

# **MiCollab Advanced Messaging Alcatel-Lucent Enterprise OmniPCX Enterprise Communication Server SIP Integration Technical Note**

For version 6.1 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 are familiar with its procedures and terminology. This document also assumes that you are familiar with the features and programming of the Alcatel-Lucent Enterprise – OmniPCX Enterprise Communication Server (CS).

This document describes how to integrate MiCollab AM with OmniPCX Enterprise CS 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.

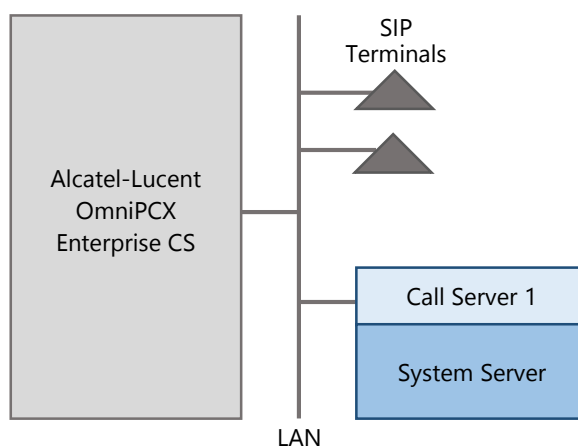


Figure 1. Alcatel-Lucent Enterprise – OmniPCX Enterprise CS Integration

The OmniPCX Enterprise CS integration is a SIP trunk integration, MiCollab AM registers itself as a single voice mail pilot number at the telephone system's SIP proxy. The proxy routes all incoming calls for MiCollab AM to that pilot number, and MiCollab AM assigns them to lines as needed. MiCollab AM also sends outbound calls and MWI messages to the pilot number so that the proxy can act on them.

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

Use this document in conjunction with *System Installation Guide*, *System Administration Guide* and with the MiCollab AM online help system.

## 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: Refer to *System Installation Guide*.

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

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

- **User Input.** Information required to be typed 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 Alcatel-Lucent Enterprise – OmniPCX Enterprise Communication Server (CS) SIP 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
Reject Call	Yes

Table 2. Integration features supported for Alcatel-Lucent Enterprise – OmniPCX Enterprise CS SIP station set

Feature	Supported	Notes
Automatic subscriber logon	Yes	
ANI/CLI	Yes	
Announce Busy greeting on forwarded calls	Yes	
Call screening	Yes	Note 1
Caller queuing	Yes	Note 2
DNIS	Yes	
End-to-end DTMF, attendant console	Yes	
End-to-end DTMF, proprietary telephones	Yes	
Fax Detection	Yes	
Internal calling party ID for reply	Yes	
Live record, integrated	Yes	
Live reply to sender	Yes	
Message notification callouts	Yes	
MWI, set/clear	Yes	
MWI, inband/outband	Outband	
Networking, analog	Yes	
Overflow from MiCollab AM to attendant	Yes	
Overflow to MiCollab AM from attendant	Yes	
PBX-provided disconnect signaling	Yes	
Revert to operator	Yes	
SRTP	No	Note 3
TLS	No	Note 3
Transfers, blind	Yes	
Transfers, confirmed	Yes	
Transfers, fully supervised	Yes	
Transfers, monitored	Yes	

Trunk ID for call routing	No	
Multiple Integrations	Yes	Note 4

## NOTES

1. Only available when using supervised transfers
2. Caller Queuing is specific to each local Call Server. Call Servers within the system are unaware of queued calls to the same subscriber on other Call Servers. For more information, refer to the [Critical Application Considerations](#) section.
3. MiCollab AM supports negotiation for SRTP media streams using the Secure RTP profile defined in RFC 3711 with the offer/answer model defined in RFC 3264. To enable SRTP, RTP, or both, see integration configuration options documentation for the switch. The default setting is RTP. Please note that MiCollab AM doesn't support RFC 5939 which is an extension of RFC 3264. Also, please note that SRTP has not been qualified for this integration, and no switch programming is available for setting up SRTP on the switch side. However SRTP may be enabled as described above, and technical support will be available on a best effort basis.
4. Refer to the [Critical Application Considerations](#) section.



# 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.

- If your MiCollab AM server platform is a component of two or more local or wide area networks (LANs or WANs), you must make sure that this integration does not interfere with the normal network operation of the server. By default, MiCollab AM uses the primary (public) network interface card (NIC) in the platform, the first NIC in the network binding order. If you want MiCollab AM to use a NIC other than the first one, you must make several required configuration changes. It is much easier to configure the Integration to use another NIC by simply setting the integration parameter Local IP Address to bind on to the address of the NIC card connected to the PBX. For more information, refer to the section, [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.
- You must configure the Hunt Group Access Code in the Switch Section Options dialog box and must match the Voice Mail Dir. No. configured in the Configure the Voicemail Settings section.
- On PBX versions prior to 9.0, the Call Server Processing Unit must be restarted after you configure the SIP gateway.
- When using Alcatel-Lucent Enterprise – OmniPCX Enterprise Communication Server, the node name of the PBX must be a fully qualified domain name (FQDN).
- In the case of call forwarding multiple times, subscriber telephones can go to **mgr > Classes of Service > Phone Feature COS** and modify the Voice Mail Forwarding parameter as follows:

Table 3. Voice Mail Forwarding options

If user wants the call goes	Then set Voice-Mail forwarding as
The first set's voice mail	Right Call Set mail
The final set's voice mail	Ring Final Set

- MiCollab AM 6.1 supports up to 10 integration types (i.e. licensed integrations) in total per system. However, the following limitations apply to each Call Server:
  - Limited to 3 integration types per Call Server
  - The 3 integration types can be any mix of TDM and SIP (e/g/ 1 TDM and 2 SIP)
  - Limited to 1 Mitel MiTAI or 1 Cisco UCM SCCP IP integration. Can be mixed with TDM, but not with SIP.
  - Connect up to 10 telephone systems total per Call Server (e.g. 2 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

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

## Telephone System Requirements

- Alcatel-Lucent Enterprise – OmniPCX Enterprise Communication Server
- CPU Type: CPU80s1 and 128MB RAM
- One SIP Gateway license for the system (#185) and one SIP Network Links license (#188) for each MiCollab AM port

## MiCollab AM Requirements

- MiCollab AM version 6.1
- Mitel software key and feature file with the Alcatel-Lucent Enterprise – OmniPCX Enterprise Communication Server SIP integration enabled and one RADVISION® SIP and RTP license enabled for each port involved in the integration
- One or two network interface cards

# Programming the Telephone System

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

The installing technician should be familiar with programming the telephone system. Programming is performed from OmniPCX Enterprise CS-programming terminal. For detailed programming information on this software version or other OmniPCX Enterprise CS software versions, refer to the appropriate OmniPCX Enterprise CS documentation.

All programming is performed through the Manager application. To use the Manager application, access the main Call Server Processing Unit through a command line program such as Telnet. Type your login and password, and then type *mgr* to enter the Manager application.

