

MiCollab Advanced Messaging SIP Routing Manager System Administrator Guide

For version 9.1 and above

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Preface

This guide describes how to install and configure the SIP Routing Manager for your integration.

This guide is written for Mitel-certified MiCollab Advanced Messaging (MiCollab AM) administrators and technicians who are familiar with MiCollab AM procedures and terminology, the MiCollab AM Configuration utility, and the Microsoft Windows® operating system, have a working knowledge of TCP/IP protocols, as well as a working knowledge of domain administration in a Windows Server environment.

Before implementing any procedures in this guide, ensure that MiCollab AM software is installed and running successfully.

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.

Frequently Used Terms

Table 1. Frequently Used Terms

Terms	Description
System Server	<p>Term refers to an organization's computer platform(s) that have MiCollab AM software installed and handles the core system functions such as storing messages, database.</p> <p>It can also refer generically to the System Server platform, the Call Server platform, or both. The term is most often used to describe a software or hardware installation or configuration practice where the role of the server platform is not specifically expressed.</p>
Call Server	<p>Term refers to an organization's computer platforms that have MiCollab AM software installed and serve as the interface to the system (PBX). The Call Server(s) interface with the System Server for the purpose of accessing messages, and database.</p>

Overview

MiCollab AM integrates with various VoIP telephone switches using the SIP (Session Initiation Protocol) to provide a seamless telephone interface through a network connection to both the subscriber and the outside caller. While the SIP protocol is an IETF standard, the methods MiCollab AM uses to communicate with the SIP Server of each telephone system can differ substantially. MiCollab AM, through its various integration configuration parameters, accommodates the diverse requirements of these SIP Servers.

The SIP Routing Manager is a Windows® Service that further enhances the integration capabilities of MiCollab AM to negotiate with the various SIP Servers with which MiCollab AM integrates by providing:

- A seamless SIP integration to multiple Call Servers
 - Load Balancing
 - Redundancy
 - Scalability
- Assistance in locating SIP Servers through DNS (Domain Name Server) SRV requests, and NAPTR (Name Authority Pointer) records
- The use of multiple SIP domains
- Keep Alive messages
- Multiple SIP routes
- Multiple hunt group numbers
- Multiple SIP integrations

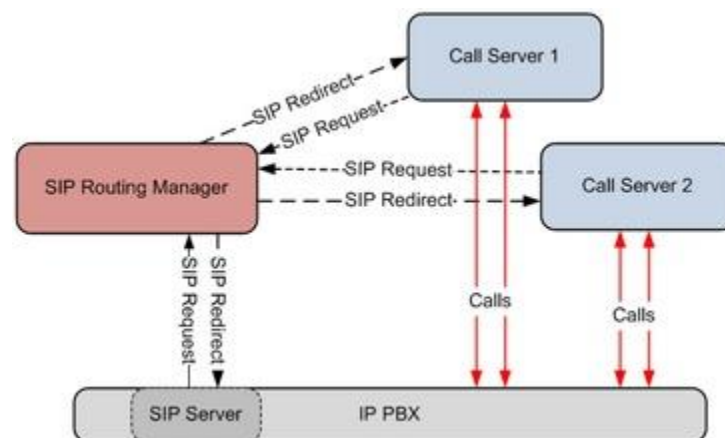


Figure 1. SIP Routing Manager Redirecting IP PBX and Call Server SIP Messages

The SIP Routing Manager acts as a redirect server. It receives the initial SIP messages sent from the IP PBX and MiCollab AM, redirects the IP PBX and MiCollab AM messages to the appropriate route or endpoint by including the appropriate destination address in the redirection response, and then steps out of the exchange, allowing the IP PBX and MiCollab AM to negotiate the call connection.

SIP Integration Types

MiCollab AM integrates with various SIP servers or IP PBXs using the SIP protocol. Though the underlying communication protocol is SIP, each SIP Server may use a different method of deployment. One of the areas in which the IP PBXs differ from one another is the registration process. Registration is a process by which a MiCollab AM port informs the SIP Server or IP PBX of its location. Once a port registers successfully, the switch can route calls to it. There are three different categories based on the type of registration process in which MiCollab AM integrations are classified. These categories are:

- **SIP Station integration (port / station / line)** - MiCollab AM registers each of its ports separately with the SIP Server (Registrar). The IP PBX performs the hunting and calls are presented to MiCollab AM seamlessly across multiple Call Servers.

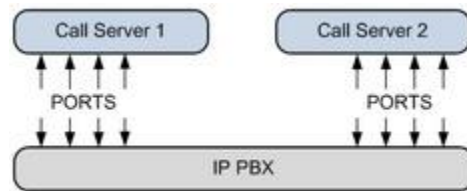


Figure 2. SIP Station Integration: PBX hunting

- **SIP Trunk integration (dynamic) (device / hunt / pilot number / user)** - MiCollab AM registers a single entity with the SIP Server (Registrar). This entity collectively represents all MiCollab AM ports on the SIP Server. Calls route to this single entity. MiCollab AM performs the hunting. Calls do not route to multiple Call Servers. Alternate routes assigned within the PBX may not forward calls in a timely fashion.
- **SIP Trunk integration (static)** - MiCollab AM does not register any of its ports. The SIP Server routes calls to a static route. This type of configuration allows the least amount of integration flexibility. The IP PBX simply routes the calls based on a single registration or static route. MiCollab AM performs the hunting. Calls do not route to multiple Call Servers. Alternate routes assigned within the PBX may not forward calls in a timely fashion.

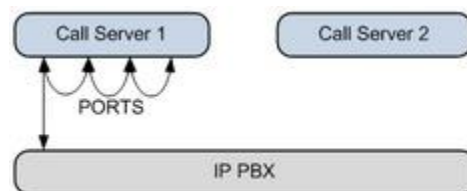


Figure 3. SIP Trunk Integration: MiCollab AM hunting

This Installation Guide documents the various types of MiCollab AM SIP integrations, the implementation techniques for each integration type, the installation of the SIP Routing Manager, and the procedures for administering the SIP Routing Manager to work with MiCollab AM SIP integrations.

Use this document in conjunction with the *System Installation and Configuration Guide*, the *System Administration Guide*, and the Integration Technical Note specific to your integration, and the MiCollab AM online help system. For information about each individual VoIP telephone system, please refer to the specific telephone system manufacturer's documentation.

What is the SIP Routing Manager

The SIP Routing Manager is an Mitel software product that acts as an intermediary between the VoIP PBX SIP Server and MiCollab AM. The SIP Routing Manager supports IP PBX switches that do not route calls to multiple MiCollab AM Call Servers. Directing all calls from the IP PBX to the SIP Routing Manager allows the SIP Routing Manager to make the routing decisions to the appropriate Call Server. This provides:

- Scalability – You can add Call Servers to the system to meet the call handling requirements of the enterprise.
- Availability – The SIP Routing Manager supports routing algorithms to perform load balancing within the system, which allows calls to distribute evenly across a group of Call Servers.
- Redundancy – The use of multiple Call Servers allows administrators to take a Call Server out of service for maintenance or to perform service upgrades without losing call-processing capabilities.

The SIP Routing Manager works as a redirect server; it redirects all SIP calls within the SIP domain. The SIP Routing Manager processes the incoming SIP requests for the SIP Domain by generating a redirect response; it does not directly send a SIP message to the destination. Instead, it provides information about the destination back to the originator. The SIP Routing Manager does not remain in the signaling path of the call session; it redirects the originator to the correct destination.

SIP Routing Manager Server Requirements

The SIP Routing Manager is a Service that runs on a Windows server platform.

- The SIP Routing Manager Service can run on the MiCollab AM System Server or a stand-alone Windows Server 2012 R2, Windows Server 2016 (Server with Desktop Experience), or Windows Server 2019 (Server with Desktop Experience) platform. (Call Services and Lines are not required to run SIP Routing Manager on the System Server).
- The server must be a member of the same network on which the VoIP SIP Server, IP PBX, and MiCollab AM communicate.
- A separate network interface card (NIC) must be installed in the server platform if the SIP integration is on a separate network subnet or VLAN than the primary network connection.

