX) with a Serving IP Group (e.g., a registrar server). The format of the ini file table parameter is as follows: [Account] FORMAT Account_Index = Account_ServedTrunkGroup, Account_ServedIPGroupName, Account_ServingIPGroupName, Account_Username, Account_Password, Account_HostName, Account_Register, Account_ContactUser, Account_ApplicationType; [\Account] For a detailed description of the table, see "Configuring Registration Accounts" on page 363.

Parameter	Description
Proxy Registration Parameters	
Use Default Proxy `enable-proxy` [IsProxyUsed]	Enables the use of Proxy Set ID 0 (for backward compatibility). ▪ [0] No = (Default) Proxy Set 0 is not used. ▪ [1] Yes = Proxy Set ID 0 is used. **Notes:** ▪ The parameter must be used only for backward compatibility. If not required for backward compatibility, make sure that the parameter is disabled, and use the Proxy Set table for configuring all your Proxy Sets (except for Proxy Set ID 0). ▪ If you are not using a proxy server, you must configure routing rules to route the call. ▪ The parameter is applicable only to the Gateway application.
Proxy Name `proxy-name` [ProxyName]	Defines the Home Proxy domain name. If specified, this name is used as the Request-URI in REGISTER, INVITE and other SIP messages, and as the host part of the To header in INVITE messages. If not specified, the Proxy IP address is used instead. The valid value is a string of up to 49 characters. **Note:** The parameter functions together with the UseProxyIPasHost parameter.
Use Proxy IP as Host `use-proxy-ip-as-host` [UseProxyIPasHost]	Enables the use of the proxy server's IP address (in dotted-decimal notation) as the host name in SIP From and To headers in REGISTER requests. ▪ **[0]** Disable (default) ▪ **[1]** Enable If the parameter is disabled and the device registers to an IP Group (i.e., proxy server), it uses the string configured by the ProxyName parameter as the host name in the REGISTER's Request-URI and uses the string configured by the IP Group table parameter, SIPGroupName as the host name in the To and From headers. If the IP Group is configured with a Proxy Set that has multiple IP addresses, all the REGISTER messages sent to these proxies are sent with the same host name. **Note:** If the parameter is disabled and the ProxyName parameter is not configured, the proxy's IP address is used as the host name in the REGISTER Request-URI.
Redundancy Mode `redundancy-mode` [ProxyRedundancyMode]	Determines whether the device switches back to the primary Proxy after using a redundant Proxy. ▪ **[0]** Parking = (Default) The device continues working with a redundant (now active) Proxy until the next failure, after which it works with the next redundant Proxy. ▪ **[1]** Homing = The device always tries to work with the primary Proxy server (i.e., switches back to the primary Proxy whenever it's available). **Note:** To use this Proxy Redundancy mechanism, you need to enable the keep-alive with Proxy option, by setting the parameter EnableProxyKeepAlive to 1 or 2.

Parameter	Description
Proxy IP List Refresh Time `proxy-ip-lst-rfrsh-time` [ProxyIPListRefreshTime]	Defines the time interval (in seconds) between each Proxy IP list refresh. The range is 5 to 2,000,000. The default interval is 60.
Enable Fallback to Routing Table `fallback-to-routing` [IsFallbackUsed]	Determines whether the device falls back to the Tel-to-IP Routing table for call routing when Proxy servers are unavailable. ▪ [0] Disable = (Default) Fallback is not used. ▪ [1] Enable = The Tel-to-IP Routing table is used when Proxy servers are unavailable. When the device falls back to the Tel-to-IP Routing table, it continues scanning for a Proxy. When the device locates an active Proxy, it switches from internal routing back to Proxy routing. **Note:** To enable the redundant Proxies mechanism, set the parameter EnableProxyKeepAlive to 1 or 2.
Prefer Routing Table `prefer-routing-table` [PreferRouteTable]	Determines whether the device's routing table takes precedence over a Proxy for routing calls. ▪ [0] No = (Default) Only a Proxy server is used to route calls. ▪ [1] Yes = The device checks the routing rules in the Tel-to-IP Routing table for a match with the Tel-to-IP call. Only if a match is not found is a Proxy used.
Always Use Proxy `always-use-proxy` [AlwaysSendToProxy]	Determines whether the device sends SIP messages and responses through a Proxy server. ▪ **[0]** Disable = (Default) Use standard SIP routing rules. ▪ **[1]** Enable = All SIP messages and responses are sent to the Proxy server. **Note:** The parameter is applicable only if a Proxy server is used (i.e., the parameter IsProxyUsed is set to 1).
SIP ReRouting Mode `sip-rerouting-mode` [SIPReroutingMode]	Determines the routing mode after a call redirection (i.e., a 3xx SIP response is received) or transfer (i.e., a SIP REFER request is received). ▪ [0] Standard = (Default) INVITE messages that are generated as a result of Transfer or Redirect are sent directly to the URI, according to the Refer-To header in the REFER message, or Contact header in the 3xx response. ▪ [1] Proxy = Sends a new INVITE to the Proxy. Note: This option is applicable only if a Proxy server is used and the parameter AlwaysSendtoProxy is set to 0. ▪ [2] Routing Table = Uses the Routing table to locate the destination and then sends a new INVITE to this destination. **Notes:** ▪ The parameter is applicable only to the Gateway application. ▪ When the parameter is set to [1] and the INVITE sent to the Proxy fails, the device re-routes the call according to the Standard mode [0]. ▪ When the parameter is set to [2] and the INVITE fails, the device re-routes the call according to the Standard mode [0]. If DNS resolution fails, the device attempts to route the call to the Proxy. If routing to the Proxy also fails, the Redirect/Transfer request is rejected.

Parameter	Description
	▪ When the parameter is set to [2], the XferPrefix parameter can be used to define different routing rules for redirect calls. ▪ The parameter is disregarded if the parameter AlwaysSendToProxy is set to 1.
DNS Query Type `dns-query` [DNSQueryType]	Enables the use of DNS Naming Authority Pointer (NAPTR) and Service Record (SRV) queries to resolve Proxy and Registrar servers and to resolve all domain names that appear in the SIP Contact and Record-Route headers. ▪ **[0]** A-Record = (Default) No NAPTR or SRV queries are performed. ▪ **[1]** SRV = If the Proxy/Registrar IP address parameter, Contact/Record-Route headers, or IP address configured in the routing tables contain a domain name, an SRV query is performed. The device uses the first host name received from the SRV query. The device then performs a DNS A-record query for the host name to locate an IP address. ▪ **[2]** NAPTR = An NAPTR query is performed. If it is successful, an SRV query is sent according to the information received in the NAPTR response. If the NAPTR query fails, an SRV query is performed according to the configured transport type. **Notes:** ▪ If the Proxy/Registrar IP address parameter, the domain name in the Contact/Record-Route headers, or the IP address configured in the routing tables contain a domain name with a port definition, the device performs a regular DNS A-record query. ▪ If a specific Transport Type is configured, a NAPTR query is not performed. ▪ To enable NAPTR/SRV queries for Proxy servers only, use the global parameter ProxyDNSQueryType, or use the proxy Set table.
Proxy DNS Query Type `proxy-dns-query` [ProxyDNSQueryType]	Global parameter that defines the DNS query record type for resolving the Proxy server's configured domain name (FQDN) into an IP address. ▪ **[0]** A-Record (default) = A-record DNS query. ▪ **[1]** SRV = If the Proxy IP address parameter contains a domain name without port definition (e.g., ProxyIP = domain.com), an SRV query is performed. The SRV query returns up to four Proxy host names and their weights. The device then performs DNS A-record queries for each Proxy host name (according to the received weights) to locate up to four Proxy IP addresses. Thus, if the first SRV query returns two domain names and the A-record queries return two IP addresses each, no additional searches are performed. ▪ **[2]** NAPTR = NAPTR query is done. If successful, an SRV query is sent according to the information received in the NAPTR response. If the NAPTR query fails, an SRV query is done according to the configured transport type. If the Proxy IP address parameter contains a domain name with port definition (e.g., ProxyIP = domain.com:5080), the device performs a regular DNS A-record query. If a specific Transport Type is defined, a NAPTR query is not performed.

Parameter	Description
	Notes: - This functionality can be configured per Proxy Set in the Proxy Set table (see "Configuring Proxy Sets" on page 352). - When enabled, NAPTR/SRV queries are used to discover Proxy servers even if the parameter DNSQueryType is disabled.
Use Gateway Name for OPTIONS `use-gw-name-for-opt` [UseGatewayNameForOptions]	Determines whether the device uses its IP address or string name ("gateway name") in keep-alive SIP OPTIONS messages (host part of the Request-URI). To configure the "gateway name", use the SIPGatewayName parameter. The device uses the OPTIONS request as a keep-alive message with its primary and redundant SIP proxy servers (i.e., the EnableProxyKeepAlive parameter is set to 1). - **[0]** No = (Default) Device's IP address is used in keep-alive OPTIONS messages. - **[1]** Yes = Device's "gateway name" is used in keep-alive OPTIONS messages. - **[2]** Server = Device's IP address is used in the From and To headers in keep-alive OPTIONS messages.
User Name `user-name-4-auth` [UserName]	Defines the username for registration and Basic/Digest authentication with a Proxy/Registrar server. By default, no value is defined. **Note:** - The parameter is applicable only to the Gateway application. - The parameter is applicable only if single device registration is used (i.e., the parameter AuthenticationMode is set to authentication per gateway).
Password `password-4-auth` [Password]	Defines the password for Basic/Digest authentication with a Proxy/Registrar server. A single password is used for all device ports. The default is 'Default_Passwd'.
Cnonce `cnonce-4-auth` [Cnonce]	Defines the Cnonce string used by the SIP server and client to provide mutual authentication. The value is free format, i.e., 'Cnonce = 0a4f113b'. The default is 'Default_Cnonce'.
Mutual Authentication Mode `mutual-authentication` [MutualAuthenticationMode]	Determines the device's mode of operation when Authentication and Key Agreement (AKA) Digest Authentication is used. - **[0]** Optional = (Default) Incoming requests that don't include AKA authentication information are accepted. - **[1]** Mandatory = Incoming requests that don't include AKA authentication information are rejected.
Challenge Caching Mode `challenge-caching` [SIPChallengeCachingMode]	Enables local caching of SIP message authorization challenges from Proxy servers. The device sends the first request to the Proxy without authorization. The Proxy sends a 401/407 response with a challenge for credentials. The device saves (caches) the response for further uses. The device sends a new request with the appropriate credentials. Subsequent requests to the Proxy are automatically sent with credentials (calculated from the

Parameter	Description
	saved challenge). If the Proxy doesn't accept the new request and sends another challenge, the old challenge is replaced with the new one. One of the benefits of the feature is that it may reduce the number of SIP messages transmitted through the network. ▪ [0] None = (Default) Challenges are not cached. Every new request is sent without preliminary authorization. If the request is challenged, a new request with authorization data is sent. ▪ [1] INVITE Only = Challenges issued for INVITE requests are cached. This prevents a mixture of REGISTER and INVITE authorizations. ▪ [2] Full = Caches all challenges from the proxies. **Note:** ▪ Challenge caching is used with all proxies and not only with the active one. ▪ For the Gateway application: The challenge can be cached per endpoint or per Account. ▪ For the SBC application: The challenge can be cached per Account or per user whose credentials are known through the User Info table.
Proxy Address Table	
Proxy IP Table `configure voip > voip-network proxy-ip` [ProxyIP]	The table defines proxy addresses per Proxy Set. The format of the ini file table parameter is as follows: [ProxyIP] FORMAT ProxyIp_Index = ProxyIp_ProxySetId, ProxyIp_ProxyIpIndex, ProxyIp_IpAddress, ProxyIp_TransportType; [\ProxyIP] For a description of the table, see "Configuring Proxy Sets" on page 352.
Proxy Sets Table	
Proxy Set Table `configure voip > voip-network proxy-set` [ProxySet]	Defines the Proxy Sets. The format of the ini file table parameter is as follows: [FORMAT ProxySet_Index = ProxySet_ProxyName, ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap, ProxySet_SRDName, ProxySet_ClassificationInput, ProxySet_TLSContextName, ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod, ProxySet_KeepAliveFailureResp, ProxySet_GWIPv4SIPInterfaceName, ProxySet_SBCIPv4SIPInterfaceName, ProxySet_GWIPv6SIPInterfaceName, ProxySet_SBCIPv6SIPInterfaceName, ProxySet_MinActiveServersLB, ProxySet_SuccessDetectionRetries, ProxySet_SuccessDetectionInterval, ProxySet_FailureDetectionRetransmissions; [\ProxySet]

Parameter	Description
	For a description of the table, see "Configuring Proxy Sets" on page 352.
Registrar Parameters	
Enable Registration `enable-registration` [IsRegisterNeeded]	Enables the device to register to a Proxy/Registrar server. ▪ [0] Disable = (Default) The device doesn't register to Proxy/Registrar server. ▪ [1] Enable = The device registers to Proxy/Registrar server when the device is powered up and at every user-defined interval (configured by the parameter RegistrationTime). **Notes:** ▪ The parameter is applicable only to the Gateway application. ▪ The device sends a REGISTER request for each channel or for the entire device (according to the AuthenticationMode parameter).
Registrar Name `registrar-name` [RegistrarName]	Defines the Registrar domain name. If specified, the name is used as the Request-URI in REGISTER messages. If it isn't specified (default), the Registrar IP address, or Proxy name or IP address is used instead. The valid range is up to 100 characters. **Note:** The parameter is applicable only to the Gateway application.
Registrar IP Address `ip-addrr-rgstrr` [RegistrarIP]	Defines the IP address (or FQDN) and port number (optional) of the Registrar server. The IP address is in dotted-decimal notation, e.g., 201.10.8.1:<5080>. **Notes:** ▪ The parameter is applicable only to the Gateway application. ▪ If not specified, the REGISTER request is sent to the primary Proxy server. ▪ When a port number is specified, DNS NAPTR/SRV queries aren't performed, even if the parameter DNSQueryType is set to 1 or 2. ▪ If the parameter RegistrarIP is set to an FQDN and is resolved to multiple addresses, the device also provides real-time switching (hotswap mode) between different Registrar IP addresses (the parameter IsProxyHotSwap is set to 1). If the first Registrar doesn't respond to the REGISTER message, the same REGISTER message is sent immediately to the next Proxy. To allow this mechanism, the parameter EnableProxyKeepAlive must be set to 0. ▪ When a specific transport type is defined using the parameter RegistrarTransportType, a DNS NAPTR query is not performed even if the parameter DNSQueryType is set to 2.
Registrar Transport Type `registrar-transport` [RegistrarTransportType]	Determines the transport layer used for outgoing SIP dialogs initiated by the device to the Registrar. ▪ [-1] Not Configured (default) ▪ [0] UDP ▪ [1] TCP ▪ [2] TLS

Parameter	Description
	Notes: ▪ The parameter is applicable only to the Gateway application. ▪ When set to 'Not Configured', the value of the parameter SIPTransportType is used.
Registration Time `registration-time` [RegistrationTime]	Defines the time interval (in seconds) for registering to a Proxy server. The value is used in the SIP Expires header. The parameter also defines the time interval between Keep-Alive messages when the parameter EnableProxyKeepAlive is set to 2 (REGISTER). Typically, the device registers every 3,600 sec (i.e., one hour). The device resumes registration according to the parameter RegistrationTimeDivider. The valid range is 10 to 2,000,000. The default is 180.
Re-registration Timing [%] `re-registration-timing` [RegistrationTimeDivider]	Defines the re-registration timing (in percentage). The timing is a percentage of the re-register timing set by the Registrar server. The valid range is 50 to 100. The default is 50. For example: If the parameter is set to 70% and the Registration Expires time is 3600, the device re-sends its registration request after 3600 x 70% (i.e., 2520 sec). **Notes**: ▪ The parameter may be overridden if the parameter RegistrationTimeThreshold is greater than 0.
Registration Retry Time `registration-retry-time` [RegistrationRetryTime]	Defines the time interval (in seconds) after which a registration request is re-sent if registration fails with a 4xx response or if there is no response from the Proxy/Registrar server. The default is 30 seconds. The range is 10 to 3600.
Registration Time Threshold `registration-time-thres` [RegistrationTimeThreshold]	Defines a threshold (in seconds) for re-registration timing. If the parameter is greater than 0, but lower than the computed re-registration timing (according to the parameter RegistrationTimeDivider), the re-registration timing is set to the following: timing set by the Registration server in the SIP Expires header minus the value of the parameter RegistrationTimeThreshold. The valid range is 0 to 2,000,000. The default is 0.
Re-register On INVITE Failure `reg-on-invite-fail` [RegisterOnInviteFailure]	Enables immediate re-registration if no response is received for an INVITE request sent by the device. ▪ [0] Disable (default) ▪ [1] Enable = The device immediately expires its re-registration timer and commences re-registration to the same Proxy upon any of the following scenarios: ✓ The response to an INVITE request is 407 (Proxy Authentication Required) without an authentication header included. ✓ The remote SIP UA abandons a call before the device has received any provisional response (indicative of an outbound proxy server failure). ✓ The remote SIP UA abandons a call and the only provisional response the device has received for the call is 100 Trying (indicative of a home proxy server failure, i.e., the failure of a proxy in the route after the outbound proxy).

Parameter	Description
	✓ The device terminates a call due to the expiration of RFC 3261 Timer B or due to the receipt of a 408 (Request Timeout) response and the device has not received any provisional response for the call (indicative of an outbound proxy server failure). ✓ The device terminates a call due to the receipt of a 408 (Request Timeout) response and the only provisional response the device has received for the call is the 100 Trying provisional response (indicative of a home proxy server failure). **Note:** The parameter is applicable only to the Gateway application.
ReRegister On Connection Failure `reg-on-conn-failure` [ReRegisterOnConnectionFailure]	Enables the device to perform SIP re-registration upon TCP/TLS connection failure. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Gateway Registration Name `gw-registration-name` [GWRegistrationName]	Defines the user name that is used in the From and To headers in SIP REGISTER messages. If no value is specified (default) for the parameter, the UserName parameter is used instead. **Notes:** ▪ The parameter is applicable only to the Gateway application. ▪ The parameter is applicable only for single registration per device (i.e., AuthenticationMode is set to 1). When the device registers each channel separately (i.e., AuthenticationMode is set to 0), the user name is set to the channel's phone number.
Registration Mode `authentication-mode` [AuthenticationMode]	Determines the device's registration and authentication method. ▪ [0] Per Endpoint = Registration and authentication is performed separately for each B-channel. ▪ [1] Per Gateway = (Default) Single registration and authentication for the entire device. This is typically used for and digital modules. **Note:** The parameter is applicable only to the Gateway application.
Set Out-Of-Service On Registration Failure `set-oos-on-reg-failure` [OOSOnRegistrationFail]	Enables setting the , trunk, or entire device (i.e., all endpoints) to out-of-service if registration fails. ▪ [0] Disable (default) ▪ [1] Enable If the registration is per endpoint (i.e., AuthenticationMode is set to 0) or per Account (see Configuring Trunk Group Settings on page 435) and a specific endpoint/Account registration fails (SIP 4xx or no response), then that endpoint is set to out-of-service until a success response is received in a subsequent registration request. When the registration is per the entire device (i.e., AuthenticationMode is set to 1) and registration fails, all endpoints are set to out-of-service. If all the Accounts of a specific Trunk Group fail registration and if the Trunk Group comprises a complete trunk, then the entire trunk is set to out-of-service. **Notes:** ▪ The parameter is applicable only to the Gateway application.

Parameter	Description
`expl-un-reg` [UnregistrationMode]	Enables the device to perform explicit unregisters. ▪ **[0]** Disable (default) ▪ **[1]** Enable = The device sends an asterisk ("*") value in the SIP Contact header, instructing the Registrar server to remove all previous registration bindings. The device removes SIP User Agent (UA) registration bindings in a Registrar, according to RFC 3261. Registrations are soft state and expire unless refreshed, but they can also be explicitly removed. A client can attempt to influence the expiration interval selected by the Registrar. A UA requests the immediate removal of a binding by specifying an expiration interval of "0" for that contact address in a REGISTER request. UA's should support this mechanism so that bindings can be removed before their expiration interval has passed. Use of the "*" Contact header field value allows a registering UA to remove all bindings associated with an address-of-record (AOR) without knowing their precise values. **Note:** The REGISTER-specific Contact header field value of "*" applies to all registrations, but it can only be used if the Expires header field is present with a value of "0".

Add Empty Authorization Header
`add-empty-author-hdr`
[EmptyAuthorizationHeader]
	Enables the inclusion of the SIP Authorization header in initial registration (REGISTER) requests sent by the device.
	▪ **[0]** Disable (default)
	▪ **[1]** Enable
	The Authorization header carries the credentials of a user agent (UA) in a request to a server. The sent REGISTER message populates the Authorization header with the following parameters:
	▪ username - set to the value of the private user identity
	▪ realm - set to the domain name of the home network
	▪ uri - set to the SIP URI of the domain name of the home network
	▪ nonce - set to an empty value
	▪ response - set to an empty value
	For example:
	`Authorization: Digest`
	`username=alice_private@home1.net,`
	`realm=”home1.net”, nonce=””,`
	`response=”e56131d19580cd833064787ecc”`
	Note: This registration header is according to the IMS 3GPP TS24.229 and PKT-SP-24.220 specifications.

Add initial Route Header `add-init-rte-hdr` [InitialRouteHeader]	Enables the inclusion of the SIP Route header in initial registration or re-registration (REGISTER) requests sent by the device. ▪ **[0]** Disable (default) ▪ **[1]** Enable When the device sends a REGISTER message, the Route header includes either the Proxy's FQDN, or IP address and port according to the configured Proxy Set, for example: `Route: <sip:10.10.10.10;lr;transport=udp>`

Parameter	Description
	or `Route: <sip: pcscf-gm.ims.rr.com;lr;transport=udp>`
[UsePingPongKeepAlive]	Enables the use of the carriage-return and line-feed sequences (CRLF) Keep-Alive mechanism, according to RFC 5626 “Managing Client-Initiated Connections in the Session Initiation Protocol (SIP)” for reliable, connection-orientated transport types such as TCP. ▪ **[0]** Disable (default) ▪ **[1]** Enable The SIP user agent/client (i.e., device) uses a simple periodic message as a keep-alive mechanism to keep their flow to the proxy or registrar alive (used for example, to keep NAT bindings open). For connection-oriented transports such as TCP/TLS this is based on CRLF. This mechanism uses a client-to-server "ping" keep-alive and a corresponding server-to-client "pong" message. This ping-pong sequence allows the client, and optionally the server, to tell if its flow is still active and useful for SIP traffic. If the client does not receive a pong in response to its ping, it declares the flow “dead” and opens a new flow in its place. In the CRLF Keep-Alive mechanism the client periodically (defined by the PingPongKeepAliveTime parameter) sends a double-CRLF (the "ping") then waits to receive a single CRLF (the "pong"). If the client does not receive a "pong" within an appropriate amount of time, it considers the flow failed. **Note:** The device sends a CRLF message to the Proxy Set only if the Proxy Keep-Alive feature (EnableProxyKeepAlive parameter) is enabled and its transport type is set to TCP or TLS. The device first sends a SIP OPTION message to establish the TCP/TLS connection and if it receives any SIP response, it continues sending the CRLF keep-alive sequences.
[PingPongKeepAliveTime]	Defines the periodic interval (in seconds) after which a “ping” (double-CRLF) keep-alive is sent to a proxy/registrar, using the CRLF Keep-Alive mechanism. The default range is 5 to 2,000,000. The default is 120. The device uses the range of 80-100% of this user-defined value as the actual interval. For example, if the parameter value is set to 200 sec, the interval used is any random time between 160 to 200 seconds. This prevents an “avalanche” of keep-alive by multiple SIP UAs to a specific server.
Max Generated Register Rate configure voip > sip-definition settings > max-gen-reg-rate [MaxGeneratedRegistersRate]	Defines the maximum number of user register requests (REGISTER messages) that the device sends (to a proxy or registrar server) at a user-defined rate configured by the GeneratedRegistersInterval parameter. The parameter is useful in that it may be used to prevent an overload on the device's CPU caused by sending many registration requests at a given time. The valid value is 30 to 300 register requests per second. The default is 150. For configuration examples, see the description of the GeneratedRegistersInterval parameter.

Parameter	Description
Generated Registers interval `gen-reg-int` [GeneratedRegistersInterval]	Defines the rate (in seconds) at which the device sends user register requests (REGISTER messages). The parameter is based on the maximum number of REGISTER messages that can be sent at this rate, configured by the MaxGeneratedRegistersRate parameter. The valid value is 1 to 5. The default is 1. Configuration examples: ▪ If you configure the MaxGeneratedRegistersRate parameter to 100 and the GeneratedRegistersInterval to 5, the device sends a maximum of 20 REGISTER messages per second (i.e., 100 messages divided by 5 sec; 100 per 5 seconds). ▪ If you configure the MaxGeneratedRegistersRate parameter to 100 and the GeneratedRegistersInterval to 1, the device sends a maximum of a 100 REGISTER messages per second.

57.6.2 Network Application Parameters

The SIP network application parameters are described in the table below.

Table 57-30: SIP Network Application Parameters

Parameter	Description
SRD Table	
SRD Table `configure voip > voip-network srd` [SRD]	Defines Signaling Routing Domains (SRD). The format of the ini file table parameter is as follows: [SRD] FORMAT SRD_Index = SRD_Name, SRD_IntraSRDMediaAnchoring, SRD_BlockUnRegUsers, SRD_MaxNumOfRegUsers, SRD_EnableUnAuthenticatedRegistrations, SRD_SharingPolicy, SRD_UsedByRoutingServer, SRD_SBCOperationMode, SRD_SBCRoutingPolicyName; [\SRD] For a detailed description of the table, see "Configuring SRDs" on page 325.
SIP Interface Table	
SIP Interface Table `configure voip > voip-network sip-interface` [SIPInterface]	Defines SIP Interfaces. The format of the ini file table parameter is as follows: [SIPInterface] FORMAT SIPInterface_Index = SIPInterface_InterfaceName, SIPInterface_NetworkInterface, SIPInterface_ApplicationType, SIPInterface_UDPPort, SIPInterface_TCPPort, SIPInterface_TLSPort, SIPInterface_SRDName, SIPInterface_MessagePolicyName, SIPInterface_TLSContext, SIPInterface_TLSMutualAuthentication, SIPInterface_TCPKeepaliveEnable, SIPInterface_ClassificationFailureResponseType, SIPInterface_PreClassificationManSet, SIPInterface_EncapsulatingProtocol, SIPInterface_MediaRealm, SIPInterface_SBCDirectMedia, SIPInterface_BlockUnRegUsers, SIPInterface_MaxNumOfRegUsers, SIPInterface_EnableUnAuthenticatedRegistrations, SIPInterface_UsedByRoutingServer; [\SIPInterface] For a detailed description of the table, see "Configuring SIP Interfaces" on page 333.
[TCPKeepAliveTime]	Defines the interval (in sec) between the last data packet sent and the first keep-alive probe to send. The valid value is 10 to 65,000. The default is 60. **Notes:** ▪ Simple ACKs such as keepalives are not considered data packets. ▪ TCP keepalive is enabled per SIP Interface in the SIP Interface table.
[TCPKeepAliveInterval]	Defines the interval (in sec) between consecutive keep-alive probes, regardless of what the connection has exchanged in the meantime. The valid value is 10 to 65,000. The default is 10. **Note:** TCP keepalive is enabled per SIP Interface in the SIP Interface table.

Parameter	Description
[TCPKeepAliveRetry]	Defines the number of unacknowledged keep-alive probes to send before considering the connection down. The valid value is 1 to 100. The default is 5. **Note:** TCP keepalive is enabled per SIP Interface in the SIP Interface table.
NAT Translation Table	
NAT Translation Table `configure voip > voip-network NATTranslation` [NATTranslation]	Defines NAT rules for translating source IP addresses per VoIP interface (SIP control and RTP media traffic) into NAT IP addresses. The format of the ini file table parameter is as follows: [NATTranslation] FORMAT NATTranslation_Index = NATTranslation_SrcIPInterfaceName, NATTranslation_TargetIPAddress, NATTranslation_SourceStartPort, NATTranslation_SourceEndPort, NATTranslation_TargetStartPort, NATTranslation_TargetEndPort; [\NATTranslation] For a detailed description of the table, see "Configuring NAT Translation per IP Interface" on page 151.
Media Realm Table	
Media Realm Table `configure voip > voip-network realm` [CpMediaRealm]	Defines Media Realms. The format of the ini file table parameter is as follows: [CpMediaRealm] FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName, CpMediaRealm_IPv4IF, CpMediaRealm_IPv6IF, CpMediaRealm_PortRangeStart, CpMediaRealm_MediaSessionLeg, CpMediaRealm_PortRangeEnd, CpMediaRealm_IsDefault, CpMediaRealm_QoeProfile, CpMediaRealm_BWProfile; [\CpMediaRealm] For a detailed description of the table, see "Configuring Media Realms" on page 317.
Remote Media Subnet Table	
Remote Media Subnet `configure voip > voip-network realm remote-media-subnet` [SubRealm]	Defines Remote Media Subnets. The format of the ini file table parameter is as follows: [RemoteMediaSubnet] FORMAT RemoteMediaSubnet_Index = RemoteMediaSubnet_Realm, RemoteMediaSubnet_RemoteMediaSubnetIndex, RemoteMediaSubnet_RemoteMediaSubnetName, RemoteMediaSubnet_PrefixLength, RemoteMediaSubnet_AddressFamily, RemoteMediaSubnet_DstIPAddress, RemoteMediaSubnet_QOEProfileName, RemoteMediaSubnet_BWProfileName; [\RemoteMediaSubnet] For a detailed description of the table, see "Configuring Remote Media Subnets" on page 320.

Parameter	Description
Media Realm Extension Table	
Media Realm Extension [MediaRealmExtension]	Defines Media Realm Extensions. The format of the ini file table parameter is as follows: [MediaRealmExtension] FORMAT MediaRealmExtension_Index = MediaRealmExtension_MediaRealmIndex, MediaRealmExtension_ExtensionIndex, MediaRealmExtension_IPv4IF, MediaRealmExtension_IPv6IF, MediaRealmExtension_PortRangeStart, MediaRealmExtension_PortRangeEnd, MediaRealmExtension_MediaSessionLeg; [\MediaRealmExtension] For a detailed description of the table, see "Configuring Media Realm Extensions" on page 322.

57.7 General SIP Parameters

The general SIP parameters are described in the table below.

Table 57-31: General SIP Parameters

Parameter	Description
Max Call Duration (min) `mx-call-duration` [MaxCallDuration]	Defines the maximum duration (in minutes) of a call. If this duration is reached, the device terminates the call. This feature is useful for ensuring available resources for new calls, by ensuring calls are properly terminated. The valid range is 0 to 35,791. The default is 0 (i.e., no limitation).
Send reject on overload `configure voip/sip-definition advanced-settings/reject-on-ovrld` [SendRejectOnOverload]	Disables the sending of SIP 503 (Service Unavailable) responses upon receipt of new SIP dialog-initiating requests when the device's CPU is overloaded and thus, unable to accept and process new SIP messages. ▪ **[0]** Disable = No SIP 503 response is sent when CPU overloaded. ▪ **[1]** Enable (default) = SIP 503 response is sent when CPU overloaded. ▪ **Note:** Even if the parameter is disabled (i.e., 503 is not sent), the device still discards the new SIP dialog-initiating requests when the CPU is overloaded.
SIP 408 Response upon non-INVITE `enbl-non-inv-408` [EnableNonInvite408Reply]	Enables the device to send SIP 408 responses (Request Timeout) upon receipt of non-INVITE transactions. Disabling this response complies with RFC 4320/4321. By default, and in certain circumstances such as a timeout expiry, the device sends a SIP 408 Request Timeout in response to non-INVITE requests (e.g., REGISTER). ▪ **[0]** Disable = SIP 408 response is not sent upon receipt of non-INVITE messages (to comply with RFC 4320). ▪ **[1]** Enable = (Default) SIP 408 response is sent upon receipt of non-INVITE messages, if necessary.
SIP Remote Reset `sip-remote-reset` [EnableSIPRemoteReset]	Enables a specific device action upon the receipt of a SIP NOTIFY request, where the action depends on the value received in the Event header.

Parameter	Description
	- **[0]** Disable (default) - **[1]** Enable The action depends on the Event header value: - 'check-sync;reboot=false': triggers the regular Automatic Update feature (if Automatic Update has been enabled on the device) - 'check-sync;reboot=true': triggers a device reset **Note:** - The Event header value is proprietary to AudioCodes. - The parameter is applicable only to the Gateway application.
Max SIP Message Length [KB] [MaxSIPMessageLength]	Defines the maximum size (in Kbytes) for each SIP message that can be sent over the network. The device rejects messages exceeding this user-defined size. The valid value range is 1 to 50. The default is 50.
[SIPForceRport]	Determines whether the device sends SIP responses to the UDP port from where SIP requests are received even if the 'rport' parameter is not present in the SIP Via header. - **[0]** = (Default) Disabled. The device sends the SIP response to the UDP port defined in the Via header. If the Via header contains the 'rport' parameter, the response is sent to the UDP port from where the SIP request is received. - **[1]** = Enabled. SIP responses are sent to the UDP port from where SIP requests are received even if the 'rport' parameter is not present in the Via header.
Reject Cancel after Connect `reject-cancel-after-connect` [RejectCancelAfterConnect]	Enables or disables the device to accept or reject SIP CANCEL requests received after the receipt of a 200 OK in response to an INVITE (i.e., call established). According to the SIP standard, a CANCEL can be sent only during the INVITE transaction (before 200 OK), and once a 200 OK response is received the call can be rejected only by a BYE request. - **[0]** Disable = (Default) Accepts a CANCEL request received during the INVITE transaction by sending a 200 OK response and terminates the call session. - **[1]** Enable = Rejects a CANCEL request received during the INVITE transaction by sending a SIP 481 (Call/Transaction Does Not Exist) response and maintains the call session.
Verify Received RequestURI `verify-rcvd-requri` [VerifyReceevedRequestUri]	Enables the device to reject SIP requests (such as ACK, BYE, or re-INVITE) whose user part in the Request-URI is different from the user part received in the Contact header of the last sent SIP request. - **[0]** Disable = (Default) Even if the user is different, the device accepts the SIP request. - **[1]** Enable = If the user is different, the device rejects the SIP request (BYE is responded with 481; re-INVITE is responded with 404; ACK is ignored).
Max Number of Active Calls `max-nb-of--act-calls` [MaxActiveCalls]	Defines the maximum number of simultaneous active calls supported by the device. If the maximum number of calls is reached, new calls are not established. The valid range is 1 to the maximum number of supported channels. The default value is the maximum available channels (i.e., no restriction on the maximum number of calls).

Parameter	Description
Number of Calls Limit [IpProfile_CallLimit,]	Defines the maximum number of concurrent calls per IP Profile (see "Configuring IP Profiles" on page 387).

QoS statistics in SIP Release Call
[QoSStatistics]
	Enables the device to include call quality of service (QoS) statistics in SIP BYE and SIP 200 OK response to BYE, using the proprietary SIP header X-RTP-Stat.
	▪ **[0]** = Disable (default)
	▪ **[1]** = Enable
	The X-RTP-Stat header provides the following statistics:
	▪ Number of received and sent voice packets
	▪ Number of received and sent voice octets
	▪ Received packet loss, jitter (in ms), and latency (in ms)
	The X-RTP-Stat header contains the following fields:
	▪ PS=<voice packets sent>
	▪ OS=<voice octets sent>
	▪ PR=<voice packets received>
	▪ OR=<voice octets received>
	▪ PL=<receive packet loss>
	▪ JI=<jitter in ms>
	▪ LA=<latency in ms>
	Below is an example of the X-RTP-Stat header in a SIP BYE message:
	`BYE sip:302@10.33.4.125 SIP/2.0`
	`Via: SIP/2.0/UDP 10.33.4.126;branch=z9hG4bKac2127550866`
	`Max-Forwards: 70`
	`From: <sip:401@10.33.4.126;user=phone>;tag=1c2113553324`
	`To: <sip:302@company.com>;tag=1c991751121`
	`Call-ID: 991750671245200001912@10.33.4.125`
	`CSeq: 1 BYE`
	`X-RTP-Stat: PS=207;OS=49680;;PR=314;OR=50240;PL=0;JI=600;LA=40;`
	`Supported: em,timer,replaces,path,resource-priority`
	`Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE`
	`User-Agent: Sip-Gateway-/v.6.80A.227.005`
	`Reason: Q.850 ;cause=16 ;text="local"`
	`Content-Length: 0`

PRACK Mode `prack-mode` [PrackMode]	Determines the PRACK (Provisional Acknowledgment) mechanism mode for SIP 1xx reliable responses. ▪ **[0]** Disable ▪ **[1]** Supported (default) ▪ **[2]** Required **Notes:** ▪ The Supported and Required headers contain the '100rel' tag.

Parameter	Description
	▪ The device sends PRACK messages if 180/183 responses are received with '100rel' in the Supported or Required headers. ▪ The parameter is applicable only to the Gateway application.
Enable Early Media `early-media` [EnableEarlyMedia]	Global parameter that enables the Early Media feature for sending media (e.g., ringing) before the call is established. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_EnableEarlyMedia) or Tel Profiles. For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387 or in the Tel Profile table, see Configuring Tel Profiles on page 384. **Notes:** ▪ If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile. ▪ The parameter is applicable only to the Gateway application.
Enable Early 183 `early-183` [EnableEarly183]	Global parameter that enables the device to send SIP 183 responses with SDP to the IP upon receipt of INVITE messages. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_EnableEarly183). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see Configuring IP Profiles on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
[IgnoreAlertAfterEarlyMedia]	Defines the device's interworking of Alerting messages for IP-to-Tel calls (ISDN). It determines whether the device sends a 180 Ringing response to the caller after the device sends a 183 Session Progress response to the caller. The 180 Ringing response indicates that the INVITE has been received by the ISDN side and that alerting is taking place (i.e., ISDN Progress message), indicating to the IP PBX to play a ringback tone. The 183 Session Progress response allows an early media session to be established prior to the call being answered, for example, to hear a ring tone, busy tone or recorded announcement. ▪ [0] = (Default) Disable. If the device sends a 183 response with SDP (due to a received ISDN Progress or Proceeding with PI messages, i.e., a ring tone, busy tone or recorded announcement played to the ISDN side) and an Alerting message is then received from the ISDN side (with or without Progress Indicator), the device also sends a 180 Ringing response to the caller. Therefore, in this case, early media is played to the ISDN side and then the ringback tone is played by the IP PBX. ▪ [1] = Enable. If the device sends a 183 response with SDP (due to a received ISDN Progress or Proceeding with PI messages) and an Alerting message is then received from the ISDN side (with or without Progress Indicator), the device does not send a 180 Ringing response to the caller and the voice channel remains open. Therefore, in this case, early media is played to the ISDN side and a ringback tone is not played by the IP PBX. **Note:** The parameter is applicable only if the EnableEarlyMedia parameter is set to 1 (i.e., enabled).
183 Message Behavior `183-msg-behavior` [SIP183Behaviour]	Defines the ISDN message that is sent when the 183 Session Progress message is received for IP-to-Tel calls. ▪ [0] Progress = (Default) The device sends a Progress message.

Parameter	Description
	- [1] Alert = The device sends an Alerting message (upon receipt of a 183 response) instead of an ISDN Progress message. **Note:** The parameter is applicable only to the Gateway application.
[ReleaseIP2ISDNCallOnProgressWithCause]	Typically, if an Q.931 Progress message with a Cause is received from the PSTN for an outgoing IP-to-ISDN call and the EnableEarlyMedia parameter is set to 1 (i.e., the Early Media feature is enabled), the device interworks the Progress to 183 + SDP to enable the originating party to hear the PSTN announcement about the call failure. Conversely, if EnableEarlyMedia is set to 0, the device disconnects the call by sending a SIP 4xx response to the originating party. However, if the ReleaseIP2ISDNCallOnProgressWithCause parameter is set to 1, then the device sends a SIP 4xx response even if the EnableEarlyMedia parameter is set to 1. - [0] = (Default) If a Progress with Cause message is received from the PSTN for an outgoing IP-to-ISDN call, the device does not disconnect the call by sending a SIP 4xx response to the originating party. - [1] = The device sends a SIP 4xx response when the EnableEarlyMedia parameter is set to 0. - [2] = The device always sends a SIP 4xx response, even if he EnableEarlyMedia parameter is set to 1.
Session-Expires Time `session-expires-time` [SIPSessionExpires]	Defines the numerical value sent in the Session-Expires header in the first INVITE request or response (if the call is answered). The valid range is 1 to 86,400 sec. The default is 0 (i.e., the Session-Expires header is disabled). **Note:** The parameter is applicable only to the Gateway application.
Minimum Session-Expires `min-session-expires` [MinSE]	Defines the time (in seconds) that is used in the Min-SE header. This header defines the minimum time that the user agent refreshes the session. The valid range is 10 to 100,000. The default is 90. **Note:** The parameter is applicable only to the Gateway application.
Session Expires Disconnect Time `session-exp-disconnect-time` [SessionExpiresDisconnectTime]	Defines a session expiry timeout. The new session expiry timeout is calculated by subtracting the configured value from the original timeout as specified in the Session-Expires header. However, the new timeout must be greater than or equal to one-third (1/3) of the Session-Expires value. If the refresher does not send a refresh request within the new timeout, the device disconnects the session (i.e., sends a SIP BYE). For example, if you configure the parameter to 32 seconds and the Session-Expires value is 180 seconds, the session timeout occurs 148 seconds (i.e., 180 minus 32) after the last session refresh. If the Session-Expires header value is 90 seconds, the timeout occurs 60 seconds after the last refresh. This is because 90 minus 32 is 58 seconds, which is less than one third of the Session-Expires value (i.e., 60/3 is 30, and 90 minus 30 is 60). The valid range is 0 to 32 (in seconds). The default is 32.
Session Expires Method `session-exp-method` [SessionExpiresMethod]	Determines the SIP method used for session-timer updates. - **[0]** Re-INVITE = (Default) Uses re-INVITE messages for session-timer updates. - **[1]** UPDATE = Uses UPDATE messages. **Notes:**

Parameter	Description
	- The parameter is applicable only to the Gateway application. - The device can receive session-timer refreshes using both methods. - The UPDATE message used for session-timer is excluded from the SDP body.
[RemoveToTagInFailureResponse]	Determines whether the device removes the ‘to’ header tag from final SIP failure responses to INVITE transactions. - **[0]** = (Default) Do not remove tag. - **[1]** = Remove tag.
[EnableRTCPAttribute]	Enables the use of the 'rtcp' attribute in the outgoing SDP. - **[0]** = Disable (default) - **[1]** = Enable **Note:** The parameter is applicable only to the Gateway application.
[OPTIONSUserPart]	Defines the user part value of the Request-URI for outgoing SIP OPTIONS requests. If no value is configured, the configuration parameter ‘Username’ valueis used. A special value is ‘empty’, indicating that no user part in the Request-URI (host part only) is used. The valid range is a 30-character string. By default, this value is not defined.
Trunk Status Reporting Mode `configure voip/gw digitalgw digital-gw-parameters/trunk-status-reporting` [TrunkStatusReportingMode]	Enables the device to not respond to received SIP OPTIONS messages from, and/or not to send keep-alive messages to, a proxy server associated with Trunk Group ID 1 if all its member trunks are down. - [0] Disable (default) = Device responds to SIP OPTIONS messages from, and sends keep-alive messages to, a proxy server associated with Trunk Group ID 1 if all its member trunks are down. - [1] Don’t reply OPTIONS = The device does not respond to SIP OPTIONS received from the proxy associated with Trunk Group 1 when all its trunks are down. - [2] Don’t send Keep-Alive = The device does not send keep-alive messages to the proxy associated with Trunk Group 1 when all its trunks are down. - [3] Don’t Reply and Send = Both options [1] and [2] are applied. **Notes:** - When the parameter is set to not respond to SIP OPTIONS received from the proxy, it is applicable only if the OPTIONS message does not include a user part in the Request-URI. - The proxy server is determined by the Proxy Set that is associated with the Serving IP Group defined for the Trunk Group in the Trunk Group Settings table.
Fax Signaling Method `fax-sig-method` [IsFaxUsed]	Global parameter that defines the SIP signaling method for establishing and transmitting a fax session when the device detects a fax. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_IsFaxUsed). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.

Parameter	Description
[HandleG711asVBD]	Enables the handling of G.711 as a G.711 Voice Band Data (VBD) coder. ▪ **[0]** = (Default) Disable. The device negotiates G.711 as a regular audio coder and sends an answer only with G.729 coder. For example, if the device is configured with G.729 and G.711 VBD coders and it receives an INVITE with an SDP offer containing G.729 and "regular" G.711 coders, it sends an SDP answer containing only the G.729 coder. ▪ **[1]** = Enable. The device assumes that the G.711 coder received in the INVITE SDP offer is a VBD coder. For example, if the device is configured with G.729 and G.711 VBD coders and it receives an INVITE with an SDP offer containing G.729 and "regular" G.711 coders, it sends an SDP answer containing G.729 and G.711 VBD coders, allowing a subsequent bypass (passthrough) session if fax/modem signals are detected during the call. **Note:** The parameter is applicable only if G.711 VBD coder(s) with regular G.711 payload types 0 or 8 are configured for the device (using the CodersGroup parameter).
`fax-vbd-behvr` [FaxVBDBehavior]	Determines the device's fax transport behavior when G.711 VBD coder is negotiated at call start. ▪ **[0]** = (Default) If the device is configured with a VBD coder (see the CodersGroup parameter) and is negotiated OK at call start, then both fax and modem signals are sent over RTP using the bypass payload type (and no mid-call VBD or T.38 Re-INVITEs occur). ▪ **[1]** = If the IsFaxUsed parameter is set to 1, the channel opens with the FaxTransportMode parameter set to 1 (relay). This is required to detect mid-call fax tones and to send T.38 Re-INVITE messages upon fax detection. If the remote party supports T.38, the fax is relayed over T.38. **Notes:** ▪ If VBD coder negotiation fails at call start and if the IsFaxUsed parameter is set to 1 (or 3), then the channel opens with the FaxTransportMode parameter set to 1 (relay) to allow future detection of fax tones and sending of T.38 Re-INVITES. In such a scenario, the FaxVBDBehavior parameter has no effect. ▪ This feature can be used only if the remote party supports T.38 fax relay; otherwise, the fax fails.
[NoAudioPayloadType]	Defines the payload type of the outgoing SDP offer. The valid value range is 96 to 127 (dynamic payload type). The default is 0 (i.e. NoAudio is not supported). For example, if set to 120, the following is added to the INVITE SDP: `a=rtpmap:120 NoAudio/8000\r\n` **Note:** For incoming SDP offers, NoAudio is always supported.

Parameter	Description
SIP Transport Type `app-sip-transport-type` [SIPTransportType]	Determines the default transport layer for outgoing SIP calls initiated by the device. - **[0]** UDP (default) - **[1]** TCP - **[2]** TLS (SIPS) **Notes:** - It's recommended to use TLS for communication with a SIP Proxy and not for direct device-to-device communication. - For received calls (i.e., incoming), the device accepts all these protocols. - The value of the parameter is also used by the SAS application as the default transport layer for outgoing SIP calls.
Display Default SIP Port `display-default-sip-port` [DisplayDefaultSIPPort]	Enables the device to add the default SIP port 5060 (UDP/TCP) or 5061 (TLS) to outgoing messages that are received without a port. This condition also applies to manipulated messages where the resulting message has no port number. The device adds the default port number to the following SIP headers: Request-Uri, To, From, P-Asserted-Identity, P-Preferred-Identity, and P-Called-Party-ID. If the message is received with a port number other than the default, for example, 5070, the port number is not changed. An example of a SIP From header with the default port is shown below: `From: <sip:+4000@10.8.4.105:`**`5060`**`;user=phone>;tag=f25419a96a;epid=009FAB8F3E` - **[0]** Disable (default) - **[1]** Enable
Enable SIPS `enable-sips` [EnableSIPS]	Enables secured SIP (SIPS URI) connections over multiple hops. - **[0]** Disable (default) - **[1]** Enable When the SIPTransportType parameter is set to 2 (i.e., TLS) and the parameter EnableSIPS is disabled, TLS is used for the next network hop only. When the parameter SIPTransportType is set to 2 or 1 (i.e., TCP or TLS) and EnableSIPS is enabled, TLS is used through the entire connection (over multiple hops). **Note:** If the parameter is enabled and the parameter SIPTransportType is set to 0 (i.e., UDP), the connection fails.
Enable TCP Connection Reuse `tcp-conn-reuse` [EnableTCPConnectionReuse]	Enables the reuse of the same TCP connection for all calls to the same destination. - **[0]** Disable = Uses a separate TCP connection for each call. - **[1]** Enable = (Default) Uses the same TCP connection for all calls. **Notes:** - - For the SAS application, this feature is configured using the SASConnectionReuse parameter.
Fake TCP alias `fake-tcp-alias` [FakeTCPalias]	Enables the re-use of the same TCP/TLS connection for sessions with the same user, even if the "alias" parameter is not present in the SIP Via header of the first INVITE.