## Verifying the Network Settings

To verify the network settings:

- 1 From the main menu select **Translator**, and then press **Enter**.
- 2 Select **Network Routing Table**, and then press **Enter**.
- 3 Select **Review/Modify**, and then press **Enter**.
- 4 Press **Enter** to select All Instances, and then press **Ctrl + V**.
- 5 For the node you have chosen, verify the settings as shown in the example below.

```
Review/Modify: Network Routing Table

Node Number (reserved)           : 1
Instance (reserved)              : 1
Network Number                   : 10

Rank of First Digit to be Sent   : 1
Incoming Identification Prefix    : -----
Protocol Type                    + ABC-F
Numbering Plan Descriptor ID     : 11
ARS Route List                   : 0
Schedule Number                  : -1
ATM Address ID                   : -1
Network Call Prefix              : -----
City/Town Name                   : -----
Send City/Town Name              : False
Associated Ext SIP Gateway       : -1
Enable UTF8 Name Sending         : True
```

## Creating the SIP Trunk Group

To create the SIP trunk group:

- 1 From the main menu select Trunk Groups, and then press **Enter**.
- 2 Select **Create**, and then press **Enter**.
- 3 Change the settings as shown in the example below, and then press **Ctrl + V** to save your changes.

```
Create: Trunk Groups
      Node Number (reserved)           : 1
Trunk Group ID                         : 10

Trunk Group Type                       + T2
Trunk Group Name                       : SIP Local
  UTF-8 Trunk Group Name               : -----
  Number Compatible With               : -1
Remote Network                        : 10
  Shared Trunk Group                   + False
  Special Services                     + Nothing

Create: Trunk Groups
      Node Number                       : 1
      Transcom Trunk Group               + False
      Auto.reserv.by Attendant           + False
      Overflow Trunk Group No.           : -1
      Tone on Seizure                   + False
      Private Trunk Group                + False
Q931 Signal Variant                    + ABC-F
      SS7 Signal Variant                 + No variant
      Number of Digits to Send           : 0
      Channel Selection Type             + Quantified
      Auto.DTMF Dialing on Outgoing Call + YES
T2 Specifications                      + SIP
      Public Network COS                 : 31
      DID Transcoding                   + False
      Can Support UUS in SETUP           + True
```

## Configuring the SIP Gateway

To configure the SIP gateway:

- 1 From the main menu select **SIP**, and then press **Enter**.
- 2 Select **SIP Gateway**, and then press **Enter**.
- 3 Select **Review/Modify**, and then press **Enter**.
- 4 Press **Enter** to select **All Instances**, and then press **Ctrl + V**.
- 5 Change the settings as shown in the example below and then press **Ctrl + V** to save your changes.

**NOTE** The machine name is the domain name for MiCollab AM. This value is case sensitive and is unique for your site.

```
Review/Modify: SIP Gateway
Node Number (reserved)           : 1
Instance (reserved)              : 1
Instance (reserved)              : 1

SIP Subnetwork                   : 10
SIP Trunk Group                  : 10
IP Address (FQDN)                : 123.45.6.789
Machine Name - Host              : alcatel-oxe (See note *)
SIP Proxy Port Number            : 5060
SIP Subscriber Min Duration      : 1800
SIP Subscriber Max Duration      : 86400
Session Timer                    : 1800
Min. Session Timer               : 900
Session Timer Method             + RE_INVITE
DNS Local Domain Name            : -----
DNS Type                         + DNS A
SIP DNS1 IP Address              : -----
SIP DNS2 IP Address              : -----
SDP IN 180                      + True
Cac SIP-SIP                     + False
INFO method for remote extension + False
Dynamic Payload type for DTMF    : 97
```

**NOTE** Alcatel-Lucent Enterprise – OmniPCX Enterprise Communication Server requires a fully qualified domain name (FQDN) as the Machine Name-Host.

## Configuring the MiCollab AM Server as a Trusted IP Address

To configure the IP Address:

- 1 From the main menu select **SIP**, and then press **Enter**.
- 2 Select **Trusted IP Addresses**, and then press **Enter**.
- 3 Select **Create**, and then press **Enter**.
- 4 Type the IP address of the MiCollab AM server in the **trusted address** line, and then press **Ctrl + V** to save your changes.

**IMPORTANT** You must restart the Call Server Processing Unit to apply your changes on PBX versions prior to 9.0.

# Configuring the SIP Registration Period

To configure the SIP Registration Period:

- 1 From the main menu select **SIP**, and then press **Enter**.
- 2 Select **SIP Registrar**, and then press **Enter**.
- 3 Select **Review/Modify**, and then press **Enter**.
- 4 Press **Enter** to select All Instances, and then press **Ctrl + V**.
- 5 Change the **SIP Min Expiration Date** as shown in the example below, and then press **Ctrl + V** to save your changes.

**NOTE** The default value of the SIP Min Expiration Date (expiry period) on OmniPCX Enterprise CS is 3600 seconds. This value must be less than or equal to the value set in the field, Registration period in seconds (default 90 seconds) in the MiCollab AM Required Parameter Settings for the MiCollab AM fail-over to work effectively.

```
Review/Modify: SIP Registrar
                Node Number (reserved)      :1
                Instance (reserved)          :1
                Instance (reserved)          :1

                SIP Min Expiration Date      :60
                SIP Max Expiration Date      :86400
```

- 6 Change the **SIP Parameters** as shown in the example below. Packetization times per codec must be set to **False**.

```
Review/Modify: SIP Parameters
                Node Number (reserved)      : 101
                Instance (reserved)          : 1
                Instance (reserved)          : 1
                System Option                Packetization times per codec

                Packetization times per codec      False
```

- 7 Press **Ctrl+V** to save your changes.

**IMPORTANT** You must restart the Call Server Processing Unit to apply your changes on PBX versions prior to 9.0.

## Restarting the Call Server Processing Unit

On PBX software versions prior to 9.0 you must restart the Call Server Processing unit to apply the programming changes.

### To restart the Call Server Processing Unit:

- 1 From the main menu press **Ctrl + C** to quit the Manager application.
- 2 In the command line, type *killall sipmotor*, and then press **Enter**.

## Configuring the Voicemail Settings

**NOTE** This procedure programs the Messages button on telephones system-wide.

### To configure the voicemail settings:

- 1 From the main menu select **Applications**, and then press **Enter**.
- 2 Select **External Voicemail**, and then press **Enter**.
- 3 Select **Create**, and then press **Enter**.
- 4 Change the settings as shown in the example below, and then press **Ctrl + V** to save your changes.

