

MiCollab Advanced Messaging 9.4 Mitel MiVoice Connect SIP Trunk Integration Technical Note

For version 9.4 and above

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Preface

This Integration Technical Note (ITN) is written for technicians who are experienced with MiCollab Advanced Messaging (MiCollab AM) and are familiar with its procedures and terminology. This document also assumes that you are familiar with the features and programming of the Mitel MiVoice Connect telephone system.

NOTE The configuration procedures in this document also apply to ShoreTel Connect Onsite PBX.

This document describes how to integrate MiCollab AM with the Mitel MiVoice Connect PBX telephone system, using the Session Initiation Protocol (SIP) integration.

The Mitel MiVoice Connect SIP integration consists of the Mitel MiVoice Connect PBX and MiCollab AM. This integration operates exclusively over an IP-based network; it uses no analog or digital voice telephony ports, but passes voice communication and signaling information over a static network connection between the Mitel MiVoice Connect PBX and MiCollab AM.

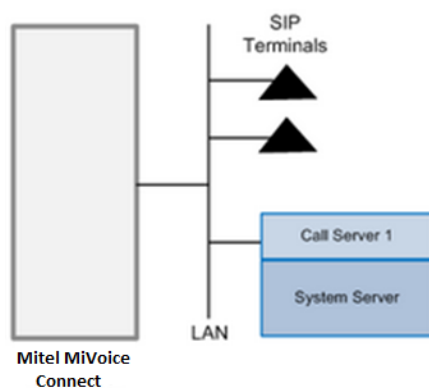


Figure 1. Diagram of Mitel MiVoice Connect System Integration

All calls for MiCollab AM are routed to the hunt group pilot number through the Mitel MiVoice Connect SIP Server interface. MiCollab AM performs the call hunting. Similarly, MiCollab AM routes all outgoing calls and MWI requests through the SIP Server interface.

This document describes the procedure for setting up the Mitel MiVoice Connect integration between MiCollab AM and a Mitel MiVoice Connect telephone system. The process consists of programming the Mitel MiVoice Connect PBX telephone system, installing MiCollab AM software and configuring MiCollab AM.

Critical application considerations are documented, as well as installation and programming procedures necessary to integrate MiCollab AM with the Mitel MiVoice Connect PBX, referred to throughout this document as MiVoice Connect.

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>
Server Documentation	<i>System Installation and Configuration Guide</i>
Online help	MiCollab AM online help system

Features Supported by this Integration

The following tables list the features supported using the Mitel MiVoice Connect SIP 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 - Limited Support – Note
Do Not Disturb	Yes (as FW All) – Note

NOTES

The Mitel MiVoice Connect does not allow loop-back calls on the SIP Server interface. This limitation may cause the following issues where a subscriber's extension is set to unconditionally forward to MiCollab AM:

- a.** If MiCollab AM initiates a consultation call or an IMN (Immediate Message Notification) callout, the extension will ring and the call does not forward immediately. The Ring No Answer (RNA) timeout will ensure that the call is eventually pulled back. The subscriber may press the *Send to Voicemail* key on the phone to forward the call to MiCollab AM immediately.
- b.** If MiCollab AM performs a supervised call transfer to a subscriber with call screening enabled, then the extension will ring and the call will not forward. If the *Send to Voicemail* key is pressed instead of letting the call be pulled back on timeout, the subscriber may receive an additional voice message containing an audio prompt asking the subscriber to accept or reject the call.

Table 3. Integration Supported Features for Mitel MiVoice Connect SIP Trunk

Feature	Supported	Notes
Automatic subscriber logon	Yes	

ANI/CLI/DNIS	Yes	
Announce Busy greeting on forwarded calls	Yes	
Call screening	Yes	Note 1, Note 2
Caller queuing	Yes	Note 1
End-to-end DTMF, from trunks	Yes	
End-to-end DTMF, proprietary telephones	Yes	
Fax Tone Detection	Yes	
Internal calling party ID for reply	Yes	
Live record, integrated	No	
Live reply to sender	Yes	
Message notification callouts	Yes	
MWI, set/clear	Yes	
MWI, inband/outband	Outband	
PBX-provided disconnect signaling	Yes	
SRTP	No	Note 4
TLS	No	Note 4
Transfers, blind	Yes	
Transfers, confirmed	Yes	
Transfers, fully supervised	Yes	Note 3
Transfers, monitored	No	
Trunk ID for call routing	No	
Multiple Integrations	Yes	Note 5
Automatic subscriber logon	Yes	
ANI/CLI/DNIS	Yes	

NOTES

1. Available only when using supervised transfers.
2. The Call Screening feature is not supported in cases where the end point proprietary phone will perform a *Send to Voicemail* action on a call that is made through the voicemail system. Subscribers that want to use the confirm feature must not manually redirect the call using phone's feature keys if the call originated from the voicemail system (no loop back calls are allowed if Call Screening feature is enabled). In the case a user uses the *Send to Voicemail* phone feature key and has Call Screening enabled, then the user might also receive a voice message containing the confirmation prompt. The caller will still be allowed to leave a message
3. Because of a MiVoice Connect limitation, if calling through the voicemail system and requesting a supervised transfer to a Forward All phone, the phone will ring instead of getting immediately to the subscriber's greeting. However, the voicemail's Ring No Answer timeout will assure that the caller will eventually hear the subscriber's greeting. The user can also choose to redirect the call to the voicemail system (before this timeout) by using the *Send to Voicemail* feature key button on Mitel MiVoice proprietary phones.
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.