IMPORTANT The primary (public) network interface card (NIC) must be the first network connection in the network binding order of the System Server.

NOTE Running the SIP Routing Manager Service on a server within a Neverfail® cluster ensures continuous operation in the event of a platform failure.

Recommendations

- If your System Server is running Call Services with Lines and is handling large call volumes, it may be practical to install the SIP Routing Manager on a separate server platform.
- If you are running MiCollab AM in a Neverfail cluster, install the SIP Routing Manager on the System Server.
- If you are running a stand-alone server in a Neverfail cluster, install the SIP Routing Manager on the stand-alone platform.

Installing SIP Routing Manager

Follow the procedures in this section to install SIP Routing Manager Service.

To install the SIP Routing Manager Service:

- 1 Log on to the platform using a Windows Administrator account.
- 2 Shut down all running programs, including MiCollab AM.
- 3 Insert the MiCollab AM Installation Media into the appropriate drive.
- 4 Do one of the following.

Table 2. Autorun options

If autorun is...	Then...
Enabled	The MiCollab AM Installation Media displays. In Server Components area, click SIP Routing Manager , and then skip to Step 6
Not Enabled	On the taskbar select Start > Run > Browse , and then continue to Step 5.

- 5 Locate and open the Server Installs\Telephony Server folder, double-click **Setup**, and then click **OK**. The MiCollab AM Installation Media splash screen displays.
- 6 Click **SIP Routing Manager** in the Server components submenu. The InstallShield Wizard Welcome page displays.
- 7 On the Welcome page, click **Next**. The **License Agreement** dialog box appears.
- 8 Click **Yes** to accept the License Agreement. The **Choose Destination** dialog box appears.

IMPORTANT You must accept the terms of the license agreement to continue with Setup.

- 9 Click **Next** to continue with the next step in the installation, or click **Browse** to select a new destination, and then click **Next**. The **Start Copying** dialog box appears.
- 10 Click **Next** to continue. The installation process runs and the **InstallShield Wizard Complete** dialog box appears when it completes.
- 11 Click **Finish** to exit the Wizard.

Configuring the SIP Routing Manager

Once you have installed the SIP Routing Manager successfully, you must configure it for use with your SIP domain and integration. To configure the SIP Routing Manager successfully you must:

- Configure the Settings for the SIP domain
- Configure the local IP Addresses and the supported transports for the SIP Routing Manager
- Configure the SIP Domain Host for the SIP domain
- Configure a Numbering Plan for the SIP Domain
- Configure the Route(s) for the Numbering Plan
- Test the New Numbering Plan and Route
- Start the SIP Routing Manager Service

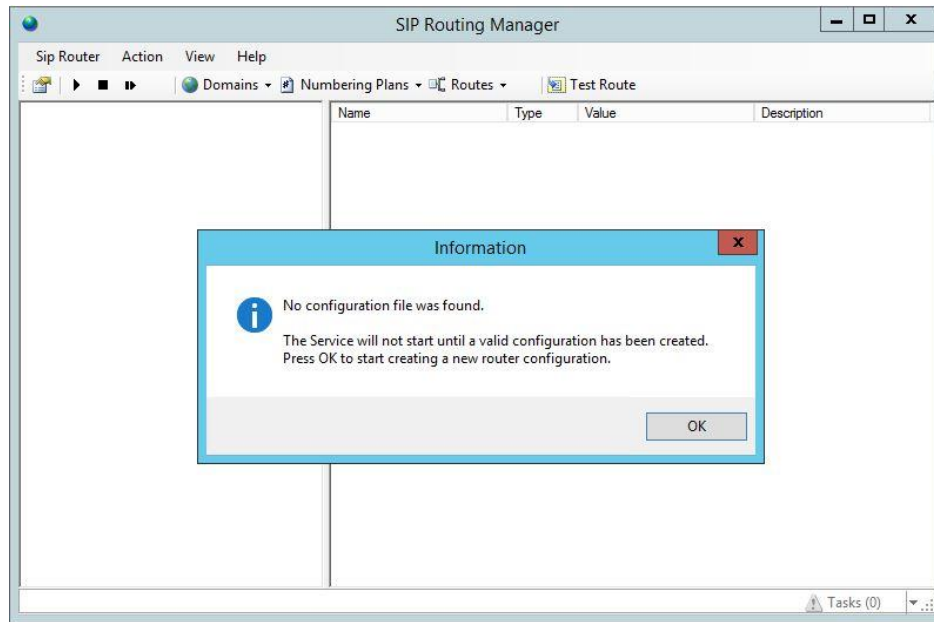
SIP Routing Manager Configuration

Once the SIP Routing Manager is installed, you must configure it for your application and environment. To start the SIP Routing Manager utility you must navigate to the drive and directory in which it was installed and locate the **AT_SIPRouterFrontEnd.exe** file. Double-click the file name to open the utility.

NOTE To alleviate the need to return to this folder each time, create a shortcut for the AT_SIPRouterFrontEnd.exe file, and then place it on the desktop.

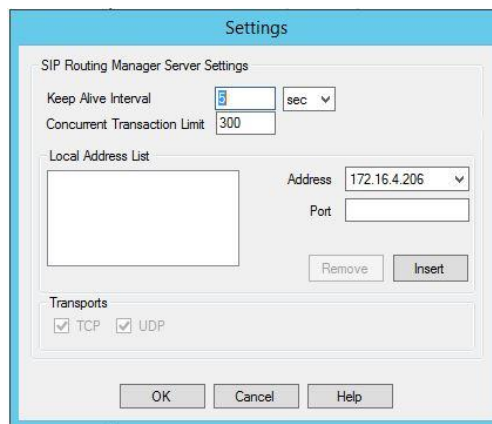
To start the SIP Routing Manager Server utility:

- 1 Navigate to the drive\directory that you installed the SIP Routing Manager. The default location is D:\Mitel\SIP Routing Manager.
- 2 Double-click the **AT_SipRouterFrontEnd.exe** file. SIP Routing Manager window appears.
- 3 If this is the first time the SIP Routing Manager utility starts, a pop-up message displays to advise you that the Service cannot start until a valid configuration is created.



- 4 Click **OK** to continue. The **Settings** dialog box appears.

NOTE The first time you start the SIP Routing Manager, the **Settings** dialog box displays in front of the SIP Routing Manager window as a configuration wizard. The wizard allows you to configure each dialog box for the application and site requirements.



- 5 The local server **Address** field is populated automatically with the IP address of the primary network connection. If you are using a separate network connection for the SIP integration, click the drop-down box, and then select the IP address that you are using for the integration.
- 6 In the **Port** field, enter a TCP port number. The default port for the SIP protocol is 5060.
- 7 Click **Insert** to add the IP address to the **Local Address List**.
- 8 Click **OK**. The **SIP Domain** dialog box appears.

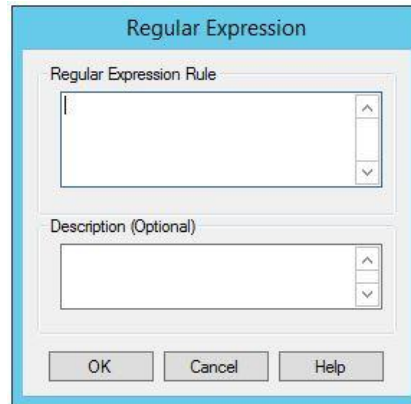
- 9 In the **SIP Domain Host** field, enter a SIP Domain Host. A valid SIP Domain Host entry is one of the following formats:
 - [Chicago.com](#) – If the PBX is configured to use a domain
 - SIPRouterAddress - If the PBX changes the original Request-URI and uses the destination IP Address
 - PBXAddress - If the PBX uses its IP Address as a SIP Domain.
- 10 In the **Description** field, enter a description of the domain. For example, Avaya/Nortel CS1000 SIP Integration describes a PBX and the integration type.
- 11 Click **Add**. The **Numbering Plan** dialog box appears.