**IMPORTANT** Do not use 0000 as your SIP password. Instead, choose a unique password.

```
Create: External Voice Mail
Node Number (reserved)      : 1
Instance (reserved)         : 1
Voice Mail Dir. No.         : 5000

Directory Name               : MiCollab AM
Connection COS               : 0
Public Network COS           : 2
Entity Number                : 1
Cost Center ID               : 255
Charging COS                 + Justified
URL UserName                 : -----
Remote SIP Domain            + False
URL Domain                   : node000000

SIP Password                 : 0000
Confirm                      : 0000

Register On Line Number      : -----
Register URL (Username)      : -----
Register URL (Remote domain) + False
Register URL (Domain)        : -----

Register Password            : -----
Confirm                      : -----
```



# Configuring MiCollab AM

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

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

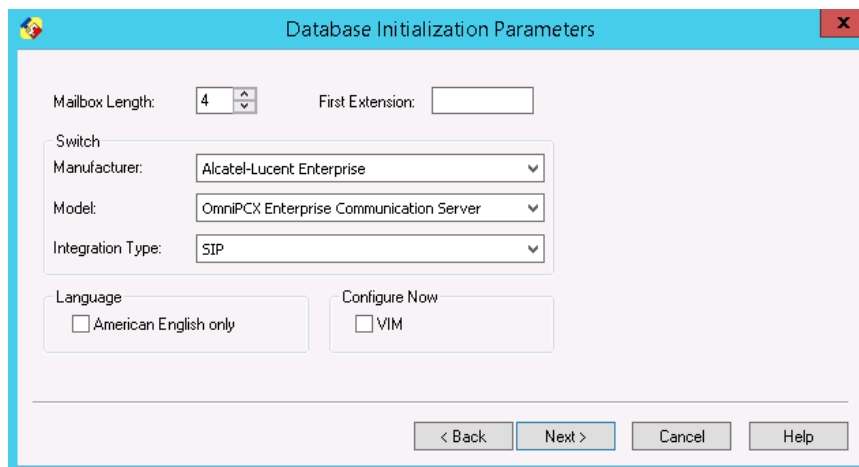
- [Configuring MiCollab AM for the Integration During Initial Installation](#): Integrate the telephone system while you install MiCollab AM for the first time.
- [Configuring Existing MiCollab AM for the Integration](#): Integrate a new telephone system on your exiting MiCollab AM system.

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

## Configuring MiCollab AM for the Integration During Initial Installation

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

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



Database Initialization Parameters

Mailbox Length: 4 First Extension:

Switch

Manufacturer: Alcatel-Lucent Enterprise

Model: OmniPCX Enterprise Communication Server

Integration Type: SIP

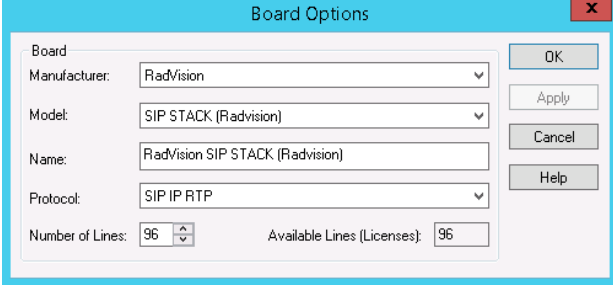
Language ☐ American English only

Configure Now ☐ VIM

< Back Next > Cancel Help

- a In the **Mailbox Length** box, enter the mailbox length in digits.
- b In the **First Extension** box, enter first extension number for the first line. You can also leave the **First Extension** box empty.

- c From the **Manufacturer** dropdown list, select **Alcatel-Lucent Enterprise**.
  - d From the **Model** dropdown list, select **OmniPCX Enterprise Communication Server**.
  - e From the **Integration Type** dropdown list, select **SIP**.
- 2 Click **Next**. The **Board Options** dialog box displays for the virtual board configuration.

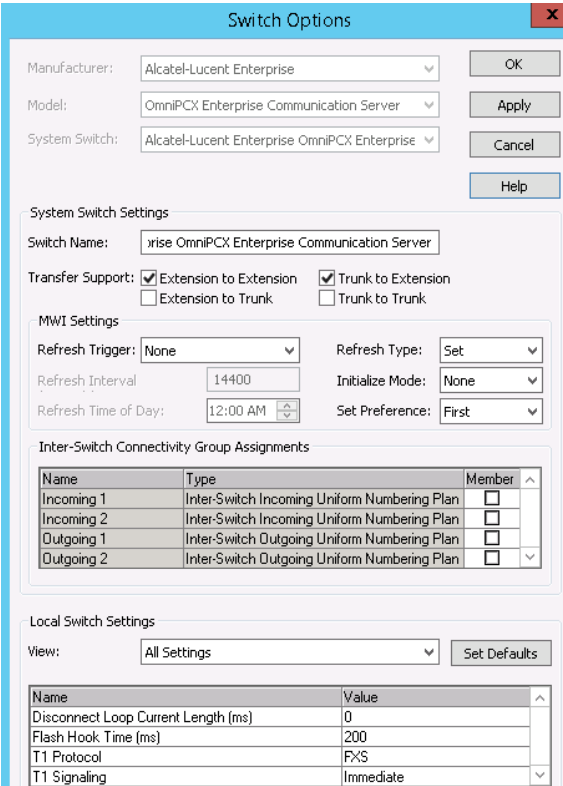


The **Board Options** dialog box is shown with the following configuration:

- Manufacturer:** RadVision
- Model:** SIP STACK (Radvision)
- Name:** RadVision SIP STACK (Radvision)
- Protocol:** SIP IP RTP
- Number of Lines:** 96
- Available Lines (Licenses):** 96

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

- 3 In the **Board Options** dialog box, configure the following options:
- a From the **Manufacturer** dropdown list, select **RadVision**.
  - b From the **Model** dropdown list, select **SIP STACK**.
  - c In the **Name** field, the name for this board is automatically generated. Enter a new name if necessary.
  - d From the **Protocol** dropdown list, select **SIP IP RTP**.
  - e In the **Number of Lines** field, enter the number of lines this board uses. The total number of lines is limited by the capacity of the board and the number of **Available Line Licenses**.
- 4 Click **OK**. The **Switch Options** dialog box displays.



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

- Manufacturer:** Alcatel-Lucent Enterprise
- Model:** OmniPCX Enterprise Communication Server
- System Switch:** Alcatel-Lucent Enterprise OmniPCX Enterprise

**System Switch Settings**

- Switch Name:** rise OmniPCX Enterprise Communication Server
- Transfer Support:**
  - ☒ Extension to Extension
  - ☐ Extension to Trunk
  - ☒ Trunk to Extension
  - ☐ 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"/>

**Local Switch Settings**

**View:** All Settings

**Set Defaults**

Name	Value
Disconnect Loop Current Length (ms)	0
Flash Hook Time (ms)	200
T1 Protocol	FXS
T1 Signaling	Immediate

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

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

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

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

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

Table 4. Required Parameters Settings

Field	Value
SIP Server Address	Enter the TCP/IP address or an FQDN. <b>For example:</b> The IP address 123.45.6.789 or Machine Name-Host as configured on the Review/Modify SIP Gateway screen. <b>NOTE</b> Alcatel-Lucent Enterprise – OmniPCX Enterprise CS requires the Machine Name to be a fully qualified domain name (FQDN). Use the Machine Name-Host in this field. <b>IMPORTANT</b> This value must match the configuration of the SIP Gateway.

