

MiCollab Advanced Messaging 9.3 MiVoice MX-ONE Service Node SIP Trunk Integration Technical Note

For version 9.3 and above

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Preface

This Integration Technical Note (ITN) is written for dealers who are experienced with MiCollab Advanced Messaging (MiCollab AM) and familiar with MiCollab AM procedures and terminology. It also assumes that you are familiar with the features and programming of MiVoice MX-ONE.

This document describes how to integrate MiCollab AM with a MiVoice MX-ONE system using the Service Node SIP trunk integration. This integration operates exclusively over a TCP/IP-based network; it does not use analog or digital voice telephony ports, but passes voice communication and signaling information over the network.

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 in this Integration

The following tables list the features supported using the MiVoice MX-ONE Service Node SIP integration.

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

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

Table 3. Integration features supported for MiVoice MX-ONE Service Node SIP

Feature	Supported	Notes
Automatic subscriber logon	Yes	
ANI/CLI	Yes	
<i>Announce Busy</i> greeting on forwarded calls	Yes	
Call screening	Yes	
Caller queuing	Yes	Note 1
DNIS	No	
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	Note 2
Live reply to sender	Yes	
Message notification callouts	Yes	
MWI, set/clear	Yes	
MWI, inband/outband	Outband	
Networking; AMIS, analog	Yes	
Overflow from MiCollab AM to attendant	Yes	
Overflow to MiCollab AM from attendant	Yes	
PBX-provided disconnect signaling	Yes	
Revert to operator from personal greeting	Yes	
Transfers, blind	Yes	
Transfers, confirmed	No	
Transfers, fully supervised	Yes	
Transfers, monitored	Yes	

Trunk ID for call routing	No	
Multiple Integrations	Yes	Note 3

NOTES

1. 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](#). Calls to MiCollab AM are not queued in the MiVoice MX-ONE Service Node.
2. Third-party conferences are not allowed on an integrated SIP-trunk.
3. 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.

MiCollab AM Application Considerations

- Configure the MiCollab AM **Incoming Hunt Mode** in the **Switch Section Options** dialog box. The hunt mode must match the type of hunting provided by the IP PBX. This helps to alleviate any *glare* conditions between the IP PBX and MiCollab AM. The default mode is Terminal.
- Configure the **Hunt Group Access Code** in the **Switch Section Options** dialog box.
- 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 Call Screening feature requires T-type supervised transfers. To use this feature without having to remove diversion programming from the subscriber telephone, set the TRAF parameter of the trunk category to restrict voice mail ports from calling other voice mail ports.
- The Windows quality of service (QoS) packet scheduler must be installed and operational on the network connection serving MiCollab AM and the telephone system. For more information about installing and configuring the QoS packet scheduler, refer to Windows Help.
- 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 one SIP trunk integration per Call Server. However, MiCollab AM supports up to 10 instances of the MiVoice MX-ONE Service Node SIP integration per Call Server.
- MiCollab AM supports G.729a with support for annex b on the incoming audio stream only. MiCollab AM does not transmit annex b packets.
- The SIP server address in the MiCollab AM **Integration Options** dialog box must match the IP Address configured in the telephone system.
- 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 supports up to 10 integration types (i.e., licensed integrations) in total per system. However, the following limitations apply to each Call Server:
 - MiCollab AM 9.3 and above:
 - 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 the Mitel TSW IP integrations are incompatible with other SIP integrations. Thus, it is not possible to have a Mitel TSW IP integration with any other SIP integration on the Call Server.
- When TLS is configured the certificates has to be configured in MiCollab AM and MiVoice MX-ONE Service Node.
- For TLS Endpoints/Terminals, a TFTP Server is required containing the root CA.pem certificate and modified startup.cfg file set to enable TLS.
- To create secure connections, use TLS 1.3 (recommended where available) or 1.2 for the System Server and Call Servers.
- When the SRTP is used on a SIP trunk to MiCollab AM in a system and terminals are configured for SRTP and when SRTP is enforced in MiCollab AM, calls from terminals with no encryption will fail.

Application Considerations for MiVoice MX-ONE Service Node

- The use of traffic-restricted voice mail ports is not compatible with blind transfers. We recommend that you use the monitored (Monitor) transfer type unless the application requires a supervised (T-type) transfer.
- When using reason code diversions from subscriber telephones, diverted calls will always go to the common diversion position. If MiCollab AM is the common diversion position, calls are always diverted there, even if individual diversions have been programmed to divert calls elsewhere.
- Because the telephone system performs the call progress detection in this integration and passes call progress as out-of-band events to MiCollab AM, MiCollab AM features that rely on analysis of the incoming audio stream do not function properly under this integration. These features include the following: detection of fax tone and call handling actions such as transfers and callouts to external telephone numbers.

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

- MiVoice MX-ONE with system software 7.3 or later.
- One or more MGU or Media Server.
- When TLS is used a certificate has to be configured in MiVoice MX-ONE Service Node, see command `mxone_certificate help`.
- For SIP endpoints to use TLS, a separate TFTP Server is required and must contain a copy of the CA.pem file, as well as a TLS-enabled configuration system file (startup.cfg).

MiCollab AM Requirements

- MiCollab AM version 9.3 or above.
- At least one 100 MB or 1000 MB network interface card and cable.
- Mitel feature file with the MiVoice MX-ONE Service Node SIP Trunk integration enabled and one Virtual SIP license enabled for each port involved in the integration.

Programming MiVoice MX-ONE

Follow the recommendations and programming examples in this section to program the MiVoice MX-ONE Service Node for integration with MiCollab AM. Programming examples show commands and parameters of MiVoice MX-ONE that are necessary for integration; they do not represent PBX programming in its entirety.

Depending on which version of the MiVoice MX-ONE Service Node software is running, the command format may be different. The format used in following examples are for MiVoice MX-ONE version 7.0.

The connection to MiCollab AM can be initiated as SIP trunk using either TCP or TLS, choose the appropriate section for the configuration.

The media encryption is optional and is only enabled if SRTP is required.

The installing technician should be familiar with programming the telephone system. For detailed programming information on this software version or other MiVoice MX-ONE Service Node software versions, refer to the appropriate ASB Basic Exchange and Extra Facility documentation and the MiVoice MX-ONE Service Node CPI documentation.

The following steps show how to create a new connection and establish proper routing to MiCollab AM.

A combination of Command Line Interface/Terminal emulation and Graphical User Interface steps are shown with examples where applicable.

The following examples show the command line interface procedure to add a number series or individual number.

Adding a Number Range/Series for the MX-ONE and Adding the Access Number for MiCollab AM

To add external destination numbers in Number Analysis for the MiCollab AM, use EX as the number type and choose a range of numbers that is appropriate for your numbering plan. Perform the following steps from a terminal emulation platform logged in with root permissions. (This can also be accomplished using the Graphical User Interface at the following location: Service Node Manager Menu: **Number Analysis > Number Series**.)

To add a number range/series for the MX-ONE and add the access number for MiCollab AM:

- 1 Log into the terminal emulation session.

The following prompt appears:

```
mxone_admin@MXONE63:~>
```

- 2 To access the root prompt, type **su -** and press **Enter**.

The prompt for the root password appears:

```
Password: 
```

- 3 Enter your root password.

The following prompt appears:

```
MXONE63:~ #
```

From here you can perform most system administration tasks. Some commands will be performed from the MDSH shell prompt identified later in this document.

- 4 To add an extension range, type the following:

```
MXONE63:~ # number_initiate -number 4140..4165 -numbertype ex
```

(Where number *4140..4165* is the range of numbers you are adding.

NOTE The *..* between the numbers is significant, indicating that you wish to create extensions that begin with the first number and end with the last number specified.

- 5 To make any group of numbers IP extensions, type the following command:

```
MXONE63:~ # ip_extension -i -d 4041..4049
```

- 6 To associate the extensions with a CSP (Common Service Profile) enter the following command:

```
MXONE63:~ # extension -i -d 4141..4165 --lim 1 --csp 50
```

This also can be performed on an individual extension as well. A group range assignment is shown.

- 7 To add a single number type the following command:

```
MXONE70:~ # number_initiate -number 2100 -numbertype ex
```

- 8 To set the number length of the external destination number to MiCollab AM, enter the following command

```
MXONE70:~ # number_data_initiate -externalnumber 2100 -minlength 4 -maxlength 4
```

NOTE This can also be accomplished using the GUI at the following location: Service Node Manager Menu: **Number Analysis > External Number Length**. This example is of extension number 2100 with a minimum and maximum length of 4 digits.