- 12 In the **Name** field, enter a name for the Numbering Plan.
- 13 Select the **Fault Tolerant** checkbox if you want to ensure all routes in this numbering plan are valid based on periodic keep-alive messages that determine if the routes are operational.
- 14 In the **Routing Mode** field, select the routing mode used for this numbering plan from the drop-down list.
 - **Round Robin** - Routes are selected from the routing list in a cyclic fashion
 - **Overflow** - Routes are selected based on their priority. Highest priority routes receive calls first

- **Least loaded** - The least loaded route is selected from the list of active routes. The SIP Routing Manager collects capacity information using the SIP OPTIONS message and the keep-alive mechanism

15 In the **Description** field, enter a description for the numbering plan.

16 In the **Rules** area, click **Add**. The **Regular Expression** dialog box appears.



The image shows a dialog box titled "Regular Expression". It contains two text input fields. The first field is labeled "Regular Expression Rule" and the second is labeled "Description (Optional)". Both fields have vertical scroll bars on their right sides. At the bottom of the dialog box, there are three buttons: "OK", "Cancel", and "Help".

17 In the **Regular Expression** area, enter a pattern-matching rule to allow the SIP Routing Manager to process the incoming message.

IMPORTANT Every numbering plan must have at least one pattern-matching rule. Numbering plans may have multiple rules. For example, rules must allow calls to all destinations within the SIP domain. For example, destinations must include MiCollab AM, the PBX, and subscriber soft phones.

For example:

The following regular expressions may be used to create a numbering plan to process calls destined to 5000.

5000

5(0{3})

To match any 4-digit number with 5 as the leading digit:

5[0-9][0-9][0-9]

5[0-9]{3}

^5\d{3}\$

To specify a range of numbers with 5 as the leading digit:

5[0]{2}[0-8] // 5001-5008

[5]{1}[0]{1}(((0-1){1}[0-9]{1})|((2){1}[0-4]{1})) // 5001-5024

[5]{1}[0]{1}(((0-8){1}[0-9]{1})|((9){1}[0-6]{1})) // 5001-5096

[5]{1}[0]{1}(((0-4){1}[9]{1})|((5-8){1}[0-9]{1})|((9){1}[0-6]{1})) // 5049-5096

To specify a range using wildcard characters:

. * // Match any number of digits/characters

.{4} // Match 4 digits/ characters only

NOTE The dialog box accepts white spaces for readability purposes. In addition, the dialog box validates the syntax of the expression before accepting the expression. It cannot, however, prevent syntactically valid but incorrect expressions. Use the SIP Routing Manager's Test Tool to test the configuration.

- 18 In the **Description** field, enter a description for this regular expression rule.
- 19 Click **OK** after you have entered all of the expression rules for this numbering plan. The **Numbering Plan** dialog box displays with the new Rules Expressions added.

- 20 In the **Routes** area, click **Add**. The **Route** dialog box appears.

- 21 In the **Destination Address** field, enter the IP Address or FQDN of the destination. In this example, the destination is *Call Server 1*.

- 22** In the **Port** field, enter the TCP port to use. The default is 5060.
- 23** In the **Description** field, enter a description for this route. This description displays in the list view of the route.
- 24** Select the Preferred transport for this route, **UDP** or **TCP**.
- 25** In the **Rules** area, click **Add** if you want to add a regular expression rule for this route.
- 26** In the **SIP Methods** area, select the SIP Message types to use on this route. The route may use multiple SIP Methods.

Table 3. SIP Methods

SIP Method	Event Package
INVITE	
REGISTER	
REFER	Refer
SUBSCRIBE	Message-summary
NOTIFY	Refer Message-summary

For example, assume the SIP Routing Manager is configured to work with a Registrar. The route that points to the registrar should only have REGISTER in the supported methods list. Similarly, when configuring it to work with a PBX that supports unsolicited NOTIFY request for message waiting, a NOTIFY method should be added to the list.

Different SIP methods implement different features. For example, you can use the SUBSCRIBE and NOTIFY methods to implement message waiting. Additionally, you can use these two methods to convey presence related information. The distinction is made with the use of another SIP header in the request. This header is referred to as the "Event-Package." "Message-summary" and "Refer" are header examples of the event package.

A SIP Method can be associated with an event package although not all SIP methods require an event package. When selecting a SIP Method, the event package drop-down menu is enabled if the selected SIP Method requires an event package associated with it.

- 27** Click **Insert** to insert the SIP method and Event Package into the **SIP Methods** area.
- 28** If you are using Fault Tolerant keep-alive messages for this route, enter the name or number used when sending these messages in the **SIP Account Name** field. The account name is used in the header of the SIP Options message. For example, the account name may be the pilot number of the hunt group.
- 29** In the **Keep Alive Interval** field, enter the time interval to send keep-alive messages in seconds or milliseconds. This field overrides the default SIP Routing Manager **Keep-Alive Interval** field of the **Settings** dialog box.

- 30** In the **Maximum Capacity** field of the **Destination Capacity** area, enter the maximum capacity (number of lines) of the destination server in the **Maximum Capacity** field. For MiCollab AM, this is the total number of SIP lines on the Call Servers.

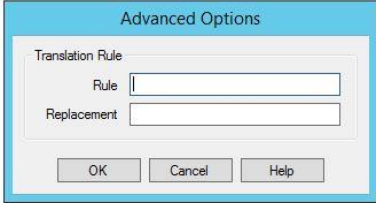
NOTE The destination capacity information allows the SIP Routing Manager to implement load-balancing algorithms prior to receiving a response from the first keep-alive request. The keep-alive response message contains the most current capacity information from the route used to route subsequent calls. The value in this field is used on routes that do not provide capacity information through keep-alive messages.

- 31** In the **Maximum Route Capacity** field, enter the maximum capacity of endpoints this route can serve to the destination.

NOTE The maximum route capacity is the number of lines configured to handle calls for this route or integration. For MiCollab AM, this is the total number of SIP lines configured to participate in the integration. If a single SIP integration is configured on the Call Server only, this value is equal to the Maximum Capacity parameter.

NOTE Steps 29 through 32 are optional steps, they are not required to configure a valid SIP domain.

- 32** If you want to add a Translation Rule, click the **Advanced** button. The **Advanced Options** dialog box appears.



You can configure a route optionally to perform a name or number translation. When a route is selected to redirect a message, the SIP Routing Manager can perform an additional name or number translation. For example, you can redirect an incoming message destined to 895000, by creating a translation rule to change the destination number 5000, 4259455000, or 14259455000.

- 33** In the **Rule** field, enter the rule you want to modify with the replacement value.
- 34** In the **Replacement** field, enter the modified output value of the rule.

For example:

Rule: "89"

Replacement: ""

Input: 895000

Output: 5000

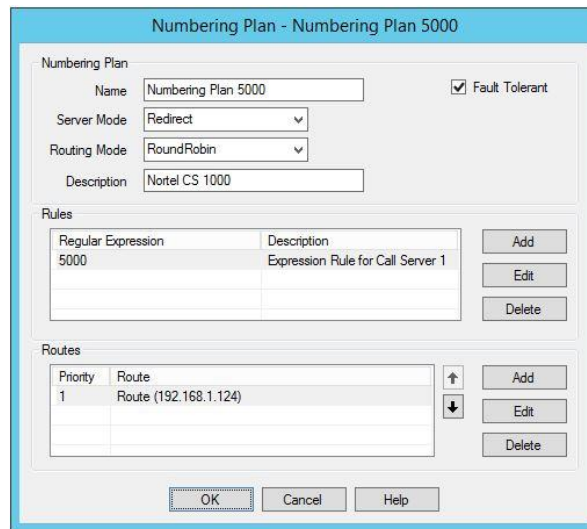
Rule: ^7[0-9]{3}\$ or ^7\d{3}\$

Replacement: +14259457\$1

Input: 7234
Output: +14259457234

Rule .{4}
Replacement: 4800
Input: 4000
Output: 4800

- 35 Click **OK** to save and close the **Advanced Options** dialog box.
- 36 Click **OK** to save and close the **Route** dialog box. The **Numbering Plan** dialog box displays with the saved route information.



