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Integration Reference Manual

SIP Server 8.1.0

1/15/2022

Table of Contents

SIP Server 8.1 Integration Reference Manual

This document introduces you to the concepts, terminology, and procedures related to integrating SIP Server with SIP softswitches and gateways. The reference information includes, but is not limited to, configuration options, limitations, and switch-specific functionality. This document is designed to be used along with the Framework 8.1 SIP Server Deployment Guide.

Switch Integrations

Find information for integrating SIP Server with the following switches.

Siemens OpenScape Voice

Asterisk

Media Gateway Integrations

Find information for integrating SIP Server with the following media gateways.

Cisco Media Gateway

AudioCodes Gateway

Network Load Balancer Integrations

Find information for integrating SIP Server with the following network load balancers.

F5 Networks BIG-IP LTM

About SIP Server

SIP Server is the Genesys software component that provides an interface between your telephony hardware and the rest of the Genesys software components in your enterprise. It translates and keeps track of events and requests that come from, and are sent to the telephony device. SIP Server is a TCP/IP-based server that can also act as a messaging interface between SIP Server clients. It is the critical point in allowing your Genesys solution to facilitate and track the contacts that flow through your enterprise.

Intended Audience

This guide is intended primarily for system administrators, certified technicians, those who are new to SIP Server and those who are familiar with it. Based on your specific contact center environment and your responsibilities in it, you may need to be familiar with a much wider range of issues as you deploy SIP Server.

In general, this document assumes that you have a basic understanding of, and familiarity with:

- Computer-telephony integration (CTI) concepts, processes, terminology, and applications.
- Network design and operation.
- Your own network configurations.
- Your telephony hardware and software.
- Genesys Framework architecture and functions.
- Configuration Manager interface and object management operations.

In particular, this document assumes that you are trained and certified on the products this guide is written for. For more information, see product-specific documentation.

The SIP Server integration solutions described in this document are not the only methods that will work; rather, they are the ones that have been tested and approved by Genesys, and that are supported by Genesys Customer Support.

Reading Prerequisites

You must read the *Framework 8.1 SIP Server Deployment Guide* before using this manual.

Siemens OpenScape Voice

This topic describes how to integrate SIP Server with the Siemens OpenScape Voice. It contains the following sections:

- Overview
- Configuring OpenScape Voice
- Configuring DN Objects
- Support for First-Party Call-Control Operations
- Support for Split-Node Deployments

Note: The instructions in this topic assume that OpenScape Voice is fully functional and is routing calls before Genesys products are installed. They also assume that SIP Server has already been configured to function properly in stand-alone mode, and that configuration between SIP Server and Universal Routing Server (URS) has already been completed.

Overview

The SIP Server and OpenScape Voice integration solution that is described in this topic is not the only method that will work. Although there are other methods, this is the only one that has been tested and approved by Genesys, and that is supported by Genesys Customer Support. This topic contains best-practice guidelines that have been determined by both Genesys and Siemens Engineering departments. Deviating from the solution that is described in this topic can have unexpected consequences.

Although this topic provides steps to log in to OpenScape Voice, login credentials are site-specific and should be different for each installation, due to the nature of the equipment.

Note: The OpenScape Voice screen captures in this topic were taken from the HiPath Assistant 3.0R0.0.0 Build 860. Depending on your onsite version, the onscreen output might differ.

Assumptions

The integration solution described in this topic makes the following assumptions about the desired call flow:

- Agent endpoints (SIP Phones) register directly with OpenScape Voice. Genesys SIP Server does not signal these endpoints directly; instead, it always goes through OpenScape Voice.
- A single instance of SIP Server is configured behind OpenScape Voice.
- If it is used for treatments, music on hold, MCU (Multipoint Conference Unit) recording, and supervisor functionality, Stream Manager is signaled only by SIP Server. No direct SIP signaling occurs between OpenScape Voice and Stream Manager. For information about configuring SIP Server to use Stream Manager, see the Framework 8.1 SIP Server Deployment Guide.

In the event that these assumptions are not valid for the required deployment, you can still configure SIP Server for integration with OpenScape Voice; however, you might have to modify the configuration that is described in this topic.

To configure multiple instances of SIP Server to work with OpenScape Voice, create a unique Numbering Plan for each SIP Server and each group of agents that is associated with it and related switch entities, as described in the table: Task Flow—Configuring OpenScape Voice. For example, to configure two SIP Servers, create two unique SIP Server Numbering Plans, two Agent Numbering Plans, and all related switch entities as required for each Numbering Plan.