Parameter	Description
	▪ **[0]** Disable = (Default) TCP/TLS connection reuse is done only if the "alias" parameter is present in the Via header of the first INVITE. ▪ **[1]** Enable **Note:** To enable TCP/TLS connection re-use, set the EnableTCPConnectionReuse parameter to 1.
Reliable Connection Persistent Mode `reliable-conn-persistent` [ReliableConnectionPersistentMode]	Enables setting of all TCP/TLS connections as persistent and therefore, not released. ▪ **[0]** = (Default) Disable. All TCP connections (except those that are set to a proxy IP) are released if not used by any SIP dialog\transaction. ▪ **[1]** = Enable - TCP connections to all destinations are persistent and not released unless the device reaches 70% of its maximum TCP resources. While trying to send a SIP message connection, reuse policy determines whether live connections to the specific destination are re-used. Persistent TCP connection ensures less network traffic due to fewer setting up and tearing down of TCP connections and reduced latency on subsequent requests due to avoidance of initial TCP handshake. For TLS, persistent connection may reduce the number of costly TLS handshakes to establish security associations, in addition to the initial TCP connection set up. **Note:** If the destination is a Proxy server, the TCP/TLS connection is persistent regardless of the settings of the parameter.
TCP Timeout `tcp-timeout` [SIPTCPTimeout]	Defines the Timer B (INVITE transaction timeout timer) and Timer F (non-INVITE transaction timeout timer), as defined in RFC 3261, when the SIP transport type is TCP. The valid range is 0 to 40 sec. The default is 64 * SipT1Rtx parameter value. For example, if SipT1Rtx is set to 500 msec, then the default of SIPTCPTimeout is 32 sec.
SIP Destination Port `sip-dst-port` [SIPDestinationPort]	Defines the SIP destination port for sending initial SIP requests. The valid range is 1 to 65534. The default port is 5060. **Note:** SIP responses are sent to the port specified in the Via header.
Use user=phone in SIP URL `user=phone-in-url` [IsUserPhone]	Determines whether the 'user=phone' string is added to the SIP URI and SIP To header. ▪ **[0]** No = 'user=phone' string is not added. ▪ **[1]** Yes = (Default) 'user=phone' string is part of the SIP URI and SIP To header.
Use user=phone in From Header `phone-in-from-hdr` [IsUserPhoneInFrom]	Determines whether the 'user=phone' string is added to the From and Contact SIP headers. ▪ **[0]** No = (Default) Doesn't add 'user=phone' string. ▪ **[1]** Yes = 'user=phone' string is part of the From and Contact headers.
Use Tel URI for Asserted Identity `uri-for-assert-id` [UseTelURIForAssertedID]	Determines the format of the URI in the P-Asserted-Identity and P-Preferred-Identity headers. ▪ **[0]** Disable = (Default) 'sip:' ▪ **[1]** Enable = 'tel:'

<table>
<tr><th>Parameter</th><th>Description</th></tr>
<tr><td>Tel to IP No Answer Timeout
`tel2ip-no-ans-timeout`
[IPAlertTimeout]</td><td>Defines the time (in seconds) that the device waits for a 200 OK response from the called party (IP side) after sending an INVITE message, for Tel-to-IP calls. If the timer expires, the call is released.
The valid range is 0 to 3600. The default is 180.</td></tr>
<tr><td>Enable Remote Party ID
`remote-party-id`
[EnableRPIheader]</td><td>Enables Remote-Party-Identity headers for calling and called numbers for Tel-to-IP calls.
▪ [0] Disable (default).
▪ [1] Enable = Remote-Party-Identity headers are generated in SIP INVITE messages for both called and calling numbers.</td></tr>
<tr><td>Enable History-Info Header
`hist-info-hdr`
[EnableHistoryInfo]</td><td>Enables usage of the History-Info header.
▪ [0] Disable (default)
▪ [1] Enable
User Agent Client (UAC) Behavior:
▪ Initial request: The History-Info header is equal to the Request-URI. If a PSTN Redirect number is received, it is added as an additional History-Info header with an appropriate reason.
▪ Upon receiving the final failure response, the device copies the History-Info as is, adds the reason of the failure response to the last entry, and concatenates a new destination to it (if an additional request is sent). The order of the reasons is as follows:
a. Q.850 Reason
b. SIP Reason
c. SIP Response code
▪ Upon receiving the final response (success or failure), the device searches for a Redirect reason in the History-Info (i.e., 3xx/4xx SIP reason). If found, it is passed to ISDN according to the following table:
<table>
<tr><th>SIP Reason Code</th><th>ISDN Redirecting Reason</th></tr>
<tr><td>302 - Moved Temporarily</td><td>Call Forward Universal (CFU)</td></tr>
<tr><td>408 - Request Timeout</td><td rowspan="3">Call Forward No Answer (CFNA)</td></tr>
<tr><td>480 - Temporarily Unavailable</td></tr>
<tr><td>487 - Request Terminated</td></tr>
<tr><td>486 - Busy Here</td><td rowspan="2">Call Forward Busy (CFB)</td></tr>
<tr><td>600 - Busy Everywhere</td></tr>
</table>
▪ If history reason is a Q.850 reason, it is translated to the SIP reason (according to the SIP-ISDN tables) and then to ISDN Redirect reason according to the table above.
User Agent Server (UAS) Behavior:
▪ The History-Info header is sent only in the final response.
▪ Upon receiving a request with History-Info, the UAS checks the policy in the request. If a 'session', 'header', or 'history' policy tag is found, the (final) response is sent without History-Info; otherwise, it is copied from the request.</td></tr>
<tr><td>Use Tgrp Information
`use-tgrp-inf`
[UseSIPTgrp]</td><td>Determines whether the SIP 'tgrp' parameter is used. This SIP parameter specifies the Trunk Group to which the call belongs (according to RFC 4904). For example, the SIP message below indicates that the call belongs to Trunk Group ID 1:</td></tr>
</table>

Parameter	Description
	INVITE sip::+16305550100;tgrp=1;trunk-context=example.com@10.1.0.3;user=phone SIP/2.0 ▪ [0] Disable = (Default) The 'tgrp' parameter isn't used. ▪ [1] Send Only = The Trunk Group number or name (configured in the Trunk Group Settings table) is added to the 'tgrp' parameter value in the Contact header of outgoing SIP messages. If a Trunk Group number / name is not associated with the call, the 'tgrp' parameter isn't included. If a 'tgrp' value is specified in incoming messages, it is ignored. ▪ [2] Send and Receive = The functionality of outgoing SIP messages is identical to the functionality described for option [1]. In addition, for incoming SIP INVITEs, if the Request-URI includes a 'tgrp' parameter, the device routes the call according to that value (if possible). The Contact header in the outgoing SIP INVITE (Tel-to-IP call) contains "tgrp=<source trunk group ID>;trunk-context=<gateway IP address>". The <source trunk group ID> is the Trunk Group ID where incoming calls from Tel is received. For IP-Tel calls, the SIP 200 OK device's response contains "tgrp=<destination trunk group ID>;trunk-context=<gateway IP address>". The <destination trunk group ID> is the Trunk Group ID used for outgoing Tel calls. The <gateway IP address> in "trunk-context" can be configured using the SIPGatewayName parameter. ▪ [3] Hotline = Interworks the hotline "Off Hook Indicator" parameter between SIP and ISDN: ✓ For IP-to-ISDN calls: - The device interworks the SIP tgrp=hotline parameter (received in INVITE) to ISDN Setup with the Off Hook Indicator IE of "Voice", and "Speech" Bearer Capability IE. Note that the Off Hook Indicator IE is described in UCR 2008 specifications. - The device interworks the SIP tgrp=hotline-ccdata parameter (received in INVITE) to ISDN Setup with an Off Hook Indicator IE of "Data", and with "Unrestricted 64k" Bearer Capability IE. The following is an example of the INVITE with tgrp=hotline-ccdata: INVITE sip:1234567;tgrp=hotline-ccdata;trunk-context=dsn.mil@example.com ✓ For ISDN-to-IP calls: - The device interworks ISDN Setup with an Off Hook Indicator of "Voice" to SIP INVITE with "tgrp=hotline;trunk-context=dsn.mil" in the Contact header. - The device interworks ISDN Setup with an Off Hook indicator of "Data" to SIP INVITE with "tgrp=hotline-ccdata;trunk-context=dsn.mil" in the Contact header. - If ISDN Setup does not contain an Off Hook Indicator IE and the Bearer Capability IE contains "Unrestricted 64k", the outgoing INVITE includes "tgrp=ccdata;trunk-context=dsn.mil". If the Bearer Capability IE contains "Speech", the INVITE in this case does not contain tgrp and trunk-context parameters. ▪ [4] Hotline Extended = Interworks the ISDN Setup message's hotline "OffHook Indicator" Information Element (IE) to SIP INVITE's Request-URI and Contact headers. (Note: For IP-to-ISDN calls, the device handles the call as described in option [3].)

Parameter	Description
	✓ The device interworks ISDN Setup with an Off Hook Indicator of “Voice” to SIP INVITE Request-URI and Contact header with “tgrp=hotline;trunk-context=dsn.mil”. ✓ The device interworks ISDN Setup with an Off Hook indicator of “Data” to SIP INVITE Request-URI and Contact header with “tgrp=hotline-ccdata;trunk-context=dsn.mil”. ✓ If ISDN Setup does not contain an Off Hook Indicator IE and the Bearer Capability IE contains “Unrestricted 64k”, the outgoing INVITE Request-URI and Contact header includes “tgrp=ccdata;trunk-context=dsn.mil”. If the Bearer Capability IE contains “Speech”, the INVITE in this case does not contain tgrp and trunk-context parameters. **Note:** IP-to-Tel configuration (using the PSTNPrefix parameter) overrides the 'tgrp' parameter in incoming INVITE messages.
TGRP Routing Precedence `tgrp-routing-prec` [TGRProutingPrecedence]	Determines the precedence method for routing IP-to-Tel calls - according to the IP to Trunk Group Routing table or according to the SIP 'tgrp' parameter. ▪ [0] = (Default) IP-to-Tel routing is determined by the IP to Trunk Group Routing table (PSTNPrefix parameter). If a matching rule is not found in this table, the device uses the Trunk Group parameters for routing the call. ▪ [1] = The device first places precedence on the 'tgrp' parameter for IP-to-Tel routing. If the received INVITE Request-URI does not contain the 'tgrp' parameter or if the Trunk Group number is not defined, the IP to Trunk Group Routing table is used for routing the call. Below is an example of an INVITE Request-URI with the 'tgrp' parameter, indicating that the IP call should be routed to Trunk Group 7: `INVITE sip:200;tgrp=7;trunk-context=example.com@10.33.2.68;user=phone SIP/2.0` **Notes:** ▪ For enabling routing based on the 'tgrp' parameter, the UseSIPTgrp parameter must be set to 2. ▪ For IP-to-Tel routing based on the 'dtg' parameter (instead of the 'tgrp' parameter), use the parameter UseBroadsoftDTG.
`use-dtg` [UseBroadsoftDTG]	Determines whether the device uses the 'dtg' parameter for routing IP-to-Tel calls to a specific Trunk Group. ▪ [0] Disable (default) ▪ [1] Enable When the parameter is enabled, if the Request-URI in the received SIP INVITE includes the 'dtg' parameter, the device routes the call to the Trunk Group according to its value. The parameter is used instead of the 'tgrp/trunk-context' parameters. The 'dtg' parameter appears in the INVITE Request-URI (and in the To header). For example, the received SIP message below routes the call to Trunk Group ID 56: `INVITE sip:123456@192.168.1.2;dtg=56;user=phone SIP/2.0` **Note:** If the Trunk Group is not found based on the 'dtg' parameter, the IP to Trunk Group Routing table is used instead for routing the call to the appropriate Trunk Group.

Parameter	Description

Enable GRUU
`enable-gruu`
[EnableGRUU]
	Determines whether the Globally Routable User Agent URIs (GRUU) mechanism is used, according to RFC 5627. This is used for obtaining a GRUU from a registrar and for communicating a GRUU to a peer within a dialog.
	▪ **[0]** Disable (default)
	▪ **[1]** Enable
	A GRUU is a SIP URI that routes to an instance-specific UA and can be reachable from anywhere. There are a number of contexts in which it is desirable to have an identifier that addresses a single UA (using GRUU) rather than the group of UA’s indicated by an Address of Record (AOR). For example, in call transfer where user A is talking to user B, and user A wants to transfer the call to user C. User A sends a REFER to user C:
	`REFER sip:C@domain.com SIP/2.0`
	`From: sip:A@domain.com;tag=99asd`
	`To: sip:C@domain.com`
	`Refer-To: (URI that identifies B's UA)`
	The Refer-To header needs to contain a URI that user C can use to place a call to user B. This call needs to route to the specific UA instance that user B is using to talk to user A. User B should provide user A with a URI that has to be usable by anyone. It needs to be a GRUU.
	▪ Obtaining a GRUU: The mechanism for obtaining a GRUU is through registrations. A UA can obtain a GRUU by generating a REGISTER request containing a Supported header field with the value “gruu”. The UA includes a “+sip.instance” Contact header parameter of each contact for which the GRUU is desired. This Contact parameter contains a globally unique ID that identifies the UA instance. The global unique ID is created from one of the following:
	✓ If the REGISTER is per the device’s client (endpoint), it is the MAC address concatenated with the phone number of the client.
	✓ If the REGISTER is per device, it is the MAC address only.
	✓ When using TP, “User Info” can be used for registering per endpoint. Thus, each endpoint can get a unique id – its phone number. The globally unique ID in TP is the MAC address concatenated with the phone number of the endpoint.
	If the remote server doesn’t support GRUU, it ignores the parameters of the GRUU. Otherwise, if the remote side also supports GRUU, the REGISTER responses contain the “gruu” parameter in each Contact header. The parameter contains a SIP or SIPS URI that represents a GRUU corresponding to the UA instance that registered the contact. The server provides the same GRUU for the same AOR and instance-id when sending REGISTER again after registration expiration. RFC 5627 specifies that the remote target is a GRUU target if its’ Contact URL has the "gr" parameter with or without a value.
	▪ Using GRUU: The UA can place the GRUU in any header field that can contain a URI. It must use the GRUU in the following messages: INVITE request, its 2xx response, SUBSCRIBE request, its 2xx response, NOTIFY request, REFER request and its 2xx response.

[IsCiscoSCEMode]	Determines whether a Cisco gateway exists at the remote side.

Parameter	Description
	▪ **[0]** = (Default) No Cisco gateway exists at the remote side. ▪ **[1]** = A Cisco gateway exists at the remote side. When a Cisco gateway exists at the remote side, the device must set the value of the 'annexb' parameter of the fmtp attribute in the SDP to 'no'. This logic is used if Silence Suppression for the used coder is configured to 2 (enable without adaptation). In this case, Silence Suppression is used on the channel but not declared in the SDP. **Note:** ▪ The parameter is applicable only to the Gateway application. ▪ The IsCiscoSCEMode parameter is applicable only when the selected coder is G.729.

User-Agent Information
`user-agent-info`
[UserAgentDisplayInfo]
	Defines the string that is used in the SIP User-Agent and Server response headers. When configured, the string <UserAgentDisplayInfo value>/software version' is used, for example:
	`User-Agent: myproduct/v.6.80A.227.005`
	If not configured, the default string, <AudioCodes product-name>/software version' is used, for example:
	`User-Agent: Audiocodes-Sip-Gateway-Mediant 500 E-SBC/v.6.80A.227.005`
	The maximum string length is 50 characters.
	Note: The software version number and preceding forward slash (/) cannot be modified. Therefore, it is recommended not to include a forward slash in the parameter's value (to avoid two forward slashes in the SIP header, which may cause problems).

SDP Session Owner `sdp-session-owner` [SIPSDPSessionOwner]	Defines the value of the Owner line ('o' field) in outgoing SDP messages. The valid range is a string of up to 39 characters. The default is "AudiocodesGW". For example: `o=AudiocodesGW 1145023829 1145023705 IN IP4 10.33.4.126`
`sdp-ver-nego` [EnableSDPVersionNegotiation]	Enables the device to ignore new SDP re-offers (from the media negotiation perspective) in certain scenarios (such as session expires). According to RFC 3264, once an SDP session is established, a new SDP offer is considered a new offer only when the SDP origin value is incremented. In scenarios such as session expires, SDP negotiation is irrelevant and thus, the origin field is not changed. Even though some SIP devices don't follow this behavior and don't increment the origin value even in scenarios where they want to re-negotiate, the device can assume that the remote party operates according to RFC 3264, and in cases where the origin field is not incremented, the device does not re-negotiate SDP capabilities. ▪ **[0]** Disable = (Default) The device negotiates any new SDP re-offer, regardless of the origin field. ▪ **[1]** Enable = The device negotiates only an SDP re-offer with an incremented origin field.
Subject `usr-def-subject` [SIPSubject]	Defines the Subject header value in outgoing INVITE messages. If not specified, the Subject header isn't included (default). The maximum length is up to 50 characters.

Parameter	Description
[CoderPriorityNegotiation]	Defines the priority for coder negotiation in the incoming SDP offer, between the device's or remote UA's coder list. ▪ [0] = (Default) Coder negotiation is given higher priority to the remote UA's list of supported coders. ▪ [1] = Coder negotiation is given higher priority to the device's (local) supported coders list. **Note:** The parameter is applicable only to the Gateway application.
Send All Coders on Retrieve `send-all-cdrs-on-rtrv` [SendAllCodersOnRetrieve]	Enables coder re-negotiation in the sent re-INVITE for retrieving an on-hold call. ▪ [0] Disable = (Default) Sends only the initially chosen coder when the call was first established and then put on-hold. ▪ [1] Enable = Includes all supported coders in the SDP of the re-INVITE sent to the call made un-hold (retrieved). The used coder is therefore, re-negotiated. The parameter is useful in the following call scenario example: 1 Party A calls party B and coder G.711 is chosen. 2 Party B is put on-hold while Party A blind transfers Party B to Party C. 3 Party C answers and Party B is made un-hold. However, as Party C supports only G.729 coder, re-negotiation of the supported coder is required. **Note:** The parameter is applicable only to the Gateway application.
Multiple Packetization Time Format `mult-ptime-format` [MultiPtimeFormat]	Determines whether the 'mptime' attribute is included in the outgoing SDP. ▪ **[0]** None = (Default) Disabled. ▪ **[1]** PacketCable = Includes the 'mptime' attribute in the outgoing SDP - PacketCable-defined format. The mptime' attribute enables the device to define a separate packetization period for each negotiated coder in the SDP. The 'mptime' attribute is only included if the parameter is enabled even if the remote side includes it in the SDP offer. Upon receipt, each coder receives its 'ptime' value in the following precedence: from 'mptime' attribute, from 'ptime' attribute, and then from default value.
[EnablePtime]	Determines whether the 'ptime' attribute is included in the SDP. ▪ **[0]** = Remove the 'ptime' attribute from SDP. ▪ **[1]** = (Default) Include the 'ptime' attribute in SDP.
3xx Behavior `3xx-behavior` [3xxBehavior]	Determines the device's behavior regarding call identifiers when a 3xx response is received for an outgoing INVITE request. The device can either use the same call identifiers (Call-ID, To, and From tags) or change them in the new initiated INVITE. ▪ **[0]** Forward = (Default) Use different call identifiers for a redirected INVITE message. ▪ **[1]** Redirect = Use the same call identifiers.
Enable P-Charging Vector `p-charging-vector` [EnablePChargingVector]	Enables the inclusion of the P-Charging-Vector header to all outgoing INVITE messages. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** The parameter is applicable only to the Gateway application.

Parameter	Description
Retry-After Time `retry-aftr-time` [RetryAfterTime]	Defines the time (in seconds) used in the Retry-After header when a 503 (Service Unavailable) response is generated by the device. The time range is 0 to 3,600. The default is 0.
Fake Retry After `fake-retry-after` [FakeRetryAfter]	Determines whether the device, upon receipt of a SIP 503 response without a Retry-After header, behaves as if the 503 response included a Retry-After header and with the period (in seconds) specified by the parameter. ▪ **[0]** Disable (default) ▪ Any positive value (in seconds) for defining the period When enabled, this feature allows the device to operate with Proxy servers that do not include the Retry-After SIP header in SIP 503 (Service Unavailable) responses to indicate an unavailable service. The Retry-After header is used with the 503 (Service Unavailable) response to indicate how long the service is expected to be unavailable to the requesting SIP client. The device maintains a list of available proxies, by using the Keep-Alive mechanism. The device checks the availability of proxies by sending SIP OPTIONS every keep-alive timeout to all proxies. If the device receives a SIP 503 response to an INVITE, it also marks that the proxy is out of service for the defined "Retry-After" period.
Enable P-Associated-URI Header `p-associated-uri-hdr` [EnablePAssociatedURIHeader]	Determines the device usage of the P-Associated-URI header. This header can be received in 200 OK responses to REGISTER requests. When enabled, the first URI in the P-Associated-URI header is used in subsequent requests as the From/P-Asserted-Identity headers value. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** P-Associated-URIs in registration responses is handled only if the device is registered per endpoint (using the User Information file).
Source Number Preference `src-nb-preference` [SourceNumberPreference]	Determines from which SIP header the source (calling) number is obtained in incoming INVITE messages. ▪ If not configured or if any string other than "From" or "Pai2" is configured, the calling number is obtained from a specific header using the following logic: **a.** P-Preferred-Identity header. **b.** If the above header is not present, then the first P-Asserted-Identity header is used. **c.** If the above header is not present, then the Remote-Party-ID header is used. **d.** If the above header is not present, then the From header is used. ▪ "**From**" = The calling number is obtained from the From header. ▪ "**Pai2**" = The calling number is obtained using the following logic: **a.** If a P-Preferred-Identity header is present, the number is obtained from it. **b.** If no P-Preferred-Identity header is present and two P-Asserted-Identity headers are present, the number is obtained from the second P-Asserted-Identity header. **c.** If only one P-Asserted-Identity header is present, the calling number is obtained from it. **Notes:**

Parameter	Description
	- The "From" and "Pai2" values are not case-sensitive. - Once a URL is selected, all the calling party parameters are set from this header. If P-Asserted-Identity is selected and the Privacy header is set to 'id', the calling number is assumed restricted.
`src-hdr-4-called-nb` [SelectSourceHeaderForCalledNumber]	Determines the SIP header used for obtaining the called number (destination) for IP-to-Tel calls. - [0] Request-URI header = (Default) Obtains the destination number from the user part of the Request-URI. - [1] To header = Obtains the destination number from the user part of the To header. - [2] P-Called-Party-ID header = Obtains the destination number from the P-Called-Party-ID header.
Enable Reason Header `reason-header` [EnableReasonHeader]	Enables the usage of the SIP Reason header. - **[0]** Disable - **[1]** Enable (default)
Gateway Name `gw-name` [SIPGatewayName]	Defines a name for the device (e.g., device123.com). This name is used as the host part of the SIP URI in the From header. If not specified, the device's IP address is used instead (default). **Notes:** - Ensure that the parameter value is the one with which the Proxy has been configured with to identify the device. - The parameter can also be configured for an IP Group (in the IP Group table).
[ZeroSDPHandling]	Determines the device's response to an incoming SDP that includes an IP address of 0.0.0.0 in the SDP's Connection Information field (i.e., "c=IN IP4 0.0.0.0"). - **[0]** = (Default) Sets the IP address of the outgoing SDP's c= field to 0.0.0.0. - **[1]** = Sets the IP address of the outgoing SDP c= field to the IP address of the device. If the incoming SDP doesn’t contain the "a=inactive" line, the returned SDP contains the "a=recvonly" line.
Enable Delayed Offer `delayed-offer` [EnableDelayedOffer]	Determines whether the device sends the initial INVITE message with or without an SDP. Sending the first INVITE without SDP is typically done by clients for obtaining the far-end's full list of capabilities before sending their own offer. (An alternative method for obtaining the list of supported capabilities is by using SIP OPTIONS, which is not supported by every SIP agent.) - **[0]** Disable = (Default) The device sends the initial INVITE message with an SDP. - **[1]** Enable = The device sends the initial INVITE message without an SDP.
[DisableCryptoLifeTimeInSDP]	Enables the device to send "a=crypto" lines without the lifetime parameter in the SDP. For example, if the SDP contains "a=crypto:12 AES_CM_128_HMAC_SHA1_80 inline:hhQe10yZRcRcpIFPkH5xYY9R1de37ogh9G1MpvNp|2^31", it removes the lifetime parameter "2^31". - **[0]** Disable (default) - **[1]** Enable
Enable Contact Restriction `contact-restriction`	Determines whether the device sets the Contact header of outgoing INVITE requests to ‘anonymous’ for restricted calls.

Parameter	Description
[EnableContactRestriction]	▪ **[0]** Disable (default) ▪ **[1]** Enable
`anonymous-mode` [AnonymousMode]	Determines whether the device's IP address is used as the URI host part instead of "anonymous.invalid" in the INVITE's From header for Tel-to-IP calls. ▪ [0] = (Default) If the device receives a call from the Tel with blocked caller ID, it sends an INVITE with From: “anonymous”<anonymous@anonymous.invalid> ▪ [1] = The device's IP address is used as the URI host part instead of "anonymous.invalid". The parameter may be useful, for example, for service providers who identify their SIP Trunking customers by their source phone number or IP address, reflected in the From header of the SIP INVITE. Therefore, even customers blocking their Caller ID can be identified by the service provider. Typically, if the device receives a call with blocked Caller ID from the PSTN side (e.g., Trunk connected to a PBX), it sends an INVITE to the IP with a From header as follows: From: “anonymous” <anonymous@anonymous.invalid>. This is in accordance with RFC 3325. However, when the parameter is set to 1, the device replaces the "anonymous.invalid" with its IP address.
`p-assrtd-usr-name` [PAssertedUserName]	Defines a 'representative number' (up to 50 characters) that is used as the user part of the Request-URI in the P-Asserted-Identity header of an outgoing INVITE for Tel-to-IP calls. The default is null.
[UseAORInReferToHeader]	Defines the source for the SIP URI set in the Refer-To header of outgoing REFER messages. ▪ **[0]** = (Default) Use SIP URI from Contact header of the initial call. ▪ **[1]** = Use SIP URI from To/From header of the initial call.
Enable User-Information Usage `user-inf-usage` [EnableUserInfoUsage]	Enables the usage of the User Information, which is loaded to the idevice> in the User Information Auxiliary file. For more nformation on User Information, see "User Information File" on page 659. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** For the parameter to take effect, a device reset is required.
[HandleReasonHeader]	Determines whether the device uses the value of the incoming SIP Reason header for Release Reason mapping. ▪ **[0]** = Disregard Reason header in incoming SIP messages. ▪ **[1]** = (Default) Use the Reason header value for Release Reason mapping.
[EnableSilenceSuppInSDP]	Determines the device's behavior upon receipt of SIP Re-INVITE messages that include the SDP's 'silencesupp:off' attribute. ▪ **[0]** = (Default) Disregard the 'silecesupp' attribute. ▪ **[1]** = Handle incoming Re-INVITE messages that include the 'silencesupp:off' attribute in the SDP as a request to switch to the Voice-Band-Data (VBD) mode. In addition, the device includes the attribute 'a=silencesupp:off' in its SDP offer. **Note:** The parameter is applicable only if the G.711 coder is used.
[EnableRport]	Enables the usage of the 'rport' parameter in the Via header. ▪ **[0]** = Disabled (default)

Parameter	Description
	▪ **[1]** = Enabled The device adds an 'rport' parameter to the Via header of each outgoing SIP message. The first Proxy that receives this message sets the 'rport' value of the response to the actual port from where the request was received. This method is used, for example, to enable the device to identify its port mapping outside a NAT. If the Via header doesn't include the 'rport' parameter, the destination port of the response is obtained from the host part of the Via header. If the Via header includes the 'rport' parameter without a port value, the destination port of the response is the source port of the incoming request. If the Via header includes 'rport' with a port value (e.g., rport=1001), the destination port of the response is the port indicated in the 'rport' parmeter.
Enable X-Channel Header `x-channel-header` [XChannelHeader]	Determines whether the SIP X-Channel header is added to SIP messages for providing information on the physical Trunk/B-channel on which the call is received or placed. ▪ **[0]** Disable = (Default) X-Channel header is not used. ▪ **[1]** Enable = X-Channel header is generated by the device and sent in INVITE messages and 180, 183, and 200 OK SIP responses. The header includes the Trunk number, Bchannel, and the device's IP address. For example, 'x-channel: DS/DS1-5/8;IP=192.168.13.1', where: ✓ 'DS/DS-1' is a constant string ✓ '5' is the Trunk number ✓ '8' is the B-channel ✓ 'IP=192.168.13.1' is the device's IP address
Progress Indicator to IP `prog-ind-2ip` [ProgressIndicator2IP]	Global parameter that defines the progress indicator (PI) sent to the IP. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_ProgressIndicator2IP) or Tel Profiles. For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see Configuring IP Profiles on page 387 or in the Tel Profile table, see Configuring Tel Profiles on page 384. **Note:** If this functionality is configured for a specific profile, the settings of this global parameter is ignored for calls associated with the profile.
[EnableRekeyAfter181]	Enables the device to send a re-INVITE with a new (different) SRTP key (in the SDP) if a SIP 181 response is received ("call is being forwarded"). The re-INVITE is sent immediately upon receipt of the 200 OK (when the call is answered). ▪ **[0]** = Disable (default) ▪ **[1]** = Enable **Note:** The parameter is applicable only if SRTP is used.
[NumberOfActiveDialogs]	Defines the maximum number of concurrent, outgoing SIP REGISTER dialogs. The parameter is used to control the registration rate. The valid range is 1 to 20. The default is 20. **Notes:** ▪ Once a 200 OK is received in response to a REGISTER message, the REGISTER message is not considered in this maximum count limit.

Parameter	Description
	▪ The parameter applies only to outgoing REGISTER messages (i.e., incoming is unlimited).
[TransparentCoderOnDataCall]	▪ [0] = (Default) Only use coders from the coder list. ▪ [1] = Use Transparent coder for data calls (according to RFC 4040). The Transparent coder can be used on data calls. When the device receives a Setup message from the ISDN with 'TransferCapabilities = data', it can initiate a call using the coder 'Transparent' (even if the coder is not included in the coder list). The initiated INVITE includes the following SDP attribute: `a=rtpmap:97 CLEARMODE/8000` The default payload type is set according to the CodersGroup parameter. If the Transparent coder is not defined, the default is set to 56. The payload type is negotiated with the remote side, i.e., the selected payload type is according to the remote side selection. The receiving device must include the 'Transparent' coder in its coder list.
Network Node ID net-node-id [NetworkNodeId]	Defines the Network Node Identifier of the device for Avaya UCID. The valid value range is1 to 0x7FFF. The default is 0. **Notes:** ▪ To use this feature, you must set the parameter to any value other than 0. ▪ To enable the generation by the device of the Avaya UCID value and adding it to the outgoing INVITE sent to the IP Group (Avaya entity), use the IP Group table's parameter 'UUI Format'.
Default Release Cause `dflt-release-cse` [DefaultReleaseCause]	Defines the default Release Cause (sent to IP) for IP-to-Tel calls when the device initiates a call release and an explicit matching cause for this release is not found. The default release cause is NO_ROUTE_TO_DESTINATION (3). Other common values include NO_CIRCUIT_AVAILABLE (34), DESTINATION_OUT_OF_ORDER (27), etc. **Notes:** ▪ The default release cause is described in the Q.931 notation and is translated to corresponding SIP 40x or 50x values (e.g., 3 to SIP 404, and 34 to SIP 503). ▪ When the Trunk is disconnected or is not synchronized, the internal cause is 27. This cause is mapped, by default, to SIP 502. ▪ For mapping SIP-to-Q.931 and Q.931-to-SIP release causes, see Configuring Release Cause Mapping on page 457. ▪ For a list of SIP responses-Q.931 release cause mapping, see Alternative Routing to Trunk upon Q.931 Call Release Cause Code on page 492.
Enable Microsoft Extension `microsoft-ext` [EnableMicrosoftExt]	Enables the modification of the called and calling number for numbers received with Microsoft's proprietary "ext=xxx" parameter in the SIP INVITE URI user part. Microsoft Office Communications Server sometimes uses this proprietary parameter to indicate the extension number of the called or calling party. ▪ **[0]** Disable (default) ▪ **[1]** Enable For example, if a calling party makes a call to telephone number 622125519100 Ext. 104, the device receives the SIP INVITE (from Microsoft's application) with the URI user part as INVITE

Parameter	Description
	sip:622125519100;ext=104@10.1.1.10 (or INVITE tel:622125519100;ext=104). If the parameter EnableMicrosofExt is enabled, the device modifies the called number by adding an "e" as the prefix, removing the "ext=" parameter, and adding the extension number as the suffix (e.g., **e**622125519100**104**). Once modified, the device can then manipulate the number further, using the Number Manipulation tables to leave only the last 3 digits (for example) for sending to a PBX.
[UseSIPURIForDiversionHeader]	Defines the URI format in the SIP Diversion header. ▪ **[0]** = 'tel:' (default) ▪ **[1]** = 'sip:'
[TimeoutBetween100And18x]	Defines the timeout (in msec) between receiving a 100 Trying response and a subsequent 18x response. If a 18x response is not received within this timeout period, the call is disconnected. The valid range is 0 to 180,000 (i.e., 3 minutes). The default is 32000 (i.e., 32 sec).
[EnableImmediateTrying]	Determines if and when the device sends a 100 Trying in response to an incoming INVITE request. ▪ [0] = 100 Trying response is sent upon receipt of a Proceeding message from the PSTN ▪ [1] = (Default) 100 Trying response is sent immediately upon receipt of INVITE request.
[TransparentCoderPresentation]	Determines the format of the Transparent coder representation in the SDP. ▪ [0] = clearmode (default) ▪ [1] = X-CCD
[IgnoreRemoteSDPMKI]	Determines whether the device ignores the Master Key Identifier (MKI) if present in the SDP received from the remote side. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Comfort Noise Generation Negotiation `com-noise-gen-nego` [ComfortNoiseNegotiation]	Enables negotiation and usage of Comfort Noise (CN) for Gateway calls. ▪ [0] Disable ▪ [1] Enable (default) The use of CN is indicated by including a payload type for CN on the media description line of the SDP. The device can use CN with a codec whose RTP time stamp clock rate is 8,000 Hz (G.711/G.726). The static payload type 13 is used. The use of CN is negotiated between sides. Therefore, if the remote side doesn't support CN, it is not used. Regardless of the device's settings, it always attempts to adapt to the remote SIP UA's request for CNG, as described below. To determine CNG support, the device uses the ComfortNoiseNegotiation parameter and the codec's SCE (silence suppression setting) using the CodersGroup parameter. If the ComfortNoiseNegotiation parameter is enabled, then the following occurs: ▪ If the device is the initiator, it sends a “CN” in the SDP only if the SCE of the codec is enabled. If the remote UA responds with a “CN” in the SDP, then CNG occurs; otherwise, CNG does not occur.

Parameter	Description
	- If the device is the receiver and the remote SIP UA does not send a "CN" in the SDP, then no CNG occurs. If the remote side sends a "CN", the device attempts to be compatible with the remote side and even if the codec's SCE is disabled, CNG occurs. If the ComfortNoiseNegotiation parameter is disabled, then the device does not send "CN" in the SDP. However, if the codec's SCE is enabled, then CNG occurs. **Note:** The parameter is applicable only to the Gateway application.
`sdp-ecan-frmt` [SDPEcanFormat]	Defines the echo canceller format in the outgoing SDP. The 'ecan' attribute is used in the SDP to indicate the use of echo cancellation. ▪ **[0]** = (Default) The 'ecan' attribute appears on the 'a=gpmd' line. ▪ **[1]** = The 'ecan' attribute appears as a separate attribute. ▪ **[2]** = The 'ecan' attribute is not included in the SDP. ▪ **[3]** = The 'ecan' attribute and the 'vbd' parameter are not included in the SDP. **Note:** The parameter is applicable only when the IsFaxUsed parameter is set to 2, and for re-INVITE messages generated by the device as result of modem or fax tone detection.
First Call Ringback Tone ID `1st-call-rbt-id` [FirstCallRBTId]	Defines the index of the first ringback tone in the CPT file. This option enables an Application server to request the device to play a distinctive ringback tone to the calling party according to the destination of the call. The tone is played according to the Alert-Info header received in the 180 Ringing SIP response (the value of the Alert-Info header is added to the value of the parameter). The valid range is -1 to 1,000. The default is -1 (i.e., play standard ringback tone). **Notes:** ▪ It is assumed that all ringback tones are defined in sequence in the CPT file. ▪ In case of an MLPP call, the device uses the value of the parameter plus 1 as the index of the ringback tone in the CPT file (e.g., if this value is set to 1, then the index is 2, i.e., 1 + 1).
PSTN Alert Timeout `pstn-alert-timeout` [PSTNAlertTimeout]	Defines the Alert Timeout (in seconds) for calls sent to the PSTN. This timer is used between the time a Setup message is sent to the Tel side (IP-to-Tel call establishment) and a Connect message is received. If an Alerting message is received, the timer is restarted. If the timer expires before the call is answered, the device disconnects the call and sends a SIP 408 request timeout response to the SIP party that initiated the call. The valid value range is 1 to 600 (in seconds). The default is 180. **Note:** If per trunk configuration (using TrunkPSTNAlertTimeout) is set to other than default, the PSTNAlertTimeout parameter value is overridden.
RTP Only Mode `rtp-only-mode` [RTPOnlyMode]	Enables the device to send and receive RTP packets to and from remote endpoints without the need to establish a SIP session. The remote IP address is determined according to the Tel-to-IP Routing table (Prefix parameter). The port is the same port as the local RTP port (configured by the BaseUDPPort parameter and the channel on which the call is received). ▪ **[0]** Disable (default)

Parameter	Description
	- **[1]** Transmit & Receive = Send and receive RTP packets. - **[2]** Transmit Only= Send RTP packets only. - **[3]** Receive Only= Receive RTP packets only. **Notes:** - To configure the RTP Only mode per trunk, use the RTPOnlyModeForTrunk_x parameter. - If per trunk configuration (using the RTPOnlyModeForTrunk_ID parameter) is set to a value other than the default, the RTPOnlyMode parameter value is ignored.
[RTPOnlyModeForTrunk_x]	Enables the RTP Only feature per trunk. The x in the parameter name denotes the trunk number, where 0 is Trunk 1. For a description of the parameter, see the RTPOnlyMode parameter. **Note:** For using the global parameter (i.e., setting the RTP Only feature for all trunks), set the parameter to -1 (default).
Media IP Version Preference `media-ip-ver-pref` [MediaIPVersionPreference]	Global parameter that defines the preferred RTP media IP addressing version (IPv4 or IPv6) for outgoing SIP calls. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_MediaIPVersionPreference). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see Configuring IP Profiles on page 387.
SIT Q850 Cause [SITQ850Cause]	Defines the Q.850 cause value specified in the SIP Reason header that is included in a 4xx response when a Special Information Tone (SIT) is detected on an IP-to-Tel call. The valid range is 0 to 127. The default is 34. **Notes:** - For mapping specific SIT tones, you can use the SITQ850CauseForNC, SITQ850CauseForIC, SITQ850CauseForVC, and SITQ850CauseForRO parameters.
SIT Q850 Cause For NC [SITQ850CauseForNC]	Defines the Q.850 cause value specified in the SIP Reason header that is included in a 4xx response when SIT-NC (No Circuit Found Special Information Tone) is detected from the Tel side for IP-to-Tel calls. The valid range is 0 to 127. The default is 34. **Notes:** - When not configured (i.e., default), the SITQ850Cause parameter is used.
SIT Q850 Cause For IC [SITQ850CauseForIC]	Defines the Q.850 cause value specified in the SIP Reason header that is included in a 4xx response when SIT-IC (Operator Intercept Special Information Tone) is detected from the Tel for IP-to-Tel calls. The valid range is 0 to 127. The default is -1 (not configured). **Notes:** - When not configured (i.e., default), the SITQ850Cause parameter is used.
SIT Q850 Cause For VC [SITQ850CauseForVC]	Defines the Q.850 cause value specified in the SIP Reason header that is included in a 4xx response when SIT-VC (Vacant Circuit - non-registered number Special Information Tone) is detected from the Tel for IP-to-Tel calls. The valid range is 0 to 127. The default is -1 (not configured). **Notes:**

Parameter	Description
	- When not configured (i.e., default), the SITQ850Cause parameter is used.
SIT Q850 Cause For RO [SITQ850CauseForRO]	Defines the Q.850 cause value specified in the SIP Reason header that is included in a 4xx response when SIT-RO (Reorder - System Busy Special Information Tone) is detected from the Tel for IP-to-Tel calls. The valid range is 0 to 127. The default is -1 (not configured). **Notes:** - When not configured (i.e., default), the SITQ850Cause parameter is used.
[GWInboundManipulationSet]	Selects the Manipulation Set ID for manipulating all inbound INVITE messages. The Manipulation Set is defined using the MessageManipulations parameter. By default, no manipulation is done (i.e. Manipulation Set ID is set to -1). **Note:** The parameter is applicable only to the Gateway application.
[GWOutboundManipulationSet]	Selects the Manipulation Set ID for manipulating all outbound INVITE messages. The Manipulation Set is defined using the MessageManipulations parameter. By default, no manipulation is done (i.e. Manipulation Set ID is set to -1). **Notes:** - The parameter is used only if the Outbound Message Manipulation Set parameter of the destination IP Group is not set. - The parameter is applicable only to the Gateway application.
Out-of-Service (Busy Out) Parameters	
Enable Busy Out `busy-out` [EnableBusyOut]	Enables the Busy Out feature. - [0] Disable (Default) - [1] Enable When Busy Out is enabled and certain scenarios exist, the device does the following: - All BRI trunks are automatically taken out-of-service by taking down the D-Channel. The above behavior is done upon one of the following scenarios: - The device is physically disconnected from the network (i.e., Ethernet cable is disconnected). - The device can't communicate with the Proxy Sets (according to the Proxy Keep-Alive mechanism) associated with the destination IP Groups for matching routing rules in the Tel-to-IP Routing table, and no other alternative route exists to send the call. - The IP Connectivity mechanism is enabled (see the AltRoutingTel2IPEnable parameter) and there is no connectivity to any destination IP address configured for matching routing rules in the Tel-to-IP Routing table. **Note:** - If the AltRoutingTel2IPEnable parameter is enabled, the Busy Out feature does not function with the Proxy Set keep-alive mechanism. To use the Busy Out feature with the Proxy Set keep-alive mechanism (for IP Groups), disable the AltRoutingTel2IPEnable parameter. - The Busy Out behavior depends on the PSTN protocol type.

Parameter	Description
	▪ The Busy Out condition is also applied per Trunk Group. This occurs if there is no connectivity to the Serving IP Group of a specific Trunk Group (configured in the Trunk Group Settings table). In such a scenario, all the physical trunks of the Trunk Group are set to the Busy Out condition. Each trunk uses the out-of-service method according to the ISDN variant. ▪ For configuring the method for taking trunks/channels out-of-service, see the DigitalOOSBehaviorForTrunk_x parameter for per trunk or the DigitalOOSBehavior parameter for all trunks.
Graceful Busy Out Timeout `graceful-bsy-out-t-out` [GracefulBusyOutTimeout]	Defines the timeout interval (in seconds) for Out-of-Service graceful shutdown mode for busy trunks (per trunk) if communication fails with a Proxy server (or Proxy Set). In such a scenario, the device rejects new calls from the PSTN (Serving Trunk Group), but maintains currently active calls for this user-defined timeout. Once this timeout elapses, the device terminates currently active calls and takes the trunk out of service (sending the PSTN busy-out signal). Trunks on which no calls are active are immediately taken out of service regardless of the timeout. The parameter is applicable to the locking of Trunk Groups feature and the Busy Out feature (see the EnableBusyOut parameter), where trunks/channels are taken out-of-service. The range is 0 to 3,600. The default is 0. **Note:** For configuring the method for taking trunks/channels out-of-service, see the DigitalOOSBehaviorForTrunk_x parameter for per trunk or the DigitalOOSBehavior parameter for all trunks.
Digital Out-Of-Service Behavior `dig-oos-behavior` [DigitalOOSBehaviorForTrunk_x]	Defines the method for setting digital trunks to out-of-service state. The parameter is defined per trunk. The parameter is applicable to the Busy Out feature (see the EnableBusyOut parameter) and the Lock/Unlock per Trunk Group feature performed in the Trunk Group Settings table of the Web interface. ▪ **[-1]** Not Configured = (Default) Use the settings of the DigitalOOSBehavior parameter ("global" parameter that applies to all trunks). ▪ **[0]** Default = ✓ ISDN: Sends ISDN Service messages to indicate out-of-service or in-service state for ISDN variants that support Service messages. For ISDN variants that do not support Service messages, the device sends an Alarm Indication Signal (AIS) alarm. ▪ **[1]** Service = Sends ISDN Service messages indicating out-of-service or in-service state. ✓ Graceful out-of-service disabled: The device rejects new incoming calls and immediately takes all channels (idle and busy) out-of-service, by sending Service messages on the B-channels. ✓ Graceful out-of-service enabled: - Fully configured trunk (all channels): The device rejects new incoming calls. If at least one busy channel exists during the graceful period, the device immediately takes all idle channels out-of-service, but sends out-of-service Service messages to the B-channels only when all channels are idle. - Partially configured trunk (only some channels configured): The device rejects new incoming calls and places all channels

Parameter	Description
	out-of-service only after the graceful period expires, by sending out-of-service Service messages to the B-channels. When connectivity is restored for the Busy Out feature or the Trunk Group is unlocked, the device brings all the trunks back into service by sending in-service Service messages to all their B-channels. ▪ **[2]** D-Channel = (Applicable only to ISDN and fully configured trunks) Takes the D-channel down or brings it up. ✓ Graceful out-of-service disabled: The device rejects new incoming calls and immediately takes the D-channel down. ✓ Graceful out-of-service enabled: The device rejects new incoming calls. Only when all channels are idle (when graceful period ends or when all channels become idle before graceful period ends, whichever occurs first), does the device take the D-channel down. When connectivity is restored for the Busy Out feature or the Trunk Group is unlocked, the device brings the D-channels up again. **Note:** For partially configured trunks (only some channels configured), this option only rejects new calls for the trunk; the D-channel remains up. ▪ **[3]** Alarm = Sends or clears a PSTN Alarm Indication Signal (AIS) alarm. ✓ Graceful out-of-service enabled: The device rejects new incoming calls and immediately sends an AIS alarm. ✓ Graceful out-of-service enabled: The device rejects new incoming calls and only when all channels are idle (when graceful period ends or when all channels become idle before graceful period ends, whichever occurs first), does the device send an alarm on the trunk. When connectivity is restored for the Busy Out feature or the Trunk Group is unlocked, the device clears the alarm. **Note:** For partially configured trunks (only some channels configured), this option only rejects new calls for the trunk; no alarm is sent. **Notes:** ▪ When configuring out-of-service behavior per trunk (DigitalOOSBehaviorForTrunk_x), you must stop the trunk (**Stop Trunk** button in the Trunk Settings page), configure the parameter, and then restart the trunk (**Apply Trunk Settings** button in the Trunk Settings page) for the settings to take effect. ▪ To define out-of-service behavior for all trunks (globally), see the DigitalOOSBehavior parameter. ▪ For locking/unlocking Trunk Groups in the Trunk Group Settings table, see Configuring Trunk Group Settings on page 435. ▪ For a description of the Busy Out feature and for enabling the feature, see the EnableBusyOut parameter. ▪ To configure the graceful out-of-service period, see the GracefulBusyOutTimeout parameter. ▪ If the ISDN variant does not support the configured out-of-service option of the parameter, the device sets the parameter to Default [0].

Parameter	Description
	▪ The x in the ini file parameter name denotes the trunk number, where 0 is Trunk 1.
Digital Out-Of-Service Behavior `dig-oos-behavior` [DigitalOOSBehavior]	Defines the method for setting all digital trunks to out-of-service state. To configure the out-of-service method per trunk, see the DigitalOOSBehaviorForTrunk_x parameter. ▪ **[0]** Default = (Default) For a detailed description, see option [0] of the DigitalOOSBehaviorForTrunk_x parameter (per trunk setting). ▪ **[1]** Service = Sends an ISDN Service message indicating out-of-service state (or in-service). For a detailed description, see option [1] of the DigitalOOSBehaviorForTrunk_x parameter (per trunk setting). ▪ **[2]** D-Channel = Takes the D-Channel down or brings it up. For a detailed description, see option [2] of the DigitalOOSBehaviorForTrunk_x parameter (per trunk setting). ▪ **[3]** Alarm = Sends or clears a PSTN Alarm Indication Signal (AIS) alarm. For a detailed description, see option [3] of the DigitalOOSBehaviorForTrunk_x parameter (per trunk setting). **Notes:** ▪ When using the parameter to configure out-of-service behavior for all trunks, you must reset the device for the settings to take effect. ▪ If the ISDN variant does not support the configured out-of-service option of the parameter, the device sets the parameter to Default [0].
Retransmission Parameters	
SIP T1 Retransmission Timer `t1-re-tx-time` [SipT1Rtx]	Defines the time interval (in msec) between the first transmission of a SIP message and the first retransmission of the same message. The default is 500. **Note:** The time interval between subsequent retransmissions of the same SIP message starts with SipT1Rtx. For INVITE requests, it is multiplied by two for each new retransmitted message. For all other SIP messages, it is multiplied by two until SipT2Rtx. For example, assuming SipT1Rtx = 500 and SipT2Rtx = 4000: ▪ The first retransmission is sent after 500 msec. ▪ The second retransmission is sent after 1000 (2*500) msec. ▪ The third retransmission is sent after 2000 (2*1000) msec. ▪ The fourth retransmission and subsequent retransmissions until SIPMaxRtx are sent after 4000 (2*2000) msec.
SIP T2 Retransmission Timer `t2-re-tx-time` [SipT2Rtx]	Defines the maximum interval (in msec) between retransmissions of SIP messages (except for INVITE requests). The default is 4000. **Note:** The time interval between subsequent retransmissions of the same SIP message starts with SipT1Rtx and is multiplied by two until SipT2Rtx.
SIP Maximum RTX `sip-max-rtx` [SIPMaxRtx]	Defines the maximum number of UDP transmissions of SIP messages (first transmission plus retransmissions). The range is 1 to 30. The default is 7.

Parameter	Description
Number of RTX Before Hot-Swap `nb-of-rtx-b4-hot-swap` [HotSwapRtx]	Defines the number of retransmitted INVITE/REGISTER messages before the call is routed (hot swap) to another Proxy/Registrar. The valid range is 1 to 30. The default is 3. For example, if configured to 3 and no response is received from an IP destination, the device attempts another three times to send the call to the IP destination. If still unsuccessful, it attempts to redirect the call to another IP destination. **Note:** The parameter is also used for alternative routing (see "Alternative Routing Based on IP Connectivity" on page 486.
SIP Message Manipulations Table	
Message Manipulations `configure voip > sbc manipulations message-manipulations` [MessageManipulations]	Defines manipulation rules for SIP header messages. The format of the ini file table parameter is as follows: [MessageManipulations] FORMAT MessageManipulations_Index = MessageManipulations_ManSetID, MessageManipulations_MessageType, MessageManipulations_Condition, MessageManipulations_ActionSubject, MessageManipulations_ActionType, MessageManipulations_ActionValue, MessageManipulations_RowRole; [\MessageManipulations] For example, the below configuration changes the user part of the SIP From header to 200: MessageManipulations 1 = 0, Invite.Request, , Header.From.Url.User, 2, 200, 0; For a detailed description of the table, see Configuring SIP Message Manipulation on page 370.
Message Policy Table	
Message Policy Table `configure voip > sbc message-policy` [MessagePolicy]	Defines SIP message policy rules for blocking (blacklist) unwanted incoming SIP messages or allowing (whitelist) receipt of desired messages. The format of the ini file table parameter is as follows: [MessagePolicy] FORMAT MessagePolicy_Index = MessagePolicy_Name, MessagePolicy_MaxMessageLength, MessagePolicy_MaxHeaderLength, MessagePolicy_MaxBodyLength, MessagePolicy_MaxNumHeaders, MessagePolicy_MaxNumBodies, MessagePolicy_SendRejection, MessagePolicy_MethodList, MessagePolicy_MethodListType, MessagePolicy_BodyList, MessagePolicy_BodyListType; [/MessagePolicy] For a detailed description of the table, see Configuring SIP Message Policy Rules.

57.8 Coders and Profile Parameters

The profile parameters are described in the table below.