- 9 To define a priority-ordered list of the audio CODECs for the Servers via the GUI, go to Service Node Manager Menus: **Telephony > Telephony Domain**.
- 10 To set the CODEC List, enter the command: *ip_domain -c --domain-name DEFAULT --codec-priority-list PCMA*

NOTE The priority-order of the audio CODECS will be set to:
PCMA,G722,PCMU,G729AB,G729A,G723. Each codec value corresponds to its matching audio CODEC.

MiCollab AM supports **G.729a** with support for **annex b** on the incoming audio stream only. MiCollab AM does not transmit **annex b** packets.

Initiating the SIP Trunk Definition

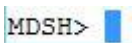
The following steps are performed from the MDSH command prompt in the Terminal Emulator. Apply the commands in the following procedure to configure SIP trunk connection to MiCollab AM using TLS. (Service Node Manager Menus: **Telephony** > **External Lines**.)

To initiate the SIP Trunk definition:

- 1 At the root prompt, enter the following command: *mdsh*



- 2 The prompt will change to:



- 3 From this prompt, you can initiate a route category for the route connection to MiCollab AM.

For example:

```
ROCAI:ROU=201,SEL=711000000000010,SIG=0111110000A0,TRAF=03151515,TRM=4,SERV=31  
10000001,BCAP=000100;
```

To verify your work, type the following command:

```
ROCAP:ROU=201;
```

- 4 Initiate a route data for the route connection to MiCollab AM.

For example:

```
RODAI:ROU=201,TYPE=TL66,VARC=00000000,VARI=00000000,VARO=00000000;
```

To verify your work, type the following command:

```
RODAP:ROU=201;
```

- 5 Initiate a route destination data for the route connection to MiCollab AM.

For example:

```
RODDI:ROU=201,DEST=75001,ADC=0005000000000250000001010000,SRT=1;
```

To verify your work, type the following command:

```
RODDP:DEST=75001;
```

- 6 To exit the MDSH screens, type **Exit** and then press **Enter**.

- 7 Initiate a SIP trunk profile for the connection to MiCollab AM. MiVoice MX-ONE Service Node SIP trunk is initiated using the profile **MiCollabAM_TCP** or **MiCollabAM_TLS_SRTP**.

The **tgrp=mxone-system** string can be changed and is used to identify the connected system and later entered as the SIP Parser Qualifier String in MiCollab AM configuration.

The **-mwnumber** parameter is the access number to MiCollab AM, will enable automatic DTMF on extensions when calling MiCollab AM.

The **-uristring0** parameter has to contain the IP of FQDN of MiCollab AM.

Parameter **-match** is the IP of MiCollab AM;

The following examples are from the Terminal Emulator at the root prompt:

For example: SIP trunk with TCP:

```
sip_route -set -route 201 -profile MiCollabAM_TCP -uristring0 "sip:?@mivoiceam.mx-one;tgrp=my-system" -accept REMOTE_IP -match 192.168.50.90 -mwinumber 75001
```

To verify your work, type the following command:

```
Sip_route -print -route 201
```

For example: SIP trunk with TLS:

```
sip_route -set -route 201 -profile MiCollabAM_TLS_SRTP -uristring0 "sip:?@mivoiceam.trust.mx-one;tgrp=mxone-system" -accept REMOTE_IP -match 192.168.50.90 -mwinumber 75001
```

To verify your work, type the following command:

```
Sip_route -print -route 201
```

Initiating the Number of Ports in MiVoice MX-ONE

To initiate the number of ports in MiVoice MX-ONE used for calls to MiCollab AM:

- 1 Go to the Service Node Manager Menus: **Telephony** > **External Lines**.
- 2 From the Terminal emulator MDSH submenu, enter the following command:

For example:

```
ROEQI:ROU=201,TRU=1-1&&1-10;
```

- 3 To verify your work, type the following command:

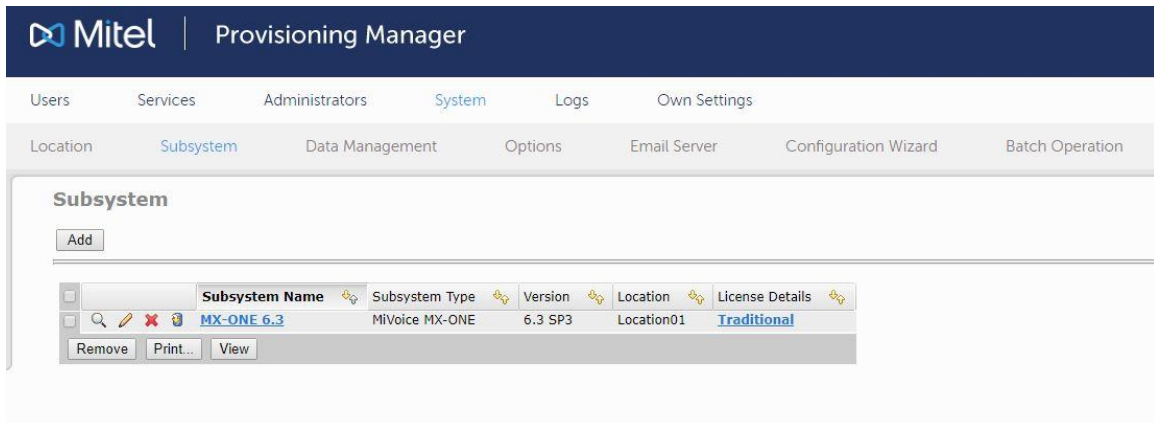
```
ROEDP:ROU=201,TRU=ALL;
```

NOTE The number of ports should be the same as the number of ports that can be connected on MiCollab AM. The example above initiates 10 ports for calls to MiCollab AM. Ports can be assigned in multiple MiVoice MX-ONE Service Nodes.

Connecting the MiVoice MX-ONE Graphical User Interface for to MiCollab AM

To connect the MiVoice MX-ONE GUI to MiCollab AM:

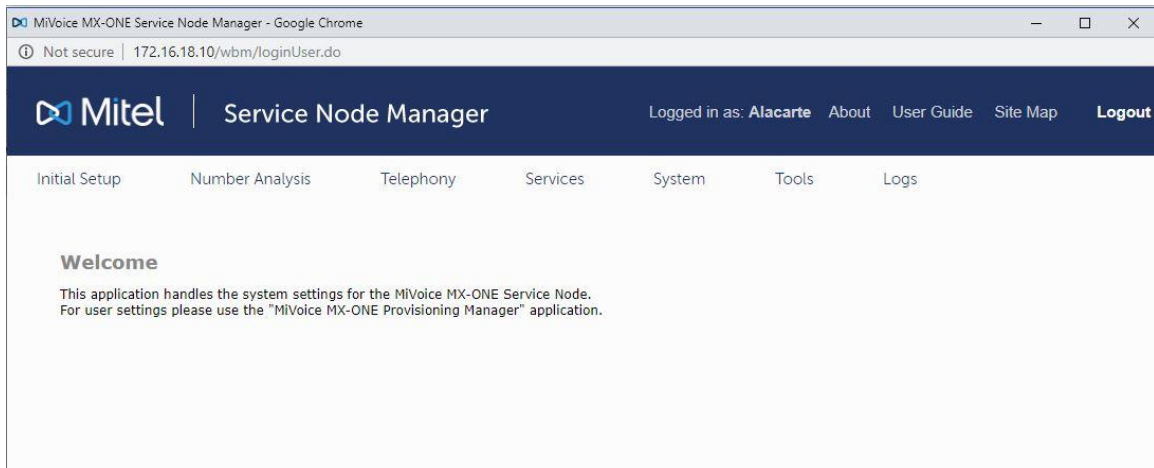
- 1 Point your browser to the IP Address assigned for your MX-ONE System and log in with the Administrator profile provided.



