Configuration Note

AudioCodes Mediant™ Family of Media Gateways & Session Border Controllers

Connecting AudioCodes' SBC with Analog Device to Microsoft Teams Direct Routing Enterprise Model

Enterprise Model

Microsoft Teams

audiocodes

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Abbreviations and Terminology

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Related Documentation

Document Name
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 2600 SBC User's Manual
Mediant 4000 SBC User's Manual
Mediant 9000 SBC User's Manual
Mediant Software SBC User's Manual
MP-11x and MP-124 SIP User's Manual
MP-20x Telephone Adapter User's Manual
SIP Message Manipulation Reference Guide

Document Revision Record

LTRT	Description
33421	Initial document release for Version 7.2. Teams Enterprise Model.
33422	Modified Section: Deploy Baltimore Trusted Root Certificate (added note for Baltimore Trusted Root Certificate and MTLS implementation).; Configure SIP Signaling Interfaces; Configure IP Groups
33423	Note removed regarding external firewall.
33424	Licenses consolidated into one section.
33425	Update to topology figures and correction for parameter “Remote Update Support” to “SIP UPDATE Support”.
33426	Update to the “Related Documentation” table to include the Mediant 1000B Gateway & E-SBC product.
33427	Updates related to the usage of LAD licenses.
33428	Typo fixes, a note regarding external firewall was removed. An additional two IP addresses were added to firewall per Microsoft request.
33429	Update for Message Manipulation rule towards Microsoft Teams.
33430	“SipSignallingPort” replaced by “SipSignalingPort”.
33431	Update to the Firewall Table Rules table with two additional IP addresses of the new infrastructure in Japan.
33432	Update to SIP Trunk IP Profile and validated firmware version.
33433	Added section for overcoming problem of not playing music on hold during conversational transfer.
33434	Remote Replaces Mode parameter with value “Handle Locally” was added to the Teams IP Profile due to new Microsoft requirements. The Classification rule was updated. Update to the Firewall Table Rules table due to new Microsoft requirements.
33435	TLS Root Certificate Authority updated by Microsoft.
33436	Updated Classification Table with stricter rules to only allow for documented Microsoft SIP Proxies.
33437	Note added detailing deployment in Office 365 GCC DoD and GCC High environments.
33438	TLS Private Key size of 1024 was removed. Microsoft subnets were updated in the Classification and Firewall tables.
33439	Teams IP Profile updated with RFC 2833 Mode parameter.

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1 Introduction

This Configuration Note describes an example setup of the AudioCodes Enterprise Session Border Controller (hereafter, referred to as *SBC*) for interworking between Company's SIP Trunk, ATA device and Microsoft's Teams Direct Routing environment.

For configuring the Office 365 side, please refer to https://docs.microsoft.com/en-us/microsoftteams/direct-routing-configure

This document is intended for IT or telephony professionals.

1.1 About Microsoft Teams Direct Routing

Teams Direct Routing allows connecting a customer-provided SBC to the Microsoft Phone System. The customer-provided SBC can be connected to almost any telephony trunk, or connect with third-party PSTN equipment. The connection allows:

- Using virtually any PSTN trunk with Microsoft Phone System
- Configuring interoperability between customer-owned telephony equipment, such as third-party PBXs, analog devices, and Microsoft Phone System

1.2 About AudioCodes SBC Product Series

AudioCodes' family of SBC devices enables reliable connectivity and security between the Enterprise's and the service provider's VoIP networks.

The SBC provides perimeter defense as a way of protecting Enterprises from malicious VoIP attacks; mediation for allowing the connection of any PBX and/or IP-PBX to any service provider; and Service Assurance for service quality and manageability.

Designed as a cost-effective appliance, the SBC is based on field-proven VoIP and network services with a native host processor, allowing the creation of purpose-built multiservice appliances, providing smooth connectivity to cloud services, with integrated quality of service, SLA monitoring, security and manageability. The native implementation of SBC provides a host of additional capabilities that are not possible with standalone SBC appliances such as VoIP mediation, PSTN access survivability, and third-party value-added services applications. This enables Enterprises to utilize the advantages of converged networks and eliminate the need for standalone appliances.

AudioCodes SBC is available as an integrated solution running on top of its field-proven Mediant Media Gateway and Multi-Service Business Router platforms, or as a software-only solution for deployment with third-party hardware. The SBC can be offered as a Virtualized SBC, supporting the following platforms: Hyper-V, AWS, AZURE, AWP, KVM and VMWare.

1.3 Validated AudioCodes SBC Version

Microsoft has successfully conducted validation tests with AudioCodes' Mediant SBC Ver. 7.40A.250. Previous certified firmware versions are 7.20A.258 and 7.40A.100. For updated list refer to *List of Session Border Controllers certified for Direct Routing*.

For implementing Microsoft Teams Direct Routing based on the configuration described in this document, AudioCodes SBC must be installed with a License Key that includes the following features:

- **MSFT** (general Microsoft license)
  Note: By default, all AudioCodes media gateways and SBCs are shipped with this license (except MSBR products, Mediant 500 SBC, and Mediant 500 Media Gateway).
- **SW/TEAMS** (Microsoft Teams license)
- **LAD** (Lync Analog Devices license), which uses AudioCodes' MP-1xx as the ATA to the Microsoft Teams environment through AudioCodes' SBC
- **Number of SBC sessions** (based on requirements)
- **Transcoding sessions** (only if media transcoding is needed)
- **Coders** (based on requirements)

For more information about the License Key, contact your AudioCodes sales representative.

2 Topology Example

Teams Direct Routing can be implemented in the *Enterprise* or *Hosting* Models.

2.1 Enterprise Model Implementation

The interoperability example between AudioCodes SBC and Company SIP Trunk with Teams Direct Routing Enterprise Model assume the following topology setup:

- Enterprise deployed with ATA, connected analog devices and the administrator's management station, located on the LAN
- Enterprise deployed with Teams Phone System Direct Routing Interface located on the WAN for enhanced communication within the Enterprise
- Enterprise wishes to offer its employees enterprise-voice capabilities and to connect the Enterprise to the PSTN network using Company's SIP Trunking service
- AudioCodes SBC is implemented to interconnect between the SIP Trunk and Teams Direct Routing located in the WAN

The figure below illustrates this topology example:

Figure 1: Connection Topology with SIP Trunk on the LAN

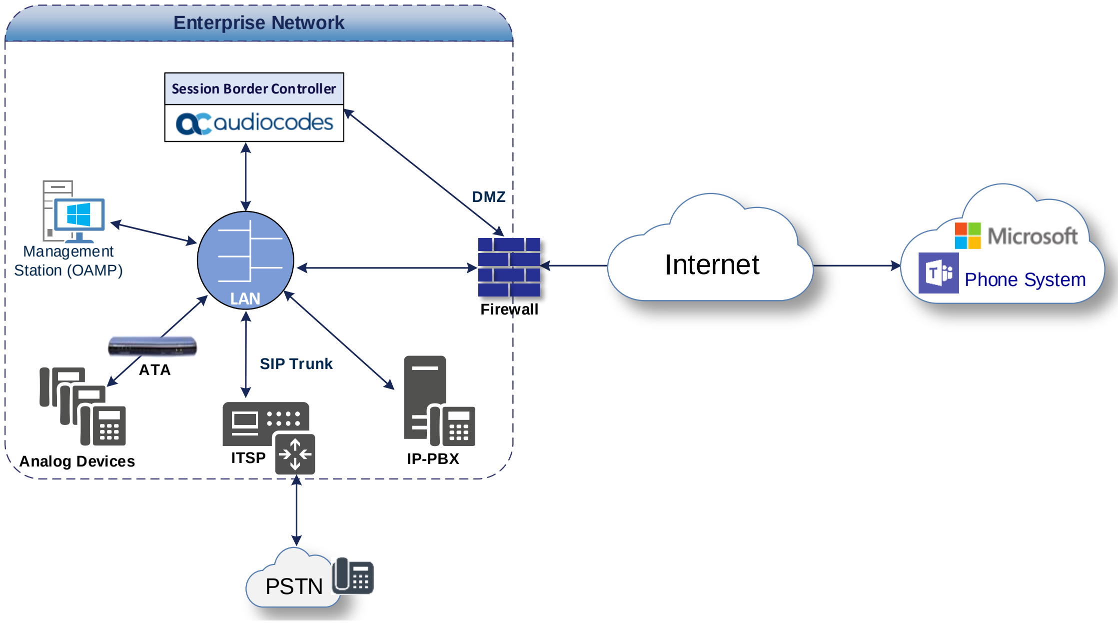
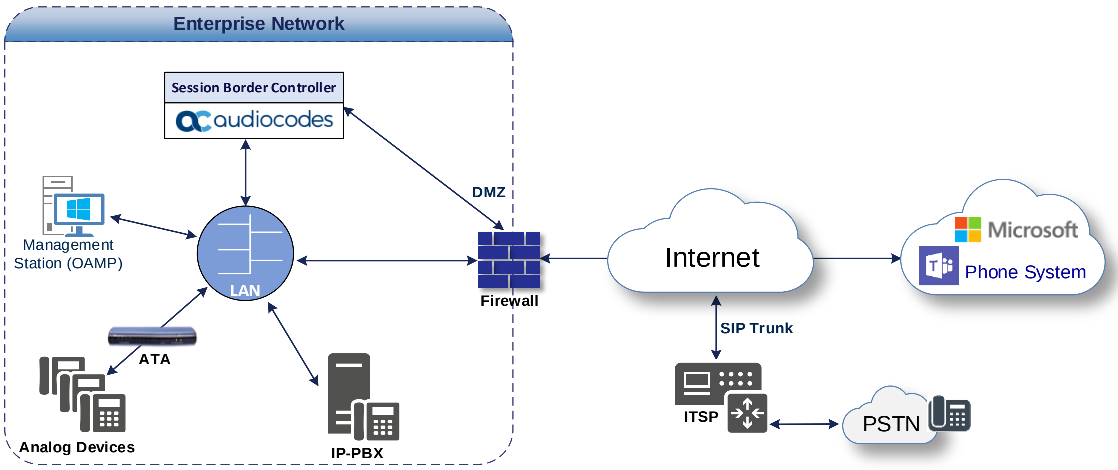


Figure 2: Connection Topology with SIP Trunk on the WAN



2.2 Environment Setup

The example topology includes the following environment setup:

Table 1: Environment Setup

Area	Setup
Network	■ Teams Direct Routing environment is located on the Enterprise's (or Service Provider's) WAN ■ Company SIP Trunk is located on the LAN
Signaling Transcoding	■ Teams Direct Routing operates with SIP-over-TLS transport type ■ Company SIP Trunk operates with SIP-over-UDP transport type
Codecs Transcoding	■ Teams Direct Routing supports G.711A-law, G.711U-law, G.729 and SILK (NB and WB) coders ■ Company SIP Trunk supports G.711A-law, G.711U-law, and G.729 coders
Media Transcoding	■ Teams Direct Routing operates with SRTP media type ■ Company SIP Trunk operates with RTP media type

2.3 Infrastructure Prerequisites

The table below shows the list of infrastructure prerequisites for deploying Teams Direct Routing.

Table 2-2: Infrastructure Prerequisites

Infrastructure Prerequisite	Details
Certified Session Border Controller (SBC)	See Microsoft's document *Plan Direct Routing.*
SIP Trunks connected to the SBC	
Office 365 Tenant	
Domains	
Public IP address for the SBC	
Fully Qualified Domain Name (FQDN) for the SBC	
Public DNS entry for the SBC	
Public trusted certificate for the SBC	
Firewall ports for Direct Routing Signaling	
Firewall IP addresses and ports for Direct Routing Media	
Media Transport Profile	
Firewall ports for Teams Clients Media	

3 Configuring Teams Direct Routing

This section describes an example of Teams Direct Routing configuration to operate with AudioCodes SBC.

3.1 Prerequisites

Before you begin configuration, make sure you have the following for every SBC you want to pair:

- Public IP address
- FQDN name matching SIP addresses of the users
- Public certificate issued by one of the supported CAs

3.2 SBC Domain Name in the Teams Enterprise Model

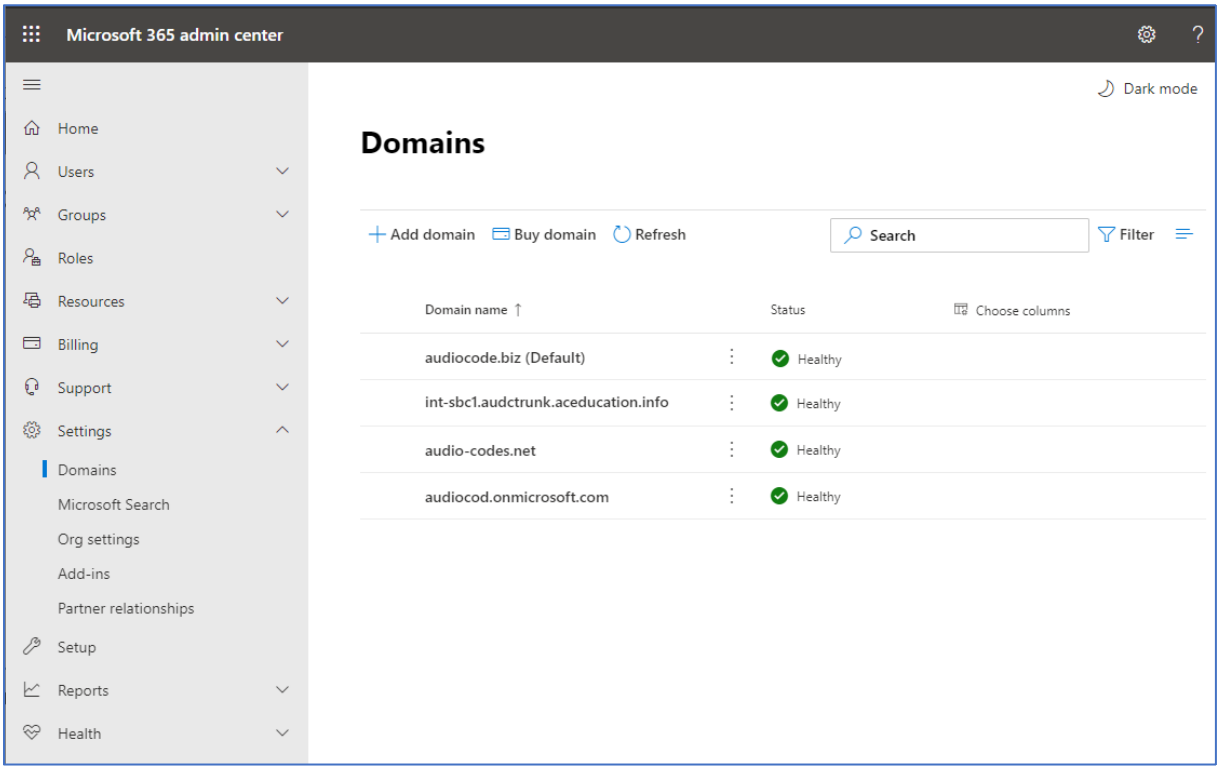
The SBC domain name must be from one of the names registered in 'Domains' of the tenant. You cannot use the ***.onmicrosoft.com** tenant for the domain name. For example, in Figure 3, the administrator registered the following DNS names for the tenant:

Table 2: DNS Names Registered by an Administrator for a Tenant

DNS name	Can be used for SBC FQDN	Examples of FQDN names
ACeducation.info	Yes	**Valid names**: ■ sbc.ACeducation.info ■ ussbcs15.ACeducation.info ■ europe.ACeducation.info **Invalid name:** sbc1.europe.ACeducation.info (requires registering domain name europe.atatum.biz in 'Domains' first)
adatumbiz.onmicrosoft.com	No	Using ***.onmicrosoft.com** domains is not supported for SBC names
hybridvoice.org	Yes	**Valid names**: ■ sbc1.hybridvoice.org ■ ussbcs15.hybridvoice.org ■ europe.hybridvoice.org **Invalid name:** sbc1.europe.hybridvoice.org (requires registering domain name europe.hybridvoice.org in 'Domains' first)

Users can be from any SIP domain registered for the tenant. For example, you can provide users user@ACeducation.info with the SBC FQDN **int-sbc1.audctrunk.aceducation.info** so long as both names are registered for this tenant.

Figure 3: Example of Registered DNS Names

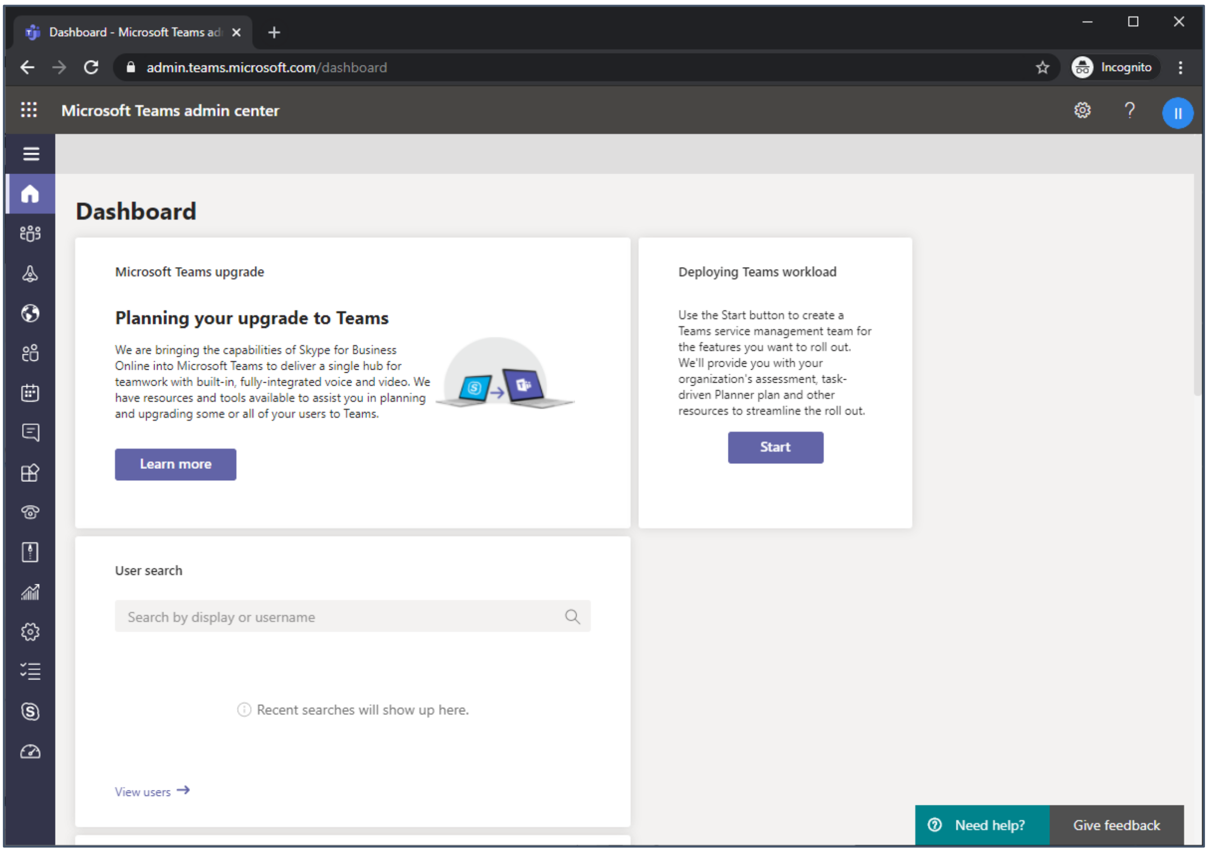


During creation of the Domain you will be forced to create public DNS record (**int-sbc1.audctrunk.aceducation.info** in our example.)

3.3 Example of the Office 365 Tenant Direct Routing Configuration

Configuration can be done using the web or with PowerShell. For the web, login to the Teams Admin Center (https://admin.teams.microsoft.com) with Tenant Administrator credentials.

