

MiCollab Advanced Messaging Microsoft Teams Direct Routing using AudioCodes Mediant SBC SIP Trunk Integration Technical Notes

For version 9.2 and above

Notice 1

Notice

The information contained in this document is believed to be accurate in all respects but is not warranted by Mitel Networks™ Corporation (MITEL®). Mitel makes no warranty of any kind with regards to this material, including, but not limited to, the implied warranties of merchantability and fitness for a particular purpose. The information is subject to change without notice and should not be construed in any way as a commitment by Mitel or any of its affiliates or subsidiaries. Mitel and its affiliates and subsidiaries assume no responsibility for any errors or omissions in this document. Revisions of this document or new editions of it may be issued to incorporate such changes.

No part of this document can be reproduced or transmitted in any form or by any means - electronic or mechanical - for any purpose without written permission from Mitel Networks Corporation.

Trademarks

The trademarks, service marks, logos and graphics (collectively "Trademarks") appearing on Mitel's Internet sites or in its publications are registered and unregistered trademarks of Mitel Networks Corporation (MNC) or its subsidiaries (collectively "Mitel") or others. Use of the Trademarks is prohibited without the express consent from Mitel. Please contact our legal department at legal@mitel.com for additional information. For a list of the worldwide Mitel Networks Corporation registered trademarks, please refer to the website: <http://www.mitel.com/trademarks>.

© Copyright 2021, Mitel Networks Corporation

All rights reserved

Contents

| | |
|---|-----------|
| Preface | 5 |
| About This Guide | 5 |
| References | 5 |
| Documentation | 6 |
| Documentation Updates | 6 |
| Help | 6 |
| Document Conventions | 7 |
| Features Supported by This Integration | 8 |
| Critical Application Considerations | 10 |
| Installation Requirements | 11 |
| Microsoft Teams Direct Routing Requirements | 11 |
| AudioCodes Mediant SBC Requirements | 11 |
| MiCollab AM Requirements | 11 |
| Programming Microsoft Teams Direct Routing | 12 |
| Add and Configure New SBC to Direct Routing | 12 |
| Add Voice Route and PSTN Usage | 13 |
| Add Voice Routing Policy | 13 |
| Enable Online User for Calling | 13 |
| Assign Online User to the Voice Routing Policy | 13 |
| Setting Up Call Forwarding Options in the Microsoft Teams Client | 14 |
| Configuring AudioCodes SBC | 16 |
| Configure Media Realms | 16 |
| Configure SIP Signaling Interfaces | 17 |
| Configure Proxy Sets | 17 |
| Configure Proxy Address | 18 |
| Configure IP Profiles | 18 |
| Configure IP Groups | 20 |
| Configure IP-to-IP Call Routing Rules | 21 |
| Configure Number Manipulation Rules | 22 |
| Configure Message Manipulation Rules | 22 |
| Configuring MiCollab AM | 24 |
| Configuring MiCollab AM for the Integration During Initial Installation | 24 |
| Configuring Existing MiCollab AM for the Integration | 28 |
| Configuring the Extension Device for Subscribers | 32 |

| | |
|---|-----------|
| Changing the Network Binding Order on the MiCollab AM Platform | 34 |
| Configuring Quality of Service (QoS) | 36 |

Preface

About This Guide

This Integration Technical Note (ITN) is written for administrators who are experienced with MiCollab Advanced Messaging (MiCollab AM) and who are familiar with its procedures and terminology. This ITN also assumes that you are familiar with the features and functionality of Microsoft Teams and Session Initiation Protocol (SIP).

This document describes how to integrate MiCollab AM with Microsoft Teams, using the Session Initiation Protocol (SIP) integration via AudioCodes Mediant SBC.

MiCollab AM uses SIP trunks to integrate with Microsoft Teams via AudioCodes SBC. This integration operates exclusively over an IP-based network; the integration uses no analog or digital voice telephony ports, but instead passes voice communication and signaling information directly over the network. The Call Server provides the hunting to an available line on the Call Server.

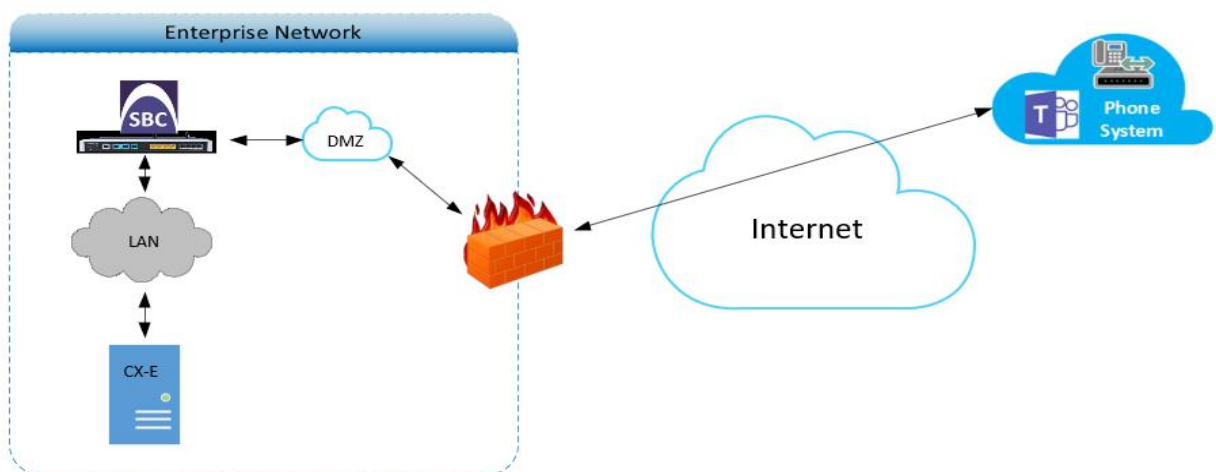


Figure 1: Interoperability Test Topology between SBC and Microsoft Teams Direct Routing with MiCollab AM

This document outlines critical application considerations you should examine before you begin work on the integration, as well as installation and programming procedures that are necessary to integrate MiCollab AM with Microsoft Teams.

Use this document in conjunction with *System Installation Guide* and *System Administration Guide* and the *MiCollab AM help*. For specific information about Microsoft Teams, refer to the Microsoft documentation.

References

A catalog of technical documentation is included on the MiCollab AM Installation Media. If you are installing any advanced applications, such as Networking and Fax Server applications, you should refer to the appropriate technical documentation for application and installation information.

Documentation

The technical documentation is produced in the PDF format and requires the PDF reader to view it. The documentation set for this MiCollab AM includes the following documents and resources:

- **Developer Resources.** Contains programming guides and API references for developers for integrating the server clients and web applications with MiCollab AM.
- **Integration Technical Notes (ITN).** Contains a set of guides that describe the integration methods and instructions for a variety of phone systems to work with MiCollab AM. The ITNs are generally used by resellers or administrators who are experienced with MiCollab AM and familiar with the integration procedures and terminology.
- **Quick Reference Card (QRC).** Contains shortcuts and quick instructions telling subscribers how to access and use the messaging system.
- **Server Documentation.** Available as a PDF only. Contains administrative guides for administrators about installing, configuring, and administering the messaging system, and user guides for subscribers about accessing the messaging system and checking and sending messages.
- **Spare Parts Documentation.** Contains a set of guides that describe the instructions for installing and configuring hardware parts to work with MiCollab AM. These documents are written for Mitel certified MiCollab AM technicians who are experienced with MiCollab AM and familiar with the procedures and terminology.
- **Software Release Notice (SRN).** This notice introduces the new features, capabilities, and hardware/software requirements for the corresponding MiCollab AM version.