2 From the **Provisioning Manager** main menu, select the **System** tab, then select the **Subsystem** tab.

3 Select the **Subsystem** you want to connect to MiCollab AM (in this case **MX-ONE 6.3**).

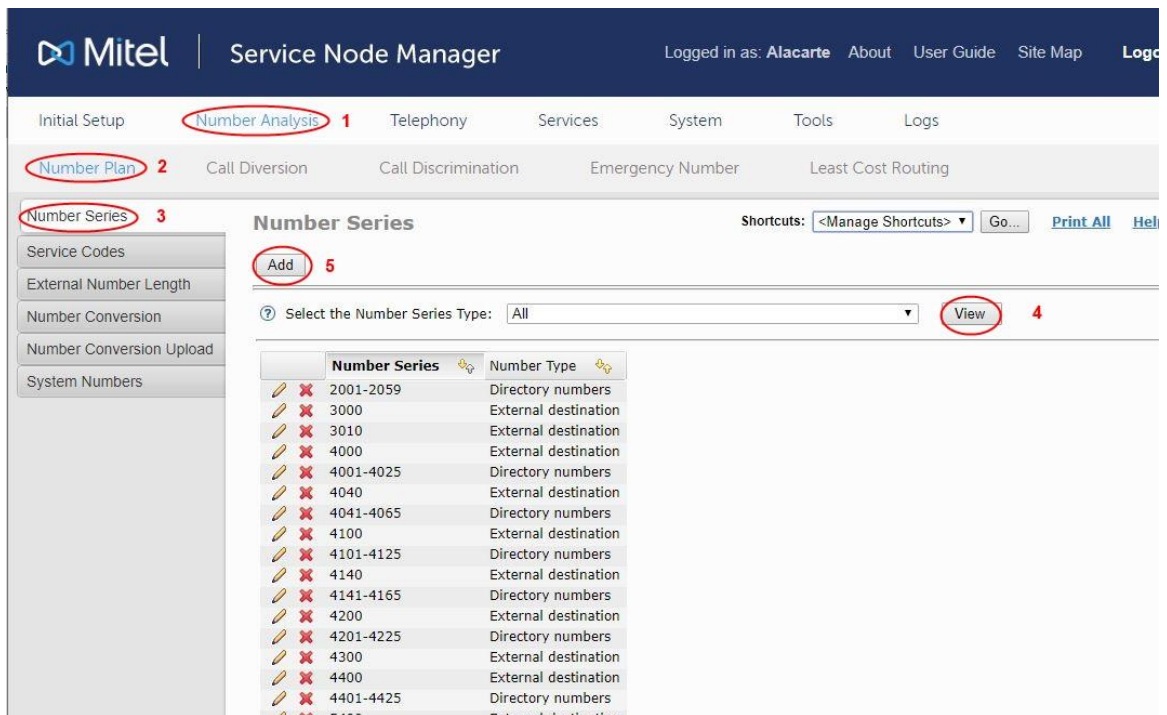
The **Service Node Manager** window will appear.



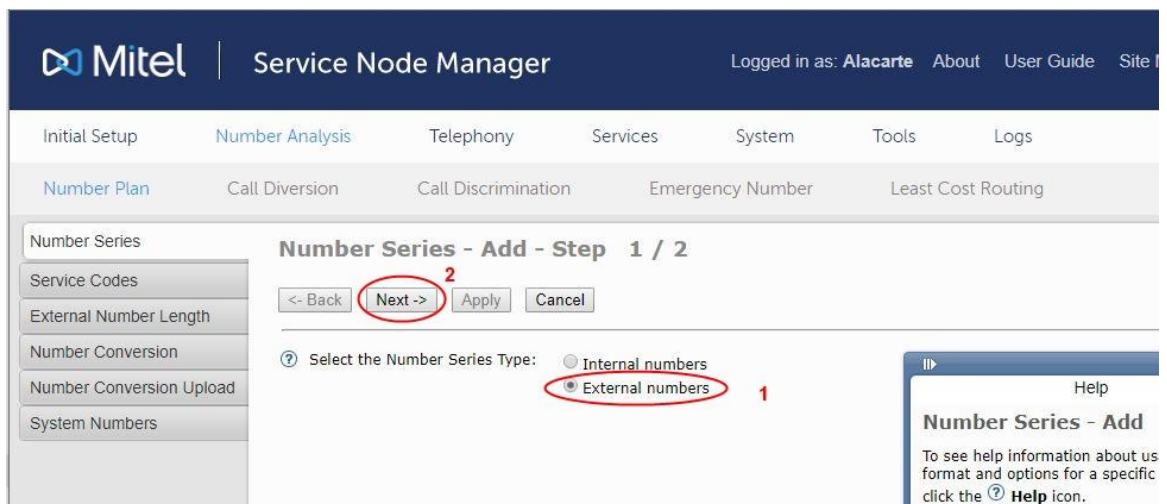
4 Click **Number Analysis**.

5 Click **Number Plan**, and then click **Number Series**.

6 Click **View** to see the existing configuration, and then click **Add**.



a On the **Number Series – Add – Step 1/2** page, select **External numbers**, and then click **Next**.



b On the **Number Series – Add – Step 1/2** page, enter the number you wish to use as your pilot number to MiCollab AM (i.e., 6000) in the **External Destination** field, and then click **Apply**.

Mitel | Service Node Manager

Logged in as: Alacarte

Initial Setup | Number Analysis | Telephony | Services | System | Tools

Number Plan | Call Diversion | Call Discrimination | Emergency Number | Least Cost Routing

Number Series

Service Codes

External Number Length

Number Conversion

Number Conversion Upload

System Numbers

Number Series - Add - Step 2 / 2

<- Back | Next -> | **Apply** | Cancel

External Number Series

- External Coordinated Destination:
- External Destination: **6000**
- Least Cost Routing Access Numbers:
- Common Direct In-Dialing Operator Numbers:
- Own Node Number:
- Common Public Directory Numbers:
- Access Numbers for Mobile Extension (without Authorization):
- Access Numbers for Mobile Extension (with Authorization):

The **Number Series – Add – Result** page appears.

c Click **Done**.

Mitel | Service Node Manager

Logged in as: Alacarte

Initial Setup | Number Analysis | Telephony | Services | System | Tools | Logs

Number Plan | Call Diversion | Call Discrimination | Emergency Number | Least Cost Routing

Number Series

Service Codes

External Number Length

Number Conversion

Number Conversion Upload

System Numbers

Number Series - Add - Result

Done

Add operation successful for:

- Number Series: 6000

Property	Value
External destination	6000

Add New... | Change This... | Remove This | Done

Now you will need to add the External Number Length for the newly created number (i.e., 6000).

- 7** Select **Number Analysis**, and then click **Number Plan**,
- 8** Select **External Number Length** tab, and then click **Add**.

External Number Length

Add 2

External Number	Minimum Length	Maximum Length
3000	4	4
3010	4	4
4000	4	4
4040	4	4
4100	4	4
4140	4	4
4200	4	4
4400	4	4
5400	4	4
5500	4	4

- a** Enter data in the **External Number**, **Minimum Length**, and **Maximum Length** fields, and then click **Apply**.

External Number Length - Add

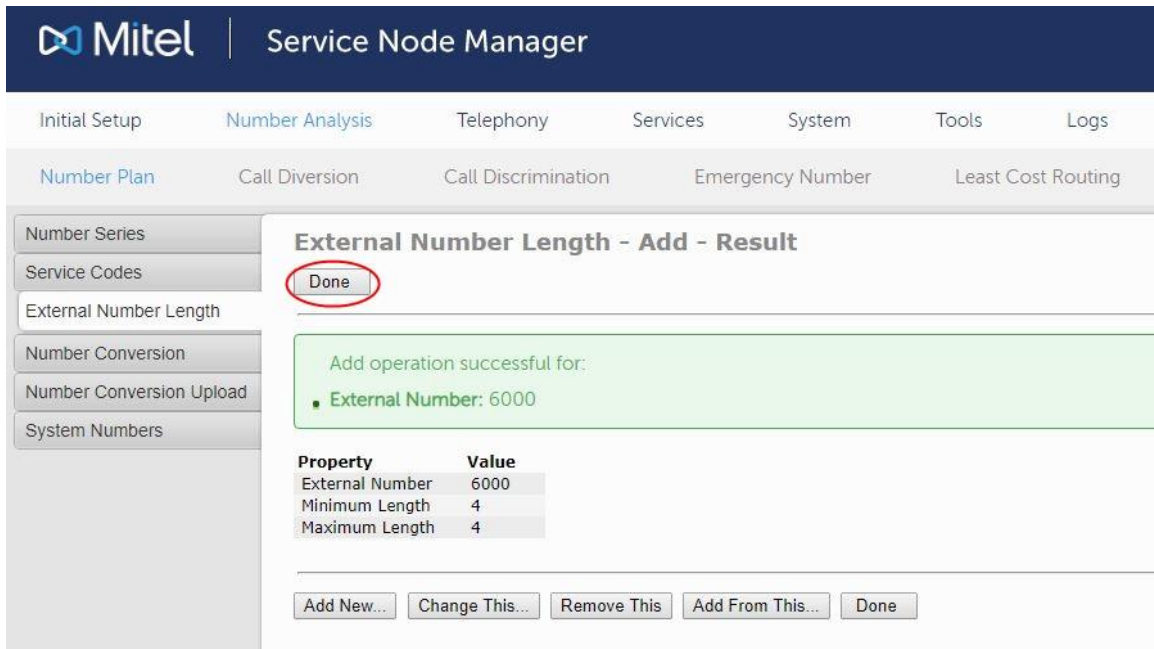
Apply **Cancel**

? External Number: * 6000

? Minimum Length: * 4

? Maximum Length: * 4

- b** The **External Number Length – Add - Result** page appears.

