

MiCollab Advanced Messaging 9.3

Audiocodes SIP Gateway SIP Trunk for

Generic PBX

Integration Technical Note

For version 9.3 and above

Notice

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Preface

This Integration Technical Note (ITN) is written for dealers who are experienced with MiCollab Advanced Messaging (MiCollab AM) and are familiar with its procedures and terminology. This document assumes that you are familiar with the features and programming of the telephone system and the AudioCodes Mediant Gateway Series.

This document describes how to integrate MiCollab AM with a telephone system through the AudioCodes Gateway Series, using the Session Initiation Protocol (SIP) integration. This integration operates exclusively over a TCP/IP-based network; it uses no analog or digital voice telephony ports between the AudioCodes Gateway and MiCollab AM, but passes voice communication and signaling information over the network. MiCollab AM can thus be located anywhere within the LAN or WAN.

Critical application considerations are documented, as well as installation and programming procedures necessary to integrate MiCollab AM with the AudioCodes Gateway Series, referred to throughout this document as AudioCodes.

The AudioCodes integration consists of three major components: the telephone system, the AudioCodes Gateway Series, and MiCollab AM. The telephone system passes all signaling to MiCollab AM through the AudioCodes Gateway. The AudioCodes Gateway then acts as a bridge between the telephone system and MiCollab AM and will convert the signaling of the telephone system into the Session Initiation Protocol (SIP) for transmission over the network to MiCollab AM.

Each AudioCodes Gateway may support up to eight T1 (six E1) QSIG trunk connections to the telephone system, depending on the specific AudioCodes Gateway, and provides a network connection to MiCollab AM. These trunk lines are programmed in the telephone system as well as on the AudioCodes Gateway, refer to appropriate system documentation for configuring these. The pilot number defined during the telephone system configuration will be associated with the MiCollab AM for integration to the AudioCodes Gateway.

The AudioCodes Gateway reads calling-party and called-party data from the telephone system, and then converts the digits and audio stream into the SIP/RTP protocol and delivers it to MiCollab AM through the network interface.

The AudioCodes integration is a SIP trunk integration. The MiCollab AM ports are configured as SIP trunks and uses static SIP endpoints to communicate with the corresponding gateway endpoints of the AudioCodes Gateway. The incoming data is matched with the ringing extension, and MiCollab AM answers with the appropriate dialog. The AudioCodes Gateway routes all outgoing calls from MiCollab AM to the telephone system. Message waiting indicator (MWI) operation is also performed through the AudioCodes Gateway.

This ITN documents the procedure for setting up the MiCollab AM integration. The process consists of programming the telephone system, programming the AudioCodes Gateway and configuring MiCollab AM.

Use this document in conjunction with *System Installation and Configuration Guide*, the *System Administration Guide* and the MiCollab AM online help system, as well as any AudioCodes-specific or telephone system-specific 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 MiCollab AM Documentation Library includes the following documents and resources:

- **Administration Documentation.** Available as a PDF only. Contains the following:
 - **Administration Guides.** Available as a PDF only. Contains administrative guides for administrators about how to manage and configure the messaging system.
 - **Quick Reference Cards (QRC).** Contains shortcuts and quick instructions telling subscribers how to access and use the messaging system.
 - **User Guides.** Available as a PDF only. Contains user guides for subscribers about accessing the messaging system and checking and sending messages.
- **Server Documentation.** Available as a PDF only. Contains the following:
 - **Developer Resources.** Contains programming guides and API references for developers for integrating the server clients and web applications with MiCollab AM.
 - **Installation and Configuration.** Available as a PDF only. Contains installation and configuration guides for server administrators about how to install and configure the messaging system.
 - **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.
 - **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 documents and program files from our partner web site: www.mitel.com

Help

The primary source of information about MiCollab AM is the online help available within any of its administrative utilities. You can access **Help** by clicking the **Help** button in the dialog box or window in which you are working.

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** Titles of other documents are shown in italics.

Example: See the *System Installation and Configuration Guide*.

- **User Interface (UI) Element Names.** Names of UI elements such as dialog boxes, windows, screens, menu items, tabs, buttons, and icons are shown in bold.

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

- **User Input.** Information required to be typed 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 MiCollab AM 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.