Documentation Updates

Documentation updates may be available from the following sources:

- Mitel certified technicians can view or download the latest/updated documents and program files from our partner web site: connect.mitel.com/connect

Help

The primary source of information about MiCollab AM is the online help available within any of its administrative utilities. You can access **Help** as follows:

- Click the **Help** button in the dialog box or window in which you are working
- Press the **F1** key at any time.

Document Conventions

The following conventions are used in this document:

- **Key Names.** Names of keys on the keyboard are shown in a box.

Example: **Enter**

When two keys must be pressed simultaneously, they are joined by a + sign.

Example: **Alt** + **Tab**

- **Reference to Document.** *Italics* fonts can also signify the titles of other documents.

Example: Refer to *System Installation Guide*.

- **UI Element Names.** Names of UI elements such as dialog windows, screens, menu items, tabs, buttons, icons, etc. are shown in bold.

Example: On the **Startup** screen, click the **Start** icon.

- **User Input.** Information required to be typed or spoken is shown in italics.

Example: Type the password *voicemail*.

- **Warning, Caution, Important, and Notes.** Text for the contents that require attention are shown as follows:

WARNING A warning paragraph advises you of circumstances that can result in the loss of data, harm to the system server platform, or personal harm.

CAUTION Failure to follow these recommendations can result in unauthorized access to the system and consequent loss of data.

IMPORTANT An important paragraph gives decision-making information or informs you of the order in which tasks need to be completed.

NOTE A note gives additional information, provides an explanation, or indicates an exception to the information in the preceding text.

Features Supported by This Integration

The following tables list the features supported with the Microsoft Teams integration.

Table 1. Call forward to personal greeting support for common call types

| Divert to MiCollab AM on | Supported | Notes |
|--------------------------|-----------|--------|
| No Answer | Yes | |
| Busy | No | Note 1 |
| Forward All | Yes | |
| Do Not Disturb | No | Note 1 |

NOTE

1. Microsoft Teams does not provide call forwarding reason.

Table 2. Integration features supported for Microsoft Teams

| Feature | Supported | Notes |
|--|-----------|--------|
| Automatic subscriber login | Yes | |
| ANI/CLI | Yes | |
| Announce Busy greeting on forward busy calls | No | Note 1 |
| Call screening | Yes | |
| Caller queuing | No | |
| DNIS | Yes | |
| End-to-end DTMF, attendant console | Yes | |
| End-to-end DTMF, proprietary telephones | Yes | |
| Fax Detection | Yes | |
| Internal calling party ID for reply | Yes | |
| Live record, integrated | No | |
| Live reply to sender | Yes | |

| Feature | Supported | Notes |
|--|-----------|--------|
| Message notification callouts | Yes | |
| MWI, set/clear | No | |
| MWI, inband/outband | No | |
| Networking, analog | No | |
| Overflow from MiCollab AM to attendant | N/A | |
| Overflow to MiCollab AM from attendant | N/A | |
| PBX-provided disconnect signaling | N/A | |
| SRTP | Yes | Note 3 |
| TLS | Yes | Note 3 |
| Transfers, blind | Yes | |
| Transfers, confirmed | Yes | |
| Transfers, supervised | Yes | |
| Transfers, monitored | Yes | |
| Trunk ID for call routing | No | |
| Multiple integrations | Yes | Note 2 |

NOTES

1. Microsoft Teams does not provide call forwarding reason.
2. See [Critical Application Considerations](#).
3. MiCollab AM supports negotiation for SRTP media streams using the Secure RTP profile defined in RFC 3711 with the offer/answer model defined in RFC 3264. To enable SRTP, RTP, or both, see integration configuration options documentation for the switch. The default setting is RTP. Please note that MiCollab AM doesn't support RFC 5939 which is an extension of RFC 3264.

To create secure connections, use TLS 1.3 (recommended where available) or 1.2 for the System Server and Call Servers.

Critical Application Considerations

The following section provides known limitations or conditions within Microsoft Teams and MiCollab AM that affect performance, and provides general recommendations to avoid these limitations.

- 911 and E.911 are not supported via the integration.
- Microsoft Teams does not provide call forwarding reason. As a result, **Call Forward Busy** and **Call Forward DND** are not supported.
- **Message Waiting Indicator (MWI)** is not supported with the Microsoft Teams Direct Routing integration.
- Each Microsoft Teams user that is supported by MiCollab AM must have a Subscriber mailbox on MiCollab AM.

In addition, each subscriber must have an extension device assigned to the integration's switch section. This number should be the same as the number assigned to the **Line URI** in Microsoft Teams for the user without the '+' sign.

IMPORTANT Please be aware that MiCollab AM does not add the '+' sign when dialing numbers.

- Subscribers must have the appropriate callout permission on MiCollab AM to enable them to place calls to numbers that are not associated with other subscribers via their devices configuration.
- It must be the first network connection in the network binding order. If your MiCollab AM server platform is a component of two or more local or wide area networks (LANs or WANs), you must make sure that this integration does not interfere with the normal network operation of the server.

For more information, refer to the [Changing the Network Binding Order on the MiCollab AM Platform](#) section later in this document.

- MiCollab AM 9.2 supports up to 10 integration types (i.e. licensed integrations) in total per system. However, the following limitations apply to each Call Server:
 - Limited to 3 integration types per Call Server
 - The 3 integration types can be any mix of TDM and SIP (e.g. 1 TDM and 2 SIP)
 - Limited to 1 Mitel MiTAI or 1 Cisco UCM SCCP IP integration. Can be mixed with TDM, but not with SIP
 - Connect up to 10 telephone systems total per Call Server (e.g. 2 Genband SIP Trunk systems using SIP + 5 Avaya IP Office systems using SIP + 3 Siemens HiPath 4000 systems using Station Set Emulation)
 - SIP timers for Aastra EETS integrations are incompatible with other SIP integrations. Thus, it is not possible to have an EETS integration with any other SIP integration on the Call Server

Installation Requirements

Before performing any of the procedures in this document, review the information in the following sections. To install this successfully, you must meet the installation requirements for Microsoft Teams Direct Routing, AudioCodes Mediant SBC and MiCollab AM.

Microsoft Teams Direct Routing Requirements

- E3 license with Microsoft Phone System add-on license or E5 license assigned to users of Direct Routing in Office 365.
- Valid domain name registered for Enterprise Office 365 tenant

NOTE You cannot use the *.onmicrosoft.com tenant for the domain name.

AudioCodes Mediant SBC Requirements

- Certified AudioCodes Mediant SBC running version 7.20A.250 or above.
- SW/TEAMS (Microsoft Teams license) installed on SBC. This license enables SBC for Microsoft Teams Direct Routing and enables audio codecs SILK and OPUS.

NOTE Refer to the following link <https://docs.microsoft.com/en-us/microsoftteams/direct-routing-border-controllers> for list of certified AudioCodes Mediant SBC for Microsoft Teams Direct Routing.

MiCollab AM Requirements

- MiCollab AM software version 21.2 or later
- At least one 100 MB or 1000 MB network interface card and cable
- MiCollab AM software key diskette or feature file with the AudioCodes SIP Gateway integration enabled and one RADVISION® SIP and RTP license enabled for each port.

Programming Microsoft Teams Direct Routing

Follow the recommendations and programming examples in this section to configure Microsoft Teams Direct Routing with AudioCodes SBC. Programming examples show commands and parameters that are necessary for integration. They do not represent programming in its entirety. Refer to the following link <https://www.audiocodes.com/media/13253/connecting-audiocodes-sbc-to-microsoft-teams-direct-routing-enterprise-model-configuration-note.pdf> for details on how to configure Direct Routing between Microsoft Teams and AudioCodes SBC.