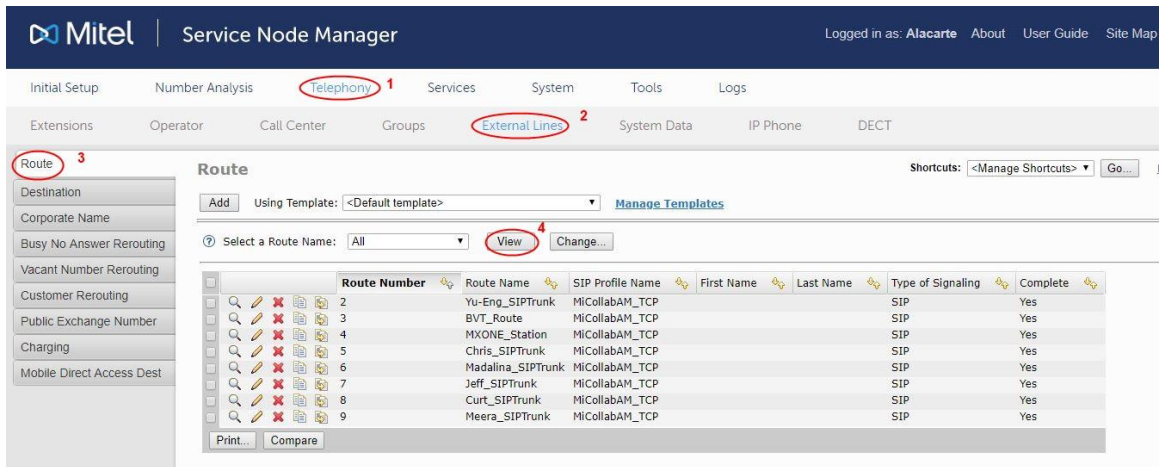


c Click **Done**.

The next step is to configure the SIP Trunk.

9 From the Service Node Manager (SNM), select **Telephony** > **External Lines** > **Route**.

10 Click **View**. You will see the list of the existing SIP Trunks defined in the system.



NOTE This is where you will need to go to modify the SIP Trunk for your MiCollab AM Integration.

You will be shown how to first add a SIP Trunk to the system, and then how to change an existing SIP Trunk to point to a different MiCollab AM Server.

11 From the **Route** tab, click **Add**.

a Set the **Type of Signaling** to **SIP** from the drop-down menu.

b Select **MiCollabAM_TCP** from the **Profile Name** drop-down menu.

c Click **Next**.

Mitel | Service Node Manager

Initial Setup | Number Analysis | **Telephony** | Services | System | Tools | Logs

Extensions | Operator | Call Center | Groups | **External Lines** | System Data | IP Ph

Route

Destination
Corporate Name
Busy No Answer Rerouting
Vacant Number Rerouting
Customer Rerouting
Public Exchange Number
Charging
Mobile Direct Access Dest

Route - Add - Step 1 / 4

<- Back **Next ->** Apply Cancel

? Type of Signaling: SIP
? Profile Name: MiCollabAM_TCP

d Enter a **Route Name** for your SIP Trunk (such as *SIP_Trunk6000*) and then select the **Route Number** (such as *10*).

e Click **Next**.

Mitel | Service Node Manager

Initial Setup | Number Analysis | **Telephony** | Services | System | Tools

Extensions | Operator | Call Center | Groups | **External Lines** | System Data

Route

Destination
Corporate Name
Busy No Answer Rerouting
Vacant Number Rerouting
Customer Rerouting
Public Exchange Number
Charging
Mobile Direct Access Dest

Route - Add - Step 2 / 4

General

<- Back **Next ->** Apply Cancel

? Route Name: * SIP_Trunk6000
? Route Number: 10

f On the **Route – Add – Step 3/4** page, specify the **Trunk Index**.

For the Trunk Index use the following format: insert a number **2** in front of the Route Number. This will make for a consistent deployment (i.e., for route 10 use 210).

g Click **Next**.

Mitel | Service Node Manager

Initial Setup Number Analysis **Telephony** Services System Tools

Extensions Operator Call Center Groups **External Lines** System Data

Route

Destination
Corporate Name
Busy No Answer Rerouting
Vacant Number Rerouting
Customer Rerouting
Public Exchange Number
Charging
Mobile Direct Access Dest

Route - Add - Step 3 / 4

Individuals

<- Back **Next ->** Apply Cancel

Server 1 Trunk Index 210

h Fill in the following fields: **Remote Proxy IP**, **VoiceMail Number**, and **trunk group name**.

The **Remote Proxy IP** will be your MiCollab AM Server's IP Address.

The **VoiceMail Number** will be the External Number you created earlier in this process.

The **trunk group name** is used to associate the trunk with the route.

i Click **Apply**.

Mitel | Service Node Manager

Initial Setup Number Analysis **Telephony** Services System Tools Logs

Extensions Operator Call Center Groups **External Lines** System Data IP Phone DECT

Route

Destination
Corporate Name
Busy No Answer Rerouting
Vacant Number Rerouting
Customer Rerouting
Public Exchange Number
Charging
Mobile Direct Access Dest

Route - Add - Step 4 / 4

Profile specific settings

<- Back Next -> **Apply** Cancel

Profile specific settings

Profile Name: MiCollabAM_TCP

Remote Proxy IP: 172.16.4.187

VoiceMail Number: 6000

trunk group name: SIPTrunk6000

Note: External Destination Number needs to initiate in the Number Analysis -> Number Series and it needs to be associated with the route in Telephony -> External Lines -> Destination.

j The **Route – Add - Result** page appears.

k Review the new Trunk values, and then click **Done**.

Mitel | Service Node Manager

Initial Setup | Number Analysis | **Telephony** | Services | System | Tools | Logs

Extensions | Operator | Call Center | Groups | **External Lines** | System Data | IP Ph

Route - Add - Result

Done

Add operation successful for:
 • **Route Name:** SIP_Trunk6000

General

Profile Name	MiCollabAM_TCP
Route Name	SIP_Trunk6000
Route Number	10

Individuals

Trunk Line Number	1-[210]
-------------------	---------

Profile specific settings

Remote Proxy IP	172.16.4.187
VoiceMail Number	6000
trunk group name	SIPTrunk6000

SIP Route Specific Data

The final step is to define the Destination.

- 12** From the Service Node Manager, select **Telephony** > **External Lines** > **Destination**, and then click **View**.

Mitel | Service Node Manager Logged in as: /

Initial Setup | Number Analysis | **Telephony** | Services | System | Tools | Logs

Extensions | Operator | Call Center | Groups | **External Lines** | System Data | IP Phone | DECT

Destination

Add Using Template: <Default template> [Manage Templates](#)

? Select Destination: All View

	Destination	Customer Name	Choice	Route Name	Fictitious Destination
<input type="checkbox"/>	3000			BVT_Route	No
<input type="checkbox"/>	3010			BVT_Route	No
<input type="checkbox"/>	4000			Yu-Eng_SIPTrunk	No
<input type="checkbox"/>	4040			Curt_SIPTrunk	No
<input type="checkbox"/>	4100			Madalina_SIPTrunk	No
<input type="checkbox"/>	4140			Meera_SIPTrunk	No
<input type="checkbox"/>	4200			Jeff_SIPTrunk	No
<input type="checkbox"/>	4400			Chris_SIPTrunk	No
<input type="checkbox"/>	5400			Chris_SIPTrunk	No
<input type="checkbox"/>	5500			MXONE_Station	No

Print... Compare

- 13** Click **Add**.

The screenshot shows the Mitel Service Node Manager web interface. The top navigation bar includes 'Initial Setup', 'Number Analysis', 'Telephony' (highlighted), 'Services', 'System', and 'Tools'. Below this, a secondary navigation bar shows 'Extensions', 'Operator', 'Call Center', 'Groups', 'External Lines' (highlighted), and 'System'. On the left, a sidebar menu lists various configuration options: 'Route', 'Destination', 'Corporate Name', 'Busy No Answer Rerouting', 'Vacant Number Rerouting', 'Customer Rerouting', 'Public Exchange Number', 'Charging', and 'Mobile Direct Access Dest'. The main content area is titled 'Destination - Add - Step 1 / 4' and 'Type of Destination'. It contains a horizontal row of buttons: '<- Back', 'Next ->', 'Apply', and 'Cancel'. Below the buttons, there is a section labeled '? Type of Destination:' with two radio button options: 'Destination' (which is selected) and 'Fictitious destination'.