The dialog box titled "Numbering Plan - Numbering Plan 5000" contains the following fields and sections:

- Numbering Plan**
 - Name: Numbering Plan 5000
 - Server Mode: Redirect
 - Routing Mode: RoundRobin
 - Description: Nortel CS 1000
 - ☒ Fault Tolerant
- Rules**

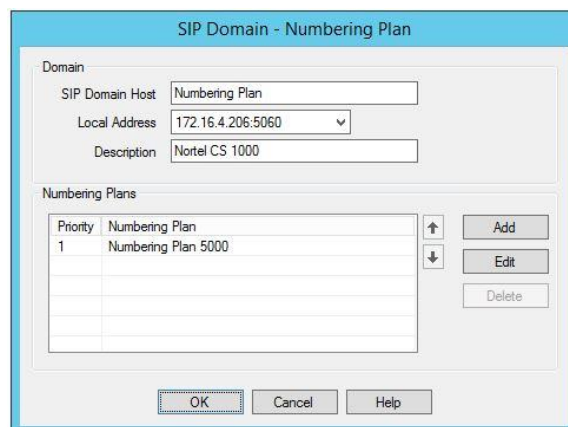
Regular Expression	Description
5000	Expression Rule for Call Server 1

Buttons: Add, Edit, Delete
- Routes**

Priority	Route
1	Route (192.168.1.124)

Buttons: Add, Edit, Delete
- Buttons: OK, Cancel, Help

- 37 To add another route to this numbering plan click **Add** in the **Routes** area and follow the procedures beginning at step 17. If you are finished adding routes to this numbering plan, click **OK**. The **SIP Domain** dialog box with the new Numbering Plan information appears.



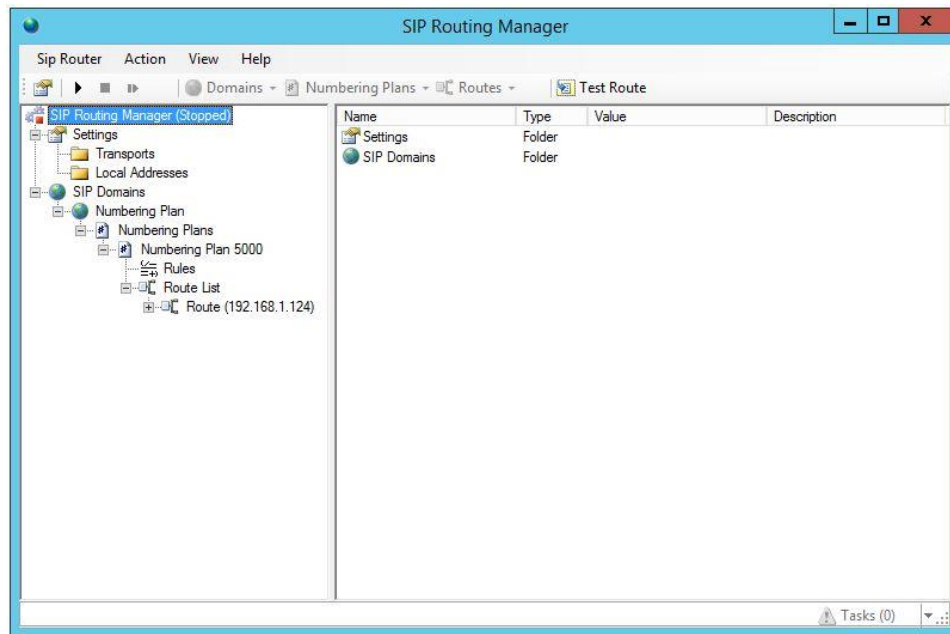
The dialog box titled "SIP Domain - Numbering Plan" contains the following fields and sections:

- Domain**
 - SIP Domain Host: Numbering Plan
 - Local Address: 172.16.4.206:5060
 - Description: Nortel CS 1000
- Numbering Plans**

Priority	Numbering Plan
1	Numbering Plan 5000

Buttons: Add, Edit, Delete
- Buttons: OK, Cancel, Help

- 38 To add another Numbering Plan click **Add** and follow the procedures beginning at step 8. Click **OK** to save the new SIP Domain configuration. The **SIP Routing Manager** window displays with the new Numbering Plan configuration.



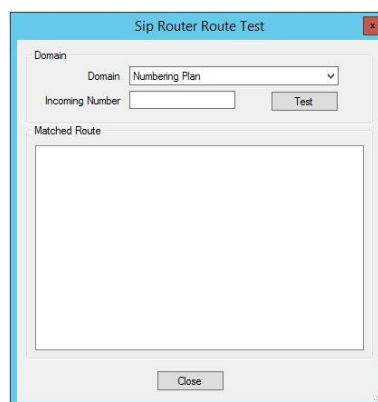
- 39 Once the initial configuration is complete it is recommended that you save a copy of the configuration file in a backup location. Navigate to the drive\directory that you installed the SIP Routing Manager. The default location is D:\Mitel\SIP Routing Manager.
- 40 Locate the file SipRouterConfig.xml, and then copy it to a backup location. This file contains the configuration settings you just created in the previous steps. You should back up this file each time you make changes to the SIP Routing Manager configuration.

Testing the New Route

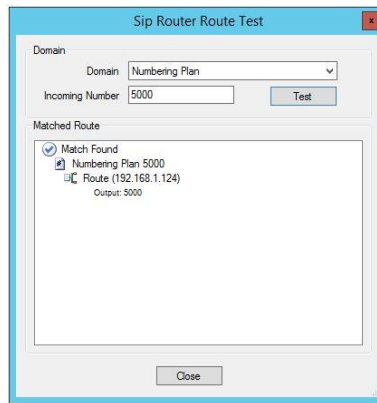
You can test the new numbering plan and route configuration using the SIP Routing Manager's Test Route Tool.

To test the new route:

- 1 On the SIP Routing Manager toolbar, click **Test Route**. The **SIP Router Route Test** dialog box appears.



- 2 In the **Incoming Number** field, enter the Regular Expression Rule you want to test.
- 3 Click **Test**. The SIP Routing Manager performs the test, and then displays the results in the **Matched Route** area.



- 4 Click **Close** to close the dialog box.

Starting the SIP Routing Manager Service

Once you have completed the configuration of the SIP Routing Manager and have tested the Numbering Plans and Routes for your integration, you must start the SIP Routing Manager Service.

From the menu bar of the SIP Routing Manager click **Action**, and then click **Start Service**. The Service is started.

NOTE Once a valid configuration exists and the Service is initially started, it is configured to start automatically by default.

Completing the Integration with the SIP Routing Manager

If you have not previously configured and programmed the IP PBX and each Call Server in the system for the integration, you must complete these tasks now. Follow the procedures in the related Integration Technical Note for your particular integration to perform these tasks. Test each route in the integration to ensure calls are directed to the correct MiCollab AM lines, verify message-waiting indicators are operational, and ensure that MiCollab AM can perform callouts to the telephone numbers that are authorized for the enterprise before you make MiCollab AM available to end users.

Administering the SIP Routing Manager

The SIP Routing Manager Server administration utility allows you to add, configure, and test SIP domains, Numbering Plans and Routes for MiCollab AM SIP integrations. In addition, you can edit existing configurations or remove them completely from the Service.

The SIP Routing Manager Server Utility

The SIP Routing Manager Server utility is the starting point for any task you perform within the SIP Routing Manager utility.