The steps you must take to configure Microsoft Teams Direct Routing with AudioCodes SBC are as follows:

- Add and configure new SBC to Direct Routing
- Add Voice Route and PSTN Usage
- Add Voice Routing Policy
- Enable Online User(s) for calling
- Assign Online User to the Voice Routing Policy

Add and Configure New SBC to Direct Routing

To add and configure new SBC to Direct Routing:

- 1 Connect to Microsoft Teams PowerShell using your Tenant Administrator credentials.
- 2 Run the following PowerShell command for creating a new PSTN Gateway:

```
New-CsOnlinePSTNGateway -Identity sbc.example.com -SipSignallingPort 5061 -  
ForwardCallHistory $True -ForwardPai $True -MediaBypass $True -Enabled $True
```

NOTE The SBC domain name must be from one of the names registered in 'Domains' of the Enterprise Office 365 tenant. You cannot use the ***.onmicrosoft.com** tenant for the domain name. For example, if your organization owns example.com then **sbc.example.com** is a good name for the SBC.

Add Voice Route and PSTN Usage

To add Voice Route and PSTN Usage

- 1 Create PSTN Usage:

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

- 2 Create new Online Voice Route and associate it with PSTN Usage:

```
New-CsOnlineVoiceRoute -Identity "audc-interop" -NumberPattern "^\\+" -  
OnlinePstnGatewayList sbc.example.com -Priority 1 -OnlinePstnUsages "Interop"
```

Add Voice Routing Policy

To add Voice Routing Policy

- 1 Connect to Microsoft Teams PowerShell using your Tenant Administrator credentials.
- 2 Run the following PowerShell command for creating a new PSTN Gateway:

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

Enable Online User for Calling

To Enable Online User for Calling

- 1 Use following PowerShell command for enabling online user:

```
Set-CsUser -Identity user1@example.com -EnterpriseVoiceEnabled $true -HostedVoiceMail  
$false -OnPremLineURI tel:+2001
```

Assign Online User to the Voice Routing Policy

To Assign Online User to the Voice Routing Policy

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

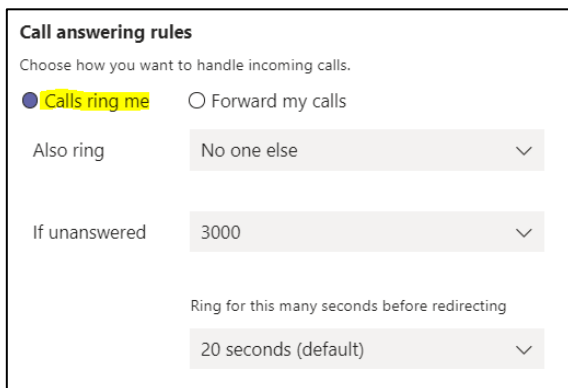
```
Grant-CsOnlineVoiceRoutingPolicy -PolicyName "audc-interop" -Identity  
user1@example.com
```

Setting Up Call Forwarding Options in the Microsoft Teams Client

Follow the steps below to setup the call forwarding options in the Microsoft Teams client to send unanswered calls to MiCollab AM

To setup Call Forward Ring-No-Answer:

- 1 At the top of the Microsoft Teams client app, click **Settings and more** **⋮** adjacent to your profile picture.
- 2 Select **Settings** > **Calls** and go to the **Call answering rules** section.
- 3 Choose **Calls ring me** and then click the drop down under **if unanswered** and select **new number or contact**.
- 4 In the **Add number or contact** box, enter your MiCollab AM pilot number. In this example, the MiCollab AM pilot number is 3000.



Call answering rules
Choose how you want to handle incoming calls.

☒ **Calls ring me** ☐ Forward my calls

Also ring


If unanswered

Ring for this many seconds before redirecting

To setup Call Forward All:

- 1 At the top of the Microsoft Teams client app, click **Settings and more** **⋮** adjacent to your profile picture.
- 2 Select **Settings** > **Calls** and go to the **Call answering rules** section.
- 3 Choose **Forward my calls** and then click the dropdown under **Forward to** and select **new number or contact**.

- 4 Enter your MiCollab AM pilot number in the **Add number or contact** box. In this example, the MiCollab AM pilot number is 3000.

Call answering rules
Choose how you want to handle incoming calls.
☐ Calls ring me ☒ Forward my calls
Forward to: 3000 

Configuring AudioCodes SBC

The tasks in the following section provide details for configuring AudioCodes SBC for internetworking between Microsoft Teams Direct Routing and MiCollab AM SIP Trunk. This configuration is done using the SBC's embedded Web server (thereafter, referred to as Web interface).

NOTE Direct Routing configuration between Microsoft Teams and AudioCodes SBC is not discussed here. You must configure Direct Routing between Microsoft Teams and AudioCodes SBC and verify that incoming requests from Microsoft Teams is able to reach SBC before proceeding with the configuration steps below. Only configurations pertinent to MiCollab AM are provided. Refer to the following link <https://www.audiocodes.com/media/13253/connecting-audiocodes-sbc-to-microsoft-teams-direct-routing-enterprise-model-configuration-note.pdf> on how to configure Direct Routing between Microsoft Teams and AudioCodes SBC.

To configure AudioCodes SBC for internetworking between Microsoft Teams Direct Routing and MiCollab AM SIP Trunk, you must perform the following steps in the following order:

- Configure Media Realms
- Configure SIP Signaling Interfaces
- Configure Proxy Sets
- Configure Proxy Address
- Configure IP Profiles
- Configure IP Groups
- Configure IP-to-IP Call Routing Rules
- Configure Number Manipulation Rules
- Configure Message Manipulation Rules

Configure Media Realms

Media Realms allow dividing the UDP port ranges for use on different interfaces.

To configure media realms:

- 1 Navigate to **Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Media Realms** and open the Media Realms table.
- 2 Configure Media Realms for MiCollab AM as follows (you can use the default Media Realm (Index 0), but modify it. Port range 6000 – 6999 is used in this example).

| Index | Name | IPv4 Interface Name | Port Range Start | Number of Media Session Legs |
|-------|------|---------------------|------------------|---|
| 0 | CX | LAN_IF_CX | 6000 | 100 (media sessions assigned with port range) |

| Media Realms (2) | | | | | | |
|---|-------|---------------------|----------------------|------------------------------|--------------------|---------------------|
| <div> + New Edit </div> <div> Page 1 of 1 Show 10 records per page </div> | | | | | | |
| INDEX | NAME | IPV4 INTERFACE NAME | UDP PORT RANGE START | NUMBER OF MEDIA SESSION LEGS | UDP PORT RANGE END | DEFAULT MEDIA REALM |
| 0 | CX | LAN_IF_CX | 6000 | 100 | 6999 | Yes |
| 1 | Teams | WAN_IF_Teams | 7000 | 100 | 7999 | No |

Configure SIP Signaling 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 and Media Realm.

To configure SIP Signaling Interfaces:

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

| 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 | CX | LAN_IF_CX | SBC | - | 5060 | 0 | Disable (Leave default value) | 500 (leave default value) | CX | - |

| SIP Interfaces (7) | | | | | | | | | |
|---|-------|--------------|-------------------|------------------|----------|----------|----------|------------------------|-------------|
| <div> + New Edit </div> <div> Page 1 of 1 Show 10 records per page </div> | | | | | | | | | |
| INDEX | NAME | SRD | NETWORK INTERFACE | APPLICATION TYPE | UDP PORT | TCP PORT | TLS PORT | ENCAPSULATION PROTOCOL | MEDIA REALM |
| 0 | CX | DefaultSRD (| LAN_IF_CX | SBC | 5060 | 0 | 0 | No encapsulation | CX |
| 1 | Teams | DefaultSRD (| WAN_IF_Teams | SBC | 0 | 0 | 5061 | No encapsulation | Teams |

Configure Proxy Sets

The Proxy Sets and Proxy Address defines the destination address (IP address or FQDN) of the IP entity server.

