Reference Guide

AudioCodes Gateway & Session Border Controller Series

Security Guidelines

Version 7.6

audiocodes

Notice

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WEEE EU Directive

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Related Documentation

SBC-Gateway Release Notes for Latest Release (LR) Versions 7.6
MP-1288 High-Density Analog Media Gateway User's Manual
Mediant 500 Gateway & E-SBC User's Manual
Mediant 500L Gateway & E-SBC User's Manual
Mediant 800 Gateway & E-SBC User's Manual
Mediant 1000B Gateway and E-SBC User's Manual
Mediant 3100 Gateway & E-SBC User's Manual
Mediant 2600 E-SBC User's Manual
Mediant 4000 SBC User's Manual
Mediant 9000 SBC User's Manual
Mediant Software SBC User's Manual

Document Revision Record

LTRT	Description
30232	Initial document release for Version 7.6; Block device Restarts after Emergency Calls; Secure Communication between HA devices; Password Obfuscation in Downloaded CLI and INI Files; Store TLS Private Keys Encrypted

Table of Contents

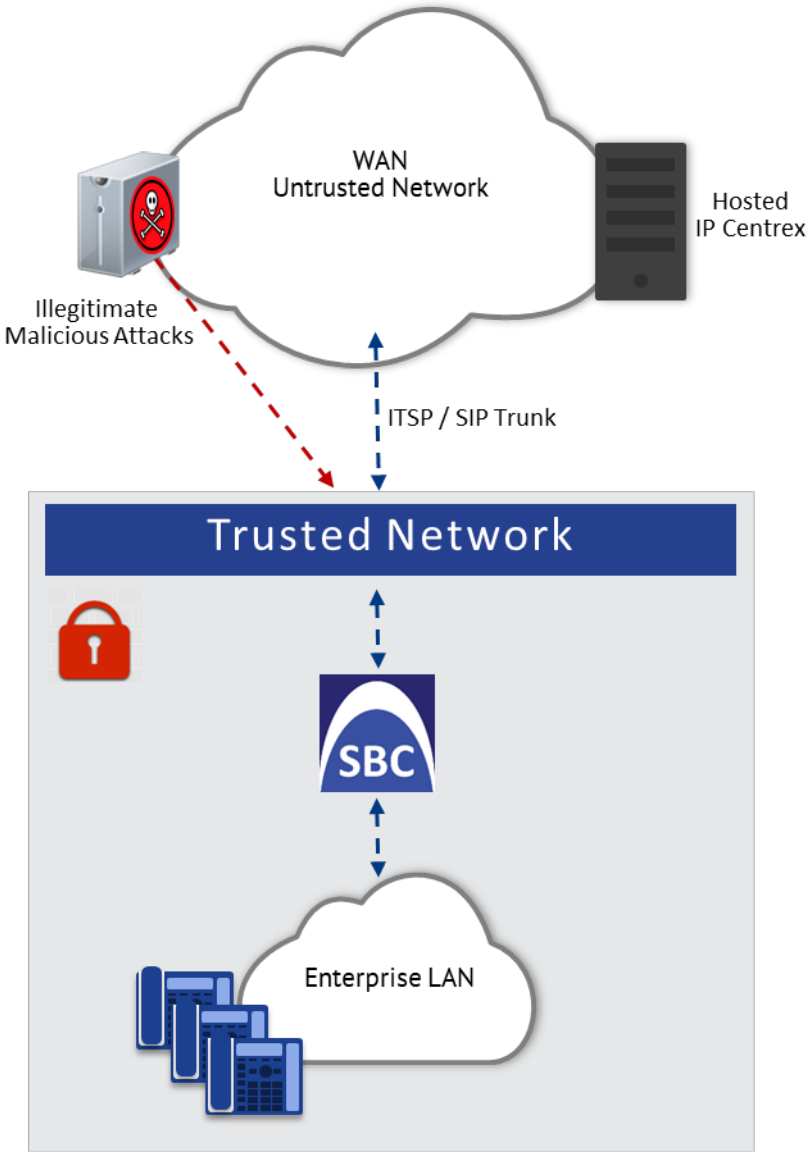
1 Introduction

This document provides recommended security guidelines for safeguarding your network and your AudioCodes device against malicious attacks.

- This document provides **only** recommended security guidelines; your network architecture may require additional and/or different security measures.
- The document includes partial configuration. For detailed configuration, refer to the device's *User's Manual*.
- The document may refer to AudioCodes products not included in Version 7.6. For a list of supported products, refer to the *Release Notes*.

Security Threats

AudioCodes devices are commonly located at the demarcation point between safe (trusted) and unsafe (untrusted) networks. A typical example of an un-trusted network would be a SIP trunk connected to an Internet Telephony Service Provider (ITSP) network; the trusted network would be the internal LAN. The following figure illustrates this basic concept of trusted and untrusted networks.



Attacks on your network from the un-trusted network may include the following:

- Denial of Service (DoS) attacks: Malicious attacks designed to cripple your VoIP network by overloading it with calls or service requests.
- Overload events: In addition to purposeful DoS attacks, non-malicious periods of intense activity can also cause an increase in call signaling rates that exceed what your infrastructure can support, resulting in network conditions that are similar in effect to DoS attacks. Successful attacks resulting in contact center downtime can result in lost revenue and diminished customer satisfaction.
- Network abuse and fraud: Malicious intrusion or service theft may take the form of an unauthorized user gaining access to your VoIP network by mimicking an authorized user or seizing control of a SIP proxy and initiating outbound calls to the PSTN for free. Another possibility is using a compromised endpoint to redirect or forward calls for eavesdropping.
- Viruses and malware: Computer viruses, worms, Trojan horses, and other malware can infect user agent phones and SIP-based ACD infrastructure - just as they can computers and

servers - and degrade performance or completely disrupt service. As devices become more sophisticated with distinct operating systems, malware also serves as a way to subjugate devices and launch DoS attacks that piggyback encrypted links.

- Identity theft: Phishing and "man-in-the-middle" can be used to acquire caller identification information to gain unauthorized access to services and information. Theft by phone (or service theft), whereby access to your corporate phone system is attempted by users posing as legitimate ones can sky-rocket your corporation's phone bill.
- Eavesdropping: The ability to listen to or record calls is easier on VoIP networks than on PSTN. This is a concern not only because of personal privacy violations, but also because sensitive information can be compromised and exploited.
- Spam over Internet Telephony (SPIT): The delivery of unsolicited calls or voicemails can inundate networks, annoy subscribers, and diminish the usefulness of VoIP networks.

These threats can exist, for example, at the following main IP network border points:

- Interconnect: SIP trunks to ITSPs, using SIP signaling for inbound and outbound calls.
- Trusted access: Private, managed IP networks that connect service providers' residential, enterprise, or mobile subscribers (as part of an emerging federation of trusted networks).
- Untrusted access: Unmanaged Internet for connections to work-at-home agents or inbound callers.

AudioCodes Security Solution

The device provides a comprehensive package of security features, which handles the following two main security areas:

- **Securing the Service**: Secures the call services by implementing separation and defense of different network entities (e.g., SIP Trunk, softswitch, and users):
  - Physical separation of networks
  - SRDs per SIP entity (user agent)
  - IP Groups per SIP entity (user agent)
- **Securing the Device:**
  - Ensures that only authorized users can access the device's management interface
  - Protection against attacks on the device from SIP signaling and media (RTP).

    For the SBC application, the device provides built-in protection from Denial of Service (DoS) and Distributed Denial of Service (DDoS) attacks:
    - Prevention of DoS/DDoS SIP flood attacks
    - Defense against TCP/IP vulnerabilities
    - Defense against ICMP flooding
    - Optimal handling of SIP user registration avalanche

- Prevents over-the-top traffic from unknown sources

Due to the vast number and types of potential attacks (some described in the previous section), security of your trusted VoIP network should be your paramount concern. The device provides a rich set of features to support perimeter defense for protecting your trusted network from the un-trusted ones. However, the device's security features and capabilities are only effective if implemented correctly. Improper use of the device for perimeter defense may render the overall security solution ineffective, thereby exposing your network to multiple threats.

The benefits of an IP-based telephony network are quite clear, but so are the threats and security implications that need to be addressed. The IP borders of the IP telephony network are the attack points and it's AudioCodes security solutions that are designed to help safeguard your trusted network from such threats.

2 Separate Network Traffic

This chapter provides recommendations for separating network traffic.

Identify Trusted and Un-trusted Networks

It's crucial that you identify the trusted network (i.e., your local LAN) and the un-trusted network (i.e., public Internet – WAN) in the environment in which the device is deployed. There may be multiple un-trusted networks in a single deployment environment. For example, far-end WAN users and a SIP trunk with an ITSP may represent two un-trusted networks.

Once identified, you need to handle the un-trusted networks with extreme caution in order to safeguard your trusted network from malicious attacks from them. One of the main precautions is to separate your trusted network from the un-trusted network, using different logical configuration entities such as SRDs etc. The precautions and security guidelines are described in detail in subsequent sections.

Implement Physical Network Separation using Ethernet Port Groups

For the devices mentioned in the note above, you can physically separate the network traffic by Ethernet ports, using Ethernet Groups. Each Ethernet Group can include up to two physical Ethernet ports. The Ethernet Device defines the VLAN per Ethernet Group. The Ethernet Device is then assigned to the network interface as an Underlying Device. The following procedure provides an example of assigning different ports per traffic type.

➢ **To implement physical network separation:**

1. Open the Ethernet Groups table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **Ethernet Groups**), and then assign ports to different Ethernet Group:

INDEX	NAME	MODE	MEMBER 1	MEMBER 2
0	GROUP_1	Single	GE_4_1	--
1	GROUP_2	Single	GE_4_2	--
2	GROUP_3	Single	GE_4_3	--

2. Open the Ethernet Devices table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **Ethernet Devices**), and then configure VLAN IDs per Ethernet Group:

INDEX	VLAN ID	UNDERLYING INTERFACE	NAME	TAGGING
0	1	GROUP_1	vlan 1	Untagged
1	2	GROUP_2	vlan 2	Untagged
2	3	GROUP_3	vlan 3	Untagged

3. Open the IP Interfaces table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **IP Interfaces**), and then assign the Ethernet Devices (VLANs) to the different traffic network interfaces:

INDEX	NAME	APPLICATION TYPE	INTERFACE MODE	IP ADDRESS	PREFIX LENGTH	DEFAULT GATEWAY	PRIMARY DNS	SECONDARY DNS	ETHERNET DEVICE
0	O+M+C	OAMP	IPv4 Manual	10.15.7.96	16	10.15.0.1	0.0.0.0	0.0.0.0	vlan 1
1	RTP	Media	IPv4 Manual	10.15.7.9	16	10.15.0.1	0.0.0.0	0.0.0.0	vlan 2
2	SIP	Control	IPv4 Manual	10.15.7.99	16	10.15.0.1	0.0.0.0	0.0.0.0	vlan 3

3 Implement Layer 3/4 (Network) Firewall

This section discusses Layer 3/4 (Network) firewall recommendations. By default, there are no firewall rules and therefore, configuring firewall rules is recommended to protect the device from external attacks.

Block Unused Network Ports

It's recommended that you disable network ports that are not needed in your deployment. For example, if you don't need TFTP in your network, then disable this network port application.

Define VoIP Traffic Firewall Rules

For packets whose source IP addresses are known, it's recommended to define VoIP firewall rules that allow receipt of calls or packets from this network and block all calls from elsewhere. These rules can be defined per source IP address, port, protocol, and network IP interface. If an incoming packet is received from an invalid source (as defined in the firewall), the call or packet is discarded.

Below is a list of recommended guidelines when configuring the VoIP firewall:

- Add firewall rules per network interface: It's recommended to configure firewall rules for packets from source IP addresses received on the OAMP interface and each SIP Control (SIP) interface (configured in the IP Interfaces table). A less recommended alternative is to define a single rule that applies to all interfaces (by configuring the 'Use Specific Interface' parameter to **Disable**).
- Define bandwidth limitation per rule: For each IP network interface, it's advised to configure a rate-limiting value (byte rate, burst bytes and maximum packet size). Bandwidth limitation prevents overloading (flooding) of your network and thereby, helps in preventing attacks such as DoS on your device (on each network).
- Define rules as specific as possible: Define the rules as detailed as possible so that they block only the intended traffic.
- Add an ICMP firewall rule: ICMP is typically used for pinging. However, malicious attackers can send over-sized (floods) ICMP packets to a specific network address. Therefore, it's recommended to define a rule for limiting these packets.
- Add a rule to block all traffic: You must define a firewall rule that blocks all incoming traffic (i.e., block all protocol traffic from all source IP addresses and ports for all interfaces). This rule must be the last rule listed in the table, so that rules above it that allow specific traffic are valid (otherwise, all traffic is blocked).

- If the 'Prefix Length' field on the Firewall Settings page is set to "0", the rule will apply to all IP addresses, regardless of whether an IP address is specified in the 'Source IP' field. Thus, if you need to apply a rule to a specific IP address, make

sure that you also set the 'Prefix Length' field to a value other than "0".

- The device provides built-in firewall rules that allow High Availability (HA) traffic between Active and Redundant devices on the Maintenance network interface.

The Layer 3-4 VoIP traffic firewall rules are configured in the Firewall table (**Setup** menu > **IP Network** tab > **Security** folder > **Firewall**). The following table shows a configuration example of firewall rules:

Table 3-1: Configuration Example of Firewall Rules in the Firewall Table

Parameter	Index				
	1	2	3	4	5
Match					
'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 / 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	None
Action					
'Byte Rate'	0	0	40000	40000	0
'Burst Bytes'	0	0	50000	50000	0
'Action Upon Match'	Allow	Allow	Allow	Allow	Block

- Index 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.

- Index 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.
- Index 4: Allows traffic from the LAN voice interface and limits bandwidth.
- Index 5: Blocks all other traffic.

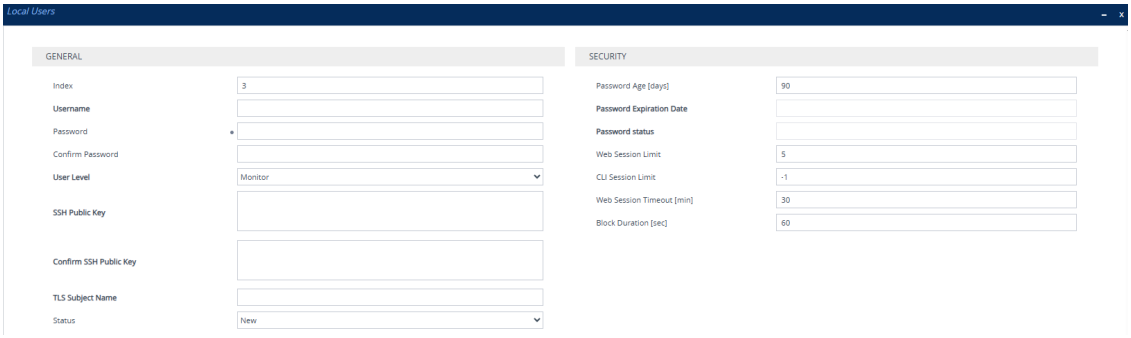
4 Secure Management Access

This section provides guidelines for securing access to the device's management interface.