To start the SIP Routing Manager utility:

From the server taskbar, select **Start > All Programs |MiCollab AM Desktop**, and then click **SIP Routing Manager**.

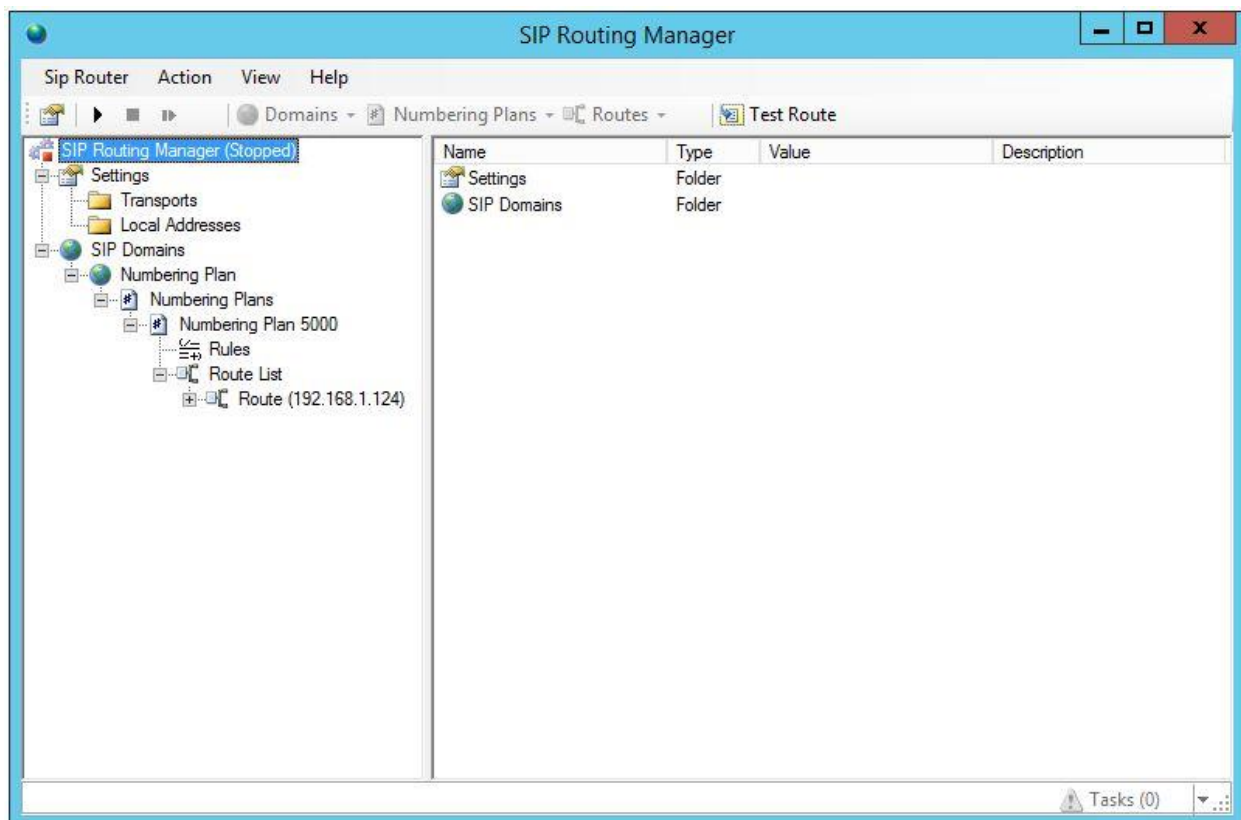


Figure 4. SIP Routing Manager

File Menu

SIP Router

Server Settings – Select to display the SIP Routing Manager Server Settings dialog box.

Exit – Select to close the SIP Routing Manager utility

Action

Start Service – Select to start the SIP Routing Manager Service

Stop Service – Select to stop the SIP Routing Manager Service

Restart Service – Select to restart the SIP Routing Manager Service

View

Expand All – Select to expand the tree view of the SIP domain list pane. The tree view displays all of the configured domains and their related numbering plans and routes.

Help

Contents – Select to open **Help**. Displays the contents of the help file and allows you to select a help topic from the contents, look-up a topic from the index, or use the search mechanism to search for a key word in a topic.

About – Select to display the **About SIP Routing Manager** dialog box. The dialog box displays the current software version and build information of the SIP Routing Manager.

Tool Bar

Buttons



- **SIP Routing Manager Server Settings** – Click to display the SIP Routing Manager dialog box.



- Click to start the SIP Routing Manager Service



- Click to stop the SIP Routing Manager Service



- Click to restart the SIP Routing Manager Service



- Highlight **Domains**, and then click to add or edit an existing Domain



- Highlight **Numbering Plans**, and then click to add or edit an existing Numbering Plan



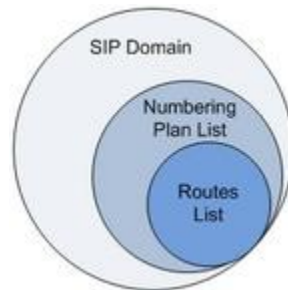
- Highlight **Routes**, and then click to add or edit an existing Route



- **Test Route button**, Click to display the **SIP Router Route Test** dialog box

SIP Domain Tree-View List Pane

The domain list pane on the left-hand side of the window displays a tree view of all of the SIP Routing Manager's Global Settings, the configured SIP Domains and their related numbering plans and routes. For a configuration to be valid, it must have at least one SIP Domain. A SIP Domain must have at least one numbering plan to which an incoming message is matched, and each numbering plan must have at least one route that points to a destination.



- To add a new SIP domain, right-click **SIP Domain** in the SIP Domain List pane, and then click **Add**. A new **SIP Domain** dialog box appears.
- To edit or delete a domain, right-click the domain to, and then click **Edit** to display the **SIP Domain** dialog box, or click **Delete** to remove it.
- To add a numbering plan, right-click **Numbering Plans**, and then click **Add**. A new **Numbering Plans** dialog box appears.
- To edit or delete a numbering plan, right-click the numbering plan, and then click **Edit** to display the **Numbering Plans** dialog box, or click **Delete** to remove it.
- To edit a rule, right-click the **Rules**, and then click **Edit**. The **Numbering Plans** dialog box appears.
- To add a route to the route list, right-click **Route List**, and then click **Add**. A new **Route** dialog box appears.
- To edit or delete a route, right-click the route, and then click **Edit** to display the **Route** dialog box, or click **Delete** to remove it.
- To edit a SIP Method, right-click the **SIP Method**, and then click **Edit**. The **SIP Method** dialog box appears.

SIP Domain Detail List Pane

The Detail List pane on the right-hand side of the window displays the details of the selected object in the Domain List pane. To view the details of an object, select the object in the Domain List pane. To edit any of the details, double-click the item you want to edit. The related dialog box appears.

Name – Displays the name of the folder, string or value of the selected object

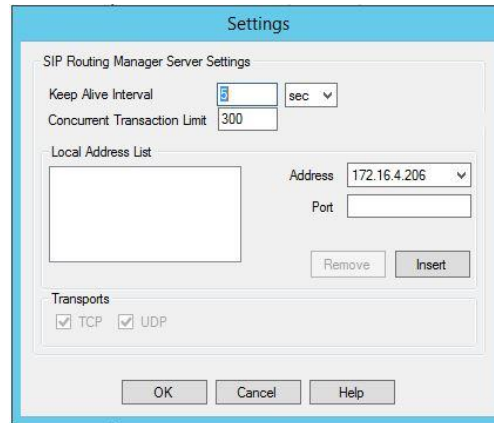
Type – Displays the type of value for the selected object

Value-Displays the value of the string or integer of the selected object

Description – Displays the description of the selected object

SIP Routing Manager Server Settings dialog box

The **Settings** dialog box allows you to configure the SIP Routing Manager Server settings. This dialog box contains global settings for the SIP Routing Manager. When you start the SIP Routing Manager for the first time this dialog box displays automatically as a configuration wizard.



