

# MiCollab Advanced Messaging Siemens® OpenScape® Voice SIP Trunk Integration Technical Notes

For version 9.0 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 Siemens OpenScape Voice telephone system.

This document describes how to integrate MiCollab AM with a Siemens® OpenScape Voice telephone system, 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, but passes voice communication and signaling information over the network.

MiCollab AM uses SIP Trunks to integrate with OpenScape Voice. The PBX routes calls to MiCollab AM using a static SIP Endpoint which is associated with a Route pointing towards MiCollab AM. Message Waiting Indicator (MWI) operations are performed using unsolicited NOTIFY requests.

This ITN documents the procedure for setting up the integration. The process consists of programming the telephone system and configuring MiCollab AM. Critical application considerations are also documented.

## References

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

## Documentation

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

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

certified MiCollab AM technicians who are experienced with MiCollab AM and familiar with the procedures and terminology.

- **Software Release Notice (SRN).** This notice introduces the new features, capabilities, and hardware/software requirements for the corresponding MiCollab AM version.

## Documentation Updates

Documentation updates may be available from the following sources:

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

## Help

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

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

## Document Conventions

The following conventions are used in this document:

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

Example: **Enter**

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

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

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

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

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

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

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

Example: Type the password *voicemail*.

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

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

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

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

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

## Features Supported by this Integration

The following tables list the features supported using the OpenScape Voice integration.

Table 1. 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	No

Table 2. Integration features supported for Siemens OpenScape Voice 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 1, Note 2
DNIS	Yes	
End-to-end DTMF, attendant console	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, inbound/outbound	Outbound	
Networking, analog	No	
Overflow from MiCollab AM to attendant	Yes	
Overflow to MiCollab AM from attendant	Yes	
PBX-provided disconnect signaling	Yes	
Revert to operator	Yes	
Transfers, blind	Yes	
Transfers, confirmed	Yes	Note 3
Transfers, fully supervised	Yes	Note 3
Transfers, monitored	Yes	
Trunk ID for call routing	No	

## NOTES

1. Only available 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 the [Critical Application Considerations](#) section of this document for limitations on these features.
3. Refer to the [Critical Application Considerations](#) section of this document for limitations on these features.
4. 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.

- You must configure the Incoming Hunt Mode in the **Switch Section Options** dialog box. This integration supports terminal, circular, reverse terminal and reverse circular hunt modes only. The default mode is Terminal.
- You must configure the Trunk Group Access Code in the **Switch Section Options** dialog box.
- If a caller is transferred into MiCollab AM and the transferring party completes the call transfer after MiCollab AM answers the call, the PBX does not generate a ring update. In this case, the envelope information (calling party information) will be incorrect. Note that for a call scenario where the call transfer is completed before the call is forwarded into MiCollab AM, this is not an issue.
- 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 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.
- 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.

- MiCollab AM 9.0 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.

# Installation Requirements

WinSCP (file transfer) and PuTTY (SSH utility) are open source programs that have been tested and approved for use with the messaging system.

These programs can be downloaded from the Internet or installed from the Installation disc.

## Telephone System

- Siemens OpenScape Voice Version 5 and above.

## MiCollab AM Requirements

- MiCollab AM software key diskette or feature file with the Siemens OpenScape Voice integration enabled and one Virtual SIP and RTP license enabled for each port involved in the integration
- One or two 10 MB, 100 MB, or 1000 MB (gigabit) network interface cards with cables

# 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.

## Programming the Endpoint Profile

To program the endpoint profile, open the Common Management Portal (CMP) and:

- 1 Go to **OpenScope Voice > Business Group > Profiles > Endpoint Profile**
- 2 Click **Add** on the Endpoint Profile.
- 3 Go to the **General** tab.

The screenshot shows a web browser window titled "[vmV41cluster] - Edit Endpoint Profile: avstepp". The browser address bar shows "https://10.153.111.10". The page has a header with a message: "Enter the profile data. Maximum number of allowed blocked number is 10." Below the header are three tabs: "General", "Endpoints", and "Services". The "General" tab is selected. The form contains the following fields and values:

- Name:** avstepp
- Remark:** (empty text area)
- Numbering Plan:** E164NANP

Below these fields is a section titled "Management Information" with a message: "Please enter the data for the following fields in the corresponding screens." This section contains the following fields and values:

- Class of Service:** (empty dropdown)
- Routing Area:** (empty dropdown)
- Calling Location:** (empty dropdown)
- Time Zone:** LOCAL
- SIP Privacy Support:** Basic
- Failed Calls Intercept Treatment:** Disabled

At the bottom right of the form are "Save" and "Cancel" buttons. The browser status bar at the bottom shows "Done" and "Internet".