For more detailed documents, refer to the following list of references:

Table 1. References

Document Type	Document Title
Administration Documentation	<i>System Administration Guide</i>
External Resource at www.AudioCodes.com	<i>AudioCodes® Gateway Series Getting Started Guides</i>
External Resource at www.AudioCodes.com	<i>AudioCodes® Gateway Installation and Configuration Integration Notes</i>
Server Documentation	<i>System Installation and Configuration Guide</i>
Spare Parts Documentation	<i>AudioCodes Mediant800B Series Media Gateway Installation and Replacement Spare Parts Document</i>
Online help	<i>MiCollab AM online help system</i>

Features Supported by This Integration

The following tables list the features supported using the AudioCodes SIP Gateway SIP Trunk integration.

Table 2. Call forward to personal greeting support for these common call types

Divert to MiCollab AM on	Supported
No Answer	Yes
Busy	Yes
Forward All	Yes
Do Not Disturb	Yes

Table 3. Integration Features Supported for AudioCodes for SIP Trunk

Feature	Supported	Notes
Automatic subscriber logon	Yes	
ANI/CLI	Yes	
Announce Busy greeting on forwarded calls	Yes	
Call screening	Yes	Note 1

Caller queuing	Yes	Note 2
DNIS	Yes	
End-to-end DTMF, attendant console	Yes	
End-to-end DTMF, proprietary telephones	Yes	
End-to-end DTMF, joined calls	No	
Fax Tone Detection	Yes	
Internal calling party ID for reply	Yes	
Live record, integrated	No	Note 3
Live reply to sender	Yes	
Message notification callouts	Yes	
MWI, set/clear	Yes	
MWI, inband/outband	Outband	
Networking, analog	Yes	
Overflow from MiCollab AM to attendant	Yes	
Overflow to MiCollab AM from attendant	Yes	
PBX-provided disconnect signaling	Yes	
Revert to operator	Yes	
SRTP	No	Note 4
TLS	No	Note 4
Transfers, blind	Yes	
Transfers, confirmed	Yes	
Transfers, fully supervised	Yes	
Transfers, monitored	Yes	
Trunk ID for call routing	Yes	
Multiple Integrations	Yes	Note 5

NOTES

1. Only available when using supervised transfers
2. Caller Queuing is specific to each local Call Server. Call Servers within the system are unaware of queued calls to the same subscriber on other Call Servers. For more information, refer to [Critical Application Considerations](#).
3. Live Record is not supported with this SIP integration.
4. 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. Also, please note that SRTP has not been qualified for this integration, and no switch programming is available for setting up SRTP on the switch side. However, SRTP may be enabled as described above, and technical support will be available on a best effort basis.

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

5. See [Critical Application Considerations](#).

Critical Application Considerations

Known limitations or conditions within the telephone system and MiCollab AM that affect the integration performance are listed here. General recommendations are provided when ways to avoid these limitations exist.

AudioCodes/MiCollab AM Related Critical Application Considerations

- The primary (public) network interface card (NIC) must be the first network connection in the network binding order. On a MiCollab AM server with two or more NICs, the NIC that supports this integration must not occupy first place in the operating system's binding order. MiCollab AM binds and communicates to other servers and subscribers on this network connection. For more information, refer to [Changing the Network Binding Order on the MiCollab AM Platform](#).
- If you make changes to the parameters on the **Integrations Options** dialog box of the Call Server, you may not need to restart the Call Server to update the AudioCodes configuration. The AudioCodes Gateway is configured separately for connectivity to MiCollab AM.
- The network interface card (NIC) supporting the AudioCodes integration must be a 100MB full duplex link or greater. The G.711 PCM codec requires approximately 80kbps for one-way conversation, or 160kbs per MiCollab AM port.
- Use the G.711 protocol is used to communicate with the AudioCodes Gateway. MiCollab AM does not support the G.729AB protocol.
- A unique name must be entered in the SIP Parser Qualifier String field for each integration to an AudioCodes device on the network. This is a required parameter in the MiCollab AM Integrations Options dialog box. If you configure the AudioCodes Gateway with a FQDN (Fully Qualified Domain Name), enter the FQDN of the AudioCodes device in the SIP Parser Qualifier String field. Otherwise, enter a unique name, such as AudioCodes_1, that distinguishes SIP messages generated by one AudioCodes device from any other AudioCodes device in the system. This allows MiCollab AM to handle incoming calls associated with each AudioCodes device correctly.
- Each AudioCodes must be running version 7.2. Previous versions of firmware and software have not been qualified or tested with MiCollab AM and may not be supported.
- Match the hunt mode in the **Switch Section Options** dialog box with the type of hunting provided by the PBX. This helps to alleviate any glare conditions between the PBX and the Call Server. The default mode is **Terminal**.
- The Call Queuing feature does not transcend the Call Server. Calls may be queued on multiple Call Servers for the same subscriber but Call Servers do not have knowledge of calls in the queue on other Call Servers within the system. Callers may be prompted with specific information about their place in the queue; however, the information pertains to the specific Call Server on which their call is queued.