SIP Routing Manager Server Settings

Keep Alive Interval – The global interval for keep-alive messages in seconds or milliseconds. Each route in a Route List has its own keep-alive parameter that may override this setting. The default setting is five seconds.

Keep Alive Mechanism - The keep-alive mechanism prevent calls from going to a route that is inactive; the SIP Routing Manager sends a keep-alive message that allows it to keep track of its configured routes. A route that fails to generate a response to the keep-alive request successfully is removed from the route selection.

The keep-alive mechanism is generated by a SIP OPTIONS request that is sent to all of the routes configured to use the Fault Tolerant setting of the **Routes** dialog box. If the destination does not respond to the request, the SIP Routing Manager assumes the route is disabled and the route is removed from the list of active routes. For example, if a Call Server is shut down for maintenance, the network connection failed, or the Call Server is down for unexpected reasons the route is disabled. It is possible that the route is mis-configured and the destination cannot respond successfully. In either event, the SIP Routing Manager continues to send an OPTIONS request periodically to the destination so that it can be put back on the active route list once it starts responding successfully again.

Concurrent Transaction Limit – The number of concurrent transactions the server supports. The default is 300. The default limit supports a maximum of 300 calls per second.

Local Address List

The Local Address list displays the list of local IP Addresses and the listening port for the SIP Routing Manager Server. The selected Address and Port are added to the list when you click Insert.

Address - Select the IP Address to use from the drop-down list. The drop-down list provides a list of all of the configured IP Addresses for the server. Select the IP address used to communicate with the integration. If you are using the SIP Routing Manager with multiple integrations, you may require multiple IP Addresses.

IMPORTANT If the SIP Routing Manager is installed on the MiCollab AM System Server; do not select the IP Address of the primary network connection unless you are communicating with the integration through the primary network connection. In most cases, a separate network interface card provides the network connection for IP integrations.

Port - The port is the local listening port for the SIP Routing Manager Server. The default port number is 5060.

Remove button - To remove an IP Address from the list, highlight the address, and then click Remove.

Insert button - To Insert an IP Address in to the list, select the IP Address from the list, select the Port, and then click Insert.

Transports

The TCP and UDP transports are available for selection within each route by default. You can define a route to use either the TCP or the UDP transport within each route configuration.

SIP Domain dialog box

The **SIP Domain** dialog box allows you to configure the SIP Domain Host, the local IP address, and allows you to create a Numbering Plan for the domain. The SIP Routing Manager requires a SIP Domain for every integration it serves.

IP PBX SIP Servers are configured with a SIP Domain name. This name distinguishes endpoints that are local to the server from those that are not. The SIP Routing Manager can communicate with multiple SIP domains. This allows it to integrate simultaneously with multiple SIP servers and to support multiple integrations.

NOTE The SIP Routing Manager uses the host portion of the "To" header of the incoming SIP message to extract the SIP domain Name.

Priority	Numbering Plan

Domain

SIP Domain Host – Enter the SIP Domain Host Address. The SIP Routing Manager receives SIP messages that are destined to an endpoint that belongs to a specific SIP Domain (integration). Typically, a SIP Domain is specified as the Host portion of the Request URI To header. The following examples are of valid SIP Domains:

5000@newport.com - If the PBX is configured to use a domain.

4000@SIPRouterAddress - If the PBX changes the original Request-URI and inserts the destination address.

2500@PBXAddress - If the PBX uses its IP Address as a SIP Domain

The SIP Router must be configured to handle SIP messages for a particular SIP Domain. If the SIP Router receives a request for a Domain for which no configuration exists, it responds with a "404 Not Found" response. If the SIP Domain is configured, message processing continues. You can configure multiple SIP Domains (integrations) on the SIP Router. Each SIP Domain (integration) must have a unique domain name.

Local Address – Select the local IP Address the SIP Routing Manager from the drop-down list to use for this SIP Domain. This address is the IP Address of the network interface card that communicates with the particular integration or SIP Domain.

Description – Enter description for this SIP Domain that describes the domain's purpose. For example, a description for an Avaya/Nortel integration may be, Newport CS1000. The description displays in the left-hand tree-view list pane of the SIP Routing Manager window under the SIP domain.

Numbering Plans

Numbering Plan – The Numbering Plan column displays the configured numbering plans for this SIP domain.

Priority – The priority column of the **Numbering Plans** area displays the priorities of the configured numbering plans. To change the priority of a numbering plan, click the up or down buttons until the numbering plans are ordered in the correct priority.

Priorities are assigned to each numbering plan. The lower the priority number, the higher the priority. For example, one is the highest priority. For routing algorithms such as Overflow, messages are redirected to a route that is active and has the highest priority.

Add button – Click Add to add a new numbering plan to the SIP domain. The **Numbering Plan** dialog box appears.

Edit button – To edit a numbering plan, highlight the numbering plan you want to edit, and then click Edit.

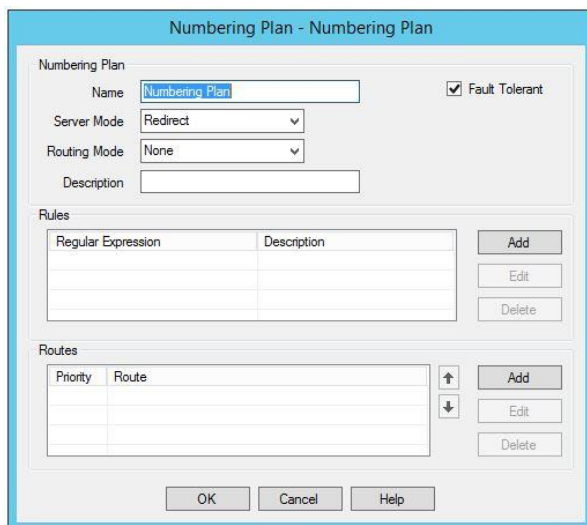
Delete button - To delete a numbering plan, highlight the numbering plan you want to delete, and then click **Delete**.

Numbering Plan dialog box

The **Numbering Plan** dialog box allows you to configure one or more numbering plans for each SIP domain. For a domain to be valid, it must contain at least one numbering plan. A numbering plan consists of one or more routes that can handle calls for a particular destination.

The intended destination user (typically the User portion of the Request-URI) is matched against the numbering plans. Each numbering plan is configured with a matching rule and the first numbering plan that matches the destination successfully is selected to further process the request. This matching process makes the order (priority) in which the numbering plans are configured very important.

For example, consider two numbering plans within a domain. One handles all calls destined for extension 5000. This numbering plan routes calls to a MiCollab AM Call Server cluster and handles all Request messages. In addition, this Numbering Plan routes calls to the PBX and is accessed whenever a Call Server makes an outgoing call. The second numbering plan can also handle calls to 5000. It may be configured as an Overflow route to another Call Server. In order for the SIP Routing Manager to route calls properly, the numbering plan for 5000 must have a higher priority over the other. If the Priority of the numbering plans is incorrect, calls do not route to the designated Call Server.



The dialog box is titled "Numbering Plan - Numbering Plan". It contains the following fields and sections:

- Numbering Plan**
 - Name:** A text field containing "Numbering Plan".
 - Server Mode:** A dropdown menu set to "Redirect".
 - Routing Mode:** A dropdown menu set to "None".
 - Description:** An empty text field.
 - Fault Tolerant:** A checked checkbox.
- Rules**
 - A table with two columns: "Regular Expression" and "Description".
 - Buttons: "Add", "Edit", and "Delete".
- Routes**
 - A table with two columns: "Priority" and "Route".
 - Buttons: "Add", "Edit", and "Delete".
- Buttons:** "OK", "Cancel", and "Help" at the bottom.

Numbering Plan

Name – Enter a name for the numbering plan. This name displays in the left-hand list pane of the SIP Routing Manager window under Numbering Plan.