- 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. The primary (public) network interface card (NIC) must be the first network connection in the network 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](#).
- MiCollab AM supports G.729a with support for annex b on the incoming audio stream only. MiCollab AM does not transmit annex b packets.
- When codec negotiation takes place between MiCollab AM and the PBX, MiCollab AM always offers the G.729a audio format as an option. You may configure G.729a as the preferred codec in MiCollab AM; however, the decision whether to use G.729a is always made by the PBX.
- The **Incoming Hunt Mode** must be configured in the **Switch Section Options** dialog box. This integration only supports terminal, circular, reverse terminal and reverse circular hunt modes. The default mode is Terminal.
- The MiVoice Connect SIP integration cannot route calls to multiple Call Servers. This integration supports one Call Server only. However, calls can be routed to multiple call servers using the Mitel SIP Routing Manager.
- Each MiVoice Connect switch supports 24 simultaneous calls through each of its SIP Server interfaces.
- Monitored transfers are not supported on the MiVoice Connect PBX.
- MiVoice Connect does not allow loop-back calls on the SIP Server interface. This limitation may cause the following issues where a subscriber's extension is set to unconditionally forward to MiCollab AM:
 - If MiCollab AM initiates a consultation call or an IMN (Immediate Message Notification) callout, the extension will ring and the call does not forward immediately. The Ring No Answer (RNA) timeout will ensure that the call is eventually pulled back. The subscriber may press the *Send to Voicemail* key on the phone to forward the call to MiCollab AM immediately.
 - If MiCollab AM performs a supervised call transfer to a subscriber with call screening enabled, then the extension will ring and the call will not forward as. If the *Send to Voicemail* key is pressed instead of letting the call be pulled back on timeout, the subscriber may receive an additional voice message containing an audio prompt asking the subscriber to accept or reject the call.
- In order for voicemail initiated Live Record to work, enough separate Conference ports must be configured on a Mitel Voice Switch (such as a SG220T1) so that VM SIP channels can be conferenced in. The call must then be conferenced in using the *Join* feature on the subscriber's phone. If this

feature is desired, please see Mitel MiVoice Connect documentation on how to program Conference trunks.

- 3-way conferencing with the voicemail SIP trunk is not allowed. Because of this, voicemail Live Record will not work. Conferencing works only when telephones are used.
- MiCollab AM 9.4 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.
- The MiCollab AM **Integration Options** parameter, **Validate Remote Hosts for Media** validates each incoming audio packet and accepts it only if it is sent from a valid endpoint. The parameter is disabled by default. Enabling this parameter causes MiCollab AM to reject RTP packets from invalid endpoints, rejects MWI packets that timeout after a specified number of times, and overcomes port lockups when callers hang up while MiCollab AM is performing a blind transfer.

IMPORTANT Enabling this parameter causes processing overhead and should only be enabled when necessary.

Installation Requirements

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

Telephone System Requirements

- Mitel MiVoice Connect version 18.1 Build 21.90.9738.0 or prior supported versions
- One External Unified Messaging SIP Link license

MiCollab AM Requirements

- MiCollab AM version 9.4
- MiCollab AM software key diskette or feature file with the Mitel MiVoice Connect SIP integration enabled and one Virtual SIP and RTP license enabled for each port involved in the integration
- One or more network interfaces

Programming the Telephone System

Follow the recommendations and programming examples in this section to program the telephone system for integration with MiCollab AM. Programming examples show commands and parameters that are necessary for integration; they do not represent PBX programming in its entirety.

The installing technician should be familiar with programming the MiVoice Connect system. For detailed information on programming and installing the telephone system, refer to the *MiVoice Connect Planning and Installation Guide* and the *MiVoice Connect System Administration Guide*. Programming is performed through the Mitel Connect Director.

Creating a Virtual IP Address (VIP) for SIP Proxy Redundancy

Creating a Virtual IP address (VIP) for SIP proxy redundancy is an optional step that improves resiliency for sites with two Mitel voice switches. MiCollab AM communicates with the PBX system through the VIP address. The VIP is accessible to MiCollab AM as long as at least one of the two Mitel voice switches configured with this VIP remains in service.

In addition, you can use the VIP at sites with a single Mitel voice switch. Creating a virtual IP address on a single Mitel voice switch keeps the configuration steps uniform and, if additional Mitel voice switches are installed later, no additional configuration changes to MiCollab AM are required.

NOTE Assigning SIP proxy resources is required to create a Virtual IP address (VIP) only. It is an optional procedure; it does not have any effect on the integration between MiVoice Connect and MiCollab AM.