14 Click **Next**.

- a** On the **Destination – Add – Step 2/4** page, for the **Destination:** select from the dropdown menu the newly created External Number (i.e., 6000-External)
- b** For the **Route Name:** select from the Dropdown menu the newly created SIP Trunk (i.e., SIP_Trunk6000).
- c** Click **Next**.

Service Node Manager

Initial Setup
Number Analysis
Telephony
Services
System

Extensions
Operator
Call Center
Groups
External Lines

Route
Destination
Corporate Name
Busy No Answer Rerouting
Vacant Number Rerouting
Customer Rerouting
Public Exchange Number
Charging
Mobile Direct Access Dest

Destination - Add - Step 2 / 4

Route Details

<- Back
Next ->
Apply
Cancel

Destination: 6000-External Edit...
Route Name: SIP_Trunk6000 View... Edit...
Customer Choice:

d On the **Destination – Add – Step 3/4** page, accept the default values and then click **Apply**.

Service Node Manager

Initial Setup
Number Analysis
Telephony
Services
System
Tools

Extensions
Operator
Call Center
Groups
External Lines
System Data

Route
Destination
Corporate Name
Busy No Answer Rerouting
Vacant Number Rerouting
Customer Rerouting
Public Exchange Number
Charging
Mobile Direct Access Dest

Destination - Add - Step 3 / 3

ADC Details

<- Back
Next ->
Apply
Cancel

Destination: 6000
Route Name: SIP_Trunk6000
Primary Choice is the sequence number for the route choice in alternative routing
Start Position for Digit Transmission: 1
Type of Seizure of External Line: Immediate seizure
Forward Switching:
Type of Called Number: Unknown public
Type of Calling Public Number: Unknown public
Type of Calling Private Number: Unknown private
Use as Emergency Destination:

Advanced...

- e Review the settings and then click **Done**.

Enabling Media Encryption and Secure Transport Methods (SRTP/TLS)

If media encryption and secure transport methods (SRTP/TLS) are used, they have to be enabled in the MiVoice MX-ONE.

To enable media encryption and secure transport methods (SRTP/TLS):

- 1 Enter the following commands on the Terminal Emulation Session with root permissions:

- a To enable encryption for extensions, enter the following command:

```
MXONE70:~ # media_encryption_enable -type extension
```

- b To enable encryption for trunks, enter the following command:

```
MXONE70:~ # media_encryption_enable -type trunk
```

- c To enable encryption for external routes, enter the following command:

```
MXONE70:~ # media_encryption_enable -type route
```

- d To display currently defined encryption values, enter the following command:

```
MXONE70:~ # media_encryption_print
```

Result of display request for encryption values is as follows:

```
Media encryption is enabled for extension
Media encryption is enabled for trunk
Media encryption is disabled for intermgw
```

To enable TLS on the Endpoint/Terminal:

- 1 Point your browser to one of the Endpoint/Terminal. It must be a valid Mitel Endpoint that is connected and registered to the MX-ONE. (i.e., 10.2.6.56)
- 2 Log in with the admin profile.

Sign in

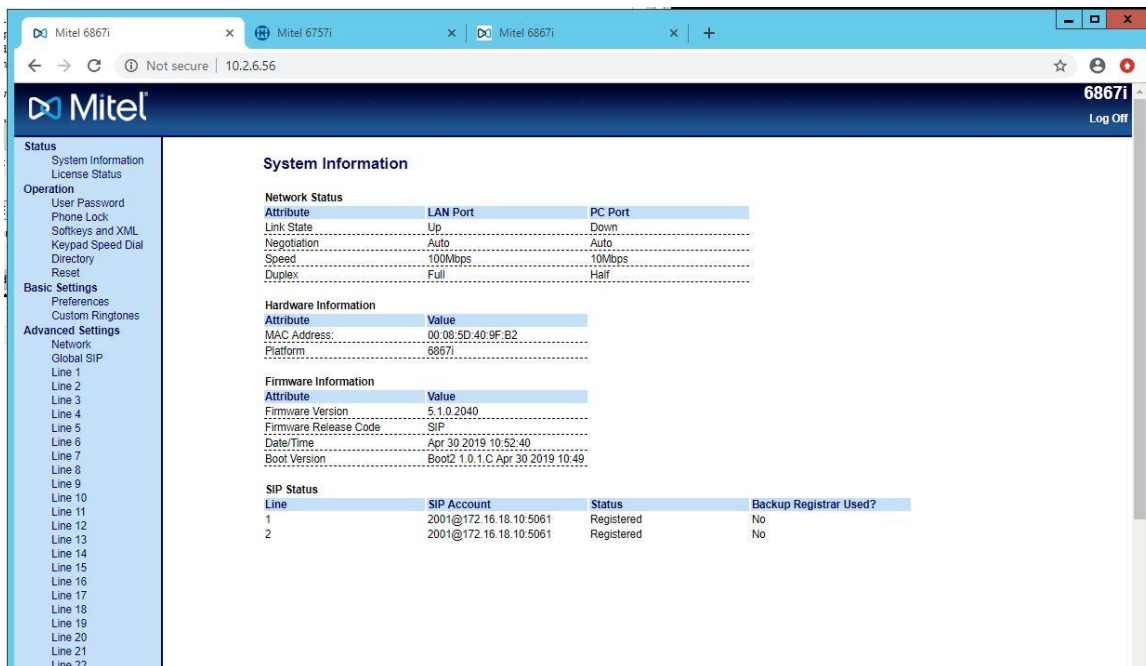
http://10.2.6.56

Your connection to this site is not private

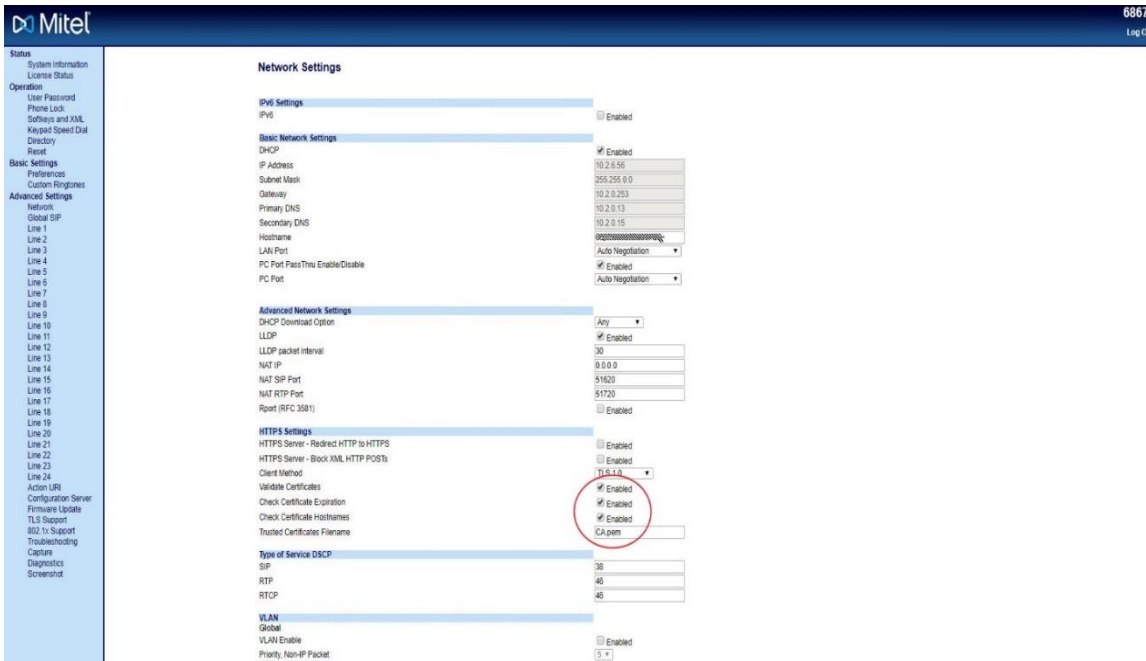
Username

Password

The main menu appears.



- 3 Under the **Advanced Settings**, select **Network**.
- 4 Under the **HTTPS Settings**, verify that TLS is enabled and that the **Trusted Certificate Filename** is specified.