- MiCollab AM 9.3 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 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 Avaya Communication Manager 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

Telephone System/AudioCodes Related Critical Application Considerations

- Non-numeric DTMF tones cannot be used as any character in the station number, nor can zero be used as a leading number of the extension. The maximum length of a station number is 10 digits.
- Telephone numbers in the name fields of either subscriber stations or MiCollab AM ports prevent forwarded calls to MiCollab AM from integrating correctly.
- If you plan to use supervised transfers (T-type), we recommend installing the Music on Hold (MOH) feature to assure callers of proper call handling and system operation. Otherwise, callers being transferred to a station by MiCollab AM experience a period of silence and might misunderstand what is happening to their calls.
- AudioCodes line ports must be assigned to a T1/E1 QSIG trunk group.

Installation Requirements

Review the following information before performing any of the procedures in this document. To successfully install this integration, you must meet the installation requirements for both the telephone system and MiCollab AM.

Telephone System Requirements

Please consult the AudioCodes website for information about telephone system (PBX) requirements. The AudioCodes website provides information on the software and hardware requirements for each telephone system that connects to the AudioCodes Gateway: www.AudioCodes.com.

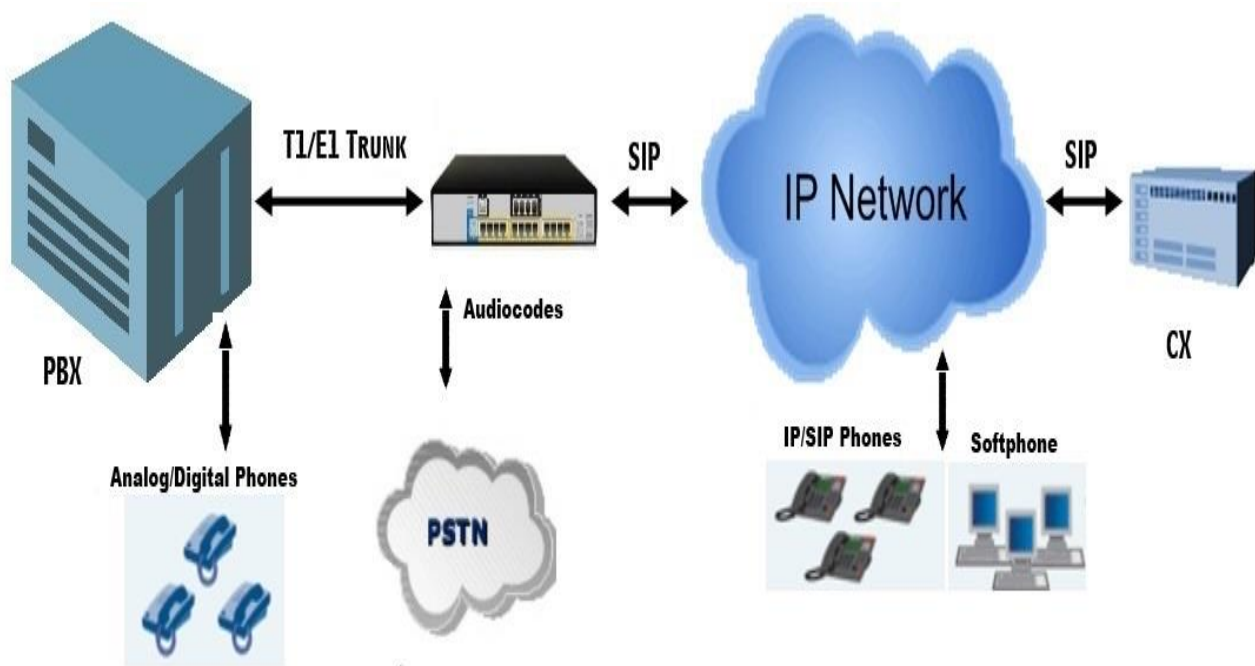


Figure 1. Typical Network Layout Example

AudioCodes Mediant Gateway Requirements

- One T1/E1 QSIG trunk for each telephone system connection
- Each AudioCodes Gateway can support from one to eight T1 (six E1) QSIG trunks depending upon the gateway series model and configured hardware. Consult the AudioCodes website for the AudioCodes model required for the PBX integration you are installing: www.AudioCodes.com.
- The AudioCodes Gateway must be running AudioCodes version 7.2 or later. To upgrade the AudioCodes Gateway, visit <http://www.AudioCodes.com>.

IMPORTANT Verify that the AudioCodes Gateway you are installing is compatible with the version 7.2 before you begin integration. If it is not, contact Mitel Technical Support.

MiCollab AM Requirements

- MiCollab AM version 9.3
- A MiCollab AM software key diskette or feature file with the AudioCodes Gateway integration enabled and one RADVISION® SIP and RTP license enabled for each port involved in the integration.
- One or two 100 MB, or 1000 MB (gigabit) network interface cards (NIC) with cables. The NIC used to communicate with the AudioCodes Gateway must be a 100MB full duplex card or greater and must be separate and distinct from other network cards in the system.

Programming the Telephone System

Follow the recommendations and programming examples provided by AudioCodes to program the telephone system for integration with AudioCodes Gateway Series.