To create a Virtual IP address and assign it to the Mitel voice switch:

- 1 In Mitel Connect Director, go to **Administration > Appliances/Servers > Platform Equipment**, and then select a voice switch to assign a Virtual IP address.
- 2 On the **Switch** properties sheet, assign SIP proxy ports to the voice switch.

The following sample image displays the SIP proxy ports assignment (see the **Built-in capacity** field) on the voice switch SG220T1.

SG220T1: SG220T1 - 172.16.30.20

GENERAL **SWITCH**

Built-in capacity:
 IP phone + SIP trunks = Total
 99 of 100 (20 SIP proxy ports)

☐ Enable Jack based Music on hold
 Jack based Music on hold gain: dB (-49 to 13)

☐ Assign digital ports as 20 SIP Trunks with Media Proxies

The following sample image displays the SIP proxy ports assignment on voice switch SG220T1.

Digital ports:						
Port	Port Type	Trunk Group	Description	Jack Number	Tx Gain (dB)	Rx Gain (dB)
1	5 IP Phones ▼			P01	0	0
2	100 SIP Proxy ▼			P02	0	0
3	5 SIP Trunks ▼			P03	0	0

NOTE Please refer to the appropriate MiVoice Connect administration guide to configure other voice switch models.

- In MiVoice Connect Director, select **Administration > System**, and then select **Sites**.
- Select a site to assign a Virtual IP Address.
- On the **General** properties sheet, enter an IP address, and then assign the voice switches to the **Virtual IP address** by selecting them from the drop-down list in the **Proxy switch 1** and **Proxy switch 2** fields.

Virtual IP address:

Proxy switch 1:

Proxy switch 2:

- Click **Save**.

Creating a SIP Profile for MiCollab AM

Create a SIP Profile for the voicemail integration.

To create a SIP Profile for MiCollab AM:

- Using the MiVoice Connect Director, go to **Administration > Appliances/Servers > Integrated Servers > SIP Profiles**, and then click **New**.

SIP Profiles

Voicemail

GENERAL

Name:

Voicemail

User agent:

.*

Priority:

100

☒ Enable

System parameters:

acceptMWI=notify
Accept302=sip
HoldSupport=no
AddrSupport=diversion
EnableSymmetricDtmf=yes
UseSipProxyOut=yes
OAEMedialessPort=8600
AllowedCodecs=PCMU/8000
OptionsPing=1

Custom parameters:

acceptMWI=notify
Accept302=sip
HoldSupport=yes
AddrSupport=diversion
EnableSymmetricDtmf=yes
OptionsPing=1
SendEarlyMedia=1
UseSipProxyOut=yes

- 2 In the **Name** field, enter a name for the SIP Profile, for example *Voicemail*.
- 3 In the **User Agent** field, enter a *period* followed by an *asterisk* (.*)
- 4 In the **Priority** field, enter *100*.
- 5 Select the **Enable** check box to enable the profile.
- 6 In the **Custom Parameters** field, enter the following custom parameters:
 - *acceptMWI=notify*
 - *Accept302=sip*
 - *HoldSupport=yes*
 - *AddrSupport=diversion*
 - *EnableSymmetricDtmf=yes*
 - *OptionsPing=1*
 - *SendEarlyMedia=1*
 - *UseSipProxyOut=yes*
- 7 Click **Save**.

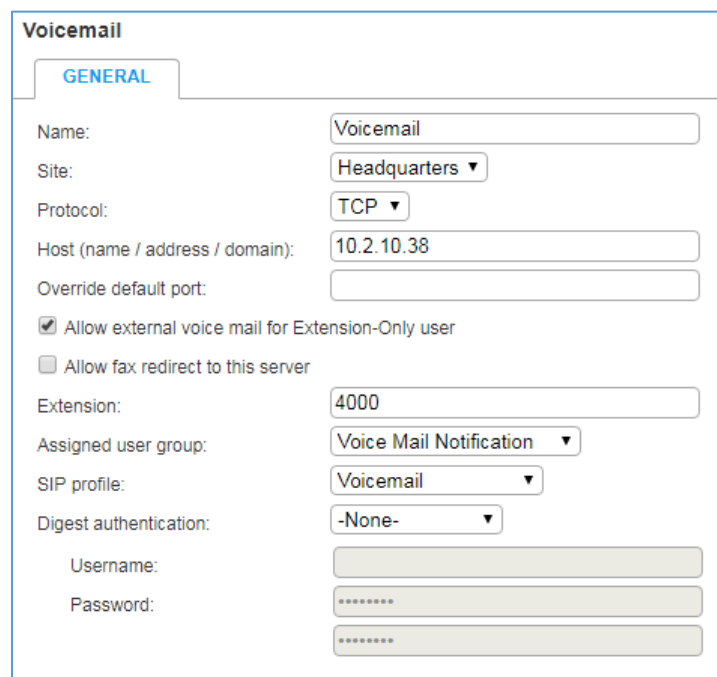
NOTE You can safely ignore warnings from the Mitel Connect Director regarding custom SIP Profile configurations.