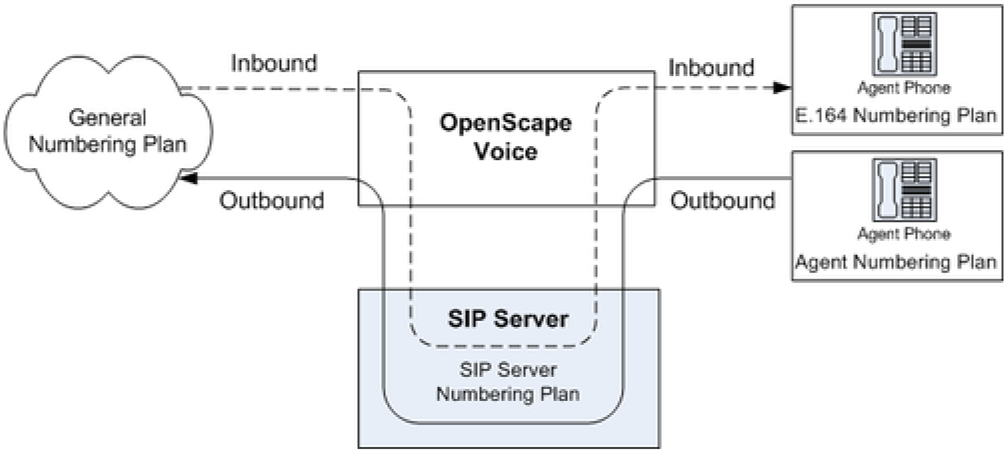
For GVP integration with SIP Server, the configuration must be performed on the SIP Server side, not on the OpenScape Voice side.

Endpoint Support

When Genesys SIP Server is integrated with Siemens OpenScape Voice, the endpoints register directly to the Siemens switch. Genesys validates the integration using a representative selection of endpoints recommended by Siemens. However, this selection is not an exhaustive list of endpoints, and Genesys defers the official endpoint support statement to Siemens. Also note that the Click-to-Answer feature requires the referenced Patchset on OpenScape Voice and a device that supports it.

Deployment Architecture

A successful implementation requires that Genesys SIP Server be in the communications path for every call in the contact center"both internal and external (see the following figure). This can be done efficiently and effectively by using multiple Numbering Plans. Note, however, that gateways should not be put into the Global Numbering Plan. Doing so can cause complications by routing gateway calls directly to the agents, bypassing SIP Server.



SIP Server - OpenScape Voice Deployment Architecture

In the General Numbering Plan (the Numbering Plan that contains the gateways), the contact center is given a range of numbers for agents (assuming that the agents have direct lines) and Routing Points. Those numbers route directly to SIP Server, which then routes the calls accordingly.

SIP Server must have its own Numbering Plan, because it will make calls on behalf of the agents. These calls are sent to the E.164 Numbering Plan (to reach internal phones) or, if necessary, to available gateways.

The Agent Numbering Plan is simple; all calls go to SIP Server. The configuration of SIP Server Numbering Plan will determine how the calls should be routed.

Accessing Configuration Tools

HiPath Assistant

The HiPath Assistant is a thin, Web-based application that runs within a browser to provide a common user experience. It is primarily intended for use as a Service Management Center that provides administrators of communications networks with provisioning information and control over their subscribers' voice services. Its purpose is to provide enterprises with a cost-effective, IP-based system that works seamlessly with OpenScape Voice.

For enterprises with more than 5,000 lines, the HiPath Assistant can be installed on an external server as a stand-alone (off-board) installation, separated from the OpenScape Voice switch.

To access the HiPath Assistant, enter the following URL in your browser:

```
https://<IP Address>
```

Command-Line Interface

OpenScape Voice also has an SSL (Secure Sockets Layer) command-line interface that you can access. SSL is the same as Telnet, except that it is encrypted to provide more security. There are many SSL client applications available on the Web for free, in addition to commercial applications. A common application for SSL is PuTTY. You can download PuTTY from the following web page:

http://www.chiark.greenend.org.uk/~sgtatham/putty/download.html.

After you have your SSL application, configure it to connect to the management IP address of OpenScape Voice.

Integration Task Summary

To integrate SIP Server with OpenScape Voice, complete the following procedures:

1. Configure OpenScape Voice.
2. Configure DN objects in the Configuration Layer.

Configuring OpenScape Voice

This page provides an overview of the main steps that are required to configure OpenScape Voice. Complete all steps in the order in which they are listed.

Configuring OpenScape Voice

1. Check that OpenScape Voice is working.

Check Minimum Functionality in OpenScape Voice

The procedures in this topic assume that OpenScape Voice is functional and routing calls appropriately. There should already be at least one Numbering Plan that has gateways and nonagent subscribers in it. For more information, see Siemens OpenScape Voice-specific documentation.

2. Configure the Numbering Plans.

Configuring Numbering Plans

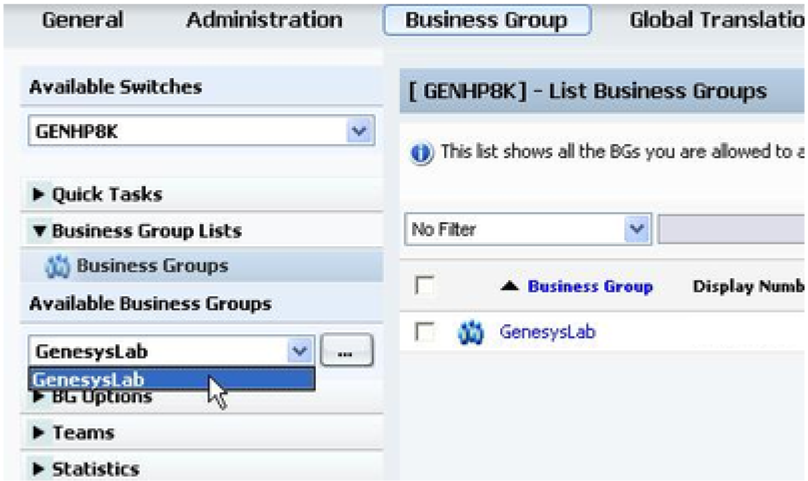
The instructions in this topic assume that OpenScape Voice is functional and routing calls appropriately. There should already be at least one Numbering Plan with configured gateways and nonagent subscribers.