Table 57-32: Profile Parameters

Parameter	Description
IP Profile Settings Table	
IP Profile Settings `configure voip > coders-and-profiles ip-profile` [IPProfile]	Defines the IP Profile table. The format of the ini file table parameter is as follows: [IPProfile] FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference, IpProfile_CodersGroupID, IpProfile_IsFaxUsed, IpProfile_JitterBufMinDelay, IpProfile_JitterBufOptFactor, IpProfile_IPDiffServ, IpProfile_SigIPDiffServ, IpProfile_SCE, IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort, IpProfile_CNGmode, IpProfile_VxxTransportType, IpProfile_NSEMode, IpProfile_IsDTMFUsed, IpProfile_PlayRBTone2IP, IpProfile_EnableEarlyMedia, IpProfile_ProgressIndicator2IP, IpProfile_EnableEchoCanceller, IpProfile_CopyDest2RedirectNumber, IpProfile_MediaSecurityBehaviour, IpProfile_CallLimit, IpProfile_DisconnectOnBrokenConnection, IpProfile_FirstTxDtmfOption, IpProfile_SecondTxDtmfOption, IpProfile_RxDTMFOption, IpProfile_EnableHold, IpProfile_InputGain, IpProfile_VoiceVolume, IpProfile_AddIEInSetup, IpProfile_SBCExtensionCodersGroupID, IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode, IpProfile_SBCAllowedMediaTypes, IpProfile_SBCAllowedCodersGroupID, IpProfile_SBCAllowedVideoCodersGroupID, IpProfile_SBCAllowedCodersMode, IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior, IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity, IpProfile_AMDSensitivityParameterSuit, IpProfile_AMDSensitivityLevel, IpProfile_AMDMaxGreetingTime, IpProfile_AMDMaxPostSilenceGreetingTime, IpProfile_SBCDiversionMode, IpProfile_SBCHistoryInfoMode, IpProfile_EnableQSIGTunneling, IpProfile_SBCFaxCodersGroupID, IpProfile_SBCFaxBehavior, IpProfile_SBCFaxOfferMode, IpProfile_SBCFaxAnswerMode, IpProfile_SbcPrackMode, IpProfile_SBCSessionExpiresMode, IpProfile_SBCRemoteUpdateSupport, IpProfile_SBCRemoteReinviteSupport, IpProfile_SBCRemoteDelayedOfferSupport, IpProfile_SBCRemoteReferBehavior, IpProfile_SBCRemote3xxBehavior, IpProfile_SBCRemoteMultiple18xSupport, IpProfile_SBCRemoteEarlyMediaResponseType, IpProfile_SBCRemoteEarlyMediaSupport, IpProfile_EnableSymmetricMKI, IpProfile_MKISize, IpProfile_SBCEnforceMKISize, IpProfile_SBCRemoteEarlyMediaRTP, IpProfile_SBCRemoteSupportsRFC3960, IpProfile_SBCRemoteCanPlayRingback, IpProfile_EnableEarly183, IpProfile_EarlyAnswerTimeout, IpProfile_SBC2833DTMFPayloadType, IpProfile_SBCUserRegistrationTime, IpProfile_ResetSRTPStateUponRekey, IpProfile_AmdMode, IpProfile_SBCReliableHeldToneSource, IpProfile_GenerateSRTPKeys, IpProfile_SBCPlayHeldTone, IpProfile_SBCRemoteHoldFormat, IpProfile_SBCRemoteReplacesBehavior, IpProfile_SBCSDPPtimeAnswer, IpProfile_SBCPreferredPTime, IpProfile_SBCUseSilenceSupp,

Parameter	Description
	IpProfile_SBCRTPRedundancyBehavior, IpProfile_SBCPlayRBTToTransferee, IpProfile_SBCRTCPMode, IpProfile_SBCJitterCompensation, IpProfile_SBCRemoteRenegotiateOnFaxDetection, IpProfile_JitterBufMaxDelay, IpProfile_SBCUserBehindUdpNATRegistrationTime, IpProfile_SBCUserBehindTcpNATRegistrationTime, IpProfile_SBCSDPHandleRTCPAttribute, IpProfile_SBCRemoveCryptoLifetimeInSDP, IpProfile_SBCIceMode, IpProfile_SBCRTCPMux, IpProfile_SBCMediaSecurityMethod, IpProfile_SBCHandleXDetect, IpProfile_SBCRTCPFeedback, IpProfile_SBCRemoteRepresentationMode, IpProfile_SBCKeepVIAHeaders, IpProfile_SBCKeepRoutingHeaders, IpProfile_SBCKeepUserAgentHeader, IpProfile_SBCRemoteMultipleEarlyDialogs, IpProfile_SBCRemoteMultipleAnswersMode, IpProfile_SBCDirectMediaTag; [\IPProfile] For a description of the table, see "Configuring IP Profiles" on page 387.
Tel Profile Table	
Tel Profile Settings `configure voip > coders-and-profiles tel-profile` [TelProfile]	Defines the Tel Profile table. Each Tel Profile ID includes a set of parameters (which are typically configured separately using their individual, "global" parameters). You can later assign these Tel Profile IDs to other elements such as in the Trunk Group table (TrunkGroup parameter). Therefore, Tel Profiles allow you to apply the same settings of a group of parameters to multiple channels, or apply specific settings to different channels. The format of the ini file table parameter is as follows: [TelProfile] FORMAT TelProfile_Index = TelProfile_ProfileName, TelProfile_TelPreference, TelProfile_CodersGroupID, TelProfile_IsFaxUsed, TelProfile_JitterBufMinDelay, TelProfile_JitterBufOptFactor, TelProfile_IPDiffServ, TelProfile_SigIPDiffServ, TelProfile_DtmfVolume, TelProfile_InputGain, TelProfile_VoiceVolume, TelProfile_EnableReversePolarity, TelProfile_EnableCurrentDisconnect, TelProfile_EnableDigitDelivery, TelProfile_EnableEC, TelProfile_MWIAnalog, TelProfile_MWIDisplay, TelProfile_FlashHookPeriod, TelProfile_EnableEarlyMedia, TelProfile_ProgressIndicator2IP, TelProfile_TimeForReorderTone, TelProfile_EnableDIDWink, TelProfile_IsTwoStageDial, TelProfile_DisconnectOnBusyTone, TelProfile_EnableVoiceMailDelay, TelProfile_DialPlanIndex, TelProfile_Enable911PSAP, TelProfile_SwapTelToIpPhoneNumbers, TelProfile_EnableAGC, TelProfile_ECNlpMode, TelProfile_DigitalCutThrough, TelProfile_EnableFXODoubleAnswer, TelProfile_CallPriorityMode; [\TelProfile] For a description of the parameter, see Configuring Tel Profiles on page 384.

57.9 Channel Parameters

This subsection describes the device's channel parameters.

57.9.1 Voice Parameters

The voice parameters are described in the table below.

Table 57-33: Voice Parameters

Parameter	Description
Input Gain `input-gain` [InputGain]	Global parameter that defines the pulse-code modulation (PCM) input (received) gain control level (in decibels). You can also configure this functionality per specific calls, using IP Profiles (IpProfile_InputGain) or Tel Profiles. For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387 or in the Tel Profile table, see Configuring Tel Profiles on page 384. **Note:** If this functionality is configured for a specific profile, the settings of this global parameter is ignored for calls associated with the profile.
Voice Volume `voice-volume` [VoiceVolume]	Global parameter that defines the voice gain control (in decibels). This defines the level of the transmitted (IP-to-Tel) signal. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_VoiceVolume) or Tel Profiles. For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387 or in the Tel Profile table, see Configuring Tel Profiles on page 384. **Note:** If this functionality is configured for a specific profile, the settings of this global parameter is ignored for calls associated with the profile.
`G726-voice-payload-format` [VoicePayloadFormat]	Determines the bit ordering of the G.726 voice payload format. ▪ **[0]** = (Default) Little Endian ▪ **[1]** = Big Endian **Note:** To ensure high voice quality when using G.726, both communicating ends should use the same endianness format. Therefore, when the device communicates with a third-party entity that uses the G.726 voice coder and voice quality is poor, change the settings of the parameter (between Big Endian and Little Endian).
MF Transport Type `MF-transport-type` [MFTransportType]	Currently, not supported.
Echo Canceler `echo-canceller-enable` [EnableEchoCanceller]	Global parameter that enables echo cancellation (i.e., echo from voice calls is removed). You can also configure this functionality per specific calls, using IP Profiles (IpProfile_EnableEchoCanceller) or Tel Profiles. For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387 or Tel Profile table, see Configuring Tel Profiles on page 384. **Note:** If this functionality is configured for a specific profile, the settings of this global parameter is ignored for calls associated with the profile.
`echo-canceller-hybrid-loss` [ECHybridLoss]	Defines the four-wire to two-wire worst-case Hybrid loss, the ratio between the signal level sent to the hybrid and the echo level returning from the hybrid.

Parameter	Description
	▪ **[0]** = (Default) 6 dB ▪ **[1]** = N/A ▪ **[2]** = 0 dB ▪ **[3]** = 3 dB
`echo-canceller-NLP-mode` [ECNLPMode]	Enables Non-Linear Processing (NLP) mode for echo cancellation. ▪ **[0]** = (Default) NLP adapts according to echo changes ▪ **[1]** = Disables NLP **Note:** The parameter can also be configured in a Tel Profile.
`echo-canceller-aggressive-NLP` [EchoCancellerAggressiveNLP]	Enables the Aggressive NLP at the first 0.5 second of the call. ▪ **[0]** = Disable ▪ **[1]** = (Default) Enable. The echo is removed only in the first half of a second of the incoming IP signal. **Note:** For the parameter to take effect, a device reset is required.
`number-of-SID-coefficients` [RTPSIDCoeffNum]	Defines the number of spectral coefficients added to an SID packet being sent according to RFC 3389. The valid values are **[0]** (default), **[4]**, **[6]**, **[8]** and **[10]**.
Answer Detector (AD) Parameters	
Enable Answer Detector [EnableAnswerDetector]	Currently, not supported.
Answer Detector Activity Delay `answer-detector-activativity-delay` [AnswerDetectorActivityDelay]	Defines the time (in 100-msec resolution) between activating the Answer Detector and the time that the detector actually starts to operate. The valid range is 0 to 1023. The default is 0.
Answer Detector Silence Time [AnswerDetectorSilenceTime]	Currently, not supported.
Answer Detector Redirection [AnswerDetectorRedirection]	Currently, not supported.
Answer Detector Sensitivity `answer-detector-sensitivity` [AnswerDetectorSensitivity]	Defines the Answer Detector sensitivity. The range is 0 (most sensitive) to 2 (least sensitive). The default is 0.

57.9.2 Coder Parameters

The coder parameters are described in the table below.

Table 57-34: Coder Parameters

Parameter	Description
Silk Tx Inband FEC `silk-tx-inband-fec` [SilkTxInbandFEC]	Enables forward error correction (FEC) for the SILK coder. ▪ [0] Disable (default) ▪ [1] Enable
Silk Max Average Bit Rate	Defines the maximum average bit rate for the SILK coder. The valid value range is 5000 to 30000. The default is 16000.

Parameter	Description
silk-max-average-bitrate [SilkMaxAverageBitRate]	The SILK coder is Skype's default audio codec used for Skype-to-Skype calls.
vbr-coder-header-format [VBRCoderHeaderFormat]	Determines the format of the RTP header for VBR coders. ▪ [0] = (Default) Payload only (no header, TOC, or m-factor) - similar to RFC 3558 Header Free format. ▪ [1] = Supports RFC 2658 - 1 byte for interleaving header (always 0), TOC, no m-factor. ▪ [2] = Payload including TOC only, allow m-factor. ▪ [3] = RFC 3558 Interleave/Bundled format.
vbr-coder-hangover [VBRCoderHangover]	Defines the required number of silence frames at the beginning of each silence period when using the VBR coder silence suppression. The range is 0 to 255. The default is 1.
AMR Payload Format [AmrOctetAlignedEnable]	Defines the AMR payload format type. ▪ [0] Bandwidth Efficient ▪ [1] Octet Aligned (default) **Note:** The AMR payload type can also be configured per Coder Group (see Configuring Coder Groups on page 382). The Coder Group configuration overrides the parameter.
[AMRCoderHeaderFormat]	Determines the payload format of the AMR header. ▪ [0] = Non-standard multiple frames packing in a single RTP frame. Each frame has a CMR and TOC header. ▪ [1] = AMR frame according to RFC 3267 bundling. ▪ [2] = AMR frame according to RFC 3267 interleaving. ▪ [3] = AMR is passed using the AMR IF2 format. **Note:** Bandwidth Efficient mode is not supported; the mode is always Octet-aligned.

57.9.3 DTMF Parameters

The dual-tone multi-frequency (DTMF) parameters are described in the table below.

Table 57-35: DTMF Parameters

Parameter	Description
DTMF Transport Type DTMF-transport-type [DTMFTransportType]	Determines the DTMF transport type. ▪ **[0]** Mute DTMF = DTMF digits are removed from the voice stream and are not relayed to remote side. ▪ **[2]** Transparent DTMF = DTMF digits remain in the voice stream. ▪ **[3]** RFC 2833 Relay DTMF = (Default) DTMF digits are removed from the voice stream and are relayed to remote side according to RFC 2833. ▪ **[7]** RFC 2833 Relay Decoder Mute = DTMF digits are sent according to RFC 2833 and muted when received. **Note:** The parameter is automatically updated if the parameters FirstTxDTMFOption or RxDTMFOption are configured.

Parameter	Description
DTMF Volume (-31 to 0 dB) `DTMF-volume` [DTMFVolume]	Defines the DTMF gain control value (in decibels) to the Tel side. The valid range is -31 to 0 dB. The default is -11 dB. **Note:** The parameter can also be configured in a Tel Profile.
DTMF Generation Twist `DTMF-generation-twist` [DTMFGenerationTwist]	Defines the range (in decibels) between the high and low frequency components in the DTMF signal. Positive decibel values cause the higher frequency component to be stronger than the lower one. Negative values cause the opposite effect. For any parameter value, both components change so that their average is constant. The valid range is -10 to 10 dB. The default is 0 dB. **Note:** For the parameter to take effect, a device reset is required.
`inter-digit-interval` [DTMFInterDigitInterval]	Defines the time (in msec) between generated DTMF digits to the Tel side (if FirstTxDTMFOption = 1, 2 or 3). The valid range is 0 to 32767. The default is 100.
[DTMFDigitLength]	Defines the time (in msec) for generating DTMF tones to the Tel side (if FirstTxDTMFOption = 1, 2 or 3). It also configures the duration that is sent in INFO (Cisco) messages. The valid range is 0 to 32767. The default is 100.
`default-dtmf-signal-duration` [RxDTMFHangOverTime]	Defines the Voice Silence time (in msec) after playing DTMF or MF digits to the Tel side that arrive as Relay from the IP side. Valid range is 0 to 2,000 msec. The default is 1,000 msec.
`digit-hangover-time-tx` [TxDTMFHangOverTime]	Defines the Voice Silence time (in msec) after detecting the end of DTMF or MF digits at the Tel side when the DTMF Transport Type is either Relay or Mute. Valid range is 0 to 2,000 msec. The default is 1,000 msec.
NTE Max Duration `telephony-events-max-duration` [NTEMaxDuration]	Defines the maximum time for sending Named Telephony Events / NTEs (RFC 4733/2833 DTMF relay) to the IP side, regardless of the DTMF signal duration on the TDM side. The range is -1 to 200,000,000 msec. The default is -1 (i.e., NTE stops only upon detection of an End event).

57.9.4 RTP, RTCP and T.38 Parameters

The RTP, RTCP and T.38 parameters are described in the table below.

Table 57-36: RTP/RTCP and T.38 Parameters

Parameter	Description
Dynamic Jitter Buffer Minimum Delay `jitter-buffer-minimum-delay` [DJBufMinDelay]	Global parameter that defines the minimum delay (in msec) of the device's dynamic Jitter Buffer. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_JitterBufMinDelay) or Tel Profiles. For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see Configuring IP Profiles on page 387, or in the Tel Profile table, see Configuring Tel Profiles on page 384.

Parameter	Description
	Note: If this functionality is configured for a specific profile, the settings of this global parameter is ignored for calls associated with the profile.
Dynamic Jitter Buffer Optimization Factor jitter-buffer-optimization-factor [DJBufOptFactor]	Global parameter that defines the Dynamic Jitter Buffer frame error/delay optimization factor. You can also configure this functionality per specific calls, using IP Profiles or Tel Profiles. For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see Configuring IP Profiles on page 387, or in the Tel Profile table, see Configuring Tel Profiles on page 384. **Note:** If this functionality is configured for a specific profile, the settings of this global parameter is ignored for calls associated with the profile.
RTP Redundancy Depth RTP-redundancy-depth [RTPRedundancyDepth]	Global parameter that enables the device to generate RFC 2198 redundant packets. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_RTPRedundancyDepth). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
Enable RTP Redundancy Negotiation rtp-rdcy-nego-enbl [EnableRTPRedundancyNegotiation]	Enables the device to include the RTP redundancy dynamic payload type in the SDP (according to RFC 2198). ▪ [0] Disable (default) ▪ [1] Enable = The device includes in the SDP message the RTP payload type "RED" and the payload type configured by the parameter RFC2198PayloadType. a=rtpmap:<PT> RED/8000 Where <PT> is the payload type as defined by RFC2198PayloadType. The device sends the INVITE message with "a=rtpmap:<PT> RED/8000" and responds with a 18x/200 OK and "a=rtpmap:<PT> RED/8000" in the SDP. **Notes:** ▪ The parameter is applicable only to the Gateway application. ▪ For this feature to be functional, you must also set the parameter RTPRedundancyDepth to 1 (i.e., enabled). ▪ Currently, the negotiation of “RED” payload type is not supported and therefore, it should be configured to the same PT value for both parties.
RFC 2198 Payload Type RTP-redundancy-payload-type [RFC2198PayloadType]	Defines the RTP redundancy packet payload type (according to RFC 2198). The valid value is 96 to 127. The default is 104. **Note:** The parameter is applicable only if the RTPRedundancyDepth parameter is set to 1.
Packing Factor [RTPPackingFactor]	N/A. Controlled internally by the device according to the selected coder.

Parameter	Description
RFC 2833 TX Payload Type `telephony-events-payload-type-tx` [RFC2833TxPayloadType]	Defines the Tx RFC 2833 DTMF relay dynamic payload type for outbound calls. The valid range is 96 to 127. The default is 96. **Note:** When RFC 2833 payload type negotiation is used (i.e., the parameter FirstTxDTMFOption is set to 4), this payload type is used for the received DTMF packets. If negotiation isn't used, this payload type is used for receive and for transmit.
RFC 2833 RX Payload Type `telephony-events-payload-type-rx` [RFC2833RxPayloadType]	Defines the Rx RFC 2833 DTMF relay dynamic payload type for inbound calls. The valid range is 96 to 127. The default is 96. **Note:** When RFC 2833 payload type negotiation is used (i.e., the parameter FirstTxDTMFOption is set to 4), this payload type is used for the received DTMF packets. If negotiation isn't used, this payload type is used for receive and for transmit.
[EnableDetectRemoteMACChange]	Determines whether the device changes the RTP packets according to the MAC address of received RTP packets and according to Gratuitous Address Resolution Protocol (GARP) messages. ▪ [0] = Nothing is changed. ▪ [1] = If the device receives RTP packets with a different source MAC address (than the MAC address of the transmitted RTP packets), then it sends RTP packets to this MAC address and removes this IP entry from the device's ARP cache table. ▪ [2] = (Default) The device uses the received GARP packets to change the MAC address of the transmitted RTP packets. ▪ [3] = Options 1 and 2 are used. **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ If the device is located in a network subnet which is connected to other gateways using a router that uses Virtual Router Redundancy Protocol (VRRP) for redundancy, then set the parameter to 0 or 2.
RTP Base UDP Port [BaseUDPport]	Global parameter that defines the lower boundary of the UDP port used for RTP, RTCP (RTP port + 1) and T.38 (RTP port + 2). For more information on configuring the UDP port range, see "Configuring RTP Base UDP Port" on page 191. The range of possible UDP ports is 6,000 to 65,535. The default base UDP port is 6000. **Note:** For the parameter to take effect, a device reset is required.
`no-operation-enable` [NoOpEnable]	Enables the transmission of RTP or T.38 No-Op packets. ▪ **[0]** = Disable (default) ▪ **[1]** = Enable This mechanism ensures that the NAT binding remains open during RTP or T.38 silence periods.

Parameter	Description
[NoOpInterval]	Defines the time interval in which RTP or T.38 No-Op packets are sent in the case of silence (no RTP/T.38 traffic) when No-Op packet transmission is enabled. The valid range is 20 to 65,000 msec. The default is 10,000. **Note:** To enable No-Op packet transmission, use the NoOpEnable parameter.
`no-operation-interval` [RTPNoOpPayloadType]	Defines the payload type of No-Op packets. The valid range is 96 to 127 (for the range of Dynamic RTP Payload Type for all types of non hard-coded RTP Payload types, refer to RFC 3551). The default is 120. **Note:** When defining the parameter, ensure that it doesn't cause collision with other payload types.
`rtcp-act-mode` [RTCPActivationMode]	Disables RTCP traffic when there is no RTP traffic. This feature is useful, for example, to stop RTCP traffic that is typically sent when calls are put on hold (by an INVITE with 'a=inactive' in the SDP). ▪ [0] Active Always = (Default) RTCP is active even during inactive RTP periods, i.e., when the media is in 'recvonly' or 'inactive' mode. ▪ [1] Inactive Only If RTP Inactive = No RTCP is sent when RTP is inactive. **Note:** The parameter is applicable only to Gateway calls (not SBC).
RTP Control Protocol Extended Reports (RTCP XR) Parameters	
Enable RTCP XR `voice-quality-monitoring-enable` [VQMonEnable]	Enables voice quality monitoring and RTCP XR, according to RFC 3611. ▪ **[0]** Disable (default) ▪ **[1]** Enable Fully = Calculates voice quality metrics, uses them for QoE calculations, reports them to SEM (if configured), and sends them to remote side using RTCP XR. ▪ **[2]** Enable Calculation Only = Calculates voice quality metrics, uses them for QoE calculations, reports them to SEM (if configured), but does not send them to remote side using RTCP XR. **Note:** For the parameter to take effect, a device reset is required.
Minimum Gap Size [VQMonGMin]	Defines the voice quality monitoring - minimum gap size (number of frames). The default is 16.
Burst Threshold [VQMonBurstHR]	Defines the voice quality monitoring - excessive burst alert threshold. The default is -1 (i.e., no alerts are issued).
Delay Threshold [VQMonDelayTHR]	Defines the voice quality monitoring - excessive delay alert threshold. The default is -1 (i.e., no alerts are issued).
R-Value Delay Threshold [VQMonEOCRValTHR]	Defines the voice quality monitoring - end of call low quality alert threshold.

Parameter	Description
	The default is -1 (i.e., no alerts are issued).
RTCP XR Packet Interval `rtcp-interval` [RTCPInterval]	Defines the time interval (in msec) between adjacent RTCP XR reports. This interval starts from call establishment. Thus, the device can send RTCP XR reports during the call, in addition to at the end of the call. If the duration of the call is shorter than this interval, RTCP XR is sent only at the end of the call. The valid value range is 0 to 65,535. The default is 5,000.
Disable RTCP XR Interval Randomization `disable-RTCP-randomization` [DisableRTCPRandomize]	Determines whether RTCP report intervals are randomized or whether each report interval accords exactly to the parameter RTCPInterval. ▪ **[0]** Disable = (Default) Randomize ▪ **[1]** Enable = No Randomize
Gateway RTCP XR Report Mode `rtcp-xr-rep-mode` [RTCPXRReportMode]	Enables the device to send RTCP XR in SIP PUBLISH messages to the Event State Compositor (ESC) server and defines the interval at which they are sent. ▪ [0] Disable = (Default) RTCP XR is not sent. ▪ [1] End Call = RTCP XR is sent at the end of the call. ▪ [2] End Call & Periodic = RTCP XR is sent at the end of the call and periodically according to the RTCPInterval parameter. ▪ [3] End Call & End Segment = RTCP XR is sent at the end of the call and at the end of each media segment of the call. A media segment is a change in media, for example, when the coder is changed or when the caller toggles between two called parties (using call hold/retrieve). The RTCP XR sent at the end of a media segment contains information only of that segment. If the segment does not contain RTP/RTCP content, the RTCP XR is not sent. For call hold, the device sends an RTCP XR each time the call is placed on hold and each time it is retrieved. In addition, the Start timestamp in the RTCP XR indicates the start of the media segment; the End timestamp indicates the time of the last sent periodic RTCP XR (typically, up to 5 seconds before reported segment ends). **Note:** The parameter is applicable only to the Gateway application.
SBC RTCP XR Report Mode `sbc-rtcpxr-report-mode` [SBCRtcpXrReportMode]	Enables the sending of RTCP XR reports of QoE metrics at the end of each call session (i.e., after a SIP BYE). The RTCP XR is sent in the SIP PUBLISH message. ▪ [0] Disable (default) ▪ [1] End of Call **Note:** The parameter is applicable only to the SBC application.
publication-ip-group-id [PublicationIPGroupID]	Defines the IP Group to where the RTCP XR is sent.

57.10 Gateway Application Parameters

57.10.1 Fax and Modem Parameters

The fax and modem parameters are described in the table below.

Table 57-37: Fax and Modem Parameters

Parameter	Description
Fax Transport Mode `fax-transport-mode` [FaxTransportMode]	Determines the fax transport mode used by the device. ▪ **[0]** Disable = transparent mode ▪ **[1]** T.38 Relay (default) ▪ **[2]** Bypass ▪ **[3]** Events Only **Note:** The parameter is overridden by the parameter IsFaxUsed. If the parameter IsFaxUsed is set to 1 (T.38 Relay) or 3 (Fax Fallback), then FaxTransportMode is always set to 1 (T.38 relay).
`V34-fax-transport-type` [V34FaxTransportType]	Determines the V.34 fax transport method (whether V34 fax falls back to T.30 or pass over Bypass). ▪ [0] = Transparent ▪ [1] = (Default) Relay ▪ [2] = Bypass ▪ [3] = Transparent with Events **Note:** To configure V34FaxTransportType to 1 (i.e., fax relay), you also need to configure FaxTransportMode to 1 (fax relay).
V.21 Modem Transport Type `V21-modem-transport-type` [V21ModemTransportType]	Determines the V.21 modem transport type. ▪ **[0]** Disable = (Default) Transparent. ▪ **[2]** Enable Bypass ▪ **[3]** Events Only = Transparent with Events. **Note:** You can also configure this functionality per specific calls, using IP Profiles (IpProfile_VxxTransportType). For more information, see "Configuring IP Profiles" on page 387.
V.22 Modem Transport Type `V22-modem-transport-type` [V22ModemTransportType]	Determines the V.22 modem transport type. ▪ **[0]** Disable = Transparent. ▪ **[2]** Enable Bypass (default) ▪ **[3]** Events Only = Transparent with Events. **Note:** You can also configure this functionality per specific calls, using IP Profiles (IpProfile_VxxTransportType). For more information, see "Configuring IP Profiles" on page 387.
V.23 Modem Transport Type `V23-modem-transport-type` [V23ModemTransportType]	Determines the V.23 modem transport type. ▪ **[0]** Disable = Transparent. ▪ **[2]** Enable Bypass (default)

Parameter	Description
	- **[3]** Events Only = Transparent with Events. **Note:** You can also configure this functionality per specific calls, using IP Profiles (IpProfile_VxxTransportType). For more information, see "Configuring IP Profiles" on page 387.
V.32 Modem Transport Type `V32-modem-transport-type` [V32ModemTransportType]	Determines the V.32 modem transport type. - **[0]** Disable = Transparent. - **[2]** Enable Bypass (default) - **[3]** Events Only = Transparent with Events. **Notes:** - The parameter applies only to V.32 and V.32bis modems. - You can also configure this functionality per specific calls, using IP Profiles (IpProfile_VxxTransportType). For more information, see "Configuring IP Profiles" on page 387.
V.34 Modem Transport Type `V34-modem-transport-type` [V34ModemTransportType]	Determines the V.90/V.34 modem transport type. - **[0]** Disable = Transparent. - **[2]** Enable Bypass (default) - **[3]** Events Only = Transparent with Events. **Note:** You can also configure this functionality per specific calls, using IP Profiles (IpProfile_VxxTransportType). For more information, see "Configuring IP Profiles" on page 387.
`bell-modem-transport-type` [BellModemTransportType]	Determines the Bell modem transport method. - **[0]** = Transparent (default) - **[2]** = Bypass - **[3]** = Transparent with events
Fax CNG Mode `fax_cng_mode` [FaxCNGMode]	Determines the device's handling of fax relay upon detection of a fax CNG tone or a V.34/Super G3 V8-CM (Call Menu) signal from originating faxes. - **[0]** Doesn't send T.38 Re-INVITE = (Default) SIP re-INVITE is not sent. - **[1]** Sends on CNG tone = Sends a SIP re-INVITE with T.38 parameters in SDP to the terminating fax upon detection of a fax CNG tone, if the CNGDetectorMode parameter is set to 1. - [2] Sends on CNG or v8-cn = Sends a SIP re-INVITE with T.38 parameters in SDP to the terminating fax upon detection of a fax CNG tone (if the CNGDetectorMode parameter is set to 1) or upon detection of a V8-CM signal. **Notes:**

Parameter	Description
	- If the parameter is set to [2] and the CNGDetectorMode parameter is set to [0], the device sends a re-INVITE only if it detects a V8-CM signal from the originating fax. - This feature is applicable only if the IsFaxUsed parameter is set to [1] or [3]. - The device also sends T.38 re-INVITE if the CNGDetectorMode parameter is set to [2], regardless of the FaxCNGMode parameter settings.
CNG Detector Mode `coder` [CNGDetectorMode]	Global parameter that enables the detection of the fax calling tone (CNG) and defines the detection method. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_CNGmode). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
Fax Detect Timeout Since Connect `configure voip > sip-definition general-settings > fax-detect-timeout-since-connect` [FaxDetectTimeoutSinceConnect]	Defines a timeout (in msec) for detecting fax from the Tel side during an established voice call. The interval starts from when the voice call is established. If the device detects a fax tone within the interval, it ends the voice session and sends a T.38 or VBD re-INVITE message to the IP side and processes the fax. If the interval expires without any received fax event, the device ignores all subsequent fax events during the voice session. The valid value is 0 to 120000. The default is 0. If set to 0, the device can detect fax during the entire voice call.
SIP T.38 Version `sip-t38-ver` [SIPT38Version]	Determines the T.38 fax relay version. - [-1] Not Configured = (Default) No T.38 - [0] Version 0 - [3] Version 3 = T.38 Version 3 (V.34 over T.38) **Note:** For a description on V.34 over T.38 fax relay, see V.34 Fax Support on page 182.
Fax Relay Enhanced Redundancy Depth `enhanced-redundancy-depth` [FaxRelayEnhancedRedundancyDepth]	Defines the number of times that control packets are retransmitted when using the T.38 standard. The valid range is 0 to 4. The default is .

Parameter	Description
Fax Relay Redundancy Depth `redundancy-depth` [FaxRelayRedundancyDepth]	Defines the number of times that each fax relay payload is retransmitted to the network. ▪ **[0]** = (Default) No redundancy ▪ **[1]** = One packet redundancy ▪ **[2]** = Two packet redundancy **Note:** The parameter is applicable only to non-V.21 packets.
Fax Relay Max Rate (bps) `max-rate` [FaxRelayMaxRate]	Defines the maximum rate (in bps) at which fax relay messages are transmitted (outgoing calls). ▪ **[0]** 2400 = 2.4 kbps ▪ **[1]** 4800 = 4.8 kbps ▪ **[2]** 7200 = 7.2 kbps ▪ **[3]** 9600 = 9.6 kbps ▪ **[4]** 12000 = 12.0 kbps ▪ **[5]** 14400 = 14.4 kbps (default) ▪ [6] 16800bps = 16.8 kbps ▪ [7] 19200bps = 19.2 kbps ▪ [8] 21600bps = 21.6 kbps ▪ [9] 24000bps = 24 kbps ▪ [10] 26400bps = 26.4 kbps ▪ [11] 28800bps = 28.8 kbps ▪ [12] 31200bps = 31.2 kbps ▪ [13] 33600bps = 33.6 kbps **Notes:** ▪ The rate is negotiated between both sides (i.e., the device adapts to the capabilities of the remote side). Negotiation of the T.38 maximum supported fax data rate is provided in SIP's SDP T38MaxBitRate parameter. The negotiated T38MaxBitRate is the minimum rate supported between the local and remote endpoints. ▪ Fax relay rates greater than 14.4 kbps are applicable only to V.34 / T.38 fax relay. For non-T.38 V.34 supporting devices, configuration greater than 14.4 kbps is truncated to 14.4 kbps.
Fax Relay ECM Enable `ecm-mode` [FaxRelayECMEnable]	Enables Error Correction Mode (ECM) mode during fax relay. ▪ **[0]** Disable ▪ **[1]** Enable (default)
Fax/Modem Bypass Coder Type [FaxModemBypassCoderType]	Determines the coder used by the device when performing fax/modem bypass. Typically, high-bit-rate coders such as G.711 should be used. ▪ **[0]** G.711Alaw= (Default) G.711 A-law 64 ▪ **[1]** G.711Mulaw = G.711 µ-law
Fax/Modem Bypass Packing Factor `packing-factor`	Defines the number (20 msec) of coder payloads used to generate a fax/modem bypass packet.

Parameter	Description
[FaxModemBypassM]	The valid range is 1, 2, or 3 coder payloads. The default is 1 coder payload.
`fax-modem-telephony-events-mode` [FaxModemNTEMode]	Determines whether the device sends RFC 2833 ANS/ANSam events upon detection of fax and/or modem Answer tones (i.e., CED tone). ▪ **[0]** = Disabled (default) ▪ **[1]** = Enabled **Note:** The parameter is applicable only when the fax or modem transport type is set to bypass or Transparent-with-Events.
Fax Bypass Payload Type `fax-bypass-payload-type` [FaxBypassPayloadType]	Defines the fax bypass RTP dynamic payload type. The valid range is 0 to 127. The default is 102.
`modem-bypass-payload-type` [ModemBypassPayloadType]	Defines the modem bypass dynamic payload type. The range is 0 to 127. The default is 103.
`volume` [FaxModemRelayVolume]	Defines the fax gain control. The range is -18 to -3, corresponding to -18 dBm to -3 dBm in 1-dB steps. The default is -6 dBm fax gain control.
Fax Bypass Output Gain `fax-bypass-output-gain` [FaxBypassOutputGain]	Defines the fax bypass output gain control. The range is -31 to +31 dB, in 1-dB steps. The default is 0 (i.e., no gain).
Modem Bypass Output Gain [ModemBypassOutputGain]	Defines the modem bypass output gain control. The range is -31 dB to +31 dB, in 1-dB steps. The default is 0 (i.e., no gain).
`modem-bypass-output-gain` [FaxModemBypassBasicRTPPacketInterval]	Defines the basic frame size used during fax/modem bypass sessions. ▪ **[0]** = (Default) Determined internally ▪ **[1]** = 5 msec (not recommended) ▪ **[2]** = 10 msec ▪ **[3]** = 20 msec **Note:** When set to 5 msec (1), the maximum number of simultaneous channels supported is 120.
`jitter-buffer-minimum-delay` [FaxModemBypasDJBufMinDelay]	Defines the Jitter Buffer delay (in milliseconds) during fax and modem bypass session. The range is 0 to 150 msec. The default is 40.
`enable-fax-modem-inband-network-detection` [EnableFaxModemInbandNetworkDetection]	Enables in-band network detection related to fax/modem. ▪ **[0]** = (Default) Disable. ▪ **[1]** = Enable. When the parameter is enabled on Bypass and transparent with events mode (VxxTransportType is set to 2 or 3), a detection of an Answer Tone from the network triggers a switch to bypass mode in addition to the local Fax/Modem tone detections. However, only a high bit-

Parameter	Description
	rate coder voice session effectively detects the Answer Tone sent by a remote endpoint. This can be useful when, for example, the payload of voice and bypass is the same, allowing the originator to switch to bypass mode as well.
`NSE-mode` [NSEMode]	Global parameter that enables Cisco's compatible fax and modem bypass mode, Named Signaling Event (NSE) packets. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_NSEMode). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
`NSE-payload-type` [NSEPayloadType]	Defines the NSE payload type for Cisco Bypass compatible mode. The valid range is 96-127. The default is 105. **Notes:** ▪ The parameter is applicable only to the Gateway application. ▪ Cisco gateways usually use NSE payload type of 100.
[T38UseRTPPort]	Defines the port (with relation to RTP port) for sending and receiving T.38 packets. ▪ **[0]** = (Default) Use the RTP port +2 to send/receive T.38 packets. ▪ **[1]** = Use the same port as the RTP port to send/receive T.38 packets. **Notes:** ▪ For the parameter to take effect, you must reset the device. ▪ When the device is configured to use V.152 to negotiate audio and T.38 coders, the UDP port published in SDP for RTP and for T38 must be different. Therefore, set the T38UseRTPPort parameter to 0.
T.38 Max Datagram Size `t38-mx-datagram-sz` [T38MaxDatagramSize]	Defines the maximum size of a T.38 datagram that the device can receive. This value is included in the outgoing SDP when T.38 is used. The valid range is 120 to 600. The default is 560.

Parameter	Description
T38 Fax Max Buffer `t38-fax-mx-buff` [T38FaxMaxBufferSize]	Defines the maximum size (in bytes) of the device's T.38 buffer. This value is included in the outgoing SDP when T.38 is used for fax relay over IP. The valid range is 500 to 3000. The default is 3000.
Detect Fax on Answer Tone `det-fax-on-ans-tone` [DetFaxOnAnswerTone]	Determines when the device initiates a T.38 session for fax transmission. ▪ **[0]** Initiate T.38 on Preamble = (Default) The device to which the called fax is connected initiates a T.38 session on receiving Preamble signal from the fax. ▪ **[1]** Initiate T.38 on CED = The device to which the called fax is connected initiates a T.38 session on receiving a CED answer tone from the fax. This option can only be used to relay fax signals, as the device sends T.38 Re-INVITE on detection of any fax/modem Answer tone (2100 Hz, amplitude modulated 2100 Hz, or 2100 Hz with phase reversals). The modem signal fails when using T.38 for fax relay. **Note:** The parameters is applicable only if the IsFaxUsed parameter is set to 1 (T.38 Relay) or 3 (Fax Fallback).
CED Transfer Mode [CEDTransferMode]	Defines the method for sending fax/modem CED (answering) tones. ▪ [0] Fax Relay or VBD = (Default) The device transfers the CED tone in Relay mode and starts the fax session immediately. ▪ [1] Voice Mode or VBD = The device transfers the CED tone in either Voice or Bypass mode and starts the fax session on V21 preamble. ▪ [2] RFC 4733 Blocking RTP VBD = The device transfers the CED tone in RFC 2833. This is applicable only to V.150.1 modem relay and fax bypass. ▪ [3] RFC 4733 Along with RTP VBD = The device transfers the CED tone in RFC 2833 and bypass, in parallel. For combined V.150.1 modem relay and fax relay, use this option. **Note:** The parameter is applicable only to the Gateway application.
T.38 Fax Session `t38-sess-imm-strt` [T38FaxSessionImmediateStart]	Enables fax transmission of T.38 "no-signal" packets to the terminating fax machine. ▪ **[0]** Disable (default) ▪ **[1]** Immediate Start on Fax = Device activates T.38 fax relay upon receipt of a re-INVITE with T.38 only in the SDP. ▪ **[2]** Immediate Start on Fax & Voice = Device activates T.38 fax relay upon receipt of a re-

Parameter	Description
	INVITE with T.38 and audio media in the SDP. The parameter is used for transmission from fax machines connected to the device and located inside a NAT. Generally, the firewall blocks T.38 (and other) packets received from the WAN, unless the device behind NAT sends at least one IP packet from the LAN to the WAN through the firewall. If the firewall blocks T.38 packets sent from the termination IP fax, the fax fails. To overcome this, the device sends No-Op (“no-signal”) packets to open a pinhole in the NAT for the answering fax machine. The originating fax does not wait for an answer, but immediately starts sending T.38 packets to the terminating fax machine. **Note:** To enable No-Op packet transmission, use the NoOpEnable and NoOpInterval parameters.
V.150.1 Modem over IP **Note:** These parameters are applicable only to the Gateway application.	
Profile Number [V1501AllocationProfile]	Defines the V.150.1 profile, which determines how many DSP channels support V.150.1. The value range is 0 to 20. The default is 0. **Note:** For the parameter to take effect, a device reset is required.
SSE Payload Type Rx `V1501-SSE-payload-type-rx` [V1501SSEPayloadTypeRx]	Defines the V.150.1 (modem relay protocol) State Signaling Event (SSE) payload type Rx. The value range is 96 to 127. The default is 105.
SSE Redundancy Depth `SSE-redundancy-depth` [V1501SSERedundancyDepth]	Defines the SSE redundancy depth. The value range is 1-6. The default is 3.
SPRT Transport Ch.0 Max Payload Size `SPRT-transport-channel0-max-payload-size` [V1501SPRTTransportChannel0MaxPayloadSize]	Defines the maximum payload size for V.150.1 SPRT Transport Channel 0. The range is 140 to 256. The default is 140.
SPRT Transport Ch.2 Max Payload Size `SPRT-transport-channel2-max-payload-size` [V1501SPRTTransportChannel2MaxPayloadSize]	Defines the maximum payload size for V.150.1 SPRT Transport Channel 2. The range is 132 to 256. The default is 132.
SPRT Transport Ch.2 Max Window Size `SPRT-transport-channel2-max-window-size` [V1501SPRTTransportChannel2MaxWindowSize]	Defines the maximum window size of SPRT transport channel 2. The value range is 8 to 32. The default is 8.
SPRT Transport Ch.3 Max Payload Size	Defines the maximum payload size for V.150.1 SPRT Transport Channel 3.

Parameter	Description
`SPRT-transport-channel3-max-payload-size` [V1501SPRTTransportChannel3MaxPayloadSize]	The range is 140 to 256. The default is 140.

57.10.2 DTMF and Hook-Flash Parameters

The DTMF and hook-flash parameters are described in the table below.

Table 57-38: DTMF and Hook-Flash Parameters

Parameter	Description
Hook-Flash Parameters	
Hook-Flash Code `hook-flash-code` [HookFlashCode]	Defines the digit pattern used by the PBX to indicate a Hook Flash event. When this pattern is detected from the Tel side, the device responds as if a Hook Flash event has occurred and sends a SIP INFO message if the HookFlashOption parameter is set to 1, 5, 6, or 7 (indicating a Hook Flash). If configured and a Hook Flash indication is received from the IP side, the device generates this pattern to the Tel side. The valid range is a 25-character string. The default is a null string. **Note:** The parameter can also be configured in a Tel Profile.

Hook-Flash Option
`hook-flash-option`
[HookFlashOption]
	Defines the hook-flash transport type (i.e., method by which hook-flash is sent and received). For digital interfaces: This feature is applicable only if the HookFlashCode parameter is configured.
	▪ **[0]** Not Supported = (Default) Hook-Flash indication is not sent.
	▪ **[1]** INFO = Sends proprietary INFO message (Broadsoft) with Hook-Flash indication. The device sends the INFO message as follows:
	Content-Type: application/broadsoft; version=1.0
	Content-Length: 17
	event flashhook
	▪ **[4]** RFC 2833 = This option is currently not supported.
	▪ **[5]** INFO (Lucent) = Sends proprietary SIP INFO message with Hook-Flash indication. The device sends the INFO message as follows:
	Content-Type: application/hook-flash
	Content-Length: 11
	signal=hf
	▪ **[6]** INFO (NetCentrex) = Sends proprietary SIP INFO message with Hook-Flash indication. The device sends the INFO message as follows:
	`Content-Type: application/dtmf-relay`
	`Signal=16`
	Where 16 is the DTMF code for hook flash.
	▪ **[7]** INFO (HUAWEI) = Sends a SIP INFO message with Hook-Flash indication. The device sends the INFO message as follows:
	`Content-Length: 17`

Parameter	Description

	`Content-Type: application/sscc`
	`event=flashhook`
	Note:
	- The device can interwork DTMF HookFlashCode to SIP INFO messages with Hook Flash indication.

DTMF Parameters	
`notify-on-sig-end` [MGCPDTMFDetectionPoint]	Determines when the detection of DTMF events is notified. - **[0]** = DTMF event is reported at the end of a detected DTMF digit. - **[1]** = (Default) DTMF event is reported at the start of a detected DTMF digit.
Declare RFC 2833 in SDP `rfc-2833-in-sdp` [RxDTMFOption]	Global parameter that enables the device to declare the RFC 2833 'telephony-event' parameter in the SDP. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_RxDTMFOption). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
First Tx DTMF Option `configure voip > gw dtmf-and-suppl dtmf-and-dialing > first-dtmf-option-type` [FirstTxDTMFOption]	Defines the first preferred transmit (Tx) DTMF negotiation method. - **[0]** Not Supported = (Default) No negotiation - DTMF digits are sent according to the parameters DTMFTransportType and RFC2833PayloadType. The RFC 2833 payload type is according to the RFC2833PayloadType parameter for transmit and receive. - **[1]** Info NORTEL = Sends DTMF digits according to IETF Internet-Draft draft-choudhuri-sip-info-digit-00. - **[2]** NOTIFY = Sends DTMF digits according to IETF Internet-Draft draft-mahy-sipping-signaled-digits-01. - **[3]** Info Cisco = Sends DTMF digits according to Cisco format. - **[4]** RFC 2833 = The device handles DTMF as follows: ✓ Negotiates RFC 2833 payload type using local and remote SDPs. ✓ Sends DTMF packets using RFC 2833 payload type according to the payload type in the received SDP. ✓ Expects to receive RFC 2833 packets with the same payload type according to the RFC2833PayloadType parameter. ✓ Removes DTMF digits in transparent mode (as part of the voice stream). - **[5]** Info KOREA = Sends DTMF digits according to Korea Telecom format. **Notes:** - When out-of-band DTMF transfer is used ([1], [2], [3], or [5]), the DTMFTransportType parameter is automatically set to [0] (DTMF digits are erased from the RTP stream). - For more information on DTMF transport, see "Configuring DTMF Transport Types" on page 189. - You can also configure the parameter per specific calls, using IP Profiles (IpProfile_FirstTxDtmfOption). For configuring IP Profiles, see "Configuring IP Profiles" on page 387.
Second Tx DTMF Option	Defines the second preferred transmit (Tx) DTMF negotiation method. The first preferred method is configured by the

Parameter	Description
`configure voip > gw dtmf-and-suppl dtmf-and-dialing > second-dtmf-option-type` [SecondTxDTMFOption]	FirstTxDTMFOption parameter. For a description of the optional values for the parameter, see the FirstTxDTMFOption parameter above. **Note:** You can also configure the parameter per specific calls, using IP Profiles (IpProfile_SecondTxDtmfOption). For configuring IP Profiles, see "Configuring IP Profiles" on page 387.
[DisableAutoDTMFMute]	Enables the automatic muting of DTMF digits when out-of-band DTMF transmission is used. ▪ **[0]** = (Default) Automatic mute is used. ▪ **[1]** = No automatic mute of in-band DTMF. When the parameter is set to 1, the DTMF transport type is set according to the parameter DTMFTransportType and the DTMF digits aren't muted if out-of-band DTMF mode is selected (FirstTxDTMFOption set to 1, 2 or 3). This enables the sending of DTMF digits in-band (transparent of RFC 2833) in addition to out-of-band DTMF messages. **Note:** Usually this mode is not recommended.
Enable Digit Delivery to IP `digit-delivery-2ip` [EnableDigitDelivery2IP]	Enables the Digit Delivery feature whereby DTMF digits are sent to the destination IP address after the Tel-to-IP call is answered. ▪ **[0]** Disable (default). ▪ **[1]** Enable = Enable digit delivery to IP. To enable this feature, modify the called number to include at least one 'p' character. The device uses the digits before the 'p' character in the initial INVITE message. After the call is answered, the device waits for the required time (number of 'p' multiplied by 1.5 seconds), and then sends the rest of the DTMF digits using the method chosen (in-band or out-of-band). **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ The called number can include several 'p' characters (1.5 seconds pause), for example, 1001pp699, 8888p9p300.
Enable Digit Delivery to Tel `digit-delivery-2tel` [EnableDigitDelivery]	Enables the Digit Delivery feature, which sends DTMF digits of the called number to the device's B-channel (phone line) after the call is answered for IP-to-Tel calls. ▪ **[0]** Disable (default) ▪ **[1]** Enable If the called number in IP-to-Tel call includes the characters 'w' or 'p', the device places a call with the first part of the called number (before 'w' or 'p') and plays DTMF digits after the call is answered. If the character 'w' is used, the device waits for detection of a dial tone before it starts playing DTMF digits. For example, if the called number is '1007766p100', the device places a call with 1007766 as the destination number, then after the call is answered it waits 1.5 seconds ('p') and plays the rest of the number (100) as DTMF digits. Additional examples: 1664wpp102, 66644ppp503, and 7774w100pp200. **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ The parameter can also be configured in a Tel Profile.
Special Digit Representation `special-digit-rep`	Defines the representation for ‘special’ digits (‘*’ and ‘#’) that are used for out-of-band DTMF signaling (using SIP INFO/NOTIFY).

Parameter	Description
[UseDigitForSpecialDTMF]	▪ **[0]** Special = (Default) Uses the strings '*' and '#'. ▪ **[1]** Numeric = Uses the numerical values 10 and 11.
`isdn-keypad-mode` [ISDNKeypadMode]	Enables the device to send DTMF digits received in the called party number from the IP side, as Keypad facility IE in ISDN INFORMATION messages to PSTN. ▪ [0] Don't send = (Default) All digits are sent as DTMF to PSTN (i.e., not sent as Keypad). ▪ [1] During Call Establishment = DTMF digits after * or # (inclusive) are sent as Keypad only during call establishment and call disconnect. During an established call, all digits are sent as DTMF. ▪ [2] Always = DTMF digits after * or # (inclusive) are always sent as Keypad (call establishment, connect, and disconnect). For more information, see Interworking Keypad DTMFs for SIP-to-ISDN Calls on page 499. **Note:** This feature is not applicable to re-INVITE messages.

57.10.3 Digit Collection and Dial Plan Parameters

The digit collection and dial plan parameters are described in the table below.

Table 57-39: Digit Collection and Dial Plan Parameters

Parameter	Description
Dial Plan Index `dial-plan-index` [DialPlanIndex]	Defines the Dial Plan index to use in the external Dial Plan file. The Dial Plan file is loaded to the device as a .dat file (converted using the DConvert utility). The Dial Plan index can be defined globally or per Tel Profile. The valid value range is 0 to 7, where 0 denotes PLAN1, 1 denotes PLAN2, and so on. The default is -1, indicating that no Dial Plan file is used. **Notes:** ▪ If the parameter is configured to select a Dial Plan index, the settings of the parameter DigitMapping are ignored. ▪ If the parameter is configured to select a Dial Plan index from an external Dial Plan file, the device first attempts to locate a matching digit pattern in the Dial Plan file, and if not found, then attempts to locate a matching digit pattern in the Digit Map rules configured by the DigitMapping parameter. ▪ The parameter is applicable also to ISDN with overlap dialing. ▪ The parameter can also be configured in a Tel Profile. ▪ For more information on the Dial Plan file, see "Dialing Plans for Digit Collection" on page 653.
`tel2ip-src-nb-map-dial-index` [Tel2IPSourceNumberMappingDialPlanIndex]	Defines the Dial Plan index in the external Dial Plan file for the Tel-to-IP Source Number Mapping feature. The valid value range is 0 to 7, defining the Dial Plan index [Plan x] in the Dial Plan file. The default is -1 (disabled). For more information on this feature, see "Modifying ISDN-to-IP Calling Party Number using Dial Plan File" on page 658.