To configure proxy sets:

- 1 Navigate to **Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**

- 2 Configure Proxy Sets as shown in the table below:

| Index | Name | SBCIPv4 SIP Interface | TLS Context Name | Proxy Keep-Alive | Proxy Hot Swap | Proxy Load Balancing Method |
|-------|------|-----------------------|------------------|------------------|----------------|-----------------------------|
| 0 | CX | CX | Default | Disable | - | - |

| Proxy Sets (7) | | | | | | | |
|---|-------|-----------------|----------------------------|------------------------|-----------------------------|-----------------|----------------|
| + New Edit Delete | | | | | | | |
| 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 | CX | DefaultSRD (#0) | -- | CX | 60 | | Disable |
| 1 | Teams | DefaultSRD (#0) | -- | Teams | 60 | | Disable |

Configure Proxy Address

The Proxy Sets and Proxy Address defines the destination address (IP address or FQDN) of the IP entity server.

To configure the proxy address:

- 1 Navigate to **Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets** to open the Proxy Sets table. Click Proxy Set **CX**, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.
- 2 Click **+ New** and configure the address of Proxy Set according to the parameters described in the table below:

| Index | Proxy Address | Transport Type | Proxy Priority | Proxy Random Weight |
|-------|---------------------------------------|----------------|----------------|---------------------|
| 0 | 10.12.49.105:5060 (CX IP and port) | TCP | 0 | 0 |

#0

| GENERAL | |
|---------------------|---------------------|
| Proxy Address | • 10.12.49.105:5060 |
| Transport Type | • TCP |
| Proxy Priority | 0 |
| Proxy Random Weight | 0 |

- 3 Click **Apply**


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).

To configure IP Profiles:

- 1 Navigate to **Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **IP Profiles** to open the IP Profiles table.
- 2 Click **New** and configure the parameters as follows:

| Parameter | Value |
|---------------------------------|---|
| General | |
| Index | 1 |
| Name | CX |
| Media Security | |
| SBC Media Security Mode | Not Secured |
| 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 |
| Remote 3xx Mode | Handle Locally |

| IP Profiles (6) | | |
|--|-------|--------------------|
| + New Edit  | | |
| Page 1 of 1 Show 10 records per page | | |
| INDEX | NAME | PROFILE PREFERENCE |
| 1 | CX | 1 |
| 2 | Teams | 1 |

#2[CX] Edit

| | | | |
|---------------------------------------|------------------------------|---------------------------------|-----------------------------|
| GENERAL | | SBC SIGNALING | |
| Name | • CX | PRACK Mode | Transparent |
| Created by Routing Server | No | P-Asserted-Identity Header Mode | • Add |
| MEDIA SECURITY | | Diversion Header Mode | As Is |
| SBC Media Security Mode | • Not Secured | History-Info Header Mode | As Is |
| Gateway Media Security Mode | Preferable | Session Expires Mode | Transparent |
| Symmetric MKI | Disable | SIP UPDATE Support | Supported |
| MKI Size | 0 | Remote re-INVITE | Supported |
| SBC Enforce MKI Size | Don't enforce | Remote Delayed Offer Support | Supported |
| SBC Media Security Method | SDES | MSRP re-INVITE/UPDATE | Supported |
| Reset SRTP Upon Re-key | Disable | MSRP Offer Setup Role | ActPass |
| Generate SRTP Keys Mode | Only if Required | MSRP Empty Message Format | Default |
| SBC Remove Crypto Lifetime in SDP | No | Remote Representation Mode | According to Operation Mode |
| SBC Remove Unknown Crypto | No | Keep Incoming Via Headers | According to Operation Mode |
| SBC EARLY MEDIA | | Keep Incoming Routing Headers | According to Operation Mode |
| Remote Early Media | Supported | Keep User-Agent Header | According to Operation Mode |
| Remote Multiple 18x | Supported | Handle X-Detect | No |
| Remote Early Media Response Type | Transparent | ISUP Body Handling | Transparent |
| Remote Multiple Early Dialogs | According to Operation Mode | ISUP Variant | ITU92 |
| Remote Multiple Answers Mode | Disable | Max Call Duration [min] | 0 |
| Remote Early Media RTP Detection M... | By Signaling | SBC REGISTRATION | |
| Remote RFC 3960 Support | Not Supported | User Registration Time | 0 |
| Remote Can Play Ringback | Yes | NAT UDP Registration Time | -1 |
| Generate RTP | None | NAT TCP Registration Time | -1 |
| SBC MEDIA | | SBC FORWARD AND TRANSFER | |
| Mediation Mode | RTP Mediation | Remote REFER Mode | • Handle Locally |
| Extension Coders Group | -- | Remote Replaces Mode | • Handle Locally |
| Allowed Audio Coders | • AllowedAudioCodersGroups_1 | Play RBT To Transferee | No |
| Allowed Coders Mode | Restriction | Remote 3xx Mode | • Handle Locally |

3 Click **Apply**.

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., MiCollab AM server, 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.

To configure the proxy address:

- 1 Navigate to **Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups** to open the IP Groups table.
- 2 Click an IP Group for MiCollab AM:

| Parameter | Value |
|-------------|--------|
| Index | 0 |
| Name | CX |
| Type | Server |
| Proxy Set | CX |
| IP Profile | CX |
| Media Realm | CX |

| IP Groups (7) | | | | | | | | | | | |
|-----------------------|-------|----------------------|--------|--------------------|-----------|-------------|-------------|--------------------------|-----------------------|---------------------------------|----------------------------------|
| + New | | Edit | | | | Page 1 of 1 | | Show 10 records per page | | <input type="text"/> | |
| INDEX | NAME | SRD | TYPE | SBC OPERATION MODE | PROXY SET | IP PROFILE | MEDIA REALM | SIP GROUP NAME | CLASSIFY BY PROXY SET | INBOUND MESSAGE MANIPULATED SET | OUTBOUND MESSAGE MANIPULATED SET |
| 0 | CX | DefaultSR | Server | Not Configur | CX | CX | CX | | Enable | -1 | -1 |
| 1 | Teams | DefaultSR | Server | Not Configur | Teams | Teams | Teams | | Disable | -1 | 4 |

Configure IP-to-IP Call Routing Rules

IP-to-IP call routing 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.

To configure IP-to-IP call routing rules:

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

| Index | Name | Source IP Group | Request Type | Call Trigger | ReRoute IP Group | Dest Type | Dest IP Group | Dest Address |
|-------|-----------------------------------|-----------------|--------------|--------------|------------------|--------------|---------------|--------------|
| 0 | Terminate OPTIONS | Any | OPTIONS | | | Dest Address | | internal |
| 1 | REFER from Teams (arbitrary name) | Any | | REFER | Teams | Request URI | CX | |
| 2 | Teams to CX (arbitrary name) | Teams | | | | IP Group | CX | |
| 3 | CX to Teams (arbitrary name) | CX | | | | IP Group | Teams | |
| 4 | REFER from CX (arbitrary name) | Any | | REFER | CX | IP Group | Teams | |

| IP-to-IP Routing (6) | | | | | | | | | | | |
|-----------------------|-------------|----------------------|---------------------------|------------------------|--------------|-------------------------|------------------------------|------------------|----------------------|---------------------------|---------------------|
| + 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 O | Default_SBCI | Route Row | Any | OPTIONS | * | * | Dest Address | -- | -- | internal |
| 1 | REFER from | Default_SBCI | Route Row | Any | All | * | * | Request URI | CX | -- | |
| 2 | Teams to CX | Default_SBCI | Route Row | Teams | All | * | * | IP Group | CX | -- | |
| 3 | CX to Teams | Default_SBCI | Route Row | CX | All | * | * | IP Group | Teams | -- | |
| 4 | REFER from | Default_SBCI | Route Row | Any | All | * | * | IP Group | Teams | -- | |

Configure Number Manipulation Rules

These rules manipulate the SIP Request-URI user part (source or destination number). The manipulation rules use the configured IP Groups to denote the source and destination of the call.