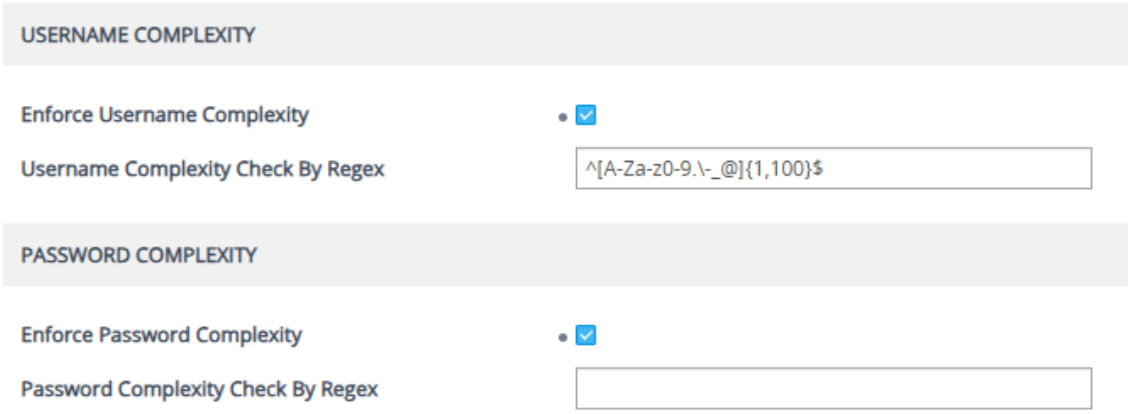
Change Default Login Passwords

To secure access to the device's Web management interface, please adhere to the following recommended guidelines:

- The device is shipped with a default **Security Administrator** access-level user account with username **Admin** and password **Admin**. This user has full read-write access privileges to the device. It's recommended to change this default password to a hard-to-hack string. You can change the username and password in the Local Users table (**Setup** menu > **Administration** tab > **WEB & CLI** folder > **Local Users**):



- Enforce username and password complexity. Instead of using the device's default complex policy, you can configure a customized complex policy based on a regular expression (regex). Username and password complexity are configured on the Local Users Settings page (**Setup** menu > **Administration** tab > **WEB & CLI** folder > **Local Users Settings**):



- The device is shipped with a default **Monitor** access-level user account with username **User** and password **User**. This user has read-only privileges to the device. The read access privilege is also limited to certain Web pages. However, this user can view certain SIP settings such as proxy server addresses. Therefore, to prevent an attacker from obtaining

sensitive SIP settings that could result in possible call theft etc., either **delete** this user account or change its default login password to a hard-to-hack string.

- If you have deployed multiple devices, use a unique password for each device.
- Change the login password periodically (e.g., once a month). It's recommended to configure users with a password age. This is done in the Local Users table ('Password Age' parameter).

User Authentication by Third-Party Server

For securing access to the device, it's recommended to implement a third-party authentication server.

LDAP-based User Authentication and Authorization

You can implement a third-party, LDAP server in your network for authenticating and authorizing the device's management users (Web and CLI). This can be done by using an LDAP-compliant server such as Microsoft Active Directory (AD). When a user attempts to log in to one of the management platforms, the device verifies the login username and password with AD. The device can also determine the user's management access level (privileges) based on the user's profile in the AD. This is configured in the LDAP pages located under **Setup** menu > **IP Network** tab > **AAA Servers** folder.

An alternative to using an LDAP server is to use a RADIUS server, as discussed in the next section.

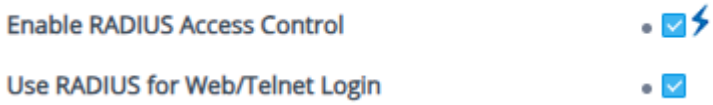
RADIUS-based User Authentication

You can implement a third-party, RADIUS server in your network for authenticating Web / Telnet management users and thereby, preventing unauthorized access. RADIUS allows you to define different passwords for different interface users, with centralized management of the password database. When RADIUS is used, logging into the Web / Telnet interfaces is performed through the RADIUS server. The device verifies the authenticity of the username and password with the RADIUS server.

An alternative is to use an LDAP server, as discussed in the previous section.

➢ **To enable RADIUS-based user authentication:**

1. Open the Authentication Settings page (**Setup** menu > **Administration** tab > **WEB & CLI** folder > **Authentication Settings**), and then select the check boxes of the following parameters:
   - 'Enable RADIUS Access Control'
   - 'Use RADIUS for Web/Telnet Login'



2. Click **Apply**, and then restart the device with a burn-to-flash for your settings to take effect.
3. Open the RADIUS Servers table (**Setup** menu > **IP Network** tab > **AAA Servers** folder > **RADIUS Servers**), and then configure the RADIUS authentication server for authenticating the device with the RADIUS server:

INDEX	IP ADDRESS	AUTHENTICATION PORT	ACCOUNTING PORT	SHARED SECRET	INTERFACE NAME
0	10.6.6.7	1645	1646	*	O+M+C

4. You can enhance security for incoming and outgoing RADIUS packets, using RADIUS attribute 80 (Message-Authenticator):
   - **Outgoing RADIUS messages:** You can enable the device (Network Access Server / NAS), using the [RadiusPapRequireMsgAuthTx] parameter, to include the Message-Authenticator attribute in all Access-Request RADIUS packets sent to the RADIUS server. This is applicable only to the Password Authentication Protocol (PAP) authentication method.
   - **Incoming RADIUS messages:** You can enable the device, using the [RadiusRequireMsgAuthRx] parameter, to require the presence of the Message-Authenticator attribute in all incoming Accept-Accept RADIUS messages from the RADIUS server. If the attribute is not present, the device rejects the message and denies user login. This is applicable to Digest and PAP authentication methods.

OAuth 2.0 Authentication using Microsoft Entra ID

You can implement Microsoft's Entra ID (formerly Azure Active Directory) to authenticate (credentials) and authorize (privilege level) users attempting to log in to the device's management interfaces (Web interface, CLI, and REST API). Authentication is done using the OAuth 2.0 protocol.

OAuth authentication is configured in the OAuth Servers table and Login OAuth Servers table. However, for full configuration details, refer to the User's Manual.

Implement Two-Way Authentication with X.509 Certificates

It's recommended to use two-way authentication (in addition to HTTPS) between the device's Web server and the client accessing it. Authentication is performed and connection to the Web interface is subsequently allowed only if the following conditions are met:

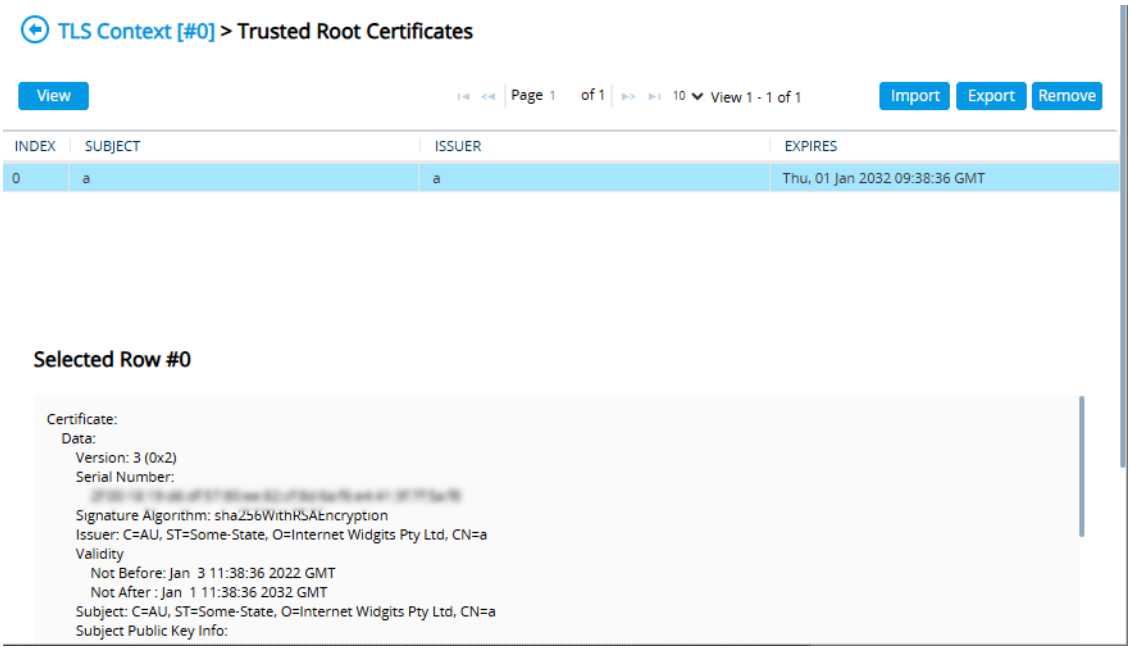
- The client possesses a client certificate from a Certification Authority (CA).
- The CA certificate is listed in the device's Trusted Root CA Store.

Otherwise, the connection is rejected, preventing unauthorized access to the Web management tool.

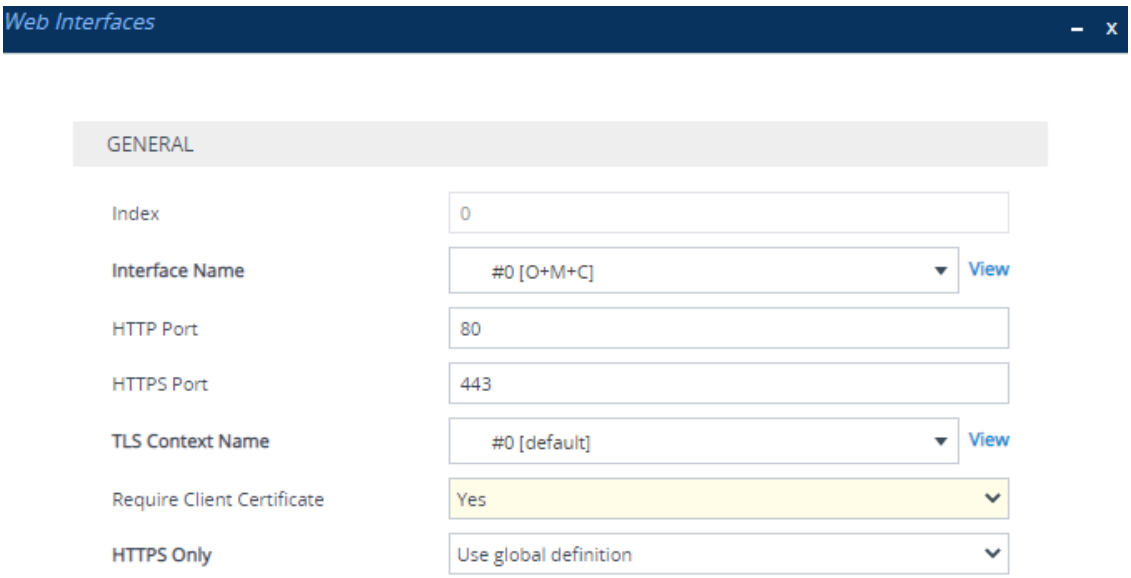
- Implementation of two-way authentication requires a third-party security equipment vendor, CA server, and security administrator personnel. These should create certificates and deploy them to all the computers in the organization.
- The device is supplied with a working TLS configuration consisting of a unique self-signed server certificate. Replace this certificate with one provided by your security administrator. For more information, refer to the User's Manual.
- For management through the device's REST API, secure access can be based on the certificate's subject name. When a client attempts to connect to the REST interface over TLS, the device checks if the subject name in the client's certificate matches a TLS subject name configured ('TLS Subject Name') for any user in the Local Users table. If a match is found, the client is automatically authenticated, without needing to provide username or password. The client is granted the user level associated with the matching user in the Local Users table.

➢ To configure client-server, two-way authentication using X.509 certificates:

1. Install a client certificate on the management station (your network administrator should provide you with a certificate).
2. Install your organization's CA certificate on the management station.
3. Open the TLS Contexts table (**Setup** menu > **IP Network** tab > **Security** folder > **TLS Contexts**).
4. In the TLS Contexts table, add a new TLS Context or select the required TLS Context row, and then click the **Trusted Root Certificates** link located at the bottom of the TLS Contexts page.
5. Click the **Import** button, select the Root CA certificate file (in base64-encoded PEM format), and then click **OK** to import the file:



6. Since X.509 certificates have an expiration date and time, the device must be configured to use Network Time Protocol (NTP) to obtain the current date and time. Without the correct date and time, client certificates cannot operate.
7. Make sure that client certificates for HTTPS connections are required. Open the Web Interfaces table (**Setup** menu > **Administration** tab > **WEB & CLI** folder > **Web Interfaces**), and then from the 'Require Client Certificate' drop-down list of the OAMP Web interface, select **Yes**:



Secure HTTP Access using HTTPS

It's recommended to allow access to the Web interface through HTTPS only. In addition, it's recommended to block port 80.

➢ **To allow Web access only through HTTPS:**

1. Open the Web Settings page (**Setup** menu > **Administration** tab > **WEB & CLI** folder > **Web Settings**).
2. From the 'Secured Web Connection (HTTPS)' drop-down list, select **HTTPS Only**:

3. Click **Apply**.

Secure Telnet Sessions

It's recommended to disable access through Telnet. However, if you do require Telnet and your management software provides a secure Telnet application, then use a secured Telnet connection (i.e., TLS). TLS protects Telnet traffic from network sniffing.

➢ **To secure Telnet:**

1. Open the Administration Settings page (**Setup** menu > **Administration** tab > **WEB & CLI** folder > **Administration Settings**).
2. From the 'Enable Telnet Server' drop-down list, select **Enable Secured**:

Secure CLI Sessions using SSH

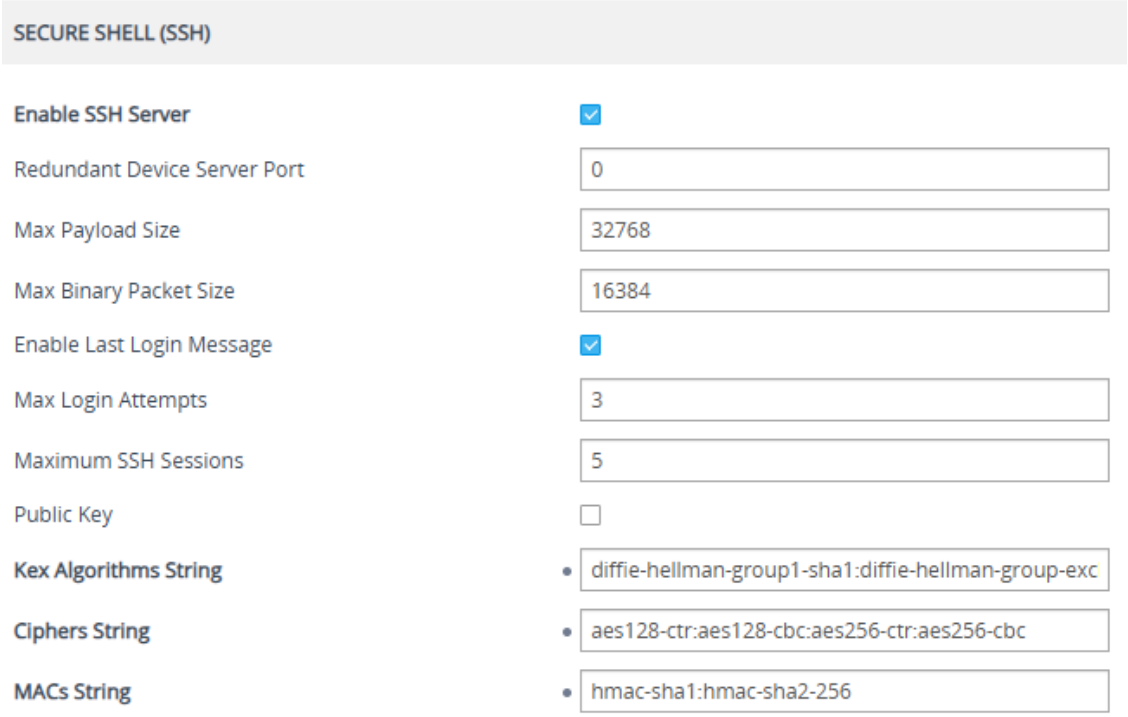
It's recommended to employ Secure SHell (SSH) for accessing the device's CLI. SSH is the de-facto standard for secure CLI. SSH 2.0 is a protocol built above TCP, providing methods for key exchange, authentication, encryption, and authorization. By default, SSH uses the same username and password as the Telnet and Web server.