Parameter	Description
Digit Mapping Rules `default-dm` [DigitMapping]	Defines the digit map pattern (used to reduce the dialing period when ISDN overlap dialing). If the digit string (i.e., dialed number) matches one of the patterns in the digit map, the device stops collecting digits and establishes a call with the collected number. The digit map pattern can contain up to 52 options (rules), each separated by a vertical bar (|). The maximum length of the entire digit pattern is 152 characters. The available notations include the following: ▪ **[n-m]:** Range of numbers (not letters). ▪ **.** (single dot): Repeat digits until next notation (e.g., T). ▪ **x:** Any single digit. ▪ **T:** Dial timeout (configured by the TimeBetweenDigits parameter). ▪ **S:** Short timer (configured by the TimeBetweenDigits parameter; default is two seconds) that can be used when a specific rule is defined after a more general rule. For example, if the digit map is 99|998, then the digit collection is terminated after the first two 9 digits are received. Therefore, the second rule of 998 can never be matched. But when the digit map is 99s|998, then after dialing the first two 9 digits, the device waits another two seconds within which the caller can enter the digit 8. An example of a digit map is shown below: 11xS|00T|[1-7]xxx|8xxxxxxx|#xxxxxxx|*xx|91xxxxxxxxxx|9011x.T In the example above, the last rule can apply to International numbers: 9 for dialing tone, 011 Country Code, and then any number of digits for the local number ('x.'). **Notes:** ▪ For ISDN interfaces, the digit map mechanism is applicable only when ISDN overlap dialing is used (ISDNRxOverlap is set to 1). ▪ If the DialPlanIndex parameter is configured (to select a Dial Plan index), then the device first attempts to locate a matching digit pattern in the Dial Plan file, and if not found, then attempts to locate a matching digit pattern in the Digit Map rules configured by the DigitMapping parameter. ▪ For more information on digit mapping, see "Digit Mapping" on page 497.
Max Digits in Phone Num `mxdig-b4-dialing` [MaxDigits]	Defines the maximum number of collected destination number digits that can be received from the Tel side when ISDN Tel-to-IP overlap dialing is performed. When the number of collected digits reaches this maximum, the device uses these digits for the called destination number. The valid range is 1 to 49. The default is 30. **Note:** ▪ Instead of using the parameter, Digit Mapping rules can be configured.
Inter Digit Timeout for Overlap Dialing `time-btwn-dial-digs` [TimeBetweenDigits]	Defines the time (in seconds) that the device waits between digits that are received from the PSTN or IP during overlap dialing. When this inter-digit timeout expires, the device uses the collected digits to dial the called destination number. The valid range is 1 to 10. The default is 4.

57.10.4 Voice Mail Parameters

The voice mail parameters are described in the table below. For more information on the Voice Mail application, refer to the *CPE Configuration Guide for Voice Mail*.

Table 57-40: Voice Mail Parameters

Parameter	Description
Voice Mail Interface `vm-interface` [VoiceMailInterface]	Enables the device's Voice Mail application and determines the communication method between the device and PBX. - **[0]** None (default) - **[1]** DTMF - **[2]** SMDI - [3] QSIG - [4] SETUP Only = Applicable only to ISDN. - [5] MATRA/AASTRA QSIG - [6] QSIG SIEMENS = QSIG MWI activate and deactivate messages include Siemens Manufacturer Specific Information (MSI) - [8] ETSI = Euro ISDN, according to ETS 300 745-1 V1.2.4, section 9.5.1.1. Enables MWI interworking from IP to Tel, typically used for BRI phones. **Note:** To disable voice mail per Trunk Group, you can use a Tel Profile with the EnableVoiceMailDelay parameter set to disabled (0). This eliminates the phenomenon of call delay on Trunks not implementing voice mail when voice mail is enabled using this global parameter.
Enable VoiceMail URI `voicemail-uri` [EnableVMURI]	Enables the interworking of target and cause for redirection from Tel to IP and vice versa, according to RFC 4468. - **[0]** Disable (default) - **[1]** Enable Upon receipt of an ISDN Setup message with Redirect values, the device maps the Redirect phone number to the SIP 'target' parameter and the Redirect number reason to the SIP 'cause' parameter in the Request-URI. Redirecting Reason >> SIP Response Code Unknown >> 404 User busy >> 486 No reply >> 408 Deflection >> 487/480 Unconditional >> 302 Others >> 302 If the device receives a Request-URI that includes a 'target' and 'cause' parameter, the 'target' is mapped to the Redirect phone number and the 'cause' is mapped to the Redirect number reason.
SMDI Parameters	
Enable SMDI [SMDI]	Enables Simplified Message Desk Interface (SMDI) interface on the device. - [0] Disable = (Default) Normal serial - [1] Enable (Bellcore) - [2] Ericsson MD-110 - [3] NEC (ICS)

Parameter	Description
	Notes: - For the parameter to take effect, a device reset is required. - When the RS-232 connection is used for SMDI messages (Serial SMDI), it cannot be used for other applications, for example, to access the Command Line Interface (CLI).
SMDI Timeout [SMDITimeOut]	Defines the time (in msec) that the device waits for an SMDI Call Status message before or after a Setup message is received. The parameter synchronizes the SMDI and analog CAS interfaces. If the timeout expires and only an SMDI message is received, the SMDI message is dropped. If the timeout expires and only a Setup message is received, the call is established. The valid range is 0 to 10000 (i.e., 10 seconds). The default is 2000.
Message Waiting Indication (MWI) Parameters	
MWI Off Digit Pattern `mwi-off-dig-ptrn` [MWIOffCode]	Defines the digit code used by the device to notify the PBX that there are no messages waiting for a specific extension. This code is added as prefix to the dialed number. The valid range is a 25-character string.
MWI On Digit Pattern `mwi-on-dig-ptrn` [MWIOnCode]	Defines the digit code used by the device to notify the PBX of messages waiting for a specific extension. This code is added as prefix to the dialed number. The valid range is a 25-character string.
MWI Suffix Pattern `mwi-suffix-pattern` [MWISuffixCode]	Defines the digit code used by the device as a suffix for 'MWI On Digit Pattern' and 'MWI Off Digit Pattern'. This suffix is added to the generated DTMF string after the extension number. The valid range is a 25-character string.
MWI Source Number `mwi-source-number` [MWISourceNumber]	Defines the calling party's phone number used in the Q.931 MWI Setup message to PSTN. If not configured, the channel's phone number is used as the calling number.

`mwi-subs-ipgrpid`
[MWISubscribeIPGroupID]
	Defines the IP Group ID used when subscribing to an MWI server. The 'The SIP Group Name' field value of the IP Group table is used as the Request-URI host name in the outgoing MWI SIP SUBSCRIBE message. The request is sent to the IP address defined for the Proxy Set that is associated with the IP Group. The Proxy Set's capabilities such as proxy redundancy and load balancing are also applied to the message.
	For example, if the 'SIP Group Name' field of the IP Group is set to "company.com", the device sends the following SUBSCRIBE message:
	`SUBSCRIBE sip:company.com...`
	Instead of:
	`SUBSCRIBE sip:10.33.10.10...`
	Note: If the parameter is not configured, the MWI SUBSCRIBE message is sent to the MWI server as defined by the MWIServerIP parameter.

[NotificationIPGroupID]	Defines the IP Group ID to which the device sends SIP NOTIFY MWI messages. **Notes:**

Parameter	Description
	▪ This is used for MWI Interrogation. For more information on the interworking of QSIG MWI to IP, see Message Waiting Indication on page 506. ▪ To determine the handling method of MWI Interrogation messages, use the TrunkGroupSettings_MWIInterrogationType, parameter (in the Trunk Group Settings table).
[MWIQsigMsgCentreldIDPartyNumber]	Defines the Message Centred ID party number used for QSIG MWI messages. If not configured (default), the parameter is not included in MWI (activate and deactivate) QSIG messages. The valid value is a string.
Digit Patterns The following digit pattern parameters apply only to voice mail applications that use the DTMF communication method. For the available pattern syntaxes, refer to the *CPE Configuration Guide for Voice Mail*.	
Forward on Busy Digit Pattern (Internal) `fwd-bsy-dig-ptrn-int` [DigitPatternForwardOnBusy]	Defines the digit pattern used by the PBX to indicate 'call forward on busy' when the original call is received from an internal extension. The valid range is a 120-character string.
Forward on No Answer Digit Pattern (Internal) `fwd-no-ans-dig-pat-int` [DigitPatternForwardOnNoAnswer]	Defines the digit pattern used by the PBX to indicate 'call forward on no answer' when the original call is received from an internal extension. The valid range is a 120-character string.
Forward on Do Not Disturb Digit Pattern (Internal) `fwd-dnd-dig-ptrn-int` [DigitPatternForwardOnDND]	Defines the digit pattern used by the PBX to indicate 'call forward on do not disturb' when the original call is received from an internal extension. The valid range is a 120-character string.
Forward on No Reason Digit Pattern (Internal) `fwd-no-rsn-dig-ptrn-int` [DigitPatternForwardNoReason]	Defines the digit pattern used by the PBX to indicate 'call forward with no reason' when the original call is received from an internal extension. The valid range is a 120-character string.
Forward on Busy Digit Pattern (External) `fwd-bsy-dig-ptrn-ext` [DigitPatternForwardOnBusyExt]	Defines the digit pattern used by the PBX to indicate 'call forward on busy' when the original call is received from an external line (not an internal extension). The valid range is a 120-character string.
Forward on No Answer Digit Pattern (External) `fwd-no-ans-dig-pat-ext` [DigitPatternForwardOnNoAnswerExt]	Defines the digit pattern used by the PBX to indicate 'call forward on no answer' when the original call is received from an external line (not an internal extension). The valid range is a 120-character string.

Parameter	Description
Forward on Do Not Disturb Digit Pattern (External) `fwd-dnd-dig-ptrn-ext` [DigitPatternForwardOnDNDExt]	Defines the digit pattern used by the PBX to indicate 'call forward on do not disturb' when the original call is received from an external line (not an internal extension). The valid range is a 120-character string.
Forward on No Reason Digit Pattern (External) `fwd-no-rsn-dig-ptrn-ext` [DigitPatternForwardNoReasonExt]	Defines the digit pattern used by the PBX to indicate 'call forward with no reason' when the original call is received from an external line (not an internal extension). The valid range is a 120-character string.
Internal Call Digit Pattern `int-call-dig-ptrn` [DigitPatternInternalCall]	Defines the digit pattern used by the PBX to indicate an internal call. The valid range is a 120-character string.
External Call Digit Pattern `ext-call-dig-ptrn` [DigitPatternExternalCall]	Defines the digit pattern used by the PBX to indicate an external call. The valid range is a 120-character string.
Disconnect Call Digit Pattern `disc-call-dig-ptrn` [TelDisconnectCode]	Defines a digit pattern that when received from the Tel side, indicates the device to disconnect the call. The valid range is a 25-character string.
Digit To Ignore Digit Pattern `dig-to-ignore-dig-pattern` [DigitPatternDigitToIgnore]	Defines a digit pattern that if received as Src (S) or Redirect (R) numbers is ignored and not added to that number. The valid range is a 25-character string.

57.10.5 Supplementary Services Parameters

This subsection describes the device's supplementary telephony services parameters.

57.10.5.1 Caller ID Parameters

The caller ID parameters are described in the table below.

Table 57-41: Caller ID Parameters

Parameter	Description
Enable Caller ID `enable-caller-id` [EnableCallerID]	Global parameter that enables Caller ID. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Asserted Identity Mode `asserted-identity-m` [AssertedIdMode]	Determines whether the SIP header P-Asserted-Identity or P-Preferred-Identity is added to the sent INVITE, 200 OK, or UPDATE request for Caller ID (or privacy). These headers are used to present the calling party's Caller ID, which is composed of a Calling Number and a Calling Name (optional). ▪ **[0]** Disabled = (Default) P-Asserted-Identity and P-Preferred-Identity headers are not added. ▪ **[1]** Add P-Asserted-Identity ▪ **[2]** Add P-Preferred-Identity The used header also depends on the calling Privacy (allowed or restricted). These headers are used together with the Privacy header. If Caller ID is restricted (i.e., P-Asserted-Identity is not sent), the Privacy header includes the value 'id' ('Privacy: id'). Otherwise, for allowed Caller ID, 'Privacy: none' is used. If Caller ID is restricted (received from Tel), the From header is set to <anonymous@anonymous.invalid>. The 200 OK response can contain the connected party CallerID - Connected Number and Connected Name. For example, if the call is answered by the device, the 200 OK response includes the P-Asserted-Identity with Caller ID. The device interworks (in some ISDN variants), the Connected Party number and name from Q.931 Connect message to SIP 200 OK with the P-Asserted-Identity header. In the opposite direction, if the ISDN device receives a 200 OK with P-Asserted-Identity header, it interworks it to the Connected party number and name in the Q.931 Connect message, including its privacy.
Use Destination As Connected Number [UseDestinationAsConnectedNumber]	Enables the device to include the Called Party Number, from outgoing Tel calls (after number manipulation), in the SIP P-Asserted-Identity header. The device includes the SIP P-Asserted-Identity header in 180 Ringing and 200 OK responses for IP-to-Tel calls. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Notes:** ▪ For this feature to function, you also need to enable the device to include the P-Asserted-Identity header in 180/200 OK responses, by setting the AssertedIDMode parameter to **Add P-Asserted-Identity.**

Parameter	Description
	▪ If the received Q.931 Connect message contains a Connected Party Number, this number is used in the P-Asserted-Identity header in 200 OK response. ▪ The parameter is applicable to ISDN interfaces.
Caller ID Transport Type `caller-ID-transport-type` [CallerIDTransportType]	Determines the device's behavior for Caller ID detection. ▪ **[0]** Disable = The caller ID signal is not detected - DTMF digits remain in the voice stream. ▪ **[1]** Relay = (Currently not applicable.) ▪ **[3]** Mute = (Default) The caller ID signal is detected from the Tel side and then erased from the voice stream.

57.10.5.2 Call Waiting Parameters

The call waiting parameters are described in the table below.

Table 57-42: Call Waiting Parameters

Parameter	Description
Enable Call Waiting `call-waiting` [EnableCallWaiting]	Enables the Call Waiting feature. ▪ **[0]** Disable ▪ **[1]** Enable (Default) If enabled and the device initiates a Tel-to-IP call to a destination that is busy, it plays a call waiting ringback tone to the caller. The tone is played only if the destination returns a 182 "Queued" SIP response. **Notes:** ▪ The device's Call Progress Tones (CPT) file must include a Call Waiting ringback tone.
[Send180ForCallWaiting]	Determines the SIP response code for indicating Call Waiting. ▪ **[0]** = (Default) Use 182 Queued response to indicate call waiting. ▪ **[1]** = Use 180 Ringing response to indicate call waiting.

57.10.5.3 Call Forwarding Parameters

The call forwarding parameters are described in the table below.

Table 57-43: Call Forwarding Parameters

Parameter	Description
Enable Call Forward `call-forward` [EnableForward]	Enables the Call Forwarding feature. ▪ **[0]** Disable ▪ **[1]** Enable (Default) **Notes:** ▪ To use this service, the devices at both ends must support this option. ▪ For the device to respond to SIP 3xx responses with a new SIP request (forwarding the original request), set the parameter to **Enable**.

57.10.5.4 Call Hold Parameters

The call hold parameters are described in the table below.

Table 57-44: Call Hold Parameters

Parameter	Description
Enable Hold `hold` [EnableHold]	Global parameter that enables the interworking of the Hold/Retrieve supplementary service from ISDN to SIP. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_EnableHold). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
Hold Format `hold-format` [HoldFormat]	Defines the format of the SDP in the sent re-INVITE hold request. ▪ **[0]** 0.0.0.0 = (Default) The SDP "c=" field contains the IP address "0.0.0.0" and the "a=inactive" attribute. ▪ **[1]** Send Only = The SDP "c=" field contains the device's IP address and the "a=sendonly" attribute. ▪ **[2]** x.y.z.t = The SDP "c=" field contains the device's IP address and the "a=inactive" attribute. **Notes:** ▪ The device does not send any RTP packets when it is in hold state. ▪ The parameter is applicable only to QSIG and Euro ISDN protocols.
Held Timeout `held-timeout` [HeldTimeout]	Defines the time interval that the device allows for a call to remain on hold. If a Resume (un-hold Re-INVITE) message is received before the timer expires, the call is renewed. If this timer expires, the call is released (terminated). ▪ **[-1]** = (Default) The call is placed on hold indefinitely until the initiator of the on hold retrieves the call again. ▪ **[0 - 2400]** = Time to wait (in seconds) after which the call is released.
`dtmf-during-hold` [PlayDTMFduringHold]	Determines whether the device sends DTMF signals (or DTMF SIP INFO message) when a call is on hold. ▪ **[0]** = (Default) Disable. ▪ **[1]** = Enable - If the call is on hold, the device stops playing the Held tone (if it is played) and sends DTMF: ✓ To Tel side: plays DTMF digits according to the received SIP INFO message(s). (The stopped held tone is not played again.) ✓ To IP side: sends DTMF SIP INFO messages to an IP destination if it detects DTMF digits from the Tel side.

57.10.5.5 Call Transfer Parameters

The call transfer parameters are described in the table below.

Table 57-45: Call Transfer Parameters

Parameter	Description
Enable Transfer `enable-transfer` [EnableTransfer]	Enables the Call Transfer feature. ▪ **[0]** Disable ▪ **[1]** Enable = (Default) The device responds to a REFER message with the Referred-To header to initiate a call transfer. **Notes:** ▪ To use call transfer, the devices at both ends must support this option. ▪ To use call transfer, set the parameter EnableHold to 1.
Transfer Prefix `transfer-prefix` [xferPrefix]	Defines the string that is added as a prefix to the transferred/forwarded called number when the REFER/3xx message is received. **Notes:** ▪ The number manipulation rules apply to the user part of the Refer-To and/or Contact URI before it is sent in the INVITE message. ▪ The parameter can be used to apply different manipulation rules to differentiate transferred / forwarded number from the originally dialed number.
Enable Semi-Attended Transfer `semi-att-transfer` [EnableSemiAttendedTransfer]	Determines the device behavior when Transfer is initiated while in Alerting state. ▪ **[0]** Disable = (Default) Send REFER with the Replaces header. ▪ **[1]** Enable = Send CANCEL, and after a 487 response is received, send REFER without the Replaces header.
Blind `blind-transfer` [KeyBlindTransfer]	Defines the keypad sequence to activate blind transfer for established Tel-to-IP calls. The Tel user can perform blind transfer by dialing the KeyBlindTransfer digits, followed by a transferee destination number. After the KeyBlindTransfer DTMF digits sequence is dialed, the current call is put on hold (using a Re-INVITE message), a dial tone is played to the channel, and then the phone number collection starts. After the destination phone number is collected, it is sent to the transferee in a SIP REFER request in a Refer-To header. The call is then terminated and a confirmation tone is played to the channel. If the phone number collection fails due to a mismatch, a reorder tone is played to the channel.
`blind-xfer-disc-tmo` [BlindTransferDisconnectTimeout]	Defines the duration (in milliseconds) for which the device waits for a disconnection from the Tel side after the Blind Transfer Code (KeyBlindTransfer) has been identified. When this timer expires, a SIP REFER message is sent toward the IP side. If the parameter is set to 0, the REFER message is immediately sent. The valid value range is 0 to 1,000,000. The default is 0.

Parameter	Description
QSIG Path Replacement Mode `qsig-path-replacement-md` [QSIGPathReplacementMode]	Enables QSIG transfer for IP-to-Tel and Tel-to-IP calls. - [0] IP2QSIGTransfer = (Default) Enables IP-to-QSIG transfer. - [1] QSIG2IPTransfer = Enables QSIG-to-IP transfer.
`replace-tel2ip-calnum-to` [ReplaceTel2IPCallingNumTimeout]	Defines the maximum duration (timeout) to wait between call Setup and Facility with Redirecting Number for replacing the calling number (for Tel-to-IP calls). The valid value range is 0 to 10,000 msec. The default is 0. The interworking of the received Setup message to a SIP INVITE is suspended when the parameter is set to any value greater than 0. This means that the redirecting number in the Setup message is not checked. When a subsequent Facility with Call Transfer Complete/Update is received with a non-empty Redirection Number, the Calling Number is replaced with the received redirect number in the sent INVITE message. If the timeout expires, the device sends the INVITE without changing the calling number. **Notes:** - The suspension of the INVITE message occurs for all calls. - The parameter is applicable to QSIG.

57.10.5.6 Multi-Line Extensions and Supplementary Services Parameters

The multi-line extensions and supplementary services parameters are described in the table below.

Table 57-46: Multi-line Extensions and Supplementary Services Parameters

Parameter	Description
Supplementary Services Table	
Supplementary Services Table `configure voip/gw digitalgw isdn-supp-serv` [ISDNSuppServ]	Defines phone extension numbers per BRI port and configures various supplementary services per endpoint. The format of the ini file table parameter is as follows: [ISDNSuppServ] FORMAT ISDNSuppServ_Index = ISDNSuppServ_PhoneNumber, ISDNSuppServ_LocalPhoneNumber, ISDNSuppServ_Module, ISDNSuppServ_Port, ISDNSuppServ_UserId, ISDNSuppServ_UserPassword, ISDNSuppServ_CallerID, ISDNSuppServ_IsPresentationRestricted, ISDNSuppServ_IsCallerIDEnabled, ISDNSuppServ_CFB2PhoneNumber, ISDNSuppServ_CFNR2PhoneNumber, ISDNSuppServ_CFU2PhoneNumber, ISDNSuppServ_NoReplyTime; [\ISDNSuppServ] For a detailed description of the table, see "Configuring Multi-Line Extensions and Supplementary Services" on page 513.

57.10.5.7 Three-Way Conferencing Parameters

The three-way conferencing parameters are described in the table below.

Table 57-47: Three-Way Conferencing Parameters

Parameter	Description
Enable 3-Way Conference `enable-3w-conf` [Enable3WayConference]	Enables the 3-Way Conference feature. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** For the parameter to take effect, a device reset is required.
3-Way Conference Mode `3w-conf-mode` [3WayConferenceMode]	Defines the mode of operation for three-way conferencing. ▪ **[0]** AudioCodes Media Server = (Default) The conference-initiating INVITE sent by the device, uses the ConferenceID concatenated with a unique identifier as the Request-URI. This same Request-URI is set as the Refer-To header value in the REFER messages that are sent to the two remote parties. This conference mode is used when operating with AudioCodes IPMedia conferencing server. ▪ **[1]** Non-AudioCodes Media Server = The conference-initiating INVITE sent by the device, uses only the ConferenceID as the Request-URI. The Conference server sets the Contact header of the 200 OK response to the actual unique identifier (Conference URI) to be used by the participants. This Conference URI is then included by the device in the Refer-To header value in the REFER messages sent by the device to the remote parties. The remote parties join the conference by sending INVITE messages to the conference using this conference URI. ▪ **[2]** On Board = On-board, three-way conference. The conference is established on the device without the need of an external Conference server. You can limit the number of simultaneous, on-board 3-way conference calls, by using the MaxInBoardConferenceCalls parameter. ▪ **[3]** Huawei Media Server = The conference is managed by an external, third-party Conferencing server. The conference-initiating INVITE sent by the device, uses only the ConferenceID as the Request-URI. The Conferencing server sets the Contact header of the 200 OK response to the actual unique identifier (Conference URI) to be used by the participants. The Conference URI is included in the URI of the REFER with a Replaces header sent by the device to the Conferencing server. The Conferencing server then sends an INVITE with a Replaces header to the remote participants. **Notes:** ▪ The parameter is applicable only to BRI interfaces. ▪ When using an external Conferencing server, a conference call with up to six participants can be established.
Max. 3-Way Conference [MaxInBoardConferenceCalls]	Defines the maximum number of simultaneous, on-board three-way conference calls. The valid range is 0 to 5. The default is 2. **Notes:** ▪ For enabling on-board, three-way conferencing, use the 3WayConferenceMode parameter. ▪ The parameter is applicable only to BRI interfaces.

Parameter	Description
Establish Conference Code `estb-conf-code` [ConferenceCode]	Defines the DTMF digit pattern, which upon detection generates the conference call when three-way conferencing is enabled (Enable3WayConference is set to 1). The valid range is a 25-character string. The default is “!” (Hook-Flash). **Note:** If the FlashKeysSequenceStyle parameter is set to 1 or 2, the setting of the ConferenceCode parameter is overridden.
Conference ID `conf-id` [ConferenceID]	Defines the Conference Identification string. The valid value is a string of up to 16 characters. The default is "conf". The device uses this identifier in the Conference-initiating INVITE that is sent to the media server when the Enable3WayConference parameter is set to 1.

57.10.5.8 MLPP and Emergency Call Parameters

The Multilevel Precedence and Preemption (MLPP) and emergency E911 call parameters are described in the table below.

Table 57-48: MLPP and Emergency E911 Call Parameters

Parameter	Description
Call Priority Mode `call-prio-mode` [CallPriorityMode]	Defines priority call handling, for all calls. ▪ **[0]** Disable (default). ▪ **[1]** MLPP = MLPP Priority Call handling is enabled. MLPP prioritizes call handling whereby the relative importance of various kinds of communications is strictly defined, allowing higher precedence communication at the expense of lower precedence communications. Higher priority calls override less priority calls when, for example, congestion occurs in a network. ▪ **[2]** Emergency = Preemption of IP-to-Tel E911 emergency calls. If the device receives an E911 call and there are unavailable channels to receive the call, the device terminates one of the channel calls and sends the E911 call to that channel. The preemption is done only on a channel pertaining to the same Trunk Group for which the E911 call was initially destined and if the channel select mode (configured by the ChannelSelectMode parameter) is set to other than By Dest Phone Number (0). The preemption is done only if the incoming IP-to-Tel call is identified as an emergency call. The device identifies emergency calls by one of the following: ✓ The destination number of the IP call matches one of the numbers defined by the EmergencyNumbers parameter. (For E911, you must define the parameter with the value "911".) ✓ The incoming SIP INVITE message contains the “emergency” value in the Priority header. **Notes:** ▪ The parameter is applicable to ISDN. ▪ MLPP and Emergency services can also be configured in a Tel Profile.

Parameter	Description
	▪ For more information, see "Pre-empting Existing Call for E911 IP-to-Tel Call" on page 509.
Emergency E911 Parameters	
E911 Gateway [E911Gateway]	Enables Enhanced 9-1-1(E9-1-1) support for ELIN handling in a Microsoft Lync Server environment and routing to a PSTN-based emergency service provider. ▪ [0] None = (Default) Disable ▪ [1] NG911 Gateway = Enables the ELIN Gateway. ▪ [2] Location Based Manipulations = Enables ELIN Gateway and location-based manipulation. For more information, see Location Based Emergency Routing on page 299. For more information on E9-1-1 in a Lync environment, see Enhanced 9-1-1 Support for Lync Server on page 289. **Note:** The parameter is applicable only to Gateway calls.
[E911CallbackTimeout]	Defines the maximum interval within which the PSAP can use the ELIN to call back the E9-1-1 caller. This interval starts from when the initial call established with the PSAP is terminated. The valid range is 1 to 60 (minutes). The default is 30.
Emergency Special Release Cause `emrg-spcl-rel-cse` [EmergencySpecialReleaseCause]	Enables the device to send a SIP 503 "Service Unavailable" response if an emergency call cannot be established (i.e., rejected). This can occur, for example, due to the PSTN (for example, the destination is busy or not found) or ELIN Gateway (for example, lack of resources or an internal error). ▪ [0] = Disable (default) ▪ [1] = Enable
Emergency Numbers `emerg-nbs` [EmergencyNumbers]	Defines a list of “emergency” numbers. For ISDN: These emergency numbers are used for the preemption of E911 IP-to-Tel calls when there are unavailable or busy channels. In this scenario, the device terminates one of the busy channels and sends the emergency call to this channel. This feature is enabled by setting the CallPriorityMode parameter to 2 (“Emergency”). For a description of this feature, see "Pre-empting Existing Call for E911 IP-to-Tel Call" on page 509. The list can include up to four different numbers, where each number can be up to four digits long. Example: EmergencyNumbers = ‘100’,’911’,’112’
Multilevel Precedence and Preemption (MLPP) Parameters	
MLPP Default Namespace `mlpp-dflt-namespace` [MLPPDefaultNamespace]	Determines the namespace used for MLPP calls received from the ISDN side without a Precedence IE and destined for an Application server. This value is used in the Resource-Priority header of the outgoing SIP INVITE request. ▪ [1] DSN (default) ▪ [2] DOD ▪ [3] DRSN ▪ [5] UC ▪ [7] CUC **Note:** If the ISDN message contains a Precedence IE, the device automatically interworks the "network identity" digits in

Parameter	Description
	the IE to the network domain subfield in the Resource-Priority header. For more information, see Multilevel Precedence and Preemption on page 509.
[ResourcePriorityNetworkDomains]	Defines up to 32 user-defined MLPP network domain names (namespaces). This value is used in the AS-SIP Resource-Priority header of the outgoing SIP INVITE request. The parameter is used in combination with the MLPPDefaultNamespace parameter, where you need to enter the table row index as its value. The parameter is also used for mapping the Resource-Priority field value of the SIP Resource-Priority header to the ISDN Precedence Level IE. The mapping is configured by the field, EnableIp2TelInterworking: ▪ Disabled: The network-domain field in the Resource-Priority header is set to "0 1 0 0" (i.e., "routine") in the Precedence Level field. ▪ Enabled: The network-domain field in the Resource-Priority header is set in the Precedence Level field according to Table 5.3.2.12-4 (Mapping of RPH r-priority Field to Precedence Level Value). The domain name can be a string of up to 10 characters. The format of this table ini file parameter is as follows: FORMAT ResourcePriorityNetworkDomains_Index = ResourcePriorityNetworkDomains_Name, ResourcePriorityNetworkDomains_EnableIp2TelInterworking; ResourcePriorityNetworkDomains 1 = dsn, 0; ResourcePriorityNetworkDomains 2 = dod, 0; ResourcePriorityNetworkDomains 3 = drsn, 0; ResourcePriorityNetworkDomains 5 = uc, 1; ResourcePriorityNetworkDomains 7 = cuc, 0; [\ResourcePriorityNetworkDomains] **Notes:** ▪ Indices 1, 2, 3, 5, and 7 cannot be modified and are defined for DSN, DOD, DRSN, UC, and CUC, respectively. ▪ If the MLPPDefaultNamespace parameter is set to -1, interworking from PSTN NI digits is done automatically.
Default Call Priority `dflt-call-prio` [SIPDefaultCallPriority]	Determines the default call priority for MLPP calls. ▪ [0] 0 = (Default) ROUTINE ▪ [2] 2 = PRIORITY ▪ [4] 4 = IMMEDIATE ▪ [6] 6 = FLASH ▪ [8] 8 = FLASH-OVERRIDE ▪ [9] 9 = FLASH-OVERRIDE-OVERRIDE If the incoming SIP INVITE request doesn't contain a valid priority value in the SIP Resource-Priority header, the default value is used in the Precedence IE (after translation to the relevant ISDN Precedence value) of the outgoing Setup message. If the incoming Setup message doesn't contain a valid Precedence Level value, the default value is used in the Resource-Priority header of the outgoing SIP INVITE request.

Parameter	Description
	In this scenario, the character string is sent without translation to a numerical value.
MLPP DiffServ mlpp-diffserv [MLPPDiffserv]	Defines the DiffServ value (differentiated services code point/DSCP) used in IP packets containing SIP messages that are related to MLPP calls. The parameter defines DiffServ for incoming and outgoing MLPP calls with the Resource-Priority header. The valid range is 0 to 63. The default is 50.
Preemption Tone Duration preemp-tone-dur [PreemptionToneDuration]	Defines the duration (in seconds) in which the device plays a preemption tone to the Tel and IP sides if a call is preempted. The valid range is 0 to 60. The default is 3. **Note:** If set to 0, no preemption tone is played.
MLPP Normalized Service Domain mlpp-norm-ser-dmn [MLPPNormalizedServiceDomain]	Defines the MLPP normalized service domain string. If the device receives an MLPP ISDN incoming call, it uses the parameter (if different from ‘FFFFFF’) as a Service domain in the SIP Resource-Priority header in outgoing INVITE messages. If the parameter is configured to ‘FFFFFF’, the Resource-Priority header is set to the MLPP Service Domain obtained from the Precedence IE. The valid value is 6 hexadecimal digits. The default is ‘000000’. **Note:** The parameter is applicable only to the MLPP NI-2 ISDN variant with CallPriorityMode set to 1.
mlpp-nwrk-id [MLPPNetworkIdentifier]	Defines the MLPP network identifier (i.e., International prefix or Telephone Country Code/TCC) for IP-to-ISDN calls, according to the UCR 2008 and ITU Q.955 specifications. The valid range is 1 to 999. The default is 1 (i.e., USA). The MLPP network identifier is sent in the Facility IE of the ISDN Setup message. For example: ▪ MLPPNetworkIdentifier set to default (i.e., USA, 1): PlaceCall- MLPPNetworkID:0100 MlppServiceDomain:123abc, MlppPrecLevel:5 Fac(1c): 91 a1 15 02 01 05 02 01 19 30 0d 0a 01 05 0a 01 01 04 05 01 00 12 3a bc ▪ MLPPNetworkIdentifier set to 490: PlaceCall- MLPPNetworkID:9004 MlppServiceDomain:123abc, MlppPrecLevel:5 Fac(1c): 91 a1 15 02 01 0a 02 01 19 30 0d 0a 01 05 0a 01 01 04 05 90 04 12 3a bc
MLPP Default Service Domain mlpp-dflt-srv-domain [MLPPDefaultServiceDomain]	Defines the MLPP default service domain string. If the device receives a non-MLPP ISDN incoming call (without a Precedence IE), it uses the parameter (if different than “FFFFFF”) as a Service domain in the SIP Resource-Priority header in outgoing (Tel-to-IP calls) INVITE messages. The parameter is used in conjunction with the parameter SIPDefaultCallPriority. If MLPPDefaultServiceDomain is set to 'FFFFFF', the device interworks the non-MLPP ISDN call to non-MLPP SIP call, and the outgoing INVITE does not contain the Resource-Priority header.

<table>
<tr><th>Parameter</th><th>Description</th></tr>
<tr><td></td><td>The valid value is a 6 hexadecimal digits. The default is "000000".
<b>Note:</b> The parameter is applicable only to the MLPP NI-2 ISDN variant with CallPriorityMode set to 1.</td></tr>
<tr><td>resource-prio-req
[RPRequired]</td><td>Determines whether the SIP resource-priority tag is added in the SIP Require header of the INVITE message for Tel-to-IP calls.
▪ <b>[0]</b> Disable = Excludes the SIP resource-priority tag from the SIP Require header.
▪ <b>[1]</b> Enable = (Default) Adds the SIP resource-priority tag in the SIP Require header.
<b>Note:</b> The parameter is applicable only to MLPP priority call handling (i.e., only when the CallPriorityMode parameter is set to 1).</td></tr>
<tr><td colspan="2"><b>Multiple Differentiated Services Code Points (DSCP) per MLPP Call Priority Level (Precedence) Parameters</b>
The MLPP service allows placement of priority calls, where properly validated users can preempt (terminate) lower-priority phone calls with higher-priority calls. For each MLPP call priority level, the DSCP can be set to a value from 0 to 63. The Resource Priority value in the Resource-Priority SIP header can be one of the following:
<table>
<tr><th>MLPP Precedence Level</th><th>Precedence Level in Resource-Priority SIP Header</th></tr>
<tr><td>0 (lowest)</td><td>routine</td></tr>
<tr><td>2</td><td>priority</td></tr>
<tr><td>4</td><td>immediate</td></tr>
<tr><td>6</td><td>flash</td></tr>
<tr><td>8</td><td>flash-override</td></tr>
<tr><td>9 (highest)</td><td>flash-override-override</td></tr>
</table></td></tr>
<tr><td>RTP DSCP for MLPP Routine
dscp-4-mlpp-rtn
[MLPPRoutineRTPDSCP]</td><td>Defines the RTP DSCP for MLPP Routine precedence call level.
The valid range is -1 to 63. The default is -1.
<b>Note:</b> If set to -1, the DiffServ value is taken from the global parameter PremiumServiceClassMediaDiffServ or as defined in IP Profiles per call.</td></tr>
<tr><td>RTP DSCP for MLPP Priority
dscp-4-mlpp-prio
[MLPPPriorityRTPDSCP]</td><td>Defines the RTP DSCP for MLPP Priority precedence call level.
The valid range is -1 to 63. The default is -1.
<b>Note:</b> If set to -1, the DiffServ value is taken from the global parameter PremiumServiceClassMediaDiffServ or as defined in IP Profiles per call.</td></tr>
<tr><td>RTP DSCP for MLPP Immediate
dscp-4-mlpp-immed
[MLPPImmediateRTPDSCP]</td><td>Defines the RTP DSCP for MLPP Immediate precedence call level.
The valid range is -1 to 63. The default is -1.
<b>Note:</b> If set to -1, the DiffServ value is taken from the global parameter PremiumServiceClassMediaDiffServ or as defined in IP Profiles per call.</td></tr>
<tr><td>RTP DSCP for MLPP Flash
dscp-4-mlpp-flsh</td><td>Defines the RTP DSCP for MLPP Flash precedence call level.
The valid range is -1 to 63. The default is -1.</td></tr>
</table>

Parameter	Description
[MLPPFlashRTPDSCP]	**Note:** If set to -1, the DiffServ value is taken from the global parameter PremiumServiceClassMediaDiffServ or as defined in IP Profiles per call.
RTP DSCP for MLPP Flash Override `dscp-4-mlpp-flsh-ov` [MLPPFlashOverRTPDSCP]	Defines the RTP DSCP for MLPP Flash-Override precedence call level. The valid range is -1 to 63. The default is -1. **Note:** If set to -1, the DiffServ value is taken from the global parameter PremiumServiceClassMediaDiffServ or as defined in IP Profiles per call.
RTP DSCP for MLPP Flash-Override-Override `dscp-4-mlpp-flsh-ov-ov` [MLPPFlashOverOverRTPDSCP]	Defines the RTP DSCP for MLPP Flash-Override-Override precedence call level. The valid range is -1 to 63. The default is -1. **Note:** If set to -1, the DiffServ value is taken from the global parameter PremiumServiceClassMediaDiffServ or as defined in IP Profiles per call.

57.10.5.9 ISDN BRI Parameters

The automatic dialing upon off-hook parameters are described in the table below.

Table 57-49: Automatic Dialing Parameters

Parameter	Description
BRI-to-SIP Supplementary Services Codes for Call Forward **Note:** Upon receipt of an ISDN Facility message for call forward from the BRI phone, the device sends a SIP INVITE to the softswitch with a user-defined code in the SIP To header, representing the reason for the call forward. For more information on BRI call forwarding, see "BRI Call Forwarding" on page 504.	
Call Forward Unconditional [SuppServCodeCFU]	Defines the prefix code for activating Call Forward Unconditional sent to the softswitch. The valid value is a string. By default, no value is defined. **Note:** The string must be enclosed in single apostrophe (e.g., ‘*72’).
Call Forward Unconditional Deactivation [SuppServCodeCFUDeact]	Defines the prefix code for deactivating Call Forward Unconditional Deactivation sent to the softswitch. The valid value is a string. By default, no value is defined. **Note:** The string must be enclosed in single apostrophe (e.g., ‘*72’).
Call Forward on Busy [SuppServCodeCFB]	Defines the prefix code for activating Call Forward on Busy sent to the softswitch. The valid value is a string. By default, no value is defined. **Note:** The string must be enclosed in single apostrophe (e.g., ‘*72’).
Call Forward on Busy Deactivation [SuppServCodeCFBDeact]	Defines the prefix code for deactivating Call Forward on Busy Deactivation sent to the softswitch. The valid value is a string. By default, no value is defined. **Note:** The string must be enclosed in single apostrophe (e.g., ‘*72’).

Parameter	Description
Call Forward on No Reply [SuppServCodeCFNR]	Defines the prefix code for activating Call Forward on No Reply sent to the softswitch. The valid value is a string. By default, no value is defined. **Note:** The string must be enclosed in single apostrophe (e.g., '*72').
Call Forward on No Reply Deactivation [SuppServCodeCFNRDeact]	Defines the prefix code for deactivating Call Forward on No Reply Deactivation sent to the softswitch. The valid value is a string. By default, no value is defined. **Note:** The string must be enclosed in single apostrophe (e.g., '*72').
use-facility-in-req [UseFacilityInRequest]	Enables the device to indicate the type of call forwarding service in the Request-URI of the outgoing SIP INVITE message, using a proprietary header parameter "facility=<call forward service>". ▪ [0] = (Default) Disable ▪ [1] = Enable
[BRICallForwardHandling]	Enables the device to handle BRI call forwarding. ▪ [0] Disable = (Default) BRI call forwarding is handled by a remote server. The device interworks Facility message from the BRI endpoint to SIP messages sent to the server. For more information, see Remote Handling of BRI Call Forwarding on page 504. ▪ [1] Enable = BRI call forwarding is handled by the device. For more information, see Local Handling of BRI Call Forwarding on page 505.

57.10.5.10 Character Conversion Parameters

The Character Conversion table parameter is described in the table below.

Table 57-50: Char Conversion Table Parameters

Char Conversion Table	
Char Conversion `configure voip > gw dtmf-and-suppl dtmf-and-dialing char-conversion display` [CharConversion]	Defines Unicode-to-ASCII character conversion rules. The format of the ini file table parameter is as follows: [CharConversion] FORMAT CharConversion_Index = CharConversion_CharName, CharConversion_FirstByte, CharConversion_SecondByte, CharConversion_ConvertedOutput; [\CharConversion] For a detailed description of the table, see "Converting Accented Characters from IP to Tel" on page 522.

57.10.6 PSTN Parameters

This subsection describes the device's PSTN parameters.

57.10.6.1 General Parameters

The general PSTN parameters are described in the table below.

Table 57-51: General PSTN Parameters

Parameter	Description
Trunk Name `name (config-voip > interface <e1|t1|bri> name` [DigitalPortInfo_x]	Defines an arbitrary name for a trunk (where *x* denotes the trunk number for the ini file parameter). This can be used to help you easily identify the trunk. The valid value is a string of up to 40 characters. The following special characters can be used (without the quotes): ▪ " " (space) ▪ "." (period) ▪ "=" (equal sign) ▪ "-" (hyphen) ▪ "_" (underscore) ▪ "#" (pound sign) By default, the value is undefined.
Protocol Type `protocol` [ProtocolType]	Defines the PSTN protocol for all the Trunks. To configure the protocol type for a specific Trunk, use the *ini* file parameter ProtocolType_x: ▪ **[0]** NONE ▪ [50] BRI EURO ISDN = Euro ISDN over BRI ▪ [51] BRI NI2 ISDN ▪ [52] BRI DMS 100 ISDN ▪ [53] BRI 5ESS 10 ISDN ▪ [54] BRI QSIG = QSIG over BRI ▪ [55] BRI VN6 = VN6 over BRI ▪ [56] BRI NTT = BRI ISDN Japan (Nippon Telegraph) **Note:** The ISDN BRI North American variants (NI-2, DMS-100, and 5ESS) are partially supported by the device. Please contact your AudioCodes sales representative before implementing this protocol.
[ProtocolType_x]	Defines the protocol type for a specific trunk ID (where x denotes the Trunk ID and 0 is the first trunk). For more information, see the ProtocolType parameter.
[ISDNTimerT310]	Defines the T310 override timer for DMS, Euro ISDN, and ISDN NI-2 variants. An ISDN timer is started when a Q.931 Call Proceeding message is received. The timer is stopped when a Q.931 Alerting, Connect, or Disconnect message is received from the other end. If no ISDN Alerting, Progress, or Connect message is received within the duration of T310 timer, the call clears. The valid value range is 0 to 600 seconds. The default is 0 (i.e., use the default timer value according to the protocol's specifications). **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ When both the parameters ISDNDmsTimerT310 and ISDNTimerT310 are configured, the value of the parameter ISDNTimerT310 prevails.
[ISDNTimerT301]	Defines the override T301 timer (in seconds). The T301 timer is started when a Q.931 Alert message is received. The timer is stopped when a Q.931 Connect/Disconnect message is received from the other side. If no Connect or Disconnect message is received within the duration of T301, the call is cleared.

Parameter	Description
	The valid range is 0 to 2400. The default is 0 (i.e., the default T301 timer value - 180 seconds - is used). If set to any value other than 0, it overrides the timer with this value. **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ The parameter is applicable only to the QSIG variant.
Trace Level [TraceLevel]	Defines the trace level: ▪ **[0]** No Trace (default) ▪ **[1]** Full ISDN Trace ▪ **[2]** Layer 3 ISDN Trace ▪ **[3]** Only ISDN Q.931 Messages Trace ▪ **[4]** Layer 3 ISDN No Duplication Trace
[TrunkLifeLineType]	Defines the scenarios upon which the device activates PSTN Fallback for digital interfaces. When PSTN Fallback is triggered, the device automatically routes incoming Tel calls to the PSTN (instead of to the IP). ▪ [0] = (Default) PSTN Fallback is activated upon the loss of power to the device, for example, due to a power outage or the unplugging of the device's power cable. PSTN Fallback is provided by two ports, where one port is connected to the PBX, for example, and the other to the PSTN. When PSTN Fallback is triggered, for example, due to a power outage, the device automatically connects the two ports using a metallic relay switch. In such a scenario, calls originating from the PBX are routed directly to the PSTN. For more information on PSTN Fallback cabling, refer to the *Hardware Installation Manual*. **Note:** ▪ For the parameter to take effect, a device reset is required. ▪ The parameter is applicable only to the Gateway application and digital interfaces. ▪ PSTN Fallback is supported only on specific hardware configurations and where dual digital ports are provided. For more information, refer to the *Hardware Installation Manual*. ▪ The PSTN Fallback feature has no relation to the PSTN Fallback License Key.
[AdminState]	Defines the administrative state for all trunks. ▪ **[0]** = Lock the trunk; stops trunk traffic to configure the trunk protocol type. ▪ **[1]** = Shutting down (read only). ▪ **[2]** = (Default) Unlock the trunk; enables trunk traffic. **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ When the device is locked from the Web interface, the parameter changes to 0. ▪ To define the administrative state per trunk, use the TrunkAdministrativeState parameter.
[TrunkAdministrativeState_x]	Defines the administrative state per trunk, where *x* denotes the trunk number. ▪ **[0]** = Lock the trunk; stops trunk traffic to configure the trunk protocol type.

Parameter	Description
	▪ **[1]** = shutting down (read only). ▪ **[2]** = (Default) Unlock the trunk; enables trunk traffic.
[TDMHairPinning]	Defines static TDM hair-pinning (cross-connection) performed at initialization. The connection is between trunks with an option to exclude a single B-channel in each trunk. Format example: T0-T1/B3,T2-T3,T4-T5/B2. **Note:** For the parameter to take effect, a device reset is required.
`iso8859-charset` [ISO8859CharacterSet]	Defines the ISO 8859 character set type (languages) for representing the alphanumeric string of the calling name (caller ID) in the forwarded message, for IP-to-Tel and Tel-to-IP calls. ▪ **[0]** No Accented = Proprietary method where incoming INVITE messages with any accented characters (e.g., ף ,ם ,י ,ב, and □), which are represented in a 2-byte unicode character, are translated to Latin-only, which are normal one-byte ASCII characters (a, e, i, o, and u, respectively). ▪ **[1]** Western European (Default) ▪ **[2]** Central European ▪ **[3]** South European ▪ **[4]** North European ▪ **[5]** Cyrillic ▪ **[6]** Arabic ▪ **[7]** Hebrew ▪ **[8]** Turkish

57.10.6.2 TDM Bus and Clock Timing Parameters

The TDM Bus parameters are described in the table below.

Table 57-52: TDM Bus and Clock Timing Parameters

Parameter	Description
TDM Bus Parameters	
PCM Law Select [PCMLawSelect]	Determines the type of pulse-code modulation (PCM) companding algorithm law in input and output TDM bus. ▪ **[1]** Alaw ▪ **[3]** MuLaw The default value is automatically selected according to the Protocol Type of the selected trunk. **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ Typically, A-Law is used for most BRI variants.
Idle PCM Pattern `idle-pcm-pattern` [IdlePCMPattern]	Defines the PCM Pattern that is applied to the timeslot (B-channel) when the channel is idle. The range is 0 to 255. The default is set internally according to the Law select 1 (0xFF for Mu-Law; 0x55 for A-law).

Parameter	Description
	Note: For the parameter to take effect, a device reset is required.
TDM Bus Clock Source [TDMBusClockSource]	Defines the clock source to which the device synchronizes. ▪ **[1]** Internal = (Default) Generate clock from local source. ▪ **[4]** Network = Recover clock from PSTN line. **Note:** ▪ For the parameter to take effect, a device reset is required.
TDM Bus Local Reference [TDMBusLocalReference]	Defines the physical Trunk ID from which the device recovers (receives) its clock synchronization. The range is 0 to the maximum number of Trunks. The default is 0. **Note:** The parameter is applicable only if the parameter TDMBusClockSource is set to 4 and the parameter TDMBusPSTNAutoClockEnable is set to 0.
TDM Bus Enable Fallback [TDMBusEnableFallback]	Defines the automatic fallback of the clock. ▪ **[0]** Manual (default) ▪ **[1]** Auto Non-Revertive ▪ **[2]** Auto Revertive
TDM Bus Fallback Clock Source [TDMBusFallbackClock]	Determines the fallback clock source on which the device synchronizes in the event of a clock failure. ▪ **[4]** Network (default) ▪ **[8]** H.110_A ▪ **[9]** H.110_B ▪ **[10]** NetReference1 ▪ **[11]** NetReference2
TDM Bus Net Reference Speed [TDMBusNetrefSpeed]	Defines the NetRef frequency (for both generation and synchronization). ▪ **[0]** 8 kHz (default) ▪ **[1]** 1.544 MHz ▪ **[2]** 2.048 MHz
TDM Bus PSTN Auto FallBack Clock [TDMBusPSTNAutoClockEnable]	Enables the PSTN trunk Auto-Fallback Clock feature. ▪ **[0]** Disable = (Default) Recovers the clock from the trunk line defined by the parameter TDMBusLocalReference. ▪ **[1]** Enable = Recovers the clock from any connected synchronized slave trunk line. If this trunk loses its synchronization, the device attempts to recover the clock from the next trunk. Note that initially, the device attempts to recover the clock from the trunk defined by the parameter TDMBusLocalReference. **Notes:** ▪ For the parameter to take effect, a device reset is required.

Parameter	Description
	▪ The parameter is applicable only if the TDMBusClockSource parameter is set to 4.
TDM Bus PSTN Auto Clock Reverting [TDMBusPSTNAutoClockRevertingEnable]	Enables the PSTN trunk Auto-Fallback Reverting feature. If enabled and a trunk returning to service has an AutoClockTrunkPriority parameter value that is higher than the priority of the local reference trunk (set in the TDMBusLocalReference parameter), the local reference reverts to the trunk with the higher priority that has returned to service for the device's clock source. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ The parameter is applicable only when the TDMBusPSTNAutoClockEnable parameter is set to 1.
Auto Clock Trunk Priority `clock-priority` [AutoClockTrunkPriority]	Defines the trunk priority for auto-clock fallback (per trunk parameter). The valid range is 0 to 100, where 0 (default) is the highest priority and 100 indicates that the device does not perform a fallback to the trunk (typically, used to mark untrusted source of clock). **Note:** Fallback is enabled when the TDMBusPSTNAutoClockEnable parameter is set to 1.

57.10.6.3 ISDN Parameters

The ISDN parameters are described in the table below.