- 5 Under the **Advanced Settings**, select **Global SIP**.
- 6 Verify that the **Basic SIP Network Settings** have the following values:
 - a **Proxy Server:** (MX-ONE System IP Address)

- b Proxy Port:** 5061
- c Registrar Server:** (MX-ONE System IP Address)
- d Registrar Port:** 5061

Global SIP Settings

Basic SIP Authentication Settings

Screen Name
Screen Name 2
Phone Number
Caller ID
Authentication Name
Password
BLA Number
Line Mode
Call Waiting

Basic SIP Network Settings

Proxy Server: 172.16.18.10
Proxy Port: 5061
Backup Proxy Server: 0.0.0.0
Backup Proxy Port: 0
Outbound Proxy Server: 0.0.0.0
Outbound Proxy Port: 0
Backup Outbound Proxy Server: 0.0.0.0
Backup Outbound Proxy Port: 0
Registrar Server: 172.16.18.10
Registrar Port: 5061
Backup Registrar Server: 0.0.0.0
Backup Registrar Port: 0
Registration Period: 0
Conference Server URI

Advanced SIP Settings

Explicit MMIV Subscription: ☒ Enabled
Explicit MMIV Subscription Period: 1800
MMIV for BLA account: ☐ Enabled
As-Feature-Event Subscription: ☐ Enabled
As-Feature-Event Subscription Period: 3600
UA-Profile-Event Subscription: Disabled
UA-Profile-Event Subscription Period: 36400
Send MAC Address in REGISTER Message: ☐ Enabled
Send Line Number in REGISTER Message: ☒ Enabled
Session Timer: 1800
T1 Timer: 0
T2 Timer: 0
Transaction Timer: 4000
Transport Protocol: Persistent TLS
Local SIP UDP/TCP Port: 5060
Local SIP TLS Port: 5061
Registration Failed Retry Timer: 300

- 7** Under **Advanced Settings**, select **Configuration Server**.
- 8** Verify that the following values are entered:
 - a Download Protocol:** Select TFTP
 - b Primary Server:** TFTP Server IP Address
 - c Pri TFTP Path:** \sipsw_MX6 (destination folder for configuration files, CA.pem file, and endpoint firmware)
 - d Alternate Server:** TFTP Server IP Address (If different than primary)
 - e Alt TFTP Path:** \sipsw_MX6 (destination folder for configuration files, CA.pem file, and endpoint firmware)
 - f Use Alt TFTP:** checked if Alternate Server is to be used for TFTP

Mitel

Status
System Information
License Status

Operation
User Password
Phone Lock
Softkeys and XML
Keypad Speed Dial
Directory
Reset

Basic Settings
Preferences
Custom Ringtones

Advanced Settings
Network
Global SIP
Line 1
Line 2
Line 3
Line 4
Line 5
Line 6
Line 7
Line 8
Line 9
Line 10
Line 11
Line 12
Line 13
Line 14
Line 15
Line 16
Line 17
Line 18
Line 19
Line 20
Line 21
Line 22
Line 23
Line 24
Action URI
Configuration Server
Firmware Update
TLS Support
802.1x Support
Troubleshooting
Capture
Diagnostics
Screenshot

Configuration Server Settings

Settings

Download Protocol: TFTP ▼

Primary Server: 172.16.1.126

Pri TFTP Path: tsipsw_MX6

Alternate Server: 172.16.20.4

Alt TFTP Path: tsipsw_MX6

Use Alt TFTP: ☒ Enabled

FTP Server:

FTP Path:

FTP Username:

FTP Password:

HTTP Server:

HTTP Path:

HTTP Port: 80

HTTPS Server:

HTTPS Path:

HTTPS Port: 443

Auto-Resync

Mode: BOTH ▼

Time (24-hour): 03:00 ▼

Maximum Delay: 60

Days: 0

XML Push Server List(Approved IP Addresses)

Save Settings

9 Click the **Save Settings** button to update the phone configuration.

10 Reboot the Endpoint/Terminal.

Depending on Terminal/Endpoints that are connected to the **MX-ONE** system, the startup.cfg and aastra.cfg files will need to be modified as well to enable TLS.

11 Modify the following lines in the .cfg file as shown in the following screenshots:

NOTE The sections highlighted in the following screenshots apply to settings for TLS.


```

-----
# Persistent SIP TLS (also requires 'sip outbound proxy' and
'sip persistent tls keep alive')
# (persistent TLS is the only SIP TLS supported natively by MX-
ONE)
#-----
-----
sips persistent tls:1
sip outbound support:1
#sip persistent tls keep alive: 15
sip transport protocol:4 # 4-TLS
sips trusted certificates:"CA.pem" #stored on Configuration
Server, A full URI path using HTTP,HTTPS,FTP,TFTP can also be
used, as for example "http://<server host>/<path>"

'sip outbound proxy' is provisioned by telephony server at
logon if TLS in 'action uri startup: https://...:22223/Startup'
is terminated in telephony server
sip outbound proxy port:5061 #5061 or 0(which will attempt SRV
and as fall back send to 5061 due to TLS)
#sip backup outbound proxy port:5061 #5061 or 0(which will
attempt SRV and as fall back send to 5061 due to TLS)
#sip backup outbound proxy:<mx-one server 2 terminating TLS>
sip srtp mode:1 #0(SRTP disabled),1(SRTP
preferred),2(SRTP only)
# When MX-one terminate the TLS also the HTTPS part is required

'sip outbound proxy' shall be configured If Session Border
Controller(SBC) is terminating TLS and forwards packets over TCP
to MX-ONE
#sip outbound proxy:<SBC>
#sip srtp mode:2 #0(SRTP disabled),1(SRTP preferred),2(SRTP
only). Most third party equipment is only aware of SRTP only
#SBC will redirect the call according to 'sip proxy ip'.

#The official host name for a LIM is limX, X=lim number. mxone-
domain the defined domain in MX-ONE
sip proxy ip:lim1.172.16.18.20
sip proxy port:5061
sip registrar ip:0.0.0.0
sip registrar port:5061

#-----
# HTTPS used for XML keys in MX-ONE and Configuration Server for
'download protocol:HTTPS'
# http and port 22222 needs to be changed to https and port
22223 in <model>.cfg as well
#-----
-----
https client method:"TLS 1.0"
https redirect http get:0 #phone webserver may redirect to
https
#0-disable, 1-enabled
#https validate certificates:0 #if disabled the server
certificate is not checked against 'https user certificates'.
#https validate expires:0 #If disabled time server is not
required
#https validate hostname:0 #Is disabled the Common Name of
the server certificate does not need to match the server name
#https user certificates: "CA.pem"
#"CA.pem" is stored on Configuration Server, A full URI path
using HTTP,HTTPS,FTP,TFTP can also be used, as for example
"http://<server host>/<path>"

```

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

- 12 Save this .cfg file on both the TFTP Server and on the MX-ONE system.
- 13 Restart the Endpoints/Terminals and verify that they register with the MX-ONE.

Programming Message Waiting and Call Diversion for Subscriber Telephones

Apply the commands in the following procedure to configure MWI and call diversion options for MiCollab AM subscribers on the Provision Manager Menus: **Services** > **Extension**.

To program Message Waiting and Call Diversion for Subscriber Telephones:

- 1 Use the Key System Function Key Change command to assign an MWI key appearance on each subscriber telephone. Digital subscriber telephones can have a lit MWI key assigned in addition to the **Message Waiting** display on their LCD telephones, and subscribers can then press that key to retrieve messages from MiCollab AM. (This is performed at the Command Line of the MDSH prompt.)

For example:

```
KSFKC:DIR=2001&&2299,KEY=2,FCN=MEW;
```

To verify your work, type the following command:

```
KSFKP:DIR=2001&&2299;
```

For IP-Extensions (IP-Phones D4 and D5):

This is set in the configuration file, located on the system software server, for the IP-phones.

For example:

```
[FunctionKeysDBC425]
```

```
MessageWaiting=5
```

- 2 Assign MiCollab AM access number as the diversion point for subscribers. Use the **diversion_common** command to create a common diversion to voice mail for subscribers, or use the **diversion** command to create individual diversions. (These commands are performed at the Terminal Emulation root menu.)

For example:

```
diversion -c -d 2001..2299--div-destination-number 75001
```

To verify your work, type the following command:

```
diversion -p -d 2001..2299
```

- 3 Create a call list for generic extensions (H323, SIP, DECT, and Remote extension) containing the MiCollab AM access number as last call position. Generic extensions do not have any individual diversion.

For example:

```
call_list -i -d 2201 --list 1 --position 1 --dest-number 2201 --ringing-time 25
```

```
call_list -i -d 2201 --list 1 --position 2 --dest-number 75001--ringing-time 5
```

If **Call Diversion/Call List** is not programmed, subscribers must use the Follow Me feature to divert calls to MiCollab AM.

NOTE If MiCollab AM is configured as the common diversion position, ICS calls are always diverted to this position, even if individual call diversion is programmed to divert calls elsewhere. In other words, reason code diversion always goes to the common diversion position. Refer to the *Voice Intercept Messaging* document for more information on programming reason code diversions.