The installing technician should be familiar with programming the telephone system. For detailed information on programming and installing the telephone system, you should refer to the documentation provided with telephone system, consult the vendor supporting the telephone system, or refer to the AudioCodes website www.AudioCodes.com for information on programming the PBX for operation with the AudioCodes Gateway.

Programming the AudioCodes Mediant Gateway

Follow the recommendations and programming examples in this section to program the AudioCodes Gateway via the AudioCodes Web interface for integration with MiCollab AM. Consult the documentation that shipped with the AudioCodes Gateway or go online to download the AudioCodes® *Gateway Series Getting Started Guides* and the *AudioCodes® Gateway Series Installation and Configuration Integration Notes* for more information on programming the AudioCodes Gateway.

The AudioCodes Gateway ships with a default IP address and requires initial setup to communicate over the same LAN as MiCollab AM. This section assumes you will use the administration console of the Call Server to initialize the AudioCodes Gateway for service with MiCollab AM.

IMPORTANT The AudioCodes Gateway must have a TCP/IP address that MiCollab AM can communicate with over the network. If you do not know this information, consult your network administrator for the correct address information required for installing both the AudioCodes Gateway and MiCollab AM.

Configuring the TCP/IP Address

Refer to the AudioCodes documentation for the initial programming and configuration of the TCP/IP Address for the AudioCodes Gateway.

NOTE The AudioCode **must** be restarted for the changes to take effect.

All AudioCode Gateways have the same default TCP/IP address at initial startup. If you are installing more than one AudioCodes Gateway in your network, you **must** connect them to the network one at a time to avoid TCP/IP address conflicts

It is recommended that you connect the AudioCodes Gateway to the LAN MiCollab AM is currently operating on. This can be anywhere on the corporate network, usually co-located at the same location as the MiCollab AM and telephone system.

AudioCodes Configuration for SIP Connection to MiCollab AM

Refer to the AudioCodes documentation for the initial programming and configuration of the SIP routing between the AudioCodes Gateway and MiCollab AM.

Configuring MiCollab AM

Once the telephone system is programmed and the T1/E1 QSIG trunk is in service and connected to the AudioCodes Gateway, 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 and Configuration Guide*, and the topic, **Integrating MiCollab AM with the Telephone System**, in the online help.

Configuring MiCollab AM for the Integration During Initial Installation

To configure MiCollab AM for the integration during the initial installation:

- 1 In the **Database Initialization Parameters** dialog box, configure the following options:
 - 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 appears.
 - a From the **Manufacturer** dropdown list, select **RadVision**.
 - b From the **Model** dropdown list, select **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**.
- 3 Click **OK**. The **Switch Options** dialog box appears.
- 4 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*.