Figure 4: Teams Admin Center



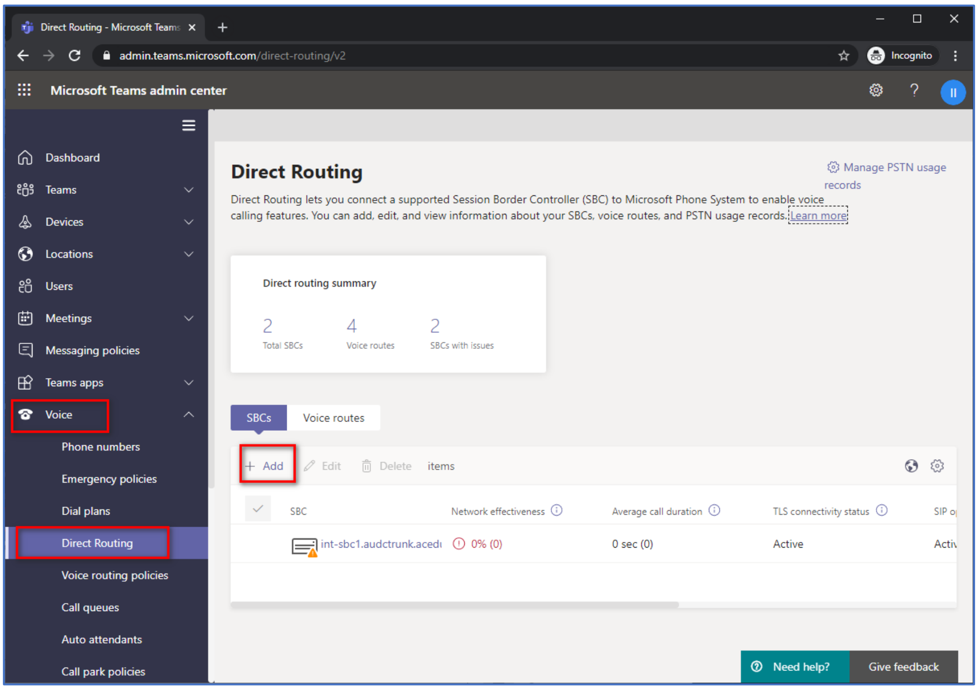
3.3.1 Adding New SBC to Direct Routing

The procedure below describes how add a new SBC to Direct Routing.

To add New SBC to Direct Routing:

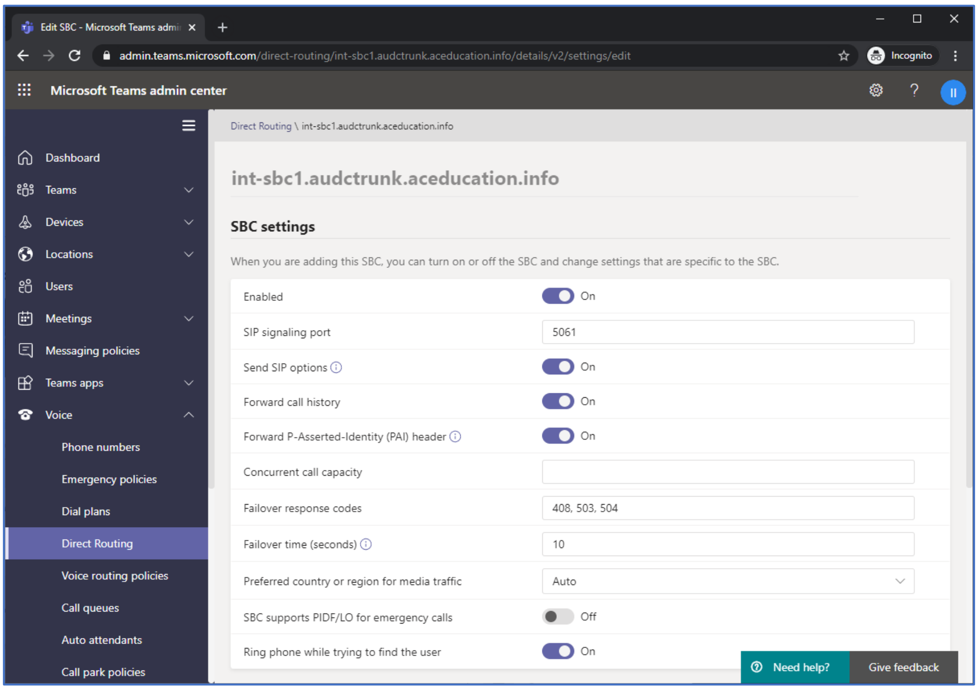
1. In the web interface, select **Voice**, and then click **Direct Routing**.
2. Under SBCs click **Add.**

Figure 5: Add new SBC to Direct Routing



3. Configure SBC.

Figure 6: Configure new SBC



You can use the following PowerShell command for creating a new Online PSTN Gateway:

```
New-CsOnlinePSTNGateway -Identity int-
sbc1.audctrunk.aceducation.info -SipSignalingPort 5061 -
ForwardCallHistory $True -ForwardPai $True -MediaBypass $True -
Enabled $True
```

Currently, enabling MediaBypass is available only through PowerShell.

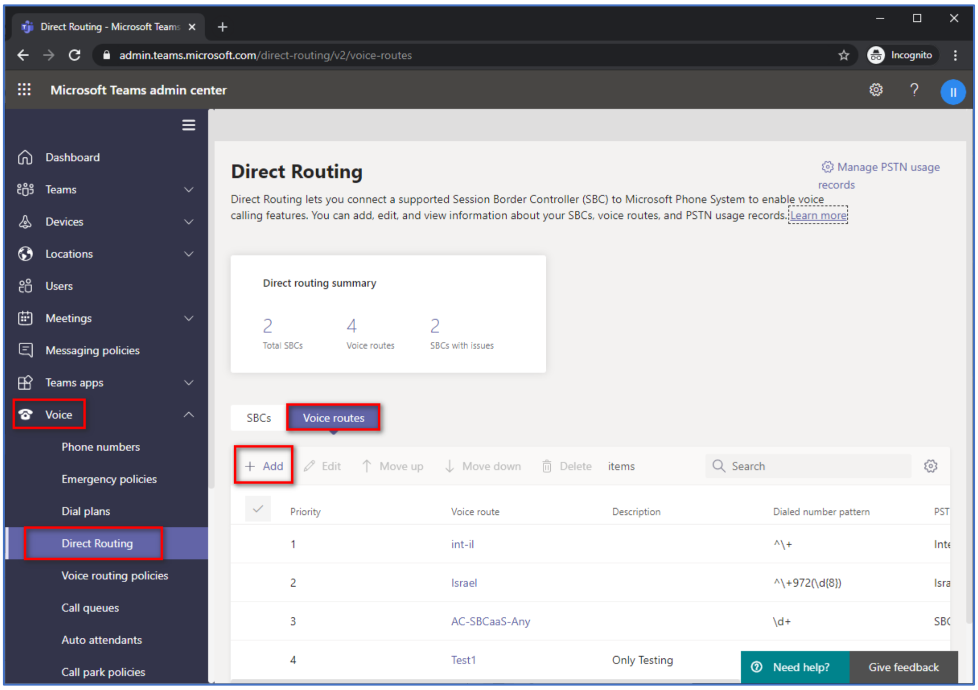
3.3.2 Adding Voice Route and PSTN Usage

The procedure below describes how add a voice route and PSTN usage.

To add voice route and PSTN usage:

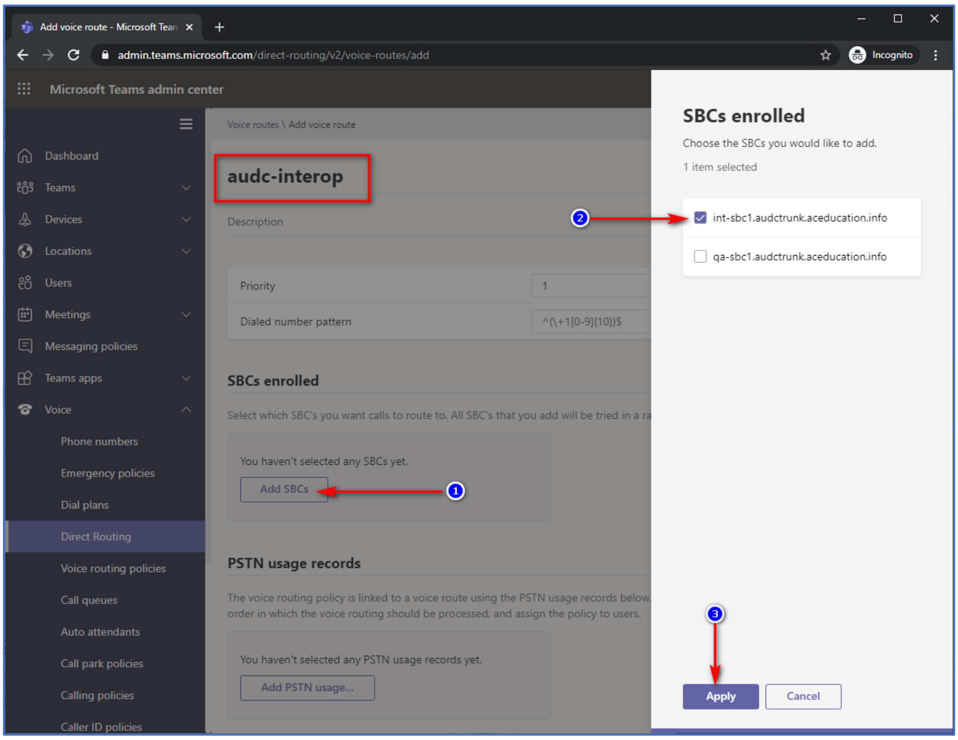
1. In the web interface, under **Direct Routing**, select **Voice routes**, and then click **Add**.

Figure 7: Add New Voice Route



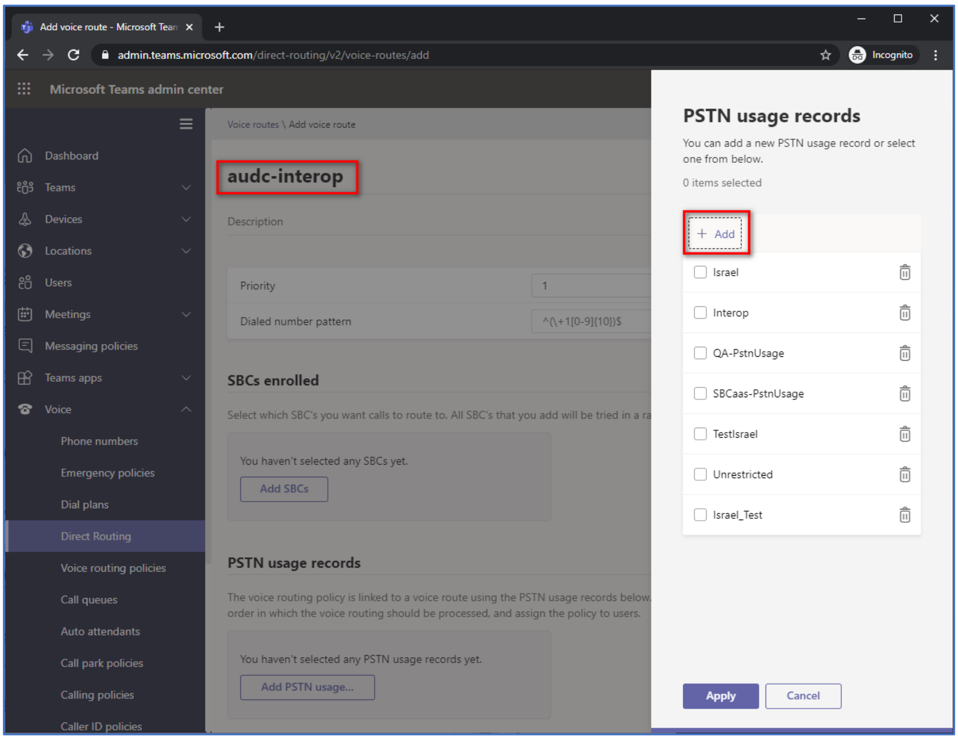
2. Create a new Voice Route and associate it with the SBC, configured in the previous step.

Figure 8: Associate SBC with new Voice Route



3. Add new (or associate existing) PSTN usage.

Figure 9: Associate PSTN Usage with New Voice Route



The same operations can be done using following PowerShell commands:

4. Creating an empty PSTN Usage:

```
Set-CsOnlinePstnUsage -Identity Global -Usage @{Add="Interop"}
```

5. Creating new Online Voice Route and associating it with PSTN Usage:

```
New-CsOnlineVoiceRoute -Identity "audc-interop" -NumberPattern
"^\+" -OnlinePstnGatewayList int-
sbc1.audctrunk.aceducation.info -Priority 1 -OnlinePstnUsages
"Interop"
```

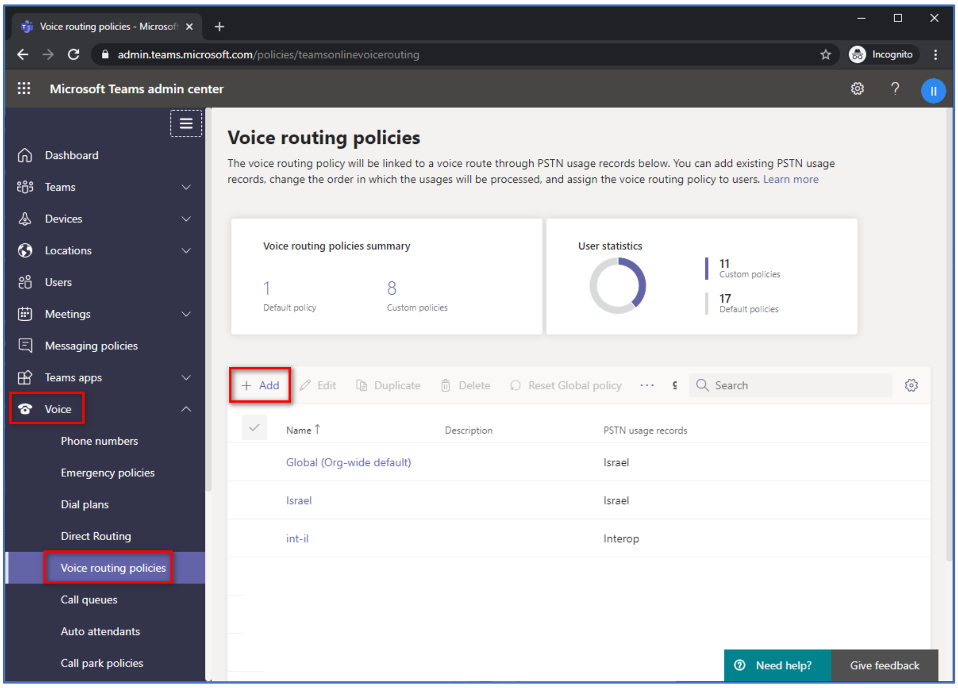
3.3.3 Adding Voice Routing Policy

The procedure below describes how add a voice routing policy

To add voice routing policy:

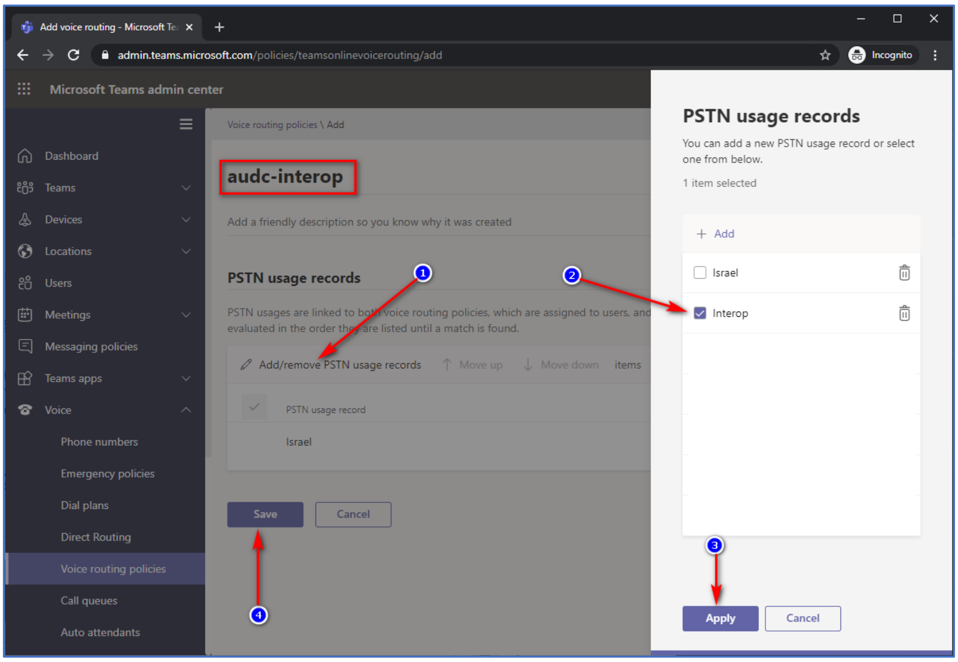
1. In the web interface, under **Voice**, select **Voice routing policies** and click **Add**.

Figure 10: Add New Voice Routing Policy



2. Create a new Voice Routing Policy and associate it with PSTN Usage, configured in the previous step.

Figure 11: Associate PSTN Usage with New Voice Routing Policy



The same operations can be done using following PowerShell command:

```
New-CsOnlineVoiceRoutingPolicy "audc-interop" -OnlinePstnUsages
"Interop"
```

> The commands specified in Sections 3.3.4 and 3.3.5, should be run **for each** Teams user (excluding ATA device users) in the company tenant. They are currently available through PowerShell **only**.

3.3.4 Enabling Online User

Use following PowerShell command for enabling online user:

```
Set-CsPhoneNumberAssignment -Identity user1@company.com -
EnterpriseVoiceEnabled $true
```

```
Set-CsPhoneNumberAssignment -Identity user1@company.com -
PhoneNumber +12345678901 -PhoneNumberType DirectRouting
```

3.3.5 Assigning Online User to the Voice Routing Policy

Use following PowerShell command for assigning online user to the Voice Route:

Grant-CsOnlineVoiceRoutingPolicy *-PolicyName "**audc-interop**" -Identity **user1@company.com***

> The command specified in Section 3.3.6 does not need to be run for each ATA device user, if the number pattern already points to the PSTNGateway and has been associated with PSTN Usage (see Section 3.3.2).

3.3.6 Analog Device Voice Route Configuration

Use the following PowerShell command for creating a new Online Voice Route and associating it with PSTN Usage:

New-CsOnlineVoiceRoute *-Identity "**audc-interop**" -NumberPattern "^\+**12345678901**" -OnlinePstnGatewayList **int-sbc1.audctrunk.aceducation.info** -Priority 1 -OnlinePstnUsages "**Interop**"*

3.3.7 Configuring with User Management Pack 365 (Optional)

As an alternative to PowerShell commands, AudioCodes recommend using User Management Pack 365 (UMP365). UMP365 provides a simple web-portal user interface for configuring and managing the Online Voice Route and associating it with PSTN Usage and PSTN Gateway. See examples below:

Figure 12: Example of Adding new Voice Route

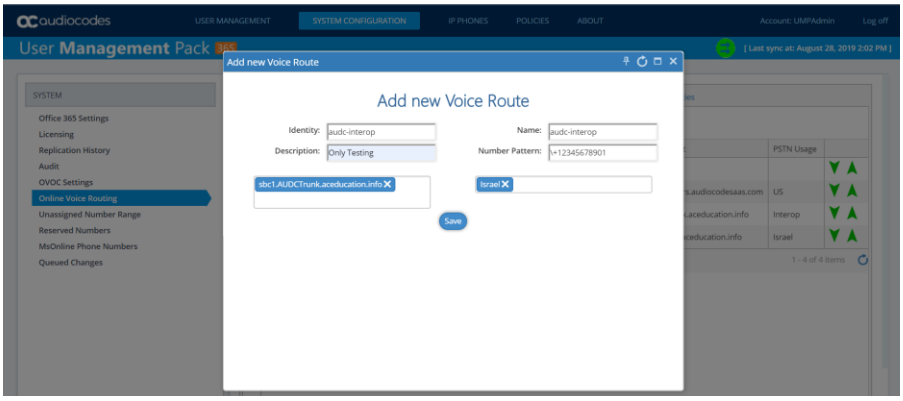
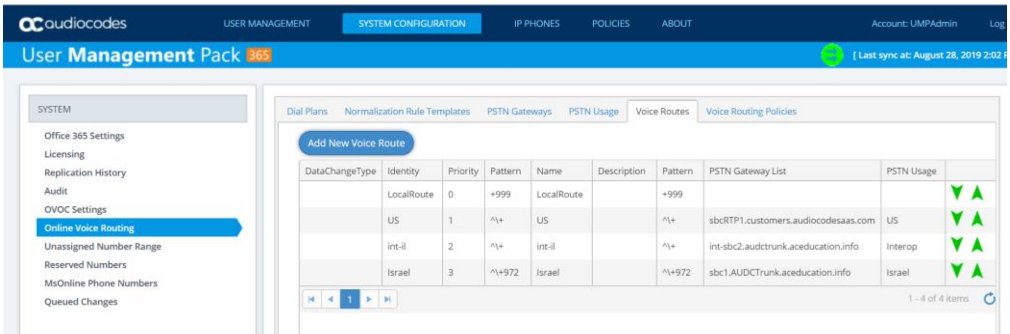


Figure 13: Example of Voice Routes Table



4 Configuring AudioCodes SBC

This section provides example of step-by-step procedures on how to configure AudioCodes SBC for interworking between Teams Direct Routing and the Company SIP Trunk. These configuration procedures are based on the topology example described in Section 2.1.1 on page 11, and includes the following main areas:

- SBC LAN interface – ATA devices environment
- SBC WAN interface - Company SIP Trunking and Teams Direct Routing environment

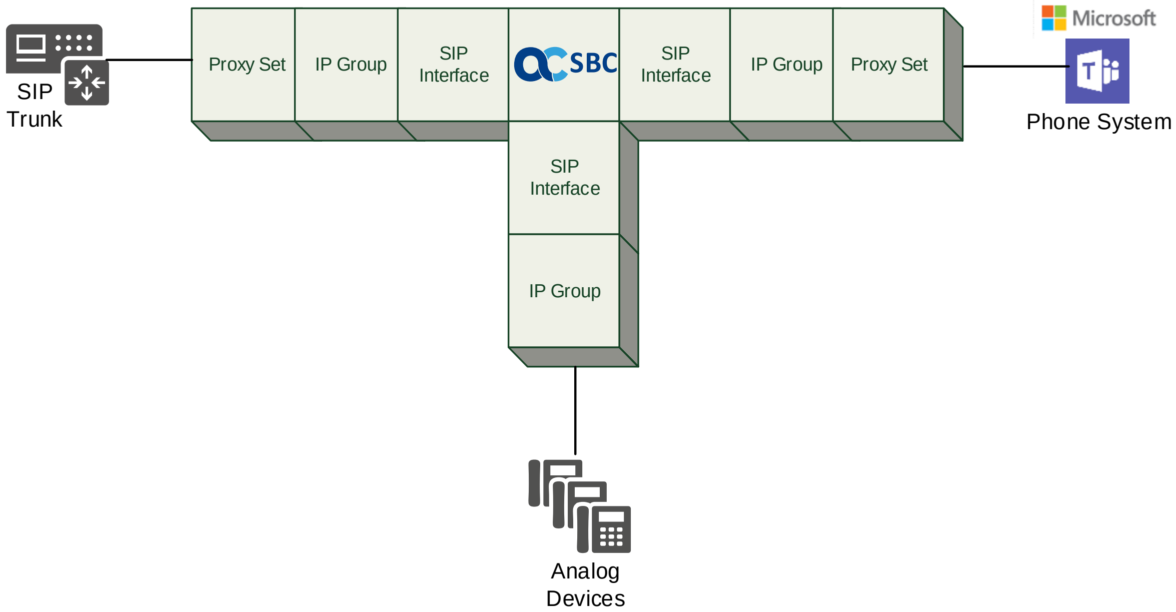
This configuration is done using the SBC's embedded Web server (hereafter, referred to as *Web interface*).