Purpose

To create the Numbering Plans that will contain the Agents and SIP Server.

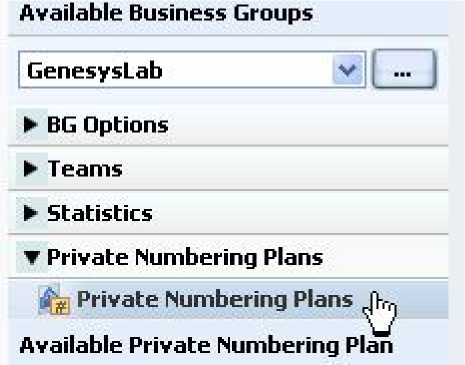
Start

1. Log in to the HiPath Assistant, and navigate to the Business Group of the contact center that you want to configure--for example, `GenesysLab` (see the following figure).



Selecting the Business Group

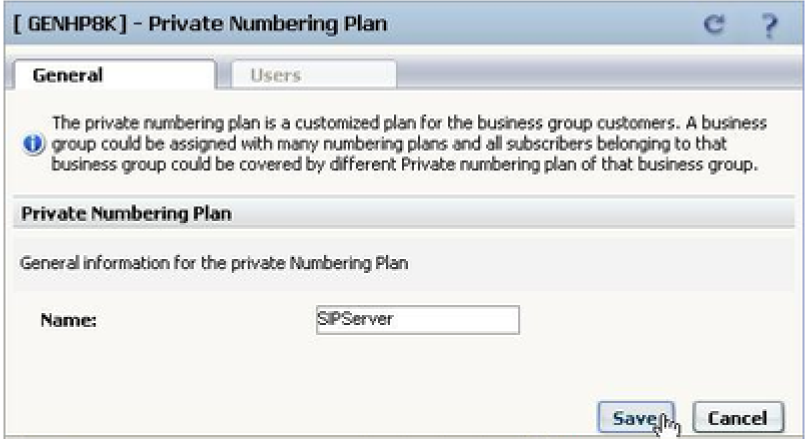
2. Click `Private Numbering Plans` (see the following figure).



Selecting Private Numbering Plans

3. In the `Private Numbering Plans` dialog box, click Add.

4. Add two new Private Numbering Plans: one for your agents and one for SIP Server itself--for example, `Agents` and `SIPServer`, respectively (see the following figure)



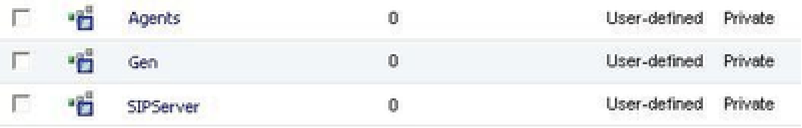
The private numbering plan is a customized plan for the business group customers. A business group could be assigned with many numbering plans and all subscribers belonging to that business group could be covered by different Private numbering plan of that business group.

General information for the private Numbering Plan

Name: SIPServer

Creating Private Numbering Plans

When you are finished, the dialog box shown in the following figure appears.



□	Agents	0	User-defined	Private
□	Gen	0	User-defined	Private
□	SIPServer	0	User-defined	Private

Private Numbering Plans

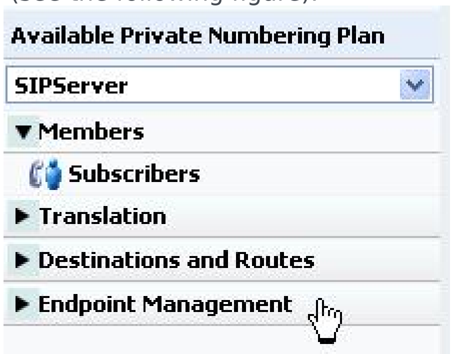
End

3. Configure the Endpoint Profile.

Configuring a SIP Server Endpoint Profile

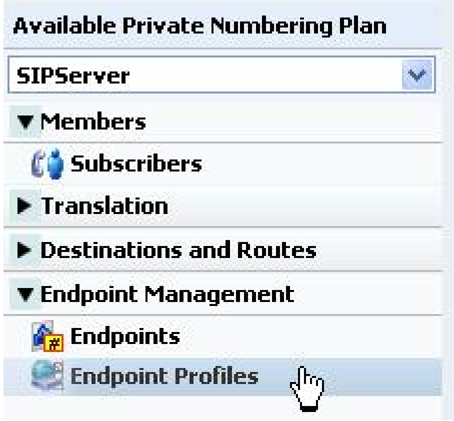
Start

1. Click `Private Numbering Plan`, and then click the SIP Server Numbering Plan—for example, `SIPServer` (see the following figure).