Creating the SIP Server for MiCollab AM

Create a SIP Server for the voicemail integration.

To create a SIP Server for MiCollab AM:

- 1 Using the Mitel Connect Director program, go to **Administration > Appliances/Servers > Integrated Servers > SIP Servers**, and then click **New**.



The screenshot shows the 'Voicemail' configuration page with the 'GENERAL' tab selected. The fields are as follows:

- Name: Voicemail
- Site: Headquarters (dropdown)
- Protocol: TCP (dropdown)
- Host (name / address / domain): 10.2.10.38
- Override default port: (empty)
- ☒ Allow external voice mail for Extension-Only user
- ☐ Allow fax redirect to this server
- Extension: 4000
- Assigned user group: Voice Mail Notification (dropdown)
- SIP profile: Voicemail (dropdown)
- Digest authentication: -None- (dropdown)
- Username: (empty)
- Password: (masked with asterisks)

- 2 In the **Name** field, enter the name of the SIP Profile previously created, for example *Voicemail*.
- 3 In the **Site** field, select the site from the list.
- 4 Select **TCP** as the **Protocol**.
- 5 In the **Host (name/address/domain)** field, enter the IP Address or FQDN of MiCollab AM. This is the IP Address or FQDN of the Call Server or System Server with Call Services to which the SIP ports are connected.
- 6 Select the **Allow external voice mail for Extension-Only user** check box.
- 7 In the **Extension** field, enter the pilot number of the MiCollab AM hunt group.
- 8 In the **Assigned user group** field, select **Voice Mail Notification** from the list.
- 9 In the **SIP profile** field, select the SIP Profile for the voicemail integration you created in the previous procedure, [Creating a SIP Profile for MiCollab AM](#).
- 10 In the **Digest authentication** field, select **None**.

11 Click **Save**.

Creating the User Group for MiCollab AM

Create a User Group for the voicemail integration.

To create the User Group for MiCollab AM:

- 1 Using the Mitel Connect Director program, go to **Administration > Users > User Groups**, and then click **New**.

Voicemail Users

GENERAL PROFILE

Name: Voicemail Users

COS - Telephony: Fully Featured [View Class of Service](#)

COS - Call Permissions: No Restrictions [View Class of Service](#)

COS - Voice Mail: Large Mail Box [View Class of Service](#)

☒ Send caller ID as caller's emergency identification (CESID)

☒ Send DID as caller's emergency identification (CESID)

Account code collection mode: None

☒ Show Mitel Connect client users a list of account codes when dialing

Voice mail interface mode: External Voice Mail, SIP

Music on hold: Mitel

Outgoing trunk groups (Access Code):

Available:				Selected:			
NAME	ACCESS CODE	TYPE	SITE	NAME	ACCESS CODE	TYPE	SITE
Analog Loop Start	9	Analog Loop Start	Headquarters				
Digital Loop Start	9	Digital Loop Start	Headquarters				
Digital Wink Start	9	Digital Wink Start	Headquarters	Digital Wink Start	9	Digital Wink Start	Headquarters

- 2 In the **Name** field, enter a User Group name, for example *Voicemail Users*.
- 3 In the **COS - Telephony** field, select **Fully Featured** from the list.
- 4 In the **COS - Call Permissions** field, select **No Restrictions** from the list.
- 5 In the **COS - Voice Mail** field, select **Large Mailbox** from the list.
- 6 Select the **Send caller ID as caller's emergency identification (CESID)** checkbox.
- 7 Select the **Send DID as caller's emergency identification (CESID)** checkbox.
- 8 In the **Account code collection mode** field, select **None**.
- 9 Select the **Show Mitel Connect client users a list of account codes when dialing** checkbox.
- 10 In the **Voice mail interface mode** field, select **External Voice Mail, SIP** from the list.
- 11 In the **Outgoing trunk groups (Access Code)** field, select **Digital Wink Start**.

NOTE The remaining fields can retain their default settings.

12 Click **Save**.

Configuring Subscriber Telephone Extensions

Configure the subscriber (users) telephone extensions for use with MiCollab AM.

To configure Subscriber Telephone Extensions:

- 1 Using the Mitel Connect Director program, go to **Administration > Users**, and then select a user.