NOTE Adapt the manipulation table according to your environment dial plan. For example, for this interoperability test topology, a manipulation is configured to remove the "+" (plus sign) from the destination number for calls from the Microsoft Teams Direct Routing IP Group.

To configure a number manipulation rule:

1. Navigate to **Setup** menu > **Signaling & Media** tab > **SBC** folder > **Manipulation** > **Inbound Manipulations** to open Inbound Manipulations table.
2. Configure the rules according to your setup.

The figure below shows an example of configured IP-to-IP inbound manipulation rules for calls from Microsoft Teams Direct Routing IP Group:

Inbound Manipulations (1)

+ New

Edit

Insert

Page 1 of 1

Show 10 records per page

| INDEX | NAME | ROUTING POLICY | ADDITION MANIPUL | MANIPUL PURPOSE | SOURCE IP GROUP | SOURCE USERNAME PATTERN | DESTINAT USERNAME PATTERN | MANIPUL ITEM | REMOVE FROM LEFT | REMOVE FROM RIGHT | LEAVE FROM RIGHT | PREFIX TO ADD | SUFFIX TO ADD |
|-------|-----------|----------------|------------------|-----------------|-----------------|-------------------------|---------------------------|--------------|------------------|-------------------|------------------|---------------|---------------|
| 0 | From Team | Default_SE | No | Normal | Teams | * | + | Destination | 1 | 0 | 255 | | |

Configure 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.

To configure SIP message manipulation rules:

1. Navigate to **Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Manipulation** to open Message Manipulations page

- 2 Configure a new manipulation rule (Manipulation Set 2) for CX. This rule applies to messages sent to the CX IP Group. This replaces the host part of the SIP Request-URI Header with the CX IP address.

| Parameter | Value |
|---------------------|------------------------------|
| Index | 0 |
| Name | Change R-URI host toward CX |
| Manipulation Set ID | 2 |
| Condition | Any.Request |
| Action Subject | Header.Request-URI.URL.Host |
| Action Type | Modify |
| Action Value | Param.Message.Address.Dst.IP |

#0[Change R-URI host toward CX]

| GENERAL | |
|---------------------|-----------------------------|
| Name | Change R-URI host toward CX |
| Manipulation Set ID | 2 |
| Row Role | Use Current Condition |

| ACTION | |
|----------------|------------------------------|
| Action Subject | Header.Request-URI.URL.Host |
| Action Type | Modify |
| Action Value | Param.Message.Address.Dst.IP |

| MATCH | |
|--------------|-------------|
| Message Type | Any.Request |
| Condition | |

Message Manipulations (4)

+ New

Edit

Insert

⬆

⬇

🗑

⏪

⏩

Page 1 of 1

⏪

⏩

Show 10 records per page

🔍

| INDEX | NAME | MANIPULATION SET ID | MESSAGE TYPE | CONDITION | ACTION SUBJECT | ACTION TYPE | ACTION VALUE | ROW ROLE |
|-------|-------------------|---------------------|--------------|-----------|--------------------|-------------|------------------------------|-----------------------|
| 0 | Change R-URI host | 2 | Any.Request | | Header.Request-URI | Modify | Param.Message.Address.Dst.IP | Use Current Condition |

The following table includes SIP message manipulation rules which are grouped together under Manipulation Set IDs 2 and which are executed for messages sent to CX IP Group. These rules are specifically required to enable proper interworking between MiCollab AM and Microsoft Teams Direct Routing.

| Rule Index | Rule Description | Reason for Introducing Rule |
|------------|--|-----------------------------|
| 0 | This rule applies to messages sent to the CX IP Group; It replaces the host part of the SIP Request-URI Header with the CX IP address. | Per CX implementation |

Configuring MiCollab AM

Once the telephone system is programmed, you must configure MiCollab AM for the integration. There are two ways you can configure MiCollab AM: (1) Configuring MiCollab AM for the telephone system integration when you are installing MiCollab AM for the first time, or (2) Configuring the existing MiCollab AM with the new telephone system integration.

Click the appropriate steps that your system requires from below and follow the steps:

- [Configuring MiCollab AM for the Integration During Initial Installation](#)—Integrate the telephone system while you install MiCollab AM for the first time.
- [Configuring Existing MiCollab AM for the Integration](#)—Integrate a new telephone system on your exiting MiCollab AM system.

NOTE For general information on integrations, refer to the **Integrating MiCollab AM with the Telephone System** chapter in *System Installation Guide*, and the topic, **Integrate the Telephony Server with the Telephone System**, in the online help.

Configuring MiCollab AM for the Integration During Initial Installation

To configure MiCollab AM with the integration for the first time:

- 1 In the **Database Initialization Parameters** dialog box, configure the following options:

Database Initialization Parameters

Mailbox Length: 4 First Extension:

Switch

Manufacturer: AudioCodes

Model: SIP Gateway

Integration Type: SIP Trunk

Language

☒ American English only

Configure Now

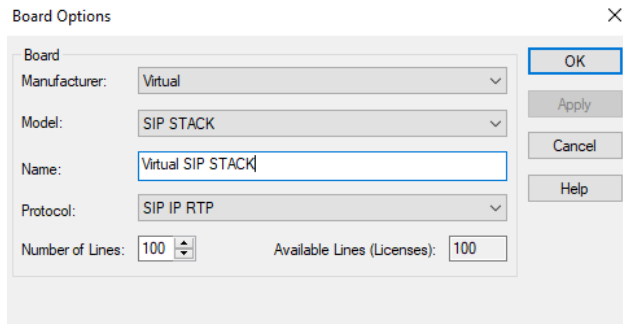
☒ VIM

< Back Next > Cancel Help

- a In the **Mailbox Length** box, enter the mailbox length in digits.

- b** In the **First Extension** box, enter first extension number for the first line. You can also leave the **First Extension** box empty.
- c** From the **Manufacturer** dropdown list, select **AudioCodes**.
- d** From the **Model** dropdown list, select **SIP Gateway**.
- e** From the **Integration Type** dropdown list, select **SIP Trunk**.

2 Click **Next**. The **Board Options** dialog box opens for the virtual board configuration.



The **Board Options** dialog box is shown with the following settings:

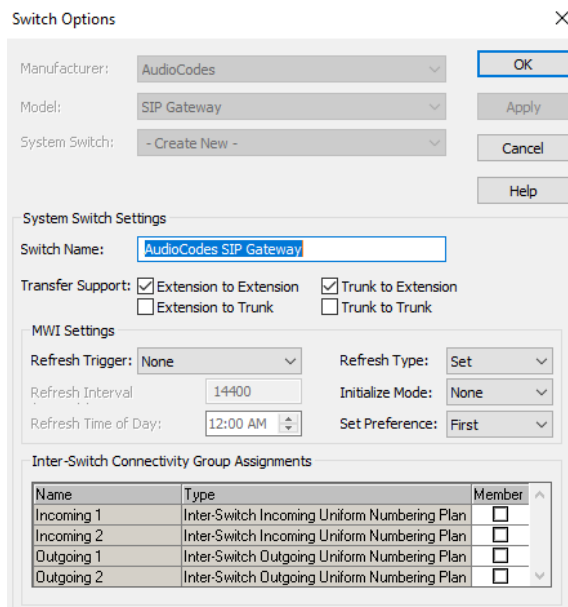
- Board:** Virtual
- Manufacturer:** Virtual
- Model:** SIP STACK
- Name:** Virtual SIP STACK
- Protocol:** SIP IP RTP
- Number of Lines:** 100
- Available Lines (Licenses):** 100

Buttons on the right: OK, Apply, Cancel, Help.