SIP Server Port	<p>The port on which MiCollab AM listens for incoming SIP messages. The default is <b>5060</b>.</p> <p><b>IMPORTANT</b> This value must match the configuration of the SIP Gateway.</p>
SIP Domain Name	<p>The domain name as configured on the switch.</p> <p><b>IMPORTANT</b> This value is case-sensitive and must match the Machine Name-Host configured on the Review/Modify screen of the SIP Gateway.</p> <p><b>For example:</b> alcatel-lucent omnipcx</p>
Transport for outgoing SIP Messages	<p>Select the transport protocol used for sending out SIP messages. The default value is <b>UDP</b>.</p>
SIP Device Name	<p>The voice mail pilot number as configured on the switch. This value match the hunt group number configured in the Alcatel-Lucent Enterprise Switch Section of MiCollab AM.</p> <p><b>IMPORTANT</b> The SIP device name must match the <b>VOICE Mail Dir. NO.</b> configured on the <b>Create: External Voice Mail</b> screen.</p>
PBX Password	<p>The password associated with the SIP Device Name.</p> <p><b>For example:</b> 0000</p> <p><b>IMPORTANT</b> This password must match the SIP password configured from the <a href="#">Configuring the Voicemail Settings</a> section.</p>
Registration period in seconds	<p>Controls the frequency of registration refreshes and not only helps the switch to route calls to voicemails but also enables MiCollab AM to detect any network switch failure and initiate the failover process.</p> <p>The smaller this value, the sooner the failover initiates. The default is <b>90</b>.</p>
Local IP Address to bind on	<p>Select the local TCP/IP address of the MiCollab AM machine. This is a drop-down box and displays all available local TCP/IP addresses.</p>
SIP Local Connection Port	<p>The port on which MiCollab AM listens for incoming SIP messages. The default is <b>5060</b>.</p>
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.</p>

This field is used by MiCollab AM to match SIP packets to the appropriate SIP integration.

In cases where there are multiple SIP integrations on the call server, use a string that is unique to the SIP integrations in use.

**For example:**

The extension that will be used as the hunt number on the PBX followed by the @ symbol and the IP of the call server, such as **5000@172.16.4.202**. The hunt number must be unique across all IP integrations.

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

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

Media packet size  
(milliseconds)

MiCollab AM sends/receives packets containing the number of milliseconds worth of audio data set here.

The default value is **20**.

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

Name	Value
Incoming Hunt Mode	Terminal
Hunt Group Access Code	

- 9 In the **Switch Section Options** dialog box, configure the following options.
- a In the **Local Switch Settings** section, select **Required Parameters** view.
  - b In the **Incoming Hunt Mode** field, select the mode appropriate for your configuration.
  - c In the **Hunt Group Access Code** field, type the pilot number or destination code that users dial to reach MiCollab AM.
  - d Click **OK**.

- 10 Continue through and complete the configuration. At the end of the configuration, a confirmation dialog box displays. 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

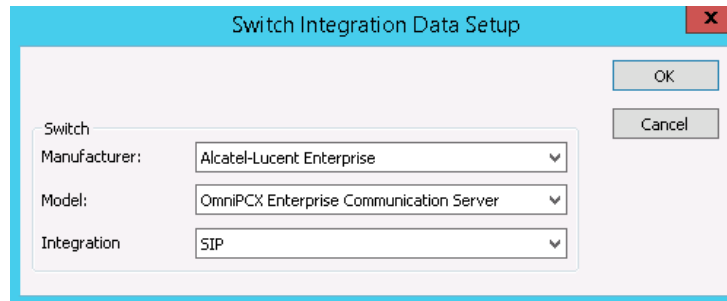
To configure exiting MiCollab AM with the integration:

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

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

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

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



Switch Integration Data Setup

Switch

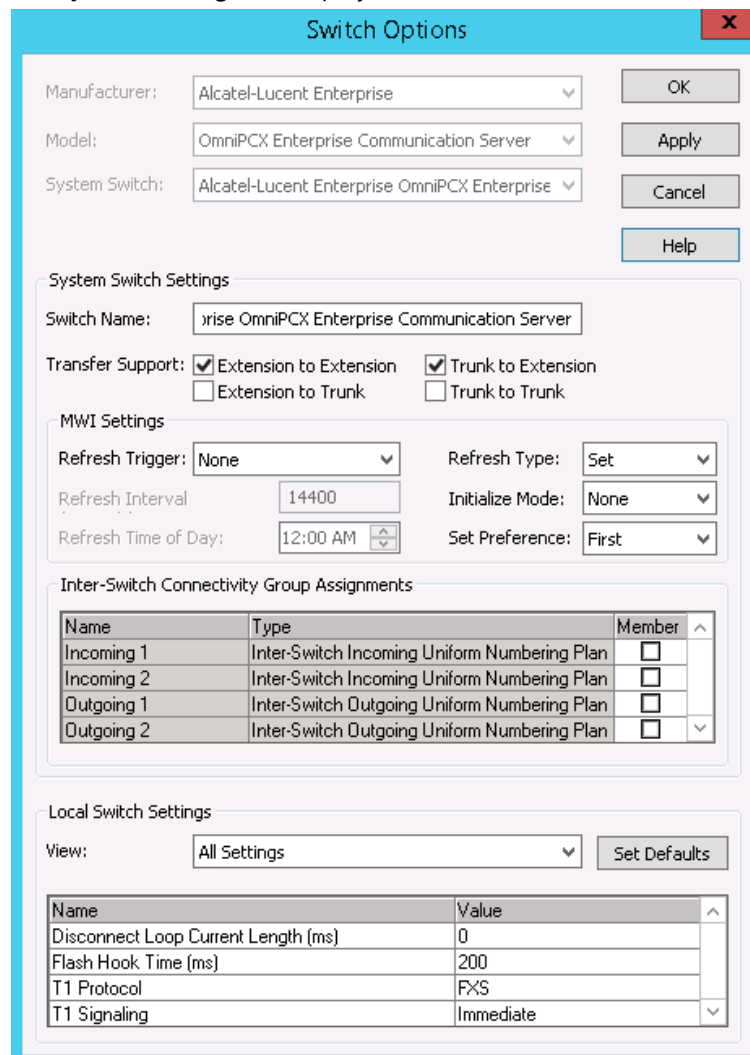
Manufacturer: Alcatel-Lucent Enterprise

Model: OmniPCX Enterprise Communication Server

Integration: SIP

OK Cancel

- a From the **Manufacturer** dropdown list, select **Avaya**.
  - b From the **Model** dropdown list, select **OmniPCX Enterprise Communication Server**.
  - c From the **Integration Type** dropdown list, select **SIP**.
- 5 Click **OK**. The **Switch Options** dialog box displays.