Server Mode – The Redirect mode is the default and the only available mode. The redirect server processes incoming SIP requests by generating a redirect response back to the originator to provide information about the destination, and then removes itself from the signaling path established between the originator and the destination.

Routing Mode - Click the drop-down box to select the routing mode for this numbering plan. The default mode is Round Robin.

The SIP Routing Manager routes calls to MiCollab AM by utilizing various algorithms. Routes grouped within a numbering plan can be configured with a unique routing mode. A fault tolerant route is selected only if it has responded to the keep-alive requests between the PBX and Call Server. This ensures that calls

are not sent to an inactive endpoint. The SIP Routing Manager uses the following algorithm modes to route calls to MiCollab AM.

- **Round Robin** - Routes are selected from the routing list in cyclic order. For example, if a numbering plan contains three routes, an incoming call is routed to the first route, the second incoming call to the second route, and the third call is routed to the third route.
- **Overflow** - Routes are selected from the routing list based on their priority. Highest priority routes receive calls first. Calls are routed to another active route only after all higher priority routes are full.
- **Least loaded** - The least loaded route is selected from the routing list of active routes. The SIP Routing Manager collects capacity information using the SIP OPTIONS message and the keep-alive mechanism. The following rules apply when selecting a route from the list of active routes in the least loaded mode:
 - A route with the highest available capacity is selected.
 - If multiple routes have the same available capacity, a route with the most ports under the Hunt is selected.
 - If multiple routes have the same available capacity and their hunt size is the same, the route that has the most ports is selected.
 - If the properties of multiple routes are the same, the route with a higher priority is selected.

NOTE No two routes within an Endpoint can have the same priority.

Description – Enter a description for the numbering plan. The description for the numbering plan displays in the right-hand list pane of the SIP Routing Manager.

Fault Tolerant – Select fault tolerant if you want to make the numbering plan fault tolerant. When enabled, all routes in the numbering plan receive periodic keep-alive messages to determine if they are up and functioning. The default is enabled.

Rules

Regular Expression – The Regular Expression column displays the list of regular expressions configured in the **Regular Expression** dialog box.

Description – The Description column displays the description of each regular expression configured in the **Regular Expression** dialog box. The rules description displays in the right-hand detail pane of the **SIP Routing Manager** window.

Add button–In the **Rules** area, click **Add** to add a regular expression for the numbering plan. The **Regular Expression** dialog box appears.

Edit button– To edit a regular expression, highlight the regular expression you want to edit, and then click **Edit**.

Delete button - To delete a regular expression, highlight the regular expression you want to delete, and then click **Delete**.

Routes

Priority – The **Priority** column displays the priority of each route. Click the up and down buttons to set the correct priority of each route.

Route – The **Route** column displays the list of routes. Routes are configured in the **Route** dialog box.

Add button–In the **Routes** area, click **Add** to add a route for the numbering plan. The **Route** dialog box appears.

Edit button– To edit a route, highlight the route you want to edit, and then click **Edit**.

Delete button - To delete a route, highlight the route you want to delete, and then click **Delete**.

Regular Expression dialog box

The **Regular Expression** dialog box allows you to configure regular expressions for the numbering plan.

The image shows a dialog box titled "Regular Expression". It has a light blue border. Inside, there are two text input fields. The first field is labeled "Regular Expression Rule" and contains a single vertical bar "|". The second field is labeled "Description (Optional)" and is empty. At the bottom of the dialog box, there are three buttons: "OK", "Cancel", and "Help".

Regular Expression – A route contains a matching rule that is similar to the numbering plan rule. This rule may or not be the destination number; it is the number the server dials to reach the destination. In some configurations, route selections are made based on whether the destination user matches the rule. For example, for SIP station integrations, it is possible that the stations span multiple MiCollab AM Call Servers. When a call is destined for a particular station, it is redirected to a route whose rule matches the destination.

Enter a valid regular expression for this numbering plan in this field. For example, the following regular expressions are valid rules for the numbering plan, 5000.

5000

5(0{3})

To match 5XXX:

5[0–9][0–9][0–9]

5[0–9]{3}

^5\d{3}\$

To specify a range:

5[0]{2}[0–8] // 5001–5008

[5]{1}[0]{1}(((0–1){1})[0–9]{1})((2){1})[0–4]{1})) // 5001–5024

[5]{1}[0]{1}(((0-8){1}[0-9]{1})|((9){1}[0-6]{1})) // 5001-5096

[5]{1}[0]{1}(((0-4){1}[9]{1})|((5-8){1}[0-9]{1})|((9){1}[0-6]{1})) // 5049-5096

To specify a range using wildcard characters:

. * // Match any number of digits/characters

{4} // Match 4 digits/ characters only

NOTE The dialog box accepts white spaces for readability purposes. In addition, the dialog box validates the syntax of the expression before accepting the expression. It cannot, however, prevent syntactically valid but incorrect expressions.

Description – The **Description** field displays the description for the regular expression. The description for the rule displays in the right-hand details pane of the SIP Routing Manager window.

Route dialog box

The **Route** dialog box allows you to configure a route for a numbering plan. Each numbering plan can have one or more routes. Each route points to a destination to which an incoming request is redirected. Routes are selected based on their configured routing mode and whether they are active.

The screenshot shows the 'Route' dialog box with the following sections and controls:

- Route**: Destination Address (text field), Port (text field), Description (text field), Preferred Transport (radio buttons for UDP and TCP).
- Rules**: A table with columns 'Regular Expression' and 'Description'. Buttons: Add, Edit, Delete.
- SIP Methods**: A table with columns 'Method' and 'Event Package'. A dropdown menu with 'INVITE' selected. Buttons: Remove, Insert.
- Fault Tolerant**: SIP Account Name (text field), Keep Alive Interval (text field with '5' and a 'sec' dropdown).
- Destination Capacity**: Maximum Capacity (text field with '48'), Maximum Route Capacity (text field with '48').
- Other (Optional)**: Advanced (button).
- Buttons**: OK, Cancel, Help.

Route

Destination Address – The Destination Address is the IP Address of the destination to which the SIP message is redirected. Typically, the IP Address of the Call Server to which calls are directed.

Preferred Transport -

Port - The port is the local listening port for this destination IP Address. The default port number is 5060.

Description – Enter a description for the route. The route description displays in the right-hand detail pane of the SIP Routing Manager window.

Preferred Transport – Select the UDP or TCP protocol as the preferred transport for this route. Each route may use its own transport preference. UDP is the default transport.

Rules

The **Rules** area displays the configured rules and descriptions as described in the **Numbering Plan** dialog box. You can add, edit, or remove a rule from the **Route** dialog box.

Regular Expression – The **Regular Expression** column displays the list of regular expressions configured in the **Regular Expression** dialog box.

Description – The **Description** column displays the description of each regular expression configured in the **Regular Expression** dialog box. The rules description displays in the right-hand detail pane of the **SIP Routing Manager** window.

Add button–In the **Rules** area, click **Add** to add a regular expression for the numbering plan. The **Regular Expression** dialog box appears.

Edit button– To edit a regular expression, highlight the regular expression you want to edit, and then click **Edit**.

Delete button - To delete a regular expression, highlight the regular expression you want to delete, and then click **Delete**.

SIP Methods

A route is configured with a list of SIP Methods that it can support; it may use multiple SIP message types. For example, consider a set of SIP methods for a SIP integration that uses aggregate subscriptions for MWI. In this configuration, SUBSCRIBE requests intended for recipients are directed towards a Call Server whereas NOTIFY requests for that user must be directed to the PBX. In this example, the configuration contains two routes under the same numbering plan, one that applies to SUBSCRIBE requests, and other that applies to NOTIFY requests.

As another example, assume the SIP Routing Manager is configured to work with a Registrar. The route that points to the registrar should only have REGISTER in the supported methods list. Similarly, when configuring the SIP Routing Manager to work with a PBX that supports unsolicited NOTIFY request for message waiting, a NOTIFY method must be added to the list.