- For implementing Teams Direct Routing based on the configuration described in this section, AudioCodes SBC must be installed with a License Key. For more information, see Section 1.3 on page 10.
- The scope of this document does **not** cover all security aspects for configuring this topology. Comprehensive security measures should be implemented per your organization's security policies. For security recommendations on AudioCodes' products, refer to the *Recommended Security Guidelines* document, which can be found at AudioCodes web site

4.1 SBC Configuration Concept in Teams Direct Routing

The diagram below represents AudioCodes' device configuration concept.

Figure 14: SBC Configuration Concept

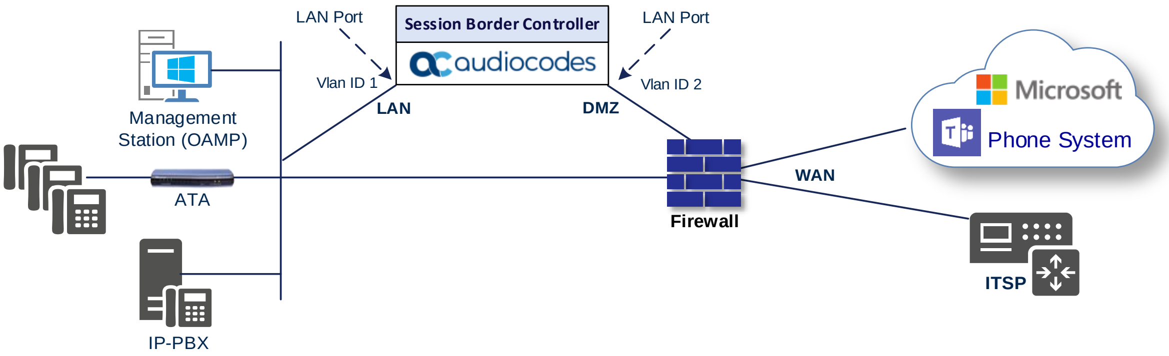


4.2 IP Network Interfaces Configuration

This section describes how to configure the SBC's IP network interfaces. There are several ways to deploy the SBC; however, this example employs the following deployment method:

- SBC interfaces with the following IP entities:
  - Teams Direct Routing and Company SIP Trunk, located on the WAN
  - IP-PBX and/or ATA, located on the LAN
- SBC connects to the WAN through a DMZ network
- Physical connection: The type of physical connection depends on the method used to connect to the Enterprise's network. In the example topology, SBC connects to the LAN and DMZ using dedicated ethernet ports (i.e., two ports and two network cables are used).
- SBC also uses two logical network interfaces:
  - LAN (VLAN ID 1)
  - DMZ (VLAN ID 2)

Figure 15: Network Interfaces in the Example Topology



4.2.1 Configuring VLANs

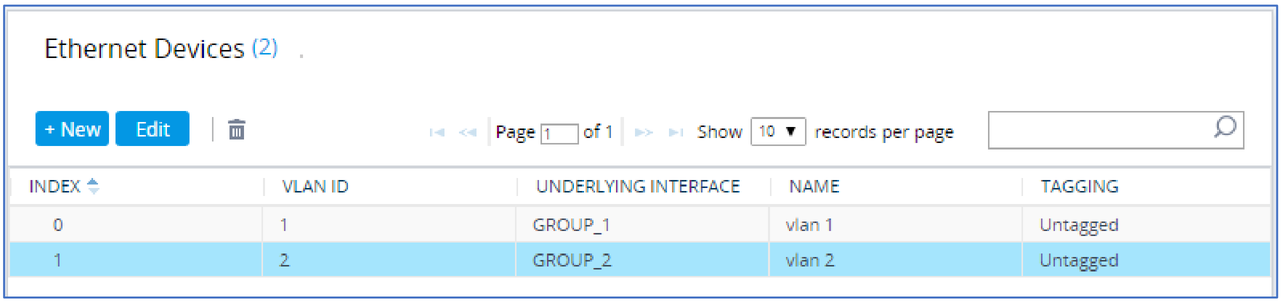
This section describes how to configure VLANs for each of the following interfaces:

- LAN VoIP (assigned the name "LAN_IF")
- WAN VoIP (assigned the name "WAN_IF")

To configure the VLANs:

1. Open the Ethernet Device table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **Ethernet Devices**).
2. There will be one existing row for VLAN ID 1 and underlying interface GROUP_1.
3. Add another VLAN ID 2 for the WAN side

Figure 16: Configured VLAN IDs in Ethernet Device



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

4.2.2 Configuring Network Interfaces

This section describes how to configure the IP network interfaces for each of the following interfaces:

- LAN Interface (assigned the name "LAN_IF")
- WAN Interface (assigned the name "WAN_IF")

To configure the IP network interfaces:

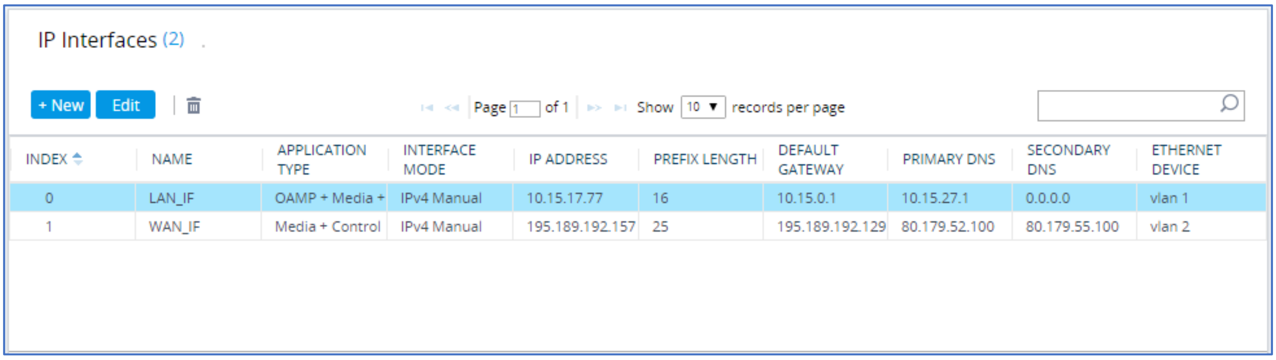
1. Open the IP Interfaces table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **IP Interfaces**).
2. Configure the IP interfaces as follows (your network parameters might be different):

Table 3: Configuration Example of the Network Interface Table

Index	Application Types	Interface Mode	IP Address	Prefix Length	Gateway	DNS	I/F Name	Ethernet Device
0	OAMP+ Media + Control	IPv4 Manual	10.15.77.77	16	10.15.0.1	10.15.27.1	LAN_IF	vlan 1
1	Media + Control (as this interface points to the internet, enabling OAMP is not recommended)	IPv4 Manual	195.189.192.157 (DMZ IP address of SBC)	25	195.189.192.129 (router's IP address)	According to your Internet provider's instructions	WAN_IF	vlan 2

The configured IP network interfaces are shown below:

Figure 17: Configured Network Interfaces in IP Interfaces Table



4.3 SIP TLS Connection Configuration

This section describes how to configure the SBC for using a TLS connection with the Teams Direct Routing Phone System. This configuration is essential for a secure SIP TLS connection. The configuration instructions example in this section are based on the following domain structure that must be implemented as part of the certificate which must be loaded to the host SBC:

- CN: int-sbc1.audctrunk.aceducation.info
- SAN: int-sbc1.audctrunk.aceducation.info

This certificate module is based on the Service Provider's own TLS Certificate. For more certificate structure options, see Microsoft Teams Direct Routing documentation.

The Phone System Direct Routing Interface allows ***only*** TLS connections from SBCs for SIP traffic with a certificate signed by one of the Trusted Certification Authorities.

Currently, supported Certification Authorities can be found in the following link:

https://docs.microsoft.com/en-us/microsoftteams/direct-routing-plan#public-trusted-certificate-for-the-sbc

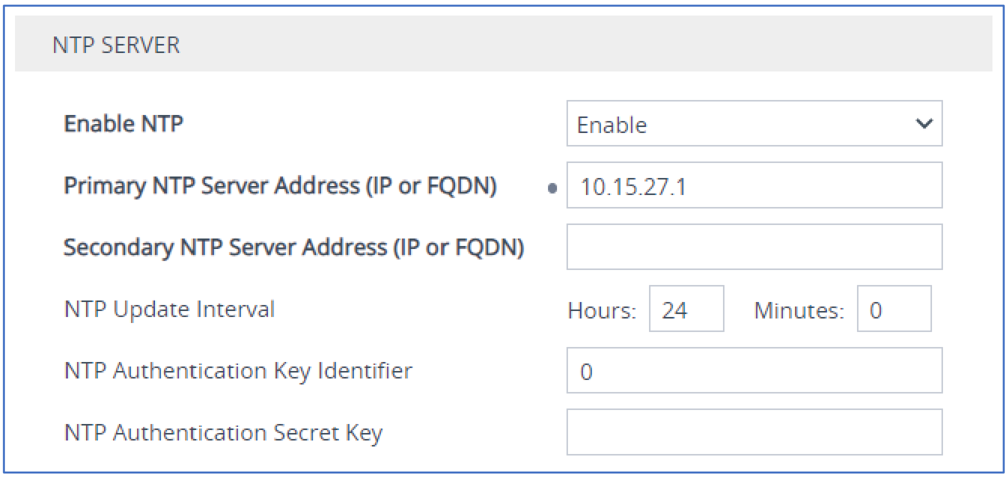
4.3.1 Configuring the NTP Server Address

This section describes how to configure the NTP server's IP address. It is recommended to implement an NTP server (Microsoft NTP server or another global server) to ensure that the SBC receives the current date and time. This is necessary for validating certificates of remote parties. It is important, that NTP Server will locate on the OAMP IP Interface (LAN_IF in our case) or will be accessible through it.

To configure the NTP server address:

1. Open the Time & Date page (**Setup** menu > **Administration** tab > **Time & Date**).
2. In the 'Primary NTP Server Address' field, enter the IP address of the NTP server (e.g., **10.15.27.1**).

Figure 18: Configuring NTP Server Address



3. Click **Apply**.

4.3.2 Creating a TLS Context for Teams Direct Routing

This section describes how to configure TLS Context in the SBC. AudioCodes recommends implementing only TLS to avoid flaws in SSL.

To configure the TLS version:

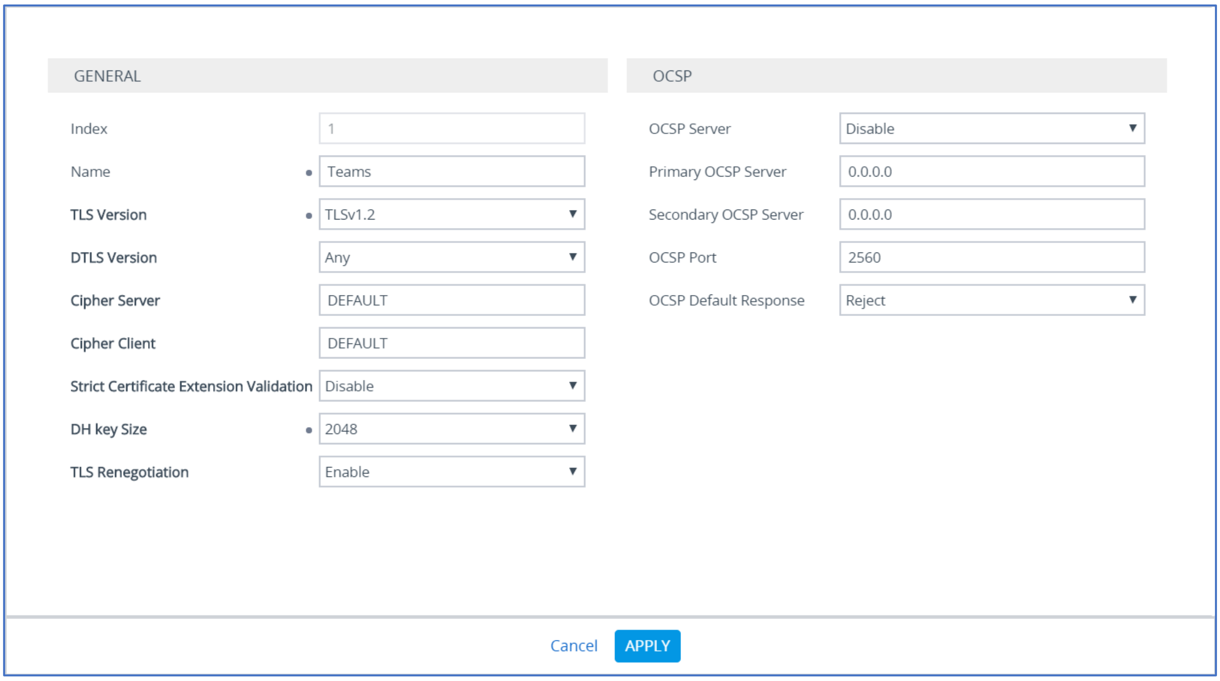
1. Open the TLS Contexts table (**Setup** menu > **IP Network** tab > **Security** folder > **TLS Contexts**).
2. Create a new TLS Context by clicking **New** at the top of the interface, and then configure the parameters using the table below as reference:

Table 4: New TLS Context

Index	Name	TLS Version
1	Teams (arbitrary descriptive name)	TLSv1.2
All other parameters can be left unchanged with their default values.		

> The table above exemplifies configuration focusing on interconnecting SIP and media. You might want to configure additional parameters according to your company's policies. For example, you might want to configure Online Certificate Status Protocol (OCSP) to check if SBC certificates presented in the online server are still valid or revoked. For more information on the SBC's configuration, see the *User's Manual*, available for download from https://www.audiocodes.com/library/technical-documents.

Figure 19: Configuring TLS Context for Teams Direct Routing



3. Click **Apply**.

4.3.3 Configuring a Certificate

This section describes how to request a certificate for the SBC and to configure it based on the example of DigiCert Global Root CA. The certificate is used by the SBC to authenticate the connection with Teams Direct Routing.

The procedure involves the following main steps:

a. Generating a Certificate Signing Request (CSR).
b. Requesting Device Certificate from CA.
c. Obtaining Trusted Root/ Intermediate Certificate from CA.
d. Deploying Device and Trusted Root/ Intermediate Certificates on SBC.

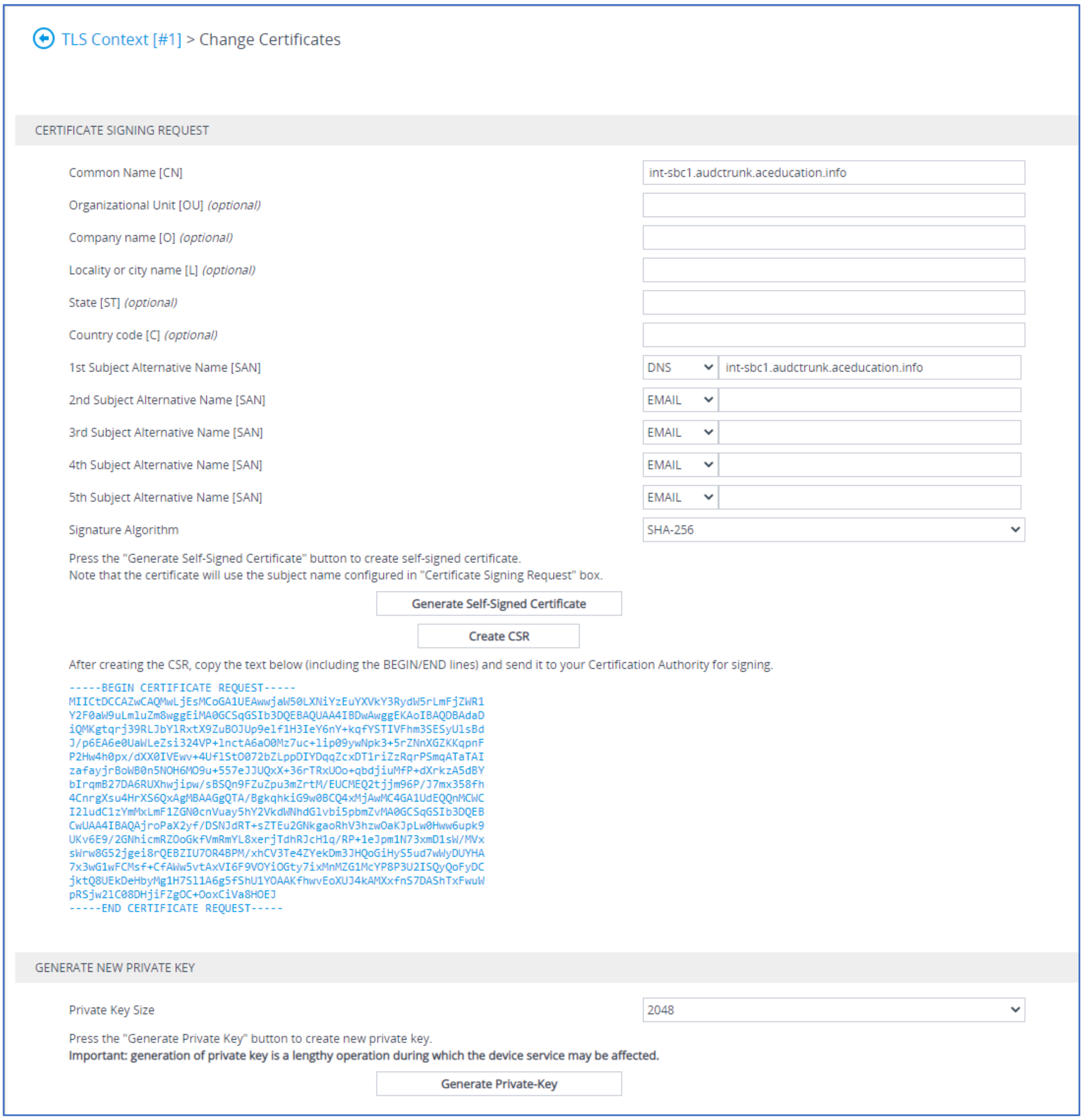
To configure a certificate:

1. Open the TLS Contexts page (**Setup** menu > **IP Network** tab > **Security** folder > **TLS Contexts**).
2. In the TLS Contexts page, select the required TLS Context index row, and then click the **Change Certificate** link located below the table; the Context Certificates page appears.
3. Under the **Certificate Signing Request** group, do the following:
    a. In the 'Subject Name [CN]' field, enter the SBC FQDN name (based on example above, **int-sbc1.audctrunk.aceducation.info**).
    b. In the '1st Subject Alternative Name [SAN]' field, change the type to 'DNS' and enter the SBC FQDN name (based on example above, **int-sbc1.audctrunk.aceducation.info**).