Switch Options

Manufacturer: Alcatel-Lucent Enterprise

Model: OmniPCX Enterprise Communication Server

System Switch: Alcatel-Lucent Enterprise OmniPCX Enterprise

OK Apply Cancel Help

System Switch Settings

Switch Name: rise OmniPCX Enterprise Communication Server

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

MWI Settings

Refresh Trigger: None Refresh Type: Set

Refresh Interval: 14400 Initialize Mode: None

Refresh Time of Day: 12:00 AM Set Preference: First

Inter-Switch Connectivity Group Assignments

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

Local Switch Settings

View: All Settings Set Defaults

Name	Value
Disconnect Loop Current Length (ms)	0
Flash Hook Time (ms)	200
T1 Protocol	FXS
T1 Signaling	Immediate

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

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

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

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

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

Table 5. Required Parameters Settings

Field	Value
SIP Server Address	<p>Enter the TCP/IP address or an FQDN.</p> <p><b>For example:</b></p> <p>The IP address 123.45.6.789 or Machine Name-Host as configured on the Review/Modify SIP Gateway screen.</p> <p><b>NOTE</b> Alcatel-Lucent Enterprise – OmniPCX Enterprise CS requires the Machine Name to be a fully qualified domain name (FQDN). Use the Machine Name-Host in this field.</p> <p><b>IMPORTANT</b> This value must match the configuration of the SIP Gateway.</p>
SIP Server Port	<p>The port on which MiCollab AM listens for incoming SIP messages.</p> <p>The default is <b>5060</b>.</p> <p><b>IMPORTANT</b> This value must match the configuration of the SIP Gateway.</p>



SIP Domain Name	<p>The domain name as configured on the switch.</p> <div> <b>IMPORTANT</b> This value is case-sensitive and must match the Machine Name-Host configured on the Review/Modify screen of the SIP Gateway.           <b>For example:</b> alcatel-lucent omnipcx       </div>
Transport for outgoing SIP Messages	A default value of UDP is already selected, but you can select TCP, depending on your configuration.
SIP Device Name	<p>The voice mail pilot number as configured on the switch. This value match the hunt group number configured in the Alcatel-Lucent Enterprise Switch Section of MiCollab AM.</p> <div> <b>IMPORTANT</b> The SIP device name must match the <b>VOICE Mail Dir. NO.</b> configured on the <b>Create: External Voice Mail</b> screen.         </div>
PBX Password	<p>The password associated with the SIP Device Name.</p> <div> <b>For example:</b> 0000         </div> <div> <b>IMPORTANT</b> This password must match the SIP password configured from the <a href="#">Configuring the Voicemail Settings</a> section.         </div>
Registration period in seconds	<p>Controls the frequency of registration refreshes and not only helps the switch to route calls to voicemails but also enables MiCollab AM to detect any network switch failure and initiate the failover process.</p> <p>The smaller this value, the sooner the failover initiates.</p> <p>The default is <b>90</b>.</p>
Local IP Address to bind on	<p>Select the local TCP/IP address of the MiCollab AM machine.</p> <p>This is a drop-down box and displays all available local TCP/IP addresses.</p>
SIP Local Connection Port	<p>The port on which MiCollab AM listens for incoming SIP messages.</p> <p>The default is <b>5060</b>.</p>
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.</p> <p>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 the SIP integrations in use.</p> <div> <b>For example:</b>   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 </div>

**5000@172.16.4.202.** The hunt number must be unique across all IP integrations.

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

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

Media packet size  
(milliseconds)

MiCollab AM sends/receives packets containing the number of milliseconds worth of audio data set here.

The default value is **20**.

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

Name	Value
Incoming Hunt Mode	Terminal
Hunt Group Access Code	

- 10 In the **Local Switch Settings** section, select **Required Parameters** View and configure the following options:
- a In the **Incoming Hunt Mode** field, select the mode appropriate for your configuration.
  - b In the **Hunt Group Access Code** field, type the pilot number or destination code that users dial to reach MiCollab AM.
  - c Click **OK**.
- 11 In **MiCollab AM Configuration**, Verify that 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 Switch Redundancy and Failover

The Alcatel-Lucent OmniPCX Enterprise Communication Server (CS) nodes can be configured for redundancy/failover in multiple ways.

- The OmniPCX Enterprise CS nodes can be configured to share an IP address.  
An OmniPCX Enterprise CS node whose current role is MAIN, owns the IP address and communicates with MiCollab AM and other devices through that IP address.  
If the roles between the OmniPCX Enterprise CS nodes switch, the ownership of the IP address is also switched in a seamless fashion. For such a configuration, the switch failover is transparent to MiCollab AM and it is not necessary to configure MiCollab AM for failover.
- The OmniPCX Enterprise CS nodes can be configured so that they share a common Fully Qualified Domain Name (FQDN) but separate IP addresses as displayed in the following PBX programming example of the DNS Delegation Properties dialog box.

**NOTE** The two OmniPCX Enterprise CS nodes may share a common domain name or SRV record instead of a FQDN.

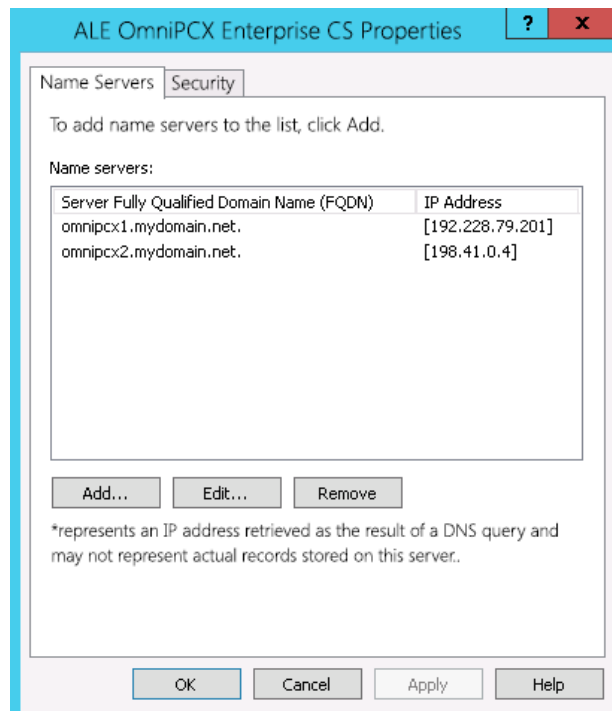


Figure 2. DNS Delegation Properties Dialog Box

This is a typical configuration when the servers are geographically separated and are located on different subnets (also known as spatial redundancy).