Different SIP methods implement different features. For example, the SUBSCRIBE and NOTIFY methods not only be used to implement message waiting but can also be used to convey presence related information. Another SIP header in the request makes the distinction. This header is called an "Event-Package." "Message-summary" and "refer" are two examples of an event package. Therefore, a SIP Method may be associated with an event package. Not all SIP methods require an event package. When selecting a SIP Method, the event package drop-down list is enabled if the selected SIP Method may require an event package to along with it.

To insert a SIP Method and associated event-package, select the SIP Method you want to use from the drop-down list. If the SIP Method requires an event-package as well, select the event-package from the list. Click **Insert**, when you have selected the appropriate SIP Method and Event-Package from the list. The SIP method and event package display in the SIP Method table.

To remove a SIP Method, click **Remove**.

Method – The Method column lists the SIP method the route uses to redirect messages.

Event-Package – The Event-Package column lists the event-package associated with the SIP Method. Not all SIP methods require an event-package.

Fault Tolerant

If the numbering plan is configured as Fault Tolerant, you must configure the route with a SIP Account name.

SIP Account Name – Enter a name in the **SIP Account Name** field for the keep-alive SIP messages. The SIP Account Name is sent in the message header of the SIP OPTIONS keep-alive messages generated for the route.

Keep-alive interval – You can configure each route with a different keep-alive interval. This parameter overrides the global keep-alive interval field value of the **Settings** dialog box.

The keep-alive mechanism is generated by a SIP OPTIONS request that is sent to all of the configured routes. If the destination does not respond to the request, the SIP Routing Manager assumes the route is disabled and the route is removed from the list of active routes. For example, if a Call Server is shut down for maintenance, the network connection failed, or the Call Server is down for unexpected reasons the route is disabled. It is possible that the route is mis-configured and the destination cannot respond successfully. In either event, the SIP Routing Manager continues to send an OPTIONS request periodically to the destination so that it can be put back on the active route list once it starts responding successfully again.

Destination Capacity

Destination capacity information helps the SIP Routing Manager implement load-balancing algorithms. These parameters allow you to enter default capacity information for the destination. The SIP Routing Manager uses the information when routing calls that arrive before the response to the first keep-alive request is received.

The response to the keep-alive requests contains the most current capacity information for the route. This information is used to route subsequent calls. For routes that cannot provide capacity information, the default values are always used.

NOTE The SIP OPTIONS message provides a way of collecting the server capacity information that is used to make an informed route selection. This message is not a standard SIP message; it is an Mitel proprietary message. The SIP Routing Manager sends the message to query the Call Servers for trunk capacity information. The SIP Routing Manager populates the outgoing SIP OPTIONS request with a "User Agent" and "Trunk-Capacity" header whose value is set to identify the destination. The presence of these two headers determines how the request is processed on the Call Servers.

For example, The SIP Routing Manager populates an outgoing SIP OPTION request with a "trunk capacity" header, the value of which is the SIP URI (Uniform Resource Identifier) of a Call Server. If the Call Server determines the message originator is an Mitel endpoint, the Call Server responds with the appropriate trunk capacity information

Maximum Capacity – Enter the maximum capacity (number of lines) of the destination server in the **Maximum Capacity** field. For MiCollab AM, this is the total number of SIP lines on the Call Servers.

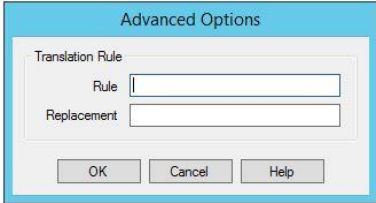
Maximum Route Capacity – Enter the maximum capacity (number of lines) configured to handle calls for this route. For MiCollab AM, this is the total number of SIP lines configured to participate in the integration. If a single SIP integration is configured on the Call Server only, this value is equal to the **Maximum Capacity** parameter.

Advanced Options

Click **Advanced** to display the **Advanced Options** dialog box.

Advanced Options dialog box

The **Advanced Options** dialog box allows you to create a translation rule for the numbering plan. A translation rule translates an input string to an output string and allows you to perform a name/number translation. When the SIP Routing Manager selects a particular route for a redirected message, it can perform an additional name/number translation. For example, assume an incoming message destined to 895000. The message is redirected to a route but the destination number is translated or changed to 5000, 4259455000, or +14259455000. The translation allows you to strip unwanted digits, or add necessary steering digits for the destination.



Translation Rule

Rule – In the **Rule** field, enter a rule for the replacement value. For example, assume a numbering plan rule of 895000. The desired translated number is 5000, so the rule becomes 89 and the desired replacement value is 5000.

Replacement – In the **Replacement** field, enter the replacement value for the translation.

For example,

Rule: "89"

Replacement: "5000"

Input: 895000

Output: 5000

Rule: ^7[0-9]{3}\$ or ^7(\d{3})\$

Replacement: +14259457\$1

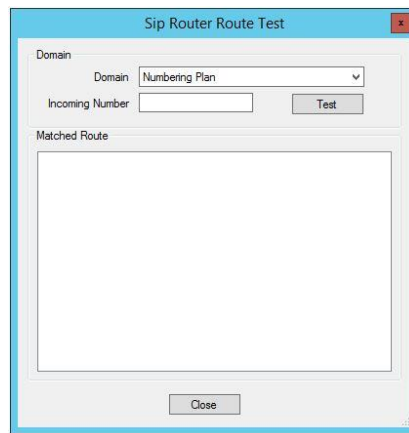
Input: 7234

Output: +14259457243

Rule {4}
Replacement: 4800
Input: 4000
Output: 4800

SIP Router Route Test Dialog Box

The **SIP Router Route Test** dialog box allows you to test each Numbering Plan and Route you configured for the SIP Domain.



Domain

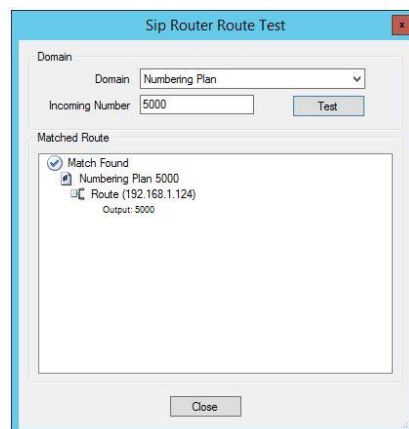
Domain – Select the SIP domain in which you want to perform the test from the drop-down list.

Incoming Number – Enter the Regular Expression of the Numbering Plan you want to test.

Test button – Click Test to test the incoming number for the domain.

Matched Route

The **Matched Route** area displays the results of the test action.



Logging

The SIP Routing Manager can log SIP messages exchanged with remote endpoints. In addition, logging is available that includes routing decisions. These log types are useful for troubleshooting purposes. Use the Diagnostics utility to enable\disable logging. Log files are stored in the \CX\Log directory.

Appendix A – Integration Specific Notes

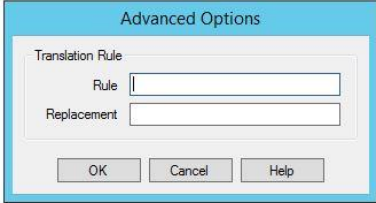
Follow the instructions in this section when you are configuring the SIP Routing Manager to work with specific integrations.

Avaya/Nortel CS1000

The Avaya/Nortel NRS does not allow two IP Addresses to be associated with the same pilot number. If you are using multiple Call Servers, each Call Server must have a unique pilot number assigned to an endpoint on the SIP Routing Manager must be unique. When adding a route, you must add a Translation Rule for each Call Server in the integration to match the corresponding endpoint.

To add a Translation Rule:

- 1 On the **Routes** dialog box, click the **Advanced** button to open the **Advanced Options** dialog box. The **Advanced Options** dialog box appears.



- 2 In the **Advanced Options** dialog box, enter the incoming digits that represent the pilot number of the Call Server to which this route is directed.
- 3 Create a translation rule for each route and Call Server in the integration.