Table 57-53: ISDN Parameters

Parameter	Description
ISDN Termination Side `isdn-termination-side` [TerminationSide]	Determines the ISDN termination side. ▪ [0] User side = (Default) ISDN User Termination Equipment (TE) side. ▪ [1] Network side = ISDN Network Termination (NT) side. **Note:** Select 'User side' when the PSTN or PBX side is configured as 'Network side' and vice versa. If you don't

Parameter	Description
	know the device's ISDN termination side, choose 'User side'. If the D-channel alarm is indicated, choose 'Network Side'. The BRI module supports the ITU-T I.430 standard, which defines the ISDN-BRI layer 1 specification. When an NT port is active, it drives a 38-V line and sends an INFO1 signal (as defined in ITU-T I.430 Table 4) on the data line to synchronize to a TE port that might be connected to it. To stop the voltage and the INFO1 signal on the line, stop the trunk using the Stop Trunk button.
[TerminationSide_x]	Same as the description for parameter TerminationSide, but for a specific trunk ID (where *x* denotes the Trunk ID and 0 is the first Trunk).
Enable ignoring ISDN Disconnect with PI `ign-isdn-disc-w-pi` [KeepISDNCallOnDisconnectWithPI]	Enables the device to ignore ISDN Disconnect messages with PI 1 or 8. ▪ **[1]** = The call (in connected state) is not released if a Q.931 Disconnect with PI (PI = 1 or 8) message is received during the call. ▪ **[0]** = (Default) The call is disconnected.
PI For Setup Message `pi-4-setup-msg` [PIForSetupMsg]	Determines whether and which Progress Indicator (PI) information element (IE) is added to the sent ISDN Setup message. Some ISDN protocols such as NI-2 or Euro ISDN can optionally contain PI = 1 or PI = 3 in the Setup message. ▪ **[0]** = PI is not added (default). ▪ **[1]** = PI 1 is added to a sent ISDN Setup message - call is not end-to-end ISDN. ▪ **[3]** = PI 3 is added to a sent ISDN Setup message - calling equipment is not ISDN.
B-channel Negotiation `b-ch-negotiation` [BchannelNegotiation]	Defines the ISDN B-channel negotiation mode. ▪ **[0]** Preferred ▪ **[1]** Exclusive (default) ▪ **[2]** Any **Notes:** ▪ For some ISDN variants, when 'Any' (2) is selected, the Setup message excludes the Channel Identification IE. ▪ The Any' (2) option is applicable only if the following conditions are met: ✓ The parameter TerminationSide is set to 0 ('User side'). ✓ The PSTN protocol type (ProtocolType) is configured as Euro ISDN.
ISDN Flexible Behavior Parameters ISDN protocol is implemented in different switches/PBXs by different vendors. Several implementations may vary slightly from the specification. Therefore, to provide a flexible interface that supports these ISDN variants, the ISDN behavior parameters can be used.	
Incoming Calls Behavior `isdn-bits-incoming-calls-behavior` [ISDNInCallsBehavior]	Determines the bit-field used to determine several behavior options that influence how the ISDN Stack INCOMING calls behave. ▪ **[32]** DATA CONN RS = The device automatically sends a Q.931 Connect (answer) message on incoming Tel calls (Q.931 Setup).

Parameter	Description
	▪ **[64]** VOICE CONN RS = The device sends a Connect (answer) message on incoming Tel calls. ▪ **[2048]** CHAN ID IN FIRST RS = The device sends Channel ID in the first response to an incoming Q.931 Call Setup message. Otherwise, the Channel ID is sent only if the device requires changing the proposed Channel ID. ▪ **[4096]** USER SETUP ACK = (Default) The Setup Ack message is sent by the SIP Gateway application layer and not automatically by the PSTN stack. ▪ **[8192]** CHAN ID IN CALL PROC = The device sends Channel ID in a Q.931 Call Proceeding message. ▪ **[65536]** PROGR IND IN SETUP ACK = (Default) The device includes Progress Indicator (PI=8) in Setup Ack message if an empty called number is received in an incoming Setup message. This option is applicable to the overlap dialing mode. The device also plays a dial tone (for TimeForDialTone) until the next called number digits are received. ▪ **[2147483648]** USER SCREEN INDICATOR = When the device receives two Calling Number IE's in the Setup message, the device, by default, uses only one of the numbers according to the following: ✓ Network provided, Network provided - the first calling number is used ✓ Network provided, User provided: the first one is used ✓ User provided, Network provided: the second one is used ✓ User provided, user provided: the first one is used When this bit is configured, the device behaves as follows: ✓ Network provided, Network provided: the first calling number is used ✓ Network provided, User provided: the second one is used ✓ User provided, Network provided: the first one is used ✓ User provided, user provided: the first one is used **Note:** ▪ In the Web interface, the parameter displays the summation of the enabled optional bit values, in hex format. For example, the default value is 0x11000 (69632 in decimal), which is the summation of the two bit options, USER SETUP ACK (0x01000 or 4096 in decimal) and PROGR IND IN SETUP ACK (0x10000 or 65536 in decimal) that are enabled by default (i.e., 4096 + 65536 = 69632). ▪ When using the *ini* file to configure the device to support several ISDNInCallsBehavior features, enter a summation of the individual feature values. For example, to support both [2048] and [65536] features, set ISDNInCallsBehavior = 67584 (i.e., 2048 + 65536).
[ISDNInCallsBehavior_x]	Same as the description for the parameter ISDNInCallsBehavior, but per trunk (i.e., where *x* denotes the Trunk ID).

Parameter	Description
Q.931 Layer Response Behavior `isdn-bits-ns-behavior` [ISDNIBehavior]	Bit-field used to determine several behavior options that influence the behaviour of the Q.931 protocol. ▪ **[0]** = Disable (default). ▪ **[1]** NO STATUS ON UNKNOWN IE = Q.931 Status message isn't sent if Q.931 received message contains an unknown/unrecognized IE. By default, the Status message is sent. Note: This value is applicable only to ISDN variants in which sending of Status message is optional. ▪ **[2]** NO STATUS ON INV OP IE = Q.931 Status message isn't sent if an optional IE with invalid content is received. By default, the Status message is sent. **Note:** This option is applicable only to ISDN variants in which sending of Status message is optional. ▪ **[4]** ACCEPT UNKNOWN FAC IE = Accepts unknown/unrecognized Facility IE. Otherwise, the Q.931 message that contains the unknown Facility IE is rejected (default). **Note:** This option is applicable only to ISDN variants where a complete ASN1 decoding is performed on Facility IE. ▪ **[128]** SEND USER CONNECT ACK = The Connect ACK message is sent in response to received Q.931 Connect; otherwise, the Connect ACK is not sent. **Note:** This option is applicable only to Euro ISDN User side outgoing calls. ▪ **[2048]** ALWAYS EXPLICIT = Always set the Channel Identification IE to explicit Interface ID, even if the B-channel is on the same trunk as the D-channel. **Note:** This value is applicable only to 4/5ESS, DMS and NI-2 variants. ▪ **[32768]** ACCEPT MU LAW =Mu-Law is also accepted in ETSI. ▪ **[65536]** EXPLICIT PRES SCREENING = The calling party number (octet 3a) is always present even when presentation and screening are at their default. **Note:** This option is applicable only to ETSI, NI-2, and 5ESS. ▪ **[131072]** STATUS INCOMPATIBLE STATE = Clears the call on receipt of Q.931 Status with incompatible state. Otherwise, no action is taken (default). ▪ **[262144]** STATUS ERROR CAUSE = Clear call on receipt of Status according to cause value. ▪ **[524288]** ACCEPT A LAW =A-Law is also accepted in 5ESS. ▪ **[2097152]** RESTART INDICATION = Upon receipt of a Restart message, acEV_PSTN_RESTART_CONFIRM is generated. ▪ **[4194304]** FORCED RESTART = On data link (re)initialization, send RESTART if there is no call. ▪ **[67108864]** NS ACCEPT ANY CAUSE = Accept any Q.850 Cause IE from ISDN. **Note:** This option is applicable only to Euro ISDN.

Parameter	Description
	▪ [134217728] NS_BRI_DL_ALWAYS_UP (0x08000000) = By default, the BRI D-channel goes down if there are no active calls. If this option is configured, the BRI D-channel is always up and synchronized. ▪ **[536870912]** = Alcatel coding for redirect number and display name is accepted by the device. Note: This option is applicable only to QSIG (and relevant for specific Alcatel PBXs such as OXE). ▪ **[1073741824]** QSI ENCODE INTEGER = If this bit is set, INTEGER ASN.1 type is used in operator coding (compliant to new ECMA standards); otherwise, OBJECT IDENTIFIER ASN.1 type is used. **Note:** This option is applicable only to QSIG. ▪ **[2147483648]** 5ESS National Mode For Bch Maintenance = Use the National mode of AT&T 5ESS for B-channel maintenance. **Notes:** ▪ To configure the device to support several ISDNIBehavior features, enter a summation of the individual feature values. For example, to support both [512] and [2048] features, set the parameter ISDNIBehavior is set to 2560 (i.e., 512 + 2048). ▪ When configuring through the Web interface, to select the options click the arrow button and then for each required option select 1 to enable. ▪ For BRI terminal endpoint identifier (TEI) configuration, instead of using the ISDNIBehavior parameter, use the following parameters: BriTEIConfigP2P_x, BriTEIConfigP2MP_x, BriTEIAssignTrigger_x, and BriTEIRemoveTrigger_x.
[ISDNIBehavior_x]	Same as the description for parameter ISDNIBehavior, but for a specific trunk ID.
General Call Control Behavior `isdn-bits-cc-behavior` [ISDNGeneralCCBehavior]	Bit-field for determining several general CC behavior options. To select the options, click the arrow button, and then for each required option, select 1 to enable. The default is 0 (i.e., disable). ▪ **[2]** = Data calls with interworking indication use 64 kbps B-channels (physical only). ▪ **[8]** REVERSE CHAN ALLOC ALGO = Channel ID allocation algorithm. ▪ **[16]** = The device clears down the call if it receives a NOTIFY message specifying 'User-Suspended'. A NOTIFY (User-Suspended) message is used by some networks (e.g., in Italy or Denmark) to indicate that the remote user has cleared the call, especially in the case of a long distance voice call. ▪ **[256]** START WITH B CHAN OOS = B-channels start in the Out-Of-Service state (OOS). ▪ **[512]** CHAN ALLOC LOWEST = CC allocates B-channels starting from the lowest available B-channel id. ▪ **[1024]** CHAN ALLOC HIGHEST = CC allocates B-channels starting from the highest available B-channel id. ▪ **[16384]** CC_TRANSPARENT_UUI bit: The UUI-protocol implementation of CC is disabled allowing the application

Parameter	Description
	to freely send UUI elements in any primitive, regardless of the UUI-protocol requirements (UUI Implicit Service 1). This allows more flexible application control on the UUI. When this bit is not set (default behavior), CC implements the UUI-protocol as specified in the ETS 300-403 standards for Implicit Service 1. ▪ **[65536]** GTD5 TBCT = CC implements the VERIZON-GTD-5 Switch variant of the TBCT Supplementary Service, as specified in FSD 01-02-40AG Feature Specification Document from Verizon. Otherwise, TBCT is implemented as specified in GR-2865-CORE specification (default behavior). **Note:** When using the *ini* file to configure the device to support several ISDNGeneralCCBehavior features, add the individual feature values. For example, to support both [16] and [32] features, set ISDNGeneralCCBehavior = 48 (i.e., 16 + 32).
Outgoing Calls Behavior `isdn-bits-outgoing-calls-behavior` [ISDNOutCallsBehavior]	Determines several behaviour options (bit fields) that influence the behaviour of the ISDN Stack outgoing calls. To select options, click the arrow button, and then for each required option, select 1 to enable. The default is 0 (i.e., disable). ▪ **[2]** USER SENDING COMPLETE =The default behavior of the device (when this bit is not set) is to automatically generate the Sending-Complete IE in the Setup message. This behavior is used when overlap dialing is not needed. When overlap dialing is needed, set this bit and the behavior is changed to suit the scenario, i.e., Sending-Complete IE is added when required in the Setup message for Enblock mode or in the last Digit with Overlap mode. ▪ **[16]** USE MU LAW = The device sends G.711-m-Law in outgoing voice calls. When disabled, the device sends G.711-A-Law in outgoing voice calls. **Note:** This option is applicable only to the Korean variant. ▪ **[128]** DIAL WITH KEYPAD = The device uses the Keypad IE to store the called number digits instead of the CALLED_NB IE. **Note:** This option is applicable only to the Korean variant (Korean network). This is useful for Korean switches that don't accept the CALLED_NB IE. ▪ **[256]** STORE CHAN ID IN SETUP = The device forces the sending of a Channel-Id IE in an outgoing Setup message even if it's not required by the standard (i.e., optional) and no Channel-Id has been specified in the establishment request. This is useful for improving required compatibility with switches. On BRI lines, the Channel-Id IE indicates ‘any channel’. ▪ **[2048]** = The device accepts any IA5 character in the called_nb and calling_nb strings and sends any IA5 character in the called_nb, and is not restricted to extended digits only (i.e., 0-9,*,#). ▪ **[16384]** DLCI REVERSED OPTION = Behavior bit used in the IUA interface groups to indicate that the reversed format of the DLCI field must be used.

Parameter	Description
	Note: When using the *ini* file to configure the device to support several ISDNOutCallsBehavior features, add the individual feature values. For example, to support both [2] and [16] features, set ISDNOutCallsBehavior = 18 (i.e., 2 + 16).
[ISDNOutCallsBehavior_x]	Same as the description for parameter ISDNOutCallsBehavior, but for a specific trunk ID.
ISDN NS Behaviour 2 `isdn-bits-ns-extension-behavior` [ISDNNSBehaviour2]	Bit-field to determine several behavior options that influence the behavior of the Q.931 protocol. ▪ **[8]** NS BEHAVIOUR2 ANY UUI = Any User to User Information Element (UUIE) is accepted for any protocol discriminator. This is useful for interoperability with non-standard switches. ▪ **[16]** NS BEHAVIOUR2 DISPLAY = The Display IE is accepted even if it is not defined in the QSIG ISDN protocol standard. This is applicable only when configuration is QSI. ▪ **[64]** NS BEHAVIOUR2 FAC REJECT = When this bit is set, the device answers with a Facility IE message with the Reject component on receipt of Facility IE with unknown/invalid Invoke component. This bit is implemented in QSIG and ETSI variants.
[PSTNExtendedParams]	Determines the bit map for special PSTN behavior parameters: ▪ **[0]** = (Default) Applicable for NI-2 ISDN and QSIG "Networking Extensions". This bit (i.e., bit #0) is responsible for the Invoke ID size: ✓ If this bit is not set (default), then the Invoke ID size is always one byte, with a value of 01 to 7f. ✓ If this bit is set, then the Invoke ID size is one or two bytes according to the Invoke ID value. ▪ **[2]** = Applicable to the ROSE format (according to the old QSIG specifications). This bit (i.e., bit #1) is responsible for the QSIG octet 3. According to the ECMA-165 new version, octet 3 in all QSIG supplementary services Facility messages should be 0x9F = Networking Extensions. However, according to the old version, the value should be 0x91 = ROSE: ✓ If this bit is not set (default): 0x9F = Networking Extensions. ✓ If this bit is set: 0x91 = ROSE. ▪ **[3]** = Use options [0] and [2] above. **Note:** For the parameter to take effect, a device reset is required.
BRI Parameters	
BRI Layer 2 Mode `isdn-layer2-mode` [BriLayer2Mode]	Defines Point-to-Point (P2P) or Point-to-Multipoint (P2MP) mode for BRI ports. ▪ [0] Point to Point (default) ▪ [1] Point to Multipoint = Must be configured for Network side.

Parameter	Description
`tei-config-p2p (config-voice > interface bri <module/port>)` [BriTEIConfigP2P_x]	Defines the BRI terminal endpoint identifier (TEI) when in point-to-point (P2P) mode. The valid value is 0 to 63, 127. The default is 0. ▪ Network Side: ✓ 0-63: Static TEI is accepted. ✓ 127: Any possible TEI is accepted. Dynamic TEI allocation is supported. ▪ User Side: ✓ 0-63: Static TEI is used. ✓ 127: Dynamic TEI allocation is supported (TEI request procedure initiated). **Note:** The value 127 replaces the previous configuration requirement to set the ISDNIBehavior parameter to NS EXPLICIT INTERFACE ID (1).
`tei-config-p2mp (config-voice > interface bri <module/port>)` [BriTEIConfigP2MP_x]	Defines the BRI TEI when in point-to-multipoint (P2MP) mode. The valid value is 0 to 63, 127. The default is 127. ▪ Network Side: Not applicable - In network side in P2MP configuration, any TEI must be accepted. ▪ User Side: ✓ 0-63: Static TEI is used. ✓ 127: Dynamic TEI allocation is supported (TEI request procedure initiated).
`tei-assign-trigger (config-voice > interface bri <module/port>)` [BriTEIAssignTrigger_x]	Defines when to start the TEI assignment procedure. The valid values are (bit-field parameter): ▪ Bit #0: LAYER1_ACTIVATION ▪ Bit #1: BRI_PORT_CONFIG ▪ Bit #2: CALL_ESTABLISH The default is 0x04 (Bit #2). **Note:** The parameter is applicable only to the User side (for Dynamic TEI).
`tei-remove-trigger (config-voice > interface bri <module/port>)` [BriTEIRemoveTrigger_x]	Defines the following: ▪ Network Side: When to "forget" all existing TEIs and wait for the User side to start a new TEI assignment procedure. This is also applicable to static TEI. ▪ User Side: When to start a new TEI assignment verification procedure. The valid values are (bit-field parameter): ▪ Bit #0: LAYER1_DEACTIVATION ▪ Bit #1: BRI_DL_RELEASED ▪ Bit #2: TEI_0_P2MP_NET_SIDE (Note: this value is used to skip TEI=0 SABMEs when the port is defined as P2MP NET side.) The default is 0x00.

57.10.7 ISDN Interworking Parameters

The ISDN interworking parameters are described in the table below.

Table 57-54: ISDN Interworking Parameters

Parameter	Description
ISDN Parameters	
Send Local Time To ISDN Connect [SendLocalTimeToISDNConnect]	Determines the device's handling of the date and time sent in the ISDN Connect message (Date / Time IE) upon receipt of SIP 200 OK messages. ▪ **[0]** Disable = (Default) If the SIP 200 OK includes the Date header, the device sends its value in the ISDN Connect Date / Time IE. If the 200 OK does not include this header, it does not add the Date / Time IE to the sent ISDN Connect message. ▪ **[1]** Enable = If the SIP 200 OK includes the Date header, the device sends its value (i.e. date and time) in the ISDN Connect Date / Time IE. If the 200 OK does not include this header, the device uses its internal, local date and time for the Date / Time IE, which it adds to the sent ISDN Connect message. ▪ **[2]** Always Send Local Date and Time = The device always sends its local date and time (obtained from its internal clock) to PBXs in ISDN Q.931 Connect messages (Date / Time IE). It does this regardless of whether or not the incoming SIP 200 OK includes the Date header. If the SIP 200 OK includes the Date header, the device ignores its value. **Notes:** ▪ This feature is applicable only to Tel-to-IP calls. ▪ For IP-to-Tel calls, the parameter is not applicable. Only if the incoming ISDN Connect message contains the Date / Time IE does the device add the Date header to the sent SIP 200 OK message.
Min Routing Overlap Digits `min-dg-b4-routing` [MinOverlapDigitsForRouting]	Defines the minimum number of overlap digits to collect (for ISDN overlap dialing) before sending the first SIP message for routing Tel-to-IP calls. The valid value range is 0 to 49. The default is 1. **Note:** The parameter is applicable when the ISDNRxOverlap parameter is set to [2] or [3].
ISDN Overlap IP to Tel Dialing `isdn-tx-overlap` [ISDNTxOverlap]	Enables ISDN overlap dialing for IP-to-Tel calls. This feature is part of ISDN-to-SIP overlap dialing according to RFC 3578. ▪ [1] Through SIP = The device sends the first received digits from the initial INVITE to the Tel side in an ISDN Setup message. For each subsequently received re-INVITE message of the same dialog session, the device sends the collected digits to the Tel side in ISDN Info Q.931 messages. For each received re-INVITE, the device sends a SIP 484 Address Incomplete response to maintain the current dialog session and to receive additional digits from subsequent re-INVITEs. ▪ [2] Through SIP INFO = The device sends the first received digits from the initial INVITE to the Tel side in an ISDN Setup message and then responds to the IP side with a SIP 183. For each subsequently received SIP INFO message with additional digits of the same dialog session, the device sends the collected digits to the Tel side in ISDN Info Q.931 messages. For each received SIP INFO, the device sends a SIP 200 OK response to maintain the current dialog session and to receive additional digits from subsequent INFOs. **Note:** When IP-to-Tel overlap dialing is enabled, to send ISDN Setup messages without the Sending Complete IE, the

Parameter	Description
	ISDNOutCallsBehavior parameter must be set to USER SENDING COMPLETE (2).
Enable Receiving of Overlap Dialing ovrlp-rcving-type [ISDNRxOverlap_x]	Determines the receiving (Rx) type of ISDN overlap dialing for Tel-to-IP calls, per trunk. ▪ **[0]** None = (Default) Disabled. ▪ **[1]** Local receiving = ISDN Overlap Dialing - the complete number is sent in the INVITE Request-URI user part. The device receives ISDN called number that is sent in the 'Overlap' mode. The ISDN Setup message is sent to IP only after the number (including the Sending Complete IE) is fully received (via Setup and/or subsequent Info Q.931 messages). In other words, the device waits until it has received all the ISDN signaling messages containing parts of the called number, and only then it sends a SIP INVITE with the entire called number in the Request-URI. ▪ **[2]** Through SIP = Interworking of ISDN Overlap Dialing to SIP according to RFC 3578. The device sends the first received digits from the ISDN Setup message to the IP side in the initial INVITE message. For each subsequently received ISDN Info Q.931 message, the device sends the collected digits to the IP side in re-INVITE messages. ▪ **[3]** Through SIP INFO =Interworking of ISDN Overlap Dialing to SIP according to RFC 3578. The device sends the first received digits from the ISDN Setup message to the IP side in the initial INVITE message. For each subsequently received ISDN Info Q.931 message, the device sends the collected digits to the IP side in INFO messages. **Notes:** ▪ When option [2] or [3] is configured, you can define the minimum number of overlap digits to collect before sending the first SIP message for routing the call, using the MinOverlapDigitsForRouting parameter. ▪ When option [2] or [3] is configured, even if SIP 4xx responses are received during this ISDN overlap receiving, the device does not release the call. ▪ The MaxDigits parameter can be used to limit the length of the collected number for ISDN overlap dialing (if Sending Complete is not received). ▪ If a digit map pattern is defined (using the DigitMapping or DialPlanIndex parameters), the device collects digits until a match is found (e.g., for closed numbering schemes) or until a timer expires (e.g., for open numbering schemes). If a match is found (or the timer expires), the digit collection process is terminated even if Sending Complete is not received. ▪ For enabling ISDN overlap dialing for IP-to-Tel calls, use the ISDNTxOverlap parameter. ▪ The *x* in the ini file parameter name denotes the trunk number, where 0 is Trunk 1. ▪ For more information on ISDN overlap dialing, see ISDN Overlap Dialing.
ovrlp-rcving-type [ISDNRxOverlap]	Same as the description for parameter ISDNRxOverlap_x, but for all trunks.
Mute DTMF In Overlap	Enables the muting of in-band DTMF detection until the device receives the complete destination number from the ISDN (for Tel-to-

Parameter	Description
[MuteDTMFInOverlap]	IP calls). In other words, the device does not accept DTMF digits received in the voice stream from the PSTN, but only accepts digits from ISDN Info messages. ▪ **[0]** Don't Mute (default). ▪ **[1]** Mute DTMF in Overlap Dialing = The device ignores in-band DTMF digits received during ISDN overlap dialing (disables the DTMF in-band detector). **Note:** The parameter is applicable to ISDN Overlap mode only when dialed numbers are sent using Q.931 Information messages.
[ConnectedNumberType]	Defines the Numbering Type of the ISDN Q.931 Connected Number IE that the device sends in the Connect message to the ISDN (for Tel-to-IP calls). This is interworked from the P-Asserted-Identity header in SIP 200 OK. The default is [0] (i.e., unknown).
[ConnectedNumberPlan]	Defines the Numbering Plan of the ISDN Q.931 Connected Number IE that the device sends in the Connect message to the ISDN (for Tel-to-IP calls). This is interworked from the P-Asserted-Identity header in SIP 200 OK. The default is [0] (i.e., unknown).
Enable ISDN Tunneling Tel to IP `isdn-tnl-tel2ip` [EnableISDNTunnelingTel2IP]	Enables ISDN Tunneling. ▪ **[0]** Disable (default). ▪ **[1]** Using Header = Enable ISDN Tunneling from ISDN to SIP using a proprietary SIP header. ▪ **[2]** Using Body = Enable ISDN Tunneling from ISDN to SIP using a dedicated message body. When ISDN Tunneling is enabled, the device sends all ISDN messages using the correlated SIP messages. The ISDN Setup message is tunneled using SIP INVITE, all mid-call messages are tunneled using SIP INFO, and ISDN Disconnect/Release message is tunneled using SIP BYE messages. The raw data from the ISDN is inserted into a proprietary SIP header (X-ISDNTunnelingInfo) or a dedicated message body (application/isdn) in the SIP messages. **Notes:** ▪ For this feature to function, you must set the parameter ISDNDuplicateQ931BuffMode to 128 (i.e., duplicate all messages). ▪ ISDN tunneling is applicable for all ISDN variants as well as QSIG.
Enable ISDN Tunneling IP to Tel `isdn-tnl-ip2tel` [EnableISDNTunnelingIP2Tel]	Enables ISDN Tunneling for IP-to-Tel calls. ▪ **[0]** Disable (default) ▪ **[1]** Enable ISDN Tunneling from IP to ISDN When ISDN Tunneling is enabled, the device extracts raw data received in the proprietary SIP header, x-isdntunnelinginfo, or a dedicated message body (application/isdn) in the SIP message and then sends the data in an ISDN message to the PSTN. If the raw data in this SIP header is suffixed with the string "ADDE", then the raw data is extracted and added as Informational Elements (IE) in the outgoing Q.931 message. The tunneling of the x-isdntunnelinginfo SIP header with IEs is converted from INVITE, 180, and 200 OK SIP messages to Q.931 SETUP, ALERT, and CONNECT respectively. For example, if the following SIP header is received,

Parameter	Description
	`x-isdntunnelinginfo: ADDE1C269FAA 06 800100820100A10F020136 0201F0A00702010102021F69` then it is added as an IE to the outgoing Q.931 message as 1C269FAA 06 800100820100A10F020136 0201F0A00702010102021F69, where, for example, "1C269F" is a 26 byte length Facility IE. **Note:** The feature is similar to that of the AddIEinSetup parameter. If both parameters are configured, the AddIEinSetup parameter is ignored.
Enable QSIG Tunneling `qsig-tunneling` [EnableQSIGTunneling]	Global parameter that enables QSIG tunneling-over-SIP for all calls. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_EnableQSIGTunneling). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
[QSIGTunnelingMode]	Defines the format of encapsulated QSIG message data in the SIP message MIME body. ▪ **[0]** = (Default) ASCII presentation of Q.931 QSIG message. ▪ **[1]** = Binary encoding of Q.931 QSIG message (according to ECMA-355, RFC 3204, and RFC 2025). **Note:** The parameter is applicable only if the QSIG Tunneling feature is enabled (using the EnableQSIGTunneling parameter).
Enable Hold to ISDN `hold-to-isdn` [EnableHold2ISDN]	Enables SIP-to-ISDN interworking of the Hold/Retrieve supplementary service. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Notes:** ▪ The parameter is applicable to Euro ISDN variants - from TE (user) to NT (network). ▪ The parameter is applicable to QSIG BRI. ▪ If the parameter is disabled, the device plays a held tone to the Tel side when a SIP request with 0.0.0.0 or "inactive" in SDP is received. An appropriate CPT file with the held tone should be used.
[ISDNDuplicateQ931BuffMode]	Determines the activation/deactivation of delivering raw Q.931 messages. ▪ **[0]** = (Default) ISDN messages aren't duplicated. ▪ **[128]** = All ISDN messages are duplicated. **Note:** For the parameter to take effect, a device reset is required.
ISDN SubAddress Format `isdn-subaddr-frmt` [ISDNSubAddressFormat]	Determines the encoding format of the SIP Tel URI parameter 'isub', which carries the encoding type of ISDN subaddresses. This is used to identify different remote ISDN entities under the same phone number (ISDN Calling and Called numbers) for interworking between ISDN and SIP networks. ▪ **[0]** = (Default) ASCII - IA5 format that allows up to 20 digits. Indicates that the 'isub' parameter value needs to be encoded using ASCII characters.

Parameter	Description
	▪ **[1]** = BCD (Binary Coded Decimal) - allows up to 40 characters (digits and letters). Indicates that the 'isub' parameter value needs to be encoded using BCD when translated to an ISDN message. ▪ **[2]** = User Specified For IP-to-Tel calls, if the incoming SIP INVITE message includes subaddress values in the 'isub' parameter for the Called Number (in the Request-URI) and/or the Calling Number (in the From header), these values are mapped to the outgoing ISDN Setup message. If the incoming ISDN Setup message includes 'subaddress' values for the Called Number and/or the Calling Number, these values are mapped to the outgoing SIP INVITE message's ‘isub’ parameter in accordance with RFC 4715.
[IgnoreISDNSubaddress]	Determines whether the device ignores the Subaddress from the incoming ISDN Called and Calling numbers when sending to IP. ▪ **[0]** = (Default) If an incoming ISDN Q.931 Setup message contains a Called/Calling Number Subaddress, the Subaddress is interworked to the SIP 'isub' parameter according to RFC. ▪ **[1]** = The device removes the ISDN Subaddress and does not include the 'isub' parameter in the Request-URI and does not process INVITEs with the parameter.
[ISUBNumberOfDigits]	Defines the number of digits (from the end) that the device takes from the called number (received from the IP) for the isub number (in the sent ISDN Setup message). This feature is applicable only for IP-to-ISDN calls. The valid value range is 0 to 36. The default is 0. This feature operates as follows: 1 If an isub parameter is received in the Request-URI, for example, INVITE sip:9565645;**isub**=1234@host.domain:user=phone SIP/2.0 then the isub value is sent in the ISDN Setup message as the destination subaddress. 2 If the isub parameter is not received in the user part of the Request-URI, the device searches for it in the URI parameters of the To header, for example, To: "Alex" <sip: 9565645@host.domain;**isub**=1234> If present, the isub value is sent in the ISDN Setup message as the destination subaddress. 3 If the isub parameter is not present in the Request-URI header nor To header, the device does the following: ✓ If the called number (that appears in the user part of the Request-URI) starts with zero (0), for example, INVITE sip:**0**5694564@host.domain:user=phone SIP/2.0 then the device maps this called number to the destination number of the ISDN Setup message, and the destination subaddress in this ISDN Setup message remains empty. ✓ If the called number (that appears in the user part of the Request-URI) does not start with zero, for example, INVITE sip:5694564@host.domain:user=phone SIP/2.0 then the device maps this called number to the destination number of the ISDN Setup message, and the destination subaddress in this ISDN Setup message then contains *y* digits from the end of the called number. The *y* number of digits can be configured using the ISUBNumberOfDigits parameter. The

Parameter	Description
	default value of ISUBNumberOfDigits is 0, thus, if the parameter is not configured, and 1) and 2) scenarios (described above) have not provided an isub value, the subaddress remains empty.
Default Cause Mapping From ISDN to SIP `dflt-cse-map-isdn2sip` [DefaultCauseMapISDN2IP]	Defines a single default ISDN release cause that is used (in ISDN-to-IP calls) instead of all received release causes, except when the following Q.931 cause values are received: Normal Call Clearing (16), User Busy (17), No User Responding (18), or No Answer from User (19). The range is any valid Q.931 release cause (0 to 127). The default is 0 (i.e., not configured - static mapping is used).

`usr2usr-hdr-frmt`
[UserToUserHeaderFormat]
	Defines the interworking between the SIP INVITE's User-to-User header and the ISDN User-to-User (UU) IE data.
	▪ **[0]** = (Default) SIP header format: X-UserToUser.
	▪ **[1]** = SIP header format: User-to-User with Protocol Discriminator (pd) attribute (according to IETF Internet-Draft draft-johnston-sipping-cc-uui-04). For example:
	`User-to-User=30303734353137343136353533b313233343b3834;pd=4`
	▪ **[2]** = SIP header format: User-to-User with encoding=hex at the end and pd embedded as the first byte (according to IETF Internet-Draft draft-johnston-sipping-cc-uui-03). For example:
	`User-to-User=0430303734353137343136353533b313233343b3834;encoding=hex`
	where "04" at the beginning of this message is the pd.
	▪ **[3]** = Interworks the SIP User-to-User header containing text format to ISDN UUIE in hexadecimal format, and vice versa. For example:
	SIP Header in text format:
	`User-to-User=01800213027b712a;NULL;4582166;`
	Translated to hexadecimal in the ISDN UUIE:
	`3031383030323133303237623731326133b4e554c4c3b343538323136363b`
	The Protocol Discriminator (pd) used in UUIE is "04" (IUA characters).
	Note: The parameter is applicable for Tel-to-IP and IP-to-Tel calls.

Remove CLI when Restricted `rmv-cli-when-restr` [RemoveCLIWhenRestricted]	Determines (for IP-to-Tel calls) whether the Calling Number and Calling Name IEs are removed from the ISDN Setup message if the presentation is set to Restricted. ▪ **[0]** No = (Default) IE's are not removed. ▪ **[1]** Yes = IE's are removed.
Remove Calling Name `rmv-calling-name` [RemoveCallingName]	Enables the device to remove the Calling Name from SIP-to-ISDN calls for all trunks. ▪ **[0]** Disable = (Default) Does not remove Calling Name. ▪ **[1]** Enable = Removes Calling Name. **Note:** Some PSTN switches / PBXs may not be configured to support the receipt of the “Calling Name” information. These switches might respond to an ISDN Setup message (including the Calling Name) with an ISDN "REQUESTED_FAC_NOT_SUBSCRIBED" failure. The

Parameter	Description
	parameter can be set to Enable (1) to remove the “Calling Name” from SIP-to-ISDN calls and allow the call to proceed.
Remove Calling Name [RemoveCallingNameForTrunk_x]	Enables the device to remove the Calling Name for SIP-to-ISDN calls, per trunk. ▪ **[-1]** Use Global Parameter = (Default) Settings of the global parameter RemoveCallingName are used. ▪ **[0]** Disable = Does not remove Calling Name. ▪ **[1]** Enable = Remove Calling Name. **Note:** The *x* in the ini file parameter name denotes the trunk number, where 0 is Trunk 1.
Progress Indicator to ISDN `pi-to-isdn` [ProgressIndicator2ISDN_x]	Determines the Progress Indicator (PI) to ISDN per trunk. ▪ **[-1]** Not Configured = (Default) The PI in ISDN messages is set according to the parameter PlayRBTone2Tel. ▪ **[0]** No PI = PI is not sent to ISDN. ▪ **[1]** PI = 1; **[8]** PI = 8: The PI value is sent to PSTN in Q.931/Proceeding and Alerting messages. Typically, the PSTN/PBX cuts through the audio channel without playing local ringback tone, enabling the originating party to hear remote Call Progress Tones or network announcements. **Note:** The *x* in the ini file parameter name denotes the trunk number, where 0 is Trunk 1.
Set PI in Rx Disconnect Message `pi-in-rx-disc-msg` [PIForDisconnectMsg_x]	Defines the device's behavior per trunk when a Disconnect message is received from the ISDN before a Connect message is received. ▪ **[-1]** Not Configured = (Default) Sends a 183 SIP response according to the received progress indicator (PI) in the ISDN Disconnect message. If PI = 1 or 8, the device sends a 183 response, enabling the PSTN to play a voice announcement to the IP side. If there isn't a PI in the Disconnect message, the call is released. ▪ **[0]** No PI = Doesn't send a 183 response to IP. The call is released. ▪ **[1]** PI = 1; **[8]** PI = 8: Sends a 183 response to IP. **Note:** The *x* in the ini file parameter name denotes the trunk number, where 0 is Trunk 1.
[ConnectOnProgressInd]	Enables the play of announcements from IP to Tel without the need to answer the Tel-to-IP call. It can be used with PSTN networks that don't support the opening of a TDM channel before an ISDN Connect message is received. ▪ **[0]** = (Default) Connect message isn't sent after SIP 183 Session Progress message is received. ▪ **[1]** = Connect message is sent after SIP 183 Session Progress message is received.
Local ISDN Ringback Tone Source `local-isdn-rbt-src` [LocalISDNRBSource_x]	Determines whether the ringback tone is played to the ISDN by the PBX/PSTN or by the device, per trunk. ▪ **[0]** PBX = (Default) PBX/PSTN plays the ringback tone. ▪ **[1]** Gateway = The device plays the ringback tone. **Notes:** ▪ The parameter is used together with the PlayRBTone2Trunk parameter.

Parameter	Description
	- The *x* in the ini file parameter name denotes the trunk number, where 0 is Trunk 1.
PSTN Alert Timeout `pstn-alrt-timeout` [TrunkPSTNAlertTimeout_x]	Defines the Alert Timeout (ISDN T301 timer) in seconds for outgoing calls to PSTN, per trunk. This timer is used between the time that an ISDN Setup message is sent to the Tel side (IP-to-Tel call establishment) and a Connect message is received. If Alerting is received, the timer is restarted. The range is 1 to 600. The default is 180. **Note:** The *x* in the ini file parameter name denotes the trunk number, where 0 is Trunk 1.
B-Channel Negotiation [BChannelNegotiationForTrunk_x]	Determines the ISDN B-channel negotiation mode, per trunk. ▪ **[-1]** Not Configured = (Default) Use per device configuration of the BChannelNegotiation parameter. ▪ **[0]** Preferred. ▪ **[1]** Exclusive. ▪ **[2]** Any. **Notes:** ▪ The option **Any** is applicable only if TerminationSide is set to 0 (i.e., User side). ▪ The *x* in the ini file parameter name denotes the trunk number, where 0 is Trunk 1.
`snd-isdn-ser-aftr-restart` [SendISDNServiceAfterRestart]	Enables the device to send an ISDN SERVice message per trunk upon device reset. The messsage (transmitted on the trunk's D-channel) indicates the availability of the trunk's B-channels (i.e., trunk in service). ▪ **[0]** = Disable (default) ▪ **[0]** = Enable
[SupportRedirectInFacility]	Determines whether the Redirect Number is retrieved from the Facility IE. ▪ **[0]** = (Default) Not supported. ▪ **[1]** = Supports partial retrieval of Redirect Number (number only) from the Facility IE in ISDN Setup messages. This is applicable to Redirect Number according to ECMA-173 Call Diversion Supplementary Services. **Note:** To enable this feature, the parameter ISDNDuplicateQ931BuffMode must be set to 1.
`call-re-rte-mode` [CallReroutingMode]	Determines whether ISDN call rerouting (call forward) is performed by the PSTN instead of by the SIP side. This call forwarding is based on Call Deflection for Euro ISDN (ETS-300-207-1) and QSIG (ETSI TS 102 393). ▪ **[0]** Disable (default). ▪ **[1]** Enable = Enables ISDN call rerouting. When the device sends the INVITE message to the remote SIP entity and receives a SIP 302 response with a Contact header containing a URI host name that is the same as the device's IP address, the device sends a Facility message with a Call Rerouting invoke method to the ISDN and waits for the PSTN side to disconnect the call. **Note:** When the parameter is enabled, ensure that you configure in the IP to Trunk Group Routing table (PSTNPrefix *ini* file parameter) a rule to route the redirected call (using the user part from the 302

Parameter	Description
	Contact header) to the same Trunk Group from where the incoming Tel-to-IP call was received.
[EnableCIC]	Enables the relay of the Carrier Identification Code (CIC) to the ISDN. ▪ **[0]** = (Default) Disabled - CIC is not relayed to the ISDN. ▪ **[1]** = Enabled - CIC (received in the INVITE Request-URI) is relayed to the ISDN in the Transit Network Selection (TNS) IE of the Setup message. For example: INVITE sip:555666;cic=2345@100.2.3.4 sip/2.0. **Notes:** ▪ This feature is supported only for SIP-to-ISDN calls. ▪ The parameter AddCicAsPrefix can be used to add the CIC as a prefix to the destination phone number for routing IP-to-Tel calls.
AoC Support [EnableAOC]	Enables the interworking of ISDN Advice of Charge (AOC) messages to SIP. ▪ **[0]** Disable (default) ▪ **[1]** Enable For more information on AOC, see "Advice of Charge Services for Euro ISDN" on page 518.
Add IE in SETUP `add-ie-in-setup` [AddIEinSetup]	Global parameter that defines an optional Information Element (IE) data (in hex format) to add to ISDN Setup messages. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_AddIEInSetup). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
Trunk Groups to Send IE `trkgrps-to-snd-ie` [SendIEonTG]	Defines Trunk Group IDs (up to 50 characters) from where the optional ISDN IE (defined by the parameter AddIEinSetup) is sent. For example: '1,2,4,10,12,6'. **Notes:** ▪ You can configure different IE data for Trunk Groups by defining the parameter for different IP Profile IDs (using the parameter IPProfile), and then assigning the required IP Profile ID in the IP to Trunk Group Routing table (PSTNPrefix). ▪ When IP Profiles are used for configuring different IE data for Trunk Groups, the parameter is ignored.
Enable User-to-User IE for Tel to IP `uui-ie-for-tel2ip` [EnableUUITel2IP]	Enables transfer of User-to-User (UU) IE from ISDN to SIP. ▪ **[0]** Disable (default) ▪ **[1]** Enable The device supports the following ISDN-to-SIP interworking: Setup to SIP INVITE, Connect to SIP 200 OK, User Information to SIP INFO, Alerting to SIP 18x response, and Disconnect to SIP BYE response messages. **Note:** The interworking of ISDN User-to-User IE to SIP INFO is applicable only to the Euro ISDN, QSIG, and 4ESS ISDN variants.
Enable User-to-User IE for IP to Tel `uui-ie-for-ip2tel`	Enables interworking of SIP user-to-user information (UUI) to User-to-User IE in ISDN Q.931 messages. ▪ **[0]** Disable = (Default) Received UUI is not sent in ISDN message.

Parameter	Description
[EnableUUIP2Tel]	▪ **[1]** Enable = The device interworks UUI from SIP to ISDN messages. The device supports the following SIP-to-ISDN interworking of UUI: ✓ SIP INVITE to Q.931 Setup ✓ SIP REFER to Q.931 Setup ✓ SIP 200 OK to Q.931 Connect ✓ SIP INFO to Q.931 User Information ✓ SIP 18x to Q.931 Alerting ✓ SIP BYE to Q.931 Disconnect **Notes:** ▪ The interworking of ISDN User-to-User IE to SIP INFO is applicable only to the Euro ISDN, QSIG, and 4ESS variants. ▪ To interwork the UUIE header from SIP-to-ISDN messages with the 4ESS ISDN variant, the ISDNGeneralCCBehavior parameter must be set to 16384.
`early-answer-timeout` [EarlyAnswerTimeout]	Global parameter that defines the duration (in seconds) that the device waits for an ISDN Connect message from the called party (Tel side), started from when it sends a Setup message. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_EarlyAnswerTimeout). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
ISDN Transfer Capabilities `isdn-xfer-cab` [ISDNTransferCapability_x]	Defines the IP-to-ISDN Transfer Capability of the Bearer Capability IE in ISDN Setup messages, per trunk (where the *x* in the ini file parameter name denotes the trunk number and where 0 is Trunk 1). ▪ **[-1]** Not Configured ▪ **[0]** Audio 3.1 (default) ▪ **[1]** Speech ▪ **[2]** Data ▪ **[3]** Audio 7 **Note:** If the parameter is not configured or set to -1, Audio 3.1 capability is used.
[TransferCapabilityForDataCalls]	Defines the ISDN Transfer Capability for data calls. ▪ **[0]** = (Default) ISDN Transfer Capability for data calls is 64k unrestricted (data). ▪ **[1]** = ISDN Transfer Capability for data calls is determined according to the ISDNTransferCapability parameter.
[ISDNTransferCompleteTimeout]	Defines the timeout (in seconds) for determining ISDN call transfer (ECT, RLT, or TBCT) failure. If the device does not receive any response to an ISDN transfer attempt within this user-defined time, the device identifies this as an ISDN transfer failure and subsequently performs a hairpin TDM connection or sends a SIP NOTIFY message with a SIP 603 response (depending whether hairpin is enabled or disabled, using the parameter DisableFallbackTransferToTDM). The valid range is 1 to 10. The default is 4.
Enable Network ISDN Transfer `network-isdn-xfer`	Determines whether the device allows interworking of network-side received ECT/TBCT Facility messages (NI-2 TBCT - Two B-channel Transfer and ETSI ECT - Explicit Call Transfer) to SIP REFER.

Parameter	Description
[EnableNetworkISDNTransfer]	▪ **[0]** Disable = Rejects ISDN transfer requests. ▪ **[1]** Enable = (Default) The device sends a SIP REFER message to the remote call party if ECT/TBCT Facility messages are received from the ISDN side (e.g., from a PBX).
[DisableFallbackTransferToTDM]	Enables "hairpin" TDM transfer upon ISDN (ECT, RLT, or TBCT) call transfer failure. When this feature is enabled and an ISDN call transfer failure occurs, the device sends a SIP NOTIFY message with a SIP 603 Decline response. ▪ **[0]** = (Default) The device performs a hairpin TDM transfer upon ISDN call transfer. ▪ **[1]** = Hairpin TDM transfer is disabled.
Enable QSIG Transfer Update `qsig-xfer-update` [EnableQSIGTransferUpdate]	Determines whether the device interworks QSIG Facility messages with CallTranferComplete or CallTransferUpdate invoke application protocol data units (APDU) to SIP UPDATE messages with P-Asserted-Identity and optional Privacy headers. This feature is supported for IP-to-Tel and Tel-to-IP calls. ▪ **[0]** Disable = (Default) Ignores QSIG Facility messages with CallTranferComplete or CallTransferUpdate invokes. ▪ **[1]** Enable For example, assume A and C are PBX call parties and B is the SIP IP phone: 1 A calls B; B answers the call. 2 A places B on hold and calls C; C answers the call. 3 A performs a call transfer (the transfer is done internally by the PBX); B and C are connected to one another. In the above example, the PBX updates B that it is now talking with C. The PBX updates this by sending a QSIG Facility message with CallTranferComplete invoke APDU. The device interworks this message to a SIP UPDATE message containing a P-Asserted-Identity header with the number and name derived from the QSIG CallTranferComplete RedirectionNumber and RedirectionName. **Notes:** ▪ For IP-to-Tel calls, the RedirectionNumber and RedirectionName in the CallTRansferComplete invoke is derived from the P-Asserted-Identity and Privacy headers in the received SIP INFO message. ▪ To include the P-Asserted-Identity header in outgoing SIP UPDATE messages, set the AssertedIDMode parameter to **Add P-Asserted-Identity.**
Release Cause Mapping from ISDN to SIP Table	
Release Cause Mapping Table `configure voip > gw manipulations CauseMapIsdn2Sip` [CauseMapISDN2SIP]	This table parameter maps ISDN Q.850 Release Causes to SIP responses. The format of the ini file table parameter is as follows: [CauseMapISDN2SIP] FORMAT CauseMapISDN2SIP_Index = CauseMapISDN2SIP_IsdnReleaseCause, CauseMapISDN2SIP_SipResponse; [\CauseMapISDN2SIP]
Release Cause Mapping from SIP to ISDN Table	
Release Cause Mapping Table	This table parameter maps SIP responses to Q.850 Release Causes. The format of the ini file table parameter is as follows:

Parameter	Description
`configure voip > gw manipulations CauseMapSip2Isdn` [CauseMapSIP2ISDN]	[CauseMapSIP2ISDN] FORMAT CauseMapSIP2ISDN_Index = CauseMapSIP2ISDN_SipResponse, CauseMapSIP2ISDN_IsdnReleaseCause; [\CauseMapSIP2ISDN]
ISDN-to-ISDN Release Cause Code Conversion Table	
Release Cause ISDN > ISDN `configure voip > gw manipulations cause-map-isdn2isdn` [CauseMapIsdn2Isdn]	Defines ISDN-to-ISDN release cause code mapping rules. The format of the ini file table parameter is as follows: [CauseMapIsdn2Isdn] FORMAT CauseMapIsdn2Isdn_Index = CauseMapIsdn2Isdn_OrigIsdnReleaseCause, CauseMapIsdn2Isdn_MapIsdnReleaseCause; [\CauseMapSip2Isdn] For a detailed description of this table, see "Configuring ISDN-to-ISDN Release Cause Mapping" on page 463.

57.10.8 Answer and Disconnect Supervision Parameters

The answer and disconnect supervision parameters are described in the table below.

Table 57-55: Answer and Disconnect Parameters

Parameter	Description
Wait before PSTN Release-Ack `wait-befor-pstn-rel-ack` [TimeToWaitForPstnReleaseAck]	Defines a timeout (in milliseconds) that the device waits for the receipt of an ISDN Q.931 Release message from the PSTN side before releasing the channel. The Release ACK is typically sent by the PSTN in response to the device's Disconnect message to end the call. If the timeout expires and a Release message has not yet been received, the device releases the call channel. The valid value is 1 to 360,000. The default is 6,000.
`configure voip > sip advanced-settings > set mn-call-duration` [MinCallDuration]	Defines the minimum call duration (in seconds) for the Tel side. If an established call is terminated by the IP side before this duration expires, the device terminates the call with the IP side, but delays the termination toward the Tel side until this timeout expires. The valid value range is 0 to 10 seconds, where 0 (default) disables this feature. For example: assume the minimum call duration is set to 10 seconds and an IP phone hangs up a call established with a BRI phone after 2 seconds. As the call duration is less than the minimum call duration, the device does not disconnect the call on the Tel side. However, it sends a SIP 200 OK immediately upon receipt of the BYE to disconnect from the IP phone. The call is disconnected from the Tel side only when the call duration is greater than or equal to the minimum call duration. **Notes:** ▪ The parameter is applicable to IP-to-Tel and Tel-to-IP calls. ▪ The parameter is applicable only to ISDN protocols.

Parameter	Description
Disconnect on Broken Connection `disc-broken-conn` [DisconnectOnBrokenConnection]	Global parameter that defines the device's handling of calls if RTP packets are not received within a user-defined timeout (configured by the BrokenConnectionEventTimeout parameter). You can also configure this functionality per specific calls, using IP Profiles (IpProfile_DisconnectOnBrokenConnection). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
Broken Connection Timeout `broken-connection-event-timeout` [BrokenConnectionEventTimeout]	Defines the time period (in 100-msec units) after which a call is disconnected if an RTP packet is not received. The valid range is from 3 (i.e., 300 msec) to an unlimited value (e.g., 20 hours). The default is 100 (i.e., 10000 msec or 10 seconds). **Notes:** ▪ The parameter is applicable only if the parameter DisconnectOnBrokenConnection is set to 1. ▪ Currently, this feature functions only if Silence Suppression is disabled.
Trunk Alarm Call Disconnect Timeout `trk-alrm-call-disc-to` [TrunkAlarmCallDisconnectTimeout]	Defines the duration (in seconds) to wait after a BRI trunk "Red" alarm (LOS / LOF) is raised, before the device disconnects the SIP call. If this timeout expires and the alarm is still raised, the device sends a SIP BYE message to terminate the call. If the alarm is cleared before this timeout expires, the call is not terminated, but continues as normal. The range is 1 to 3600. The default is 20.
Disconnect Call on Busy Tone Detection (ISDN) `disc-on-bsy-tone-i` [ISDNDisconnectOnBusyTone]	Determines whether a call is disconnected upon detection of a busy tone (for ISDN). ▪ [0] Disable = (Default) Do not disconnect call upon detection of busy tone. ▪ [1] Enable = Disconnect call upon detection of busy tone. **Notes:** ▪ The parameter is applicable only to ISDN protocols. ▪ IP-to-ISDN calls are disconnected on detection of SIT tones only in call alert state. If the call is in connected state, the SIT does not disconnect the calls. Detection of busy or reorder tones disconnects the IP-to-ISDN calls also in call connected state.