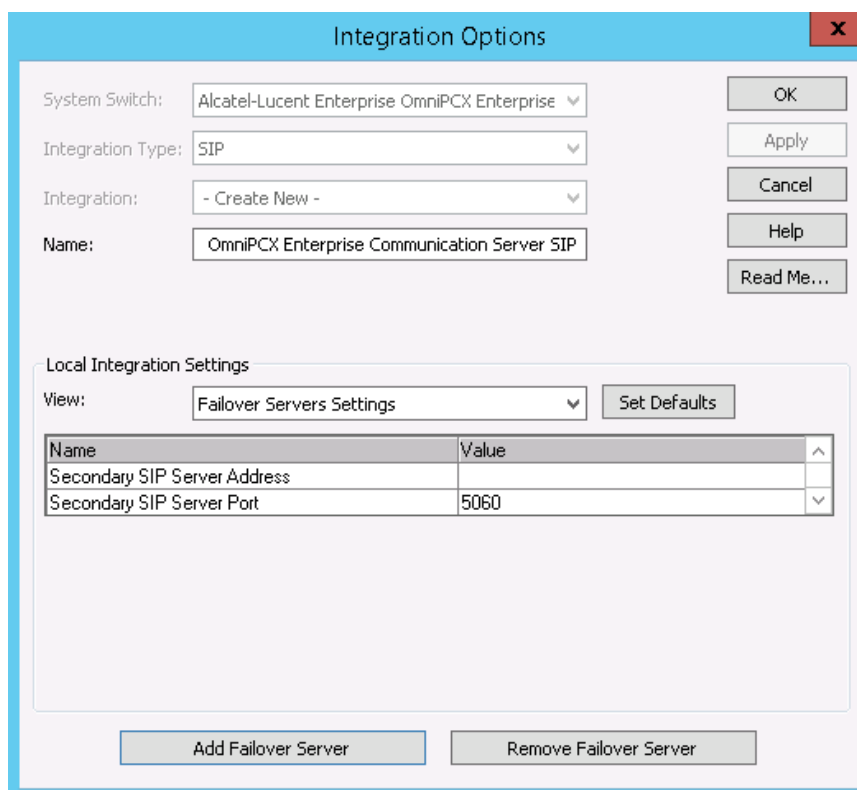
The DNS server on the IP network is responsible to resolve the FQDN to an IP address. The DNS server is configured to delegate the request to the unique OmniPCX Enterprise CS nodes for the FQDN associated with the OmniPCX Enterprise CS. The OmniPCX Enterprise CS node whose current role is MAIN responds to the DNS queries and provides its IP address.

If the OmniPCX Enterprise CS role switches, the node responding to the DNS queries also switches. Therefore, the switch failover is transparent to MiCollab AM as long as it is configured to work with FQDN instead of an IP address. The failover server configuration should be configured to use the FQDN.

MiCollab AM can support switch failover even if the OmniPCX Enterprise CS nodes are not configured to use a FQDN. In this case, follow the steps outlined below to configure the IP addresses of the primary and failover OmniPCX Enterprise CS nodes.

## To add a failover server:

- 1 From **MiCollab AM Configuration**, click the **Integrations** tab.
- 2 From the **Integrations** list, select **Alcatel-Lucent Enterprise OmniPCX Enterprise Communication Server SIP**, and then click **Edit**.
- 3 In the **Integration Options** dialog box, go to the **Local Integration Settings** section.
- 4 From the **View** dropdown list, select **Failover Server Settings**.



The screenshot shows the 'Integration Options' dialog box. The 'System Switch' is set to 'Alcatel-Lucent Enterprise OmniPCX Enterprise', 'Integration Type' is 'SIP', and 'Integration' is '- Create New -'. The 'Name' field contains 'OmniPCX Enterprise Communication Server SIP'. On the right, there are buttons for 'OK', 'Apply', 'Cancel', 'Help', and 'Read Me...'. The 'Local Integration Settings' section has a 'View' dropdown set to 'Failover Servers Settings' and a 'Set Defaults' button. Below this is a table with two columns: 'Name' and 'Value'.

Name	Value
Secondary SIP Server Address	
Secondary SIP Server Port	5060

At the bottom of the dialog, there are two buttons: 'Add Failover Server' and 'Remove Failover Server'.

- 5 Click the **Add Failover Server** button. Two new rows are added to configure the secondary SIP server.
- 6 In the **Secondary SIP Server Address** and **Secondary SIP Server Port** rows, enter the appropriate value as follows:

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

Field	Value
Secondary SIP Server Address	Enter the TCP/IP address or an FQDN of the secondary node.

**For example:**

The IP address 123.45.6.789 as displayed on the Review/Modify SIP Gateway screen.

**NOTE** Alcatel-Lucent OmniPCX Enterprise CS 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.

**IMPORTANT** This value must match the configuration on the Gateway of the secondary node.

Secondary SIP Server Port      The default value is **5060**.

- 7 From the **View** dropdown list, select **Integration Specific Parameters**. The **Integration Specific Parameters** view displays.

The screenshot shows the 'Integration Options' dialog box with the 'Integration Specific Parameters' view selected. The 'View' dropdown is set to 'Integration Specific Parameters'. The table below lists the parameters and their values as shown in the interface.

Name	Value
Enable SIP server failover	<input type="checkbox"/>
Delay (in MS) between Failover attempts	1000
Incoming off hook delay	800
Outgoing off hook delay	0
On hook delay	300
Session Expires value in seconds	1800
Minimum acceptable value of Session Expires	90
Trim After DTMF	0
Disconnect Trim (Out-of-Band)	0
Base ASR Sensitivity (Internal)	5
Base ASR Sensitivity (External)	5
Local Contact Address	
Type of call progress to use for external calls	Digital
Use Single Channel for Monitor Transfers	<input type="checkbox"/>
Use RFC5589 semi-attended transfer	<input checked="" type="checkbox"/>

- 8 In the **Integration Specific Parameters** list, enter the information as shown in the following table. If you don't see the option(s) explained in this table, scroll down the list until you find the parameter.

Table 7. Integration Specific Parameters

Field	Value
Enable SIP server failover	Select this check box to allow for failover and to enable the failover server setting changes.
Delay (in ms) between Failover attempts	The delay in milliseconds before MiCollab AM attempts to register its port with the SIP server. The default is <b>1000</b> ms.
Incoming off hook delay	800
Outgoing off hook delay	0
On hook delay	300
Media streaming port number	48004
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 Failover Server:

- 1 From **MiCollab AM Configuration**, click the **Integrations** tab.
- 2 From the **Integrations** list, select **Alcatel-Lucent Enterprise OmniPCX Enterprise Communication Server SIP**, and then click **Edit**.
- 3 In the **Integration Options** dialog box, go to the **Local Integration Settings** section.
- 4 From the **View** dropdown 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. 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 card connected to the PBX.

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

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

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

## Windows Server 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 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**.



# Load Balancing Two or More Alcatel-Lucent Enterprise Nodes

If you are trying to load balance between two or more Alcatel-Lucent Enterprise nodes, follow the steps described in this chapter.