3 In the **Board Options** dialog box, configure the following options:

- a** From the **Manufacturer** dropdown list, select **Virtual**.
- b** From the **Model** dropdown list, select **Virtual SIP STACK**.
- c** In the **Name** field, the name for this board is automatically generated. Enter a new name if necessary.
- d** From the **Protocol** dropdown list, select **SIP IP RTP**.
- e** In the **Number of Lines** field, enter the number of lines this board uses. The total number of lines is limited by the capacity of the board and the number of **Available Line Licenses**.

4 Click **OK**. The **Switch Options** dialog box opens.



The **Switch Options** dialog box is shown with the following settings:

- Manufacturer:** AudioCodes
- Model:** SIP Gateway
- System Switch:** - Create New -
- System Switch Settings:**
 - Switch Name:** AudioCodes SIP Gateway
 - Transfer Support:**
 - ☒ Extension to Extension
 - ☒ Trunk to Extension
 - ☐ Extension to Trunk
 - ☐ Trunk to Trunk
- MWI Settings:**
 - Refresh Trigger:** None
 - Refresh Type:** Set
 - Refresh Interval:** 14400
 - Initialize Mode:** None
 - Refresh Time of Day:** 12:00 AM
 - Set Preference:** First
- Inter-Switch Connectivity Group Assignments:**

| Name | Type | Member |
|------------|--|--------------------------|
| Incoming 1 | Inter-Switch Incoming Uniform Numbering Plan | <input type="checkbox"/> |
| Incoming 2 | Inter-Switch Incoming Uniform Numbering Plan | <input type="checkbox"/> |
| Outgoing 1 | Inter-Switch Outgoing Uniform Numbering Plan | <input type="checkbox"/> |
| Outgoing 2 | Inter-Switch Outgoing Uniform Numbering Plan | <input type="checkbox"/> |

Buttons on the right: OK, Apply, Cancel, Help.

- 5 If necessary, make any changes to the default settings your site requires in the **Switch Options** dialog box.

NOTE The settings related to the telephone system in the **Switch Options** dialog box are filled in automatically when you select the correct telephone system during setup.

If you need to customize settings on the **Switch Options** dialog box to meet requirements specific to your site, refer to the documentation accompanying the telephone system, the online help, and the guide, *System Installation Guide*.

- 6 Click **OK**. The **Integration Options** dialog box opens.

- 7 In the **Integration Options** dialog box, configure the following options:

- a In the **Local Integration Settings** section, select the **Required Parameters** View and configure the settings as follows:
- b In the **Local Integration Settings** section, select the **Required Parameters** View and configure the settings as follows:

Table 3. Required Parameter Settings for Integration Options

| Field | Value |
|-------------------------------------|---|
| SIP Server Address | Enter the IP address of the AudioCodes SBC |
| SIP Server Port | Enter the listening port of the AudioCodes SBC. The default port number is 5060 . |
| Transport for outgoing SIP Messages | This is the transport medium that will be used to send outgoing SIP messages. The default is TCP . |
| Local IP Address to bind on | Select the IP address of the NIC on the Call Server platform that communicates with the AudioCodes SBC. The drop-down box lists all available local IP addresses. |

| Field | Value |
|-----------------------------|---|
| SIP Local Connection Port | Enter the port number where CX-E listens for incoming SIP messages. The default value is 5060 . |
| SIP parser qualifier string | <ul style="list-style-type: none"> • Single SIP integration on the call server—Enter the local IP address to which the integration is bound. This field is used by MiCollab AM to match SIP packets to the appropriate SIP integration. • Multiple SIP integrations on the call server—Use a string that is unique to each SIP integration. For example: <ol style="list-style-type: none"> a. The extension that will be used as the hunt number on the PBX followed by the @ symbol and the IP of the call server, such as 5000@172.16.4.202. <i>The hunt number must be unique across all IP integrations.</i> b. The Fully Qualified Domain Name (FQDN) of the switch, such as pbx1.sipdomain.com. <p>NOTE This setting must match a string in the SIP header that is unique to this particular integration.</p> |

- c** In the **Local Integration Settings** section, select the **Integration Specific Parameters** View, and configure the following options:

- In the **SIP Domain Name** field, enter your Microsoft Teams domain.
- Find **Type of Call Progress to use for External Calls** and set the value as how the gateway is used for the integration.

NOTE How this should be set depends on the gateway used for the integration.

- **Digita**—Select if the gateway supports call progress through to the endpoint.
- **Media**—Select if the gateway reports early that the call is connected, such as before the phone rings or while the phone is ringing.

- 8 Click **OK**. The **Switch Section Options** dialog box opens.

| Name | Value |
|------------------------|----------|
| Incoming Hunt Mode | Terminal |
| Hunt Group Access Code | 3000 |

- 9 In the **Switch Section Options** dialog box, configure the following options:
- a In the **Local Switch Section Settings** section, select the **Required Parameters** View.
 - b In **Incoming Hunt Mode**, select **Terminal**.
 - c In **Hunt Group Access Code** box, type the telephone number (pilot number) for this integration.
 - d Click **OK**.
- 10 Continue through and complete the configuration. At the end of the configuration, a confirmation dialog box opens. Click **OK**.
- 11 If **MiCollab AM Configuration** does not open automatically after the configuration completes, open **MiCollab AM Configuration**, and select the **Lines** tab.
- 12 In the table from the **Lines** tab, configure callouts for the application. For information on configuring callout settings, see the topic *Configuring Callout Settings*, in the online help system.
- 13 Click **OK** to save all changes.

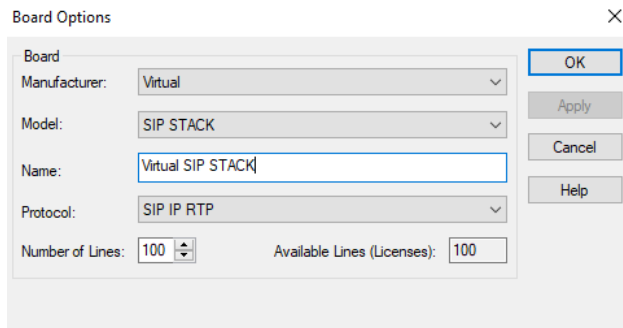
Configuring Existing MiCollab AM for the Integration

To configure exiting MiCollab AM for the telephone integration:

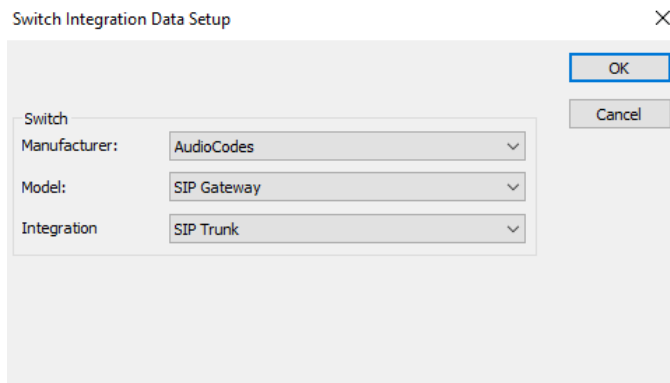
- 1 Open **MiCollab AM Configuration**, and go to the **Main** tab.
- 2 In the **Main** tab, click **Shutdown** to stop the system. Wait until the **Current Status** shows **Stopped**.