The device's embedded SSH server supports SHA-256 (rsa-sha2-256) and SHA-512 (rsa-sha2-512) signature algorithms for public-key client authentication that utilizes RSA keys:

- Server host key algorithms (refer to RFC 4253, Section 7.1)
- Algorithm for client authentication (refer to RFC 8303, Section 3.1 and RFC 8332, Section 3.2)

➢ **To enable SSH:**

1. Open the SSH Settings page (**Setup** menu > **Administration** tab > **WEB & CLI** folder > **Administration Settings**).
2. Configure the following parameters:
   - 'Enable SSH Server': Select the check box to enable SSH.
   - 'Kex Algorithms String': Define the Key Exchange Method (e.g., Diffie-Hellman-Group-Exchange-SHA256).
   - 'Ciphers String': Define the cipher string (e.g., AES128-CTR).
   - 'MACs String': Define the HMAC (e.g., HMAC-SHA2-256).



For additional security, you can configure a public key for RSA key negotiation (instead of or in addition to using a username and password) when accessing through SSH.

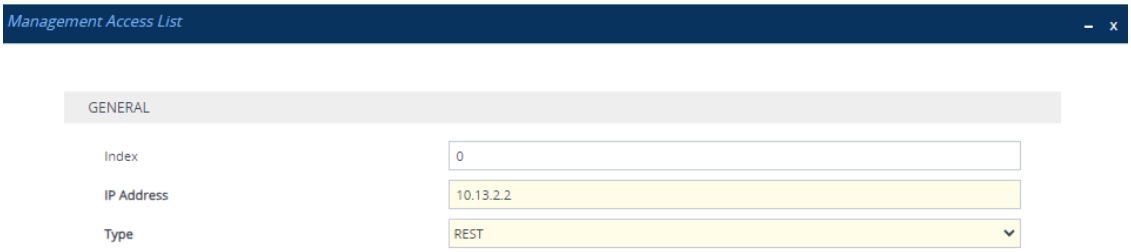
Define Web, REST, Telnet, and SSH Authorized Management Access List

Allow only specific management clients (IP addresses) to access the Web, REST, Telnet, and/or SSH management interface types. The device denies access from undefined IP addresses.

- The first authorized IP address in the list must be your computer's (terminal) IP address; otherwise, access from your computer will be denied.
- The management access list relates to OSI Layer 5 (Session). However, you can also add firewall rules for Layer 3 (Network) and Layer 4 (Transport) with bandwidth limitation to limit access to management interfaces (see Block Unused Network Ports on page 7).

➢ To configure management access list:

1. Open the Management Access List table (**Setup** menu > **Administration** tab > **WEB & CLI** folder > **Management Access List**).
2. Click **New** and configure a management access list rule:



3. Click **Apply**.

Define a Hostname for Accessing Web Interface

It's recommended to configure a hostname for the device's Web interface and enforce access through the hostname only. When enforced, all attempts to access the device through its IP address is blocked. Accessing the device with a hostname instead of an IP address helps protect against HTTP Host header (manipulation) attacks and DNS rebinding attacks.

When there is an attempt to access the device with a hostname, the device checks that the value of the Host header matches the configured hostname. If there is no match, the device rejects the request with an HTTP 403 Forbidden response.

> If you configure a hostname, you also need to define it on a DNS server so that it can be resolved into an IP address.

➢ **To configure hostname for Web interface:**

1. Open the Web Settings page (**Setup** menu > **Administration** tab > **Web & CLI** folder > **Web Settings**).
2. In the 'Web Server Name' field [WebHostname], type a hostname for the device:

3. Select the 'Enforce Web Host Name' check box:

4. Click **Apply**.

Secure SNMP Interface Access

This section discusses recommended security guidelines relating to Simple Network Management Protocol (SNMP).

Prefer SNMPv3 over SNMPv2

It's recommended to use SNMP Version 3 (SNMPv3) instead of SNMPv1 and SNMPv2c, if possible. SNMPv3 provides secure access to the device using a combination of authentication (e.g., MD5, SHA-1 or SHA-2) and encryption (e.g., DES, 3DES, AES-128, AES-192, or AES-256) of packets over the network. It's also recommended that you periodically change the SNMPv3 authentication and privacy keys.

➢ To configure SNMPv3 users:

1. Open the SNMPv3 Users table (**Setup** menu > **Administration** tab > **SNMP** folder > **SNMPv3 Users**).
2. Click **New**, and then configure an SNMPv3 user:

INDEX	USER NAME	AUTHENTICATION PROTOCOL	PRIVACY PROTOCOL	AUTHENTICATION KEY	PRIVACY KEY	GROUP
0	JoeD	MD5	3DES	*	*	Read-Write

Secure SMNPv2 Access

If you are using SNMPv2, change the community strings from their default values as they can easily be guessed by hackers. The default read-write community string is "private" and the read-only is "public".

In addition, by default, the SNMPv2 agent accepts SNMP Get and Set requests from any IP address if the correct community string is used in the request. Therefore, to enhance security with SNMPv2, implement Trusted Managers. A Trusted Manager is an IP address (management station) from which the SNMP agent accepts and processes Get and Set requests. It's also recommended that you periodically change these SNMP community string values.

> It's recommended to use SNMPv3 users, as described in Prefer SNMPv3 over SNMPv2 above.

➢ To secure SNMPv2:

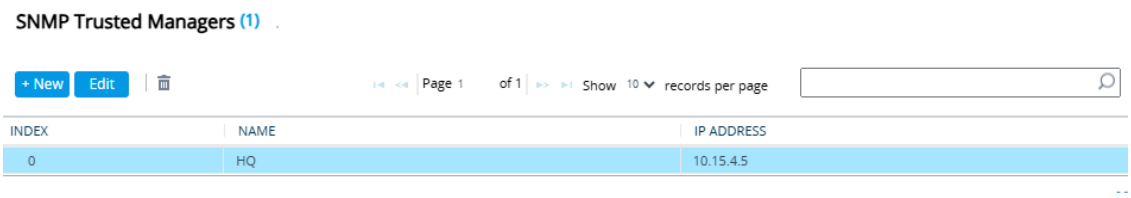
1. Open the SNMP Community Strings table (**Setup** menu > **Administration** tab > **SNMP** folder > **SNMP Community Strings**).
2. Click **New**, and then configure the SNMPv2 community strings:

SNMP Community Strings (1)

+ New | Edit | Page 1 of 1 | Show 10 records per page

INDEX	NAME	PASSWORD	GROUP
0	HQ	*	Read-Write

3. Open the SNMP Trusted Managers table (**Setup** menu > **Administration** tab > **SNMP** folder > **SNMP Trusted Managers**).
4. Click **New**, and then configure an SNMPv2 management station:

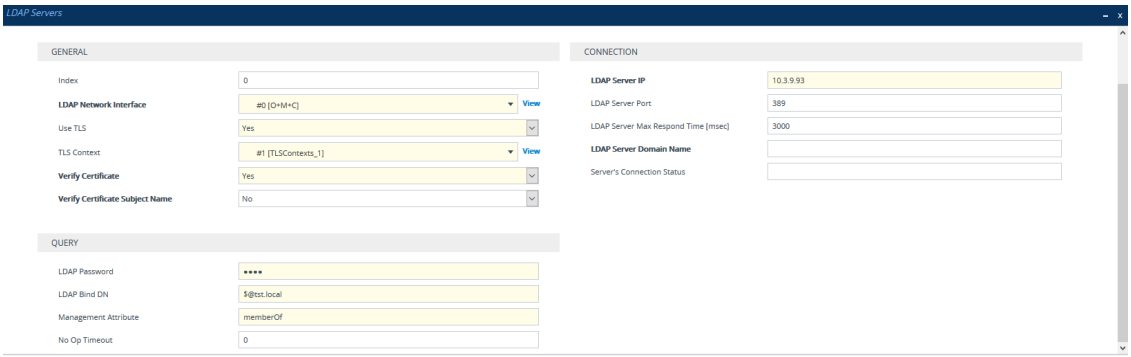


Secure LDAP Communication

If you are using LDAP-based login management (username-password) and/or LDAP-based SIP routing in your deployment, it's recommended to employ TLS for secure device communication with the LDAP server. This ensures that the device encrypts the username and password sent to the LDAP server.

➢ To secure LDAP-based applications:

1. Open the LDAP Servers table (**Setup** menu > **IP Network** tab > **AAA Servers** folder > **LDAP Servers**).
2. For the relevant LDAP server, configure the following:
   - From the 'Use TLS' drop-down list, select **Yes**.
   - From the 'TLS Context' drop-down list, select the TLS Context from the TLS Contexts table.



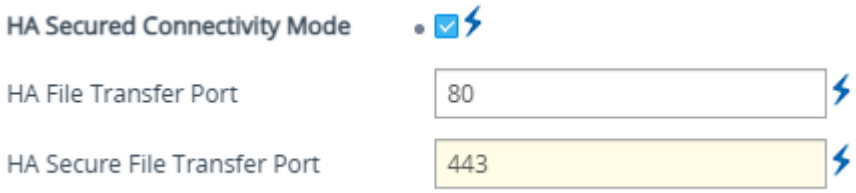
Secure Communication between HA devices

It's recommended to secure communication between the active and redundant devices in the High-Availability (HA) system. HA communication occurs through the device's Maintenance IP Interface, which is used for HA synchronization, including file transfer between devices.

➢ To secure file transfer between HA devices:

1. Open the HA Settings page (**Setup** menu > **IP Network**tab > **Core Entities** folder > **HA Settings**).
2. Select the 'HA Secured Connectivity Mode' check box.

3. In the 'HA Secure File Transfer Port' field, type the local port number for secured communication.



4. Click **Apply**, and then restart the device with a burn-to-flash for your settings to take effect.

This section is applicable only to devices supporting HA mode.

Store TLS Private Keys Encrypted

You can enable the device to store all TLS private keys in encrypted format, enhancing security.

- **CLI:** `configure network > security-settings > encrypt-private-key-files`
- **Ini File:** [EncryptPrivateKeyFiles]

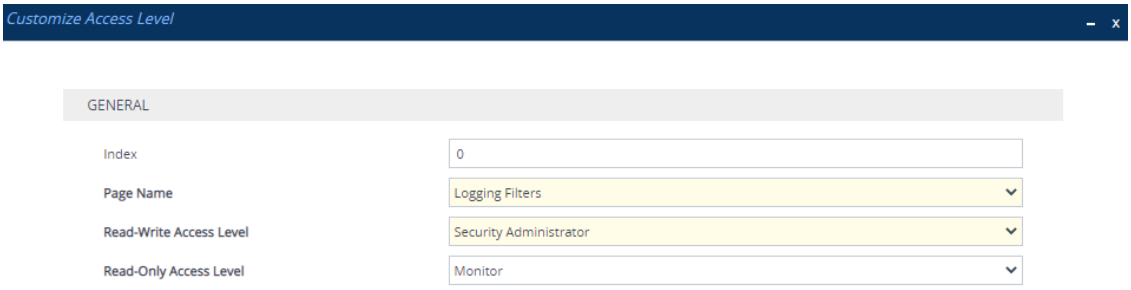
This section is applicable only to Mediant Software.

Customizing Access Levels per Web Page

You can overwrite the default access privileges (read-only or read-write) per user level (**Monitor**, **Administrator**, or **Security Administrator**) per Web interface page.

➢ **To customize access levels per Web page:**

1. Open the Customize Access Level table (**Setup** menu > **Administration** tab > **Web & CLI** folder > **Customize Access Level**).
2. Configure customization rules. For example, the configuration below allows only **Security Administrator** users to configure the Logging Filters page, while allowing users with **Monitor** level to view only.



5 Secure SIP using TLS (SIPS)

It's crucial that you implement the TLS-over-TCP protocol to secure the device's SIP signaling connections. TLS provides encryption and authentication of SIP signaling for your VoIP traffic, preventing tampering of calls. Use it whenever possible for far-end users and ITSPs.

The device's TLS feature supports the following:

- TLS: TLS 1.0, TLS 1.1, TLS 1.2, and TLS 1.3
- DTLS: DTLS 1.0 and DTLS 1.2
- Cipher: TLS cipher suites for server and client roles (per OpenSSL syntax)
- Authentication: X.509 certificates
- Certificate revocation checking: OCSP (CRLs are currently not supported)
- Receipt of wildcards ('*') in X.509 Certificates when establishing TLS connections. These wildcards can be part of the CN attribute of the Common Name field or the DNSName attribute of the Subject Alternative Name field.
- Authority Information Access (AIA) field in X.509 Certificate Signing Requests (CSRs). The AIA extension provides a URL that clients can use during the TLS handshake to verify the validity of the device's certificate

Recommended security guidelines for ensuring TLS for SIP signaling are described in the subsequent subsections.

Use Strong Authentication Passwords

Always use strong authentication passwords, which are more difficult to detect than weak ones. A strong password typically includes at least six characters with a combination of upper and lower-case letters, numbers and symbols.

Use TLS Version 1.2 or 1.3

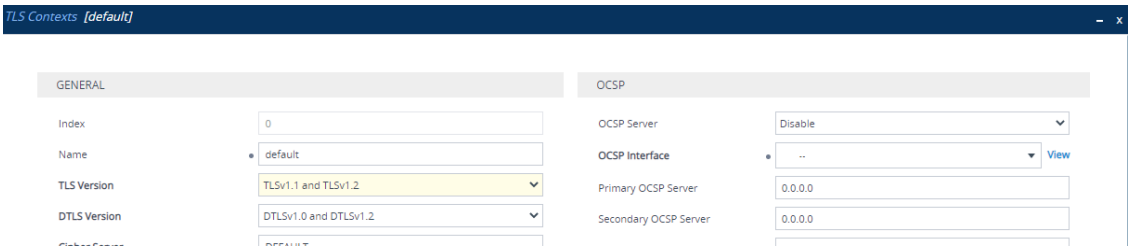
It's recommended to use the highest TLS version that is supported by all your network entities to achieve the best communication security, based on cryptographic algorithms. The device accepts only connections that adhere to the specified TLS version.