## MiCollab AM Setup

In **MiCollab AM Configuration** > **Integrations** tab, configure the Alcatel-Lucent Enterprise OmniPCX Enterprise Communication Server SIP integration as follows:

### Integration Options – Required Parameters View

**Integration Options**

System Switch: Alcatel-Lucent Enterprise OmniPCX Enterprise

Integration Type: SIP

Integration: Alcatel-Lucent Enterprise OmniPCX Enterprise

Name: OmniPCX Enterprise Communication Server SIP

Local Integration Settings

View: Required Parameters

Name	Value
SIP Server Address	
SIP Server Port	5060
SIP Domain Name	
Transport for outgoing SIP messages	UDP
SIP Device Name	
PBX Registration password	
Registration period in seconds	90
Local IP Address to bind on	fdbb:552b:a82a:500a:40c4:aec1:798c:d...
SIP Local Connection Port	5060

SIP Domain Name	
Transport for outgoing SIP messages	UDP
SIP Device Name	
PBX Registration password	
Registration period in seconds	90
Local IP Address to bind on	fdbb:552b:a82a:500a:40c4:aec1:798c:d...
SIP Local Connection Port	5060
SIP parser qualifier string	
Media packet size (milliseconds)	20

- **SIP Server Address:** Enter the Fully Qualified Domain Name of the Alcatel-Lucent Enterprise PBX.
- **SIP Domain Name:** Enter the Alcatel-Lucent Enterprise PBX SIP domain.
- **SIP Device Name:** Enter the external voicemail directory number.
- **SIP Parser qualifier string:** Enter the MiCollab AM domain name used by the DNS SRV records (more information about the DNS configuration is provided in the [DNS Configuration](#) section).

## Integration Options – Integration Specific Parameters View

The screenshot shows the 'Integration Options' dialog box with the 'Integration Specific Parameters' view selected. The 'View' dropdown is set to 'Integration Specific Parameters' and the 'Set Defaults' button is visible. The table below lists the parameters and their values:

Name	Value
Minimum acceptable value of Session Expires	90
Trim After DTMF	0
Disconnect Trim (Out-of-Band)	0
Base ASR Sensitivity (Internal)	5
Base ASR Sensitivity (External)	5
Local Contact Address	
Type of call progress to use for external calls	Digital
Use Single Channel for Monitor Transfers	<input type="checkbox"/>
Use RFC5589 semi-attended transfer	<input checked="" type="checkbox"/>

- **Local Contact Address:** MiCollab AM domain name used by the DNS SRV records (more information about the DNS configuration is provided in the [DNS Configuration](#) section).

## Alcatel-Lucent Enterprise PBX Setup

Configure a SIP gateway (in mgr: SIP→SIP Gateway):

```

+-Review/Modify: SIP Gateway-----+
|
|      Node Number (reserved) : 1
|      Instance (reserved) : 1
|      Instance (reserved) : 1
|
|      SIP Subnetwork : 10
|      SIP Trunk Group : 11
|      IP Address : 172.16.22.10
|      Machine name - Host : alcatel-oxe
|      SIP Proxy Port Number : 5060
|      SIP Subscribe Min Duration : 1800
|      SIP Subscribe Max Duration : 86400
|

```

```

|           Session Timer : 1800
|           Min Session Timer : 900
|           Session Timer Method + RE_INVITE
|           DNS local domain name : blvu.avstlabs.local
|           DNS type + DNS SRV
|           SIP DNS1 IP Address : 172.16.1.22
|           SIP DNS2 IP Address : -----
|           SDP in 18x + True
|           Cac SIP-SIP + False
|           INFO method for remote extension + False
|           Dynamic Payload type for DTMF : 97
|-----+

```

- **DNS local domain name:** The name of the local domain.
- **SIP DNS1 (or 2) IP Address:** Address(es) of the local DNS server(s)

Configure an external SIP gateway (in mgr: SIP→ SIP Ext Gateway):

```

+--Review/Modify: SIP Ext Gateway-----+
|
|           Node Number (reserved) : 1
|           Instance (reserved) : 1
|           SIP External Gateway ID : 10
|
|           Gateway Name : AVST
|           SIP Remote domain : CXSIP.blvu.avstlabs.local
|           PCS IP Address : -----
|           SIP Port Number : 5060
|           Transport type + UDP
|           Belonging Domain : -----
|           Registration ID : -----
|           Registration ID P Asserted + False
|           Registration timer : 0
|           SIP Outbound Proxy : -----
|           Supervision timer : 0
|           Trunk group number : 11
|           Pool Number : -1
|           Outgoing realm : -----
|           Outgoing username : -----
|
|           Outgoing Password : -----
|           Confirm : -----
|
|           Incoming username : -----
|
|           Incoming Password : -----
|           Confirm : -----
|
|           RFC 3325 supported by the distant + True
|           DNS type + DNS SRV
|           SIP DNS1 IP Address : 172.16.1.22
|           SIP DNS2 IP Address : -----
|           SDP in 18x + False
|           Minimal authentication method + SIP None
|           INFO method for remote extension + False
|           Send only trunk group algo + False
|           To EMS + False
|           SRTP + RTP only
|           Routing Application + False
|           Ignore inactive/black hole + False
|

```

```

|           Contact with IP address + False
|       Dynamic Payload type for DTMF : 97
|           100 REL for Outbound Calls + Supported
|           100 REL for Incoming Calls + Not Requested
|           Gateway type + Standard type
|       Re-Trans No. for REGISTER/OPTIONS : 2
|           P-Asserted-ID in Calling Number + False
|           Trusted P-Asserted-ID header + True
|       Diversion Info to provide through + History Info
|           Trusted From header + False
|-----+

```

- **SIP External Gateway ID:** ID of the SIP gateway configured previously.
- **SIP Remote domain:** the subdomain of the MiCollab AM system.
- **Trunk group number:** the SIP Trunk Group in the previously configured SIP gateway.
- **SIP DNS1(or 2) IP Address:** Address(es) of the local DNS server(s).

Configure the external voicemail (in mgr: Applications→ External Voice Mail):

```

+-Review/Modify: External Voice Mail-----+
|
|       Node Number (reserved) : 1
|       Instance (reserved) : 1
|       Voice Mail Dir.No. : 5000
|
|           Sub Type + Private
|       Directory Name : AVST_SIP
|       Connection COS : 0
|       Public Network COS : 2
|       Entity Number : 1
|       Cost Center ID : 255
|       Charging COS + Justified
|       URL UserName : 5000
|       URL Domain : alcatel-oxe
|       PCS IP Address : -----
|       SIP Authentication : 5000
|
|           SIP Passwd : ****
|           Confirm : ****
|
|       Register On Line Number : -----
|       Register URL (Username) : -----
|       Register URL (Domain) : -----
|       Register Authentication : -----
|
|           Register Password : -----
|           Confirm : -----
|
|       External Gateway Number : 10
|       Subscription on registration + False
|-----+