Setting up Certificate when using TLS

To configure certificate in MiVoice MX-ONE Service Node, see the command `mxone_certificate` help to perform the appropriate configuration.

The following is an example of steps needed to setup a certificate and signing it with your own certificate when no previous certificate configuration exists. The procedure may not be the same for a different type of certificate. If certificate is already configured only an update of configuration file and Server Certificate is necessary.

To configure root and server certificate:

- 1 Start configuration with command **mxone_certificate** from the menus, and then select **Manage Root Certificate**.
 - Select **Create Root Certificate**.
 - Select **Install Root Certificate**.
- 2 Start configuration with command **mxone_certificate**, and then select **Manage Server Certificate**.
 - Select **Create configuration file with default settings**.
 - Edit the **Default file** and enter the **FQDN** and **IP** of MiCollab AM and MiVoice MX-ONE Service Nodes.
 - Select **Create Server Certificate** using configuration file.
 - Select **Install Server Certificate** and follow the instruction.

To sign the MiCollab AM certificate:

- 1 The CSR (Certificate Signing Request) from MiCollab AM is signed with command **mxone_certificate**.
 - Select **Manage Root Certificate**.
 - Select **Sign a server Certificate**.

NOTE Make sure the configuration file is configured with **FQDN** and **IP** of MiCollab AM and MiVoice MX-ONE Service Nodes.

- 2 Copy the resulting file to MiCollab AM that contains two certificates, **Local Certificate** and **Remote Trusted Certificate**.

Installing the Network Interface

The Ethernet network adapter card and TCP/IP protocol may have been installed during initial installation of the operating system. Alternatively, you can install both the network adapter and the required TCP/IP protocol now. Consult the site system administrator for specific information on how to configure the network environment for the MiCollab AM platform. Refer to the operating system documentation or online help for information on installing network adapter cards and network protocols.

Once the network environment is configured and MiCollab AM has joined the same network as the MiVoice MX-ONE Service Node, verify that MiCollab AM can communicate with the PBX via TCP/IP. At the MiCollab AM Call Server, open a command prompt window. Type the Ping command followed by the TCP/IP address assigned to the PBX. If the TCP/IP protocol and network interface is installed properly, the PBX will reply. The following is an example of how to use the Ping command:

Examples:

```
C:\>ping 245.17.41.1
```

```
Pinging 245.17.41.1 with 32 bytes of data:
```

```
Reply from 245.17.41.1: bytes=32 time<10ms TTL=128
```

```
Reply from 245.17.41.1: bytes=32 time<10ms TTL=128
```

```
Reply from 245.17.41.1: bytes=32 time<10ms TTL=128
```

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:

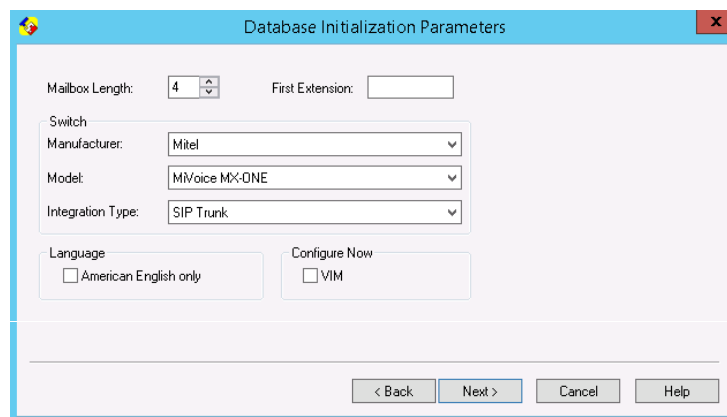
- [Configuring MiCollab AM for the Integration During Initial Installation](#): Integrate the telephone system while you install MiCollab AM for the first time.
- [Configuring Existing MiCollab AM for the Integration](#): Integrate a new telephone system on your existing MiCollab AM system.

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 for 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 MX-ONE
Integration Type: SIP Trunk

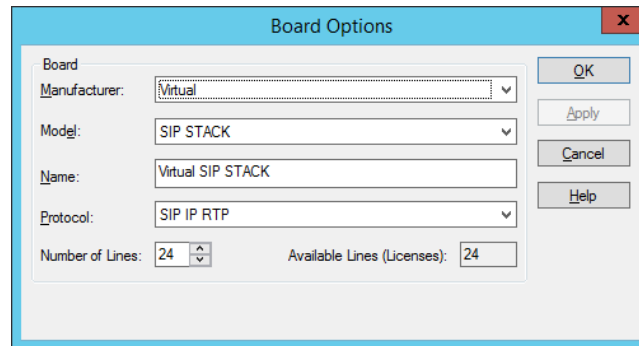
Language
☐ American English only

Configure Now
☐ VIM

< Back Next > Cancel Help

- a In the **Mailbox Length** box, enter the mailbox length in digits.
- b Leave the **First Extension** box empty.
- c From the **Manufacturer** drop-down list, select **Mitel**.
- d From the **Model** drop-down list, select **MiVoice MX-ONE**.
- 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:



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

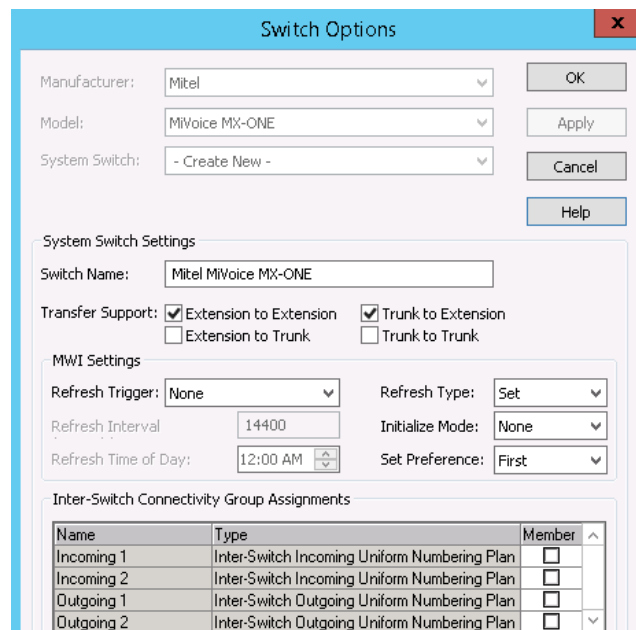
- Manufacturer:** Virtual
- Model:** SIP STACK
- Name:** Virtual SIP STACK
- Protocol:** SIP IP RTP
- Number of Lines:** 24
- Available Lines (Licenses):** 24

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

- a From the **Manufacturer** drop-down list, select **Virtual**.
 - b From the **Model** drop-down list, select **SIP STACK**.
 - c From the **Protocol** drop-down list, select **SIP IP RTP**.
 - d 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*.



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

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

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

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

Image continued on next page

Image continued from previous page

Local Switch Settings

View: All Settings Set Defaults

Name	Value
Disconnect Loop Current Length (ms)	100
Flash Hook Time (ms)	100
T1 Protocol	E&M
T1 Signaling	Wink

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

Integration Options

System Switch: Mitel MiVoice MX-ONE OK Apply Cancel Help ITN...

Integration Type: SIP Trunk

Integration: - Create New -

Name: Mitel MiVoice MX-ONE SIP Trunk

Local Integration Settings

View: Required Parameters Set Defaults

Name	Value
SIP Server Address	
SIP Server Port	5060
Use configured Domain Name	<input checked="" type="checkbox"/>
SIP Domain Name	MX-ONE
Transport for outgoing SIP messages	TCP
Local IP Address to bind on	fd8b:552b:a82a:15f8:eda:669:219b
SIP Local Connection Port	5060
SIP parser qualifier string	

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

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

Table 4. Required Parameters View – Integration Options

Field	Setting
SIP Server Address	Enter the IP address of the MiVoice MX-ONE Service Node. This can be the IP address (in the form of IPv4) or a Fully Qualified Domain Name (FQDN).
SIP Server Port	This is the port on which MiVoice MX-ONE Service Node is listening for SIP messages. MiCollab AM will send all SIP messages to this port. The default port is 5060 . When configured for TLS , the default is 5061 .