- 4 Enter a name for the profile in the Name field.
- 5 Click ... to choose the entry for the **Numbering Plan** field.
- 6 Enter the default numbering plan for OpenScope Voice (usually E164NANP). Make sure that SIP Privacy Support is set to **Basic** and Failed Calls Intercept Treatment is set to **Disabled**.

- 7 Click **Save**.
- 8 Go to the **Services** tab.

The screenshot shows a web browser window titled "[vmV41cluster] - Edit Endpoint Profile: avstepp". The address bar shows "https://10.153.111.10 - [vmV41cluster]". The page has a tabbed interface with "General", "Endpoints", and "Services" tabs. The "Services" tab is active. It contains four settings, each with a radio button and a dropdown menu: "Voice mail:" (Yes), "Call Transfer:" (Yes), "Call Forward Invalid Destination:" (No), and "Toll and Call Restrictions:" (No). There is a "Save" button and a "Cancel" button at the bottom right. The status bar at the bottom shows "Done" and "Internet".

- 9 Set Voice Mail to **Yes**.
- 10 Set Call Transfer to **Yes**.
- 11 Click **Save**.

## Programming the Endpoint Settings

To program the endpoint settings, open the Common Management Portal (CMP) and:

- 1 Go to **OpenScape Voice > Business Group > Members > Endpoints**
- 2 Click **Add** on the Endpoint screen.
- 3 Select the **General** tab.

**[vmV41cluster] - Edit Endpoint : AVStep**

General SIP Attributes Aliases Routes Accounting

Endpoint

Define the connection data of an endpoint, e.g. you may use this to add a gateway to a switch.

Name: AVStep

Remark:

Registered: ☒

Profile: avstepp

Associated Endpoint:

Default Home DN:

Location Domain:

Endpoint Template:

Endpoint Type:

Max number of users:

Save Cancel

- 4 Enter the name of the endpoint in the **Name** field.
- 5 Check the **Registered** check box.
- 6 Click **Save**.
- 7 Select the **SIP** tab.

**[vmV41cluster] - Edit Endpoint : AVStep**

General SIP Attributes Aliases Routes Accounting

SIP Private Networking: ☐

SIP Trunking: ☒

SIP-Q Signaling: ☐

SIP Signaling

For the static Endpoints the address of the SIP signaling interface can be specified in IP or FQDN format.  
Note that the address of the signaling interface cannot be modified unless the entry in the security section has first been removed.

Type: Static

Signaling Address Type: IP Address or FQDN

Endpoint Address: 172.31.119.50

Port: 5060

Transport protocol: TCP

Security

Set the Realm, Username and Password for digest authentication or configure the signaling address as a trusted one.