57.10.9 Tone Parameters

This subsection describes the device's tone parameters.

57.10.9.1 Telephony Tone Parameters

The telephony tone parameters are described in the table below.

Table 57-56: Tone Parameters

Parameter	Description
Dial Tone Duration `dt-duration` [TimeForDialTone]	Defines the duration (in seconds) that the dial tone is played to an ISDN terminal. The parameter is applicable for overlap dialing when ISDNInCallsBehavior is set to 65536. The dial tone is played if the ISDN Setup message doesn't include the called number. The valid range is 0 to 60. The default is 5.
Reorder Tone Duration `reorder-tone-duration` [TimeForReorderTone]	Defines the duration (in seconds) that the device plays a busy or reorder tone before releasing the line. The valid range is 0 to . The default is 10 seconds. Note that the Web interface denotes the default value as a string value of "255". **Notes:** ▪ The selected busy or reorder tone is according to the SIP release cause code received from IP. ▪ The parameter is also applicable for ISDN when the PlayBusyTone2ISDN parameter is set to 2. ▪ The parameter can also be configured for a Tel Profile (in the Tel Profile table).
Play Busy Tone to Tel `play-bsy-tone-2tel` [PlayBusyTone2ISDN]	Enables the device to play a busy or reorder tone to the PSTN after a Tel-to-IP call is released. ▪ [0] Don't Play = (Default) Immediately sends an ISDN Disconnect message. ▪ [1] Play when Disconnecting = Sends an ISDN Disconnect message with PI = 8 and plays a busy or reorder tone to the PSTN (depending on the release cause). ▪ [2] Play before Disconnect = Delays the sending of an ISDN Disconnect message for a user-defined time (configured by the TimeForReorderTone parameter) and plays a busy or reorder tone to the PSTN. This is applicable only if the call is released from the IP [Busy Here (486) or Not Found (404)] before it reaches the Connect state; otherwise, the Disconnect message is sent immediately and no tones are played.
`configure voip > gw digitalgw digital-gw-parameters > q850-reason-code-2play-user-tone` [Q850ReasonCode2PlayUserTone]	Defines an ISDN Q.8931 release cause code(s), which if mapped to the SIP release reason received from the IP side, causes the device to play a user-defined tone from the installed PRT file to the Tel side. For example, if the the received SIP release cause is 480 Temporarily Unavailable and you configure the parameter with Q.931 release code 18 (No User Responding), the device plays the user-defined tone to the Tel side. The user-defined tone is configured when creating the PRT file, using AudioCodes DConvert utility. The tone must be assigned to the "acSpecialConditionTone" (Tone Type 21) option in DConvert. The parameter can be configured with up to 10 release codes. When configuring multiple codes, separate the codes by commas (without spaces). For example: `Q850ReasonCode2PlayUserTone = 1,18,24` If the SIP release reason received from the IP side is mapped to the Q.931 release code specified by the parameter, the

Parameter	Description
	device plays the user-defined tone. Otherwise, if not specified and the release code is 17 (User Busy), the device plays the busy tone and for all other release codes, the device plays the reorder tone. **Note:** To enable the feature, the 'Play Busy Tone to Tel' (PlayBusyTone2ISDN) parameter must be enabled (set to 1 or 2).
Play Ringback Tone to Tel `play-rbt2tel` [PlayRBTone2Tel]	Determines the playing method of the ringback tone to the Trunk (for digital interfaces) side.The parameter applies to all trunks that are not configured by the PlayRBTone2Trunk parameter (which defines ringback tone per Trunk). ▪ **[0]** Don't Play = ✓ The device doesn't play a ringback tone. No PI is sent to the ISDN unless the ProgressIndicator2ISDN_x parameter is configured differently. ▪ **[1]** Play on Local = ✓ The device operates according to the LocalISDNRBSource parameter: 1) If the device receives a 180 Ringing response (with or without SDP) and the LocalISDNRBSource parameter is set to 1, it plays a ringback tone and sends an ISDN Alert with PI = 8 (unless the ProgressIndicator2ISDN_x parameter is configured differently). 2) If the LocalISDNRBSource parameter is set to 0, the device doesn't play a ringback tone and an Alert message without PI is sent to the ISDN. In this case, the PBX / PSTN plays the ringback tone to the originating terminal. Note that the receipt of a 183 response does not cause the device configured for ISDN to play a ringback tone; the device issues a Progress message (unless SIP183Behaviour is set to 1). If the SIP183Behaviour parameter is set to 1, the 183 response is handled the same way as a 180 Ringing response. ▪ **[2]** Prefer IP = (Default): ✓ Plays according to 'Early Media'. If a SIP 180 response is received and the voice channel is already open (due to a previous 183 early media response or due to an SDP in the current 180 response), the device doesn't play the ringback tone; PI = 8 is sent in an ISDN Alert message (unless the ProgressIndicator2ISDN_x parameter is configured differently. The device operates according to the LocalISDNRBSource parameter: 1) If LocalISDNRBSource is set to 1, the device plays a ringback tone and sends an ISDN Alert with PI = 8 to the ISDN (unless the ProgressIndicator2ISDN_x parameter is configured differently). 2) If LocalISDNRBSource is set to 0, the device doesn't play a ringback tone. No PI is sent in the ISDN Alert message (unless the ProgressIndicator2ISDN_x parameter is configured differently). In this case, the PBX / PSTN plays a ringback tone to the originating terminal. Note that the receipt of a 183 response

Parameter	Description
	results in an ISDN Progress message (unless SIP183Behaviour is set to 1). If SIP183Behaviour is set to 1 (183 is handled the same way as a 180 + SDP), the device sends an Alert message with PI = 8, without playing a ringback tone. ▪ **[3]** Play Local Until Remote Media Arrive = Plays a ringback tone according to received media. The behaviour is similar to [2]. If a SIP 180 response is received and the voice channel is already open (due to a previous 183 early media response or due to an SDP in the current 180 response), the device plays a local ringback tone if there are no prior received RTP packets. The device stops playing the local ringback tone as soon as it starts receiving RTP packets. At this stage, if the device receives additional 18x responses, it does not resume playing the local ringback tone. Note that for ISDN trunks, this option is applicable only if the LocalISDNRBSource parameter is set to 1. **Note:** The parameter is applicable only to the Gateway application.
Play Ringback Tone to Trunk `play-rbt-to-trk` [PlayRBTone2Trunk_x]	Determines the playing method of the ringback tone to the trunk side, per trunk. ▪ [-1] Not configured = (Default) The settings of the PlayRBTone2Tel parameter is used. ▪ [0] Don't Play = When the device is configured for ISDN / CAS, it doesn't play a ringback tone. No Progress Indicator (PI) is sent to the ISDN unless the ProgressIndicator2ISDN_x parameter is configured differently. ▪ [1] Play on Local = When the device is configured for CAS, it plays a local ringback tone to the PSTN upon receipt of a SIP 180 Ringing response (with or without SDP). Note that the receipt of a SIP 183 response does not cause the device configured for CAS to play a ringback tone (unless the SIP183Behaviour parameter is set to 1). When the device is configured for ISDN, it operates according to the LocalISDNRBSource parameter, as follows: ✓ If the device receives a SIP 180 Ringing response (with or without SDP) and the LocalISDNRBSource parameter is set to 1, it plays a ringback tone and sends an ISDN Alert with PI = 8 (unless the ProgressIndicator2ISDN_x parameter is configured differently). ✓ If the LocalISDNRBSource parameter is set to 0, the device doesn't play a ringback tone and an Alert message without PI is sent to the ISDN. In this case, the PBX / PSTN plays the ringback tone to the originating terminal. Note that the receipt of a 183 response does not cause the device to play a ringback tone; the device sends a Progress message (unless SIP183Behaviour is set to 1). If the SIP183Behaviour parameter is set to 1, the 183 response is handled the same way as a 180 Ringing response.

Parameter	Description
	▪ [2] Prefer IP = Plays according to 'Early Media'. If a SIP 180 response is received and the voice channel is already open (due to a previous 183 early media response or due to an SDP in the current 180 response), the device configured for ISDN / CAS doesn't play the ringback tone; PI = 8 is sent in an ISDN Alert message (unless the ProgressIndicator2ISDN_x parameter is configured differently). If a 180 response is received, but the 'early media' voice channel is not opened, the device configured for CAS plays a ringback tone to the PSTN. The device configured for ISDN operates according to the LocalISDNRBSource parameter: ✓ If LocalISDNRBSource is set to 1, the device plays a ringback tone and sends an ISDN Alert with PI = 8 to the ISDN (unless the ProgressIndicator2ISDN_x parameter is configured differently). ✓ If LocalISDNRBSource is set to 0, the device doesn't play a ringback tone. No PI is sent in the ISDN Alert message (unless the ProgressIndicator2ISDN_x parameter is configured differently). In this case, the PBX / PSTN plays a ringback tone to the originating terminal. Note that the receipt of a 183 response results in an ISDN Progress message (unless SIP183Behaviour is set to 1). If SIP183Behaviour is set to 1 (183 is handled the same way as a 180 with SDP), the device sends an Alert message with PI = 8 without playing a ringback tone. ▪ [3] Play Local Until Remote Media Arrive = Plays tone according to received media. The behaviour is similar to option [2]. If a SIP 180 response is received and the voice channel is already open (due to a previous 183 early media response or due to an SDP in the current 180 response), the device plays a local ringback tone if there are no prior received RTP packets. The device stops playing the local ringback tone as soon as it starts receiving RTP packets. At this stage, if the device receives additional 18x responses, it does not resume playing the local ringback tone. Note that for ISDN trunks, this option is applicable only if LocalISDNRBSource is set to 1. **Notes:** ▪ The parameter is applicable only to the Gateway (GW) application. ▪ The x in the ini file parameter name denotes the trunk number, where 0 is Trunk 1.
Play Ringback Tone to IP `play-rbt-2ip` [PlayRBTone2IP]	Global parameter that enables the device to play a ringback tone to the IP side for IP-to-Tel calls. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_PlayRBTone2IP). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** ▪ If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.

Parameter	Description
Play Local RBT on ISDN Transfer `play-l-rbt-isdn-trsfr` [PlayRBTOnISDNTransfer]	Determines whether the device plays a local ringback tone for ISDN's Two B Channel Transfer (TBCT), Release Line Trunk (RLT), or Explicit Call Transfer (ECT) call transfers to the originator when the second leg receives an ISDN Alerting or Progress message. - [0] Don't Play (default) - [1] Play **Notes:** - For Blind transfer, the local ringback tone is played to first call PSTN party when the second leg receives the ISDN Alerting or Progress message. - For Consulted transfer, the local ringback tone is played when the second leg receives ISDN Alerting or Progress message if the Progress message is received after a SIP REFER. - The parameter is applicable only if the parameter SendISDNTransferOnConnect is set to 1.
MFC R2 Category `mfcr2-category` [R2Category]	Defines the tone for MFC R2 calling party category (CPC). The parameter provides information on the calling party such as National or International call, Operator or Subscriber and Subscriber priority. The value range is 1 to 15 (defining one of the MFC R2 tones). The default is 1.

57.10.9.2 Tone Detection Parameters

The signal tone detection parameters are described in the table below.

Table 57-57: Tone Detection Parameters

Parameter	Description
`DTMF-detector-enable` [DTMFDetectorEnable]	Enables the detection of DTMF signaling. - **[0]** = Disable - **[1]** = Enable (default)
`MFR1-detector-enable` [MFR1DetectorEnable]	Enables the detection of MF-R1 signaling. - **[0]** = Disable (default) - **[1]** = Enable
[R1DetectionStandard]	Determines the MF-R1 protocol used for detection. - [0] = ITU (default) - [1] = R1.5 **Note:** For the parameter to take effect, a device reset is required.
`user-defined-tones-detector-enable` [UserDefinedToneDetectorEnable]	Enables the detection of User Defined Tones signaling, applicable for Special Information Tone (SIT) detection. - **[0]** = Disable (default) - **[1]** = Enable
`sit-detector-enable` [SITDetectorEnable]	Enables SIT detection according to the ITU-T recommendation E.180/Q.35. - **[0]** = Disable (default)

Parameter	Description
	- **[1]** = Enable To disconnect IP-to-ISDN calls when a SIT tone is detected, the following parameters must be configured: - SITDetectorEnable = 1 - UserDefinedToneDetectorEnable = 1 - ISDNDisconnectOnBusyTone = 1 (applicable for Busy, Reorder and SIT tones) Another parameter for handling the SIT tone is SITQ850Cause, which determines the Q.850 cause value specified in the SIP Reason header that is included in a 4xx response when a SIT tone is detected on an IP-to-Tel call. **Notes:** - For the parameter to take effect, a device reset is required. - The IP-to-ISDN call is disconnected on detection of a SIT tone only in call alert state. If the call is in connected state, the SIT does not disconnect the call. Detection of busy or reorder tones disconnect these calls also in call connected state.
`udt-detector-frequency-deviation` [UDTDetectorFrequencyDeviation]	Defines the deviation (in Hz) allowed for the detection of each signal frequency. The valid range is 1 to 50. The default is 50. **Note:** For the parameter to take effect, a device reset is required.
`cpt-detector-frequency-deviation` [CPTDetectorFrequencyDeviation]	Defines the deviation (in Hz) allowed for the detection of each CPT signal frequency. The valid range is 1 to 30. The default is 10. **Note:** For the parameter to take effect, a device reset is required.

57.10.9.3 Metering Tone Parameters

The metering tone parameters are described in the table below.

Table 57-58: Metering Tone Parameters

Parameter	Description
Generate Metering Tones `gen-mtr-tones` [PayPhoneMeteringMode]	Defines the method for configuring metering tones that are generated to the Tel side. - **[0]** Disable = (Default) Metering tones are not generated. - **[1]** Internal Table = Metering tones are generated by the device according to the Charge Code table and sent to the Tel side. - [2] SIP Interval Provided = (Proprietary method of TELES Communications Corporation) Advice-of-Charge service toward the PSTN. Periodic generation of AOC-D and AOC-E toward the PSTN. Calculation is based on seconds. The time interval is calculated according to the scale and tariff provided in the proprietary formatted file included in SIP INFO messages, which is always sent before 200 0K. The device ignores tariffs sent after the call is established. - [3] SIP RAW Data Provided = (Proprietary method of Cirpack) Advice-of-Charge service toward the PSTN. The received AOC-D messages contain a subtotal. When receiving AOC-D in raw format,

Parameter	Description
	provided in the header of SIP INFO messages, the device parses AOC-D raw data to obtain the number of units. This number is sent in the Facility message with AOC-D. In addition, the device stores the latest number of units in order to send them in AOC-E IE when the call is disconnected. ▪ [4] SIP RAW Data Incremental Provided = (Proprietary method of Cirpack) Advice-of-Charge service toward the PSTN. The AOC-D message in the payload is an increment. When receiving AOC-D in raw format, provided in the header of SIP INFO messages, the device parses AOC-D raw data to obtain the number of units. This number is sent in the Facility message with AOC-D. The device generates the AOC-E. Parsing every AOC-D received and summing the values is required to obtain the total sum (that is placed in the AOC-E). ▪ [5] SIP 2 TEL INTERWORKING = Enables IP-to-Tel AOC, using AudioCodes' proprietary SIP header, AOC. **Note:** The parameter is applicable only to ISDN Euro trunks for sending AOC Facility messages (see Advice of Charge Services for Euro ISDN on page 518).
Charge Codes Table	
`configure voip > gw analoggw ChargeCode` [ChargeCode]	Defines metering tones and their time intervals that the Euro ISDN trunk sends in AOC Facility messages to the PSTN (i.e., PBX). The format of the ini file table parameter is as follows: [ChargeCode] FORMAT ChargeCode_Index = ChargeCode_EndTime1, ChargeCode_PulseInterval1, ChargeCode_PulsesOnAnswer1, ChargeCode_EndTime2, ChargeCode_PulseInterval2, ChargeCode_PulsesOnAnswer2, ChargeCode_EndTime3, ChargeCode_PulseInterval3, ChargeCode_PulsesOnAnswer3, ChargeCode_EndTime4, ChargeCode_PulseInterval4, ChargeCode_PulsesOnAnswer4; [\ChargeCode] **Notes:** ▪ To associate a configured Charge Code to an outgoing Tel-to-IP call, use the Tel-to-IP Routing table.

57.10.10 Trunk Groups and Routing Parameters

The routing parameters are described in the table below.

Table 57-59: Routing Parameters

Parameter	Description
Trunk Group Table	
Trunk Group Table `configure voip > gw hunt-or-trunk-group TrunkGroup` [TrunkGroup]	Defines and activates Trunk Groups. The format of the ini file table parameter is as follows: [TrunkGroup] FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum, TrunkGroup_FirstTrunkId, TrunkGroup_FirstBChannel, TrunkGroup_LastBChannel, TrunkGroup_FirstPhoneNumber, TrunkGroup_ProfileName,

Parameter	Description
	TrunkGroup_LastTrunkId, TrunkGroup_Module; [\TrunkGroup] For a description of the table, seeConfiguring Trunk Groups on page 433. **Note:** Trunk Group ID 1 is denoted as 0 in the table.
Trunk Group Settings Table	
Trunk Group Settings `configure voip > gw hunt-or-trunk-group trunk-group-setting` [TrunkGroupSettings]	Defines the rules for channel allocation per Trunk Group. The format of the ini file table parameter is as follows: [TrunkGroupSettings] FORMAT TrunkGroupSettings_Index = TrunkGroupSettings_TrunkGroupId, TrunkGroupSettings_ChannelSelectMode, TrunkGroupSettings_RegistrationMode, TrunkGroupSettings_GatewayName, TrunkGroupSettings_ContactUser, TrunkGroupSettings_ServingIPGroupName, TrunkGroupSettings_MWIInterrogationType, TrunkGroupSettings_TrunkGroupName, TrunkGroupSettings_UsedByRoutingServer; [\TrunkGroupSettings] For a description of the table, see "Configuring Trunk Group Settings" on page 435.
Channel Select Mode `ch-select-mode` [ChannelSelectMode]	Defines the method for allocating incoming IP-to-Tel calls to a channel. The parameter applies to the following: ▪ All Trunk Groups configured without a channel select mode in the Trunk Group Settings table (see "Configuring Trunk Group Settings" on page 435). ▪ All channels and trunks configured without a Trunk Group ID. for all Trunk Groups channels that are configured without a Trunk Group ID,. ▪ **[0]** By Dest Phone Number ▪ **[1]** Cyclic Ascending (default) ▪ **[2]** Ascending ▪ **[3]** Cyclic Descending ▪ **[4]** Descending ▪ **[5]** Dest Number + Cyclic Ascending. ▪ **[6]** By Source Phone Number ▪ [7] Trunk Cyclic Ascending ▪ [8] Trunk & Channel Cyclic Ascending ▪ **[11]** Dest Number + Ascending For a detailed description of the parameter's options, see "Configuring Trunk Group Settings" on page 435.
Default Destination Number `dflt-dest-nb` [DefaultNumber]	Defines the default destination phone number, which is used if the received message doesn't contain a called party number and no phone number is configured in the Trunk Group table (see Configuring the Trunk Groups on page 433). The parameter is used as a starting number for the list of channels comprising all the device's Trunk Groups. The default is 1000.

Parameter	Description
Source IP Address Input `src-ip-addr-input` [SourceIPAddressInput]	Determines which IP address the device uses to determine the source of incoming INVITE messages for IP-to-Tel routing. ▪ **[-1]** = (Default) Auto Decision - the parameter is automatically set to SIP Contact Header (1). ▪ **[0]** SIP Contact Header = The IP address in the Contact header of the incoming INVITE message is used. ▪ **[1]** Layer 3 Source IP = The actual IP address (Layer 3) from where the SIP packet was received is used.
Use Source Number As Display Name `src-nb-as-disp-name` [UseSourceNumberAsDisplayName]	Determines the use of Tel Source Number and Display Name for Tel-to-IP calls. ▪ **[0]** No = (Default) If a Tel Display Name is received, the Tel Source Number is used as the IP Source Number and the Tel Display Name is used as the IP Display Name. If no Display Name is received from the Tel side, the IP Display Name remains empty. ▪ **[1]** Yes = If a Tel Display Name is received, the Tel Source Number is used as the IP Source Number and the Tel Display Name is used as the IP Display Name. If no Display Name is received from the Tel side, the Tel Source Number is used as the IP Source Number and also as the IP Display Name. ▪ **[2]** Overwrite = The Tel Source Number is used as the IP Source Number and also as the IP Display Name (even if the received Tel Display Name is not empty). ▪ **[3]** Original = Similar to option [2], except that the operation is done before regular calling number manipulation.
Use Display Name as Source Number `disp-name-as-src-nb` [UseDisplayNameAsSourceNumber]	Defines how the display name (caller ID) received from the IP side (in the SIP From header) effects the source number sent to the Tel side, for IP-to-Tel calls. ▪ [0] No = (Default) If a display name is received from the IP side, the source number of the IP side is used as the Tel source number. ▪ [1] Yes = If a display name is received from the IP side, the display name of the IP side is used as the Tel source number and Presentation is set to Allowed (0). If no display name is received from the IP side, the source number of the IP side is used as the Tel source number and Presentation is set to Restricted (1). For example: ✓ If 'From: 100 <sip:200@201.202.203.204>' is received from the IP side, the outgoing source number (and display name) are set to "100" and Presentation is set to Allowed (0). ✓ If 'From: <sip:400@101.102.103.104>' is received from the IP side, the outgoing source number is set to "400" and Presentation is set to Restricted (1). ▪ [2] Preferred = If a display name is received from the IP side, the display name of the IP side is used as the Tel source number. If no display name is received from the IP side, this setting does not affect the Tel source number.

Parameter	Description
ENUM Resolution `enum-service-domain` [EnumService]	Defines the ENUM service for translating telephone numbers to IP addresses or domain names (FQDN), for example, e164.arpa, e164.customer.net, or NRENum.net. The valid value is a string of up to 50 characters. The default is "e164.arpa". **Note:** ENUM-based routing is configured in the Tel-to-IP Routing table using the "ENUM" string value as the destination address to denote the parameter's value.
Use Routing Table for Host Names and Profiles `rte-tbl-4-host-names` [AlwaysUseRouteTable]	Determines whether to use the device's routing table to obtain the URI host name and optionally, an IP profile (per call) even if a Proxy server is used. ▪ **[0]** Disable = (Default) Don't use the Tel-to-IP Routing table. ▪ **[1]** Enable = Use the Tel-to-IP Routing table. **Notes:** ▪ The parameter appears only if the 'Use Default Proxy' parameter is enabled. ▪ The domain name is used instead of a Proxy name or IP address in the INVITE SIP URI.
Tel to IP Routing Mode onfigure voip > gw routing general-setting > tel2ip-rte-mode [RouteModeTel2IP]	Determines whether to route Tel calls to an IP destination before or after manipulation of the destination number. This applies to Tel-to-IP routing rules configured in the Tel-to-IP Routing table. ▪ **[0]** Route calls before manipulation = Calls are routed before the number manipulation rules are applied (default). ▪ **[1]** Route calls after manipulation = Calls are routed after the number manipulation rules are applied. **Notes:** ▪ The parameter is not applicable if outbound proxy routing is used. ▪ For number manipulation, see "Configuring Source/Destination Number Manipulation" on page 441. ▪ For configuring Tel-to-IP routing rules, see "Configuring Tel-to-IP Routing Rules" on page 467.
Tel-to-IP Routing table	
Tel-to-IP Routing table `configure voip > gw routing tel2ip-routing` [Prefix]	Defines Tel-to-IP routing rules for routing Tel-to-IP calls. The format of the ini file table parameter is: [PREFIX] FORMAT PREFIX_Index = PREFIX_RouteName, PREFIX_DestinationPrefix, PREFIX_DestAddress, PREFIX_SourcePrefix, PREFIX_ProfileName, PREFIX_MeteringCode, PREFIX_DestPort, PREFIX_DestIPGroupName, PREFIX_TransportType, PREFIX_SrcTrunkGroupID, PREFIX_DestSIPInterfaceName, PREFIX_CostGroup, PREFIX_ForkingGroup, PREFIX_CallSetupRulesSetId, PREFIX_ConnectivityStatus; [\PREFIX] For a detailed description of the table, see "Configuring Tel-to-IP Routing Rules" on page 467.

Parameter	Description
IP to Trunk Group Routing Table	
IP to Trunk Group Routing `configure voip > gw routing ip2tel-routing` [PSTNPrefix]	Defines the routing of IP-to-Trunk Groups. The format of the ini file table parameter is as follows: [PSTNPrefix] FORMAT PstnPrefix_Index = PstnPrefix_RouteName, PstnPrefix_DestPrefix, PstnPrefix_TrunkGroupId, PstnPrefix_SourcePrefix, PstnPrefix_SourceAddress, PstnPrefix_ProfileName, PstnPrefix_SrcIPGroupName, PstnPrefix_DestHostPrefix, PstnPrefix_SrcHostPrefix, PstnPrefix_SrcSIPInterfaceName, PstnPrefix_TrunkId, PstnPrefix_CallSetupRulesSetId, PstnPrefix_DestType; [\PSTNPrefix] For a detailed description of the table, see "Configuring IP-to-Trunk Group Routing Rules" on page 476.
IP to Tel Routing Mode `ip2tel-rte-mode` [RouteModeIP2Tel]	Determines whether to route IP calls to the Trunk Group before or after manipulation of the destination number (configured in "Configuring Source/Destination Number Manipulation Rules" on page 441). ▪ **[0]** Route calls before manipulation = (Default) Calls are routed before the number manipulation rules are applied. ▪ **[1]** Route calls after manipulation = Calls are routed after the number manipulation rules are applied.
IP Security `ip-security` [SecureCallsFromIP]	Determines the device's policy on accepting or blocking SIP calls (IP-to-Tel calls). This is useful in preventing unwanted SIP calls, SIP messages, and/or VoIP spam. ▪ **[0]** Disable = (Default) The device accepts all SIP calls. ▪ **[1]** Secure Incoming calls = The device accepts SIP calls (i.e., calls from the IP side) only from IP addresses that are configured in the Tel-to-IP Routing table or Proxy Set table, or IP addresses resolved from DNS servers from FQDN values configured in the Proxy Set table. All other incoming calls are rejected. ▪ **[2]** Secure All calls = The device accepts SIP calls only from IP addresses (in dotted-decimal notation format) that are defined in the Tel-to-IP Routing table or Proxy Set table, and rejects all other incoming calls. In addition, if an FQDN is defined in the routing table or Proxy Set table, the call is allowed to be sent only if the resolved DNS IP address appears in one of these tables; otherwise, the call is rejected. Therefore, the difference between this option and option [1] is that this option is concerned only about numerical IP addresses that are defined in the tables. **Note:** If the parameter is set to [0] or [1], when using Proxies or Proxy Sets, it is unnecessary to configure the Proxy IP addresses in the routing table. The device allows SIP calls received from the Proxy IP addresses even if these addresses are not configured in the routing table.
Filter Calls to IP `filter-calls-to-ip` [FilterCalls2IP]	Enables filtering of Tel-to-IP calls when a Proxy Set is used. ▪ **[0]** Don't Filter = (Default) The device doesn't filter calls when using a proxy. ▪ **[1]** Filter = Filtering is enabled.

Parameter	Description
	When the parameter is enabled and a proxy is used, the device first checks the Tel-to-IP Routing table before making a call through the proxy. If the number is not allowed (i.e., number isn't listed in the table or a call restriction routing rule of IP address 0.0.0.0 is applied), the call is released. **Note:** When no proxy is used, the parameter must be disabled and filtering is according to the Tel-to-IP Routing table.
IP-to-Tel Tagging Destination Dial Plan Index `ip2tel-tagging-dst` [IP2TelTaggingDestDialPlanIndex]	Defines the Dial Plan index in the Dial Plan file for called prefix tags for representing called number prefixes in Inbound Routing rules. The valid values are 0 to 7, where 0 denotes PLAN1, 1 denotes PLAN2, and so on. The default is -1 (i.e., no dial plan file used). For more information on this feature, see Dial Plan Prefix Tags for IP-to-Tel Routing on page 655.
IP to Tel Tagging Source Dial Plan Index `configure voip/gw routing general-setting/ip-to-tel-tagging-src` [IP2TelTaggingSourceDialPlanIndex]	Defines the Dial Plan index in the Dial Plan file for calling prefix tags for representing calling number prefixes in Inbound Routing rules. The valid values are 0 to 7, where 0 denotes PLAN1, 1 denotes PLAN2, and so on. The default is -1 (i.e., no dial plan file used). For more information on this feature, see Dial Plan Prefix Tags for IP-to-Tel Routing on page 655.
`etsi-diversion` [EnableETSIDiversion]	Determines the method in which the Redirect Number is sent to the Tel side. ▪ [0] = (Default) Q.931 Redirecting Number Information Element (IE). ▪ [1] = ETSI DivertingLegInformation2 in a Facility IE.
Add CIC `add-cic` [AddCicAsPrefix]	Determines whether to add the Carrier Identification Code (CIC) as a prefix to the destination phone number for IP-to-Tel calls. When the parameter is enabled, the 'cic' parameter in the incoming SIP INVITE can be used for IP-to-Tel routing decisions. It routes the call to the appropriate Trunk Group based on the parameter's value. ▪ **[0]** No (default) ▪ **[1]** Yes The SIP 'cic' parameter enables the transmission of the 'cic' parameter from the SIP network to the ISDN. The 'cic' parameter is a three- or four-digit code used in routing tables to identify the network that serves the remote user when a call is routed over many different networks. The 'cic' parameter is carried in the SIP INVITE and maps to the ISDN Transit Network Selection Information Element (TNS IE) in the outgoing ISDN Setup message (if the EnableCIC parameter is set to 1). The TNS IE identifies the requested transportation networks and allows different providers equal access support, based on customer choice. For example, as a result of receiving the below INVITE, the destination number after number manipulation is

Parameter	Description
	cic+167895550001: INVITE sip:5550001;cic=+16789@172.18.202.60:5060;user=phone SIP/2.0 **Note:** After the cic prefix is added, the IP to Trunk Group Routing table can be used to route this call to a specific Trunk Group. The Destination Number IP to Tel Manipulation table must be used to remove this prefix before placing the call to the ISDN.
[FaxReroutingMode]	Enables the re-routing of incoming Tel-to-IP calls that are identified as fax calls. If a CNG tone is detected on the Tel side of a Tel-to-IP call, the device adds the string, "FAX" as a prefix to the destination number before routing and manipulation. A routing rule in the Tel-to-IP Routing table having the value "FAX" (case-sensitive) as the destination number is then used to re-route the call to a fax destination and the destination number manipulation mechanism is used to remove the "FAX" prefix before sending the fax, if required. If the initial INVITE used to establish the voice call (not fax) was already sent, a CANCEL (if not connected yet) or a BYE (if already connected) is sent to release the voice call. ▪ **[0]** Disable (default) ▪ **[1]** Rerouting without Delay = Upon detection of a CNG tone, the device immediately releases the call of the initial INVITE and then sends a new INVITE to a specific IP Group or fax server according to the Tel-to-IP Routing table. To enable this feature, set the CNGDetectorMode parameter to 2 and the IsFaxUsed parameter to 1, 2, or 3. ▪ [2] Progress and Delay = (Applicable only to ISDN). Incoming ISDN calls are delayed until a CNG tone detection or timeout, set by the FaxReroutingDelay parameter. If the EnableComfortTone parameter is set to 1, a Q.931 Progress message with Protocol Discriminator set to 1 is sent to the PSTN and a comfort tone is played accordingly to the PSTN. When the timeout expires, the device sends an INVITE to a specific IP Group or to a fax server, according to the Tel-to-IP Routing table rules. ▪ [3] Connect and Delay = (Applicable only to ISDN). Incoming ISDN calls are delayed until a CNG tone detection or timeout, set by the FaxReroutingDelay parameter. A Q.931 Connect message is sent to the PSTN. If the EnableComfortTone parameter is set to 1, a comfort tone is played to the PSTN. When the timeout expires, the device sends an INVITE to a specific IP Group or to a fax server according to the Tel-to-IP Routing table rules. **Note:** The parameter has replaced the EnableFaxRerouting parameter. For backward compatibility, the EnableFaxRerouting parameter set to 1 is equivalent to the FaxReroutingMode parameter set to 1.
[FaxReroutingDelay]	Defines the maximum time interval (in seconds) that the device waits for CNG detection before re-routing calls

Parameter	Description
	identified as fax calls to fax destinations (terminating fax machine). The valid value range is 1-10. The default is 5.
Call Forking Parameters	
Forking Handling Mode `forking-handling` [ForkingHandlingMode]	Determines how the device handles the receipt of multiple SIP 18x forking responses for Tel-to-IP calls. The forking 18x response is the response with a different SIP to-tag than the previous 18x response. These responses are typically generated (initiated) by Proxy / Application servers that perform call forking, sending the device's originating INVITE (received from SIP clients) to several destinations, using the same Call ID. ▪ **[0]** Parallel handling = (Default) If SIP 18x with SDP is received, the device opens a voice stream according to the received SDP and disregards any subsequently received 18x forking responses (with or without SDP). If the first response is 180 without SDP, the device responds according to the PlayRBTone2TEL parameter and disregards the subsequent forking 18x responses. ▪ **[1]** Sequential handling = If 18x with SDP is received, the device opens a voice stream according to the received SDP. The device re-opens the stream according to subsequently received 18x responses with SDP, or plays a ringback tone if 180 response without SDP is received. If the first received response is 180 without SDP, the device responds according to the PlayRBTone2TEL parameter and processes the subsequent 18x forking responses. **Note:** Regardless of the parameter setting, once a SIP 200 OK response is received, the device uses the RTP information and re-opens the voice stream, if necessary.
Forking Timeout `forking-timeout` [ForkingTimeOut]	Defines the timeout (in seconds) that is started after the first SIP 2xx response has been received for a User Agent when a Proxy server performs call forking (Proxy server forwards the INVITE to multiple SIP User Agents). The device sends a SIP ACK and BYE in response to any additional SIP 2xx received from the Proxy within this timeout. Once this timeout elapses, the device ignores any subsequent SIP 2xx. The number of supported forking calls per channel is 20. In other words, for an INVITE message, the device can receive up to 20 forking responses from the Proxy server. The valid range is 0 to 30. The default is 30.
Tel2IP Call Forking Mode `tel2ip-call-forking-mode` [Tel2IPCallForkingMode]	Enables Tel-to-IP call forking, whereby a Tel call can be routed to multiple IP destinations. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** Once enabled, routing rules must be assigned Forking Groups in the Tel-to-IP Routing table.
`configure voip/sip-definition advanced-`	Defines the interval (in seconds) to wait before sending INVITE messages to the other members of the forking group. The INVITE is immediately sent to the first member.

Parameter	Description
`settings/forking-delay-time-invite` [ForkingDelayTimeForInvite]	The valid value range is 0 to 40. The default is 0 (i.e., sends immediately).
Gateway Routing Policy Table	
Gateway Routing Policy `configure voip > gw routing gw-routing-policy` [GWRoutingPolicy]	Edits the Gateway Routing Policy. The format of the ini file table parameter is as follows: [GwRoutingPolicy] FORMAT GwRoutingPolicy_Index = GwRoutingPolicy_Name, GwRoutingPolicy_LCREnable, GwRoutingPolicy_LCRAverageCallLength, GwRoutingPolicy_LCRDefaultCost, GwRoutingPolicy_LdapServersGroupName; [\GwRoutingPolicy] For a description of the table, see "Configuring a Gateway Routing Policy Rule" on page 482.

57.10.11 IP Connectivity Parameters

The IP connectivity parameters are described in the table below.

Table 57-60: IP Connectivity Parameters

Parameter	Description
Enable Alt Routing Tel to IP `alt-routing-tel2ip` [AltRoutingTel2IPEnable]	Enables the Alternative Routing feature for Tel-to-IP calls. ▪ **[0]** Disable = (Default) Disables the Alternative Routing feature. ▪ **[1]** Enable = Enables the Alternative Routing feature. ▪ **[2]** Status Only = The Alternative Routing feature is disabled, but read-only information on the QoS of the destination IP addresses is provided. **Note:** If the parameter is enabled, the Busy Out feature (see EnableBusyOut parameter) does not function with the Proxy Set keep-alive mechanism. To use the Busy Out feature with the Proxy Set keep-alive mechanism (for IP Groups), disable the parameter.
Alt Routing Tel to IP Mode `alt-rte-tel2ip-mode` [AltRoutingTel2IPMode]	Determines the IP Connectivity event(s) reason for triggering Alternative Routing. ▪ **[0]** None = Alternative routing is not used. ▪ **[1]** Connectivity = Alternative routing is performed if SIP OPTIONS message to the initial destination fails (determined according to the AltRoutingTel2IPConnMethod parameter). ▪ **[2]** QoS = Alternative routing is performed if poor QoS is detected. ▪ **[3]** Both = (Default) Alternative routing is performed if either SIP OPTIONS to initial destination fails, poor QoS is detected, or the DNS host name is not resolved. **Notes:** ▪ QoS is quantified according to delay and packet loss calculated according to previous calls. QoS statistics are reset if no new data is received within two minutes.

Parameter	Description
	▪ To receive quality information (displayed in the 'Quality Status' and 'Quality Info.' fields in "Viewing IP Connectivity" on page 731) per destination, the parameter must be set to 2 or 3.
Alt Routing Tel to IP Connectivity Method `alt-rte-tel2ip-method` [AltRoutingTel2IPConnMethod]	Determines the method used by the device for periodically querying the connectivity status of a destination IP address. ▪ **[0]** ICMP Ping = (Default) Internet Control Message Protocol (ICMP) ping messages. ▪ **[1]** SIP OPTIONS = The remote destination is considered offline if the latest OPTIONS transaction timed out. Any response to an OPTIONS request, even if indicating an error, brings the connectivity status to online. **Note:** ICMP Ping is currently not supported for the IP Connectivity feature.
Alt Routing Tel to IP Keep Alive Time `alt-rte-tel2ip-keep-alive` [AltRoutingTel2IPKeepAliveTime]	Defines the time interval (in seconds) between SIP OPTIONS Keep-Alive messages used for the IP Connectivity application. The valid range is 5 to 2,000,000. The default is 60.
Max Allowed Packet Loss for Alt Routing [%] `mx-pkt-loss-4-alt-rte` [IPConnQoSMaxAllowedPL]	Defines the packet loss (in percentage) at which the IP connection is considered a failure and Alternative Routing mechanism is activated. The default is 20%.
Max Allowed Delay for Alt Routing `mx-all-dly-4-alt-rte` [IPConnQoSMaxAllowedDelay]	Defines the transmission delay (in msec) at which the IP connection is considered a failure and the Alternative Routing mechanism is activated. The range is 100 to 10,000. The default is 250.

57.10.12 Alternative Routing Parameters

The alternative routing parameters are described in the table below.

Table 57-61: Alternative Routing Parameters

Parameter	Description
3xx Use Alt Route Reasons `configure voip/sip-definition advanced-settings/3xx-use-alt-route` [UseAltRouteReasonsFor3xx]	Defines the handling of received SIP 3xx responses regarding call redirection to listed contacts in the Contact header. ▪ **[0]** No = (Default) Upon receipt of a 3xx response, the device tries each contact, one by one, listed in the Contact headers, until a successful destination is found. However, if a contact responds with a 486 or 600, the device does not try to redirect the call to next contact, and drops the call. ▪ **[1]** No if 6xx = Upon receipt of a 3xx response, the device tries each contact, one by one, listed in the Contact headers. However, if a 6xx Global Failure response is received during this process (e.g., 600 Busy Everywhere) the device does not try to redirect the call to the next contact, and drops the call. ▪ **[2]** Yes = Upon receipt of a 3xx response, the device redirects the call to the first contact listed in the Contact header. If the contact responds with a SIP response that is defined in the Reasons for Tel-to-IP Alternative Routing table, the device tries

Parameter	Description
	to redirect the call to the next contact, and so on. If a contact responds with a response that is not configured in the table, the device does not try to redirect the call to the next contact, and drops the call.
Redundant Routing Mode `redundant-routing-m` [RedundantRoutingMode]	Determines the type of redundant routing mechanism when a call can't be completed using the main route. ▪ **[0]** Disable = No redundant routing is used. If the call can't be completed using the main route (using the active Proxy or the first matching rule in the Routing table), the call is disconnected. ▪ **[1]** Routing Table = (Default) Internal routing table is used to locate a redundant route. ▪ **[2]** Proxy = Proxy list is used to locate a redundant route. **Note:** To implement the Redundant Routing Mode mechanism, you first need to configure the parameter AltRouteCauseTEL2IP (Reasons for Alternative Routing table).
[DisconnectCallwithPIifAlt]	Defines when the device sends the IP-to-Tel call to an alternative route (if configured) when it receives an ISDN Q.931 Disconnect message from the Tel side. ▪ [0] (Default) = The device forwards early media to the IP side if Disconnect includes PI, and disconnects the call when a Release message is received. Only after the call is disconnected does the device send the call to an alternative route. ▪ [1] = The device immediately sends the call to the alternative route. For more information, see Alternative Routing upon ISDN Disconnect on page 495.
[EnableAltMapTel2IP]	Enables different Tel-to-IP destination number manipulation rules per routing rule when several (up to three) Tel-to-IP routing rules are defined and if alternative routing using release causes is used. For example, if an INVITE message for a Tel-to-IP call is returned with a SIP 404 Not Found response, the call can be re-sent to a different destination number (as defined using the parameter NumberMapTel2IP). ▪ **[0]** = Disable (default) ▪ **[1]** = Enable
Reasons for Alternative Tel-to-IP Routing Table	
Reasons for Alternative Routing `configure voip > gw manipulations general-setting alt-route-cause-tel2ip` [AltRouteCauseTel2IP]	Defines SIP call failure reason values received from the IP side. If an IP call is released as a result of one of these reasons, the device attempts to locate an alternative IP route for the call in the Tel-to-IP Routing table (if a Proxy is not used) or used as a redundant Proxy (you need to set the parameter RedundantRoutingMode to 2). The release reason for Tel-to-IP calls is provided in SIP 4xx, 5xx, and 6xx response codes. The format of the ini file table parameter is as follows: [AltRouteCauseTel2IP] FORMAT AltRouteCauseTel2IP_Index = AltRouteCauseTel2IP_ReleaseCause; [\AltRouteCauseTel2IP] For example: AltRouteCauseTel2IP 0 = 486; (Busy Here) AltRouteCauseTel2IP 1 = 480; (Temporarily Unavailable) AltRouteCauseTel2IP 2 = 408; (No Response)

Parameter	Description
	For a detailed description of the table, see "Alternative Routing Based on SIP Responses" on page 487.
Reasons for Alternative IP-to-Tel Routing Table	
Reasons for Alternative IP-to-Tel Routing `configure voip > gw manipulations general-setting alt-route-cause-ip2tel` [AltRouteCauseIP2Tel]	Defines call failure reason values received from the Tel side (in Q.931 presentation). If a call is released as a result of one of these reasons, the device attempts to locate an alternative Trunk Group for the call in the IP to Trunk Group Routing table. The format of the ini file table parameter is as follows: [AltRouteCauseIP2Tel] FORMAT AltRouteCauseIP2Tel_Index = AltRouteCauseIP2Tel_ReleaseCause; [\AltRouteCauseIP2Tel] For example: AltRouteCauseIP2Tel 0 = 3 (No Route to Destination) AltRouteCauseIP2Tel 1 = 1 (Unallocated Number) AltRouteCauseIP2Tel 2 = 17 (Busy Here) AltRouteCauseIP2Tel 2 = 27 (Destination Out of Order) For a detailed description of the table, see "Alternative Routing to Trunk upon Q.931 Call Release Cause Code" on page 492.
Forward On Busy Trunk Destination Table	
Forward On Busy Trunk Destination `configure voip > gw routing fwd-on-bsy-trk-dest` [ForwardOnBusyTrunkDest]	Defines the Forward On Busy Trunk Destination table. This table allows you to define an alternative IP destination if a trunk is busy for IP-to-Tel calls. The format of the ini file table parameter is as follows: [ForwardOnBusyTrunkDest] FORMAT ForwardOnBusyTrunkDest_Index = ForwardOnBusyTrunkDest_TrunkGroupId, ForwardOnBusyTrunkDest_ForwardDestination; [\ForwardOnBusyTrunkDest] For example, the below configuration forwards IP-to-Tel calls to destination user “112” at host IP address 10.13.4.12, port 5060, using transport protocol TCP, if Trunk Group ID 2 is unavailable: ForwardOnBusyTrunkDest 1 = 2, 112@10.13.4.12:5060;transport=tcp; For a detailed description of the table, see "Alternative Routing to IP Destination upon Busy Trunk" on page 494.

57.10.13 Number Manipulation Parameters

The number manipulation parameters are described in the table below.

Table 57-62: Number Manipulation Parameters

Parameter	Description
[ManipulateIP2PSTNReferTo]	Enables the manipulation of the called party (destination) number according to the SIP Refer-To header received by the device for TDM (PSTN) blind transfer. The number in the SIP Refer-To header is manipulated for all types of blind transfers to the PSTN (TBCT, ECT, RLT, QSIG, FXO, and CAS). ▪ [0] Disable (default)

Parameter	Description
	▪ [1] Enable During the blind transfer, the device initiates a new call to the PSTN and the destination number of this call can be manipulated if the parameter is enabled. When enabled, the manipulation is done as follows: 1 If you configure a value for the xferPrefix parameter, then this value (string) is added as a prefix to the number in the Refer-To header. 2 This called party number is then manipulated using the IP-to-Tel Destination Phone Number Manipulation table. The source number of the transferred call is taken from the original call, according to its initial direction: ✓ Source number of the original call if it is a Tel-to-IP call ✓ Destination number of the original call if it is an IP-to-Tel call This source number can also be used as the value for the 'Source Prefix' field in the IP-to-Tel Destination Phone Number Manipulation table. The local IP address is used as the value for the 'Source IP Address' field. **Note:** This manipulation does not affect IP-to-Trunk Group routing rules.
Use EndPoint Number As Calling Number Tel2IP `epn-as-cpn-tel2ip` [UseEPNumAsCallingNumTel2IP]	Enables the use of the B-channel number as the calling number (sent in the From field of the INVITE) instead of the number received in the Q.931 Setup message, for Tel-to-IP calls. ▪ [0] Disable (default) ▪ [1] Enable For example, if the incoming calling party number in the Q.931 Setup message is "12345" and the B-channel number is 17, then the outgoing INVITE From header is set to "17" instead of "12345". **Note:** When enabled, this feature is applied before routing and manipulation on the source number.
Use EndPoint Number As Calling Number IP2Tel `epn-as-cpn-ip2tel` [UseEPNumAsCallingNumIP2Tel]	Enables the use of the B-channel number as the calling party number (sent in the Q.931 Setup message) instead of the number received in the From header of the INVITE, for IP-to-Tel calls. ▪ [0] Disable (default) ▪ [1] Enable For example, if the incoming INVITE From header contains "12345" and the destined B-channel number is 17, then the outgoing calling party number in the Q.931 Setup message is set to "17" instead of "12345". **Note:** When enabled, this feature is applied after routing and manipulation on the source number (i.e., just before sending to the Tel side).
Tel2IP Default Redirect Reason `tel-to-ip-dflt-redir-rsn` [Tel2IPDefaultRedirectReason]	Determines the default redirect reason for Tel-to-IP calls when no redirect reason (or “unknown”) exists in the received Q931 ISDN Setup message. The device includes this default redirect reason in the SIP History-Info header of the outgoing INVITE. If a redirect reason exists in the received Setup message, the parameter is ignored and the device sends the INVITE message with the reason according to the received Setup message. If the parameter is not configured (-1), the outgoing INVITE is sent with

Parameter	Description
	the redirect reason as received in the Setup message (if none or "unknown" reason, then without a reason). ▪ [-1] Not Configured = (Default) Received redirect reason is not changed ▪ [1] Busy = Call forwarding busy ▪ [2] No Reply = Call forwarding no reply ▪ [9] DTE Out of Order = Call forwarding DTE out of order ▪ [10] Deflection = Call deflection ▪ [15] Systematic/Unconditional = Call forward unconditional
Redirect Number SIP to TEL `redir-nb-si-2tel` [SetIp2TelRedirectScreeningInd]	Defines the value of the Redirect Number screening indicator in ISDN Setup messages. ▪ [-1] Not Configured (default) ▪ [0] User Provided ▪ [1] User Passed ▪ [2] User Failed ▪ [3] Network Provided
Set IP-to-TEL Redirect Reason `ip2tel-redir-reason` [SetIp2TelRedirectReason]	Defines the redirect reason for IP-to-Tel calls. If redirect (diversion) information is received from the IP, the redirect reason is set to the value of the parameter before the device sends it on to the Tel. ▪ [-1] Not Configured (default) ▪ [0] Unkown ▪ [1] Busy ▪ [2] No Reply ▪ [3] Network Busy ▪ [4] Deflection ▪ [9] DTE out of Order ▪ [10] Forwarding DTE ▪ [13] Transfer ▪ [14] PickUp ▪ [15] Systematic/Unconditional

Parameter	Description
Set TEL-to-IP Redirect Reason `tel2ip-redir-reason` [SetTel2IpRedirectReason]	Defines the redirect reason for Tel-to-IP calls. If redirect (diversion) information is received from the Tel, the redirect reason is set to the value of the parameter before the device sends it on to the IP. ▪ [-1] Not Configured (default) ▪ [0] Unkown ▪ [1] Busy ▪ [2] No Reply ▪ [3] Network Busy ▪ [4] Deflection ▪ [9] DTE out of Order ▪ [10] Forwarding DTE ▪ [13] Transfer ▪ [14] PickUp ▪ [15] Systematic/Unconditional
Send Screening Indicator to IP [ScreeningInd2IP]	Overrides the calling party's number (CPN) screening indication in the received ISDN SETUP message for Tel-to-IP calls. ▪ [-1] Not Configured = (Default) Not configured (interworking from ISDN to IP). ▪ [0] User Provided = CPN set by user, but not screened (verified). ▪ [1] User Passed = CPN set by user, verified and passed. ▪ [2] User Failed = CPN set by user, and verification failed. ▪ [3] Network Provided = CPN set by network. **Note:** The parameter is applicable only if the Remote Party ID (RPID) header is enabled.
Send Screening Indicator to ISDN [ScreeningInd2ISDN]	Overrides the screening indicator of the calling party's number for IP-to-Tel ISDN calls. ▪ [-1] Not Configured = (Default) Not configured (interworking from IP to ISDN). ▪ [0] User Provided = user provided, not screened. ▪ [1] User Passed = user provided, verified and passed. ▪ [2] User Failed = user provided, verified and failed. ▪ [3] Network Provided = network provided
Copy Destination Number to Redirect Number `cp-dst-nb-2-redir-nb` [CopyDest2RedirectNumber]	Enables the device to copy the received ISDN called number to the outgoing SIP Diversion header for Tel-to-IP calls (even if a Redirecting Number IE is not received in the ISDN Setup message). Therefore, the called number is used as a redirect number. Call redirection information is typically used for Unified Messaging and voice mail services to identify the recipient of a message. ▪ **[0]** Don't copy = (Default) Disable. ▪ **[1]** Copy after phone number manipulation = Copies the called number after manipulation. The device first performs Tel-to-IP destination phone number manipulation (i.e., on the SIP To header), and only then copies the manipulated called number to the SIP Diversion header for the Tel-to-IP call. Therefore, with this option, the called and redirect numbers are identical.