NOTE If you have not configured the virtual board with your MiCollab AM system yet, complete **Step 3**. If your MiCollab AM already has the virtual board configured, skip to **Step 4**.

- 3 [Optional] Select the **Board** tab, and then click the **Add** button. The **Board** dialog box opens.

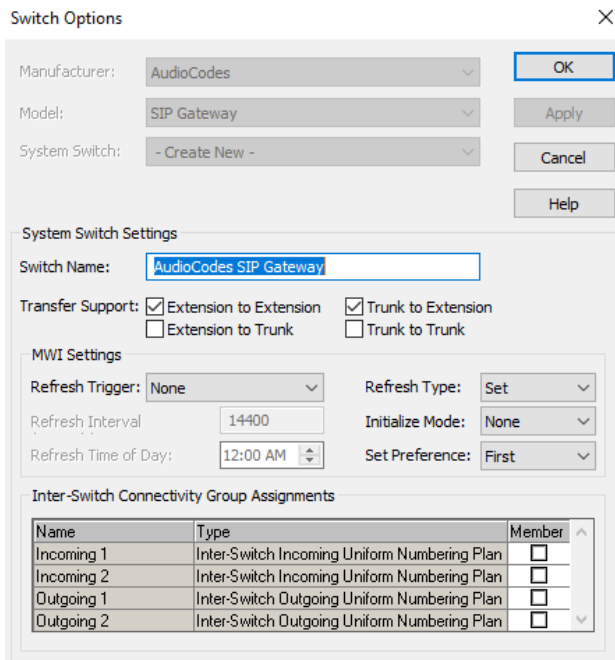
The 'Board Options' dialog box is shown. It has a title bar with a close button (X). The dialog is divided into two sections: 'Board' and 'Switch'. The 'Board' section contains the following fields: 'Manufacturer' (dropdown menu with 'Virtual' selected), 'Model' (dropdown menu with 'SIP STACK' selected), 'Name' (text field with 'Virtual SIP STACK' entered), 'Protocol' (dropdown menu with 'SIP IP RTP' selected), 'Number of Lines' (spin box with '100' entered), and 'Available Lines (Licenses)' (spin box with '100' entered). On the right side of the dialog, there are four buttons: 'OK' (highlighted with a blue border), 'Apply', 'Cancel', and 'Help'.

- a From the **Manufacturer** dropdown list, select **Virtual**.
 - b From the **Model** dropdown list, select **Virtual SIP STACK**.
 - c In the **Name** field, the name for this board is automatically generated. Enter a new name if necessary.
 - d From the **Protocol** dropdown list, select **SIP IP RTP**.
 - e In the **Number of Lines** field, enter the number of lines this board uses. The total number of lines is limited by the capacity of the board and the number of **Available Line Licenses**.
 - f Click **OK**.
- 4 Select the **Switch** tab, and click the **Add** button. The **Switch Integration Data Setup** dialog box opens.

The 'Switch Integration Data Setup' dialog box is shown. It has a title bar with a close button (X). The dialog is divided into two sections: 'Board' and 'Switch'. The 'Switch' section contains the following fields: 'Manufacturer' (dropdown menu with 'AudioCodes' selected), 'Model' (dropdown menu with 'SIP Gateway' selected), and 'Integration' (dropdown menu with 'SIP Trunk' selected). On the right side of the dialog, there are two buttons: 'OK' (highlighted with a blue border) and 'Cancel'.

- a From the **Manufacturer** dropdown list, select **AudioCodes**.
- b From the **Model** dropdown list, select **SIP Gateway**.
- c From the **Integration Type** dropdown list, select **SIP Trunk**.

- 5 Click **OK**. The **Switch Options** dialog box opens.



The **Switch Options** dialog box is shown. It has a title bar with a close button (X). The main area contains several sections:

- Manufacturer:** AudioCodes (dropdown)
- Model:** SIP Gateway (dropdown)
- System Switch:** - Create New - (dropdown)
- Buttons:** OK, Apply, Cancel, Help
- System Switch Settings:**
 - Switch Name:** AudioCodes SIP Gateway (text field)
 - Transfer Support:**
 - ☒ Extension to Extension
 - ☒ Trunk to Extension
 - ☐ Extension to Trunk
 - ☐ Trunk to Trunk
 - MWI Settings:**
 - Refresh Trigger:** None (dropdown)
 - Refresh Type:** Set (dropdown)
 - Refresh Interval:** 14400 (text field)
 - Initialize Mode:** None (dropdown)
 - Refresh Time of Day:** 12:00 AM (time picker)
 - Set Preference:** First (dropdown)
- Inter-Switch Connectivity Group Assignments:**

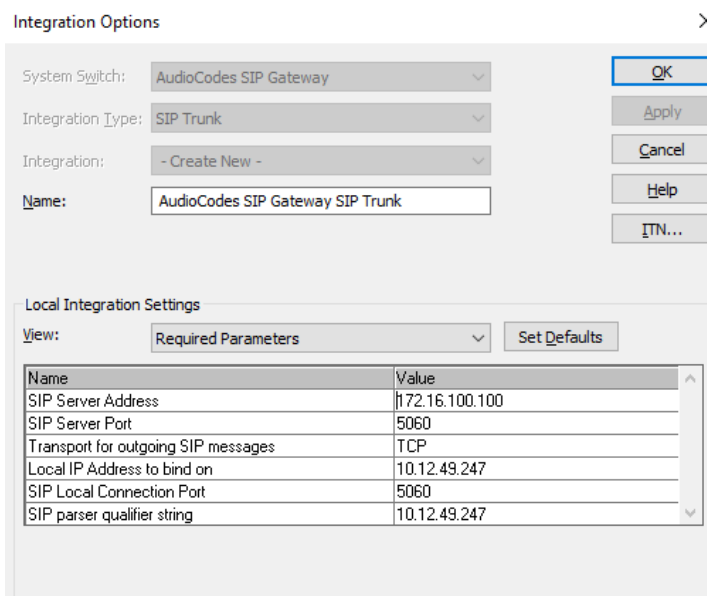
| Name | Type | Member |
|------------|--|--------------------------|
| Incoming 1 | Inter-Switch Incoming Uniform Numbering Plan | <input type="checkbox"/> |
| Incoming 2 | Inter-Switch Incoming Uniform Numbering Plan | <input type="checkbox"/> |
| Outgoing 1 | Inter-Switch Outgoing Uniform Numbering Plan | <input type="checkbox"/> |
| Outgoing 2 | Inter-Switch Outgoing Uniform Numbering Plan | <input type="checkbox"/> |

- 6 If necessary, make any changes to the default settings your site requires in the **Switch Options** dialog box.

NOTE The settings related to the telephone system in the **Switch Options** dialog box are filled in automatically when you select the correct telephone system during setup.

If you need to customize settings on the **Switch Options** dialog box to meet requirements specific to your site, refer to the documentation accompanying the telephone system, the online help, and the guide, *System Installation Guide*.

- 7 Click **OK**. The **Integration Options** dialog box opens.