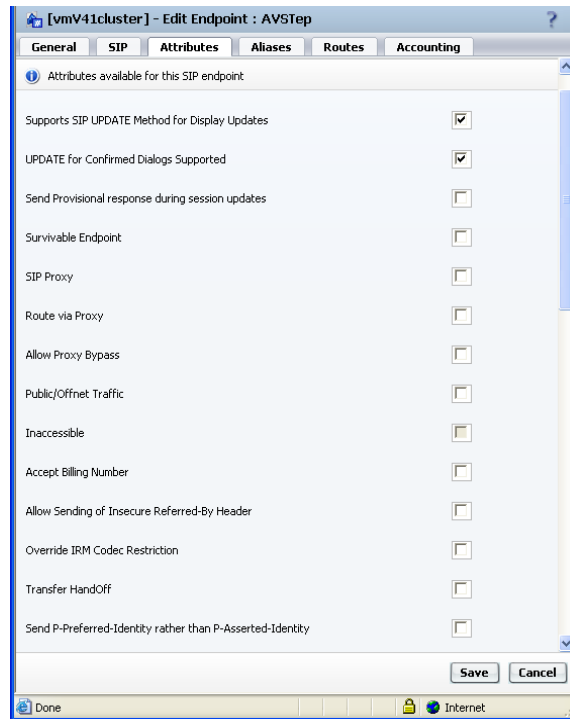
1 Item

Trusted

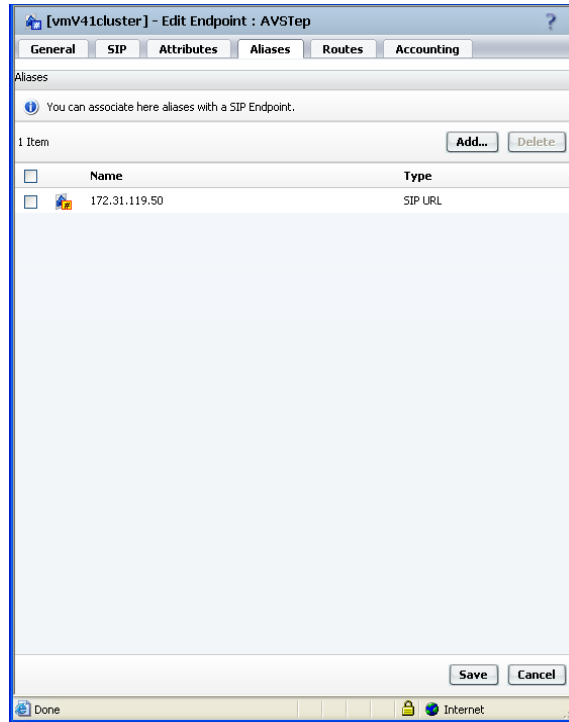
Save Cancel

- 8 Click the **SIP Trunking** radio button.

- 9 In the **SIP Signaling** section, set the **Type** to **Static**.
- 10 Set the **Signaling Address Type** to **IP Address or FQDN**.
- 11 Enter the IP address of the MiCollab AM server in the **Endpoint Address field**.
- 12 Enter the MiCollab AM server's port in the **Port** field.
- 13 Set the Transport protocol to **TCP**.
- 14 Make sure that the **Trusted** check box under **Security** is not selected.
- 15 Click **Save**.
- 16 Select the **Attributes** tab.

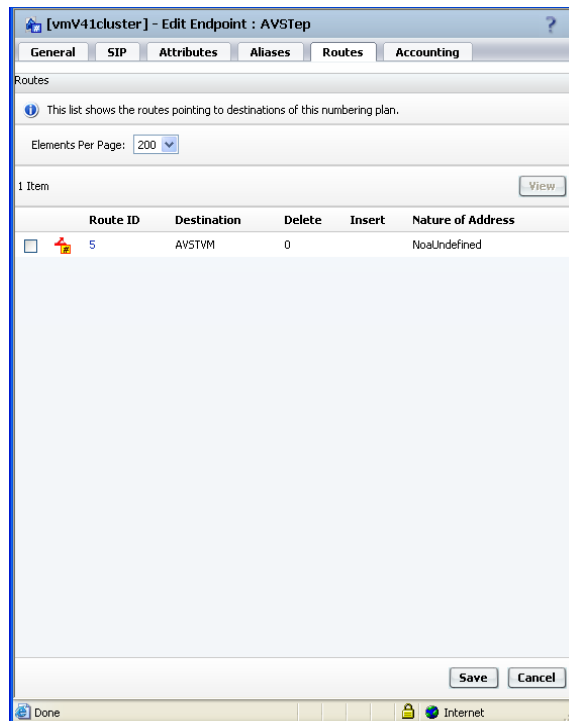


- 17 Select the **Supports SIP UPDATE Method for Display Updates** check box.
- 18 Select the **UPDATE for Confirmed Dialogs Supported** check box.
- 19 Scroll down and set **Support of Best Effort SRTP** to **Enabled**.
- 20 Scroll down and select the **Enable Session Timer** check box.
- 21 Make sure all other boxes are un-checked.
- 22 Click **Save**.
- 23 Select the **Aliases** tab.



24 Click **Add...** and add an alias for your MiCollab AM server.

25 Select the **Routes** tab.



26 Click **Add...** and add an alias for your MiCollab AM server.

27 Click **Save**.

## Programming the Destination

To program the endpoint profile:

- 1 Open the Common Management Portal (CMP).
- 2 Go to **OpenScape Voice > Business Group > Destinations and Routes > Destinations**.
- 3 In the **General** tab, give the destination a name.
- 4 Go to the **Routes** tab.