Parameter	Description
	▪ **[2]** Copy before phone number manipulation = Copies the called number before manipulation. The device first copies the original called number to the SIP Diversion header, and then performs Tel-to-IP destination phone number manipulation. Therefore, this allows you to have different numbers for the called (i.e., SIP To header) and redirect (i.e., SIP Diversion header) numbers. **Notes:** ▪ If the incoming ISDN-to-IP call includes a Redirect Number, this number is overridden by the new called number if the parameter is set to [1] or [2]. ▪ You can also use this feature for IP-to-Tel calls, by configuring the parameter per IP Profile (IpProfile_CopyDest2RedirectNum). For more information, see Configuring IP Profiles on page 387.
`rep-calling-w-redir disc-on-bsy-tone-i` [ReplaceCallingWithRedirectNumber]	Enables the replacement of the calling number with the redirect number for ISDN-to-IP calls. ▪ [0] = Disable (default) ▪ [1] = The calling name is removed and left blank. The outgoing INVITE message excludes the redirect number that was used to replace the calling number. The replacement is done only if a redirect number is present in the incoming Tel call. ▪ [2] = Manipulation is done on the new calling party number (after manipulation of the original calling party number, using the Tel2IPSourceNumberMappingDialPlanIndex parameter), but before the regular calling or redirect number manipulation: ✓ If a redirect number exists, it replaces the calling party number. If there is no redirect number, the calling number is left unchanged. ✓ If there is a calling “display” name, it remains unchanged. ✓ The redirect number remains unchanged and is included in the SIP Diversion header.
Add Trunk Group ID as Prefix `trkgrpid-prefix` [AddTrunkGroupAsPrefix]	Determines whether the Trunk Group ID is added as a prefix to the destination phone number (i.e., called number) for Tel-to-IP calls. ▪ **[0]** No = (Default) Don't add Trunk Group ID as prefix. ▪ **[1]** Yes = Add Trunk Group ID as prefix to called number. **Notes:** ▪ This option can be used to define various routing rules. ▪ To use this feature, you must configure the Trunk Group IDs (see Configuring Trunk Groups on page 433).
Add Trunk ID as Prefix `trk-id-as-prefix` [AddPortAsPrefix]	Determines whether or not the port numberTrunk ID is added as a prefix to the called (destination) number for Tel-to-IP calls. ▪ **[0]** No (Default) ▪ **[1]** Yes If enabled, the device adds the following prefix to the called phone number: port numberTrunk ID (single digit in the range 1 to 8). This option can be used to define various routing rules.
Add Trunk Group ID as Prefix to Source `trkgrpid-pref2source`	Determines whether the device adds the Trunk Group ID (from where the call originated) as the prefix to the calling number (i.e. source number).

Parameter	Description
[AddTrunkGroupAsPrefixToSource]	▪ **[0]** No (default) ▪ **[1]** Yes
Replace Empty Destination with B-channel Phone Number `empty-dst-w-bch-nb` [ReplaceEmptyDstWithPortNumber]	Determines whether the internal channel number is used as the destination number if the called number is missing. ▪ [0] No (default) ▪ [1] Yes **Note:** The parameter is applicable only to Tel-to-IP calls and if the called number is missing.
[CopyDestOnEmptySource]	Determines whether the destination number is copied to the source number if no source number is present, for Tel-to-IP calls. ▪ [0] = (Default) Source Number is left empty. ▪ [1] = If the Source Number of a Tel-to-IP call is empty, the Destination Number is copied to the Source Number.
Add NPI and TON to Calling Number `npi-n-ton-to-cng-nb` [AddNPIandTON2CallingNumber]	Determines whether the Numbering Plan Indicator (NPI) and Type of Numbering (TON) are added to the Calling Number for Tel-to-IP calls. ▪ [0] No = (Default) Do not change the Calling Number. ▪ [1] Yes = Add NPI and TON to the Calling Number ISDN Tel-to-IP call. For example: After receiving a Calling Number of 555, NPI of 1, and TON of 3, the modified number becomes 13555. This number can later be used for manipulation and routing.
Add NPI and TON to Called Number `npi-n-ton-to-cld-nb` [AddNPIandTON2CalledNumber]	Determines whether NPI and TON are added to the Called Number for Tel-to-IP calls. ▪ [0] No = (Default) Do not change the Called Number. ▪ [1] Yes = Add NPI and TON to the Called Number of ISDN Tel-to-IP call. For example: After receiving a Called Number of 555, NPI of 1 and TON of 3, the modified number becomes 13555. This number can later be used for manipulation and routing.
Add NPI and TON to Redirect Number `np-n-ton-2-redirnb` [AddNPIandTON2RedirectNumber]	Determines whether the NPI and TON values are added as the prefix to the Redirect number in INVITE messages' Diversion or History-Info headers, for ISDN Tel-to-IP calls. ▪ [0] Yes (Default) ▪ [1] No
IP to Tel Remove Routing Table Prefix `ip2tel-rmv-rte-tbl` [RemovePrefix]	Determines whether or not the device removes the prefix, as configured in the IP to Trunk Group Routing table (see "Configuring IP-to-Trunk Group Routing Rules" on page 476) from the destination number for IP-to-Tel calls, before sending it to the Tel. ▪ **[0]** No (default) ▪ **[1]** Yes For example: To route an incoming IP-to-Tel call with destination number "21100", the IP to Trunk Group Routing table is scanned for a matching prefix. If such a prefix is found (e.g., "21"), then before the call is routed to the corresponding Trunk Group, the prefix "21" is removed from the original number, and therefore, only "100" remains. **Notes:**

Parameter	Description
	▪ The parameter is applicable only if number manipulation is performed after call routing for IP-to-Tel calls (i.e., RouteModeIP2Tel parameter is set to 0). ▪ Similar operation (of removing the prefix) is also achieved by using the usual number manipulation rules.
Swap Redirect and Called Numbers `swap-rdr-n-called-nb` [SwapRedirectNumber]	▪ [0] No = (Default) Don't change numbers. ▪ [1] Yes = Incoming ISDN call that includes a redirect number (sometimes referred to as 'original called number') uses the redirect number instead of the called number.
[UseReferredByForCallingNumber]	Determines whether the device uses the number from the URI in the SIP Referred-By header as the calling number in the outgoing Q.931 Setup message, when SIP REFER messages are received. ▪ [0] = (Default) No ▪ [1] = Yes **Notes:** ▪ The parameter is applicable to all ISDN (TBCT, RLT, ECT) and blind call transfers (except for in-band) and when the device receives SIP REFER messages with a Referred-By header. ▪ This manipulation is done before regular IP-to-Tel source number manipulation.
[SwapTel2IPCalled&CallingNumbers]	Determines whether the device swaps the calling and called numbers received from the Tel side (for Tel-to-IP calls). The SIP INVITE message contains the swapped numbers. ▪ **[0]** = (Default) Disabled ▪ **[1]** = Swap calling and called numbers **Note:** The parameter can also be configured for a Tel Profile (in the Tel Profile table).
Add Prefix to Redirect Number `add-pref-to-redir-nb` [Prefix2RedirectNumber]	Defines a string prefix that is added to the Redirect number received from the Tel side. This prefix is added to the Redirect Number in the SIP Diversion header. The valid range is an 8-character string. By default, no value is defined.
Add Number Plan and Type to RPI Header `np-n-type-to-rpi-hdr` [AddTON2RPI]	Determines whether the TON/PLAN parameters are included in the Remote-Party-ID (RPID) header. ▪ **[0]** No ▪ **[1]** Yes (default) If the Remote-Party-ID header is enabled (EnableRPIHeader = 1) and AddTON2RPI = 1, it's possible to configure the calling and called number type and number plan using the Number Manipulation tables for Tel-to-IP calls.
Source Manipulation Mode `src-manipulation` [SourceManipulationMode]	Determines the SIP headers containing the source number after manipulation: ▪ **[0]** = (Default) The SIP From and P-Asserted-Identity headers contain the source number after manipulation. ▪ **[1]** = Only SIP From header contains the source number after manipulation, while the P-Asserted-Identity header contains the source number before manipulation.

Parameter	Description
Calling Name Manipulations IP-to-Tel Table	
`configure voip > gw manipulations calling-name-map-ip2tel` [CallingNameMapIp2Tel]	Configures rules for manipulating the calling name (caller ID) in the received SIP message for IP-to-Tel calls. This can include modifying or removing the calling name. The format of this table ini file parameter is as follows: [CallingNameMapIp2Tel] FORMAT CallingNameMapIp2Tel_Index = CallingNameMapIp2Tel_ManipulationName, CallingNameMapIp2Tel_DestinationPrefix, CallingNameMapIp2Tel_SourcePrefix, CallingNameMapIp2Tel_CallingNamePrefix, CallingNameMapIp2Tel_SourceAddress, CallingNameMapIp2Tel_RemoveFromLeft, CallingNameMapIp2Tel_RemoveFromRight, CallingNameMapIp2Tel_LeaveFromRight, CallingNameMapIp2Tel_Prefix2Add, CallingNameMapIp2Tel_Suffix2Add; [\CallingNameMapIp2Tel] For a detailed description of the table, see "Configuring SIP Calling Name Manipulation" on page 448.
Calling Name Manipulations Tel-to-IP Table	
`configure voip > gw manipulations calling-name-map-tel2ip` [CallingNameMapTel2Ip]	Defines rules for manipulating the calling name (caller ID) for Tel-to-IP calls. This can include modifying or removing the calling name. [CallingNameMapTel2Ip] FORMAT CallingNameMapTel2Ip_Index = CallingNameMapTel2Ip_ManipulationName, CallingNameMapTel2Ip_DestinationPrefix, CallingNameMapTel2Ip_SourcePrefix, CallingNameMapTel2Ip_CallingNamePrefix, CallingNameMapTel2Ip_SrcTrunkGroupID, CallingNameMapTel2Ip_RemoveFromLeft, CallingNameMapTel2Ip_RemoveFromRight, CallingNameMapTel2Ip_LeaveFromRight, CallingNameMapTel2Ip_Prefix2Add, CallingNameMapTel2Ip_Suffix2Add; [\CallingNameMapTel2Ip] For a detailed description of the table, see "Configuring SIP Calling Name Manipulation" on page 448.
Destination Phone Number Manipulation for IP-to-Tel Calls Table	
Destination Phone Number Manipulation Table for IP-to-Tel Calls `configure voip > gw manipulations NumberMapIp2Tel2` [NumberMapIP2Tel]	This table parameter manipulates the destination number of IP-to-Tel calls. The format of the ini file table parameter is as follows: [NumberMapIp2Tel] FORMAT NumberMapIp2Tel_Index = NumberMapIp2Tel_ManipulationName, NumberMapIp2Tel_DestinationPrefix, NumberMapIp2Tel_SourcePrefix, NumberMapIp2Tel_SourceAddress, NumberMapIp2Tel_NumberType, NumberMapIp2Tel_NumberPlan, NumberMapIp2Tel_RemoveFromLeft, NumberMapIp2Tel_RemoveFromRight, NumberMapIp2Tel_LeaveFromRight,

Parameter	Description
	NumberMapIp2Tel_Prefix2Add, NumberMapIp2Tel_Suffix2Add, NumberMapIp2Tel_IsPresentationRestricted; [\NumberMapIp2Tel] For a detailed description of the table, see "Configuring Source/Destination Number Manipulation" on page 441.
`prfm-ip-to-tel-dst-map` [PerformAdditionalIP2TELDestinationManipulation]	Enables additional destination number manipulation for IP-to-Tel calls. The additional manipulation is done on the initially manipulated destination number, and this additional rule is also configured in the manipulation table (NumberMapIP2Tel parameter). This enables you to configure only a few manipulation rules for complex number manipulation requirements (that generally require many rules). ▪ **[0]** = Disable (default) ▪ **[1]** = Enable
Destination Phone Number Manipulation for Tel-to-IP Calls Table	
Destination Phone Number Manipulation Table for Tel-to-IP Calls `configure voip > gw manipulations NumberMapTel2Ip` [NumberMapTel2IP]	This table parameter manipulates the destination number of Tel-to-IP calls. The format of the ini file table parameter is as follows: [NumberMapTel2Ip] FORMAT NumberMapTel2Ip_Index = NumberMapTel2Ip_ManipulationName, NumberMapTel2Ip_DestinationPrefix, NumberMapTel2Ip_SourcePrefix, NumberMapTel2Ip_SourceAddress, NumberMapTel2Ip_NumberType, NumberMapTel2Ip_NumberPlan, NumberMapTel2Ip_RemoveFromLeft, NumberMapTel2Ip_RemoveFromRight, NumberMapTel2Ip_LeaveFromRight, NumberMapTel2Ip_Prefix2Add, NumberMapTel2Ip_Suffix2Add, NumberMapTel2Ip_IsPresentationRestricted, NumberMapTel2Ip_SrcTrunkGroupID, NumberMapTel2Ip_ SrcIPGroupID; [\NumberMapTel2Ip] For a detailed description of the table, see "Configuring Source/Destination Number Manipulation" on page 441.
Source Phone Number Manipulation for IP-to-Tel Calls Table	
Source Phone Number Manipulation Table for IP-to-Tel Calls `configure voip > gw manipulations SourceNumberMapIp2Tel` [SourceNumberMapIP2Tel]	The p*arame*ter table manipulates the source number for IP-to-Tel calls. The format of the ini file table parameter is as follows: [SourceNumberMapIp2Tel] FORMAT SourceNumberMapIp2Tel_Index = SourceNumberMapIp2Tel_ManipulationName, SourceNumberMapIp2Tel_DestinationPrefix, SourceNumberMapIp2Tel_SourcePrefix, SourceNumberMapIp2Tel_SourceAddress, SourceNumberMapIp2Tel_NumberType, SourceNumberMapIp2Tel_NumberPlan, SourceNumberMapIp2Tel_RemoveFromLeft, SourceNumberMapIp2Tel_RemoveFromRight, SourceNumberMapIp2Tel_LeaveFromRight, SourceNumberMapIp2Tel_Prefix2Add, SourceNumberMapIp2Tel_Suffix2Add, SourceNumberMapIp2Tel_IsPresentationRestricted; [\SourceNumberMapIp2Tel]

Parameter	Description
	For a detailed description of the table, see "Configuring Source/Destination Number Manipulation" on page 441.
`prfm-ip-to-tel-src-map` [PerformAdditionalIP2TELSourceManipulation]	Enables additional source number manipulation for IP-to-Tel calls. The additional manipulation is done on the initially manipulated source number, and this additional rule is also configured in the manipulation table (SourceNumberMapIP2Tel parameter). This enables you to configure only a few manipulation rules for complex number manipulation requirements (that generally require many rules). ▪ **[0]** = Disable (default) ▪ **[1]** = Enable
Source Phone Number Manipulation for Tel-to-IP Calls Table	
Source Phone Number Manipulation Table for Tel-to-IP Calls `configure voip > gw manipulations SourceNumberMapTel2Ip` [SourceNumberMapTel2IP]	This table parameter manipulates the source phone number for Tel-to-IP calls. The format of the ini file table parameter is as follows: [SourceNumberMapTel2Ip] FORMAT SourceNumberMapTel2Ip_Index = SourceNumberMapTel2Ip_ManipulationName, SourceNumberMapTel2Ip_DestinationPrefix, SourceNumberMapTel2Ip_SourcePrefix, SourceNumberMapTel2Ip_NumberType, SourceNumberMapTel2Ip_NumberPlan, SourceNumberMapTel2Ip_RemoveFromLeft, SourceNumberMapTel2Ip_RemoveFromRight, SourceNumberMapTel2Ip_LeaveFromRight, SourceNumberMapTel2Ip_Prefix2Add, SourceNumberMapTel2Ip_Suffix2Add, SourceNumberMapTel2Ip_IsPresentationRestricted, SourceNumberMapTel2Ip_SrcTrunkGroupID; [\SourceNumberMapTel2Ip] For a detailed description of the table, see "Configuring Source/Destination Number Manipulation" on page 441.
Redirect Number IP -to-Tel Table	
Redirect Number IP -> Tel `configure voip > gw manipulations redirect-number-map-ip2tel` [RedirectNumberMapIp2Tel]	This table parameter manipulates the redirect number for IP-to-Tel calls. The format of the ini file table parameter is as follows: [RedirectNumberMapIp2Tel] FORMAT RedirectNumberMapIp2Tel_Index = RedirectNumberMapIp2Tel_ManipulationName, RedirectNumberMapIp2Tel_DestinationPrefix, RedirectNumberMapIp2Tel_RedirectPrefix, RedirectNumberMapIp2Tel_SourceAddress, RedirectNumberMapIp2Tel_SrcHost, RedirectNumberMapIp2Tel_DestHost, RedirectNumberMapIp2Tel_NumberType, RedirectNumberMapIp2Tel_NumberPlan, RedirectNumberMapIp2Tel_RemoveFromLeft, RedirectNumberMapIp2Tel_RemoveFromRight, RedirectNumberMapIp2Tel_LeaveFromRight, RedirectNumberMapIp2Tel_Prefix2Add, RedirectNumberMapIp2Tel_Suffix2Add, RedirectNumberMapIp2Tel_IsPresentationRestricted; [\RedirectNumberMapIp2Tel]

Parameter	Description
	For a description of the table, see Configuring Redirect Number Manipulation on page 451.
Redirect Number Tel-to-IP Table	
Redirect Number Tel -> IP `configure voip > gw manipulations redirect-number-map-tel2ip` [RedirectNumberMapTel2IP]	This table parameter manipulates the Redirect Number for Tel-to-IP calls. The format of the ini file table parameter is as follows: [RedirectNumberMapTel2Ip] FORMAT RedirectNumberMapTel2Ip_Index = RedirectNumberMapTel2Ip_ManipulationName, RedirectNumberMapTel2Ip_DestinationPrefix, RedirectNumberMapTel2Ip_RedirectPrefix, RedirectNumberMapTel2Ip_NumberType, RedirectNumberMapTel2Ip_NumberPlan, RedirectNumberMapTel2Ip_RemoveFromLeft, RedirectNumberMapTel2Ip_RemoveFromRight, RedirectNumberMapTel2Ip_LeaveFromRight, RedirectNumberMapTel2Ip_Prefix2Add, RedirectNumberMapTel2Ip_Suffix2Add, RedirectNumberMapTel2Ip_IsPresentationRestricted, RedirectNumberMapTel2Ip_SrcTrunkGroupID; [\RedirectNumberMapTel2Ip] For a description of the table, see "Configuring Redirect Number Manipulation" on page 451.
Phone Context Table	
Phone Context Table `configure voip > gw manipulations phone-context-table` [PhoneContext]	Defines the Phone Context table. The parameter maps NPI and TON to the SIP 'phone-context' parameter, and vice versa. The format for the parameter is as follows: [PhoneContext] FORMAT PhoneContext_Index = PhoneContext_Npi, PhoneContext_Ton, PhoneContext_Context; [\PhoneContext] For example: PhoneContext 0 = 0,0,unknown.com PhoneContext 1 = 1,1,host.com PhoneContext 2 = 9,1,na.e164.host.com For a detailed description of the table, see "Configuring NPI/TON-SIP Phone-Context Mapping Rules" on page 456.
Add Phone Context As Prefix `add-ph-cntxt-as-pref` [AddPhoneContextAsPrefix]	Determines whether the received Phone-Context parameter is added as a prefix to the outgoing ISDN Setup message withCalled and Calling numbers. ▪ **[0]** Disable (default) ▪ **[1]** Enable

57.11 SBC Parameters

The SBC and CRP parameters are described in the table below.

Table 57-63: SBC and CRP Parameters

Parameter	Description
CRP-specific Parameters	
CRP Application `enable-crp` [EnableCRPApplication]	Enables the CRP application. ▪ [0] Disable (default) ▪ [1] Enable **Note:** For the parameter to take effect, a device reset is required.
CRP Survivability Mode `crp-survivability-mode` [CRPSurvivabilityMode]	Defines the CRP mode. ▪ [0] Standard Mode (default) ▪ [1] Always Emergency Mode ▪ [2] Auto-answer REGISTER
`crp-gw-fallback` [CRPGatewayFallback]	Enables fallback routing from the proxy server to the Gateway (PSTN). ▪ [0] = Disable (default) ▪ [1] = Enable
SBC-specific Parameters	
Enable SBC `enable-sbc` [EnableSBCApplication]	Enables the Session Border Control (SBC) application. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ In addition to enabling the parameter, the number of maximum SBC/IP-to-IP sessions must be included in the Software License Key.
SBC and CRP Parameters	
Unclassified Calls `unclassified-calls` [AllowUnclassifiedCalls]	Determines whether incoming calls that cannot be classified (i.e. classification process fails) to a Source IP Group are rejected or processed. ▪ **[0]** Reject = (Default) Call is rejected if classification fails. ▪ **[1]** Allow = If classification fails, the incoming packet is assigned to a source IP Group (and subsequently processed) as follows: ✓ The source SRD is determined according to the SIP Interface to where the SIP-initiating dialog request is sent. The source IP Group is set to the default IP Group associated with this SRD. ✓ If the source SRD is ID 0, then source IP Group ID 0 is chosen. In case of any other SRD, then the first IP Group associated with this SRD is chosen as the source IP Group or the call. If no IP Group is associated with this SRD, the call is rejected.

Parameter	Description
SBC No Answer Timeout `sbc-no-arelt-timeout` [SBCAlertTimeout]	Defines the timeout (in seconds) for SBC outgoing (outbound IP routing) SIP INVITE messages. If the called IP party does not answer the call within this user-defined interval, the device disconnects the session. The device starts the timeout count upon receipt of a SIP 180 Ringing response from the called party. If no other SIP response (for example, 200 OK) is received thereafter within this timeout, the call is released. The valid range is 0 to 3600 seconds. the default is 600.
`configure voip/sbc general-setting/num-of-subscribes` [NumOfSubscribes]	Defines the maximum number of concurrent SIP SUBSCRIBE sessions permitted on the device. The valid value is any value between 0 and the maximum supported SUBSCRIBE sessions. When set to -1, the device uses the default value. For more information, contact your AudioCodes sales representative. **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ The maximum number of SUBSCRIBE sessions can be increased by reducing the maximum number of SBC channels in the Software License Key. For every reduced SBC session, the device gains two SUBSCRIBE sessions.
`configure voip/sbc general-setting/sbc-dialog-subsc-route-mode` [SBCInDialogSubscribeRouteMode]	Enables the device to route in-dialog, refresh SIP SUBSCRIBE requests to the "working" (has connectivity) proxy. ▪ **[0]** = (Default) Disable – the device sends in-dialog, refresh SUBSCRIBES according to the address in the Contact header of the 200 OK response received from the proxy to which the initial SUBSCRIBE was sent (as per the SIP standard). ▪ **[1]** = Enable – the device routes in-dialog, refresh SUBSCRIBES to the "working" proxy (regardless of the Contact header). The "working" proxy (address) is determined by the device's keep-alive mechanism for the Proxy Set that was used to route the initial SUBSCRIBE. **Note:** For this feature to be functional, ensure the following: ▪ Keep-alive mechanism is enabled for the Proxy Set ('Proxy Keep-Alive' parameter is set to any value other than **Disable**). ▪ Load-balancing between proxies is disabled ('Proxy Load Balancing Method' parameter is set to **Disable**).
`sbc-max-fwd-limit` [SBCMaxForwardsLimit]	Defines the Max-Forwards SIP header value. The Max-Forwards header is used to limit the number of servers (such as proxies) that can forward the SIP request. The Max-Forwards value indicates the remaining number of times this request message is allowed to be forwarded. This count is decremented by each server that forwards the request. The parameter affects the Max-Forwards header in the received message as follows:

Parameter	Description
	▪ If the received header's original value is 0, the message is not passed on and is rejected. ▪ If the received header's original value is less than the parameter's value, the header's value is decremented before being sent on. ▪ If the received header's original value is greater than the parameter's value, the header's value is replaced by the user-defined parameter's value. The valid value range is 1-70. The default is 10.
SBC Session-Expires `sbc-sess-exp-time` [SBCSessionExpires]	Defines the SBC session refresh timer (in seconds) in the Session-Expires header of outgoing INVITE messages. The valid value range is 90 (according to RFC 4028) to 86400. The default is 180.
Minimum Session-Expires `min-session-expires` [SBCMinSE]	Defines the minimum amount of time (in seconds) between session refresh requests in a dialog before the session is considered timed out. This value is conveyed in the SIP Min-SE header. The valid range is 0 (default) to 1,000,000, where 0 means that the device does not limit Session-Expires.
`configure voip/sbc general-setting/sbc-session-refresh-policy` [SBCSessionRefreshingPolicy]	Defines the SIP user agent responsible for periodically sending refresh requests for established sessions (active calls). The session refresh allows SIP UAs or proxies to determine the status of the SIP session. When a session expires, the session is considered terminated by the UAs, regardless of whether a SIP BYE was sent by one of the UAs. The SIP Session-Expires header conveys the lifetime of the session, which is sent in re-INVITE or UPDATE requests (session refresh requests). The 'refresher=' parameter in the Session-Expires header (sent in the initial INVITE or subsequent 2xx response) indicates who sends the session refresh requests. If the parameter contains the value 'uac', the device performs the refreshes; if the parameter contains the value 'uas', the remote proxy performs the refreshes. An example of the Session-Expires header is shown below: `Session-Expires: 4000;refresher=uac` Thus, the parameter is useful when a UA does not support session refresh requests or does not support the indication of who performs session refresh requests. In such a scenario, the device can be configured to perform the session refresh requests. ▪ **[0]** Remote Refresher = (Default) The UA (proxy) performs the session refresh requests. The device indicates this to the UA by sending the SIP message with the 'refresher=' parameter in the Session-Expires header set to 'uas'. ▪ **[1]** SBC Refresher = The device performs the session refresh requests. The device indicates this to the UA by sending the SIP message with the 'refresher=' parameter in the Session-Expires header set to 'uac'. **Note:** The time values of the Session-Expires (session refresh interval) and Min-SE (minimum session refresh

Parameter	Description
	interval) headers can be configured using the SBCSessionExpires and SBCMinSE parameters, respectively.
User Registration Grace Time `configure voip/sbc general-setting/sbc-usr-reg-grace-time` [SBCUserRegistrationGraceTime]	Defines additional time (in seconds) to add to the registration expiry time users that are registered in the device's Users Registration database. The valid value is 0 to 2,000,000. The default is 0. For more information, see Registration Refreshes on page 535.
SBC DB Routing Search Mode configure voip > sbc general-setting > set sbc-db-route-mode [SBCDBRoutingSearchMode]	Defines the method for searching a registered user in the device's User Registration database when a SIP INVITE message is received for routing to a user. If the registered user is found (i.e., destination URI in INVITE), the device routes the call to the user's corresponding contact address specified in the database. ▪ [0] All permutations = (Default) Device searches for the user in the database using the entire Request-URI (user@host). If not found, it searches for the user part of the Request-URI. For example, it first searches for "4709@joe.company.com" and if not found, it searches for "4709". ▪ [1] Dest URI dependant = Device searches for the user in the database using the entire Request-URI (user@host) only. For example, it searches for "4709@joe.company.com". **Note:** If the Request-URI contains the "tel:" URI or "user=phone" parameter, the device searches only for the user part.
Handle P-Asserted-Identity `p-assert-id` [SBCAssertIdentity]	Global parameter that defines the handling of the SIP P-Asserted-Identity header. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_SBCAssertIdentity). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
Keep original user in Register [SBCKeepContactUserinRegister]	Defines the device's handling of the SIP Contact header in REGISTER requests which it forwards as the outgoing message. ▪ [0] Do not keep user; override with unique identifier = (Default) The device replaces the user part of the Contact header with a unique value, for example: ✓ Incoming Contact Header: <sip:123@domain.com> ✓ Outgoing Contact Header: <sip:FEU1-7-1-3@SBC> ▪ [1] keep user without unique identifier = The device retains the original user part value of the Contact header in the outgoing REGISTER request. ▪ [2] Keep user; add unique identifier as URI parameter = The device retains the original user part value of the

Parameter	Description
	Contact header in the outgoing REGISTER request. In addition, it adds the special URI parameter "ac-feu=<identifier>" to the Contact header, which is used to differentiate between two SIP entities with the same user part. The identifier value is generated by the device. ✓ Incoming Contact Header: <sip:123@domain.com> ✓ Outgoing Contact Header: <sip:123@SBC;ac-feu=1-7-1-3> **Note:** ▪ The parameter is applicable only to REGISTER messages received from User-type IP Groups which are sent to Server-type IP Groups. ▪ Depending on the 'Remote Representation Mode' parameter of the IP Profiles table (IpProfile_SBCRemoteRepresentationMode), the host part in the SIP Contact header can replaced by the device's IP address or by the value of the 'SIP Group Name' parameter (configured in the IP Groups table).
SBC Remote Refer Behavior `sbc-refer-bhvr` [SBCReferBehavior]	Global parameter that defines the handling of SIP REFER requests. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_SBCRemoteReferBehavior). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
`sbc-xfer-prefix` [SBCXferPrefix]	When the SBCReferBehavior is set to 1, the device, while interworking the SIP REFER message, adds the prefix "T~&R-" to the user part of the URI in the Refer-To header. After this, the device can receive an INVITE with such a prefix (the INVITE is sent by the UA that receives the REFER message or 302 response). If the device receives an INVITE with such a prefix, it replaces the prefix with the value defined for the SBCXferPrefix parameter. By default, no value is defined. **Note:** This feature is also applicable to 3xx redirect responses. The device adds the prefix "T~&R-" to the URI user part in the Contact header if the SBC3xxBehavior parameter is set to 1.
`sbc-3xx-bhvt` [SBC3xxBehavior]	Global parameter that defines the handling of SIP 3xx redirect responses. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_SBCRemote3xxBehavior). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387.

Parameter	Description
	Note: If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.

[SBCEnforceMediaOrder]	Enables the device to include all previously negotiated media lines within the current session ('m=' line) in the SDP offer-answer exchange (RFC 3264).
	▪ **[0]** Disable (default)
	▪ **[1]** Enable
	For example, assume a call (audio) has been established between two endpoints and one endpoint wants to subsequently send an image in the same call session. If the parameter is enabled, the endpoint includes the previously negotiated media type (i.e., audio) with the new negotiated media type (i.e., image) in its SDP offer:
	`v=0`
	`o=bob 2890844730 2890844731 IN IP4 host.example.com`
	`s=`
	`c=IN IP4 host.example.com`
	`t=0 0`
	`m=audio 0 RTP/AVP 0`
	`m=image 12345 udptl t38`
	If the parameter is disabled, the only ‘m=’ line included in the SDP is the newly negotiated media (i.e., image).

SBC Diversion URI Type `sbc-diversion-uri-type (configure voip > sbc general-setting)` [SBCDiversionUriType]	Defines the URI type to use in the SIP Diversion header of the outgoing SIP message. ▪ **[0]** Transparent = (Default) The device does not change the URI and leaves it as is. ▪ **[1]** Sip = The "sip" URI is used. ▪ **[2]** Tel = The "tel" URI is used. **Note:** The parameter is applicable only if the Diversion header is used. The SBCDiversionMode and SBCHistoryInfoMode parameters in the IP Profile table determine the call redirection (diversion) SIP header to use - History-Info or Diversion.
SBC Server Auth Mode `sbc-server-auth-mode` [SBCServerAuthMode]	Defines whether authentication of the SIP client is done locally (by the device) or by a RADIUS server. ▪ **[0]** (default) = Authentication is done by the device (locally). ▪ **[1]** = Authentication is done by the RFC 5090 compliant RADIUS server ▪ **[2]** = Authentication is done according to the Draft Sterman-aaa-sip-01 method. **Note:** Currently, option [1] is not supported.
Lifetime of the nonce in seconds `lifetime-of-nonce` [AuthNonceDuration]	Defines the lifetime (in seconds) that the current nonce is valid for server-based authentication. The device challenges a message that attempts to use a server nonce beyond this period. The parameter is used to provide replay protection (i.e., ensures that old communication streams are not used in replay attacks). The valid value range is 30 to 600. The default is 300.

Parameter	Description
Authentication Challenge Method `auth-chlng-mthd` [AuthChallengeMethod]	Defines the type of server-based authentication challenge. ▪ **[0]** 0 = (Default) Send SIP 401 "Unauthorized" with a WWW-Authenticate header as the authentication challenge response. ▪ **[1]** 1 = Send SIP 407 "Proxy Authentication Required" with a Proxy-Authenticate header as the authentication challenge response.
Authentication Quality of Protection `auth-qop` [AuthQOP]	Defines the authentication and integrity level of quality of protection (QoP) for digest authentication offered to the client. When the device challenges a SIP request (e.g., INVITE), it sends a SIP 401 response with the Proxy-Authenticate header or WWW-Authenticate header containing the 'qop' parameter. The QoP offered in the 401 response can be 'auth', 'auth-int', both 'auth' and 'auth-int', or the 'qop' parameter can be omitted from the 401 response. In response to the 401, the client needs to send the device another INVITE with the MD5 hash of the INVITE message and indicate the selected auth type. ▪ **[0]** 0 = The device sends 'qop=auth' in the SIP response, requesting authentication (i.e., validates user by checking user name and password). This option does not authenticate the message body (i.e., SDP). ▪ **[1]** 1 = The device sends 'qop=auth-int' in the SIP response, indicating required authentication and authentication with integrity (e.g., checksum). This option restricts the client to authenticating the entire SIP message, including the body, if present. ▪ **[2]** 2 = (Default) The device sends 'qop=auth, auth-int' in the SIP response, indicating either authentication or integrity. This enables the client to choose 'auth' or 'auth-int'. If the client chooses 'auth-int', then the body is included in the authentication. If the client chooses 'auth', then the body is not authenticated. ▪ **[3]** 3 = No 'qop' parameter is offered in the SIP 401 challenge message.
SBC User Registration Time `sbc-usr-rgstr-time` [SBCUserRegistrationTime]	Global parameter that defines the duration (in seconds) of the periodic registrations that occur between the user and the device (the device responds with this value to the user). You can also configure this functionality per specific calls, using IP Profiles (IpProfile_SBCUserRegistrationTime). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
SBC Proxy Registration Time `sbc-prxy-rgstr-time` [SBCProxyRegistrationTime]	Defines the duration (in seconds) for which the user is registered in the proxy database (after the device forwards the REGISTER message). This value is sent in the Expires header. When set to 0, the device sends the Expires header's value as received from the user to the proxy.

Parameter	Description
	The valid range is 0 to 2,000,000 seconds. The default is 0.
`config-voip>sbc general-setting sbc-rand-expire` [SBCRandomizeExpires]	Defines a value (in seconds) that is used to calculate a new value for the expiry time in the Expires header of SIP 200 OK responses for user registration and subscription requests from users. The expiry time value appears in the Expires header in REGISTER and SUBSCRIBE SIP messages. When the device receives such a request from a user, it forwards it to the proxy or registrar server. Upon a successful registration or subscription, the server sends a SIP 200 OK response. If the expiry time was unchanged by the server, the device applies this feature and changes the expiry time in the SIP 200 OK response before forwarding it to the user; otherwise, the device does not change the expiry time. This feature is useful in scenarios where multiple users may refresh their registration or subscription simultaneously, thereby causing the device to handle many such sessions at a given time. This may result in an overload of the device (reaching maximum session capacity), thereby preventing the establishment of new calls or preventing the handling of some user registration or subscription requests. When this feature is enabled, the device assigns a random expiry time to each user registration or subscription and thus, ensuring future user registration and subscription requests are more distributed over time (i.e., do not all occur simultaneously). The device takes any random number between 0 and the value configured by the parameter, and then subtracts this random number from the original expiry time value. For example, assume that the original expiry time is 120 and the parameter is set to 10. If the device randomly chooses the number 5 (i.e., between 0 and 10), the resultant expiry time will be 115 (120 minus 5). The valid value is 0 to 20. The default is 10. If set to 0, the device does not change the expiry time. **Notes:** ▪ The lowest expiry time that the device sends in the 200 OK, regardless of the resultant calculation, is 10 seconds. For example, if the original expiry time is 12 seconds and the parameter is set to 5, theoretically, the new expiry time can be less than 10 (e.g., 12 – 4 = 8). However, the expiry time will be set to 10. ▪ The expiry time received from the user can be changed by the device before forwarding it to the proxy. This is configured by the SBCUserRegistrationTime parameter.
SBC Survivability Registration Time `sbc-surv-rgstr-time` [SBCSurvivabilityRegistrationTime]	Defines the duration of the periodic registrations between the user and the device, when the device is in survivability state (i.e., when REGISTER requests cannot be forwarded to the proxy and are terminated by the device). When set to 0, the device uses the value set by the

Parameter	Description
	SBCUserRegistrationTime parameter for the device's response. The valid range is 0 to 2,000,000 seconds. The default is 0.

[SBCEnableSurvivabilityNotice]	Enables the device to notify Aastra IP phones that the device is currently operating in Survivability mode. When this occurs, the Aastra IP phones display the message, "Stand Alone Mode" on their LCD screens. Survivability mode occurs when connectivity with the WAN fails and as a result, the device enables communication between IP phone users within the LAN enterprise.
	▪ **[0]** = Disable
	▪ **[1]** = Enable
	When this feature is enabled and the SBC device is in Survivability mode, it responds to SIP REGISTER messages from the IP phones with a SIP 200 OK containing the following XML body:
	`Content-Type: application/xml`
	`<?xml version="1.0" encoding="utf-8"?>`
	`<LMIDocument version="1.0">`
	`<LocalModeStatus>`
	
	`<LocalModeActive>true</LocalModeActive>`
	`   <LocalModeDisplay>StandAlone Mode</LocalModeDisplay>`
	`</LocalModeStatus>`
	`</LMIDocument>`

SBC Dialog-Info Interworking `configure voip/sbc general-setting/sbc-dialog-info-interwork` [EnableSBCDialogInfoInterworking]	Enables the interworking of dialog information (parsing of call identifiers in XML body) in SIP NOTIFY messages received from a remote application server. ▪ **[0]** Disable (default) ▪ **[1]** Enable For more information, see "Interworking Dialog Information in SIP NOTIFY Messages" on page 557.
`sbc-keep-call-id` [SBCKeepOriginalCallId]	Enables the device to use the same call identification value received in incoming messages for the call identification in outgoing messages. The call identification value is contained in the SIP Call-ID header. ▪ **[0]** = (Default) Disable - the device creates a new Call-ID value for the outgoing message. ▪ **[1]** = Enable - the device uses the received Call-ID value of the incoming message in the outgoing message. **Note:** When the device sends an INVITE as a result of a REFER/3xx termination, the device always creates a new Call-ID value and ignores the parameter's settings.
SBC GRUU Mode `sbc-gruu-mode` [SBCGruuMode]	Determines the Globally Routable User Agent (UA) URI (GRUU) support, according to RFC 5627. ▪ **[0]** None = No GRUU is supplied to users.

Parameter	Description

	▪ **[1]** As Proxy = (Default) The device provides same GRUU types as the proxy provided the device's GRUU clients.
	▪ **[2]** Temporary only = Supply only temporary GRUU to users. (Currently not supported.)
	▪ **[3]** Public only = The device provides only public GRUU to users.
	▪ **[4]** Both = The device provides temporary and public GRUU to users. (Currently not supported.)
	The parameter allows the device to act as a GRUU server for its SIP UA clients, providing them with public GRUU's, according to RFC 5627. The public GRUU provided to the client is denoted in the SIP Contact header parameters, "pub-gruu". Public GRUU remains the same over registration expirations. On the other SBC leg communicating with the Proxy/Registrar, the device acts as a GRUU client.
	The device creates a GRUU value for each of its registered clients, which is mapped to the GRUU value received from the Proxy server. In other words, the created GRUU value is only used between the device and its clients (endpoints).
	`Public-GRUU:`
	`sip:userA@domain.com;gr=unique-id`

Bye Authentication `sbc-bye-auth` [SBCEnableByeAuthentication]	Enables authenticating a SIP BYE request before disconnecting the call. This feature prevents, for example, a scenario in which the SBC SIP client receives a BYE request from a third-party imposer assuming the identity of a participant in the call and as a consequence, the call between the first and second parties is inappropriately disconnected. ▪ **[0]** Disable (default) ▪ **[1]** Enable = The device forwards the SIP authentication response (for the BYE request) to the request sender and waits for the user to authenticate it. The call is disconnected only if the authenticating server responds with a 200 OK.
SBC Enable Subscribe Trying `configure voip > sbc general-setting > set sbc-subs-try` [SBCSendTryingToSubscribe]	Enables the device to send SIP 100 Trying responses upon receipt of SUBSCRIBE or NOTIFY messages. ▪ **[0]** Disable (Default) ▪ **[1]** Enable
[SBCExtensionsProvisioningMode]	Enables SBC user registration for interoperability with BroadSoft's BroadWorks server, to provide call survivability in case of connectivity failure with the BroadWorks server. ▪ **[0]** = (Default) Normal processing of REGISTER messages. ▪ **[1]** = Registration method for BroadWorks server. In a failure scenario with BroadWorks, the device acts as a backup SIP proxy server, maintaining call continuity between the enterprise LAN users (subscribers) and between the subscribers and the PSTN (if provided).

Parameter	Description
	Note: For a detailed description of this feature, see "Auto-Provisioning of Subscriber-Specific Information for BroadWorks Server" on page 620.
SBC Direct Media `sbc-direct-media` [SBCDirectMedia]	Enables the Direct Media feature (i.e., no Media Anchoring) for all SBC calls, whereby SIP signaling is handled by the device without handling the RTP/SRTP (media) flow between the user agents (UA). The RTP packets do not traverse the device. Instead, the two SIP UAs establish a direct RTP/SRTP flow between one another. Signaling continues to traverse the device with minimal intermediation and involvement to enable certain SBC abilities such as routing ▪ **[0]** Disable = (Default) All calls traverse the device (i.e., no direct media). ▪ **[1]** Enable = Direct media flow between endpoints for all SBC calls. **Notes:** ▪ The setting of direct media in the SIP Interface table overrides this global parameter. In other words, even if the parameter is disabled for direct media (i.e., Media Anchoring is enabled), if direct media is enabled for a SIP Interface (in the SIP Interface table), calls between endpoints belonging to the SIP Interface employ direct media. ▪ For more information on No Media Anchoring, see "Direct Media" on page 538.
SBC RTCP Mode `sbc-rtcp-mode` [SBCRTCPMode]	Global parameter that defines the handling of RTCP packets. You can also configure this functionality per specific calls, using IP Profiles (IPProfile_SBCRTCPMode). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
SBC Send Invite To All Contacts `sbc-send-invite-to-all-contacts` [SBCSendInviteToAllContacts]	Enables call forking of INVITE message received with a Request-URI of a specific contact registered in the device's database, to all users under the same AOR as the contact. ▪ **[0]** Disable (default) = Sends the INVITE only to the contact of the received Request-URI. ▪ **[1]** Enable To configure call forking initiated by the device, see "Initiating SIP Call Forking" on page 619.
SBC Shared Line Registration Mode `sbc-shared-line-reg-mode` [SBCSharedLineRegMode]	Enables the termination on the device of SIP REGISTER messages from secondary lines that belong to the Shared Line feature. ▪ **[0]** Disable = (Default) Device forwards the REGISTER messages as is (i.e., not terminated on the device). ▪ **[1]** Enable = REGISTER messages of secondary lines are terminated on the device.

Parameter	Description
	Note: The device always forwards REGISTER messages of the primary line.
SBC Forking Handling Mode `sbc-forking-handling-mode` [SBCForkingHandlingMode]	Defines the handling of SIP 18x responses that are received due to call forking of an INVITE. ▪ **[0]** Latch On First = (Default) Only the first 18x is forwarded to the INVITE-initiating UA. If SIP 18x with SDP is received, the device opens a voice stream according to the received SDP and disregards any subsequent 18x forking responses (with or without SDP). If the first response is 180 without SDP, the device sends it to the other side. ▪ **[1]** Sequential = All 18x responses are forwarded, one at a time (sequentially) to the INVITE-initiating UA. If a 18x arrives with an offer only, then only the first offer is forwarded to the INVITE-initiating UA and subsequent 18x responses are discarded.
Gateway Direct Route Prefix `configure voip/sbc general-setting/gw-direct-route-prefix` [GWDirectRoutePrefix]	Defines the prefix destination Request-URI user part that is appended to the original user part for alternative IP-to-IP call routing from SBC to Gateway (Tel) interfaces. The valid value is a string of up to 16 characters. The default is "acgateway-<original prefix destination number>". For example, "acgateway-200". For more information, see Configuring SBC IP-to-IP Routing Rules on page 578.
`sbc-media-sync` [EnableSBCMediaSync]	Enables synchronization of media between two SIP user agents when a call is established between them. Media synchronization means that the media is properly negotiated (SDP offer/answer) between the user agents. In some scenarios, the call is established despite the media not being synchronized. This may occur, for example, in call transfer (SIP REFER) where the media between the transfer target and transferee are not synchronized. The device performs media synchronization by sending a re-INVITE immediately after the call is established in order for the user agents to negotiate the media (SDP offer/answer). ▪ [0] Disable = (Default) Media synchronization is performed only if the RTP mode (e.g., a=sendrecv, a=sendrecv, a=sendonly, a=recvonly, and a=inactive) between the user agents are different and synchronization is required. ▪ [1] Enable = Media synchronization is performed if the media, including RTP mode or any other media such as coders, is different and has not been negotiated between the user agents. ▪ [2] Never = Media synchronization is never performed.
[SBCRemoveSIPSFromNonSecuredTransport] `configure voip > sbc settings > sbc-remove-sips-non-sec-transp`	Defines the SIP headers for which the device replaces “sips:” with “sip:” in the outgoing SIP-initiating dialog request (e.g., INVITE) when the destination transport type is unsecured (e.g., UDP). (The “sips:” URI scheme indicates secured transport, for example, TLS.)

Parameter	Description
	- [0] = (Default) The device replaces “sips:” with “sip:” for the Request-URI and Contact headers only (and retains “sips:” for all other headers). - [1] = The device replaces “sips:” with “sip:” for the Request-URI, Contact, From, To, P-Asserted, P-Preferred, and Route headers.
Admission Control Table	
Admission Control `configure voip > sbc sbc-admission-control` [SBCAdmissionControl]	Defines Call Admission Control (CAC) rules. The format of the ini file table parameter is as follows: [SBCAdmissionControl] FORMAT SBCAdmissionControl_Index = SBCAdmissionControl_AdmissionControlName, SBCAdmissionControl_LimitType, SBCAdmissionControl_IPGroupName, SBCAdmissionControl_SRDName, SBCAdmissionControl_SIPInterfaceName, SBCAdmissionControl_RequestType, SBCAdmissionControl_RequestDirection, SBCAdmissionControl_Limit, SBCAdmissionControl_LimitPerUser, SBCAdmissionControl_Rate, SBCAdmissionControl_MaxBurst, SBCAdmissionControl_Reservation; [\SBCAdmissionControl] For a description of the table, see "Configuring Admission Control" on page 561.
Allowed Audio Coders Table	
Allowed Audio Coders `configure voip > sbc allowed-coders-group allowedcodersgroup0` [AllowedCodersGroupX]	Defines Allowed Coders Groups, which determine the audio (voice) coders that can be used for a specific SIP entity. The format of the ini file table parameter is as follows: [AllowedCodersGroupX] FORMAT AllowedCodersGroup_Index = AllowedCodersGroup_Name; [\AllowedCodersGroup] Where *X* represents the index number. For a description of the table, see "Configuring Allowed Audio Coder Groups" on page 565.
Allowed Video Coders Table	
`configure voip/sbc allowed-video-coders-group group-X` [AllowedVideoCodersGroupX]	Defines Allowed Video Coders Groups, which determine the video coders that can be used for a specific SIP entity. The format of the ini file table parameter is as follows: [AllowedVideoCodersGroup0] FORMAT AllowedVideoCodersGroup_Index = AllowedVideoCodersGroup_Name; [\AllowedVideoCodersGroup] Where *X* represents the index number. For a description of the table, see "Configuring Allowed Video Coder Groups" on page 566.

Parameter	Description
Classification Table	
Classification Table `configure voip > sbc routing classification` [Classification]	Defines call Classification rules. The format of the ini file table parameter is as follows: [Classification] FORMAT Classification_Index = Classification_ClassificationName, Classification_MessageConditionName, Classification_SRDName, Classification_SrcSIPInterfaceName, Classification_SrcAddress, Classification_SrcPort, Classification_SrcTransportType, Classification_SrcUsernamePrefix, Classification_SrcHost, Classification_DestUsernamePrefix, Classification_DestHost, Classification_ActionType, Classification_SrcIPGroupName, Classification_DestRoutingPolicy, Classification_IpProfileName; [\Classification] For a description of the table, see "Configuring Classification Rules" on page 569.
Condition Table	
Condition Table `configure voip > sbc routing condition-table` [ConditionTable]	Defines SIP Message Condition rules. [ConditionTable] FORMAT ConditionTable_Index = ConditionTable_Condition, ConditionTable_Description; [\ConditionTable] For a description of the table, see "Configuring Message Condition Rules" on page 576.
SBC IP-to-IP Routing Table	
IP-to-IP Routing Table `configure voip > sbc routing ip2ip-routing` [IP2IPRouting]	Defines SBC IP-to-IP routing rules. The format of the ini file table parameter is as follows: [IP2IPRouting] FORMAT IP2IPRouting_Index = IP2IPRouting_RouteName, IP2IPRouting_RoutingPolicyName, IP2IPRouting_SrcIPGroupName, IP2IPRouting_SrcUsernamePrefix, IP2IPRouting_SrcHost, IP2IPRouting_DestUsernamePrefix, IP2IPRouting_DestHost, IP2IPRouting_RequestType, IP2IPRouting_MessageConditionName, IP2IPRouting_ReRouteIPGroupName, IP2IPRouting_Trigger, IP2IPRouting_CallSetupRulesSetId, IP2IPRouting_DestType, IP2IPRouting_DestIPGroupName, IP2IPRouting_DestSIPInterfaceName, IP2IPRouting_DestAddress, IP2IPRouting_DestPort, IP2IPRouting_DestTransportType, IP2IPRouting_AltRouteOptions, IP2IPRouting_GroupPolicy, IP2IPRouting_CostGroup,

Parameter	Description
	IP2IPRouting_DestTags, IP2IPRouting_SrcTags; [\IP2IPRouting] For a description of the table, see "Configuring SBC IP-to-IP Routing Rules" on page 578.
SBC Alternative Routing Reasons Table	
SBC Alternative Routing Reasons `configure voip > sbc routing sbc-alternative-routing-reasons` [SBCAlternativeRoutingReasons]	Defines SBC alternative routing reason rules. The format of the ini file table parameter is as follows: [SBCAlternativeRoutingReasons] FORMAT SBCAlternativeRoutingReasons_Index = SBCAlternativeRoutingReasons_ReleaseCause; [\SBCAlternativeRoutingReasons] For a description of the table, see "Configuring SIP Response Codes for Alternative Routing Reasons" on page 588.
IP to IP Inbound Manipulation Table	
IP to IP Inbound Manipulation `configure voip > sbc manipulations ip-inbound-manipulation` [IPInboundManipulation]	Defines IP-to-IP inbound manipulation rules. The format of the ini file table parameter is as follows: [IPInboundManipulation] FORMAT IPInboundManipulation_Index = IPInboundManipulation_ManipulationName IPInboundManipulation_IsAdditionalManipulation, IPInboundManipulation_ManipulatedURI, IPInboundManipulation_ManipulationPurpose, IPInboundManipulation_SrcIPGroupName, IPInboundManipulation_SrcUsernamePrefix, IPInboundManipulation_SrcHost, IPInboundManipulation_DestUsernamePrefix, IPInboundManipulation_DestHost, IPInboundManipulation_RequestType, IPInboundManipulation_RemoveFromLeft, IPInboundManipulation_RemoveFromRight, IPInboundManipulation_LeaveFromRight, IPInboundManipulation_Prefix2Add, IPInboundManipulation_Suffix2Add; [\IPInboundManipulation] For a description of the table, see "Configuring IP-to-IP Inbound Manipulations" on page 597.
IP to IP Outbound Manipulation Table	
IP to IP Outbound Manipulation `configure voip > sbc manipulations ip-outbound-manipulation` [IPOutboundManipulation]	Defines IP-to-IP outbound manipulation rules. The format of the ini file table parameter is as follows: [IPOutboundManipulation] FORMAT IPOutboundManipulation_Index = IPOutboundManipulation_ManipulationName, IPOutboundManipulation_RoutingPolicyName, IPOutboundManipulation_IsAdditionalManipulation, IPOutboundManipulation_SrcIPGroupName, IPOutboundManipulation_DestIPGroupName, IPOutboundManipulation_SrcUsernamePrefix, IPOutboundManipulation_SrcHost, IPOutboundManipulation_DestUsernamePrefix,

Parameter	Description
	IPOutboundManipulation_DestHost, IPOutboundManipulation_CallingNamePrefix, IPOutboundManipulation_MessageConditionName, IPOutboundManipulation_RequestType, IPOutboundManipulation_ReRouteIPGroupName, IPOutboundManipulation_Trigger, IPOutboundManipulation_ManipulatedURI, IPOutboundManipulation_RemoveFromLeft, IPOutboundManipulation_RemoveFromRight, IPOutboundManipulation_LeaveFromRight, IPOutboundManipulation_Prefix2Add, IPOutboundManipulation_Suffix2Add, IPOutboundManipulation_PrivacyRestrictionMode, IPOutboundManipulation_DestTags, IPOutboundManipulation_SrcTags; [\IPOutboundManipulation] For a description of the table, see "Configuring IP-to-IP Outbound Manipulations" on page 601.
SBC Routing Policy Table	
SBC Routing Policy `configure voip > sbc routing sbc-routing-policy` [SBCRoutingPolicy]	Defines SBC Routing Policies. The format of the ini file table parameter is as follows: [SBCRoutingPolicy] FORMAT SBCRoutingPolicy_Index = SBCRoutingPolicy_Name, SBCRoutingPolicy_LCREnable, SBCRoutingPolicy_LCRAverageCallLength, SBCRoutingPolicy_LCRDefaultCost, SBCRoutingPolicy_LdapServerGroupName; [\SBCRoutingPolicy] For a description of the table, see "Configuring SBC Routing Policy Rules" on page 590.
Dial Plan Table	
Dial Plan `configure voip > sbc dial-plan` [DialPlans]	Defines the name of the Dial Plan. The format of the ini file table parameter is as follows: [DialPlan] FORMAT DialPlan_Index = DialPlan_Name; [\DialPlan] For a description of the table, see Configuring Dial Plans on page 607.
Dial Plan Rule Table	
Dial Plan Rule `configure voip > sbc dial-plan-rule` [DialPlanRule]	Defines the dial plan rules per Dial Plan. For a description of the table, see Configuring Dial Plans. **Note:** ▪ The table is hidden in the ini file. ▪ To configure Dial Plan rules from a file, see Importing and Exporting Dial Plans on page 611.