The **Integration Options** dialog box is shown. It has a title bar with a close button (X). The main area contains several sections:

- System Switch:** AudioCodes SIP Gateway (dropdown)
- Integration Type:** SIP Trunk (dropdown)
- Integration:** - Create New - (dropdown)
- Name:** AudioCodes SIP Gateway SIP Trunk (text field)
- Buttons:** OK, Apply, Cancel, Help, ITN...
- Local Integration Settings:**
 - View:** Required Parameters (dropdown)
 - Set Defaults** (button)
 - | Name | Value |
|-------------------------------------|----------------|
| SIP Server Address | 172.16.100.100 |
| SIP Server Port | 5060 |
| Transport for outgoing SIP messages | TCP |
| Local IP Address to bind on | 10.12.49.247 |
| SIP Local Connection Port | 5060 |
| SIP parser qualifier string | 10.12.49.247 |

8 In the **Integration Options** dialog box, configure the following options:

- a In the **Local Integration Settings** section, select the **Required Parameters** View and configure the settings as follows:

Table 4. Required Parameter Settings for Integration Options

| Field | Value |
|-------------------------------------|---|
| SIP Server Address | Enter the IP address of the AudioCodes SBC. |
| SIP Server Port | Enter the listening port of the AudioCodes SBC. The default port number is 5060 . |
| Transport for outgoing SIP Messages | This is the transport medium that will be used to send outgoing SIP messages. The default is TCP . |
| Local IP Address to bind on | Select the IP address of the NIC on the Call Server platform that communicates with the AudioCodes SBC. The drop-down box lists all available local IP addresses. |
| SIP Local Connection Port | Enter the port number where CX-E listens for incoming SIP messages. The default value is 5060 . |
| SIP parser qualifier string | <ul style="list-style-type: none">• Single SIP integration on the call server—Enter the local IP address to which the integration is bound. This field is used by MiCollab AM to match SIP packets to the appropriate SIP integration.• Multiple SIP integrations on the call server—Use a string that is unique to each SIP integration. For example:<ul style="list-style-type: none">a. The extension that will be used as the hunt number on the PBX followed by the @ symbol and the IP of the call server, such as 5000@172.16.4.202. <i>The hunt number must be unique across all IP integrations.</i>b. The Fully Qualified Domain Name (FQDN) of the switch, such as pbx1.sipdomain.com. <p>NOTE This setting must match a string in the SIP header that is unique to this particular integration.</p> |

- b In the **Local Integration Settings** section, select the **Integration Specific Parameters** View, and configure the following options:

- In the **SIP Domain Name** field, enter your Microsoft Teams domain.
- Find **Type of Call Progress to use for External Calls** and set the value as how the gateway is used for the integration.

NOTE How this should be set depends on the gateway used for the integration.

- **Digital**—Select if the gateway supports call progress through to the endpoint.

- **Media**—Select if the gateway reports early that the call is connected, such as before the phone rings or while the phone is ringing.

9 Click **OK**. The **Switch Section Options** dialog box opens.

| Name | Value |
|------------------------|----------|
| Incoming Hunt Mode | Terminal |
| Hunt Group Access Code | 3000 |

- 10 In the **Switch Section Options** dialog box, configure the following options:
 - a In the **Local Switch Section Settings** section, select the **Required Parameters** View.
 - b In **Incoming Hunt Mode**, select **Terminal**.
 - c In **Hunt Group Access Code** box, type the telephone number (pilot number) for this integration.
 - d Click **OK**.
- 11 Continue through and complete the configuration. At the end of the configuration, a confirmation dialog box opens. Click **OK**.
- 12 If **MiCollab AM Configuration** does not open automatically after the configuration completes, open **MiCollab AM Configuration**, and select the **Lines** tab.
- 13 In the table from the **Lines** tab, configure callouts for the application. For information on configuring callout settings, see the topic *Configuring Callout Settings*, in the online help system.
- 14 Click **OK** to save all changes.

Configuring the Extension Device for Subscribers

You must add an extension device that matches the LineURI of the Microsoft Teams user and in which conforms to the Microsoft Teams Dial Plan/Normalization Rules.

To add an extension device to the subscriber's device:

- 1 Log on to **MiCollab AM Administration**, select a Subscriber Mailbox to edit, and then click the **Devices** tab.
- 2 Select the device you want to modify from the **Device List**.

- 3 In the **Properties**, add the subscriber extension device number in the **Number** field. In this example, the subscriber extension number is **2011**.

The screenshot shows the 'Properties' tab of a configuration window. The 'Device List' on the left contains a single entry 'Extension'. The 'Properties' section on the right is divided into several fields and checkboxes. The 'Number/Username' field is set to '2011'. The 'Type/Capabilities' dropdown is set to 'Phone: logon, can receive calls'. The 'Category' dropdown is set to 'Extension'. The 'Category Default' checkbox is checked. The 'Mailboxes Sharing this Number' field is empty. The 'Extension Properties' section includes a 'MWI' checkbox (unchecked), a 'Switch Section' dropdown set to 'AudioCodes SIP Gateway Section', a 'Direct Dial' field, an 'SMDI Prefix' field, and an 'Enable Fax Tone Detection' checkbox (unchecked). On the right side of the 'Properties' section, there are checkboxes for 'Primary Device' (checked), 'Primary Mobile Device' (unchecked), 'Ring Timeout (sec)' set to '14', and 'Active' (checked). A 'Barge In Sensitivity' slider is also present, ranging from -3 to +5, with the slider positioned at 0. At the bottom right, there are 'OK', 'Cancel', and 'Help' buttons.

- 4 Click **OK**.

Changing the Network Binding Order on the MiCollab AM Platform

If your MiCollab AM server platform is a component of two or more local or wide area networks (LANs or WANs), you must make sure that this integration does not interfere with the normal network operation of the server. By default, MiCollab AM uses the primary (public) network interface card (NIC) in the platform, the first NIC in the network binding order.

If you want MiCollab AM to use a NIC other than the first one, you must make several required configuration changes. It is much easier to configure the Integration to use another NIC by simply setting the integration parameter Local IP Address to bind on to the address of the NIC card connected to the PBX.

NOTE The operating system gives precedence to the first network connection in the list followed by the remaining connections based on their position in the list.

The instructions in this section ensure that the binding order is correct when you set up the integration. If you replace a NIC on the MiCollab AM server platform later, the platform's operating system registers the new adapter at the bottom of its binding order. Restoring the original binding order should correct any problems caused by the change.

IMPORTANT The following procedure shifts the binding order of the network interface cards. To determine which NIC is associated with a specific network connection, right-click the connection in the Network Connections window, and then select Properties.

The following task is intended for the following Windows Servers:

- Windows Server 2008 R2 with Service Pack 1
- Windows Server 2012 R2
- Windows Server 2016
- Windows Server 2019

To change the binding order of multiple NICs:

- 1 From the taskbar, click **Start > Control Panel**.
- 2 In the **Control Panel**, click **Network and Sharing Center**.
- 3 On the left pane, select **Change Adapter Settings**.
- 4 Press **Alt** to display the menu bar.
- 5 On the menu bar, select **Advanced**, and then click **Advanced Settings**.
- 6 On the **Adapters and Bindings** tab of **Advanced Settings**, click the network connection that serves MiCollab AM.
- 7 Click the up arrow button to the right of the **Connections** list as many times as needed to move the connection to the top of the list.

- 8 Click **OK**, and then close the **Network Connections** window and the **Control Panel**.

Configuring Quality of Service (QoS)

As of version 6.0, MiCollab AM has no internal support for QoS. QoS must now be implemented externally via group policies as Policy-Based QoS. Refer to your operating system's documentation for details.

Table 5. QoS Configuration

| Field | Setting |
|------------------|---|
| Application Name | At_TelephonyServer.exe |
| Protocol | Match the setting used for the integration UDP or TCP |
| Source Port | <p>MiCollab AM requires a range of ports for audio support. The MiCollab AM audio ports start at the Local Media Base UDP Port configured in the Server tab. Each MiCollab AM line reserves 10 ports. Hence, the port range starts from the number configured there, and goes to the last port of the last line. The formula for calculating the highest port number in the range is as follows:</p> $\text{BasePortNumber} + (\text{NumberOfCXPorts} * 10) - 1.$ <p>Hence, if the base port is 10000, and MiCollab AM has 8 lines, then the port range to use would be:</p> <p>10000:10079</p> |
| DSCP Value | 46 |