Selecting the Numbering Plan

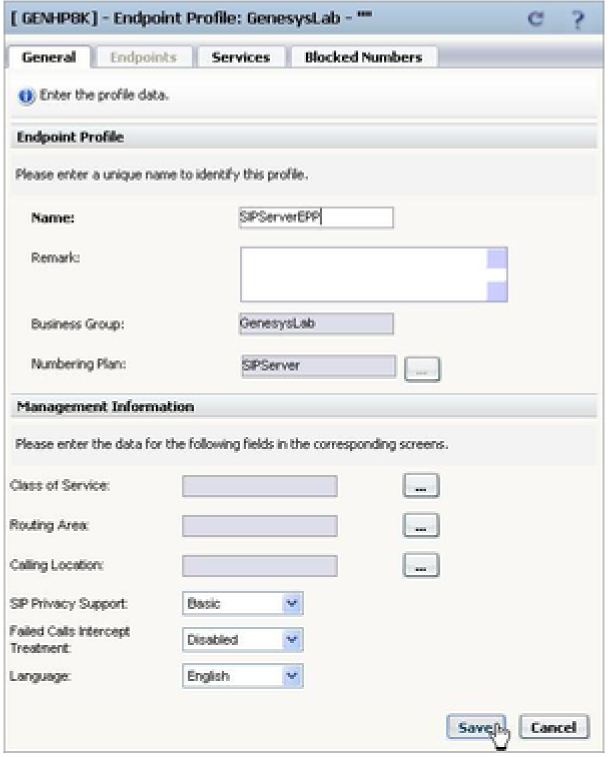
2. Click `Endpoint Management`, and then click `Endpoint Profiles` (see the following figure).



Selecting Endpoint Profiles

3. In the `Endpoint Profile`: <`Business Group`> dialog box on the `General` tab, enter a name for this

configured Endpoint Profile in the Name text box. This will associate the endpoint that uses it with the Numbering Plan in which the Endpoint Profile was created (see the following figure).



[GENHP8K] - Endpoint Profile: GenesysLab - ""

Enter the profile data.

Endpoint Profile

Please enter a unique name to identify this profile.

Name: SIPServerEPP

Remark:

Business Group: GenesysLab

Numbering Plan: SIPServer

Management Information

Please enter the data for the following fields in the corresponding screens.

Class of Service:

Routing Area:

Calling Location:

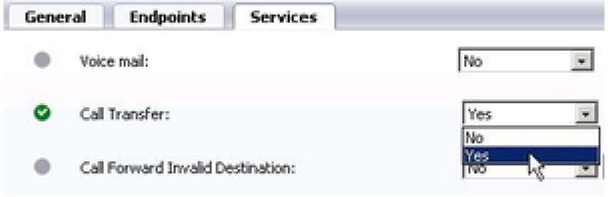
SIP Privacy Support: Basic

Failed Calls Intercept Treatment: Disabled

Language: English

Configuring an Endpoint Profile

4. (Optional) If there are existing dialing rules and conventions that require the use of Class of Service and Routing Areas, enter that information. As a general rule, give this Endpoint Profile the same calling access as you would give to your agents
5. When you are finished, click Save.
6. In the `Endpoint Profile`: <`Business Group`> dialog box on the `Services` tab, enable the `Call Transfer` service, by selecting Yes from the drop-down menu (see the following figure).



Voice mail: No

Call Transfer: Yes

Call Forward Invalid Destination: No

Enabling the Call Transfer Service

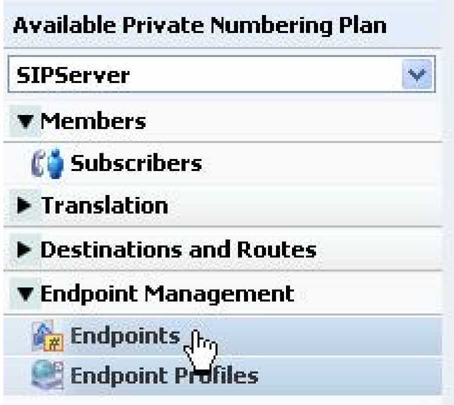
End

4. Configure the Endpoint.

Configuring a SIP Server Endpoint

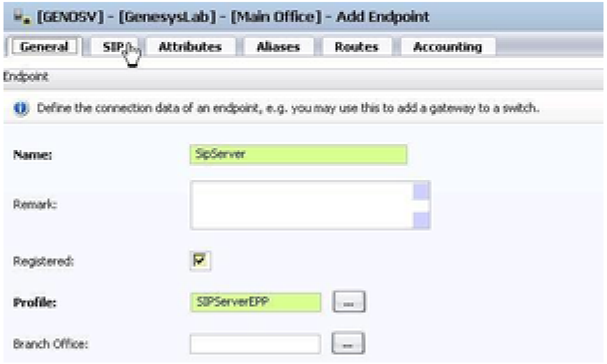
Start

1. Click Private Numbering Plan, and then click the SIP Server Numbering Plan—for example, SIPServer.
2. Click Endpoints, and then click Add (see the following figure).



Selecting Endpoints

3. In the Endpoint: <Business Group> dialog box, click the General tab, and do the following:
   a. In the Name text box, enter a unique name for this configured Endpoint.
   b. Select the Registered check box.
   c. Set the Profile text box to the Endpoint Profile that you created for SIP Server, by clicking the browse (...) button.