Users

Extension 2101: Sally Jones [View Escalation Profile](#) [View Programmable Buttons](#)

GENERAL TELEPHONY VOICE MAIL ROUTING MEMBERSHIP APPLICATIONS DNIS

First name: Sally Last name: Jones

Extension: 2101 [SHOW REFERENCES](#)

Email address: sjones [Edit System Directory record](#)

Client username: SJones

☒ Include in System Dial by Name directory

☐ Make extension private

DID Settings: (not configured) [change settings...](#)

PSTN failover: None

Caller ID (overwrite DID): (e.g. +1 (408) 331-3300)

License type: Extension-Only

Access license: Connect Client

User group: Voicemail Users [Go to this user group](#)

Site: Headquarters [Go to this site](#)

Language: English(US)

Primary phone port: SoftSwitch: Headquarters [change settings...](#)

Current port: Headquarters [GO PRIMARY PHONE](#)

Jack #:

Mailbox server: Voicemail

Client password: (6 - 26 characters)

☒ must change on next login

SIP phone password: (6 - 26 characters)

Note:

- 2 In the **License type** field, select **Extension-Only**.
- 3 In the **User group** field, select **Voicemail Users**, or the name used when you created the User Group.
- 4 In the **Mailbox server** field on the **General** properties sheet of each subscriber (user) extension, select the SIP Server you created in the procedure, [Creating the SIP Server for MiCollab AM](#) from the list.
- 5 Click **Save**.
- 6 Repeat this procedure for each telephone extension that is used by a MiCollab AM subscriber.

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:

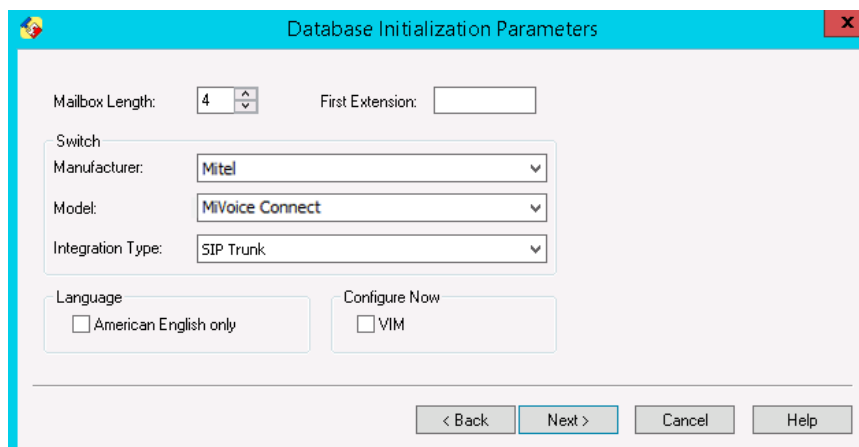
- [Configuring MiCollab AM for the Integration During Initial Installation](#): Integrate the telephone system while you install MiCollab AM for the first time.
- [Configuring Existing MiCollab AM for the Integration](#): Integrate a new telephone system on your existing MiCollab AM system.

NOTE For general information on integrations, refer to the **Integrating MiCollab AM with the Telephone System** chapter in the *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 with the integration during the initial installation:

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



Database Initialization Parameters

Mailbox Length: 4 First Extension:

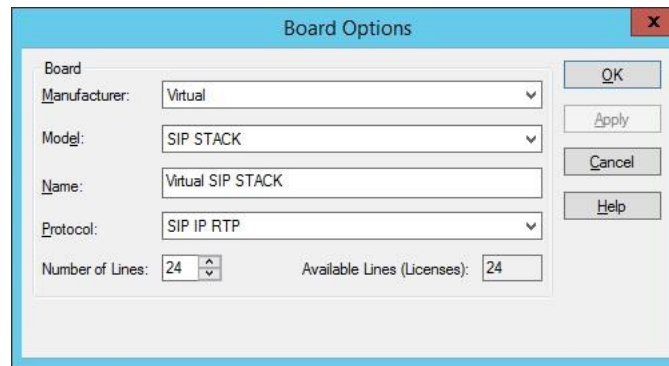
Switch
Manufacturer: Mitel
Model: MiVoice Connect
Integration Type: SIP Trunk

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** drop-down list, select **Mitel**.
- d From the **Model** drop-down list, select **MiVoice Connect**.
- e From the **Integration Type** drop-down list, select **SIP Trunk**.

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



- 3 In the **Board Options** dialog box, configure the following options:
- a From the **Manufacturer** drop-down list, select **Virtual**.
 - b From the **Model** drop-down 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** drop-down 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 appears.

Switch Options

Manufacturer:

Model:

System Switch:

System Switch Settings

Switch Name:

Transfer Support: ☒ Extension to Extension ☒ Trunk to Extension
☐ Extension to Trunk ☐ Trunk to Trunk

MWI Settings

Refresh Trigger: Refresh Type:

Refresh Interval: Initialize Mode:

Refresh Time of Day: Set Preference:

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"/>

Local Switch Settings

View:

Name	Value
Disconnect Loop Current Length (ms)	450
Flash Hook Time (ms)	500
T1 Protocol	FXS
T1 Signaling	Wink

- 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 *System Installation and Configuration Guide*.