It's also recommended not to configure the device to use any TLS version (**Any TLS1.x**). However, if some network entities use SSL 3.0 handshakes and some use a higher TLS version (e.g., TLS 1.1), then you need to configure the device to use any version.

➢ **To configure the TLS version:**

1. Open the TLS Contexts table (**Setup** menu > **IP Network** tab > **Security** folder > **TLS Contexts**).

2. For the relevant TLS Context, from the 'TLS Version' drop-down list, select the required TLS version. The example below assumes that the highest TLS versions supported by the network entities are 1.1 and 1.2.



Block Multiple Client-Initiated TLS Renegotiations

The device can block client-initiated TLS renegotiations (handshakes). This is useful for preventing DoS attacks on the device caused by multiple TLS renegotiations per second of the encrypted key initiated by the attacker.

➢ **To block multiple client-initiated TLS renegotiations:**

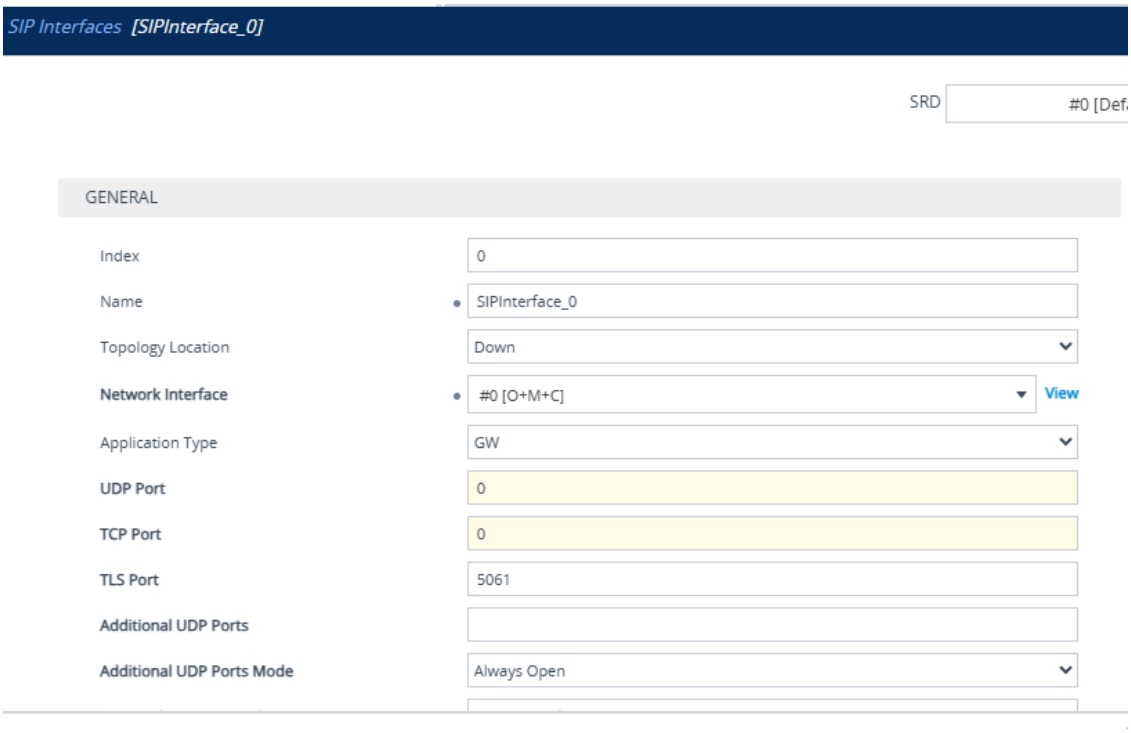
1. Open the TLS Contexts table (**Setup** menu > **IP Network** tab > **Security** folder > **TLS Contexts**).
2. For the relevant TLS Context, from the 'TLS Renegotiation' drop-down list, select **Disable:**

Use TLS for SIP Interfaces and Block TCP/UDP Ports

Each port can be vulnerable to attacks. Therefore, it's highly recommended that your SIP interfaces use only TLS. When configuring your SIP Interfaces, define the TLS port number, but set the UDP and TCP ports to zero ("0"). This configuration blocks (disables) the UDP and TCP ports. In other words, to disable UDP and TCP ports, you must define SIP Interfaces. In addition, to increase security, define only SIP Interfaces that are necessary.

➢ **To configure TLS for SIP Interfaces:**

1. Open the SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **SIP Interfaces**).
2. For the relevant SIP Interface, configure the following:
   - In the 'UDP Port' field, enter 0.
   - In the 'TCP Port' field, enter 0.
   - In the 'TLS Port' field, enter a port number (non-zero).

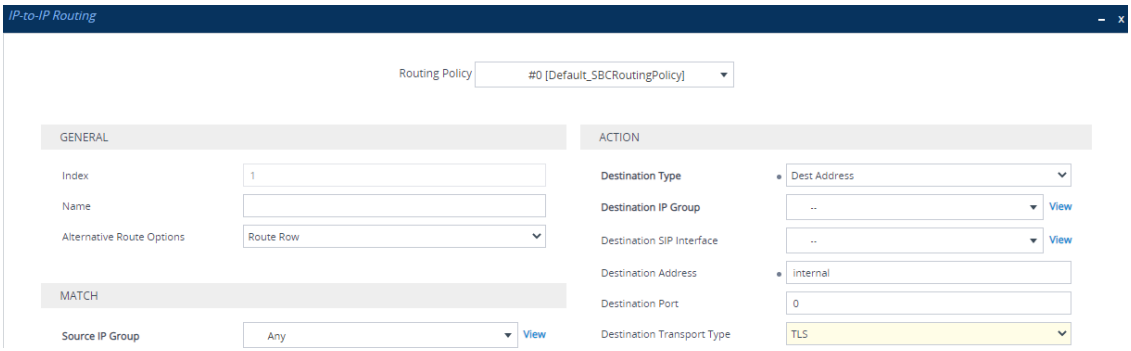


Use TLS for Routing Rules

It's recommended that your routing rules use TLS as the transport type.

➢ **To enable TLS for routing rules:**

1. Open the IP-to-IP Routing table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Routing** > **IP-to-IP Routing**).
2. For the relevant IP-to-IP Routing rule, from the 'Destination Transport Type' drop-down list, select **TLS**:



Implement X.509 Certificates for SIPS (TLS) Sessions

It's highly recommended to implement the X.509 certificate authentication mechanism for enhancing and strengthening TLS. X.509 is an ITU-T standard for Public Key Infrastructure (PKI).

The device supports the configuration of multiple TLS certificates, referred to as TLS Contexts. TLS Contexts are assigned to Proxy Sets and/or SIP Interfaces, thereby enabling specific calls to use specific TLS certificates.

The device is shipped with a working TLS configuration (TLS Context ID 0), consisting of a unique Self-Signed Server Certificate. Self-Signed Certificate is the simplest form of an X.509 Certificate that is issued by the device itself without the use of any certificate signer (CA). The Self-Signed Certificate consists of the Public Key of the device that is signed by the Private Key of the device itself. However, use of this certificate is strongly discouraged. The Self-Signed Certificate is typically used in testing environments or for a low-scale deployment where solution security may be sacrificed in favor of simplified configuration procedures. The Self-Signed Certificate does not utilize CA trust relationships and its authenticity cannot be reliably verified. Instead, you should establish a PKI for your organization (provided by your security administrator) and use certificates signed by genuine CAs.

In a typical PKI scheme, Certificates are issued by a CA and provide an attestation by the CA that the identity information and the public key belong together. Each party has a list of Trusted Root Certificates – certificates of the CAs (or their roots) that are well-known and trusted by the party. When the certificate from the other party is received, its signing entity (CA) is compared with the Trusted Root Certificates list and if a match is found, the certificate is accepted.

The device uses the following files to implement X.509 PKI:

- **Private Key File:** This file contains a private key that is used to perform decryption. It's the most sensitive part of security data and should never be disclosed to other entities.
- **Certificate File:** This file contains a digital signature that binds together the Public Key with identity information. The Certificate may be issued by a CA or self-signed (issued by the device itself, which is not recommended – see above).
- **Trusted Root Certificate File:** This file is the certificate of the Trusted Root CA used to authorize certificates received from remote parties, based on the identity of the CA that issued it. If the root certificate of this CA matches one of the Trusted Root Certificates, the remote party is authorized.

Use an NTP Server

It's recommended to implement a third-party NTP server so that the device receives the correct current date and time. This is necessary for validating certificates of remote parties. It's also recommended to enable the device to authenticate and validate messages received from the NTP server. Authentication is done using an authentication key with the MD5 cryptographic hash algorithm. NTP messages that are received without authentication are ignored.

➢ To implement NTP server:

1. Open the Time & Date page (**Setup** menu > **Administration** tab > **Time & Date** home icon).
2. select the 'Enable NTP' check box:

OAuth 2.0-based Authentication for SIP Requests using Microsoft Entra ID

You can implement Microsoft Entra ID (formerly Azure Active Directory) to authenticate SIP User Agents (UA) of incoming SIP messages (including WebRTC), based on the OAuth 2.0 protocol.

Entra ID is Microsoft's cloud-based identity and access management service, designed for Internet-based applications. As Entra ID doesn't support OAuth Token Introspection, the device validates the received token using its embedded NGINX server, which simulates an OAuth 2.0 Introspection endpoint.

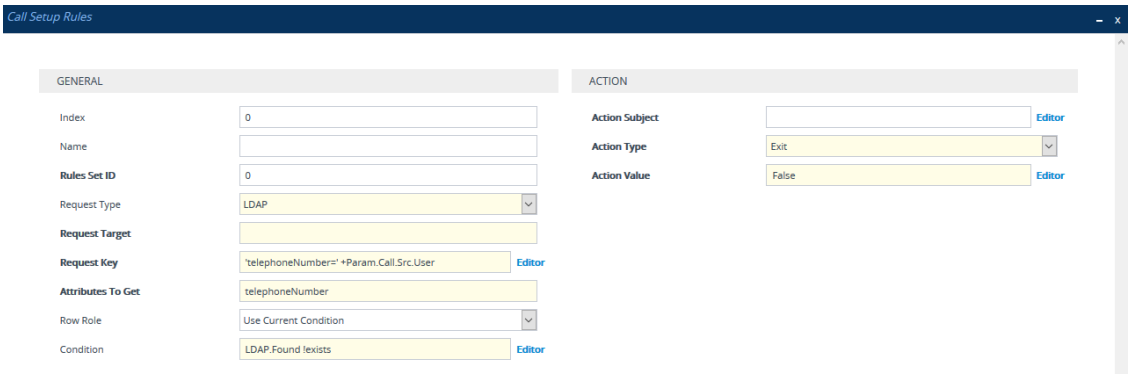
For configuring OAuth 2.0-based authentication of SIP messages, refer to the User's Manual.

6 Implement LDAP-based Conditional Call Routing

It's recommended that you implement a third-party, LDAP server in your network for determining whether a call from a specific source is permitted to be routed to its destination. This setup uses Call Setup rules, configured in the Call Setup Rules table, to define a condition-based script that queries an LDAP server for the caller's number (for example) in a specific LDAP attribute. If the number exists, the device routes the call to the destination; otherwise, the call is dropped. The device executes a Call Setup rule upon the receipt of an incoming call (dialog) at call setup if a matching routing rule exists in the IP-to-IP Routing table, before the <device> routes the call to its destination.

➢ To configure LDAP-based conditional routing:

1. For configuring LDAP, use the LDAP Settings page, LDAP Server Groups table, and LDAP Servers table (**Setup** menu > **IP Network** tab > **AAA Servers** folder).
2. For configuring Call Setup rules, use the Call Setup Rules table (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Call Setup Rules**). The below Call Setup rule example routes the incoming call only if the source number of the incoming call 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 routes the call to the destination as specified in the IP-to-IP Routing table. If the query fails (i.e., source number doesn't exist in AD server), the device rejects the call.



Make sure that you implement secure LDAP communication, as discussed in Section Secure LDAP Communication on page 19.

7 Define SIP Message Blocklist and Allowlist

It's recommended to configure SIP message policy rules for blocking (blocklist) unwanted incoming SIP messages or allowing (allowlist) receipt of desired messages. This allows you to define legal and illegal characteristics of a SIP message.

SIP message policy 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 an oversized parameter or too many occurrences of a parameter.

Each SIP message policy rule can be configured with, for example, maximum message length, header length, body length, number of headers, and number of bodies. Each rule is then set as a blocklist or allowlist.

➢ To configure SIP message blocklists and allowlists:

1. Open the Message Policies table (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Policies**).
2. Click **New** to configure a rule.

   The following displays an example of a configured rule that defines maximum SIP messages to 32,768 characters, maximum header length to 512 characters, and bodies to 1024 characters. Invalid requests are rejected. Only INVITE and BYE requests are permitted.

INDEX	NAME	MAX MESSAGE LENGTH	MAX HEADER LENGTH	MAX BODY LENGTH	SEND REJECTION
0	Malicious Signature DB Pr	-1	-1	-1	Policy Drop
1	MessagePolicy_1	32768	512	1024	Policy Reject

8 Monitor and Log Events

It's recommended that you log and monitor device events (including device operations and calls). The importance of monitoring device events is that you can quickly detect unauthorized access and subsequently take counter measures to effectively terminate the attacker before any potential damage is done to your network.

Implement Dynamic Blocklisting of Malicious Activity (IDS)

It's important to use the device's Intrusion Detection System feature (IDS) to enable the device to detect malicious attacks targeted on the device (e.g., DoS, SPAM, and Theft of Service). It's crucial to be aware of any attacks to ensure that the legitimate call service is always maintained. If any user-defined attacks are identified, the device can do the following:

- Block (blocklist) remote hosts (IP addresses / ports) considered as malicious. The device automatically blocks the malicious source for a user-defined period after which it's removed from the blocklist.
- Send SNMP traps to notify of the malicious activity and/or whether an attacker has been added to or removed from the blocklist.

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 (alarm threshold) during an interval (threshold window) 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).