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

Name: SipServer

Remark:

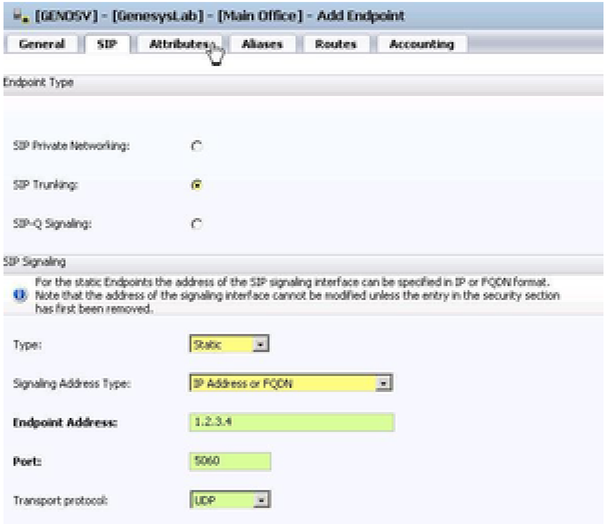
Registered: ☑

Profile: SIPServerEPP ...

Branch Office: ...

Configuring Endpoints: General Tab

4. In the Endpoint: <Business Group> dialog box, click the SIP tab, and do the following:
   a. Make sure that the Type text box is set to Static.
   b. In the Endpoint Address text box, enter the IP address of SIP Server.
   c. From the Transport protocol drop-down box, select UDP or TCP, depending on SIP Server.



[GENDSV] - [GenesysLab] - [Main Office] - Add Endpoint

Endpoint Type

SIP Private Networking: ○

SIP Trunking: ●

SIP-Q Signaling: ○

SIP Signaling

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

Type: Static

Signaling Address Type: IP Address or FQDN

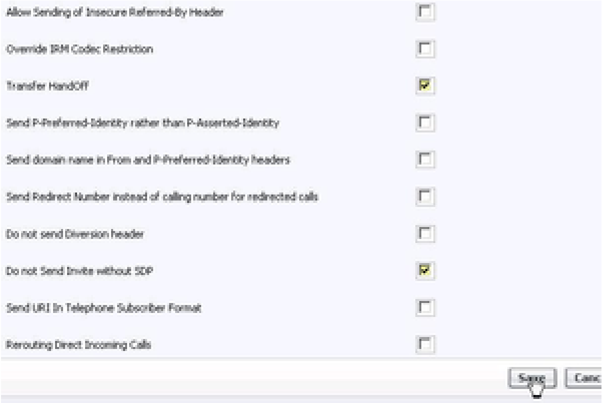
Endpoint Address: 1.2.3.4

Port: 5060

Transport protocol: UDP

Configuring Endpoints: SIP Tab

5. Click the `Attributes` tab, and do the following:
    a. Select the `Transfer HandOff` check box.
       There is a known limitation of the Transfer HandOff feature. The full number must be used to transfer a call when this feature is activated.
    b. Select the `Do not Send Invite without SDP` check box.
    c. When you are done, click Save.



Allow Sending of Insecure Referred-By Header □

Override IRM Codec Restriction □

Transfer HandOff ☑

Send P-Preferred-Identity rather than P-Asserted-Identity □

Send domain name in From and P-Preferred-Identity headers □

Send Redirect Number instead of calling number for redirected calls □

Do not send Diversion header □

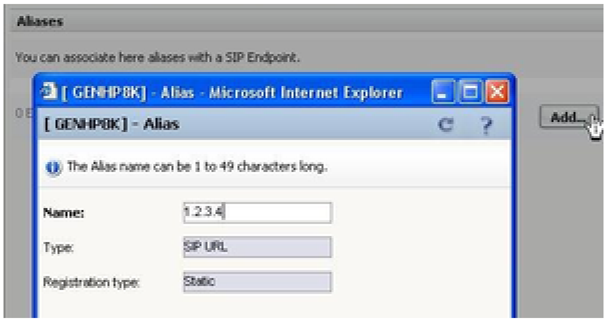
Do not Send Invite without SDP ☑

Send URI In Telephone Subscriber Format □

Rerouting Direct Incoming Calls □

Configuring Endpoints: Attributes Tab

6. Click the `Aliases` tab, and then click Add.
7. In the `Alias` dialog box, do the following:
    a. In the Name text box, enter the IP address that you entered in the `Endpoint Address` text box in Step 4.
    b. Unless you have OpenScape Voice version 5 and later, set the Type text box to `SIP URL`. (This is done automatically in version 5.)
    c. Click OK.



You can associate here aliases with a SIP Endpoint.

The Alias name can be 1 to 49 characters long.

Name: 1.2.3.4

Type: SIP URL

Registration type: Static

Configuring Endpoints: Aliases Tab

8. In the Endpoint dialog box, click Save.
9. When the confirmation message box appears, informing you that the Endpoint was created successfully, click `Close`.

End

5. Configure Gateway Destinations.

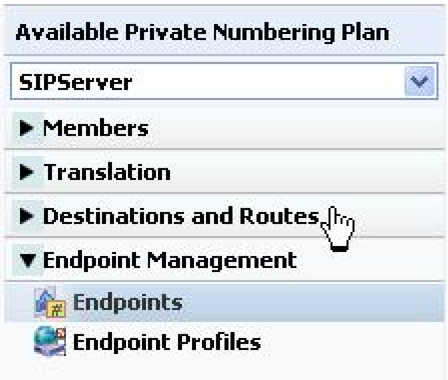
Configuring SIP Server Destinations for Gateways

Purpose

To create Gateway Destinations for SIP Server to route calls. The Endpoints of such Gateway Destinations must already be configured in OpenScape Voice. SIP Server routes calls to Gateways and to phones. Because calls to the phones are routed via the E.164 Numbering Plan, no Destinations have to be configured for them.