57.11.1 Supplementary Services

The SBC and CRP supplementary services parameters are described in the table below.

Table 57-64: SBC and CRP Supplementary Services Parameters

Parameter	Description
Emergency Call Preemption Parameters For more information on SBC emergency call preemption, "Configuring Call Preemption for SBC Emergency Calls" on page 617.	
SBC Preemption Mode `configure voip > sbc general-setting > sbc-preemption-mode` [SBCPreemptionMode]	Enables SBC emergency call preemption. - [0] Disable (default) - [1] Enable
SBC Emergency Message Condition `configure voip > sbc general-setting > sbc-emerg-condition` [SBCEmergencyCondition]	Defines the index of the Message Condition rule in the Message Condition table that is used to identify emergency calls. **Note:** The device applies the rule only after call classification (but before inbound manipulation).
SBC Emergency RTP DiffServ `configure voip > sbc general-setting > sbc-emerg-rtp-diffserv` [SBCEmergencyRTPDiffServ]	Defines DiffServ bits sent in the RTP for SBC emergency calls. The valid value is 0 to 63. The default is 46.
SBC Emergency Signaling DiffServ `configure voip > sbc general-setting > sbc-emerg-sig-diffserv` [SBCEmergencySignalingDiffServ]	Defines DiffServ bits sent in SIP signaling messages for SBC emergency calls. This is included in the SIP Resource-Priority header. The valid value is 0 to 63. The default is 40.

57.12 Standalone Survivability Parameters

The Stand-alone Survivability (SAS) parameters are described in the table below. For a detailed description of SAS, refer to the *SAS Configuration Guide*.

Table 57-65: SAS Parameters

Parameter	Description
Enable SAS `enable-sas` [EnableSAS]	Enables the Stand-Alone Survivability (SAS) feature. - **[0]** Disable (default) - **[1]** Enable When enabled, the device receives the registration requests from different SIP entities in the local network and then forwards them to the defined proxy. If the connection to the proxy fails ('Emergency Mode'), the device serves as a proxy by allowing calls internal to the local network or outgoing to PSTN.

Parameter	Description
	Note: For the parameter to take effect, a device reset is required.
SAS Default Gateway IP `sas-default-gw-ip` [SASDefaultGatewayIP]	Defines the Default Gateway used in SAS 'Emergency Mode'. When an incoming SIP INVITE is received and the destination Address-Of-Record is not included in the SAS database, the request is immediately sent to this default gateway. The address can be configured as an IP address (dotted-decimal notation) or as a domain name (up to 49 characters). You can also configure the IP address with a destination port, e.g., "10.1.2.3:5060". The default is a null string, i.e., the local IP address of the gateway.
SAS Registration Time `sas-registration-time` [SASRegistrationTime]	Defines the value of the SIP Expires header that is sent in a 200 OK response to an incoming REGISTER message when in SAS 'Emergency Mode'. The valid range is 10 to 2,000,000. The default is 20.
SAS Connection Reuse `sas-connection-reuse` [SASConnectionReuse]	Enables the re-use of the same TCP connection for sessions with the same user in the SAS application. ▪ **[0]** Disable ▪ **[1]** Enable (default) The device can use the same TCP connection for multiple SIP requests / responses for a specific SIP UA. For example, assume the following: ▪ User A sends a REGISTER message to SAS with transport=TCP. ▪ User B sends an INVITE message to A using SAS. In this scenario, the SAS application forwards the INVITE request using the TCP connection that User A initially opened with the REGISTER message.
Enable Record-Route `record-route` [SASEnableRecordRoute]	Determines whether the device's SAS application adds the SIP Record-Route header to SIP requests. This ensures that SIP messages traverse the device's SAS agent by including the SAS IP address in the Record-Route header. ▪ **[0]** Disable (default) ▪ **[1]** Enable The Record-Route header is inserted in a request by a SAS proxy to force future requests in the dialog session to be routed through the SAS agent. Each traversed proxy in the path can insert this header, causing all future dialogs in the session to pass through it as well. When this feature is enabled, the SIP Record-Route header includes the URI "lr" parameter, indicating loose routing, for example: `Record-Route: <sip:server10.biloxi.com;lr>`
SAS Proxy Set `sas-proxy-set` [SASProxySet]	Defines the Proxy Set (index number) used in SAS Normal mode to forward REGISTER and INVITE requests from users that are served by the SAS application. The valid range is 0 to 5. The default is 0 (i.e., default Proxy Set).

Parameter	Description
Redundant SAS Proxy Set `rdcy-sas-proxy-set` [RedundantSASProxySet]	Defines the Proxy Set (index number) used in SAS Emergency mode for fallback when the user is not found in the Registered Users database. Each time a new SIP request arrives, the SAS application checks whether the user is listed in the registration database. If the user is located in the database, the request is sent to the user. If the user is not found, the request is forwarded to the next redundant SAS defined in the Redundant SAS Proxy Set. If that SAS Proxy IP appears in the Via header of the request, it is not forwarded (thereby, preventing loops in the request's course). If no such redundant SAS exists, the SAS sends the request to its default gateway (configured by the parameter SASDefaultGatewayIP). The valid range is -1 to 5. The default is -1 (i.e., no redundant Proxy Set).
SAS Block Unregistered Users `sas-block-unreg-usrs` [SASBlockUnRegUsers]	Determines whether the device rejects SIP INVITE requests received from unregistered SAS users. This applies to SAS Normal and Emergency modes. ▪ **[0]** Un-Block = (Default) Allow INVITE from unregistered SAS users. ▪ **[1]** Block = Reject dialog-establishment requests from un-registered SAS users.
`sas-contact-replace` [SASEnableContactReplace]	Enables the device to change the SIP Contact header so that it points to the SAS host and therefore, the top-most SIP Via header and the Contact header point to the same host. ▪ **[0]** (default) = Disable - when relaying requests, the SAS agent adds a new Via header (with the SAS IP address) as the top-most Via header and retains the original Contact header. Thus, the top-most Via header and the Contact header point to different hosts. ▪ **[1]** = Enable - the device changes the Contact header so that it points to the SAS host and therefore, the top-most Via header and the Contact header point to the same host. **Note:** Operating in this mode causes all incoming dialog requests to traverse the SAS, which may cause load problems.
SAS Survivability Mode `sas-survivability` [SASSurvivabilityMode]	Determines the Survivability mode used by the SAS application. ▪ **[0]** Standard = (Default) Incoming INVITE and REGISTER requests are forwarded to the defined Proxy list of SASProxySet in Normal mode and handled by the SAS application in Emergency mode. ▪ **[1]** Always Emergency = The SAS application does not use Keep-Alive messages towards the SASProxySet, instead it always operates in Emergency mode (as if no Proxy in the SASProxySet is available). ▪ **[2]** Ignore Register = Use regular SAS Normal/Emergency logic (same as option [0]), but when in Normal mode incoming REGISTER requests are ignored. ▪ **[3]** Auto-answer REGISTER = When in Normal mode, the device responds to received REGISTER requests by sending a SIP 200 OK (instead of relaying the registration

Parameter	Description
	requests to a Proxy), and enters the registrations in its SAS database. - **[4]** Use Routing Table only in Normal mode = The device uses the IP-to-IP Routing table to route IP-to-IP SAS calls only when in SAS Normal mode (and is unavailable when SAS is in Emergency mode). This allows routing of SAS IP-to-IP calls to different destinations (and not only to the SAS Proxy Set).
SAS Subscribe Response `sas-subscribe-resp` [SASSubscribeResponse]	Defines the SIP response upon receipt of a SUBSCRIBE message when SAS is in Emergency mode. For example, if the parameter is set to "200", then SAS sends a SIP 200 OK in response to a SUBSCRIBE message, when in Emergency mode. The valid value is 200 to 699. The default is 489.
Enable ENUM `enable-enum` [SASEnableENUM]	Enables SAS to perform ENUM (E.164 number to URI mapping) queries when receiving INVITE messages in SAS emergency mode. - **[0]** Disable (default) - **[1]** Enable
SAS Binding Mode `sasbindingmode` [SASBindingMode]	Determines the SAS application database binding mode. - **[0]** URI = (Default) If the incoming AoR in the INVITE requests is using a 'tel:' URI or 'user=phone' is defined, the binding is performed according to the user part of the URI only. Otherwise, the binding is according to the entire URI, i.e., User@Host. - **[1]** User Part only = The binding is always performed according to the User Part only.
SAS Emergency Numbers `sas-emerg-nb` [SASEmergencyNumbers]	Defines emergency numbers for the device's SAS application. When the device's SAS agent receives a SIP INVITE (from an IP phone) that includes one of the emergency numbers (in the SIP user part), it forwards the INVITE to the default gateway (configured by the parameter SASDefaultGatewayIP), i.e., the device itself, which sends the call directly to the PSTN. This is important for routing emergency numbers such as 911 (in North America) directly to the PSTN. This is applicable to SAS operating in Normal and Emergency modes. Up to four emergency numbers can be defined, where each number can be up to four digits.
`sas-emerg-prefix` [SASEmergencyPrefix]	Defines a prefix that is added to the Request-URI user part of the INVITE message that is sent by the device's SAS agent when in Emergency mode to the default gateway or to any other destination (using the IP-to-IP Routing table). The parameter is required to differentiate between normal SAS calls routed to the default gateway and emergency SAS calls. Therefore, this allows you to define different manipulation rules for normal and emergency calls. This valid value is a character string. By default, no value is defined.

Parameter	Description
SAS Entering Emergency Mode `sas-enter-emg-mode` [SASEnteringEmergencyMode]	Determines for which sent SIP message types the device enters SAS Emergency mode if no response is received for them from the proxy server. ▪ **[0]** = (Default) SAS enters Emergency mode only if no response is received from sent SIP OPTIONS messages. ▪ **[1]** = SAS enters Emergency mode if no response is received from sent SIP OPTIONS, INVITE, or REGISTER messages. **Note:** If the keep-alive mechanism is disabled for the Proxy Set (in the Proxy Set table) and the parameter is set to [1], SAS enters Emergency mode only if no response is received from sent INVITE or REGISTER messages.
`sas-indialog-mode` [SASInDialogRequestMode]	Defines how the device sends incoming SIP dialog requests received from users when not in SAS Emergency mode. ▪ **[0]** = (Default) Send according to the SIP Request-URI. ▪ **[1]** = Send to Proxy server.

`SAS Inbound Manipulation Mode`
`sas-inb-manipul-md`
[SASInboundManipulationMode]
	Enables destination number manipulation of incoming INVITE messages when SAS is in Emergency mode. The manipulation rule is done in the IP to IP Inbound Manipulation table.
	▪ **[0]** None (default)
	▪ **[1]** Emergency Only
	Note: Inbound manipulation applies only to INVITE requests.

SAS Registration Manipulation Table	
SAS Registration Manipulation `configure voip > sas sasregistrationmanipulation` [SASRegistrationManipulation]	Defines the SAS Registration Manipulation table. This table is used by the SAS application to manipulate the SIP Request-URI user part of incoming INVITE messages and of incoming REGISTER request AoR (To header), before saving it to the registered users database. The format of The table is as follows: [SASRegistrationManipulation] FORMAT SASRegistrationManipulation_Index = SASRegistrationManipulation_RemoveFromRight, SASRegistrationManipulation_LeaveFromRight; [\SASRegistrationManipulation] For example, the manipulation rule below routes an INVITE with Request-URI header "sip:7184002@10.33.4.226" to user "4002@10.33.4.226" (i.e., keep only four digits from right of user part): `SASRegistrationManipulation 0 = 0, 4;`

Parameter	Description
SAS IP-to-IP Routing Table	
[IP2IPRouting]	Defines the IP-to-IP Routing table for SAS routing rules. The format of the ini file table parameter is as follows: [IP2IPRouting] FORMAT IP2IPRouting_Index = IP2IPRouting_SrcIPGroupName, IP2IPRouting_SrcUsernamePrefix, IP2IPRouting_SrcHost, IP2IPRouting_DestUsernamePrefix, IP2IPRouting_DestHost, IP2IPRouting_DestType, IP2IPRouting_DestIPGroupName, IP2IPRouting_DestSRDID, IP2IPRouting_DestAddress, IP2IPRouting_DestPort, IP2IPRouting_DestTransportType, IP2IPRouting_AltRouteOptions; [\IP2IPRouting] For example: IP2IPRouting 1 = -1, *, *, *, *, 0, -1, -1, , 0, -1, 0;

57.13 IP Media Parameters

The IP media parameters are described in the table below.

Table 57-66: IP Media Parameters

Parameter	Description
IPMedia Detectors `IPM-detectors-enable` [EnableDSPIPMDetectors]	Enables the device's DSP detectors for detection features such as AMD. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ The DSP Detectors feature is available only if the device is installed with a Software License Key that includes this feature. For installing a Software License Key, see "Software License Key" on page 668. ▪ When enabled (1), the number of available channels is reduced.
Answering Machine Detector (AMD) Parameters For more information on AMD, see "Answering Machine Detection (AMD)" on page 198.	
Answer Machine Detector Sensitivity Parameter Suit `amd-sensitivity-parameter-suit` [AMDSensitivityParameterSuit]	Global parameter that defines the AMD Parameter Suite to use. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_AMDSensitivityParameterSuit). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.

Parameter	Description
Answer Machine Detector Sensitivity Level `amd-sensitivity-level` [AMDSensitivityLevel]	Global parameter that defines the AMD detection sensitivity level of the selected AMD Parameter Suite. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_AMDSensitivityLevel). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
AMD Sensitivity File [AMDSensitivityFileName]	Defines the name of the AMD Sensitivity file that contains the AMD Parameter Suites. **Notes:** ▪ This file must be in binary format (.dat). You can use the DConvert utility to convert the original file format from XML to .dat. ▪ You can load this file using the Web interface (see "Loading Auxiliary Files" on page 647).
[AMDSensitivityFileUrl]	Defines the URL path to the AMD Sensitivity file for downloading from a remote server.
[AMDMinimumVoiceLength]	Defines the AMD minimum voice activity detection duration (in 5-ms units). Voice activity duration below this threshold is ignored and considered as non-voice. The valid value range is 10 to 100. The default is 42 (i.e., 210 ms).
[AMDMaxGreetingTime]	Global parameter that defines the maximum duration that the device can take to detect a greeting message. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_AMDMaxGreetingTime). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
[AMDMaxPostGreetingSilenceTime]	Global parameter that defines the maximum duration of silence from after the greeting time is over (defined by AMDMaxGreetingTime) until the device's AMD decision. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_AMDMaxPostSilenceGreetingTime). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
[AMDTimeout]	Defines the timeout (in msec) between receiving Connect messages from the ISDN and sending AMD results. The valid range is 1 to 30,000. The default is 2,000 (i.e., 2 seconds).

Parameter	Description
AMD Beep Detection Mode `amd-beep-detection` [AMDBeepDetectionMode]	Determines the AMD beep detection mode. This mode detects the beeps played at the end of an answering machine message, by using the X-Detect header extension. The device sends a SIP INFO message containing the field values Type=AMD and SubType=Beep. This feature allows users of certain third-party, Application server to leave a voice message after an answering machine plays the "beep". ▪ [0] Disabled (default) ▪ [1] Start After AMD ▪ [2] Start Immediately
Answer Machine Detector Beep Detection Timeout `amd-beep-detection-timeout` [AMDBeepDetectionTimeout]	Defines the AMD beep detection timeout (i.e., the duration that the beep detector functions from when detection is initiated). This is used for detecting beeps at the end of an answering machine message. The valid value is in units of 100 milliseconds, from 0 to 1638. The default is 200 (i.e., 20 seconds).
Answer Machine Detector Beep Detection Sensitivity `amd-beep-detection-sensitivity` [AMDBeepDetectionSensitivity]	Defines the AMD beep detection sensitivity for detecting beeps at the end of an answering machine message. The valid value is 0 to 3, where 0 (default) is the least sensitive.
`early-amd` [EnableEarlyAMD]	Enables AMD detection to be activated upon receipt of an ISDN Alerting or Connect message. ▪ [0] = (Default) Disable - AMD is activated upon receipt of ISDN Connect message. ▪ [1] = Enable - AMD is activated upon receipt of ISDN Alerting message.
AMD mode `amd-mode` [AMDmode]	Global parameter that enables the device to disconnect the IP-to-Tel call upon detection of an answering machine on the Tel side. You can also configure this functionality per specific calls, using IP Profiles (IpProfile_AmdMode). For a detailed description of the parameter and for configuring this functionality in the IP Profile table, see "Configuring IP Profiles" on page 387. **Note:** If this functionality is configured for a specific IP Profile, the settings of this global parameter is ignored for calls associated with the IP Profile.
Energy Detector Parameters	

`Enable Energy Detector`
`energy-detector-enable`
`[EnableEnergyDetector]`
	Enables the Energy Detector feature. This feature generates events (notifications) when the signal received from the PSTN is higher or lower than a user-defined threshold (defined by the EnergyDetectorThreshold parameter).
	▪ [0] Disable (default)
	▪ [1] Enable

Energy Detector Quality Factor `energy-detector-sensitivity` [EnergyDetectorQualityFactor]	Defines the Energy Detector's sensitivity level. The valid range is 0 to 10, where 0 is the lowest sensitivity and 10 the highest sensitivity. The default is 4.

Parameter	Description
Energy Detector Threshold `energy-detector-threshold` [EnergyDetectorThreshold]	Defines the Energy Detector's threshold. A signal below or above this threshold invokes an 'Above' or 'Below' event. The threshold is calculated as follows: Actual Threshold = -44 dBm + (EnergyDetectorThreshold * 6) The valid value range is 0 to 7. The default is 3 (i.e., -26 dBm).

57.14 Services

57.14.1 SIP-based Media Recording Parameters

The SIP-based media recording parameters are described in the table below.

Table 57-67: SIP-based Media Recording Parameters

Parameter	Description
SIP Recording Application `configure voip/services sip-recording general-setting/enable-sip-rec` [EnableSIPRec]	Enables the SIP-based Media Recording feature: ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** For the parameter to take effect, a device reset is required.
Recording Server (SRS) Destination Username `configure voip/services sip-recording general-setting/siprec-server-dest-username` [SIPRecServerDestUsername]	Defines the SIP user part for the recording server. This user part is added in the SIP To header of the INVITE message that the device sends to the recording server. The valid value is a string of up to 50 characters. By default, no user part is defined.
SIP Recording Time Stamp Format `configure voip/services sip-recording general-setting/siprec-time-stamp` [SIPRecTimeStamp]	Defines the format of the device's time (timestamp) in SIP messages that are sent to the SIP Recording Server (SRS). ▪ [0] Local Time = (Default) The device's local time (without the UTC time zone) is used for the timestamp. ▪ [1] UTC = The device's UTC time is used for the timestamp. **Note:** The timestamp is contained in the XML body of the SIP message. If the timestamp uses the UTC time, the time is suffixed with the letter "Z", for example: <associate-time>2017-09-07T06:33:38**Z**</associate-time>

Parameter	Description
SIP Recording Table	
SIP Recording table `configure voip/services sip-recording sip-rec-routing` [SIPRecRouting]	Defines SIP Recording Routing rules (for siprec). The format of the ini file table parameter is as follows: [SIPRecRouting] FORMAT SIPRecRouting_Index = SIPRecRouting_RecordedIPGroupName, SIPRecRouting_RecordedSourcePrefix, SIPRecRouting_RecordedDestinationPrefix, SIPRecRouting_PeerIPGroupName, SIPRecRouting_PeerTrunkGroupID, SIPRecRouting_Caller, SIPRecRouting_SRSIPGroupName; [\SIPRecRouting] For a description of the table, see "Configuring SIP Recording Rules" on page 221.

57.14.2 RADIUS and LDAP Parameters

57.14.2.1 General Parameters

The general RADIUS and LDAP parameters are described in the table below.

Table 57-68: General RADIUS and LDAP Parameters

Parameter	Description
Use Local Users Database `configure system > mgmt-auth > use-local-users-db` [MgmtUseLocalUsersDatabase]	Defines when the device uses its local management-users database (Web Users table) or an LDAP/RADIUS server for authenticating the login credentials (username-password) of users when logging into the device's management interface (e.g., Web or CLI). ▪ **[0]** When No Auth Server Defined = (Default) The device authenticates the users usng the Web Users table in the following scenarios: ✓ If no LDAP/RADIUS server is configured. ✓ If an LDAP/RADIUS server is configured, but connectivity with the server is down. If there is connectivity with the server, the device uses the server to authenticate the user. ▪ **[1]** Always = The device first attempts to authenticate the user using the Web Users table. If no user is found (based on the username-password combination), it attempts to authenticate the user using the LDAP/RADIUS server.
Behavior upon Authentication Server Timeout `configure system > mgmt-auth > timeout-behavior` [MgmtBehaviorOnTimeout]	Defines the device's response when a connection timeout occurs with the LDAP/RADIUS server. ▪ **[0]** Deny Access = User is denied access to the management platform. ▪ **[1]** Verify Access Locally = (Default) Device verifies the user's credentials in its Web Users table (local database).

Parameter	Description
	Note: The parameter is applicable to LDAP- and RADIUS-based management-user login authentication.
Default Access Level `default-access-level` [DefaultAccessLevel]	Defines the default access level for the device when the LDAP/RADIUS response doesn't include an access level attribute for determining the user's management access level. The valid range is 0 to 255. The default is 200 (i.e., Security Administrator). **Note:** The parameter is applicable to LDAP- or RADIUS-based management-user login authentication and authorization.

57.14.2.2 RADIUS Parameters

The RADIUS parameters are described in the table below.

Table 57-69: RADIUS Parameters

Parameter	Description
General RADIUS Parameters	
Enable RADIUS Access Control `enable` [EnableRADIUS]	Enables the RADIUS application. ▪ **[0]** Disable (Default) ▪ **[1]** Enable **Note:** For the parameter to take effect, a device reset is required.
[RadiusTrafficType]	Defines the device's network interface for communicating (RADIUS traffic) with the RADIUS server(s). ▪ [0] OAMP (default) ▪ [1] Control **Note:** If set to Control, only one Control interface must be configured in the Interface table; otherwise, RADIUS communication will fail.
RADIUS VSA Vendor ID `configure system > radius > vsa-vendor-id` [RadiusVSAVendorID]	Defines the vendor ID that the device accepts when parsing a RADIUS response packet. The valid range is 0 to 0xFFFFFFFF. The default is 5003.
[MaxRADIUSSessions]	Defines the number of concurrent calls that can communicate with the RADIUS server (optional). The valid range is 0 to 240. The default is 240.
RADIUS Packets Retransmission [RADIUSRetransmission]	Defines the number of RADIUS retransmission retries when no response is received from the RADIUS server. See also the RadiusTo parameter. The valid range is 1 to 10. The default is 1.
RADIUS Response Time Out [RadiusTO]	Defines the time interval (in seconds) that the device waits for a response before it performs a RADIUS retransmission. See also the RADIUSRetransmission parameter. The valid range is 1 to 30. The default is 2.

Parameter	Description
RADIUS Accounting Parameters	
RADIUS Accounting Type `radius-accounting` [RADIUSAccountingType]	Determines when the RADIUS accounting messages are sent to the RADIUS accounting server. ▪ **[0]** At Call Release = (Default) Sent at call release only. ▪ **[1]** At Connect & Release = Sent at call connect and release. ▪ **[2]** At Setup & Release = Sent at call setup and release.
AAA Indications `aaa-indications` [AAAIndications]	Determines the Authentication, Authorization and Accounting (AAA) indications. ▪ **[0]** None = (Default) No indications. ▪ **[3]** Accounting Only = Only accounting indications are used.
RADIUS User Authentication Parameters	
Use RADIUS for Web/Telnet Login `enable-mgmt-login` [WebRADIUSLogin]	Enables RADIUS queries for Web and Telnet login authentication. When enabled, logging into the device's Web and Telnet embedded servers is done through a RADIUS server. The device communicates with a user-defined RADIUS server and verifies the given username and password against a remote database, in a secure manner. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Notes:** ▪ For RADIUS login authentication to function, you must also configure the EnableRADIUS parameter to 1 (Enable). ▪ RADIUS authentication requires HTTP basic authentication, where the username and password are transmitted in clear text over the network. Therefore, it's recommended to set the HTTPSOnly parameter to 1 to force the use of HTTPS, since the transport is encrypted.
Password Local Cache Mode `local-cache-mode` [RadiusLocalCacheMode]	Defines the device's mode of operation regarding the timer (configured by the parameter RadiusLocalCacheTimeout) that determines the validity of the username and password (verified by the RADIUS server). ▪ **[0]** Absolute Expiry Timer = When you access a Web page, the timeout doesn't reset, instead it continues decreasing. ▪ **[1]** Reset Timer Upon Access = (Default) Upon each access to a Web page, the timeout always resets (reverts to the initial value configured by RadiusLocalCacheTimeout).
Password Local Cache Timeout `local-cache-timeout` [RadiusLocalCacheTimeout]	Defines the time (in seconds) the locally stored username and password (verified by the RADIUS server) are valid. When this time expires, the username and password become invalid and a must be re-verified with the RADIUS server. The valid range is 1 to 0xFFFFFF. The default is 300 (5 minutes). ▪ **[-1]** = Never expires. ▪ **[0]** = Each request requires RADIUS authentication.
RADIUS VSA Access Level Attribute `vsa-access-level` [RadiusVSAAccessAttribute]	Defines the code that indicates the access level attribute in the Vendor Specific Attributes (VSA) section of the received RADIUS packet. The valid range is 0 to 255. The default is 35.

57.14.2.3 LDAP Parameters

The Lightweight Directory Access Protocol (LDAP) parameters are described in the table below.

Table 57-70: LDAP Parameters

Parameter	Description
LDAP Service `configure voip/ldap/enable` [LDAPServiceEnable]	Enables the LDAP feature. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** For the parameter to take effect, a device reset is required.
LDAP Authentication Filter `configure voip > ldap > auth-filter` [LDAPAuthFilter]	Defines the LDAP search filter attribute for searching the login username in the directory's subtree for LDAP-based user authentication and authorization. You can use the dollar ($) sign to represent the username. For example, if the parameter is set to "(sAMAccountName=$)" and the user logs in with the username "SueM", the LDAP query is run for sAMAccountName=SueM.
Use LDAP for Web/Telnet Login `configure voip > ldap > enable-mgmt-login` [MgmtLDAPLogin]	Enables LDAP-based management-user login authentication and authorization. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** For the parameter to take effect, a device reset is required.
[LDAPDebugMode]	Determines whether to enable the LDAP task debug messages. This is used for providing debug information regarding LDAP tasks. The valid value range is 0 to 3. The default is 0.
LDAP Numeric Attribute `configure voip > sip-definition advanced-settings > ldap-numeric-attr` [LDAPNumericAttributes]	Defines up to five LDAP Attributes (separated by commas) for which the device employs LDAP query searches in the AD for numbers that may have characters between the digits. For more information, see Enabling LDAP Searches for Numbers with Characters on page 253.
MS LDAP OCS Number attribute name `ldap-ocs-nm-attr` [MSLDAPOCSNumAttributeName]	Defines the name of the attribute that represents the user's Lync number in the Microsoft AD database. The valid value is a string of up to 49 characters. The default is "msRTCSIP-Line".
MS LDAP PBX Number attribute name `ldap-pbx-nm-attr` [MSLDAPPBXNumAttributeName]	Defines the name of the attribute that represents the user PBX number in the Microsoft AD database. The valid value is a string of up to 49 characters. The default is "telephoneNumber".
MS LDAP MOBILE Number attribute name `ldap-mobile-nm-attr` [MSLDAPMobileNumAttributeName]	Defines the name of the attribute that represents the user Mobile number in the Microsoft AD database. The valid value is a string of up to 49 characters. The default is "mobile".

Parameter	Description
`ldap-private-nm-attr` [MSLDAPPrivateNumAttributeName]	Defines the name of the attribute that represents the user's private number in the AD. If this value equals the value of the MSLDAPPrimaryKey or MSLDAPSecondaryKey parameter, then the device queries the AD for the destination number in this private attribute name; otherwise, the parameter is not used as a search key. The default is "msRTCSIP-PrivateLine".
MS LDAP DISPLAY Name Attribute Name `ldap-display-nm-attr` [MSLDAPDisplayNameAttributeName]	Defines the attribute name that represents the Calling Name in the AD for LDAP queries based on calling number. The valid value is a string of up to 49 characters. The default is "displayName".
`ldap-primary-key` [MSLDAPPrimaryKey]	Defines the name of the attribute used as a query search key for the destination number in the AD. This is used instead of the "PBX" attribute name (configured by the MSLDAPPBXNumAttributeName parameter). The default is not configured.
`ldap-secondary-key` [MSLDAPSecondaryKey]	Defines the name of the attribute used as the second query search key for the destination number in the AD, if the primary search key or PBX search is not found.
LDAP Cache Service `cache` [LDAPCacheEnable]	Enables the LDAP cache service. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Notes:** ▪ For the parameter to take effect, a device reset is required. ▪ For more information on LDAP caching, see "Configuring the Device's LDAP Cache" on page 244.
LDAP Configuration Table	
LDAP Configuration Table `configure voip > ldap > ldap-configuration` [LdapConfiguration]	Defines LDAP servers. The format of the ini file table parameter is as follows: [LdapConfiguration] FORMAT LdapConfiguration_Index = LdapConfiguration_Group, LdapConfiguration_LdapConfServerIp, LdapConfiguration_LdapConfServerPort, LdapConfiguration_LdapConfServerMaxRespondTime, LdapConfiguration_LdapConfServerDomainName, LdapConfiguration_LdapConfPassword, LdapConfiguration_LdapConfBindDn, LdapConfiguration_Interface, LdapConfiguration_MngmAuthAtt, LdapConfiguration_useTLS, LdapConfiguration_ConnectionStatus; [\LdapConfiguration] For a description of the table, see "Configuring LDAP Servers" on page 236.
LDAP Server Search Base DN Table	

Parameter	Description
LDAP Server Search Base DN Table `configure voip > ldap > ldap-servers-search-dns` [LdapServersSearchDNs]	Defines the full base path (i.e., distinguished name / DN) to the objects in the AD where the query is done, per LDAP server. The format of the ini file table parameter is as follows: [LdapServersSearchDNs] FORMAT LdapServersSearchDNs_Index = LdapServersSearchDNs_Base_Path, LdapServersSearchDNs_LdapConfigurationIndex, LdapServersSearchDNs_SearchDnInternalIndex; [\LdapServersSearchDNs] For a detailed description of the table, see "Configuring LDAP DNs (Base Paths) per LDAP Server" on page 241.
Management LDAP Groups Table	
Management LDAP Groups Table `configure voip > ldap > mgmt-ldap-groups` [MgmntLDAPGroups]	Defines the users group attribute in the AD and corresponding management access level. The format of the ini file table parameter is as follows: [MgmntLDAPGroups] FORMAT MgmntLDAPGroups_Index = MgmntLDAPGroups_LdapConfigurationIndex, MgmntLDAPGroups_GroupIndex, MgmntLDAPGroups_Level, MgmntLDAPGroups_Group; [\MgmntLDAPGroups] For a description of the table, see "Configuring Access Level per Management Groups Attributes" on page 243.
LDAP Server Groups Table	
LDAP Server Groups Table `config-voip > ldap > ldap-servers-group` [LDAPServersGroup]	Defines LDAP Server Groups. The format of the ini file table parameter is as follows: [LdapServersGroup] FORMAT LdapServersGroup_Index = LdapServersGroup_Name, LdapServersGroup_ServerType, LdapServersGroup_SearchMethod, LdapServersGroup_CacheEntryTimeout, LdapServersGroup_CacheEntryRemovalTimeout, LdapServersGroup_SearchDnsMethod; [\LdapServersGroup] For a description of the table, see "Configuring LDAP Server Groups" on page 234.

57.14.3 Least Cost Routing Parameters

The Least Cost Routing (LCR) parameters are described in the table below.

Table 57-71: LCR Parameters

Parameter	Description
Cost Group Table `configure voip > services least-cost-routing cost-group` [CostGroupTable]	Defines the Cost Groups for LCR, where each Cost Group is configured with a name, fixed call connection charge, and a call rate (charge per minute). [CostGroupTable] FORMAT CostGroupTable_Index = CostGroupTable_CostGroupName, CostGroupTable_DefaultConnectionCost, CostGroupTable_DefaultMinuteCost; [\CostGroupTable] For example: CostGroupTable 2 = "Local Calls", 2, 1; For a description of the table, see "Configuring Cost Groups" on page 263.
Cost Group > Time Band Table `configure voip > services least-cost-routing cost-group-time-bands` [CostGroupTimebands]	Defines time bands and associates them with Cost Groups. [CostGroupTimebands] FORMAT CostGroupTimebands_TimebandIndex = CostGroupTimebands_StartTime, CostGroupTimebands_EndTime, CostGroupTimebands_ConnectionCost, CostGroupTimebands_MinuteCost; [\CostGroupTimebands] For a description of the table, see "Configuring Time Bands for Cost Groups" on page 264.

57.14.4 Call Setup Rules Parameters

The Call Setup Rules parameters are described in the table below.

Table 57-72: Call Setup Rules Parameters

Parameter	Description
Call Setup Rules `configure voip/services call-setup-rules` [CallSetupRules]	Defines Call Setup Rules that the device runs at call setup for LDAP-based routing and other advanced routing logic requirements including manipulation. [CallSetupRules] FORMAT CallSetupRules_Index = CallSetupRules_RulesSetID, CallSetupRules_QueryTarget, CallSetupRules_AttributesToQuery, CallSetupRules_AttributesToGet, CallSetupRules_RowRole, CallSetupRules_Condition, CallSetupRules_ActionSubject, CallSetupRules_ActionType, CallSetupRules_ActionValue; [\CallSetupRules] For a description of the table, see "Configuring Call Setup Rules" on page 283.

57.14.5 HTTP-based Services

The HTTP-based service parameters are described in the table below.

Table 57-73: HTTP-based Service Parameters

Parameter	Description
HTTP Remote Services [HTTPRemoteServices]	Defines HTTP-based services. The format of the ini file table parameter is as follows: [HTTPRemoteServices] FORMAT HTTPRemoteServices_Index = HTTPRemoteServices_Name, HTTPRemoteServices_Path, HTTPRemoteServices_HTTPType, HTTPRemoteServices_Policy, HTTPRemoteServices_LoginNeeded, HTTPRemoteServices_PersistentConnection, HTTPRemoteServices_NumOfSockets, HTTPRemoteServices_AuthUserName, HTTPRemoteServices_AuthPassword, HTTPRemoteServices_TLSContext, HTTPRemoteServices_VerifyCertificate, HTTPRemoteServices_TimeOut, HTTPRemoteServices_KeepAliveTimeOut, HTTPRemoteServices_ServiceStatus; [\HTTPRemoteServices] For a description of the table, see "Configuring HTTP Services" on page 267.
HTTP Remote Hosts [HTTPRemoteHosts]	Defines remote HTTP hosts per HTTP-based service. The format of the ini file table parameter is as follows: [HTTPRemoteHosts] FORMAT HTTPRemoteHosts_Index = HTTPRemoteHosts_HTTPRemoteServiceIndex, HTTPRemoteHosts_RemoteHostIndex, HTTPRemoteHosts_Name, HTTPRemoteHosts_Address, HTTPRemoteHosts_Port, HTTPRemoteHosts_Interface, HTTPRemoteHosts_HTTPTransportType, HTTPRemoteHosts_HostStatus; [\HTTPRemoteHosts] For a description of the table, see "Configuring Remote HTTP Hosts" on page 271.
Topology Status [RoutingServerGroupStatus]	Enables the reporting of the device's topology status (using the REST TopologyStatus API command) to HTTP remote hosts. ▪ [0] Disable (default) ▪ [1] Enable For more information, see "Configuring HTTP Services" on page 267.
GW Routing Server configure voip > gw routing general-setting > gw-routing-server [GWRoutingServer]	Enables routing by a routing server. ▪ [0] Disable = (Default) ▪ [1] Enable For more information, see Centralized Third-Party Routing Server or ARM on page 272. **Note:** The parameter is applicable only to the Gateway application.

57.14.6 HTTP Proxy Parameters

The HTTP Proxy service parameters are described in the table below.

HTTP Proxy Service Parameters

Parameter	Description
HTTP Proxy Application `configure system > http-proxy > http-proxy-app` [HTTPProxyApplication]	Enables the HTTP Proxy application. - [0] Disable (default) - [1] Enable **Note:** For the parameter to take effect, a device reset is required.
HTTP Interfaces Table `configure system > http-proxy > http-interface` [HTTPInterface]	Defines local listening interfaces for receiving HTTP/S requests from Web clients for HTTP/S-based services. The format of the ini file table parameter is as follows: [HTTPInterface] FORMAT HTTPInterface_Index = HTTPInterface_InterfaceName, HTTPInterface_NetworkInterface, HTTPInterface_Protocol, HTTPInterface_Port, HTTPInterface_TLSContext, HTTPInterface_VerifyCert; [\HTTPInterface] For a description of the table, see 'Configuring HTTP Interfaces' on page 276.
HTTP Proxy Services Table `configure system > http-proxy > http-proxy-serv` [HTTPProxyService]	Defines HTTP Proxy based services. The format of the ini file table parameter is as follows: [HTTPProxyService] FORMAT HTTPProxyService_Index = HTTPProxyService_ServiceName, HTTPProxyService_ListeningInterface, HTTPProxyService_URLPrefix, HTTPProxyService_KeepAliveMode; [\HTTPProxyService] For a description of the table, see 'Configuring HTTP Proxy Services' on page 277.
HTTP Proxy Hosts Table `configure system > http-proxy > http-proxy-host` [HTTPProxyHost]	Defines HTTP Proxy hosts. The table is a "child" of the HTTP Proxy Services table (HTTPProxyService). An HTTP Proxy Host represents the HTTP-based managed equipment (e.g., IP Phone). The format of the ini file table parameter is as follows: [HTTPProxyHost] FORMAT HTTPProxyHost_Index = HTTPProxyHost_HTTPProxyServiceId, HTTPProxyHost_HTTPProxyHostId, HTTPProxyHost_NetworkInterface, HTTPProxyHost_IpAddress, HTTPProxyHost_Protocol, HTTPProxyHost_Port, HTTPProxyHost_TLSContext, HTTPProxyHost_VerifyCert; [\HTTPProxyHost] For a description of the table, see 'Configuring HTTP Proxy Hosts' on page 279.

Parameter	Description
EMS Services Table `configure system > http-proxy > ems-serv` [EMSService]	Defines an HTTP-based EMS Service so that the device can act as an HTTP Proxy that enables AudioCodes EMS to manage AudioCodes equipment (such as IP Phones) over HTTP when the equipment is located behind NAT (e.g., in the LAN) and EMS is located in a public domain (e.g., in the WAN). The format of the ini file table parameter is as follows: [EMSService] FORMAT EMSService_Index = EMSService_ServiceName, EMSService_PrimaryServer, EMSService_SecondaryServer, EMSService_DeviceLoginInterface, EMSService_EMSInterface; [\EMSService] For a description of the table, see 'Configuring an HTTP-based EMS Service' on page 281.

58 SBC and DSP Channel Capacity

This chapter lists the supported DSP firmware templates and channel capacity.

Notes:

- Installation and use of voice coders is subject to obtaining the appropriate license and royalty payments.
- The number of channels refers to the maximum channel capacity of the device.
- For additional DSP templates, contact your AudioCodes sales representative.

58.1 Signaling-Media Sessions & User Registrations

The table below lists the maximum capacity figures for SIP signaling, media sessions, and registered users.

Table 58-1: Maximum Signaling, Media Sessions and Registered Users

Signaling Sessions	Media Sessions			Registered Users
	RTP-to-RTP	SRTP-RTP or SRTP-TDM	Codec Transcoding	
60	60	60	Not Supported	200

Notes:

- The figures listed in the table are accurate at the time of publication of this document. However, these figures may change due to a later software update. For the latest figures, please contact your AudioCodes sales representative.
- The RTP-to-RTP column represents maximum media sessions when all media sessions are RTP-to-RTP only. The same applies to the SRTP-RTP or SRTP-TDM column.
- *Registered Users* is the maximum number of users that can be registered with the device. This applies to the supported application (SBC or CRP).
- Regarding signaling, media, and transcoding session resources:
  - √ A signaling session is a SIP dialog session between two SIP entities, traversing the SBC and using one signaling session resource.
  - √ A media session is an audio (RTP or SRTP), fax (T.38), or video session between two SIP entities, traversing the SBC and using one media session resource.
  - √ A gateway session (i.e. TDM-RTP or TDM-SRTP) is also considered as a media session for the calculation of media sessions. In other words, the maximum Media Sessions specified in the table refer to the sum of Gateway and SBC sessions.
  - √ In case of direct media (i.e., anti-tromboning / Non-Media Anchoring), where only SIP signaling traverses the SBC and media flows directly between the SIP entities, only a signaling session resource is used. Thus, for products with a greater signaling session capacity than media, even when media session resources have been exhausted, additional signaling sessions can still be handled for direct-media calls.

58.2 Session Capacity per Configuration

The device's SBC session capacity and DSP channel capacity are listed in the tables below.

Table 58-2: E-SBC (Non Hybrid) SBC Capacity

Hardware Configuration	TDM-RTP Sessions			RTP-RTP Sessions
	DSP Channels Allocated for PSTN	Wideband Coders		Max. SBC Sessions
		G.722	AMR-WB	
SBC	N/A	N/A	N/A	60

Table 58-3: Hybrid E-SBC (with Gateway) Media & SBC Capacity

Hardware Configuration	DSP Channels Allocated for PSTN	Additional Coders				Max. SBC Sessions
		Narrowband	Wideband			
		Opus-NB	G.722	AMR-WB	Opus-WB	
2 x BRI / 4 x BRI	4/8	-	-	-	-	56/52
	4/8	-	√	-	-	56/52
	4/6	√	-	√	-	56/54
	4	-	-	-	√	56

58.3 Resetting the Device

The Maintenance Actions page allows you to remotely reset the device. Before resetting the device, you can also choose the following options:

- "Burn" (save) the device's current configuration to the device's flash memory (non-volatile).
- Graceful Shutdown, whereby the device resets only after a user-defined time (i.e., timeout) or after there is no traffic currently processed by the device (the earliest thereof).

Notes:

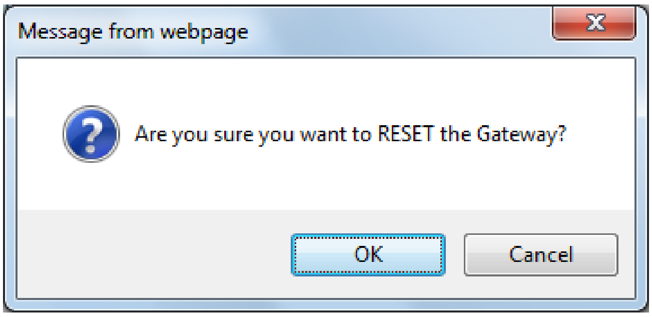
- Throughout the Web interface, parameters displayed with a lightning ⚡ symbol are not applied on-the-fly and require that you reset the device for them to take effect.
- When you modify parameters that require a device reset, once you click the **Submit** button in the relevant page, the toolbar displays "Reset" (see "Toolbar Description" on page 50) to indicate that a device reset is required.
- After you reset the device, the Web GUI is displayed in Basic view (see "Displaying Navigation Tree in Basic and Full View" on page 51).

➢ **To reset the device:**

1. Open the Maintenance Actions page (see "Basic Maintenance" on page 641).
2. Under the 'Reset Configuration' group, from the 'Burn To FLASH' drop-down list, select one of the following options:
   - **Yes:** The device's current configuration is saved (*burned*) to the flash memory prior to reset (default).
   - **No:** Resets the device without saving the current configuration to flash (discards all unsaved modifications).
3. Under the 'Reset Configuration' group, from the 'Graceful Option' drop-down list, select one of the following options:
   - **Yes:** Reset starts only after the user-defined time in the 'Shutdown Timeout' field (see Step 4) expires or after no more active traffic exists (the earliest thereof). In addition, no new traffic is accepted.
   - **No:** Reset starts regardless of traffic, and any existing traffic is terminated at once.
4. Under the 'Reset Configuration' group, in the 'Shutdown Timeout' field (relevant only if the 'Graceful Option' in the previous step is set to **Yes**), enter the time after which the device resets. Note that if no traffic exists and the time has not yet expired, the device resets.

5. Click the **Reset** button; a confirmation message box appears, requesting you to confirm.

Figure 58-1: Reset Confirmation Message Box



Message from webpage

Are you sure you want to RESET the Gateway?

6. Click **OK** to confirm device reset; if the parameter 'Graceful Option' is set to **Yes** (in Step 3), the reset is delayed and a screen displaying the number of remaining calls and time is displayed. When the device begins to reset, a message appears notifying you of this.

59 Technical Specifications

The device's technical specifications are listed in the table below.

Notes:

- All specifications in this document are subject to change without prior notice.
- The compliance and regulatory information can be downloaded from AudioCodes Web site at http://www.audiocodes.com/library.

Table 59-1: Technical Specifications

Function	Specification
Networking Interfaces	
Ethernet	4 FE interfaces configured in 1+1 redundancy or as individual ports
Security	
Encryption and Authentication	TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication, RADIUS Digest
Access Control	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting
VoIP Firewall	RTP pinhole management, rogue RTP detection and prevention, SIP message policy, advanced RTP latching
Privacy	Topology hiding, user privacy
Traffic Separation	VLAN/physical interface separation for multiple media, control and OAMP interfaces
Intrusion Detection System	Detection and prevention of VoIP attacks, theft of service and unauthorized access
Interoperability	
SIP B2BUA	Full SIP transparency, mature & broadly deployed SIP stack
SIP Interworking	3xx redirect, REFER, PRACK, session timer, early media, call hold, delayed offer
Registration and Authentication	User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users
Transport Mediation	SIP over UDP/TCP/TLS, IPv4 to IPv6, RTP to SRTP (SDES)
Message Manipulation	Ability to add/modify/delete SIP headers and message body using advanced regular expressions (regex)
URI and Number Manipulations	URI user and host name manipulations, ingress and egress digit manipulation
Transcoding and Vocoders	Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.726, G.729, GSM-FR, AMR-NB, AMR-WB (G.722.2), SILK-NB/WB, Opus-NB/WB
NAT	Local and far-end NAT traversal for support of remote workers
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, V.34, packet-time conversion

Function	Specification
Voice Quality and SLA	
Call Admission Control	Based on bandwidth, session establishment rate, number of connections/registrations
Packet Marking	802.1p/Q VLAN tagging, DiffServ, TOS
Standalone Survivability	Maintain local calls in the event of WAN failure, Outbound calls can use PSTN fallback for external connectivity (including E911).
Impairment Mitigation	Packet Loss Concealment, Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation, RTP redundancy, broken connection detection
Voice Enhancement	Transrating, RTCP-XR, acoustic echo cancellation, replacing voice profile due to impairment detection, fixed & dynamic voice gain control
Direct Media (No Media Anchoring)	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption
Voice Quality Monitoring	RTCP-XR, AudioCodes Session Experience Manager (SEM)
Quality of Experience	Access control and media quality enhancements based on QoE and bandwidth utilization
Test Agent	Ability to remotely verify connectivity, voice quality and SIP message flow between SIP UAs
SIP Routing	
Routing Methods	Request URL, IP Address, FQDN, ENUM, advanced LDAP, third-party routing control through REST API
Advanced Routing Criteria	QoE, bandwidth, SIP message (SIP request, coder type, etc.), Layer-3 parameters
Redundancy	Detection of proxy failures and subsequent routing to alternative proxies
Routing Features	Least-cost routing, call forking, load balancing, E911 gateway support, emergency call detection and prioritization
SIPRec	IETF standard SIP recording interface
Management	
OAM&P	Browser-based GUI, CLI, SNMP, EMS, INI Configuration file, REST API
Hardware Specifications	
Power Supply	Single universal AC power supply 100-240V, 3A, 50-60 Hz
Physical Dimensions (HxWxD)	51 x 296 x 160 mm (2 x 11.65 x 6.3 in.)
Weight	670 g (1.5 lbs.)
Mounting	Desktop
Environmental	▪ Operational: 5 to 40°C (41 to 104°F) ▪ Storage: -25 to 85°C (-13 to 185°F) ▪ Humidity: 10 to 90% non-condensing

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