➢ **To configure IDS:**

1. Open the IDS General Settings page (**Setup** menu > **Signaling & Media** tab > **Intrusion Detection** folder > **IDS General Settings**), and then select the 'Intrusion Detection System (IDS)' check box:

2. Open the IDS Policies table (**Setup** menu > **Signaling & Media** tab > **Intrusion Detection** folder > **IDS Policies**), and then configure an IDS policy "ITSP DoS", as shown selected below:

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
3	ITSP DoS	Denial of Service

3. Open the IDS Rule table by clicking the **IDS Rule** link located below the IDS Policies table, and then configure IDS rules for the "ITSP DoS" IDS policy:

INDEX	REASON	THRESHOLD SCOPE	THRESHOLD WINDOW	MINOR-ALAF THRESHOLD	MAJOR-ALAF THRESHOLD	CRITICAL-AL THRESHOLD	DENY THRESHOLD	DENY PERIOD
0	Malformed message	Global	30	10	15	30	-1	-1
1	Connection abuse	Global	20	-1	70	-1	-1	-1
2	Authentication failure	Global	1	-1	5	-1	-1	-1

4. Open the IDS Matches table (**Setup** menu > **Signaling & Media** tab > **Intrusion Detection** folder > **IDS Matches**), and then assign the IDS Policy to a specific SIP interface and subnet:

INDEX	SIP INTERFACE ID	PROXY SET ID	SUBNET	POLICY
0	3			ITSP DoS
1			10.33.0.0/16	ITSP DoS

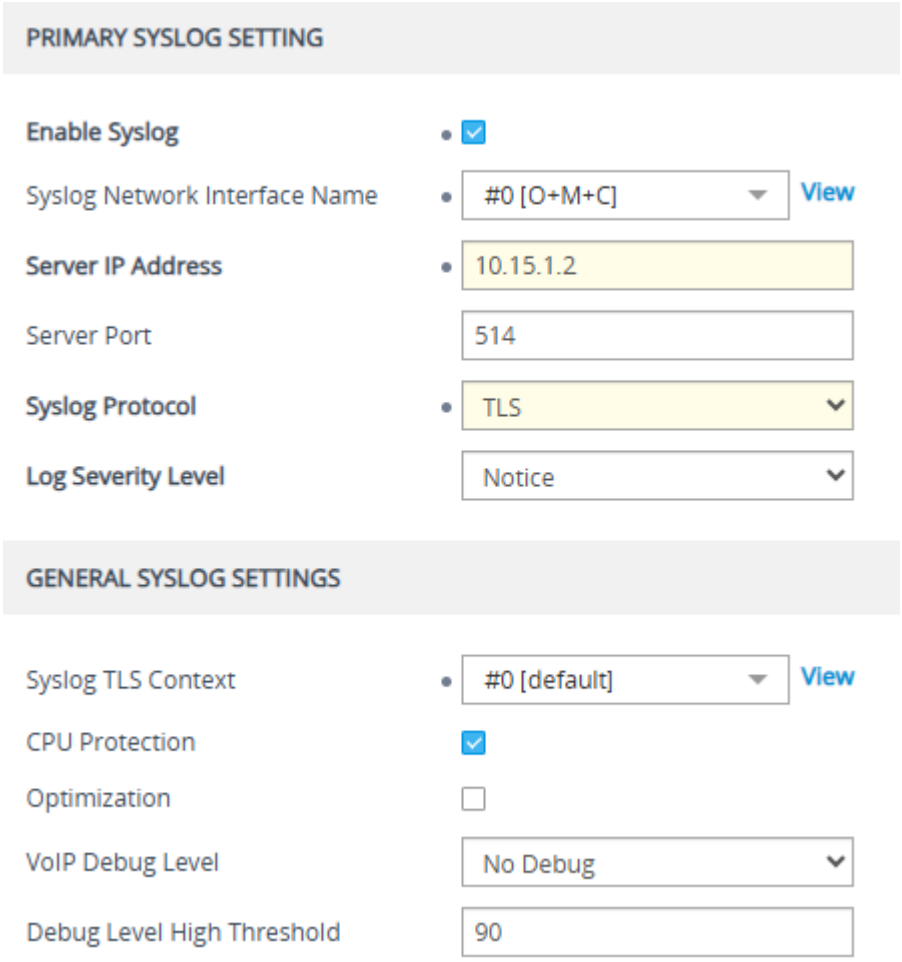
Enable Syslog

The device supports generation and reporting of syslog messages and SNMP traps to external logging servers. It's crucial that you enable one or both these features (preferably syslog) so that you can monitor events on your device. In addition, as the device does not retain logged reports (SNMP is limited), it's recommended that you make sure that your syslog server saves all logged events for future analysis and reference.

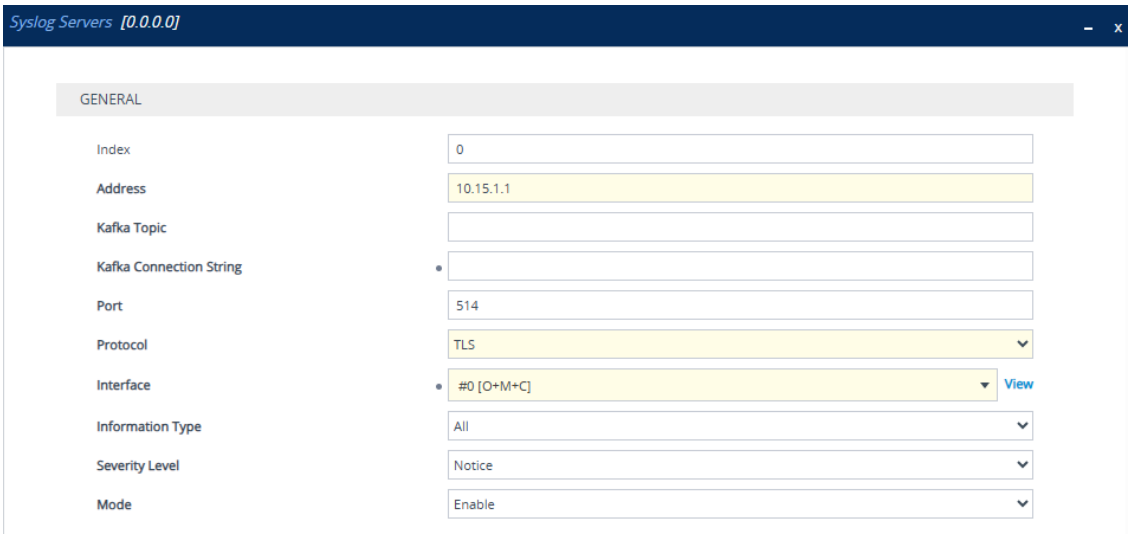
It's also recommended that you configure the device to communicate with the syslog server over TLS to secure the connection.

➢ To enable syslog:

1. Open the Logging Settings page (**Troubleshoot** menu > **Troubleshoot** tab > **Logging** folder > **Logging Settings**).
2. Select the 'Enable Syslog' check box, and then configure the relevant parameters:



3. Open the Syslog Servers table (**Troubleshoot** menu > **Troubleshoot** tab > **Logging** folder > **Syslog Servers**), and then configure a syslog server(s):



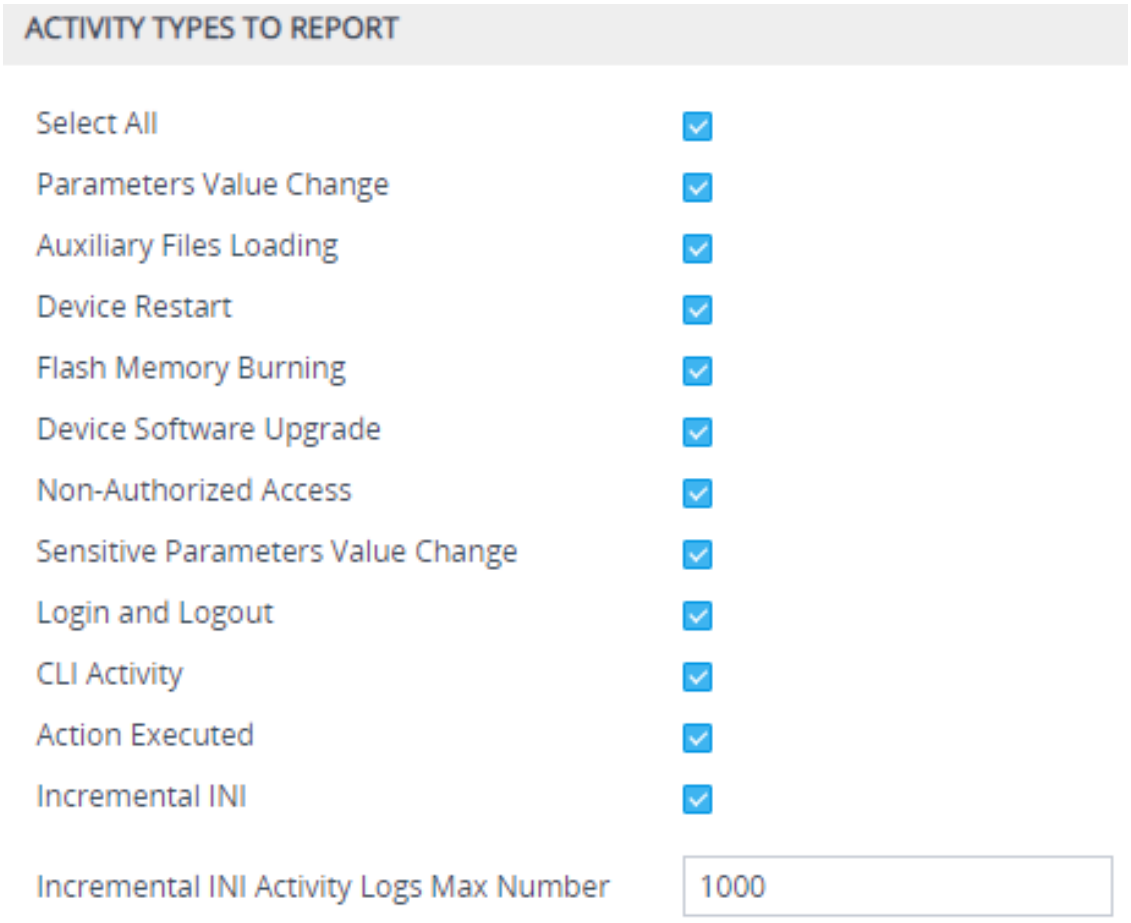
Enable Logging of Management-Related Events

Through syslog you can log and monitor management-related events to help you detect and identify unauthorized management-related activities such as:

- Unauthorized Web login attempts (attempts to access the Web interface with a false or empty user name or password)
- Access to restricted Web pages such as the page on which firewall rules are defined
- Modifications to parameter values (for example, deletion of firewall rules, allowing future unauthorized access)
- Modifications to "sensitive" parameters - changes made to important parameters such as IP addresses
- Unauthorized SIP messages (logged SIP messages)

➢ **To log management-related events:**

1. Open the Logging Settings page (**Troubleshoot** menu > **Troubleshoot** tab > **Logging** folder > **Logging Settings**).
2. Select the type of events that you want logged:



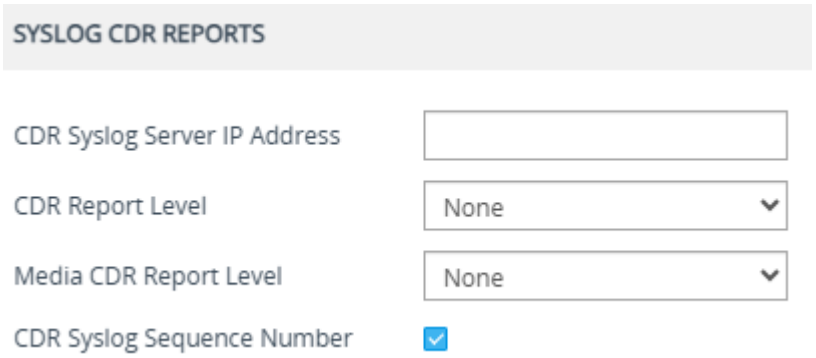
Enable Call Detail Records

Call Detail Records (CDR) provide vital information on SIP calls made through the device. This information includes numerous attributes related to the SIP call such as port number, physical channel number, source IP address, call duration, and termination reason. The device can be

configured to generate and report CDRs for various stages of the call (beginning, initial connection, and end of the call). Once generated, the CDR logs are sent to a user-defined logging server.

➢ To enable CDR generation:

1. Open the Advanced Parameters page (**Troubleshoot** menu > **Troubleshoot** tab > **Call Detail Record** folder > **Call Detail Record Settings**).
2. Configure the relevant parameters:



For CDRs, you must enable syslog functionality.

9 GDPR for Protecting Personal Information

To help you comply with the European Union's (EU) General Data Protection Regulation (GDPR) to protect and respect personal data processed by the device, the device offers various means to mask (hide) personally identifiable information (PII).

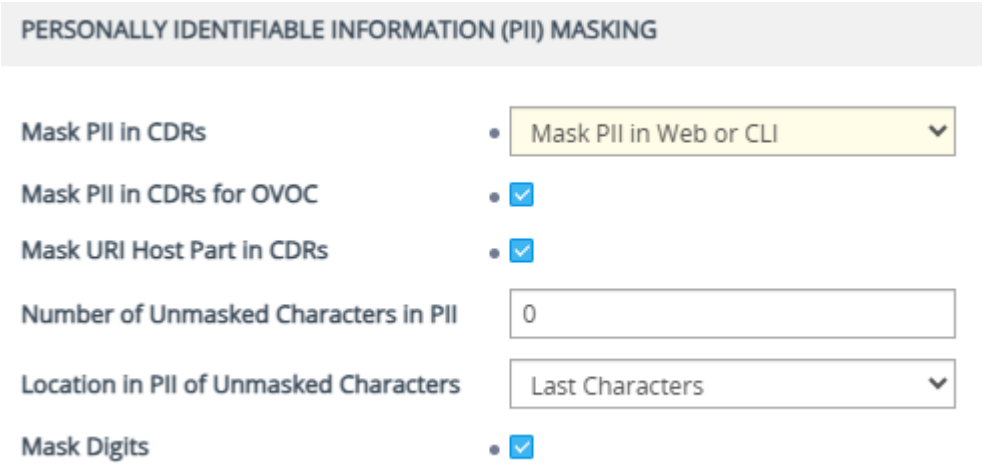
Masking PII

- AudioCodes PII Log Scrubber Tool: This tool is based on a Python script that masks PII from syslog files created by the device. You can run this tool on any computer or server that has Python 3 installed. To download the tool from AudioCodes website, click here.
- Masking PII in CDRs and SDRs: You can mask PII in CDRs and SDRs that are displayed in the Web interface and CLI, and mask PII in CDRs that are sent to syslog, REST, RADIUS, Local Storage, or OVOC (depending on configuration). In addition, you can mask digits in syslog and DR.

  a. Open the SIP Definitions General Settings page (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **SIP Definitions General Settings**).

  b. Use the following parameters:

     - 'Mask PII in CDRs': Defines where the masking is done – CDRs/SDRs displayed in the Web interface and CLI only, or also in those that are sent to local storage and remote servers (e.g., syslog).
     - 'Mask PII in CDRs for OVOC': Enables masking in CDRs (QoE) sent to OVOC.
     - 'Mask URI Host Part in CDRs': Masks the host part of URIs (including IP addresses) in CDRs.
     - 'Number of Unmasked Characters in PII' and 'Location in PII of Unmasked Characters': Defines the number of characters to not mask, starting from the end or beginning of the PII element (e.g., phone number).
     - 'Mask Digits': Masks digits (typically, in-band DTMF) sent as events and detected by the device, including SIP messages (INFO and NOTIFY) in syslog and Debug Recording (message body) generated by the device.



Deleting Locally Stored CDRs and SDRs

If you have enabled local storage of CDRs or SDRs on the device, you can delete them from storage at any time through CLI:

- **CDRs:**

```
# clear storage-history cdr-storage-history
```

- **SDRs:**

```
# clear storage-history sdr-storage-history
```

Deleting Persistent Logs

The device automatically stores logged system event messages in its memory, where they persist even if the device undergoes a reset or powers off. To make sure that the device does not store these logged files indefinitely, allowing personal information to always be available, it's recommended that you configure an "age" period for the file rotation process (i.e., creation of new file and deletion of oldest file).