> The domain portion of the Common Name [CN] and 1st Subject Alternative Name [SAN] must match the SIP suffix configured for Office 365 users.

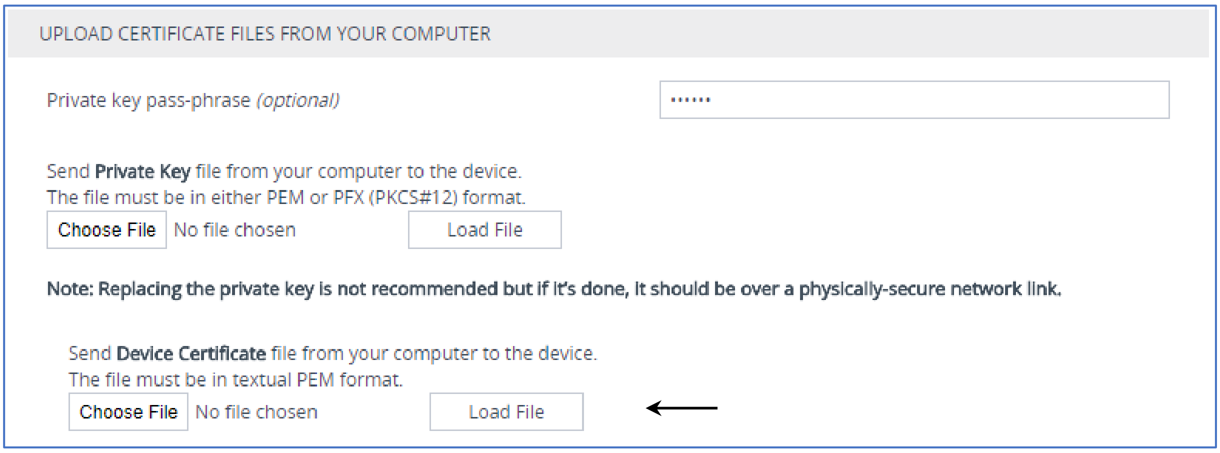
    c. Fill in the rest of the request fields according to your security provider's instructions.
    d. Click the **Create CSR** button; a textual certificate signing request is displayed in the area below the button:

Figure 20: Example of Certificate Signing Request – Creating CSR



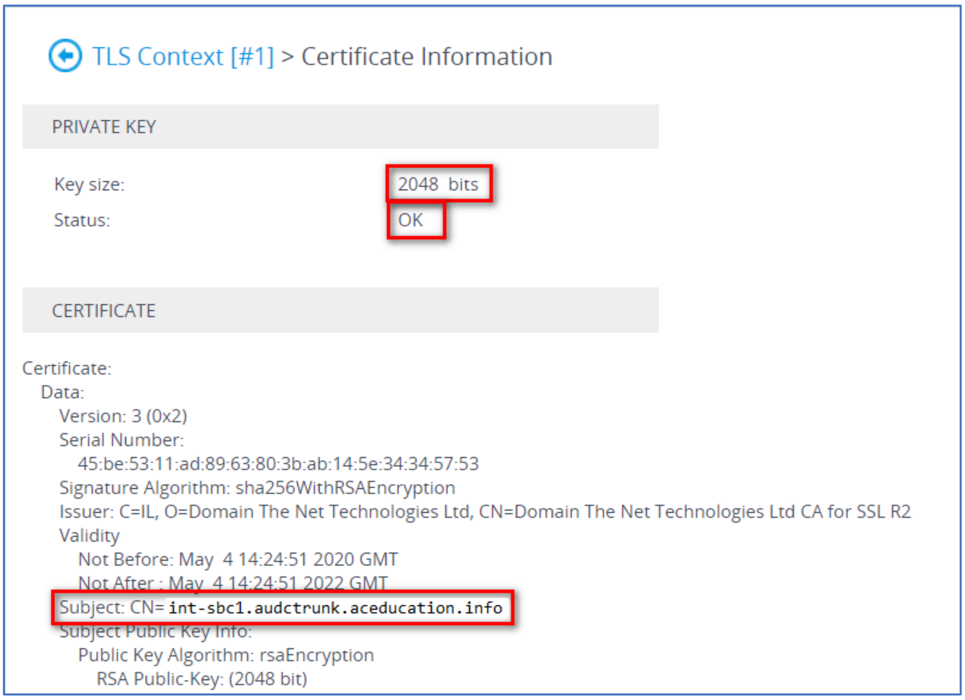
4. Copy the CSR from the line **"----BEGIN CERTIFICATE" to "END CERTIFICATE REQUEST----"** to a text file (such as Notepad), and then save it to a folder on your computer with the file name, for example *certreq.txt*.
5. Send *certreq.txt* file to the Certified Authority Administrator for signing.
6. After obtaining an SBC signed and Trusted Root/Intermediate Certificate from the CA, in the SBC's Web interface, return to the **TLS Contexts** page and do the following:
   a. In the TLS Contexts page, select the required TLS Context index row, and then click the **Change Certificate** link located below the table; the Context Certificates page appears.
   b. Scroll down to the **Upload certificates files from your computer** group, click the **Choose File** button corresponding to the '**Send Device Certificate**...' field, navigate to the certificate file obtained from the CA, and then click **Load File** to upload the certificate to the SBC.

Figure 21: Uploading the Certificate Obtained from the Certification Authority



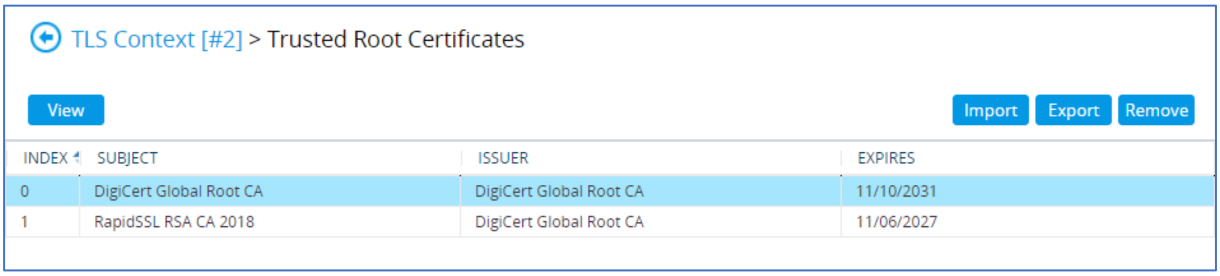
7. Confirm that the certificate was uploaded correctly. A message indicating that the certificate was uploaded successfully is displayed in blue in the lower part of the page.
8. In the SBC's Web interface, return to the **TLS Contexts** page, select the required TLS Context index row, and then click the **Certificate Information** link, located at the bottom of the TLS. Then validate the Key size, certificate status and Subject Name:

Figure 22: Certificate Information Example



9. In the SBC's Web interface, return to the **TLS Contexts** page.
   a. In the TLS Contexts page, select the required TLS Context index row, and then click the **Trusted Root Certificates** link, located at the bottom of the TLS Contexts page; the Trusted Certificates page appears.
   b. Click the **Import** button, and then select all Root/Intermediate Certificates obtained from your Certification Authority to load.
10. Click **OK**; the certificate is loaded to the device and listed in the Trusted Certificates store:

Figure 23: Example of Configured Trusted Root Certificates



4.3.4 Method for Generating and Installing the Wildcard Certificate

To use the same certificate on multiple devices, you may prefer using 3rd party application (e.g. DigiCert Certificate Utility for Windows) to process the certificate request from your Certificate Authority on another machine, with this utility installed.

After you've processed the certificate request and response using the DigiCert utility, test the certificate private key and chain and then export the certificate with private key and assign a password.

To install the certificate:

1. Open the TLS Contexts page (**Setup** menu > **IP Network** tab > **Security** folder > **TLS Contexts**).
2. In the TLS Contexts page, select the required TLS Context index row, and then click the **Change Certificate** link located below the table; the Context Certificates page appears.
3. Scroll down to the **Upload certificates files from your computer** group and do the following:
    a. Enter the password assigned during export with the DigiCert utility in the **'Private key pass-phrase**' field.
    b. Click the **Choose File** button corresponding to the 'Send **Private Key**...' field and then select the SBC certificate file exported from the DigiCert utility.

4.3.5 Deploying Trusted Root Certificate for MTLS connection

Loading Trusted Root Certificates into AudioCodes' SBC mandatory when implementing MTLS connection with Microsoft.

Microsoft 365 is updating services powering messaging, meetings, telephony, voice, and video to use TLS certificates from a different set of Root Certificate Authorities (CAs). For more details of the new Root CAs, refer to Microsoft technical guidance at Office TLS Certificate Changes. Services began transitioning to the new Root CAs (e.g., DigiCert) beginning in January 2022 and will continue through October 2022. During this migration period, it's possible to load both the old (Baltimore) and the new (DigiCert) Root certificate to the same TLS Context.

The DNS name of the Teams Direct Routing interface is **sip.pstnhub.microsoft.com**. In this interface, a certificate is presented which is signed by **DigiCert** with:

Serial Number: 0x033af1e6a711a9a0bb2864b11d09fae5,
SHA-1 Thumbprint: DF3C24F9BFD666761B268073FE06D1CC8D4F82A4 and
SHA-256 Thumbprint: CB3CCBB76031E5E0138F8DD39A23F9DE47FFC35E43C1144CEA27D46A5AB1CB5F.

To trust this certificate, your SBC *must* have the certificate in Trusted Certificates storage. Download the **DigiCert Global Root G2** (df3c) certificate in **PEM format** from https://www.digicert.com/kb/digicert-root-certificates.htm and follow the steps above to import the certificate to the Trusted Root storage.

Before importing the DigiCert Root Certificate into AudioCodes' SBC, make sure it's in .PEM or .PFX format. If it isn't, you need to convert it to .PEM or .PFX format. Otherwise, you will receive a 'Failed to load new certificate' error message. To convert to PEM format, use the Windows local store on any Windows OS and then export it as 'Base-64 encoded X.509 (.CER) certificate'.

4.4 Configuring Media Realms

This section describes how to configure Media Realms. Media Realms allow the dividing of UDP port ranges for use on different interfaces. The simplest configuration is to create Media Realms for internal (ATA) and external (Teams and SIP Trunk) traffic.

To configure Media Realms:

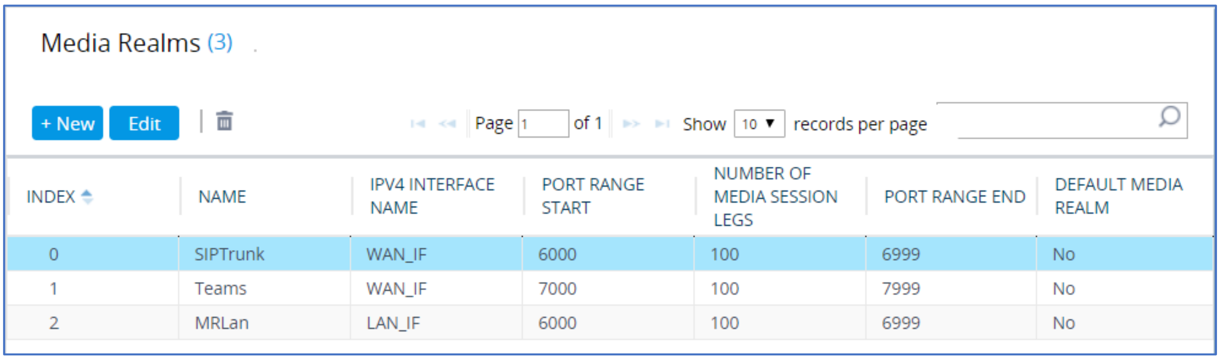
1. Open the Media Realms table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Media Realms**).
2. Configure Media Realms as follows (you can use the default Media Realm (Index 0), but modify it):

Table 5: Configuration Example Media Realms in Media Realm Table

Index	Name	Topology Location	IPv4 Interface Name	Port Range Start	Number of Media Session Legs
0	SIPTrunk (arbitrary name)	Up	WAN_IF	6000	100 (media sessions assigned with port range)
1	Teams (arbitrary name)	Up	WAN_IF	7000	100 (media sessions assigned with port range)
2	MRLan (arbitrary name)		LAN_IF	6000	100 (media sessions assigned with port range)

The configured Media Realms are shown in the figure below:

Figure 24: Configured Media Realms in Media Realm Table


Media Realms (3)

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

INDEX	NAME	IPV4 INTERFACE NAME	PORT RANGE START	NUMBER OF MEDIA SESSION LEGS	PORT RANGE END	DEFAULT MEDIA REALM
0	SIPTrunk	WAN_IF	6000	100	6999	No
1	Teams	WAN_IF	7000	100	7999	No
2	MRLan	LAN_IF	6000	100	6999	No

4.5 Configuring SIP Signaling Interfaces

This section describes how to configure SIP Interfaces. A SIP Interface defines a listening port and type (UDP, TCP, or TLS) for SIP signaling traffic on a specific logical IP network interface (configured in the Interface Table above) and Media Realm.

Note that the configuration of a SIP interface for the SIP Trunk and ATA device shows an example, which may be different to your configuration. For specific configuration of interfaces relating to SIP trunks and/or a third-party PSTN environment connected to the SBC, see the trunk / environment vendor documentation.

To configure SIP Interfaces:

1. Open the SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **SIP Interfaces**).
2. Configure SIP Interfaces. You can use the default SIP Interface (Index 0), however, modify it as shown in the table below. The table below shows an example of the configuration. You can change some of the parameters according to your requirements.

> The Direct Routing interface can only use TLS for a SIP port. It does not support using TCP due to security reasons. The SIP port might be any port of your choice. When pairing the SBC with Office 365, the chosen port is specified in the pairing command.

Table 6: Configuration Example of SIP Signaling Interfaces

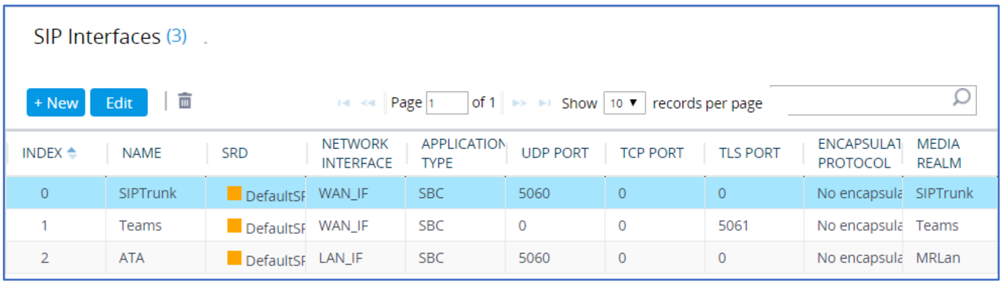
Index	Name	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	Enable TCP Keepalive	Classification Failure Response Type	Media Realm	TLS Context Name
0	**SIPTrunk** (arbitrary name)	**WAN_IF**	**SBC**	**5060** (according to Service Provider requirement)	**0**	**0**	**Disable** (leave default value)	**500** (leave default value)	**SIPTrunk**	-
1	**Teams** (arbitrary name)	**WAN_IF**	**SBC**	**0** (Phone System does not use UDP or TCP for SIP signaling)	**0**	**5061** (as configured in the Office 365)	**Enable**	**0** (Recommended to prevent DoS attacks)	**Teams**	**Teams**
2	**ATA** (arbitrary name)	**LAN_IF**	**SBC**	**5060** (according to Service Provider requirement)	**0**	**0**	**Disable** (leave default value)	**500** (leave default value)	**MRLan**	-

> For implementing an MTLS connection with the Microsoft Teams network, configure ‘TLS Mutual Authentication’ to “Enable” for Teams SIP Interface.

> Loading DigiCert Trusted Root Certificates to AudioCodes' SBC is mandatory for implementing an MTLS connection with the Microsoft Teams network. Refer to Section 4.3.5 on page 34.

The configured SIP Interfaces are shown in the figure below:

Figure 25: Configured SIP Interfaces in SIP Interface Table



INDEX	NAME	SRD	NETWORK INTERFACE	APPLICATION TYPE	UDP PORT	TCP PORT	TLS PORT	ENCAPSULAT PROTOCOL	MEDIA REALM
0	SIPTrunk	DefaultSF	WAN_IF	SBC	5060	0	0	No encapsula	SIPTrunk
1	Teams	DefaultSF	WAN_IF	SBC	0	0	5061	No encapsula	Teams
2	ATA	DefaultSF	LAN_IF	SBC	5060	0	0	No encapsula	MRLan

4.6 Configuring Proxy Sets and Proxy Address

4.6.1 Configuring Proxy Sets

This section describes how to configure Proxy Sets. The Proxy Set defines the destination address (IP address or FQDN) of the IP entity server. Proxy Sets can also be used to configure load balancing between multiple servers.

For the example topology, Proxy Sets need to be configured for the following IP entities:

- Company SIP Trunk
- Teams Direct Routing

The Proxy Sets will later be applied to the VoIP network by assigning them to IP Groups.

To configure Proxy Sets:

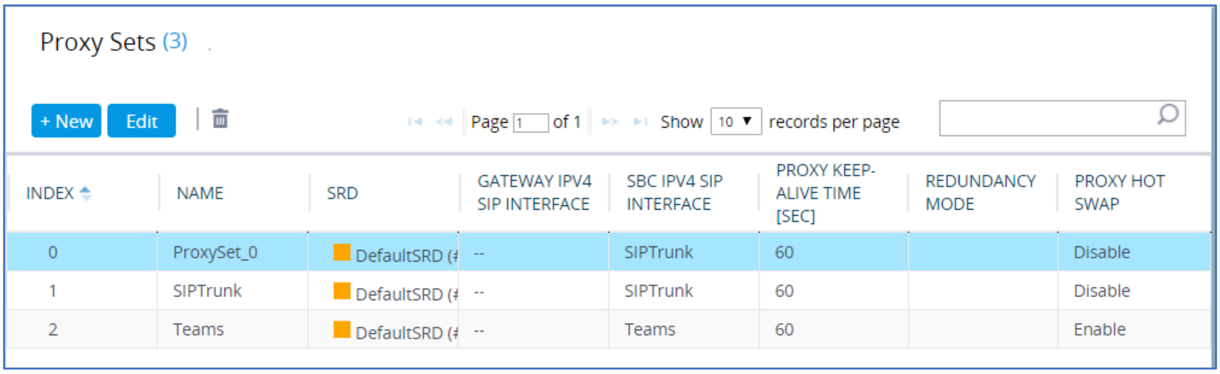
1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**).
2. Configure Proxy Sets as shown in the table below:

Table 7: Configuration Example Proxy Sets in Proxy Sets Table

Index	Name	SBC IPv4 SIP Interface	TLS Context Name	Proxy Keep-Alive	Proxy Hot Swap	Proxy Load Balancing Method
1	SIPTrunk (arbitrary name)	SIPTrunk	Default	Using Options	-	-
2	Teams (arbitrary name)	Teams	Teams	Using Options	Enable	Random Weights

The configured Proxy Sets are shown in the figure below:

Figure 26: Configured Proxy Sets in Proxy Sets Table


Proxy Sets (3)

+ New Edit

Page 1 of 1 Show 10 records per page

INDEX	NAME	SRD	GATEWAY IPV4 SIP INTERFACE	SBC IPV4 SIP INTERFACE	PROXY KEEP-ALIVE TIME [SEC]	REDUNDANCY MODE	PROXY HOT SWAP
0	ProxySet_0	DefaultSRD (#	--	SIPTrunk	60		Disable
1	SIPTrunk	DefaultSRD (#	--	SIPTrunk	60		Disable
2	Teams	DefaultSRD (#	--	Teams	60		Enable

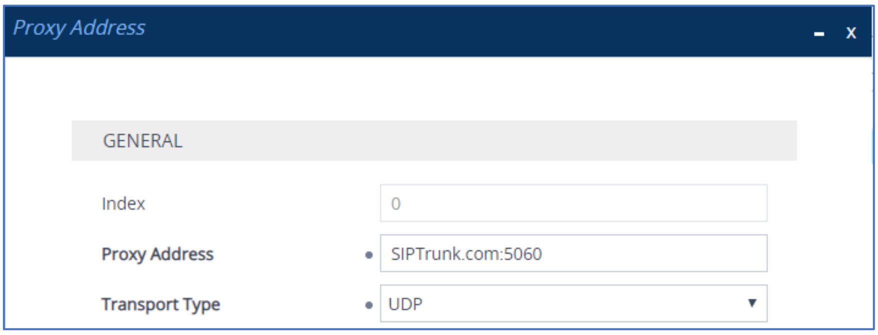
4.6.2 Configuring a Proxy Address

This section shows how to configure a Proxy Address for the SIP Trunk and Teams entities.