The screenshot shows a configuration window titled "[vmV41cluster] - View Route: R\_E523\_129". It contains several sections with informational icons and instructions:

- ID:** A text field containing the value "5".
- Type:** A dropdown menu set to "SIP Endpoint".
- SIP Endpoint:** A text field containing the value "AVSTep".
- Originator Attributes:** A section with a note: "Restricts the traffic according to specified settings. Routes with the same restrictions can be prioritized." It contains:
  - Signaling Type:** A dropdown menu set to "Undefined".
  - Bearer Capability:** A dropdown menu set to "Unassigned".
- Destination Directory Number:** A section with a note: "Number of digits to delete: Leading digits are cut off from the Directory Number. Digits to insert: the digit string is added to the beginning of the remaining digits." It contains:
  - Number of digits to delete:** A text field containing "0".
  - Digits to insert:** An empty text field.
  - Nature of Address:** A dropdown menu set to "Undefined".

An "OK" button is located at the bottom right of the window.

- 5 Enter the route ID in the **ID** field.
- 6 Enter the name of the SIP endpoint created above.
- 7 Click **Ok**.

## Configuring the Subscriber Settings

To test the subscriber settings, open the Common Management Portal (CMP) and:

- 1 Go to **OpenScape Voice > Business Group > Members > Subscribers**.
- 2 Select the **General** tab, define the Business Group name in the **Business Group** field.



[vmV41cluster] - [BG1] - [Main Office] - Edit Subscriber : 15616723182

Subscriber Description

General Displays Routing Connection Security Keyset Groups Features Applications

Subscriber Information

Business Group: BG1

Branch Office: ... Clear

Directory Number: 1 (561) 67-23182 ...

Type of Number: Public

Attendant Number: ☐

Localization

Time Zone: LOCAL ...

OpenScope Mobile

OpenScope Mobile Device: ☐

Main Device(ONS): ...

Mobile Device(Cellular): ... Clear

Ring Duration

Mobile(WIFI): ... second(s)

Main Device(ONS): ... second(s)

Save Cancel

- 3 Define the Directory Number by clicking the ... button.
- 4 Set the **Type of Number** as **Public**.
- 5 Click **Save**.
- 6 Select the **Displays** tab.

[vmV41cluster] - [BG1] - [Main Office] - Edit Subscriber : 15616723182

Subscriber Description

General Displays Routing Connection Security Keyset Groups Features Applications

Extension

This is the default extension number which is displayed for internal calls to or from this subscriber in case the Display Number Modification tables are not provisioned to return a number.

Displayed Extension Number: 23182

Special Identities

The External Caller ID, if provisioned, is the subscriber's identity which is used for all external calls.

External Caller ID:

Use Main Pilot DN as identity for external calls: ☐

Use Main Pilot DN as identity for internal calls: ☐

Display Information

Display Name: AVST 23182

Unicode Display Name:

External Display Name: AVST 23182

Unicode External Display Name:

Save Cancel

- 7 Enter a test **Displayed Extension Number** in the text box.
- 8 Enter any required further test information, including **Display Name** and **External Display Number** in the appropriate text box.
- 9 Click **Save**.
- 10 Select the **Connections** tab.

The screenshot shows a web-based configuration interface for a telephone system. The title bar reads "[vmV41cluster] - [BG1] - [Main Office] - Edit Subscriber : 15616723182". Below the title bar is a "Subscriber Description" section with a help icon. A tabbed interface is present with tabs for "General", "Displays", "Routing", "Connection", "Security", "Keyset", "Groups", "Features", and "Applications". The "Connection" tab is selected. Under "Connection Settings", there are several fields: "Connection Information:" with a dropdown menu set to "SIP"; "Type:" with a dropdown menu set to "Dynamic"; "Transport Protocol:" with a dropdown menu set to "TCP"; "IP Address:" with a text box containing "172.31.119.100" and a "Port:" text box containing "5060"; and "Associated Endpoint:" with an empty text box, a button with three dots, and a "Clear" button. Below the settings is an "Aliases" section with a message: "You can associate aliases with a subscriber." Below this message is a table with "0 Items" and buttons for "Add" and "Delete". The table has a header row with a checkbox and a "Name" column. At the bottom of the window are "Save" and "Cancel" buttons. The status bar at the very bottom shows "Done" and "Internet" icons.