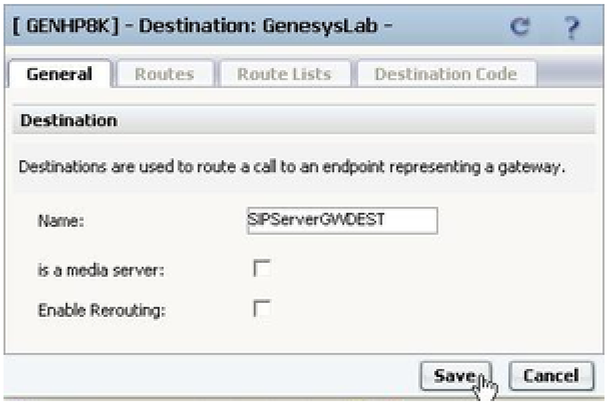
Start

1. Click `Private Numbering Plan`, and then click the SIP Server Numbering Plan—for example, `SIPServer`.
2. Click `Destinations and Routes`, then `Destinations`, and then click Add (see the following figure).



Selecting Destinations

3. In the `Destination` dialog box, on the `General` tab, do the following:
    a. In the `Name` text box, enter a unique name for the Destination—for example, `SIPServerGWDEST`. The name must be unique within the switch configuration database.
    b. Make sure that all check boxes are cleared.
    c. When you are finished, click Save.



Destinations are used to route a call to an endpoint representing a gateway.

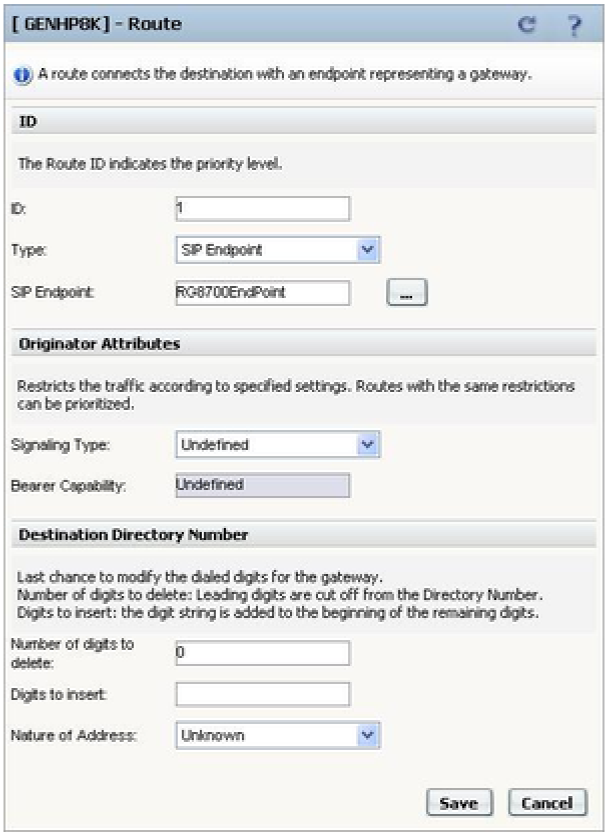
Name: SIPServerGWDEST

is a media server: □

Enable Rerouting: □

Configuring a Gateway Destination

4. In the `Destination - <Business Group>` dialog box, click the Destination that you just created.
5. Click the `Routes` tab, and then click Add.
6. In the `Route` dialog box, do the following:
    a. In the `ID` text box, enter 1 for this particular route.
    b. Set the `Type` text box to `SIP Endpoint`.
    c. Set the `SIP Endpoint` text box to the Endpoint that you created in Configuring a SIP Server Endpoint by clicking the browse (...) button, selecting the Numbering Plan that contains the Endpoint for the gateway to which you will be routing (for example, the general Numbering Plan), and then selecting the Endpoint.
    d. Do not modify the digit string for calls that are being routed from SIP Server. All modifications to the digit string should be completed before the calls arrive to SIP Server.



[GENHP8K] - Route

A route connects the destination with an endpoint representing a gateway.

ID

The Route ID indicates the priority level.

ID: 1

Type: SIP Endpoint

SIP Endpoint: RG8700EndPoint

Originator Attributes

Restricts the traffic according to specified settings. Routes with the same restrictions can be prioritized.

Signaling Type: Undefined

Bearer Capability: Undefined

Destination Directory Number

Last chance to modify the dialed digits for the gateway.
Number of digits to delete: Leading digits are cut off from the Directory Number.
Digits to insert: the digit string is added to the beginning of the remaining digits.

Number of digits to delete: 0

Digits to insert:

Nature of Address: Unknown

Configuring a Route for a Gateway Destination

5. When you are finished, click Save.
6. When the confirmation message box appears, informing you that the Route was added successfully, click `Close`.
7. In the `Destination` dialog box, click `OK`. You will now be able to view the Route that you just created in the `Routes` dialog box.
8. Repeat Steps 2-9 to create other gateway Destinations for SIP Server, as necessary.