To configure a Proxy Address for SIP Trunk:

1. Open the Proxy Sets table (Setup menu > **Signaling & Media** tab > **Core Entities** folder > Proxy Sets) and then click the Proxy Set **SIPTrunk**, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.
2. Click **+New**; the following dialog box appears:

Figure 27: Configuring Proxy Address for SIP Trunk



3. Configure the address of the Proxy Set according to the parameters described in the table below:

Table 8: Configuration Proxy Address for SIP Trunk

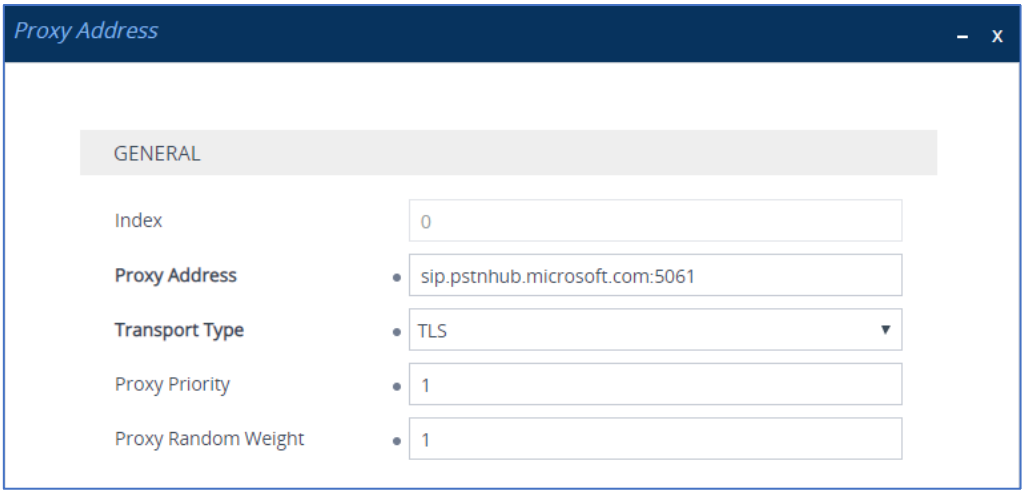
Index	Proxy Address	Transport Type	Proxy Priority	Proxy Random Weight
0	SIPTrunk.com:5060 (SIP Trunk IP / FQDN and port)	UDP	0	0

4. Click **Apply**.

To configure a Proxy Address for Teams:

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**) and then click the Proxy Set **Teams**, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.
2. Click **+New**; the following dialog box appears:

Figure 28: Configuring Proxy Address for Teams Direct Routing Interface



3. Configure the address of the Proxy Set according to the parameters described in the table below:

Table 9: Configuration Proxy Address for Teams Direct Routing

Index	Proxy Address	Transport Type	Proxy Priority	Proxy Random Weight
0	sip.pstnhub.microsoft.com:5061	TLS	1	1
1	sip2.pstnhub.microsoft.com:5061	TLS	2	1
2	sip3.pstnhub.microsoft.com:5061	TLS	3	1

4. Click Appl**y**.

If the SBC is deployed in Office 365 GCC DoD or GCC High environments, please contact AudioCodes deployment services, since these environments have different configurations (FQDNs) than the public Office 365 environment.

4.7 Configuring Coders

This section describes how to configure coders (termed *Coder Group*). As Teams Direct Routing supports the SILK and OPUS coders while the network connection to Company SIP Trunk may restrict operation with a dedicated coders list, you need to add a Coder Group with the supported coders for each leg, the Teams Direct Routing and the Company SIP Trunk.

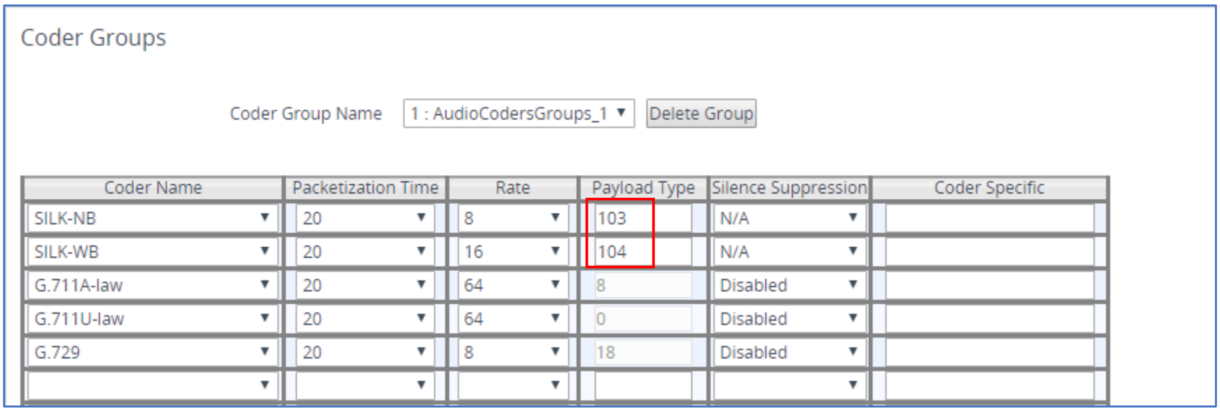
Note that the Coder Group ID for this entity will be assigned to its corresponding IP Profile in the next step.

To configure coders:

1. Open the Coder Groups table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **Coder Groups**).
2. Configure a Coder Group for Teams Direct Routing:

Parameter	Value
Coder Group Name	**AudioCodersGroups_1**
Coder Name	■ **SILK-NB** ■ SILK-WB ■ G.711 A-law ■ G.711 U-law ■ G.729

Figure 29: Configuring Coder Group for Teams Direct Routing



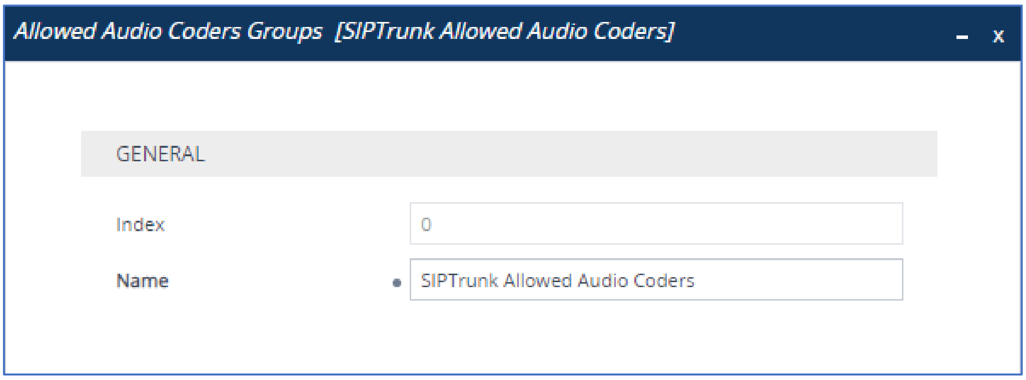
3. Click **Apply,** and then confirm the configuration change in the prompt that pops up.

The procedure below describes how to configure an Allowed Coders Group to ensure that voice sent to the Company SIP Trunk uses the dedicated coders list whenever possible. Note that this Allowed Coders Group ID will be assigned to the IP Profile belonging to the Company SIP Trunk in the next step.

To set a preferred coder for the Company SIP Trunk:

1. Open the Allowed Audio Coders Groups table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **Allowed Audio Coders Groups**).
2. Click **New** and configure a name for the Allowed Audio Coders Group for Company SIP Trunk.

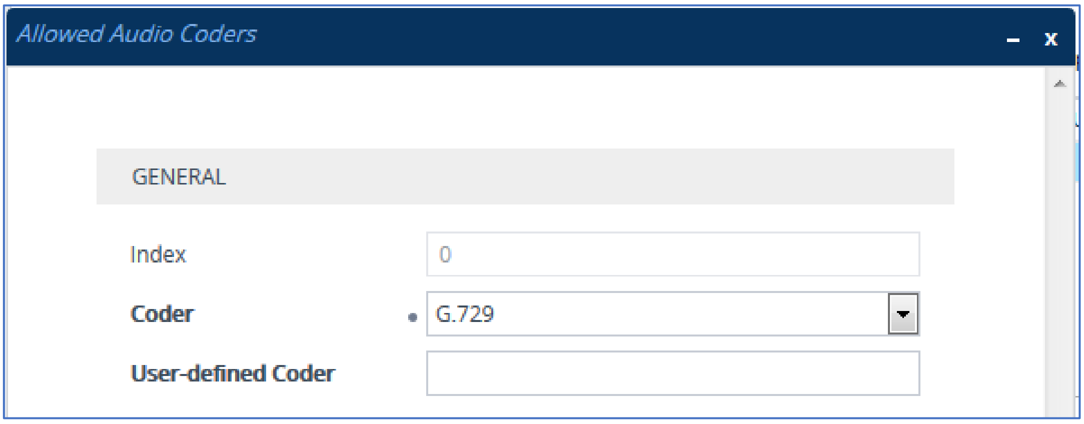
Figure 30: Configuring Allowed Coders Group for Company SIP Trunk



3. Click **Apply.**
4. Select the new row that you configured, and then click the **Allowed Audio Coders** link located below the table; the Allowed Audio Coders table opens.
5. Click **New** and configure an Allowed Coders as follows:

Parameter	Value
Index	**0**
Coder	**G.729**

Figure 31: Configuring Allowed Coders for Company SIP Trunk



6. Open the Media Settings page (**Setup** menu > **Signaling & Media** tab > **Media** folder > **Media Settings**).
7. From the '**Extended Coders Behavior**' drop-down list, select **Include Extensions**.
8. Click **Apply**.

4.8 Configuring IP Profiles

This section describes how to configure IP Profiles. The IP Profile defines a set of call capabilities relating to signaling (e.g., SIP message terminations such as REFER) and media (e.g., coder and transcoding method).

In this example topology, IP Profiles need to be configured for the following IP entities:

- Company SIP trunk – to operate in non-secure mode using RTP and SIP over UDP
- Teams Direct Routing – to operate in secure mode using SRTP and SIP over TLS
- ATA device – to operate in non-secure mode using RTP and SIP over UDP

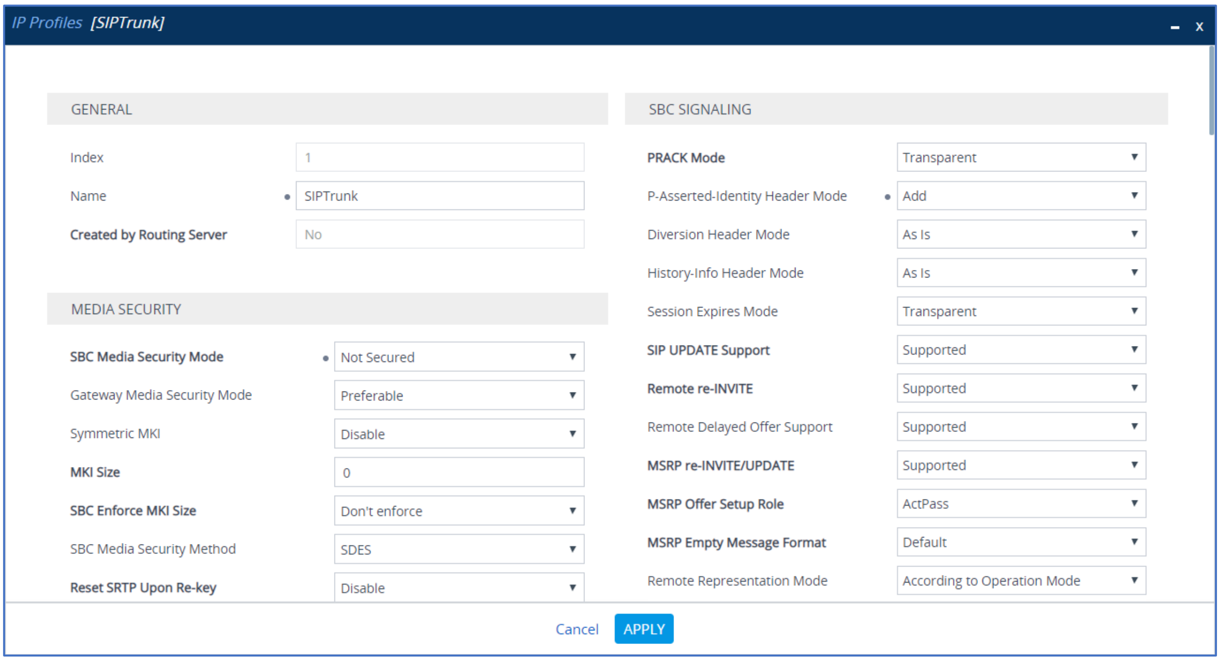
To configure an IP Profile for the Company SIP Trunk:

1. Open the IP Profiles table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **IP Profiles**).
2. Click **New**, and then configure the parameters as follows:

Parameter	Value
General	
Index	**1**
Name	**SIPTrunk**
Media Security	
SBC Media Security Mode	**Not Secured**
SBC Media	
Allowed Audio Coders	**SIPTrunk Allowed Coders**
Allowed Coders Mode	**Preference** (lists Allowed Coders first and then original coders in received SDP offer)
SBC Signaling	
P-Asserted-Identity Header Mode	**Add** (required for anonymous calls)
SBC Forward and Transfer	
Remote REFER Mode	**Handle Locally**
Remote Replaces Mode	**Handle Locally**
Play RBT To Transferee	**Yes** (required, as some SIP Trunks do not play ring-back tone during transfer)
Remote 3xx Mode	**Handle Locally**

Teams Hold music is not supported by Microsoft in consultative transfer of a PSTN call. The transferee will hear silence during the transfer. To overcome this issue, it is possible to configure SBC to play music during a consultative transfer. To do this, refer to Section 4.16.

Figure 32: Configuration example: Company SIP Trunk IP Profile



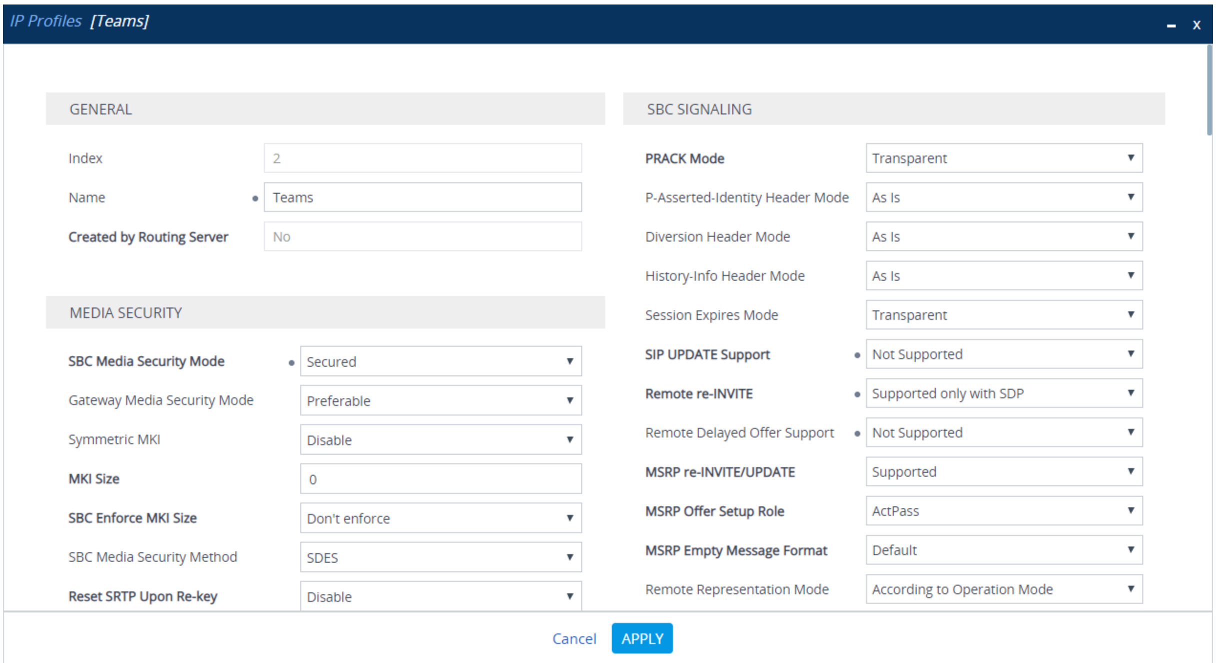
3. Click **Apply**.

To configure IP Profile for the Teams Direct Routing:

1. Open the IP Profiles table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **IP Profiles**).
2. Click **New**, and then configure the parameters as follows:

Parameter	Value
General	
Index	**2**
Name	**Teams** (arbitrary descriptive name)
Media Security	
SBC Media Security Mode	**Secured**
SBC Early Media	
Remote Early Media RTP Detection Mode	**By Media** (required, as Teams Direct Routing does not send RTP immediately to remote side when it sends a SIP 18x response)
SBC Media	
Extension Coders Group	**AudioCodersGroups_1**
RFC 2833 Mode	**Extend**
RTCP Mode	**Generate Always** (required, as some ITSPs do not send RTCP packets during while in Hold mode, but Microsoft expected to them)
ICE Mode	**Lite** (required only when Media Bypass enabled on Teams)
SBC Signaling	
SIP UPDATE Support	**Not Supported**
Remote re-INVITE Support	**Supported Only With SDP**
Remote Delayed Offer Support	**Not Supported**
SBC Forward and Transfer	
Remote REFER Mode	**Handle Locally**
Remote Replaces Mode	**Handle Locally**
Remote 3xx Mode	**Handle Locally**
SBC Hold	
Remote Hold Format	**Inactive** (some SIP Trunk may answer with a=inactive and IP=0.0.0.0 in response to the Re-Invite with Hold request from Teams. Microsoft Media Stack doesn't support this format. So, SBC will replace 0.0.0.0 with its IP address)
All other parameters can be left unchanged at their default values.	

Figure 33: Configuration example: Teams Direct Routing IP Profile



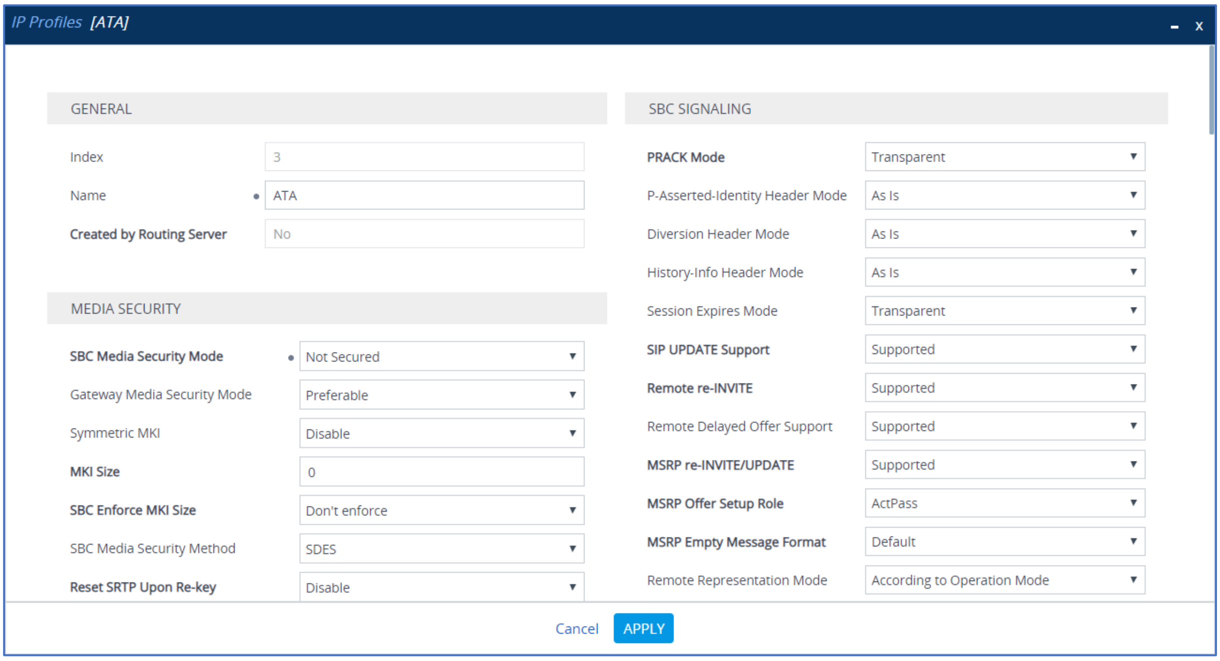
3. Click **Apply**.