- 11 Set the **Type** to **Dynamic**.
- 12 Set the **Transport Protocol** to **TCP**.
- 13 Click **Save**.
- 14 Select the **Features** tab.

[vmV41cluster] - [BG1] - [Main Office] - Edit Subscriber : 15616723182

Subscriber Description

General Displays Routing Connection Security Keyset Groups Features Applications

Feature Profile: FP\_BG1 ... Clear

Preserve Subscriber Settings: ☐

Subscriber Features

Feature Name: Please select a Feature... Add

Elements Per Page: 200

14 Items Edit... Delete

<input type="checkbox"/> Name	Active	Assignment
<input type="checkbox"/> Call Forwarding Dependable	●	Assigned
<input type="checkbox"/> Call Forwarding No Reply	●	Assigned
<input type="checkbox"/> Call Forwarding on Busy	●	Assigned
<input type="checkbox"/> Call Forwarding to Voice Mail	✓	Assigned
<input type="checkbox"/> Call Forwarding Unconditional	●	Assigned
<input type="checkbox"/> Call Transfer	✓	Inherited
<input type="checkbox"/> CSTA Access	✓	Inherited
<input type="checkbox"/> Do Not Disturb	●	Inherited
<input type="checkbox"/> Malicious Call Trace	✓	SwitchWide
<input type="checkbox"/> Music On Hold	✓	Assigned
<input type="checkbox"/> Name Permanent Presentation Status	✓	Inherited
<input type="checkbox"/> Number Permanent Presentation Status	✓	Inherited
<input type="checkbox"/> One Number Service	✓	Inherited
<input type="checkbox"/> Park to Server	✓	Inherited

Save Cancel

Done Internet

- 15 Enter a **Feature Profile** name in the text box.
- 16 Define the available features for the feature profile.
- 17 Click **Save**.
- 18 Test the 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: (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:

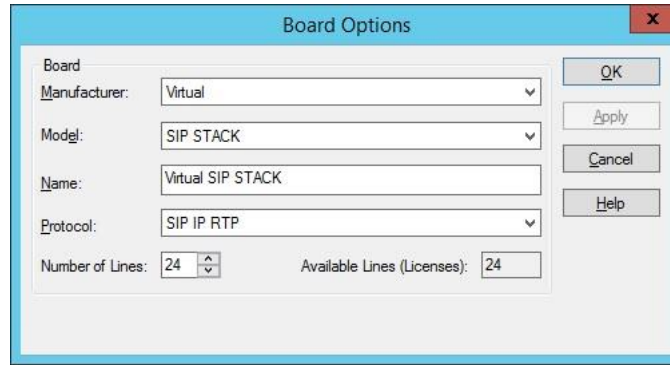
- [Configuring MiCollab AM for the Integration During Initial Installation](#): Integrate the telephone system while you install MiCollab AM for the first time.
- [Configuring Existing MiCollab AM for the Integration](#): Integrate a new telephone system on your exiting MiCollab AM system.

**NOTE** For general information on integrations, refer to the **Integrating MiCollab AM with the Telephone System** chapter in 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 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** drop-down list, select **Siemens**.
  - d From the **Model** drop-down list, select **OpenScape Voice**.
  - e From the **Integration Type** drop-down list, select **SIP Trunk**.
- 2 Click **Next**. The **Board Options** dialog box appears.



- 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.
- 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 options:

Table 3. Integrations Options-Required Parameters

Field	Value
SIP Server Address	Enter the Siemens Signaling Server IP address.
SIP Server Port	Enter the listen port configured on the Siemens Signaling Server. The default port number is <b>5060</b> .
Transport for outgoing SIP messages	Depending on the telephone system's configuration, enter UDP or TCP. This must match the one populated in

[Programming the Endpoint Settings](#) section for **Transport Protocol** field.

Local IP Address to bind on	Enter the IP address of the NIC on the MiCollab AM platform that connects to the Siemens Signaling Server and the Siemens Call Server. This IP address must match the one populated in <a href="#">Programming the Endpoint Settings</a> section for <b>Endpoint Address</b> field.
SIP Local Connection Port	Enter the TCP port MiCollab AM listens for incoming SIP messages. The default port is 5060. This port must match the one populated in <a href="#">Programming the Endpoint Settings</a> section for the <b>Port</b> field.
SIP parser qualifier string	<p>In cases of a 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.</p> <p>In cases where there are multiple SIP integrations on the call server, use a string that is unique to each SIP integration.</p> <p><b>For example:</b></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><b>NOTE</b> 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 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.
- Select the **Populate User-Agent Header** box.