End

6. Configure Prefix Access Codes.

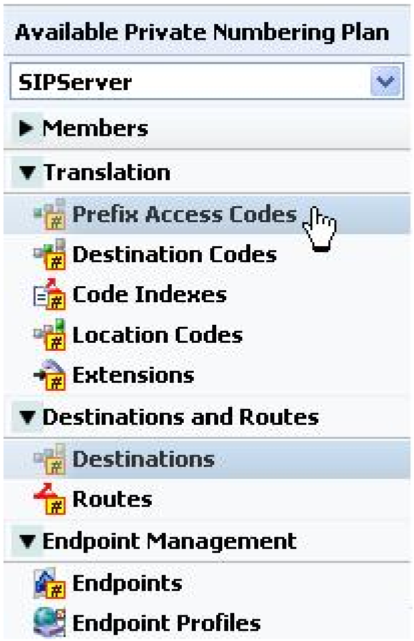
Configuring SIP Server Prefix Access Codes

Purpose

To configure Prefix Access Codes that SIP Server will dial to reach Subscribers and Gateways.

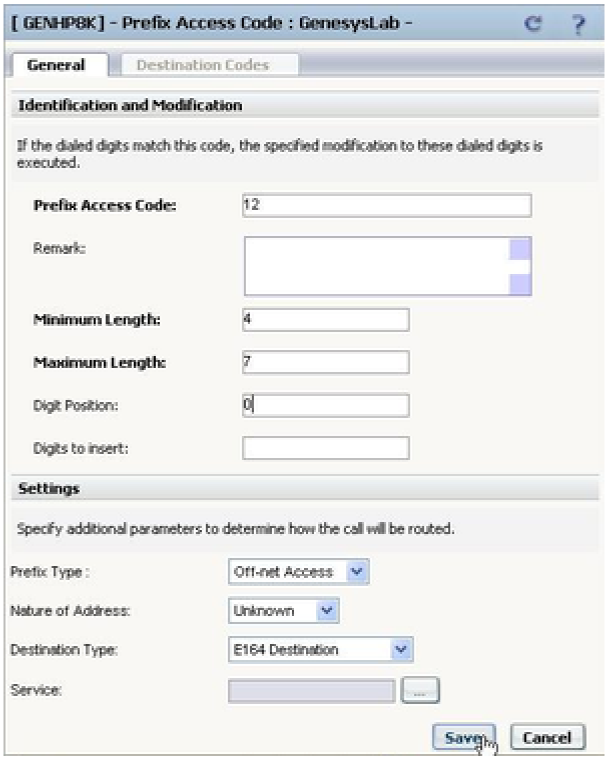
Start

1. Click `Private Numbering Plan`, and then click the SIP Server Numbering Plan—for example, `SIPServer`.
2. Click `Translation`, click `Prefix Access Codes`, and then click Add (see the following figure).



Selecting Prefix Access Codes

3. For calls that are to be routed to Subscribers: In the `Prefix Access Code: <Business Group>` dialog box, do the following:
   a. In the `Prefix Access Code` text box, enter the digits you want to use to route calls to Subscribers. **Note:** For the SIP Server Numbering Plan, minimal modifications should be required. Dialed numbers should be modified before they reach SIP Server. This convention should be followed at all sites, to simplify the solution as much as possible.
   b. Set the `Prefix Type` text box to `Off-net Access`.
   c. Set the `Nature of Address` text box to `Unknown`.
   d. Set the `Destination Type` text box to `E164 Destination`.
   e. Click Save.



[GENHP8K] - Prefix Access Code : GenesysLab -

Identification and Modification

If the dialed digits match this code, the specified modification to these dialed digits is executed.

Prefix Access Code:	12
Remark:	
Minimum Length:	4
Maximum Length:	7
Digit Position:	0
Digits to insert:	

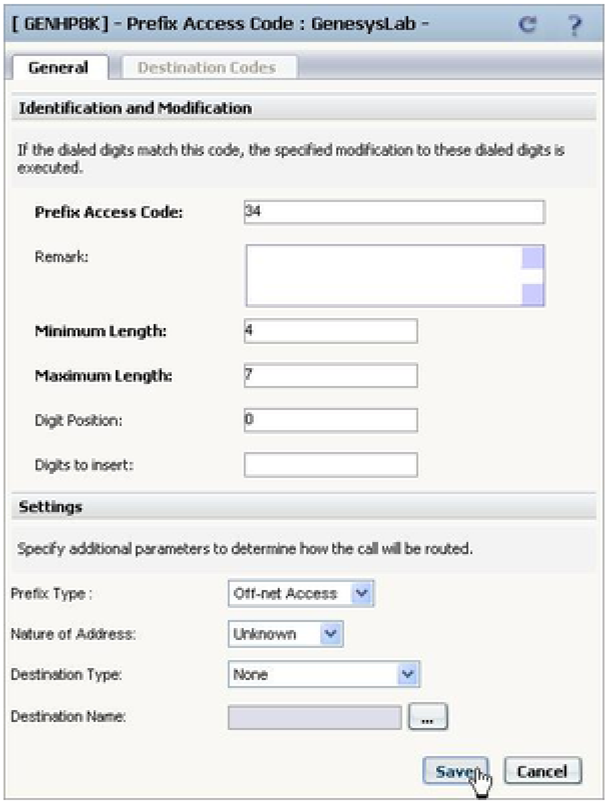
Settings

Specify additional parameters to determine how the call will be routed.