To configure an IP Profile for the ATA device:

1. Open the IP Profiles table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **IP Profiles**).
2. Click **New**, and then configure the parameters as follows:

Parameter	Value
General	
Index	**3**
Name	**ATA**
Media Security	
SBC Media Security Mode	**Not Secured**
SBC Forward and Transfer	
Remote REFER Mode	**Handle Locally**
Remote Replaces Mode	**Handle Locally**
Remote 3xx Mode	**Handle Locally**

Figure 34: Configuration example: ATA device IP Profile



3. Click **Apply**.

4.9 Configuring IP Groups

This section describes how to configure IP Groups. The IP Group represents an IP entity on the network with which the SBC communicates. This can be a server (e.g., IP PBX or ITSP) or it can be a group of users (e.g., LAN IP phones). For servers, the IP Group is typically used to define the server's IP address by associating it with a Proxy Set. Once IP Groups are configured, they are used to configure IP-to-IP routing rules for denoting source and destination of the call.

In this example topology, IP Groups must be configured for the following IP entities:

- Company SIP Trunk located on WAN
- Teams Direct Routing located on WAN
- ATA device located on LAN

To configure IP Groups:

1. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
2. Configure an IP Group for the Company SIP Trunk:

Parameter	Value
Index	**1**
Name	**SIPTrunk**
Type	**Server**
Proxy Set	**SIPTrunk**
IP Profile	**SIPTrunk**
Media Realm	**SIPTrunk**
SIP Group Name	(according to ITSP requirement)
All other parameters can remain unchanged with their default values.	

3. Configure an IP Group for the Teams Direct Routing:

Parameter	Value
Index	**2**
Name	**Teams**
Topology Location	**Up**
Type	**Server**
Proxy Set	**Teams**
IP Profile	**Teams**
Media Realm	**Teams**
Classify By Proxy Set	**Disable**
Local Host Name	**< FQDN name of the SBC in the enterprise Teams tenant >** (For example, *sbc.ACeducation.info*)
Teams Direct Routing Mode	**Enable** (Enables the SBC to include Microsoft's proprietary X-MS-SBC header in outgoing SIP INVITE and OPTIONS messages in a Microsoft Teams Direct Routing environment. The header is used by Microsoft Teams to identify vendor equipment. The header's value is in the format 'Audiocodes/<model>/<firmware>').
Always Use Src Address	**Yes**
Outbound Message Manipulation Set	**1**
Proxy Keep-Alive using IP Group settings	**Enable**
All other parameters can be left unchanged with their default values.	

4. Configure an IP Group for the ATA device:

Parameter	Value
Index	**3**
Name	**ATA**
Topology Location	**Up**
Type	**User**
IP Profile	**ATA**
Media Realm	**MRLan**
SIP Group Name	(according to ITSP requirement)
All other parameters can remain unchanged with their default values.	

4.10 Configuring SRTP

This section describes how to configure media security. The Direct Routing Interface needs to use of SRTP only, so you need to configure the SBC to operate in the same manner.

To configure media security:

1. Open the Media Security page (**Setup menu > Signaling & Media tab > Media folder > Media Security**).
2. From the 'Media Security' drop-down list, select **Enable** to enable SRTP.
3. Click **Apply**.

4.11 Configuring Message Manipulation Rules

This section describes how to configure SIP message manipulation rules. SIP message manipulation rules can include insertion, removal, and/or modification of SIP headers. Manipulation rules are grouped into Manipulation Sets, enabling you to apply multiple rules to the same SIP message (IP entity).

Once you have configured the SIP message manipulation rules, you need to assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

> Implementation of the Message Manipulation rule with Microsoft Teams (shown below) is optional according to site deployment requirements.

To configure SIP message manipulation rule for Teams:

1. Open the Message Manipulations page (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Manipulations**).
2. Configure a new manipulation rule (Manipulation Set 2) for Teams IP Group. This rule applies to messages sent towards the Teams IP Group. This rule adds a routing policy rule toward Microsoft for handling different call forwarding scenarios (according to the action values shown below).

Parameter	Value
Index	**0**
Name	**Teams Routing Policy** (arbitrary name)
Manipulation Set ID	**1**
Condition	
Action Subject	**header.X-MS-RoutingPolicies**
Action Type	**Add**
Action Value	One of the following values: **"none", "no_missed_call", "disable_forwarding", "disable_forwarding_except_phone**

4.12 Configuring Message Condition Rules

This section describes how to configure the Message Condition Rules. A Message Condition defines special conditions (pre-requisites) for incoming SIP messages. These rules can be used as additional matching criteria for the IP-to-IP routing rules in the IP-to-IP Routing table.

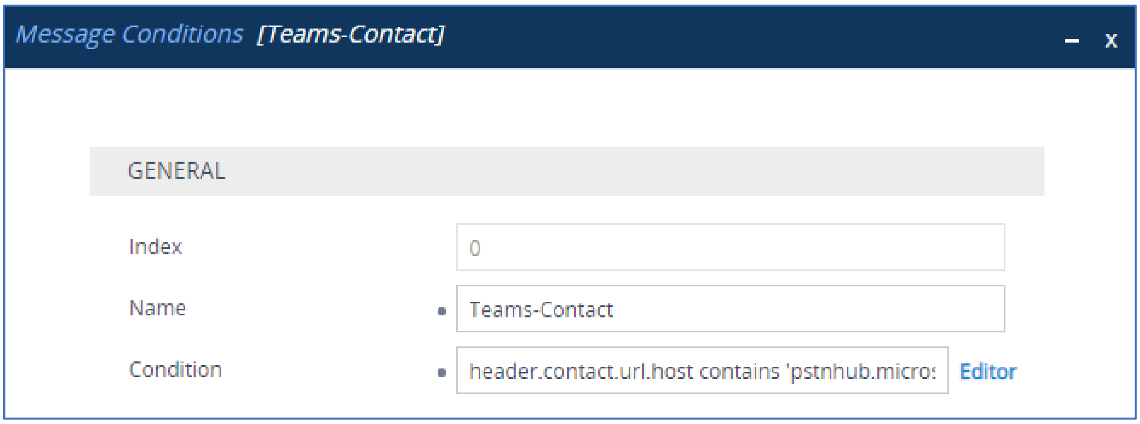
The following condition verifies that the Contact header contains Microsoft Teams FQDN.

To configure a Message Condition rule:

1. Open the Message Conditions table (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Conditions**).
2. Click **New**, and then configure the parameters as follows:

Parameter	Value
Index	**0**
Name	**Teams-Contact** (arbitrary descriptive name)
Condition	**header.contact.url.host contains 'pstnhub.microsoft.com'**

Figure 35: Configuring Condition Table



3. Click **Apply**.

4.13 Configuring Classification Rules

This section describes how to configure Classification rules. A Classification rule classifies incoming SIP dialog-initiating requests (e.g., INVITE messages) to a 'source' IP Group. The source IP Group is the SIP entity that sent the SIP dialog request. Once classified, the device uses the IP Group to process the call (manipulation and routing).

You can also use the Classification table for employing SIP-level access control for successfully classified calls, by configuring Classification rules with whitelist and blacklist settings. If a Classification rule is configured as a whitelist ("Allow"), the device accepts the SIP dialog and processes the call. If the Classification rule is configured as a blacklist ("Deny"), the device rejects the SIP dialog.

To configure a Classification rule:

1. Open the Classification table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Classification Table**).
2. Configure Classification rules as shown in the table below:

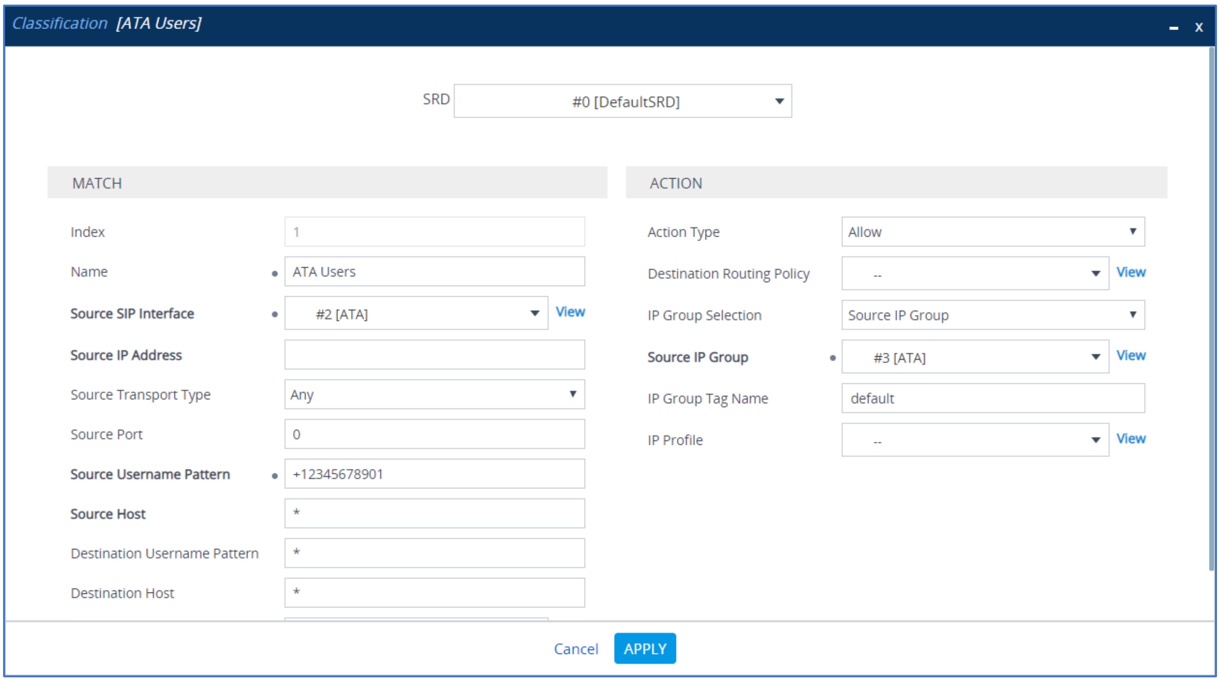
Table 10: Classification Rules

Index	Name	Source SIP Interface	Source IP Address	Destination Host	Message Condition	Action Type	Source IP Group
0	Teams_52_112 (arbitrary name)	Teams	52.112.*.*	sbc.ACeducation.info (example)	Teams-Contact	Allow	Teams
1	Teams_52_113 (arbitrary name)	Teams	52.113.*.*	sbc.ACeducation.info (example)	Teams-Contact	Allow	Teams
2	Teams_52_114 (arbitrary name)	Teams	52.114.*.*	sbc.ACeducation.info (example)	Teams-Contact	Allow	Teams
3	Teams_52_115 (arbitrary name)	Teams	52.115.*.*	sbc.ACeducation.info (example)	Teams-Contact	Allow	Teams
4	Teams_52_122 (arbitrary name)	Teams	52.122.*.*	sbc.ACeducation.info (example)	Teams-Contact	Allow	Teams
5	Teams_52_123 (arbitrary name)	Teams	52.123.*.*	sbc.ACeducation.info (example)	Teams-Contact	Allow	Teams

3. Click **Apply**.
4. Click **New**, and then configure classification rule for messages from ATA device as follows:

Parameter	Value
Index	**10**
Name	**ATA Users**
Source SIP Interface	**ATA**
Source Username Pattern	**+12345678901**
Action Type	**Allow**
Source IP Group	**ATA**

Figure 36: Configuring Classification Rule for ATA users



5. Click **Apply**.

4.14 Configuring IP-to-IP Call Routing Rules

This section describes how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The SBC selects the rule whose configured input characteristics (e.g., IP Group) match those of the incoming SIP message. If the input characteristics do not match the first rule in the table, they are compared to the second rule, and so on, until a matching rule is located. If no rule is matched, the message is rejected. The routing rules use the configured IP Groups (as configured in Section 4.9 on page 47) to denote the source and destination of the call.

For the example topology, the following IP-to-IP routing rules need to be configured to route calls between Teams Direct Routing and Company SIP Trunk:

- Terminate SIP OPTIONS messages on the SBC that are received from any entity
- REGISTER requests from ATA device
- Re-Route REFER messages to Teams Direct Routing
- Calls from Teams Direct Routing to Company SIP Trunk
- Calls from Company SIP Trunk to ATA device
- Calls from Company SIP Trunk to Teams Direct Routing
- Calls from ATA device to Teams Direct Routing
- Calls from ATA device to Company SIP Trunk

To configure IP-to-IP routing rules:

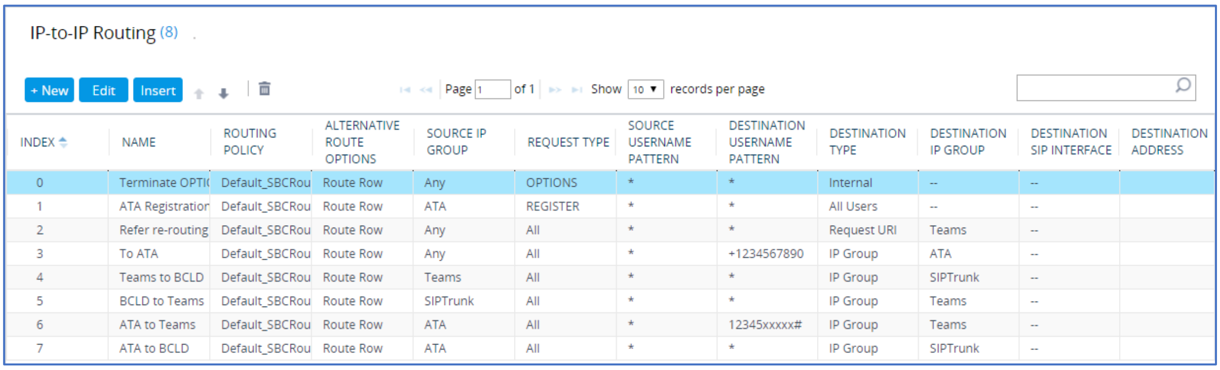
1. Open the IP-to-IP Routing table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Routing** > **IP-to-IP Routing**).
2. Configure routing rules as shown in the table below:

Table 11: Configuration Example: IP-to-IP Call Routing Rules

Index	Name	Source IP Group	Request Type	Dest Username Pattern	Call Triger	ReRoute IP Group	Dest Type	Dest IP Group	Internal Action
0	Terminate OPTIONS	Any	OPTIONS				Internal		Reply(Response='200')
1	ATA Registration	ATA	REGISTER				All Users		
2	Refer re-routing (arbitrary name)	Any			REFER	Teams	Request URI	Teams	
3	To ATA	Any		+1234567890			IP Group	ATA	
4	Teams to SIP Trunk	Teams					IP Group	SIPTrunk	
5	SIP Trunk to Teams	SIPTrunk					IP Group	Teams	
6	ATA to Teams	ATA		12345xxxxx#			IP Group	Teams	
7	ATA to SIP Trunk	ATA					IP Group	SIPTrunk	

The configured routing rules are shown in the figure below:

Figure 37: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table



IP-to-IP Routing (8)

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

INDEX	NAME	ROUTING POLICY	ALTERNATIVE ROUTE OPTIONS	SOURCE IP GROUP	REQUEST TYPE	SOURCE USERNAME PATTERN	DESTINATION USERNAME PATTERN	DESTINATION TYPE	DESTINATION IP GROUP	DESTINATION SIP INTERFACE	DESTINATION ADDRESS
0	Terminate OPTI(	Default_SBCRou	Route Row	Any	OPTIONS	*	*	Internal	--	--	
1	ATA Registratior	Default_SBCRou	Route Row	ATA	REGISTER	*	*	All Users	--	--	
2	Refer re-routing	Default_SBCRou	Route Row	Any	All	*	*	Request URI	Teams	--	
3	To ATA	Default_SBCRou	Route Row	Any	All	*	+1234567890	IP Group	ATA	--	
4	Teams to BCLD	Default_SBCRou	Route Row	Teams	All	*	*	IP Group	SIPTrunk	--	
5	BCLD to Teams	Default_SBCRou	Route Row	SIPTrunk	All	*	*	IP Group	Teams	--	
6	ATA to Teams	Default_SBCRou	Route Row	ATA	All	*	12345xxxxx#	IP Group	Teams	--	
7	ATA to BCLD	Default_SBCRou	Route Row	ATA	All	*	*	IP Group	SIPTrunk	--	

> The routing configuration may change according to your specific deployment topology.

4.15 Configuring Firewall Settings

As an extra security, there is an option to configure traffic filtering rules (*access list*) for incoming traffic on AudioCodes SBC. For each packet received on the configured network interface, the SBC searches the table from top to bottom until the first matching rule is found. The matched rule can permit (*allow*) or deny (*block*) the packet. Once a rule in the table is located, subsequent rules further down the table are ignored. If the end of the table is reached without a match, the packet is accepted. Please note that the firewall is stateless. The blocking rules will apply to all incoming packets, including UDP or TCP responses.

To configure a firewall rule:

1. Open the Firewall table (**Setup** menu > **IP Network** tab > **Security** folder> **Firewall**).
2. Configure the following Access list rules for Teams Direct Rout IP Interface:

Table 12: Firewall Table Rules

Index	Source IP	Subnet Prefix	Start Port	End Port	Protocol	Use Specific Interface	Interface ID	Allow Type
0	<Public DNS Server IP> (e.g., 8.8.8.8)	32	0	65535	Any	Enable	WAN_IF	Allow
1	52.112.0.0	14	0	65535	TCP	Enable	WAN_IF	Allow
2	52.122.0.0	15	0	65535	TCP	Enable	WAN_IF	Allow
3	xxx.xxx.xxx.xxx	32	0	65535	UDP	Enable	WAN_IF	Allow
49	0.0.0.0	0	0	65535	Any	Enable	WAN_IF	Block

> For information about prerequisites and planning your deployment, refer to Plan Direct Routing.
>
> Be aware that if in your configuration, connectivity to the SIP Trunk (or other entities) is performed through the same IP Interface as Teams (WAN_IF in our example), you <u>must</u> add rules to allow traffic from these entities. See an example in the row of index 3.

4.16 Configuring SBC To Play Music On Hold (Optional)

Teams Hold music is not supported by Microsoft in consultative transfer of a PSTN call. The transferee will hear silence during the transfer. To overcome this issue, it is possible to configure SBC to play music during a consultative transfer. To do this, a Prerecorded Tones (PRT) file needs to be prepared and loaded to the SBC. This section shows how to load a PRT file to the SBC. For a detailed procedure how to create a Prerecorded Tones (PRT) file, refer to appropriated AudioCodes' device *User Manual* document.

Update configuration of the SIP Trunk IP Profile:

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **IP Profiles**).
2. Choose SIP Trunk IP Profile, created in the Section 4.8 on the page 43. Configure the parameters using the table below as reference.

Table 13: Update Configuration of the SIP Trunk IP Profile

Parameter	Value
SBC Hold	
Remote Hold Format	**Send Only**
Reliable Held Tone Source	**No**
Play Held Tone	**Internal**

3. Click **Apply**, and then save your settings to flash memory.

To load PRT file to the device using the Web interface:

1. Open the Auxiliary Files page:
   - Toolbar: From the **Actions** drop-down menu, choose **Auxiliary Files**.
   - Navigation tree: **Setup** menu > **Administration** tab > **Maintenance** folder > **Auxiliary Files**.
2. Click the **Browse** button corresponding to the **Prerecorded Tones** file type that you want to load, navigate to the folder in which the file is located, and then click **Open**; the name and path of the file appear in the field next to the **Browse** button.
3. Click the **Load File** button corresponding to the file you want to load.
4. Save the loaded auxiliary files to flash memory.

5 Verifying the Pairing Between the SBC and Direct Routing

After you have paired the SBC with Teams Direct Routing using the *New-CsOnlinePSTNGateway* PowerShell command, validate that the SBC can successfully exchange OPTIONS with Direct Routing.