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

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

- a** In the **Local Integration Settings** section, select the **Required Parameters** view.
- b** For the **Incoming Hunt Mode** value, select the mode for this integration.

**NOTE** This integration supports **terminal**, **circular**, **reverse terminal**, and **reverse circular** hunt modes only.

- c** In the **Hunt Group Access Code** field, enter the Trunk Group Access Code.
  - 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, configure callouts for the application. For information on configuring callout settings, see the topic *Configuring Callout Settings*, in the online help system.
  - 13** Click **OK** to save all changes.

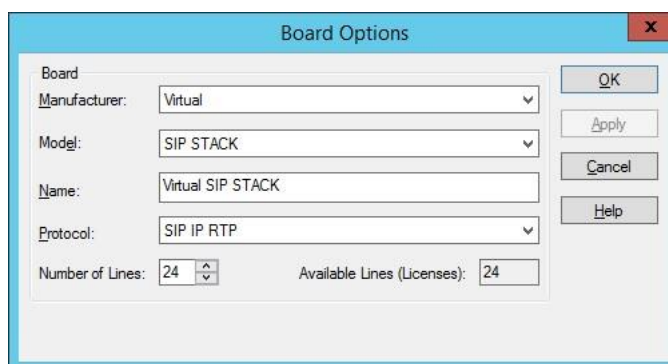
# Configuring Existing MiCollab AM for the Integration

To configure 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.
    - a From the **Manufacturer** drop-down list, select **Siemens**.
    - b From the **Model** drop-down list, select **OpenScape Voice**.
    - c From the **Integration Type** drop-down 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 following options:
  - a In the **Local Integration Settings** section, select the **Required Parameters** view, and configure the following options:

Table 4. Integrations Options-Required Parameters

Field	Value
SIP Server Address	Enter the Siemens Signaling Server IP address.
SIP Server Port	Enter the listen port configured on the Siemens Signaling Server. The default port number is 5060.
Transport for outgoing SIP messages	Depending on the telephone system's configuration, enter UDP or TCP. This must match the one populated in <a href="#">Programming the Endpoint Settings</a> section for <b>Transport Protocol</b> field.
Local IP Address to bind on	Enter the IP address of the NIC on the MiCollab AM platform that connects to the Siemens Signaling Server and the Siemens Call Server. This IP address must match the one populated in <a href="#">Programming the Endpoint Settings</a> section for <b>Endpoint Address</b> field.
SIP Local Connection Port	Enter the TCP port MiCollab AM listens for incoming SIP messages. The default port is 5060. This port must match the one populated in <a href="#">Programming the Endpoint Settings</a> section for the <b>Port</b> field.
SIP parser qualifier string	<p>In cases of a 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.</p> <p>In cases where there are multiple SIP integrations on the call server, use a string that is unique to each SIP integration.</p> <p><b>For example:</b></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>

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.

- 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.
  - Select the **Populate User-Agent Header** box.
- 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** For the **Incoming Hunt Mode** value, select the mode for this integration.

**NOTE** This integration supports **terminal**, **circular**, **reverse terminal**, and **reverse circular** hunt modes only.
  - c** In the **Hunt Group Access Code** field, enter the Trunk Group Access Code.
  - 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, configure callouts for the application. For information on configuring callout settings, see the topic *Configuring Callout Settings*, in the online help system.
- 14** Click **OK** to save all changes.

## Configuring 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 5. 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><b>For example:</b> The IP address 123.45.6.789 as displayed on the Review/Modify SIP Gateway screen.</p> <p><b>NOTE</b> 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><b>IMPORTANT</b> This value must match the configuration on the Gateway of the secondary node.</p>
Secondary SIP Server Port	Enter the port number of the secondary node. The default value is <b>5060</b> .

- 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 6. 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 <b>1000 ms</b> .

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"> <li>• If the gateway supports call progress through to the endpoint, set to <b>Digital</b>.</li> <li>• If the gateway reports early that the call is connected, such as before the phone rings or while the phone is ringing, set to <b>Media</b>.</li> </ul>

- 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

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

**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 2008 R2 with Service Pack 1

To change the binding order of multiple NICs:

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

## 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

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 <b>Server</b> 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