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

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 following settings:

Table 4. Required Parameters View – Integration Options

Field	Value
SIP Server Address	<p>Enter the Virtual IP Address (VIP) configured on the MiVoice Connect switch.</p> <p>This value must match the value entered in the Creating a Virtual IP Address (VIP) for SIP Proxy Redundancy section.</p> <p>If a VIP is not created, enter the FQDN or IP Address of MiVoice Connect PBX.</p> <p>This address must match the value selected in the Creating the SIP Server for MiCollab AM section.</p>
SIP Server Port	<p>Enter the listening port of the MiVoice Connect PBX.</p> <p>The default value is 5060.</p> <p>This port number must match the port number configured on the MiVoice Connect PBX.</p>
Transport for outgoing SIP messages	<p>Select the transport protocol used for sending out SIP messages. This value must match the value selected in the Creating the SIP Server for MiCollab AM section.</p>
SIP Device Name	<p>Enter the MiCollab AM extension number you configured on the MiVoice Connect PBX in the Creating the SIP Server for MiCollab AM section.</p>

	This number should match the extension number programmed on MiVoice Connect for the MiCollab AM ports.
PBX Registration Password	Enter the password configured for the Extension number on the MiVoice Connect. In this example, no password is configured (blank).
Local IP Address to bind on	Select the Call Server's IP Address for this integration.
SIP Local Connection Port	Enter the local port on which SIP messages are expected. 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. <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. The hunt number must be unique across all IP integrations.</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>
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 .

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

Switch Section Options

Local Switch: Mitel MiVoice Connect

System Switch Section: - Create New -

System Switch Section Settings

Name: Mitel MiVoice Connect Section

Assigned to Tenant: DOCTEST System

Node Code:

Location Code:

Location: DOCTEST

MWI Integration: Mitel MiVoice Connect SIP Trunk

Local Switch Section Settings

View: Required Parameters

Name	Value
Incoming Hunt Mode	Terminal
Hunt Group Access Code	

- 9 In the **Switch Section Options** dialog box, configure the following options.
 - a In the **Local Switch Settings** section, select **Required Parameters** view.
 - b In the **Incoming Hunt Mode** field, select **Terminal**.
 - c In the **Hunt Group Access Code** field, enter the extension number that you programmed previously in the [Creating the SIP Server for MiCollab AM](#) section.
 - d Click **OK**.
- 10 Continue through and complete the configuration. At the end of the configuration, a confirmation dialog box appears. 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, enter the extension number of each integrated line on the Call Server.
- 13 Click **OK** to save all changes.

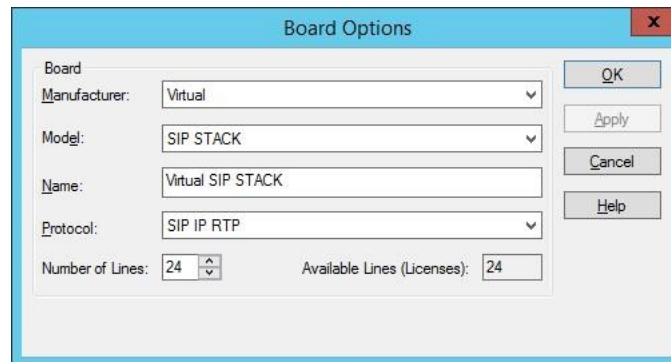
Configuring Existing MiCollab AM for the Integration

To configure existing MiCollab AM for the telephone integration:

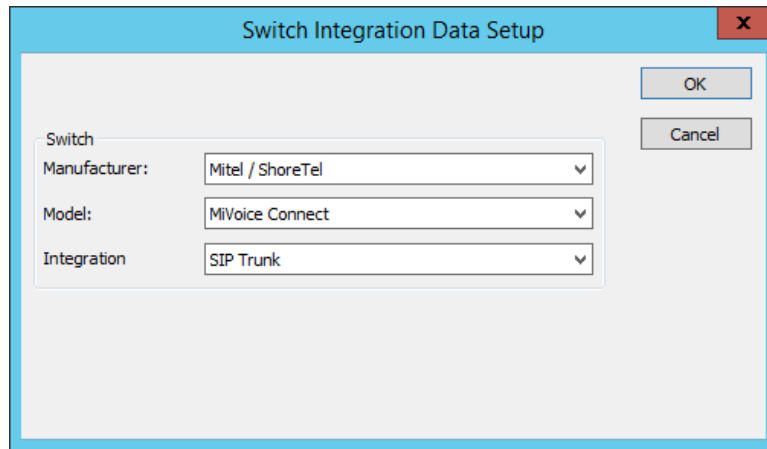
- 1 Open **MiCollab AM Configuration**, and go to the **Main** tab.
- 2 In the **Main** tab, click **Shutdown** to stop the system. Wait until the **Current Status** shows **Stopped**.

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**.

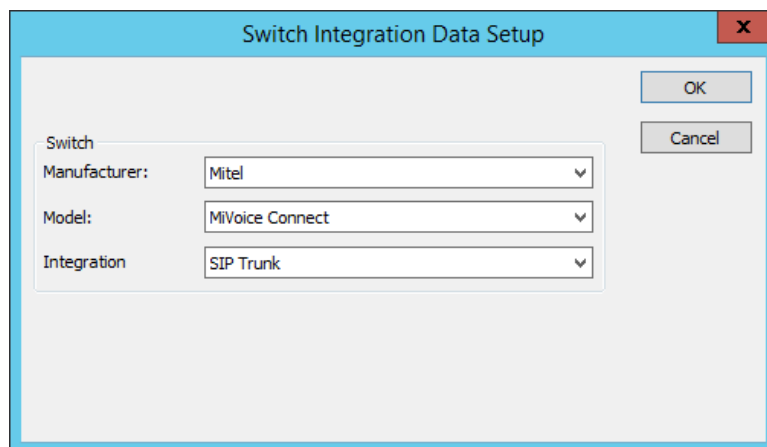
- 3 **[Optional]** Select the **Boards** tab, and then click the **Add** button. The **Board Options** dialog box appears.