To validate the pairing using SIP OPTIONS:

1. Open the Proxy Set Status page (**Monitor** menu > **VoIP Status** tab> **Proxy Set Status**).
2. Find the Direct SIP connection and verify that 'Status' is online. If you see a failure, you need to troubleshoot the connection first, before configuring voice routing.

Figure 38: Proxy Set Status

Proxy Sets Status

This page refreshes every 60 seconds

PROXY SET ID	NAME	MODE	KEEP ALIVE	ADDRESS	PRIORITY	WEIGHT	SUCCESS COUNT	FAILURE COUNT	STATUS
0	ProxySet_0	Parking	Disabled						NOT RESOLVED
1	SIPTrunk	Parking	Enabled						ONLINE
				nn6300southsipconnect.adpt-tech.com(199.19.196.17:8933)(*)	1	50.00	4816	8	ONLINE
2	Teams	Load Balancing	Enabled						ONLINE
				sip.pstnhub.microsoft.com(52.114.75.24:5061)(*)	1	1.00	1	0	ONLINE
				sip2.pstnhub.microsoft.com(52.114.132.46:5061)(*)	2	1.00	1	0	ONLINE
				sip3.pstnhub.microsoft.com(52.114.14.70:5061)(*)	3	1.00	1	0	ONLINE

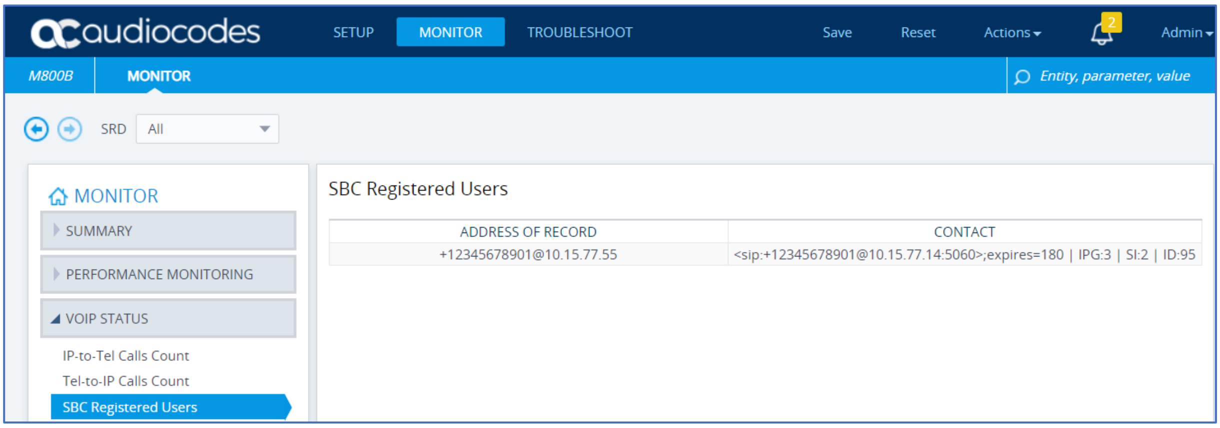
6 Verifying ATA Registered Users in the SBC

You can view SBC users that are registered with the device. For each user, the Address of Record (AOR) and the corresponding contacts are shown.

To view registered SBC users:

1. Open the SBC Registered Users page (Monitor menu > Monitor tab > VoIP Status folder > SBC Registered Users).

Figure 39: SBC Registered Users



A Configuring MP-1xx ATA for Connecting Analog Devices

This section describes how to configure AudioCodes MediaPack™ Series (MP-1xx) VoIP Gateways for connecting analog devices. The ATA device must be configured to send all calls to the AudioCodes SBC.

> This section shows partial configuration. For detailed configuration of the MediaPack MP-1xx Series refer to the device's *User's Manual* (https://www.audiocodes.com/library/technical-documents?query=MP-11x).

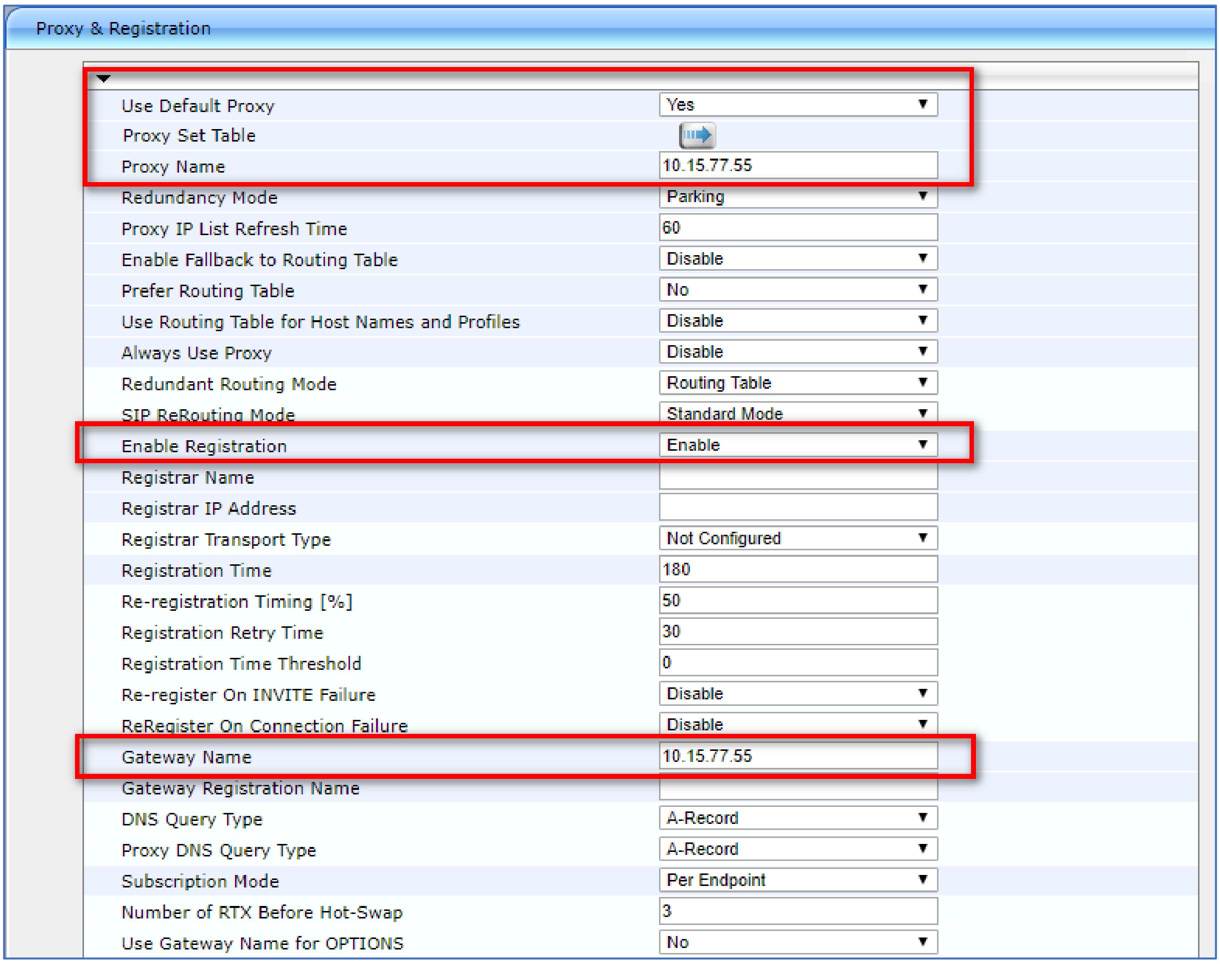
A.1 Configuring Proxy Server and Registration

This section describes how to configure the proxy server and registration. The configuration below uses the example of an ATA device registered to the SBC device (10.15.77.55).

To configure Proxy Server and Registration:

1. Open the Proxy & Registration page (**Configuration** tab > **VoIP** menu > **SIP Definitions** sub-menu > **Proxy & Registration**).

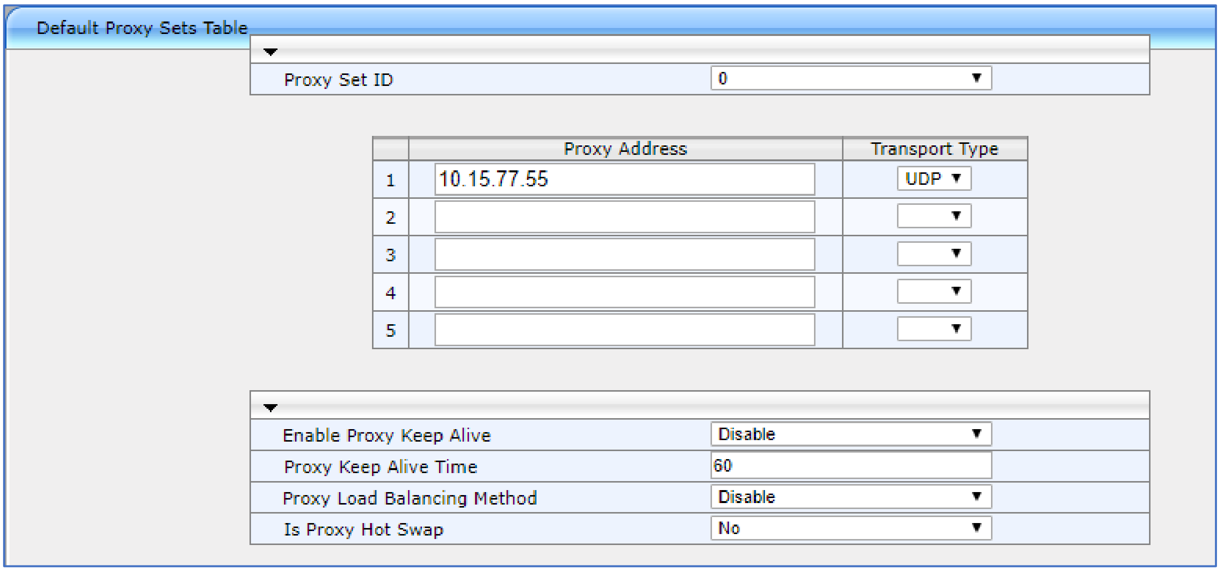
Figure 40: Proxy and Registration



2. From the 'Use Default Proxy' drop-down list, select **Yes**.

3. In the 'Proxy Name' field, enter the SBC IP address.
4. From the 'Enable Registration' drop-down list, select **Enable**.
5. In the 'Gateway Name' field, enter the SBC IP address.
6. Click the **Proxy Set Table** button, the following page is displayed:

Figure 41: Default Proxy Sets Table



7. In the 'Proxy Address' field, enter the SBC IP address.
8. Click the **Apply** button.

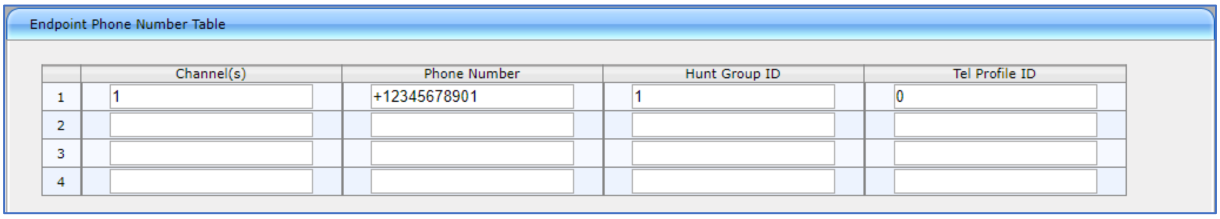
A.2 Configuring the Endpoint Phone Number Table

The 'Endpoint Phone Number Table' page allows you to activate the MP-1xx ports (endpoints) by defining telephone numbers. The configuration below uses the example of ATA destination phone number '+12345678901' with all routing directed to the SBC device (10.15.77.55).

To configure the Endpoint Phone Number table:

1. Open the Endpoint Phone Number Table page (**Configuration** tab > **VoIP** menu > **GW and IP to IP submenu** > **Hunt Group** sub-menu > **Endpoint Phone Number**).

Figure 42: Endpoint Phone Number Table Page



Endpoint Phone Number Table

	Channel(s)	Phone Number	Hunt Group ID	Tel Profile ID
1	1	+12345678901	1	0
2				
3				
4				

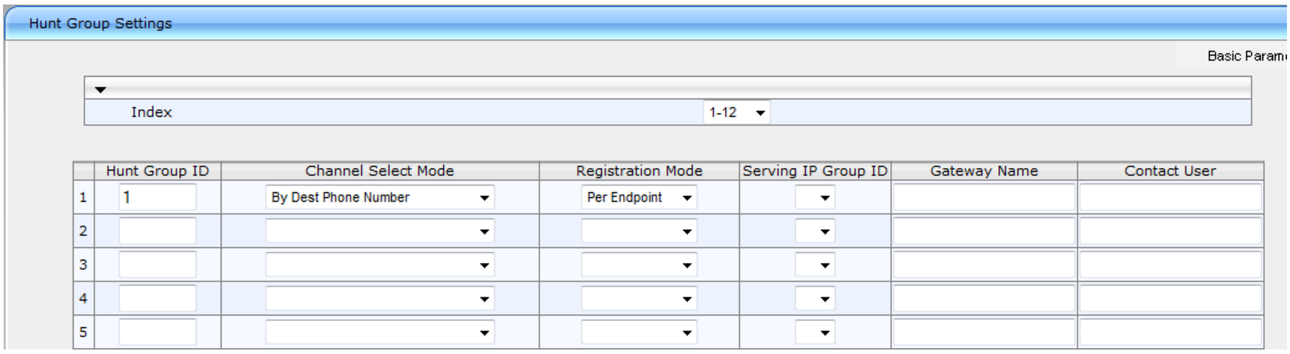
A.3 Configuring the Hunt Group

This section describes how to configure the Hunt Group.

To configure Hunt Group:

1. Open the Hunt Group Settings page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** sub-menu > **Hunt Group** > **Hunt Group Settings**).

Figure 43: Hunt Group Settings



Hunt Group Settings

Basic Param

Index 1-12

	Hunt Group ID	Channel Select Mode	Registration Mode	Serving IP Group ID	Gateway Name	Contact User
1	1	By Dest Phone Number	Per Endpoint			
2						
3						
4						
5						

2. From the 'Channel Select Mode' drop-down list, select **By Dest Phone Number**.
3. From the 'Registration Mode' drop-down list, select **Per Endpoint**.
4. Click the **Apply** button.

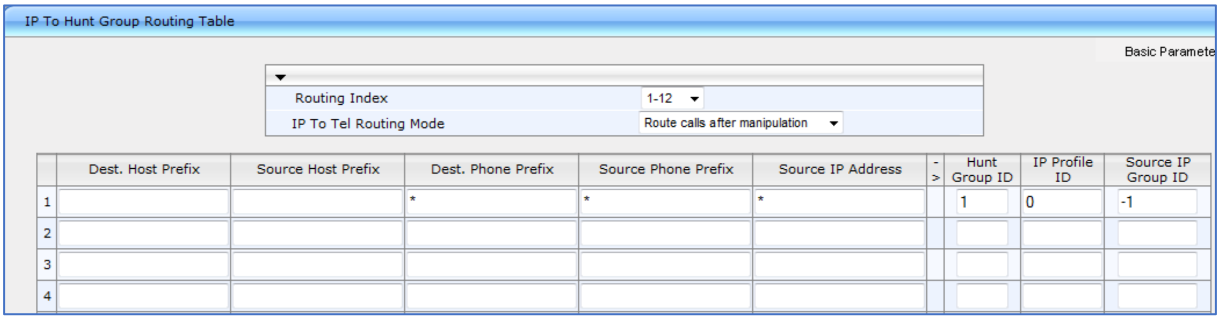
A.4 Configuring IP-to-Hunt Group Routing

This section describes how to configure the IP-to-Hunt Group routing rules.

To configure the IP to Hunt Group Routing table:

1. Open the Tel to IP Routing page (Configuration tab > VoIP menu > GW and IP to IP sub-menu > Routing > IP to Hunt Group Routing).

Figure 44: IP to Hunt Group Routing Page



2. Configure the entry as shown in the screen above.
3. Click the **Apply** button.

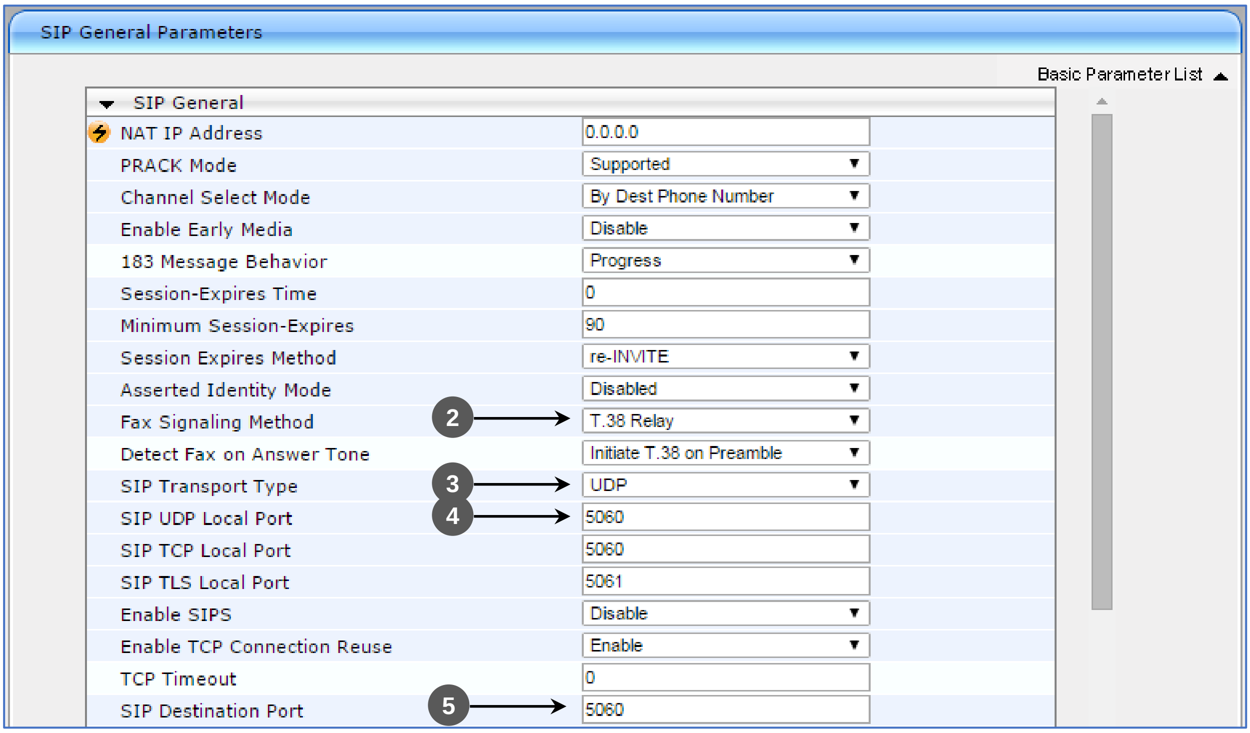
B Configuring SIP UDP Transport Type and Fax Signaling Method

In most cases ATA device is used for interconnection fax devices. This step describes how to configure the fax signaling method for the MP-1xx device.

To configure the fax signaling method:

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

Figure 45: SIP General Parameters Page



2. From the 'FAX Signaling Method' drop-down list, select **G.711 Transport** for G.711 fax support and select **T.38 Relay** for T.38 fax support.
3. From the 'SIP Transport Type' drop-down list, select **UDP**.
4. In the 'SIP UDP Local Port' field, enter **5060** (corresponding to the SBC configuration).
5. In the 'SIP Destination Port', enter **5060** (corresponding to the SBC configuration).