- 5 Click **OK**. The **Integration Options** dialog box appears.
- 6 In the **Integration Options** dialog box, configure the options as follows:
 - a In the **Local Integration Settings** section, select the **Required Parameters** view and configure the following parameters:

Table 4. Required Parameters for Integration Options

Field	Value
Telephony Switch Type	Enter the correct telephone switch type. This field cannot be blank. The default is Lucent. Contact Mitel Technical Support if you do not know the switch type to enter in this field.
SIP Server Address	Enter the TCP/IP address or FQDN of the AudioCodes device.
SIP Server Port	Enter the SIP port on which the AudioCodes device is listening for SIP messages. The default value is 5060 .
SIP Domain Name	Enter the TCP/IP address of MiCollab AM.
Transport for outgoing SIP messages	Enter the transport protocol to be used for sending SIP messages to the AudioCodes device. Enter TCP . The default value is UDP .
SIP Device Name	Enter the hunt group number for the integration. This must match the hunt group number set in the switch section associated with this integration.
Local IP Address to Bind On	Select the local TCP/IP address of the MiCollab AM machine. This is a drop-down box and displays all available local TCP/IP addresses.
SIP Local Connection Port	The port where MiCollab AM listens for incoming SIP messages. The default 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:

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. *The hunt number must be unique across all IP integrations.*

The Fully Qualified Domain Name (FQDN) of the switch, such as pbx1.sipdomain.com.

NOTE This setting must match a string in the SIP header that is unique to this particular integration

Media Packet Size
(milliseconds)

MiCollab AM sends/receives packets containing the number of milliseconds worth of audio data set here. The default value is **20**.

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

- Set the **Type of Call Progress to use for External Calls** value. How this should be set depends on the gateway used for the integration as follows:
 - **Digital:** Select Digital if the gateway supports call progress through to the endpoint.
 - **Media:** Select Media if the gateway reports early that the call is connected, such as before the phone rings or while the phone is ringing.

- 7** Click **OK**. The **Switch Section Options** dialog box appears.

- 8** In the **Switch Section Options** dialog box, configure the following options:

- a** In the **Local Integration Settings** section, select the **Required Parameters** view.

- b** In the **Incoming Hunt Mode** field, enter the mode for this integration.

NOTE Select the hunt mode that matches the hunt mode type in the telephone system (PBX) programming.

- c** In the **Hunt Group Access Code** field, type the code that was programmed in the telephone system.

- d** Click **OK**.

- 9** Continue through and complete the configuration. At the end of the configuration, a confirmation dialog box appears. Click **OK**.

- 10** If **MiCollab AM Configuration** does not open automatically after the configuration completes, open **MiCollab AM Configuration**, and select the **Lines** tab.

- 11 In the table from the **Lines** tab, enter the extension numbers on the Lines in which they appear. Verify that the line numbers match the hunt group member DNs defined in the telephone system programming.

For information on configuring callout settings, see the topic *Configuring Callout Settings*, in the online help system.
- 12 Click **OK** to save all changes.

Configuring Existing MiCollab AM for the Integration

To configure existing 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 appears.
 - a From the **Manufacturer** dropdown list, select **RadVision**.
 - b From the **Model** dropdown list, select **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 **Switches** tab and click the **Add** button. The **Switch Integration Data Setup** dialog box appears.
 - 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 appears.
- 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 *System Installation and Configuration Guide*.

- 7 Click **OK**. The **Integration Options** dialog box appears.
- 8 In the **Integration Options** dialog box, configure the options as follows:
 - a In the **Local Integration Settings** section, select the **Required Parameters** view and configure the following parameters:

Table 5. Required Parameters for Integration Options

Field	Value
SIP Server Address	Enter the TCP/IP address or FQDN of the AudioCodes device.
SIP Server Port	Enter the SIP port on which the AudioCodes device is listening for SIP messages. The default value is 5060 .
Transport for outgoing SIP messages	Enter the transport protocol to be used for sending SIP messages to the AudioCodes device. Enter TCP or UDP . The default value is TCP .