If you configure an "age" period, the device creates a new file when either the configured file size is reached ('Persistent Log Size' parameter) or the "age" is reached ('Persistent Log Period' parameter) -- whichever occurs first. Therefore, even if the configured file size for file rotation is not reached (even empty), when the period expires the device creates a new file, deleting the oldest persistent log file from storage.

➢ **To configure persistent log period:**

1. Open the Logging Settings page (**Troubleshoot** menu > **Troubleshoot** tab > **Logging** folder > **Logging Settings**).
2. In the 'Persistent Log Size' field, enter the log size for file rotation.

3. In the 'Persistent Log Period' field, enter the age period for file rotation.

Persistent logging is applicable only to Mediant 90xx and Mediant Software SBCs.

Encrypting the SIP Header Value

For enhanced security, you can configure the device to encrypt the value of a specific SIP header. Encryption is done using the AES-256 key algorithm. This feature is typically used between two AudioCodes devices, where one encrypts the SIP header value before sending the SIP message, while the other decrypts the value when it receives the SIP message.

This feature is intended for SIP headers that are **not** used by the device for classification or routing. For example, you may want to encrypt the value of a proprietary SIP header called "P-Access-Network-Info" that may contain sensitive information.

➢ **To configure SIP header value encryption:**

1. Open the SIP Definitions General Settings page (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **SIP Definitions General Settings**), and then in the 'AES-256 Encryption Key' parameter, configure the encryption key.

- The key must be 32 characters.
- Configure both devices with the same key.

2. Open the Message Manipulations table (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Manipulations**), and then configure a Message Manipulation rule to specify the SIP header to encrypt. Use the Funct.Encrypt and Funct.Decrypt keywords in the 'Action Value' field to encrypt and decrypt the header, respectively. For more information, refer to the device's *User's Manual*.
3. Open the IP Groups table, and then assign the Manipulation Set ID (configured in the previous step) to the relevant IP Group.

10 Password Obfuscation in Downloaded CLI and INI Files

You can enhance security by obfuscating passwords in downloaded ini and CLI Script files. Password encryption is achieved using the AES-256 algorithm with a 16-bit random CFB initialization vector (IV) cipher mode, using an encryption key. This method offers robust protection of sensitive data.

Obscured passwords are displayed in the following syntax:

- **ini File:**

  Syntax: *2<obfuscated password>*

  Example:

  ```
  WSTunPassword = $2$8EGYm+FG+JJT/p8ZOytU64uplPMKcw==
  ```

- **CLI Script File:**

  Syntax: *<obscured password>== encrypted*

  Example:

  ```
  password B55osyLT1t7+oorwkaNB3bxEX4Bl8g== encrypted
  ```

You can manually define the encryption key for password obfuscation or you can trigger the device to automatically generate a key. If you want to manually configure the key, it must be at least 32 characters long, and it can contain a combination of the following characters:

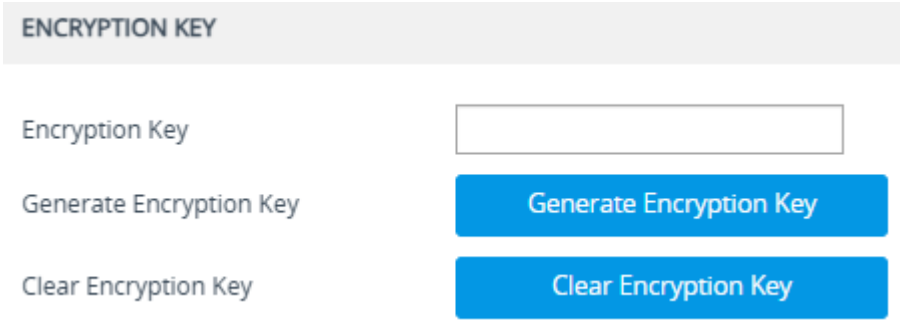
- Letters (A-Z and a-z)
- Numbers (0-9)
- Special characters: !, #, $, %, &, (,), *, +, ,, -, ., /, <, =, >, ?, @, [,], ^, _, `, {, }, ~. A-Z, a-z, 0-9, !, #, $, %, &, (, ), *, +, ,, -, ., /, <, =, >, ?, @, [, ], ^, _, `, {, }, ~

➢ **To configure encryption key for password obfuscation:**

- **Web interface:**

  a. Open the Security Settings page (**Setup** menu > **IP Network** tab > **Security** folder > **Security Settings**).

  b. Scroll down the page to the **Encryption Key** group:



c. Configure the encryption key, using one of the following methods:

- **Manually:** In the 'Encryption Key' field, enter your encryption key.
- **Automatically Generated by Device:** Click the **Generate Encryption Key** button; a message is displayed at the bottom of the page indicating that the key was successfully generated and copied to your clipboard. In addition, the key is partially displayed in the 'Encryption Key' field, showing the first four characters followed by three asterisks (*), for example, "F/sZ***". For future use, you can paste the key from your clipboard to a safe location.

CLI:

- **Manually:** `configure network > security-settings > encryption-key assign <your key>`
- **Automatically Generated by Device:** `configure network > security-settings > encryption-key generate`

11 Passwords Hidden in Management Interfaces

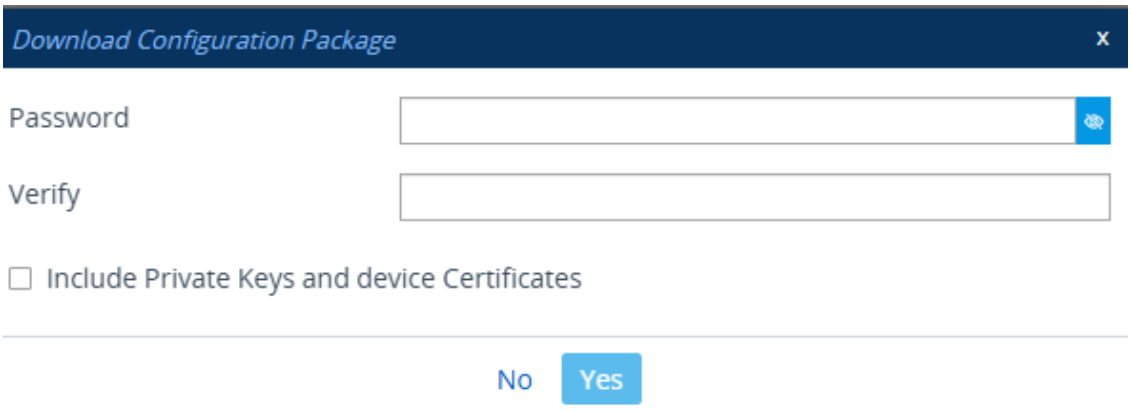
By default, the device hides all passwords in the different management interfaces, by replacing them with asterisks (*). The password is only displayed when you are typing it in the field and you click the show eye icon next to the field.

12 Password-Protect (Encrypt) Configuration Package File

If you want to download the device's Configuration Package file (which includes configuration, Auxiliary files and optionally, certificates), it's recommended that you password-protect the file. Password protection also encrypts the file using the AES-256 algorithm.

➢ **To password-protect Configuration Package file:**

1. Open the Configuration File page (**Setup** menu > **Administration** tab > **Maintenance** folder > **Configuration File**).
2. Select the 'Encrypted Configuration Package' check box.
3. Click the **Download Configuration Package** button; the following dialog box appears:



4. In the 'Password' and 'Verify' fields, type a password to protect the file.
5. To include TLS private keys and certificates, select the 'Include Private Keys and device Certificates' check box.
6. Click **Yes**.

13 SBC-Specific Security Guidelines

This section provides basic SBC security guidelines that should be implemented in your network deployment.

This section is applicable only to the Session Border Controller (SBC) application.

General Guidelines

It's crucial that you separate trusted from un-trusted networks:

- Separate un-trusted networks from trusted networks, by using different SRDs, IP Groups, SIP Interfaces, and SIP Media Realms (with limited port range).
- Similarly, separate un-trusted networks from one another. In particular, far-end users must be separated from the ITSP SIP trunk, using a different SRD, IP Group, SIP interface, and Media Realms. This separation helps in preventing attacks targeted on far-end user ports from affecting other users.
- For un-trusted networks, use strict classification rules over vague rules. For example, if the ITSP's proxy IP address, port and host name are known, then use them in the classification rules. This ensures that all other potentially malicious SIP traffic is rejected.
- Unclassified packets must be discarded (rejected).

Secure Media (RTP) Traffic using SRTP

It's recommended to use Secured RTP (SRTP) for encrypting the media (RTP and RTCP) path and thereby, protecting the voice traffic. The device supports SRTP according to RFC 3711. SRTP performs a Key Exchange mechanism (according to RFC 4568). This is done by adding a 'crypto' attribute to the SDP. This attribute is used (by both sides) to declare the supported cipher suites and to attach the encryption key. If negotiation of the encryption data is successful, the call is established. The device's SRTP feature supports various suites, for example, AES_CM_128_HMAC_SHA1_32.

➢ **To secure RTP traffic:**

- **Globally (all calls):** Open the Media Security page (**Setup** menu > **Signaling & Media** tab > **Media** folder > **Media Security**), and then select the 'SBC Media Security Mode' check box:

- **Per specific calls using IP Profiles:** SRTP is enforced on the SBC legs of an IP Profile (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **IP Profiles**). For each IP Profile associated with a leg, configure the 'SBC Media Security Mode' parameter to **SRTP**. This

enforces the SBC legs to negotiate only SRTP media lines; RTP media lines are removed from the incoming SDP offer \ answer.

Enable Rate-Limit of ICMP Echo Requests

By default, the device limits incoming ICMP echo requests to 100 packets per second, protecting it from possible ping flooding (i.e., DDoS attacks). It's recommended to leave this feature at its default (enabled).

➢ **To enable or disable rate-limiting of ICMP echo requests:**

1. Open the Network Settings page (**Setup** menu > **IP Network** tab > **Advanced** folder > **Network Settings**).
2. Select the 'Rate-Limit ICMP Echo Requests' check box:

3. Click **Apply**.

This feature is applicable only to Mediant 90xx and Mediant Software.

Blocking ICMP Timestamp Requests

The ICMP protocol allows for network timing measurements by sending an ICMP timestamp request to a remote peer and receiving an ICMP timestamp in reply. However, sending ICMP timestamp replies may expose peers to security vulnerabilities.

By default, the device accepts ICMP timestamps and replies accordingly. You can block incoming ICMP timestamp requests and thereby prevent the device from replying to them, by uploading an ini file to the device with the ini file parameter [BlockIcmpTimeStamp] set to 1.

This feature is applicable only to Mediant 90xx and Mediant Software.

Implement SIP Authentication and Encryption

It's paramount that your network implements authentication and encryption to secure the network and ensure integrity and confidentiality of sensitive communications over un-trusted networks. Some of the main authentication and encryption guidelines are discussed in the subsequent sections.

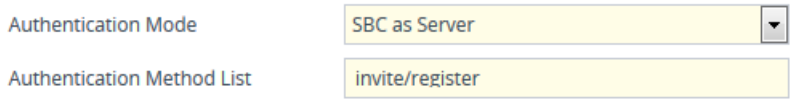
Authenticating Users as an Authentication Server

Instead of relying on external, third-party authentication servers, the device can be configured to act as an Authentication server, performing authentication and validation challenges with SIP UAs. The SIP method (INVITE or REGISTER) on which it challenges can be defined. If the message is received without an Authorization header, the device challenges the client by sending a 401 or 407 SIP response. The client then resends the request with an Authorization header containing its username and password. The device validates the SIP message and if it fails, the message is rejected and the device sends a 403 "Forbidden" response. If the SIP message is validated, the device verifies identification of the UA by checking whether the username and password received from the user is correct. The usernames and passwords are obtained from the User Information table. If after three attempts the UA is not successfully authenticated, the device sends a 403 "Forbidden" response. The device can also perform authentication on behalf of its UAs with an external third-party server.

The cryptographic hash algorithm used when the device sends the authentication challenge in the SIP 401 or 407 response can be configured, using the [SIPServerDigestAlgorithm] parameter.

➢ **To authenticate users:**

1. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Identities** folder > **IP Groups**).
2. For the IP Group (**User**-type) of the UAs, configure the following:
   - From the 'Authentication Mode' drop-down list, select **SBC as Server**.
   - In the 'Authentication Method List' field, enter the SIP message(s) to authenticate (e.g., "INVITE\REGISTER").



3. Configure the authentication usernames and passwords of the users:
   a. Open the Proxy & Registration page (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Proxy & Registration**), and then select the 'User-Information Usage' check box to enable the SBC User Information table and its feature:

> The 'User-Information Usage' parameter is available only if your device's License Key includes a license for far-end users ("FEU").

   b. Click **Apply**, and then restart the device with a burn-to-flash for your settings to take effect.

c. Open the SBC User Information table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **SBC User Information**), and then add users with authentication usernames and passwords:

INDEX	LOCAL USER	USERNAME	PASSWORD	IP GROUP	STATUS
0	John Dee	johnd	*	ITSP	Not Registered

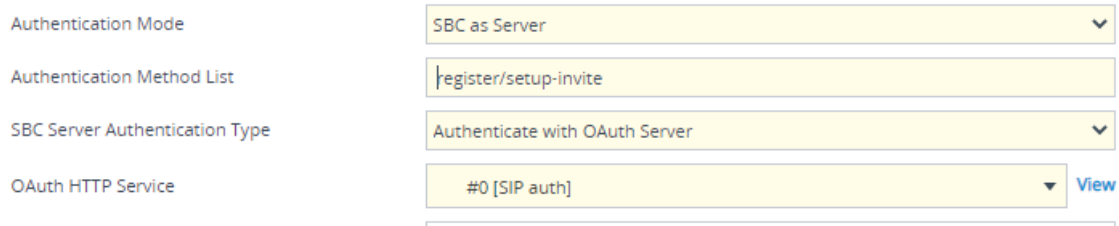
OAuth 2.0 Token-based SIP Authentication

The device can authenticate any incoming SIP requests (e.g., REGISTER and INVITE) from client applications, based on access tokens with an OAuth 2.0 Authorization Server (internal or external).

When the device receives a SIP request (with an OAuth access token) from a client application (e.g., WebRTC client), the device introspects the token with the OAuth Authorization server (HTTP server). Upon successful introspection, the device allows the client access to the device's resources (e.g., registration and calls) and continues to handle and process the SIP request as usual.

➢ To configure OAuth-based SIP authentication:

1. Open the Remote Web Services table (**Setup** menu > **IP Network** tab > **Web Services** folder > **Remote Web Services**), add then configure a Remote Web Service to represent the OAuth Authentication server.
2. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Identities** folder > **IP Groups**), and then configure the following parameters:
   - 'Authentication Mode': **SBC as Server**
   - 'Authentication Method List': "register/setup-invite"
   - 'SBC Server Authentication Type': **Authenticate with OAuth Server**
   - 'OAuth HTTP Service': Assign the Remote Web Service that you configured in Step 1



Authenticating Users by RADIUS Server

Instead of authenticating calls locally by the device, digest authentication of SIP users can be done by a RADIUS server (according to RFC 5090). In this way, the device offloads the MD5 calculation (validation) to a RADIUS server, where the device is classed as a RADIUS client.