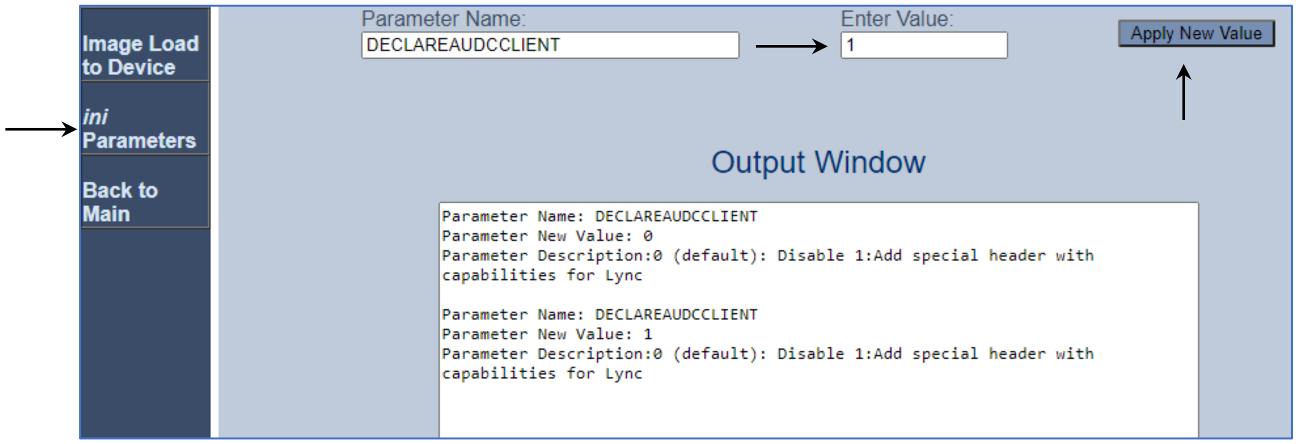
B.1 Configuring MP-1xx for LAD

This section describes what need to be configured to enable MP-1xx to work as Analog Device with Microsoft through AudioCodes' SBC.

To configure MP-1xx for LAD:

1. Open the Admin page: Append the case-sensitive suffix 'AdminPage' to the device's IP address in your Web browser's URL field (e.g., http://10.15.77.14/AdminPage).
2. In the left pane of the page that opens, click ***ini* Parameters**.

Figure 46: Configuring MP-1xx for LAD in AdminPage



3. Enter these values in the 'Parameter Name' and 'Enter Value' fields:

Parameter	Value
DECLAREAUDCCLIENT	**1** (adds a special header with capabilities for Lync)

4. Click the **Apply New Value** button for each field.

> This parameter classifies the MP-1xx's clients as LAD in the SBC.

C Configuring MP-20x ATA for Connecting Analog Devices

This section describes how to configure AudioCodes MediaPack™ Series (MP-20x) Telephony Adapter for connecting analog devices. The ATA device must be configured to send all calls to the AudioCodes SBC.

> This section shows partial configuration. For detailed configuration of the MediaPack MP-20x Series refer to the device's *User's Manual* *(*https://www.audiocodes.com/library/technical-documents?query=MP-20x*).*

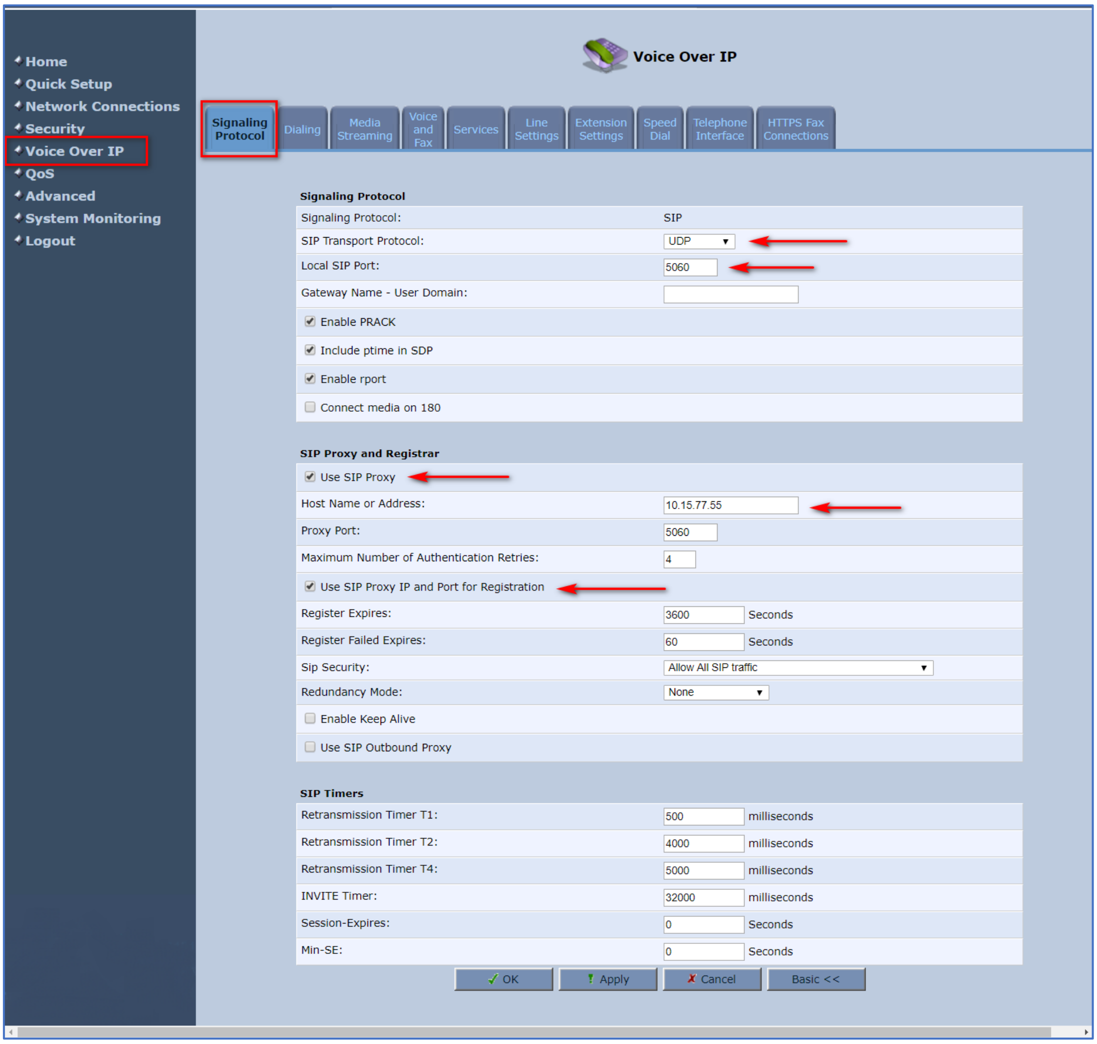
C.1 Configuring SIP Interface Settings

This section describes how to configure SIP Signaling Protocol.

To configure SIP Interface Settings:

1. Click the **Voice Over IP** menu in the side menu bar; the Voice Over IP screen appears.

Figure 47: Signaling Protocol Page



2. On the **Signaling Protocol** page, the following parameters enable configuration:
   a. From the 'SIP Transport Type' drop-down list, select **UDP**.
   b. In the 'Local SIP Port**'** field, enter **5060** (corresponding to the SBC configuration)
   c. In the 'Use SIP Proxy' check the check box.
   d. In the 'Host Name or Address' field, set the IP address of the SBC.
   e. In the 'Use SIP Proxy IP and Port for Registration' check the check box.

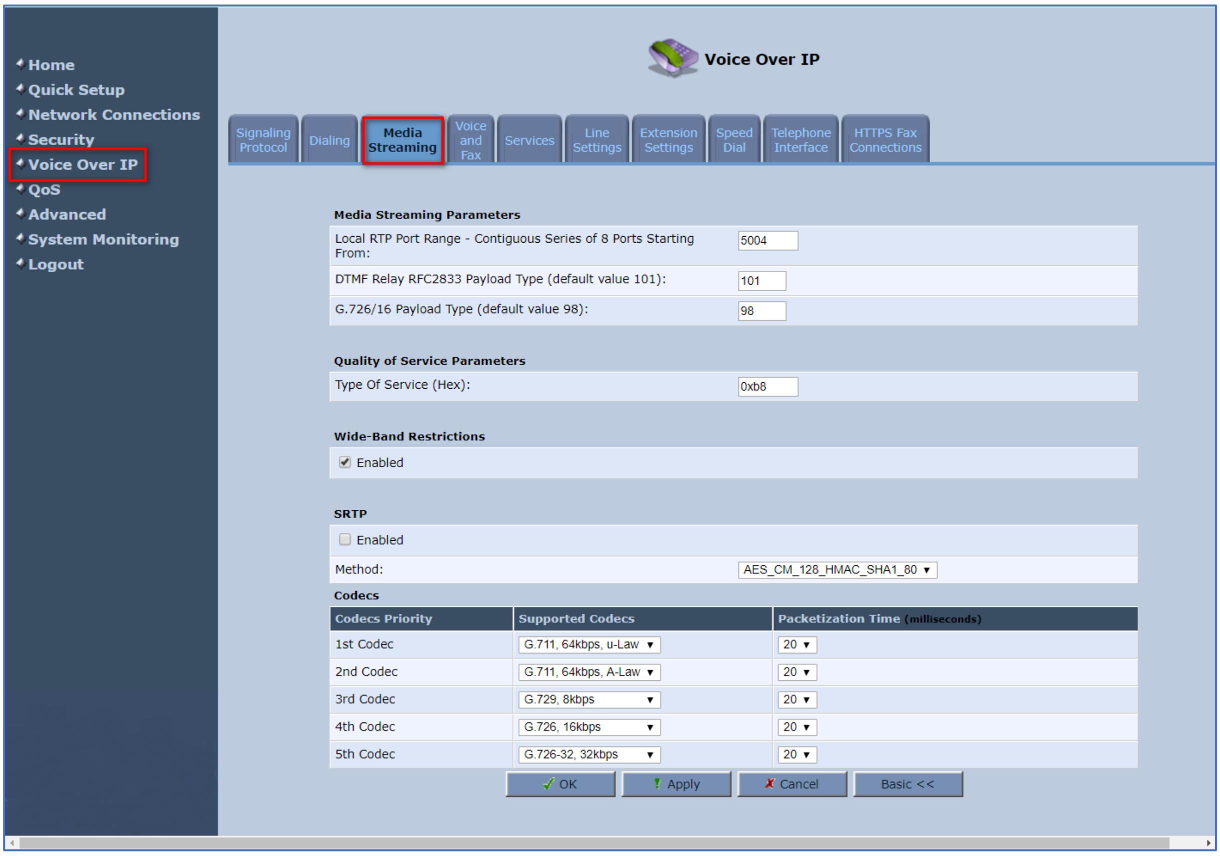
C.2 Configuring Media Streaming Parameters

The section describes how to configure Media Streaming Parameters.

To configure Media Streaming Parameters:

1. Click the **Media Streaming** tab. The Media Streaming screens opens, which enables you to configure the following:
   - Supported Codecs
   - Codecs Priority
   - Packetization Time

Figure 48: Media Streaming Page



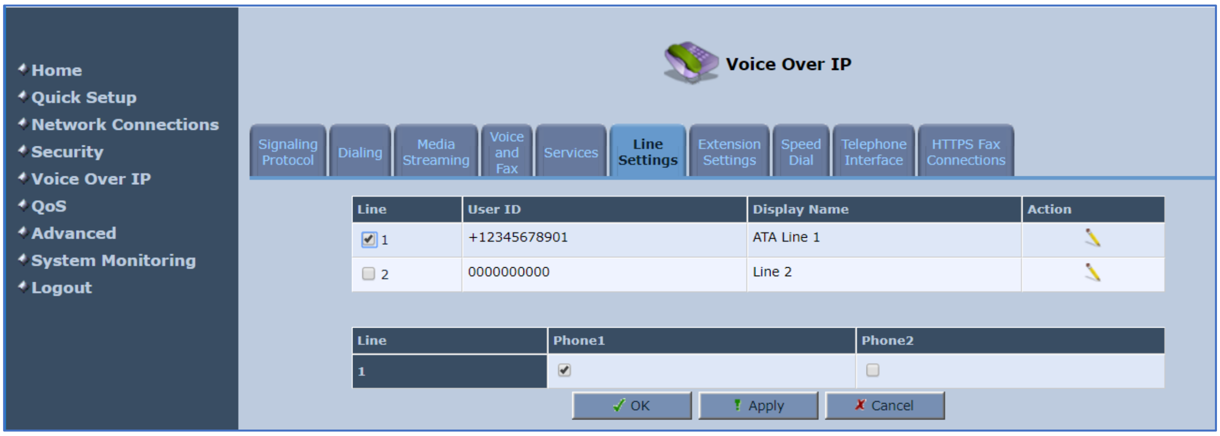
C.3 Configuring Line Settings

Before you can make phone calls, you need to configure lines. Lines are logical SIP ID numbers (i.e., telephone numbers) which are registered to a SIP proxy server and for which you are charged for calls you make on it. With a MP-20x line setting configuration, you can associate any phone extension to any line.

To configure lines:

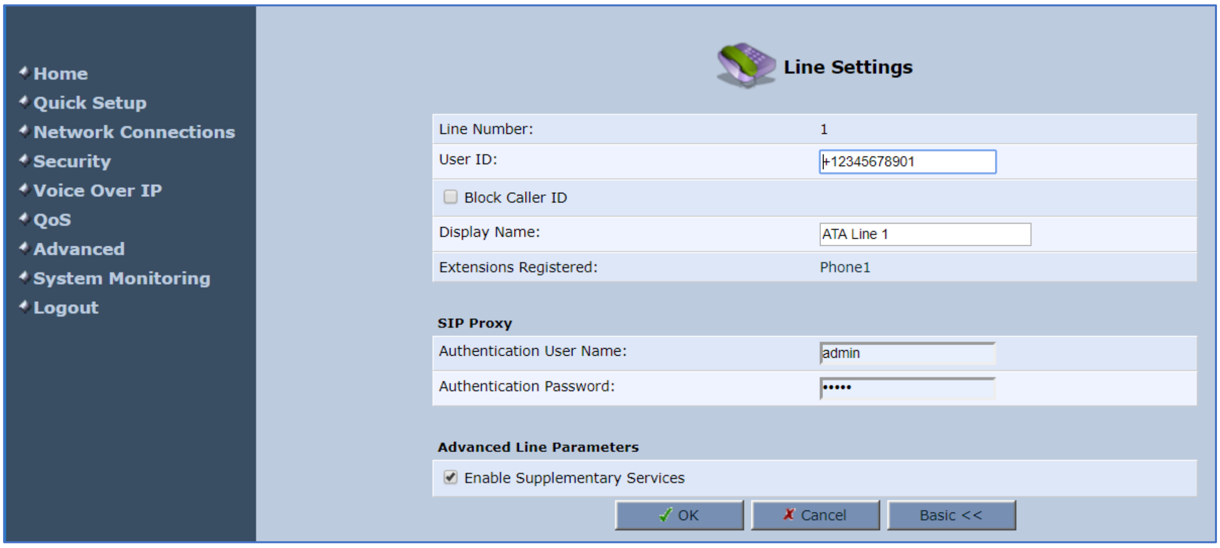
1. On the 'Voice Over IP' screen, click the **Line Settings** tab; the following screen appears.

Figure 49: Line Settings Tab Screen



2. For each line, click the corresponding **Edit** icon to configure the line; the following screen appears:

Figure 50: Line Settings Screen for a New Line



3. In the 'User ID' field, enter phone's VoIP user ID used for identification to initiate and accept calls.
4. To hide the phone's ID from the remote party, select the 'Block Caller ID' check box.
5. In the 'Display Name' field, enter a name to intuitively identify the line. This is also displayed to remote parties as your caller ID.
6. Click **OK** to save your settings.

D Syntax Requirements for SIP Messages 'INVITE' and 'OPTIONS'

The syntax of SIP messages must conform with Teams Direct Routing requirements.

This section covers the high-level requirements for the SIP syntax used in 'INVITE' and 'OPTIONS' messages. You can use the information presented here as a first step when troubleshooting unsuccessful calls. AudioCodes has found that most errors are related to incorrect syntax in SIP messages.

D.1 Terminology

Must	Strictly required. The deployment does not function correctly without the correct configuration of these parameters.

D.2 Syntax Requirements for 'INVITE' Messages

Figure 51: Example of an 'INVITE' Message

```
INVITE sip:+97249888108@10.15.40.55;user=phone SIP/2.0
Via: SIP/2.0/TLS sbc.ACeducation.info:5068;alias;branch=z9hG4bKac496289557
Max-Forwards: 69
From: <sip:+97239762000@10.15.77.12>;tag=1c1642854452
To: <sip:+97249888108@10.15.40.55;user=phone>
Call-ID: 1167963076285201992217@ACeducation.info
CSeq: 1 INVITE
Contact: <sip:+97239762000@sbc.ACeducation.info:5068;transport=tls>
Supported: em,100rel,timer,replaces,path,resource-priority,sdp-anat
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE
User-Agent: 10.15.40.55/v.7.20A.250.273
Content-Type: application/sdp
Content-Length: 1114
```

- **Contact** header
  - **MUST:** When placing calls to the Direct Routing interface, the 'CONTACT' header must have the SBC FQDN in the URI hostname
  - **Syntax:** *Contact: <phone number>@<FQDN of the SBC>:<SBC Port>;<transport type>*
  - If the parameter is not configured correctly, calls are rejected with a '403 Forbidden' message.

D.3 Requirements for 'OPTIONS' Messages Syntax

Figure 52: Example of 'OPTIONS' message

```
OPTIONS sip:195.189.192.171 SIP/2.0
Via: SIP/2.0/TLS sbc.ACeducation.info:5068;alias;branch=z9hG4bKac1385438539
Max-Forwards: 70
From: <sip:195.189.192.171>;tag=1c1890841146
To: <sip:195.189.192.171>
Call-ID: 595855232295201931O3@ACeducation.info
CSeq: 1 OPTIONS
Contact: <sip:sbc.ACeducation.info:5068;transport=tls>
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE
User-Agent: 10.15.40.55/v.7.20A.250.273
Accept: application/sdp, application/simple-message-summary, message/sipfrag
Content-Length: 0
```

- **Contact** header
  - **MUST:** When placing calls to the Direct Routing interface, the 'CONTACT' header must have the SBC FQDN in the URI hostname
  - **Syntax:** *Contact: <phone number>@<FQDN of the SBC>:<SBC Port>;<transport type>*
  - If the parameter is not configured correctly, the calls are rejected with a '403 Forbidden' message

The table below navigates to the path in the Web interface where the parameters are configured and refers to the relevant location in this document including the configuration instructions.

Table 14: Syntax Requirements for an 'OPTIONS' Message

Parameter	Where Configured	How to Configure
Contact	**Setup** > **Signaling and Media** > **Core Entities** > **IP Groups** > <Group Name> > **Local Host Name** In IP Groups, 'Contact' must be configured. In this field ('Local Host Name'), define the local host name of the SBC as a string, for example, *sbc.ACeducation.info*. The name changes the host name in the call received from the IP group.	See Section 4.9 Configure IP Groups

D.4 Connectivity Interface Characteristics

The table below shows the technical characteristics of the Direct Routing interface.

In most cases, Microsoft uses RFC standards as a guide during development, but does not guarantee interoperability with SBCs - even if they support all the parameters in the table below - due to the specifics of the implementation of the standards by SBC vendors.

Microsoft has a partnership with some SBC vendors and guarantees their devices' interoperability with the interface. All validated devices are listed on Microsoft's website. Microsoft only supports devices *that are validated* in order to connect to the Direct Routing interface.

AudioCodes is one of the vendors who are in partnership with Microsoft.

AudioCodes' SBCs are validated by Microsoft to connect to the Direct Routing interface.

Table 15: Teams Direct Routing Interface - Technical Characteristics

Category	Parameter	Value	Comments
Ports and IP ranges	SIP Interface FQDN Name	See Microsoft's document *Deploying Direct Routing Guide.*	-
	IP Addresses range for SIP interfaces	See Microsoft's document *Deploying Direct Routing Guide.*	-
	SIP Port	5061	-
	IP Address range for Media	See Microsoft's document *Deploying Direct Routing Guide.*	-
	Media port range on Media Processors	See Microsoft's document *Deploying Direct Routing Guide.*	-
	Media Port range on the client	See Microsoft's document *Deploying Direct Routing Guide.*	-
Transport and Security	SIP transport	TLS	-
	Media Transport	SRTP	-
	SRTP Security Context	DTLS, SIPS **Note:** Support for DTLS is pending. Currently, SIPS must be configured. When support for DTLS will be announced, it will be the recommended context.	https://tools.ietf.org/html/rfc5763
	Crypto Suite	AES_CM_128_HMAC_SHA1_80, non-MKI	-
	Control protocol for media transport	SRTCP (SRTCP-Mux recommended)	Using RTCP MUX helps reduce the number of required ports
	Supported Certification Authorities	See the *Deployment Guide*	-
	Transport for Media Bypass (of configured)	■ ICE-lite (RFC5245) – recommended ■ Client also has Transport Relays	-
	Audio codecs	■ G711 ■ Silk (Teams clients) ■ Opus (WebRTC clients) - only if Media Bypass is used ■ G729	-
Codecs	Other codecs	■ CN ■ Required narrowband and wideband ■ RED - Not required ■ DTMF - Required ■ Events 0-16 ■ Silence Suppression - Not required	-

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