- a From the **Manufacturer** drop-down list, select **Virtual**.
 - b From the **Model** drop-down 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** drop-down 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.
- 5 If you have a ShoreTel Connect Onsite PBX:
 - a From the **Manufacturer** drop-down list, select **Mitel/ShoreTel**.
 - b From the **Model** drop-down list, select **MiVoice Connect**.
 - c From the **Integration Type** drop-down list, select **SIP Trunk**.



- 6 If you have a Mitel MiVoice Connect PBX:
- a From the **Manufacturer** drop-down list, select **Mitel**.
 - b From the **Model** drop-down list, select **MiVoice Connect**.
 - c From the **Integration Type** drop-down list, select **SIP Trunk**.



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

Switch Options

Manufacturer: Mitel / ShoreTel
Model: MiVoice Connect
System Switch: - Create New -

System Switch Settings
Switch Name: Mitel / ShoreTel MiVoice Connect

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"/>

Local Switch Settings
View: All Settings Set Defaults

Name	Value
Disconnect Loop Current Length (ms)	450
Flash Hook Time (ms)	500
T1 Protocol	FXS
T1 Signaling	Wink

Switch Options

Manufacturer: Mitel
Model: MiVoice Connect
System Switch: - Create New -

System Switch Settings
Switch Name: Mitel MiVoice Connect

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"/>

Local Switch Settings
View: All Settings Set Defaults

Name	Value
Disconnect Loop Current Length (ms)	450
Flash Hook Time (ms)	500
T1 Protocol	FXS
T1 Signaling	Wink

OR

- 8 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*.

- 9 Click **OK**. The **Integration Options** dialog box appears.

Integration Options

System Switch: Mitel / ShoreTel MiVoice Connect
Integration Type: SIP Trunk
Integration: - Create New -
Name: Mitel / ShoreTel MiVoice Connect SIP Trunk

Local Integration Settings
View: Required Parameters Set Defaults

Name	Value
SIP Server Address	
SIP Server Port	5060
Transport for outgoing SIP messages	TCP
SIP Device Name	
PBX Registration password	- Please Select -
Local IP Address to bind on	
SIP Local Connection Port	5060
SIP parser qualifier string	

Integration Options

System Switch: Mitel MiVoice Connect
Integration Type: SIP Trunk
Integration: - Create New -
Name: Mitel MiVoice Connect SIP Trunk

Local Integration Settings
View: Required Parameters Set Defaults

Name	Value
SIP Server Address	
SIP Server Port	5060
Transport for outgoing SIP messages	TCP
SIP Device Name	
PBX Registration password	- Please Select -
Local IP Address to bind on	
SIP Local Connection Port	5060
SIP parser qualifier string	

OR

10 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 following settings:

Table 5. Required Parameters View – Integration Options

Field	Value
SIP Server Address	<p>Enter the Virtual IP Address (VIP) configured on the MiVoice Connect switch.</p> <p>This value must match the value entered in the Creating a Virtual IP Address (VIP) for SIP Proxy Redundancy section.</p> <p>If a VIP is not created, enter the FQDN or IP Address of MiVoice Connect PBX.</p> <p>This address must match the value selected in the Creating the SIP Server for MiCollab AM section.</p>
SIP Server Port	<p>Enter the listening port of the MiVoice Connect PBX.</p> <p>The default value is 5060.</p> <p>This port number must match the port number configured on the MiVoice Connect PBX.</p>
Transport for outgoing SIP messages	<p>Select TCP from the list. This value must match the value selected in the Creating the SIP Server for MiCollab AM section.</p>
SIP Device Name	<p>Enter the MiCollab AM extension number you configured on the MiVoice Connect PBX in the Creating the SIP Server for MiCollab AM section.</p> <p>This number should match the extension number programmed on MiVoice Connect for the MiCollab AM ports.</p>
PBX Registration Password	<p>Enter the password configured for the Extension number on the MiVoice Connect.</p> <p>In this example, no password is configured (blank).</p>
Local IP Address to bind on	<p>Select the Call Server's IP Address for this integration.</p>
SIP Local Connection Port	<p>Enter the local port on which SIP messages are expected.</p> <p>The default value is 5060.</p>
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** Click **OK**. The **Switch Section Options** dialog box appears.