Use Configured Domain Name	Select this checkbox if you want to use the domain name configured in the SIP Domain Name field instead of SIP Server Address .
SIP Domain Name	If Use Configured Domain Name is enabled, enter a string that will be used to populate the host portion of the Request-URI , To , and From headers.
Local SIP Domain Name (if available)	<p>If Use Configured Domain Name is enabled, enter a string that will be used to populate the host portion of the Request-URI, To, and From headers of the outgoing SIP requests.</p> <p>This field overrides the value of SIP Domain Name (but for the From headers only).</p> <p>This field was added to the MiCollab AM configuration to support the environments where the MiCollab AM extensions (or the pilot number) are placed in a SIP domain that is different from the SIP domain for the rest of the extensions.</p>
Transport for outgoing SIP messages	<p>This is the transport medium that will be used to send outgoing SIP messages.</p> <ul style="list-style-type: none"> • Enter <i>TCP</i> when using the TCP SIP trunk profile. • Enter <i>TLS</i> when using the TLS_S RTP SIP trunk profile.
Local IP Address to bind on	Enter the IP address of the network interface card (NIC) on the MiCollab AM Call Server platform that supports the media server. If there is only one NIC on the Call Server platform, this field typically contains the IP address of that NIC.
SIP Local Connection Port	Enter the port number where MiCollab AM listens for incoming SIP messages. The default value is 5060 .
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 MiCollab AM, use a string that is unique to each SIP integration.</p> <p>For example:</p> <p>The SIP trunk profile is initiated with string tgrp=mxone-system. This string is entered as a unique string to identify the integration.</p> <p>NOTE This setting must match a string in the SIP header that is unique to this particular integration</p>

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

Table 5. Configuration Connection Security Settings Values

Field	Setting
Enable TLS	Indicates whether or not to enable TLS . NOTE Only enabled if SIP trunk profile with TLS is used.
SIP Server Address	Enter the IP addresses of all MiVoice MX-ONE Service Nodes that will be connecting to MiCollab AM. These are the addresses for trusted remote SIP servers that will process the TLS messages. NOTE To add more lines for this field, click the Add Trusted Address button.
SIP Server TLS Port	Enter the port number on the remote MiVoice MX-ONE Service Nodes that is configured to receive the TLS messages. The default value is 5061 .
SIP Local TLS Port	Enter the port number on the MiCollab AM system that will receive TLS messages. Specify a port that is not already in use by MiCollab AM (or another application). The default value is 5061 .
Local Certificate File Name	Path to the local security certificate file.
Local Private Key File Name	Path to the local private key file.
Local Issuer Certificate File Name	Path to a local issuer certificate key file (may not be needed in some configurations).
Remote Trusted Certificate File Name	Path to a remote trusted certificate.

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

Image continued on next page

Image continued from previous page

Local Switch Section Settings

View: Incoming Call Settings Set Defaults

Name	Value
Incoming Hunt Mode	Terminal
Hunt Group Access Code	
Incoming Line Reserve	48
Maximum Ports on Hold	48

- 9 In the **Switch Section Options** dialog box, configure the following options:
 - a In the **Local Integration Settings** section, select the **Incoming Call Settings** view.
 - b For the **Incoming Hunt Mode** value, select **Terminal**.
 - c In the **Hunt Group Access Code** field, enter the value for this integration.

NOTE Select the hunt mode that matches the hunt mode type in IP PBX programming.
 - 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 exiting MiCollab AM for the telephone integration:

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

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

- 3 **[Optional]** Select the **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 **Mitel**.
 - b From the **Model** drop-down list, select **MiVoice MX-ONE**.
 - 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*.

Switch Options ✕

Manufacturer: OK

Model: Apply

System Switch: Cancel

[Help](#)

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: Set Defaults

Name	Value
Disconnect Loop Current Length (ms)	100
Flash Hook Time (ms)	100
T1 Protocol	E&M
T1 Signaling	Wink

7 Click **OK**. The **Integration Options** dialog box appears.

Integration Options ✕

System Switch: OK

Integration Type: Apply

Integration: Cancel

Name: Help

[ITN...](#)

Local Integration Settings

View: Set Defaults

Name	Value
SIP Server Address	
SIP Server Port	5060
Use configured Domain Name	<input checked="" type="checkbox"/>
SIP Domain Name	MX-ONE
Transport for outgoing SIP messages	TCP
Local IP Address to bind on	fddb:552b:a82a:5004:15f8:eda:669:219b
SIP Local Connection Port	5060
SIP parser qualifier string	

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

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

Table 6. Required Parameter View – Integration Options

Field	Setting
SIP Server Address	Enter the IP address of the MiVoice MX-ONE Service Node. This can be the IP address (in the form of IPv4) or a Fully Qualified Domain Name (FQDN).
SIP Server Port	This is the port on which MiVoice MX-ONE Service Node is listening for SIP messages. MiCollab AM will send all SIP messages to this port. The default port is 5060 . When configured for TLS , the default is 5061 .
Use Configured Domain Name	Select this checkbox if you want to use the domain name configured in the SIP Domain Name field instead of SIP Server Address .
SIP Domain Name	If Use Configured Domain Name is enabled, enter a string that will be used to populate the host portion of the Request-URI , To , and From headers.
Local SIP Domain Name (if available)	If Use Configured Domain Name is enabled, enter a string that will be used to populate the host portion of the Request-URI , To , and From headers of the outgoing SIP requests. This field overrides the value of SIP Domain Name (but for the From headers only). This field was added to the MiCollab AM configuration to support the environments where the MiCollab AM extensions (or the pilot number) are placed in a SIP domain that is different from the SIP domain for the rest of the extensions.
Transport for outgoing SIP messages	This is the transport medium that will be used to send outgoing SIP messages. <ul style="list-style-type: none">• Enter <i>TCP</i> when using the TCP SIP trunk profile.• Enter <i>TLS</i> when using the TLS_SRTP SIP trunk profile.
Local IP Address to bind on	Enter the IP address of the network interface card (NIC) on the MiCollab AM Call Server platform that supports the media server. If there is only one NIC on the Call Server platform, this field typically contains the IP address of that NIC.
SIP Local Connection Port	Enter the port number where MiCollab AM listens for incoming SIP messages. The default value is 5060 .

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 MiCollab AM, use a string that is unique to each SIP integration.</p> <p>For example:</p> <p>The SIP trunk profile is initiated with string tgrp=mxone-system. This string is entered as a unique string to identify the integration.</p> <p>NOTE This setting must match a string in the SIP header that is unique to this particular integration</p>
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- b** In the **Local Integration Settings** section, select the **Connection Security Parameters** View and configure the settings as follows:

Table 7. Configuration Connection Security Settings Values

Field	Setting
Enable TLS	<p>Indicates whether or not to enable TLS.</p> <p>NOTE Only enabled if SIP trunk profile with TLS is used.</p>
SIP Server Address	<p>Enter the IP addresses of all MiVoice MX-ONE Service Nodes that will be connecting to MiCollab AM. These are the addresses for trusted remote SIP servers that will process the TLS messages.</p> <p>NOTE To add more lines for this field, click the Add Trusted Address button.</p>
SIP Server TLS Port	<p>Enter the port number on the remote MiVoice MX-ONE Service Nodes that is configured to receive the TLS messages. The default value is 5061.</p>
SIP Local TLS Port	<p>Enter the port number on the MiCollab AM system that will receive TLS messages. Specify a port that is not already in use by MiCollab AM (or another application). The default value is 5061.</p>
Local Certificate File Name	<p>Path to the local security certificate file.</p>
Local Private Key File Name	<p>Path to the local private key file.</p>
Local Issuer Certificate File Name	<p>Path to a local issuer certificate key file (may not be needed in some configurations).</p>

Remote Trusted Certificate Path to a remote trusted certificate.
File Name

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

Switch Section Options

Local Switch: Mitel MiVoice MX-ONE

System Switch Section: - Create New -

System Switch Section Settings

Name: Mitel MiVoice MX-ONE Section

Node Code:

Location Code:

Location: Auto Location

MWI Integration: Mitel MiVoice MX-ONE SIP Trunk

Local Switch Section Settings

View: Incoming Call Settings

Name	Value
Incoming Hunt Mode	Terminal
Hunt Group Access Code	
Incoming Line Reserve	48
Maximum Ports on Hold	48

- 10 In the **Switch Section Options** dialog box, configure the following options:
- a In the **Local Integration Settings** section, select the **Incoming Call Settings** view.
 - b For the **Incoming Hunt Mode** value, select **Terminal**.
 - c In the **Hunt Group Access Code** field, enter the value for this integration.

NOTE Select the hunt mode that matches the hunt mode type in IP PBX programming.

- 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 8. Secondary SIP Server Address and the Secondary SIP Server Port example

Field	Value
Secondary SIP Server Address	<div>Enter the TCP/IP address or an FQDN of the secondary node.</div> <div>For example: The IP address 123.45.6.789 as displayed on the Review/Modify SIP Gateway screen.</div> <div>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.</div> <div>IMPORTANT This value must match the configuration on the Gateway of the secondary node.</div>
Secondary SIP Server Port	Enter the port number of the secondary node. The default value is 5060 .

- 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 9. 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 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 10. 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