Local IP Address to Bind On	Select the local TCP/IP address of the MiCollab AM machine. This is a drop-down box and displays all available local TCP/IP addresses.
SIP Local Connection Port	Enter the port where MiCollab AM listens for incoming SIP messages. The default 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. <p>For example:</p> <p>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></p> <p>The Fully Qualified Domain Name (FQDN) of the switch, such as pbx1.sipdomain.com.</p> <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:

Table 6. Integration Specific Parameters for Integration Options

Field	Value
Use configured name	Select the check box.
SIP Domain Name	Enter the TCP/IP address of MiCollab AM.
Local SIP Domain Name	Enter the TCP/IP address of MiCollab AM.
Type of Call Progress to use for External Calls	<p>How this should be set depends on the gateway used for the integration as follows:</p> <ul style="list-style-type: none"> • Digital: Select Digital if the gateway supports call progress through to the endpoint. • Media: Select Media 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 appears.

- 10** In the **Switch Section Options** dialog box, configure the following options:

- a** In the **Local Integration Settings** section, select the **Required Parameters** view.

- b** In the **Incoming Hunt Mode** field, enter the mode for this integration.

NOTE Select the hunt mode that matches the hunt mode type in telephone system (PBX) programming.

- c** In the **Hunt Group Access Code** field, type the code that was programmed in the telephone system.

- d** Click **OK**.

- 11** In **MiCollab AM Configuration**, verify that the telephone system is properly added and configured in the **Switches**, **Switch Sections**, and **Integrations** tabs.

- 12** Select the **Lines** tab.

- 13** In the table from the **Lines** tab, associate the lines with the Integration, this should equal the number of lines in the T1 and/or the number of licensed ports on the MiCollab AM. These would be defined in the telephone system programming.

Line	Extension	Switch Integration Name	Section	Callouts	Open	Tenant
1				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo
2				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo
3				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo
4				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo
5		AudioCodes SIP Gateway SIP Trunk	AudioCodes SIP Gateway Section	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Buffalo
6		AudioCodes SIP Gateway SIP Trunk	AudioCodes SIP Gateway Section	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Buffalo
7		AudioCodes SIP Gateway SIP Trunk	AudioCodes SIP Gateway Section	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Buffalo
8		AudioCodes SIP Gateway SIP Trunk	AudioCodes SIP Gateway Section	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	Buffalo
9				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo
10				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo
11				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo
12				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo
13				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo
14				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo
15				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo
16				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo
17				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo
18				<input type="checkbox"/>	<input type="checkbox"/>	Buffalo

For information on configuring callout settings, see the topic *Configuring Callout Settings*, in the online help system.

- 14** Click **OK** to save all changes.

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 document ensure that the binding order is correct when you set up the integration. However, 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**.

Windows Server 2012 R2

To change the binding order of multiple NICs:

- 1 From the taskbar, click **Start** > **Control Panel**.
- 1 In the **Control Panel**, click **Network and Internet** > **Network and Sharing Center**.
- 2 On the left pane, select **Change Adapter Settings**.
- 3 Press **Alt** to display the menu bar.
- 4 On the menu bar, select **Advanced**, and then click **Advanced Settings**.
- 5 On the **Adapters and Bindings** tab of **Advanced Settings**, click the network connection that serves MiCollab AM.
- 6 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.
- 7 Click **OK**, and then close the **Network Connections** window and the **Control Panel**.

Windows Server 2016 / 2019

To change the binding order of multiple NICs:

- 1 From the taskbar, select **Start > Control Panel**.
- 2 In the **Control Panel**, click **Network and Internet > Network and Sharing Center**.
- 3 On the left pane, select **Change Adapter Settings**.
- 4 Right-click the network connection that serves MiCollab AM and then select **Properties**.
- 5 On the **Networking** tab of the **Local Area Connection Properties** dialog box, select **Internet Protocol Version 4 (TCP/IPv4)**, and then click **Properties**.
- 6 On the **General** tab of the **Internet Protocol Version 4 (TCP/IPv4) Properties** dialog box, click the **Advanced** button.
- 7 On the **IP Settings** tab of the **Advanced TCP/IP Settings** dialog box, clear the **Automatic metric** check box and then type in a low value in the **Interface metric** field. The lower the value, the higher the priority.

NOTE For all Windows systems, the value 1 is reserved for the loopback adapter. It is recommended to use a value of 2 or higher for the network connection that serves MiCollab AM.

- 8 Click **OK** on all of the dialog boxes to save the settings, and then close the **Local Area Connection Properties** dialog box.
- 9 Repeat steps 4 through 8 to assign an Interface metric value to all other network adapters.

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 7. 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