```

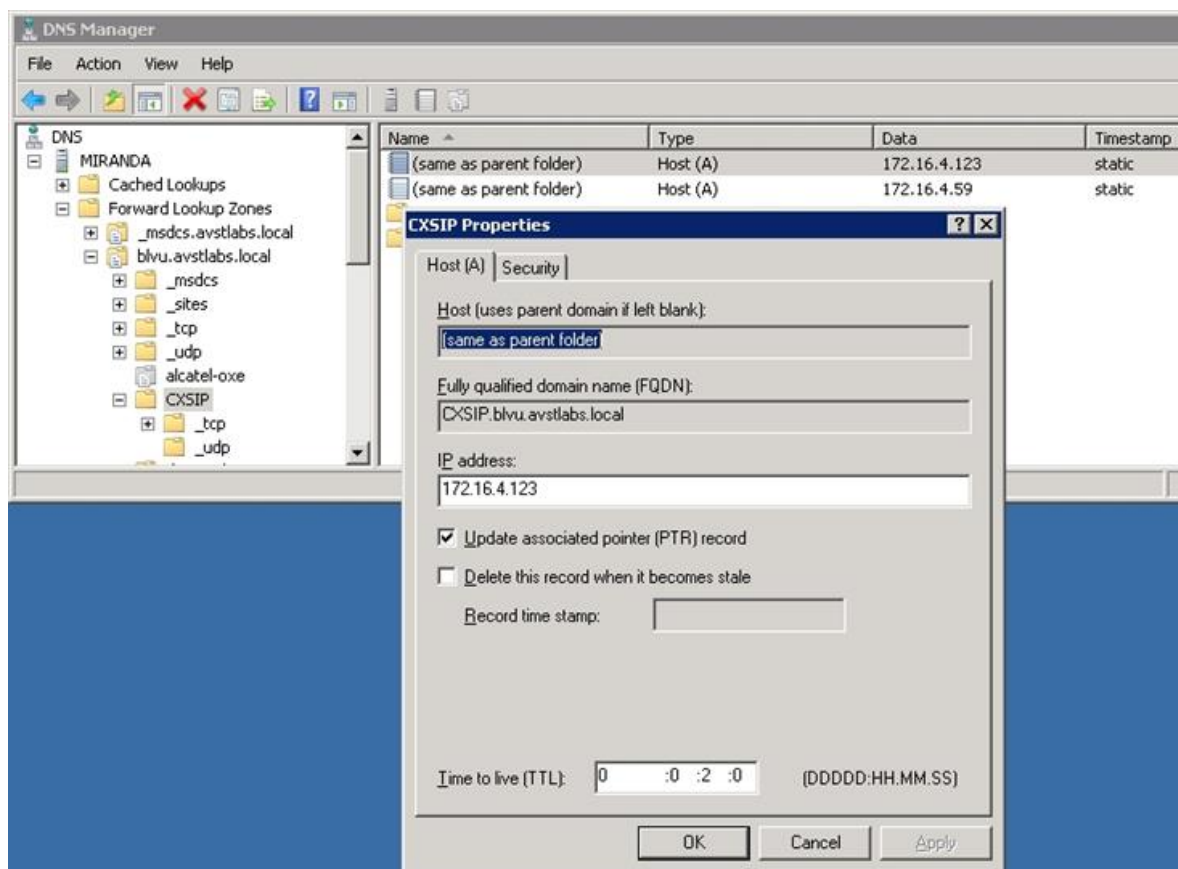
- **Voice Mail Dir. No:** The hunt group access code of the MiCollab AM system.
- **URL UserName:** The hunt group access code of the MiCollab AM system.

- **URL Domain:** The subdomain of the Alcatel PBX.
- **External Gateway Number:** The SIP External Gateway ID of the previously configured external SIP gateway.

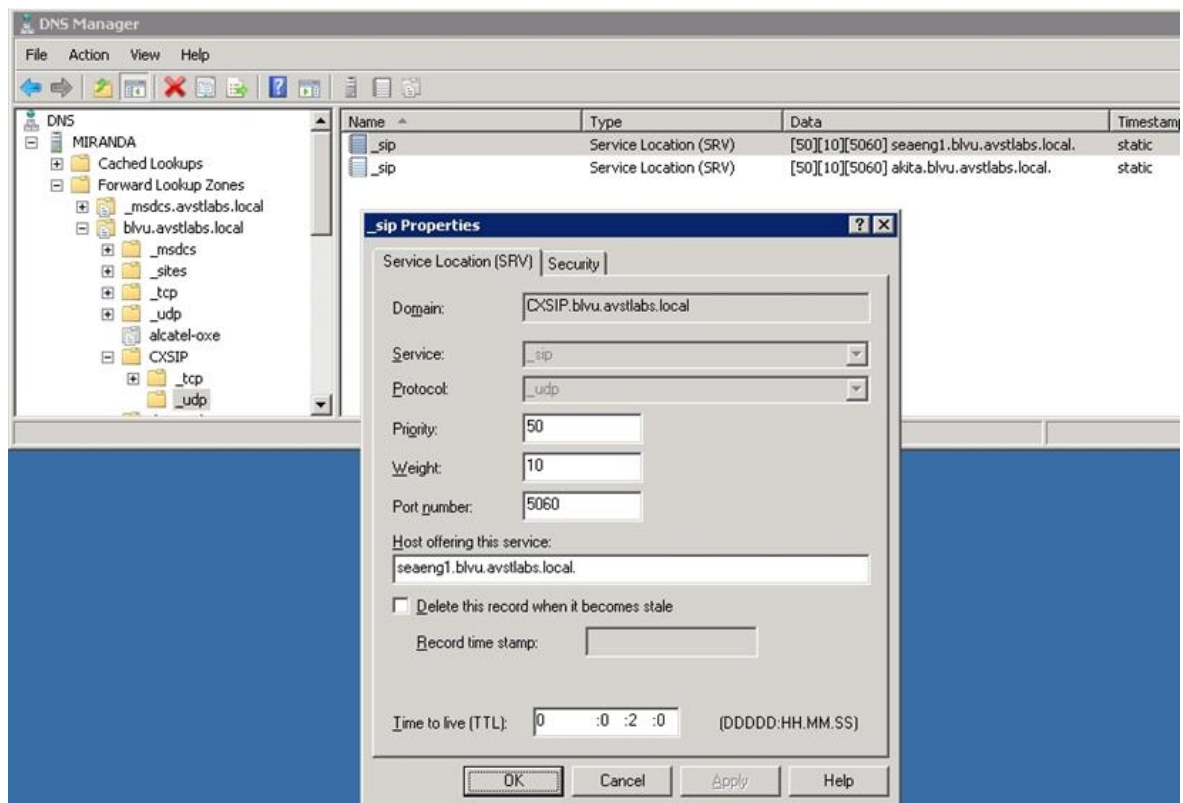
## DNS Configuration

To configure DNS:

- 1 Configure a subdomain for the MiCollab AM systems (in this example, CXSIP.blvu.avstlabs.local).
- 2 Create DNS A entries to resolve the subdomain to the IP addresses of the MiCollab AM Call Servers. Set the TTL to 2 minutes.



- 3 Create DNS SRV entries to resolve **\_udp.\_sip.CXSIP.blvu.avstlabs.local**. Set the TTL to 2 minutes.



**NOTE** The TTL for the DNS A and SRV records have to be the same; the Alcatel-Lucent Enterprise PBX will use that TTL to update its DNNS cache. In case of a MiCollab AM Call Server going down inside the TTL window, it will be blacklisted by the Alcatel-Lucent Enterprise PBX will start routing the calls to it again.

# Configuring Quality of Service (QoS)

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

Table 8. QoS Configuration

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