Switch Section Options

Local Switch: Mitel / ShoreTel MiVoice Connect

System Switch Section: - Create New -

System Switch Section Settings

Name: Mitel / ShoreTel MiVoice Connect Section

Assigned to Tenant: DOCTEST System

Node Code:

Location Code:

Location: DOCTEST

MWI Integration: Mitel / ShoreTel MiVoice Connect SIP Trunk

Local Switch Section Settings

View: Required Parameters

Set Defaults

Name	Value
Incoming Hunt Mode	Terminal
Hunt Group Access Code	

OR

Switch Section Options

Local Switch: Mitel MiVoice Connect

System Switch Section: - Create New -

System Switch Section Settings

Name: Mitel MiVoice Connect Section

Assigned to Tenant: DOCTEST System

Node Code:

Location Code:

Location: DOCTEST

MWI Integration: Mitel MiVoice Connect SIP Trunk

Local Switch Section Settings

View: Required Parameters

Set Defaults

Name	Value
Incoming Hunt Mode	Terminal
Hunt Group Access Code	

- 11** In the **Switch Section Options** dialog box, configure the following options.

- In the **Local Switch Settings** section, select **Required Parameters** view.
- In the **Incoming Hunt Mode** field, select **Terminal**.
- In the **Hunt Group Access Code** field, enter the extension number that you programmed previously in the [Creating the SIP Server for MiCollab AM](#) section.
- Click **OK**.

- 12 In **MiCollab AM Configuration**, verify that the telephone system is properly added and configured in the **Switches**, **Switch Sections**, and **Integrations** tabs.
- 13 Select the **Lines** tab.
- 14 In the table from the **Lines** tab, enter the extension number of each integrated line on the Call Server.
- 15 Click **OK** to save all changes.

Configuring MiCollab AM for SIP Failover

MiCollab AM can be configured for automatic failover to the secondary SIP server in the event of the primary/host SIP server failure. Use the instructions provided in this section to add or remove secondary SIP server(s) for failover.

To add a SIP failover server:

- 1 From **MiCollab AM Configuration**, click the **Integrations** tab.
- 2 From the **Integrations** list, select your integration, and then click **Edit**.
- 3 In the **Integration Options** dialog box, go to the **Local Integration Settings** section.
- 4 From the **View** drop-down list, select **Failover Server Settings**.
- 5 Click the **Add Failover Server** button. Two new rows are added to configure the secondary SIP server.
- 6 In the **Secondary SIP Server Address** and **Secondary SIP Server Port** rows, enter the appropriate value as follows:

Table 6. Secondary SIP Server Address and the Secondary SIP Server Port example

Field	Value
Secondary SIP Server Address	<p>Enter the TCP/IP address or an FQDN of the secondary node.</p> <p>For example: The IP address 123.45.6.789 as displayed on the Review/Modify SIP Gateway screen.</p> <p>NOTE This integration requires the machine name to be a fully qualified domain name. Therefore, use the Machine Name field as displayed on the Review/Modify SIP Gateway screen during the integration process.</p> <p>IMPORTANT This value must match the configuration on the Gateway of the secondary node.</p>
Secondary SIP Server Port	<p>Enter the port number of the secondary node. The default value is 5060.</p>

- 7 From the **View** drop-down list, select **Integration Specific Parameters**. The **Integration Specific Parameters** view appears.
- 8 In the **Integration Specific Parameters** list, enter the information as shown in the following table:

NOTE The parameters in the following table is listed in alphabetical order. The actual Integration Specific Parameters on your system may not be listed in the same order presented in the table below.

Table 7. Integration Specific Parameters

Field	Value
Enable SIP server failover	Select this check box to allow for failover and to enable the failover server setting changes.
Delay (in ms) between Failover attempts	The delay in milliseconds before MiCollab AM attempts to register its port with the SIP server. The default is 1000 ms.
Incoming off hook delay	800
Outgoing off hook delay	0
On hook delay	300
Type of Call Progress to use for External Calls	<p>How this should be set depends on the gateway used for the integration.</p> <ul style="list-style-type: none">• If the gateway supports call progress through to the endpoint, set to Digital.• If the gateway reports early that the call is connected, such as before the phone rings or while the phone is ringing, set to Media.

- 9 Click **Apply** to save the changes.
- 10 To add another failover server repeat **Steps 4-9**.
- 11 Click **OK** to close the **Integration Options** dialog box.

To remove a SIP Failover Server:

- 1 From **MiCollab AM Configuration**, click the **Integrations** tab.
- 2 From the **Integrations** list, select your integration, and then click **Edit**.
- 3 In the **Integration Options** dialog box, go to the **Local Integration Settings** section.
- 4 From the **View** drop-down list, select **Failover Server Settings**.
- 5 In the **Failover Server Settings** view, click the **Remove Failover Server** button.
- 6 At the confirmation prompt, click **Yes** to confirm the deletion.

NOTE If multiple servers are listed, the last server address and port pair on the list is deleted first.

- 7 Click **Apply** to save the changes, and then click **OK** to close the **Integration Options** dialog box.

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 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**.

Windows Server 2012 R2

To change the binding order of multiple NICs:

- 1 From the taskbar, click **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 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**.

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 8. 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