➢ **To authenticate users by RADIUS server:**

1. Open the RADIUS Servers table (**Setup** menu > **IP Network** tab > **AAA Servers** folder > **RADIUS Servers**), and then configure the RADIUS sever (IP address, port and shared secret password):

INDEX	IP ADDRESS	AUTHENTICATION PORT	ACCOUNTING PORT	SHARED SECRET
0	202.100.0.2	1645	1645	*

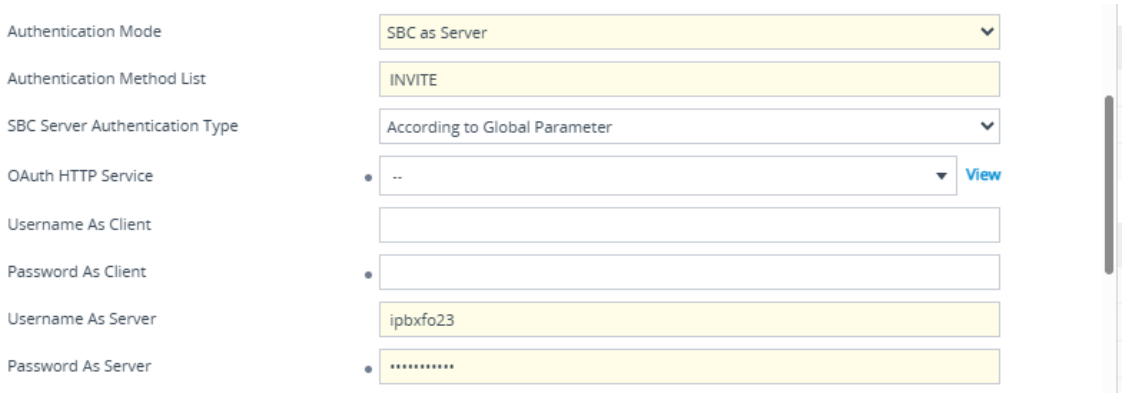
2. Configure the [SBCServerAuthMode] parameter to 1 to enable authentication by an RFC 5090 compliant RADIUS server.

Authenticating SIP Servers as an Authentication Server

It's recommended to enable the device (acting as an authentication server) to authenticate remote SIP servers (e.g., SIP proxy servers). This provides protection from rogue SIP servers, preventing unauthorized usage of the device's resources and functionality. The device authenticates remote servers by challenging them with a username and password that is shared with the remote server. From such a challenge, the device can check if the server's identity is genuine. The type of SIP message (e.g., INVITE) to authenticate can also be specified.

➢ **To configure SIP server authentication:**

1. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Identities** folder > **IP Groups**).
2. For the IP Group of the SIP server, configure the following:
   - From the 'Authentication Mode' drop-down list, select **SBC as Server**.
   - In the 'Authentication Method List' field, enter the SIP message(s) to authenticate.
   - In the 'Username As Server' field, enter the authentication username.
   - In the 'Password As Server' field, enter the authentication password.



Enforce SIP Client Authentication by SIP Proxy

When the device is located between a SIP client and a third-party SIP proxy server and SIP Digest Authentication is used, the device relays authentication messages between these

entities. Although the device gathers and maintains some information in its registration database (Address of Record / AOR) it does not actively participate in the authentication process. Instead, it's the SIP proxy that handles and enforces SIP client authentication. Therefore, it's imperative that your SIP proxy server be configured to enforce SIP client authentication.

Enforce SIP Digest Authentication by IP PBX

If TLS cannot be configured (for whatever reason) and if you are using an on-premises IP PBX, it's crucial that your IP PBX implements SIP Digest Authentication for remote users. In addition, authentication should be applied to as many SIP methods as possible (i.e., not only on REGISTER messages, but also INVITEs, re-INVITEs, etc.).

Secure Routing Rules

This section provides recommended security guidelines regarding routing rules.

Classify by Classification Rules versus Proxy Set

An important security functionality of the device is to make sure that incoming SIP dialog-initiating requests (e.g., INVITE messages) from malicious attackers are not mistakenly identified as belonging to a configured Server-type IP Group entity.

The device provides two optional mechanisms that can be employed to identify incoming dialogs as coming from a specific Server-type IP Group:

- **Classification by Classification rules (Classification table):** Identifies incoming dialogs based on the characteristics of the SIP message such as host part in the INVITE message (Layer 4-7) and source IP address (Layer 3).

  Recommended usage:

  If the IP address of the IP Group entity is known, it's recommended to employ classification based on a Classification rule, where the rule is configured with not only the IP address, but also with SIP message characteristics to increase strictness of the classification process.

  When Classification rules are used and classify by Proxy Set is disabled (see below), it's recommended to enable the 'Validate Source IP' parameter in the IP Groups table. This setting verifies that the incoming dialog was sent from one of the IP addresses (including DNS-resolved IP addresses) of the Proxy Set associated with the classified IP Group (see Validate Source IP Address of Incoming SIP Dialog Requests on page 48). IP address validation is also typically needed when multiple IP Groups are assigned to the same Proxy Set and therefore, Classification rules are necessary to produce the desired mapping (classification) of the incoming SIP dialogs to the different IP Groups.

The device uses the Classification table for classification only if the following classification stages fail (listed chronologically):

1. The incoming SIP dialog is not from a SIP UA that is registered with the device (i.e., not in user registration database).
2. The Classify by Proxy Set feature is disabled for the IP Group (i.e., source IP address of incoming dialog is matched with a Proxy Set associated with the IP Group but Classify by Proxy Set is disabled for the IP Group).

- **Classification by Proxy Set:** Identifies incoming dialogs based on source IP address (Layer 3) only. The Proxy Set defines the address of the IP Group. For this method, the device searches for a Proxy Set that has the same source IP address as the incoming dialog, and then classifies it to the IP Group that is assigned to this Proxy Set. Classification by Proxy Set is enabled in the IP Groups table, using the 'Classify By Proxy Set' parameter:

Recommended usage:

If the IP address is unknown, in other words, the Proxy Set associated with the IP Group is configured with an FQDN, it's recommended to employ SIP dialog classification based on Proxy Set. This allows the SBC to classify the incoming dialog based on the DNS-resolved IP address. The reason for classifying by Proxy Set 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.

Define Strict Classification Rules

Classification rules are used to identify incoming SIP dialog-initiating requests (e.g., INVITE messages) and bond them to IP Groups. In other words, these rules identify the source of the call. Once the source IP Group is identified, the traffic can then be routed to its destination according to IP-to-IP routing rules.

When defining Classification rules, adhere to the following recommendations:

- For Server-type IP Groups, use Classification rules only if the IP address of the IP Group is known. If known, include the IP address in the Classification rule ('Source IP Address' parameter). In addition, to increase classification strictness, configure SIP message characteristics in the rule as well.

If the IP address is unknown (i.e., the Proxy Set associated with the IP Group is configured with an FQDN), it's recommended to employ SIP dialog classification based on Proxy Set (see Classify by Classification Rules versus Proxy Set on the previous page).

- It's recommended to enable the 'Validate Source IP' parameter in the IP Groups table. This setting verifies that the incoming dialog was sent from one of the IP addresses (including DNS-resolved IP addresses) of the Proxy Set associated with the classified IP Group (see Validate Source IP Address of Incoming SIP Dialog Requests on page 48). IP address

validation is also typically needed when multiple IP Groups are assigned to the same Proxy Set and therefore, Classification rules are necessary to produce the desired mapping (classification) of the incoming SIP dialogs to the different IP Groups.

- For Server-type IP Groups whose IP addresses are known, it's recommended to also configure VoIP firewall rules (see Block Unused Network Ports on page 7).
- Use strict Classification rules over vague ones so that all other potentially malicious SIP traffic is rejected. In other words, configure the rule with as much information as possible that accurately characterizes the incoming SIP dialog (e.g., source and destination host name).
- Define a range for the source and destination prefix numbers.
- Define a combination of Classification rules to guarantee correct and accurate identity of sender of call.
- Make sure that you configure the device to block unclassified calls, as described in Section Validate Source IP Address of Incoming SIP Dialog Requests on the next page.
- Use Message Condition rules to increase the strictness of the Classification process. Message Condition rules enhance the process of classifying incoming SIP dialogs to an IP Group. When a Classification rule is associated with a Message Condition rule, the Classification rule is used only if its' associated Message Condition rule are matched. Message Condition rules are SIP message conditions based on the same syntax used in the Message Manipulations table. You can define complex rules using the "AND" or "OR" Boolean operands. You can also use regular expressions (regex) as Message Condition rules, 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

  To implement message conditions:

  a. Configure a Message Condition rule in the Message Conditions table (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Conditions**). The following shows a Message Condition rule example for P-Asserted-Identity headers that contain "abc":

INDEX	NAME	CONDITION
0	P-Asserted-Identity header with "sbc"	header.p-asserted-identity.url.user contains 'abc'

  b. Assign the Message Condition rule to the Classification rule in the Classification table, using the 'Message Condition' parameter:

Classification rules are configured in the Classification table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Classification**). The following figure shows an example of two Classification rules:

INDEX	NAME	SRD	SOURCE SIP INTERFACE	SOURCE USERNAME PREFIX	SOURCE HOST	DESTINATION USERNAME PREFIX	DESTINATION HOST	ACTION TYPE	SOURCE IP GROUP
0	ITSP	DefaultSRD	Any	[2-4]	domain.com	[1-7]	*	Allow	ITSP
1	Deny	DefaultSRD	Any	*	*	*	*	Deny	--

- Index 0 "ITSP": Classifies received calls to Server-type IP Group "ITSP" if they have the following incoming matching characteristics:
  - 'Source IP Address': 10.15.7.96
  - 'Source Username Prefix': 2 through 4
  - 'Source Host': domain.com
  - 'Destination Username Prefix': 1 through 7
  - 'Message Condition': SIP message with P-Asserted-Identity header containing "abc" (Message Condition rule described previously in this section)
- Index 2 "Deny": Denies calls that cannot be classified (unknown calls).

Validate Source IP Address of Incoming SIP Dialog Requests

When classification is according to Classification rules and you need to classify SIP dialogs originating from the same Proxy Set into multiple IP Groups, and where Classification rules are necessary to produce the desired mapping (classification) to the different IP Groups, it's recommended that you configure the device to validate the source IP address of incoming SIP dialog-initiating requests (e.g., INVITE).

The device checks that it matches an IP address (or DNS-resolved IP address) of the Proxy Set that is associated with the IP Group to which it's classified.

➢ **To validate source IP addresses:**

1. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
2. From the 'Validate Source IP' drop-down list, select **Enable**:

Validation is done for the IP address only (not port, transport, or SIP Interface).

Block Unclassified Calls

It's recommended that you block incoming calls that can't be classified to an IP Group, based on the rules in the Classification table (discussed in the previous section). If unclassified calls are

not blocked, they are sent to the default SRD / IP Group and therefore, illegitimate calls can be established.

➢ **To block unclassified calls:**

1. Open the SBC General Settings page (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **SBC General Settings**).
2. From the 'Unclassified Calls' drop-down list, select **Reject**:

Allow Calls Only with Specific SIP User-Agent Header Value

The SIP User-Agent header contains information about the User Agent Client (UAC) initiating the SIP dialog request. This information is unique to the Enterprise and therefore, it's recommended to configure the device so that it accepts only calls with a specific User-Agent header value.

➢ **To configure security based on SIP User-Agent header:**

1. Open the Message Conditions table (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Conditions**), and then add a rule that specifies a value for the User-Agent header.

   The following figure shows a rule where the SIP User-Agent header value is "abc.com":

INDEX	NAME	CONDITION
0	Only sbc.com calls	header.user-agent='abc.com'

2. Open the Classification table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Classification**), and then assign the Message Condition rule to the relevant Classification rule.

Define Strict Routing Rules

It's crucial that you adhere to the following guidelines when configuring IP-to-IP Routing rules:

- Make sure that your routing rules are accurate and correctly defined. Inaccurate or weak routing rules can easily result in Service Theft.
- Make sure that your routing rules from source IP Group to destination IP Group are accurately defined for the desired call routing outcome.
- If possible, avoid using the asterisk (*) symbol to indicate "any" for a specific parameter in your routing rule. This constitutes weak routing rules that can be vulnerable to attackers. For strong routing rules, enter specific alphanumerical values instead of the asterisk.

Define Call Admission Control Rules

It's recommended to configure Call Admission Control (CAC) rules for regulating VoIP traffic volume. CAC rules can assist in limiting the rate of call requests, preventing excessive signaling requests originating from malicious and legitimate sources from overwhelming your network resources.

CAC rules can limit the number of concurrent calls (SIP dialogs) per IP Group, SIP Interface or SRD. The call limitation can be defined per SIP-dialog initiating request type (e.g., INVITE or REGISTER messages), request direction (inbound, outbound, or both), and user. Requests that exceed the user-defined limits are rejected (with SIP 480 "Temporarily Unavailable" responses). You can also limit the incoming packet rate based on the "token bucket" mechanism.

Adhere to the following CAC recommendations:

- It's crucial that your CAC rules include call limitations per user. This ensures that a user doesn't make unlimited, simultaneous calls.
- Define rules as specific as possible. For example, instead of defining one rule for all SIP request types, create rules per request type.

> **Note:** If call routing to a specific IP Group is blocked due to a CAC rule, the device searches for an alternative route (if configured) in the SBC IP-to-IP Routing table. If this alternative route doesn't exceed the CAC rule limitation, the device uses it to route the call.

➢ To configure CAC rules:

1. Open the Call Admission Control Profile table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Call Admission Control Profile**).
2. Click **New** to add a new CAC Profile name.
3. Select the new CAC Profile row, and then click the **Call Admission Control Rule** link, located below the table.
4. Click **New** to add a CAC rule.

   The following displays an example of a CAC rule that defines a maximum of 100 concurrent SIP INVITE requests. SIP requests received above this threshold are rejected:

INDEX	REQUEST TYPE	REQUEST DIRECTION	LIMIT	LIMIT PER USER
0	INVITE	Both	100	-1

Define Maximum Call Duration

It's recommended to configure the maximum call duration (in minutes) to prevent SBC calls from utilizing valuable device resources that could otherwise be used for additional new calls. If a call exceeds this duration, the device terminates the call.

➢ **To configure maximum SBC call duration:**

1. Open the Gateway Advanced Settings page (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **SBC General Settings**).
2. In the 'Max Call Duration' field, enter the maximum call duration:

Secure SIP User Agent Registration

Service theft can result from a lack of security in the SIP user registration process. This section provides recommended guidelines regarding user registration.

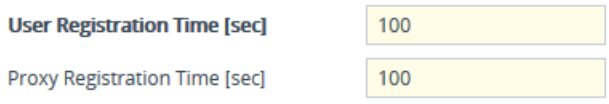
Configure Identical Registration Intervals

Scenarios in which the device doesn't forward user registrations to a server (e.g., a PBX) and the device receives a new SIP REGISTER request from the same number (i.e., same AOR) but without an Authentication header, the device still sends a SIP 200 OK response to the user. This is because the AOR already exists in the device's registration database. Therefore, if an illegitimate user attempts to connect with a legitimate IP address and phone number (without authentication), the malicious user can connect and steal calls.

To overcome this issue and prevent stealing of calls, make sure that you configure the user and proxy registration times with identical values.

➢ **To configure identical registration intervals for user and proxy:**

1. Open the SBC General Settings page (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Proxy & Registration**).
2. In the 'User Registration Time' field, configure the duration of the periodic registrations between the user and the device.
3. In the 'Proxy Registration Time' field, configure the time interval (in seconds) that the device must register to the server (e.g., PBX).



Limit SBC Registered Users per IP Group, SIP Interface or SRD

It's recommended that you define a maximum number of allowed registered users per IP Group (User-type IP Group), SIP Interface, or SRD. This ensures that illegitimate users are blocked from registering with the IP Group.

➢ **To limit SBC registered users:**

1. Open any of the following tables, depending on your specific configuration requirements:

- IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Identities** folder > **IP Groups**)
- SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Identities** folder > **SIP Interfaces**)
- SRDs table (**Setup** menu > **Signaling & Media** tab > **Core Identities** folder > **SRDs**)

2. In the 'Max. Number of Registered Users' field, enter the maximum number of registered users:

Block Calls from Unregistered Users

Make sure that calls from unregistered users are blocked (rejected) and that calls from only registered users are allowed.

➢ **To block calls from unregistered users:**

1. Open any of the following tables, depending on your specific configuration requirements:
   - SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Identities** folder > **SIP Interfaces**)
   - SRDs table (**Setup** menu > **Signaling & Media** tab > **Core Identities** folder > **SRDs**)
2. From the 'User Security Mode' drop-down list, select **Accept Registered Users**:

Block Registration from Un-Authenticated New Users

Typically, when a SIP proxy (registrar) server is available, the device forwards SIP REGISTER requests from new users 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. However, if the proxy becomes unavailable at any time (e.g., due to network connectivity loss), the REGISTER requests can't be authenticated. For these scenarios, make sure that the device is configured to reject such unauthenticated request messages from new users.

➢ **To block registration of un-authenticated users:**

1. Open any of the following tables, depending on your specific configuration requirements:
   - SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Identities** folder > **SIP Interfaces**)
   - SRDs table (**Setup** menu > **Signaling & Media** tab > **Core Identities** folder > **SRDs**)
2. From the 'Enable Un-Authenticated Registrations' drop-down list, select **Disable**:

The device accepts registration refreshes from users already in its database.

Authenticate SIP BYE Messages

It's recommended to enable the device to authenticate incoming SIP BYE requests before it releases the call. This prevents, for example, a scenario in which the device receives a BYE request from a third-party imposter assuming the identity of a participant in the call and therefore, the call is inappropriately disconnected.

When the device is configured to authenticate BYE messages, it sends a SIP authentication response to the sender of the BYE request and waits for the sender (user) to authenticate it. The call is disconnected only if the authenticating server responds with a 200 OK.

➢ **To authenticate SIP BYE messages:**

1. Open the Proxy & Registration page (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Proxy & Registration**).
2. From the 'BYE Authentication' drop-down list, select **Enable**:

Use SIP Message Manipulation for Topology Hiding

The device intrinsically employs topology hiding, limiting the amount of topology information displayed to external parties (i.e., un-trusted networks). This anonymous information minimizes the chances of directed attacks on your network.

The device employs topology hiding 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
- Generates a new SIP Call-ID header value (different between legs)
- Changes the SIP Contact header to the device's address
- Performs Layer-3 topology hiding by modifying the source IP address in the SIP IP header (for example, IP addresses of ITSPs equipment such as proxies, gateways, and application servers can be hidden from outside parties)

In addition, to enhance topology hiding, you can modify the SIP To header, From header, and/or Request-URI host name. This can be done using the Message Manipulation table or the IP Group (for SIP URI host part manipulations). The Message Manipulation table also supports Regular Expressions (Regex).

Define Malicious Signatures

To protect the device from malicious attacks on SBC calls, it's recommended to employ the device's Malicious Signature feature, which defines malicious signature patterns. The Malicious Signature feature identifies and protects against SIP (Layer 5) threats by examining new inbound SIP dialog messages. Once the device identifies an attack based on the configured malicious signature patterns, it marks the SIP message as invalid and discards it (or alternatively, rejects it with a SIP response). Malicious signatures are typically based on the SIP User-Agent header, which attackers often use as their identification string (e.g., "User-Agent: VaxSIPUserAgent").

➢ **To configure malicious signatures:**

1. Open the Malicious Signature table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Malicious Signature**), and then configure malicious signatures. The device provides preconfigured malicious signatures.

INDEX	NAME	PATTERN
0	SIPVicious	Header.User-Agent.content prefix 'friendly-scanner'
1	SIPScan	Header.User-Agent.content prefix 'sip-scan'
2	Smap	Header.User-Agent.content prefix 'smap'
3	Sipsak	Header.User-Agent.content prefix 'sipsak'

2. Open the Message Policies table (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Policies**), and then configure a Message Policy and enable it to use the malicious signatures, by configuring 'Malicious Signature Database' to **Enable**:

3. Open the SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **SIP Interfaces**), and then for the required SIP Interface, from the 'Message Policy' drop-down list, select the Message Policy that you configured in Step 1:

Secure Media Cluster Management Interface

By default, connectivity between the Signaling Component and Media Components for management of the Media Components is secured using TLS. Configuration is done using the [TpncpEncryptionEnable] parameter.

This section is applicable only to Mediant CE.

14 Gateway-Specific Security Guidelines

This section describes recommended security guidelines for the Gateway application (IP-to-Tel and Tel-to-IP call routing).

Block Calls from Unknown IP Addresses

Make sure that the device accepts incoming calls only from source IP addresses that appear in the Proxy Sets or Tel-to-IP Routing tables. In addition, if an FQDN appears in these tables, the call is accepted only if the resolved DNS IP address of the call appears in any one of these tables. The device rejects calls whose source IP addresses don't appear in these tables. This is useful in preventing unwanted SIP calls, SIP messages, or VoIP spam.

➢ **To block calls from unknown IP addresses:**

1. Open the Gateway Advanced Settings page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Gateway Advanced Settings**).
2. From the 'IP Security' drop-down list, select **Secure All calls**:

Enable Secure SIP (SIPS)

Make sure that you enable Secure SIP (SIPS) so that the device initiates TLS all the way to the destination (i.e., over multiple hops). SIPS runs SIP-over-TLS on a hop-by-hop basis. This is important as using TLS as a transport by itself guarantees only encryption over a single hop. Since it's very common for a SIP call to traverse multiple proxy servers from one end to the other, there is a need to guarantee end-to-end security for SIP traffic. A call to a SIPS URI is guaranteed to be encrypted from end to end. All SIP traffic within this call is secured using TLS from the sender to the domain of the final recipient.

➢ **To enable SIPS:**

1. Open the Transport Settings page (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Transport Settings**).
2. Select the 'SIPS' check box:

It's recommended to use the 'SIPS' parameter and not the 'SIP Transport Type' parameter to define TLS. The 'SIP Transport Type' parameter provides only a TLS connection to the next network hop whereas the 'SIPS' parameter provides TLS to the final destination (over multiple hops).

3. Configure the local SIP TLS port for the SIP Interface in the SIP Interfaces table.

Define Strict Routing Rules

When defining IP-to-Tel (IP-to-Trunk Group Routing table) and Tel-to-IP (Tel-to-IP Routing table) routing rules, it's crucial that you adhere to the following security guidelines:

- Make sure that your routing rules are accurate and correctly defined for the desired routing outcome. Inaccurate or “loose” routing rules can easily result in service theft.
- Avoid, if possible, using the asterisk "*" symbol and Any option to indicate any for a specific parameter in your routing rules. This constitutes weak routing rules that can be vulnerable to attackers. For strong routing rules, enter specific alphanumerical values instead of the asterisk.

Define Call Admission Control

Make sure that you configure the maximum number of concurrent calls per routing rule or IP Group.

➢ **To configure maximum concurrent calls:**

1. Open the IP Profiles table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **IP Profiles**).
2. For the relevant IP Profile, in the 'Number of Calls Limit' field, enter the maximum number of concurrent calls:

3. Assign the IP Profile in the IP-to-Tel Routing table, Tel-to-IP Routing table, or IP Groups table.

The maximum number of concurrent calls considers incoming and outgoing calls (i.e., summation of all calls).

Define Maximum Call Duration

It's recommended to configure maximum call duration (in minutes) to prevent Gateway calls from utilizing valuable device resources that could otherwise be used for additional new calls. If a call exceeds this duration, the device terminates the call.

➢ **To configure maximum Gateway call duration:**

1. Open the Gateway Advanced Settings page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Gateway Advanced Settings**).
2. In the 'Max Call Duration' field, enter the maximum call duration:

Block Device Restarts after Emergency Calls

By default, the device blocks restarts that are triggered through the CLI (`reload` command) during emergency calls (IP-to-Tel or Tel-to-IP). You can block restarts for a user-defined period after the emergency call ends (whether successfully established or failed). This may be important to allow emergency services to contact the caller, which would not be possible during a device restart.

To configure the duration for blocking device restarts after an emergency call ends, use the following parameter:

- **CLI:** `configure voip > sip-definition settings > reload-timeout-for-emergency-call`
- **ini file:**[ReloadTimeoutForEmergencyCall]

This section is applicable only to device supporting analog or digital Gateway interfaces.

15 Network Port Assignment

The following table lists the device's network port assignments. This table also shows whether these ports are enabled or disabled by default and how to configure them.

For enhanced security, it's highly recommended to:

- Disable any default-enabled ports that are not required for your deployment.
- Change the default port numbers for required services (especially for the SIP application).

Table 15-1: Network Port Assignments

Interface Type	Application	Protocol	Port	Default	Port Configuration
OAMP	SSH server	TCP	22	Enabled	■ Enabling: 'Enable SSH Server' ■ Port Number: SSH Interfaces table ('Port') ■ Access Control: ✓ Firewall table (Layer 3/4) ✓ Management Access List table
	Telnet server	TCP	23	Disabled	■ Enabling: 'Enable Telnet Server' (**Enable Secured** or **Enable Unsecured**) ■ Port Number: Telnet Interfaces table ('Port') ■ Access Control: ✓ Firewall table (Layer 3/4) ✓ Management Access List

Interface Type	Application	Protocol	Port	Default	Port Configuration
					table

Interface Type	Application	Protocol	Port	Default	Port Configuration
	Web server (HTTP)	TCP	80	Enabled	■ Enabling: Web Interfaces table ('HTTPS Only' - (**HTTP and HTTPS**) ■ Port Number: Web Interfaces table ('HTTP Port') ■ Access Control: ✓ Firewall table (Layer 3/4) ✓ Management Access List table
	Web server (HTTPS)	TCP	443	Enabled	■ Enabling: Web Interfaces table ('HTTPS Only' - (**HTTPS Only** or **HTTP and HTTPS**) ■ Port Number: Web Interfaces table ('HTTPS Port') ■ Access Control: ✓ Firewall table (Layer 3/4) ✓ Management Access List table
	SNMP server (GET / SET) and client (trap sender)	UDP	161	Enabled (Disabled for Mediant 90xx and Mediant	■ Enabling: 'Disable SNMP' (**No**) ■ Port Number: ✓ SNMP GET / SET: 'SNMP

Interface Type	Application	Protocol	Port	Default	Port Configuration
				Software)	Port' ✓ SNMP traps: SNMP Trap Destinations table ('Port') ■ Access Control: ✓ Firewall table (Layer 3/4) ✓ SNMP Trusted Managers table
Any	DHCP server	UDP	67	Disabled	■ Enabling: DHCP Servers table ■ Local interface: DHCP Servers table ('Interface Name') ■ Port Number: Not configurable ■ Access Control: Firewall table (Layer 3/4)
Control	SIP traffic	UDP / TCP	5060	Enabled	■ Enabling: SIP Interfaces table – 'UDP Port' or 'TCP Port' ■ Port Number: SIP Interfaces table – 'UDP Port' or 'TCP Port' ■ Access Control: Firewall table (Layer 3/4)
	SIPS traffic	TCP	5061	Enabled	■ Enabling: SIP

Interface Type	Application	Protocol	Port	Default	Port Configuration
					Interfaces table ('TLS Port') ■ Port Number: SIP Interfaces table ('TLS Port') ■ Access Control: Firewall table (Layer 3/4)
	SIP over WebSocket (e.g., WebRTC)	TCP	443	Enabled	■ Enabling: SIP Interfaces table ('TLS Port') ■ Port Number: SIP Interfaces table ('TLS Port' and 'Encapsulating Protocol' - **WebSocket**) ■ Access Control: Firewall table (Layer 3/4)
Media	Media traffic (RTP, RTCP, T.38)	UDP	6000-65535	Enabled	■ Enabling: Enabled during SIP session establishment ■ Port Number: Media Realms table ('UDP Port Range Start' and 'Number Of Media Session Legs') ■ Access Control: n/a
Maintenance (HA)	HA status	UDP	669	Disabled	■ Enabling: 'HA Remote Address' ■ Port Number: n/a **Note:** Applicable to

Interface Type	Application	Protocol	Port	Default	Port Configuration
					the following: ■ Standalone SBC ■ Signaling Component in Mediant CE ■ Signaling Component in Media Transcoding Cluster
	HA keep-alive	UDP	680	Disabled	■ Enabling: 'HA Remote Address' ■ Port Number: n/a **Note:** Applicable to the following: ■ Standalone SBC ■ Signaling Component in in Mediant CE ■ Signaling Component in Media Transcoding Cluster
	HA file sync	TCP	80	Disabled	■ Enabling: 'HA Remote Address' ■ Port Number: n/a **Note:** Applicable to the following: ■ Standalone SBC ■ Signaling Component in in Mediant CE ■ Signaling Component in

Interface Type	Application	Protocol	Port	Default	Port Configuration
					Media Transcoding Cluster
	HA data sync	TCP	2442	Disabled	■ Enabling: 'HA Remote Address' ■ Port Number: n/a **Note:** Applicable to the following: ■ Standalone SBC ■ Signaling Component in in Mediant CE ■ Signaling Component in Media Transcoding Cluster
Cluster	Cluster control	TCP	2424	Disabled	■ Enabling: 'Cluster Mode' ■ Port Number: n/a
	Cluster keep-alive	UDP	3900	Disabled	■ Enabling: 'Cluster Mode' ■ Port Number: n/a **Note:** Applicable to the following: ■ Signaling Component in in Mediant CE ■ Media Component in in Mediant CE ■ Signaling Component in Media Transcoding

Interface Type	Application	Protocol	Port	Default	Port Configuration
					Cluster ■ Media Component in Media Transcoding Cluster

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International Headquarters

6 Ofra Haza Street

Naimi Park

Or Yehuda, 6032303, Israel

Tel: +972-3-976-4000

Fax: +972-3-976-4040

AudioCodes Inc.

80 Kingsbridge Rd

Piscataway, NJ 08854, USA

Tel: +1-732-469-0880

Fax: +1-732-469-2298

Contact us: https://www.audiocodes.com/corporate/offices-worldwide

Website: https://www.audiocodes.com/

Documentation Feedback: https://online.audiocodes.com/documentation-feedback

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