Prefix Type :	Off-net Access
Nature of Address:	Unknown
Destination Type:	E164 Destination
Service:	

Configuring a Prefix Access Code for Calls Routed to Subscribers

6. When the confirmation message box appears, informing you that the Prefix Access Code was created successfully, click `Close`.
7. If agents will be allowed to make external calls: In the `Prefix Access Code` dialog box, click `Add` again.
8. In the `Prefix Access Code` dialog box, do the following:
   a. In the `Prefix Access Code` text box, enter the digits that you want to use to route calls to Gateways. The matched digits will be site-specific, and there should be minimal modification of the digit string.
   b. Set the `Prefix Type` text box to `Off-net Access`.
   c. Set the `Nature of Address` text box to `Unknown`.
   d. Set the `Destination Type` text box to `None`, so you will be able to route the call from a Destination Code.
   e. Click `OK`.



[GENHP8K] - Prefix Access Code : GenesysLab -

Identification and Modification

If the dialed digits match this code, the specified modification to these dialed digits is executed.

Prefix Access Code:	34
Remark:	
Minimum Length:	4
Maximum Length:	7
Digit Position:	0
Digits to insert:	

Settings

Specify additional parameters to determine how the call will be routed.

Prefix Type :	Off-net Access
Nature of Address:	Unknown
Destination Type:	None
Destination Name:	

Configuring a Prefix Access Code for Calls Routed to Gateways

6. When the confirmation message box appears, informing you that the Prefix Access Code was created successfully, click `Close`.

End

Next Steps

Continue with the following procedure, unless calls are routed only to Subscribers:

- Procedure: Configuring SIP Server Destination Codes

7. Configure Destination Codes.

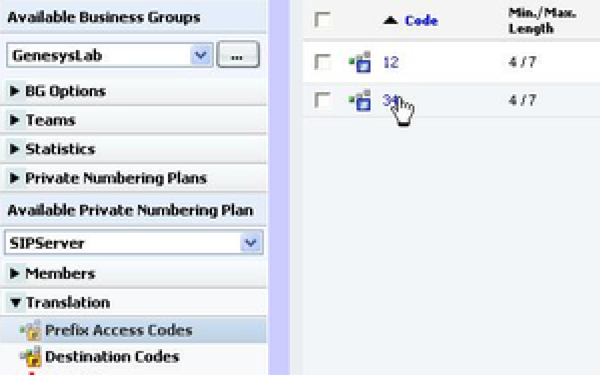
Configuring SIP Server Destination Codes

Purpose

To configure SIP Server Destination Codes to route calls to non-Subscriber devices.

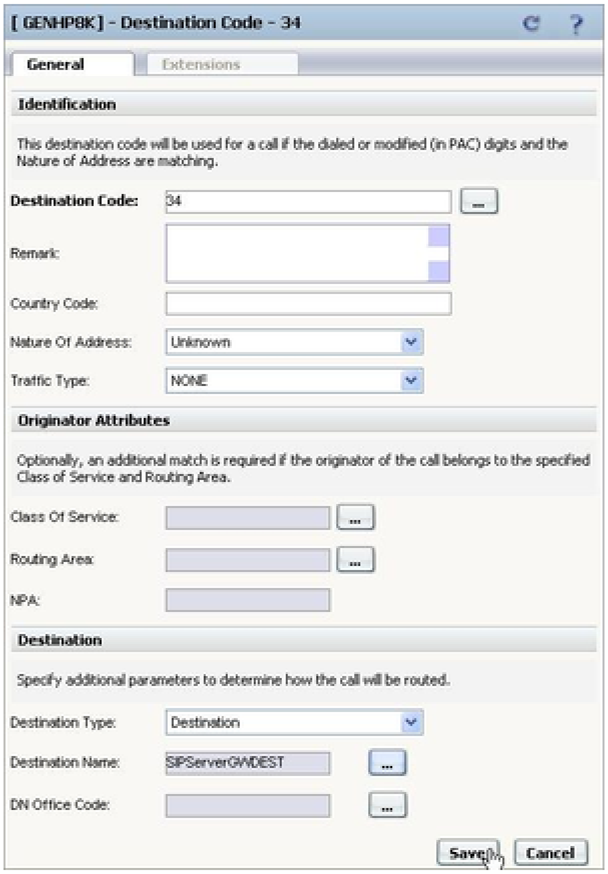
Start

1. Click `Private Numbering Plan`, and then click the SIP Server Numbering Plan—for example, `SIPServer`.
2. Click `Prefix Access Codes`.
3. Click the Prefix Access Code that you created for non-Subscriber devices (see the following figure).



Selecting a Prefix Access Code

4. In the `Prefix Access Code` dialog box, click the `Destination Codes` tab.
5. In the `Destination Code` dialog box, do the following:
   a. Set the `Destination Type` text box to `Destination`.
   b. Set the `Destination Name` text box to the Destination that you created for SIP Server in Configuring SIP Server Destinations for Gateways, by clicking the browse (...) button.


[GENHP8K] - Destination Code - 34

Identification

This destination code will be used for a call if the dialed or modified (in PAC) digits and the Nature of Address are matching.

Destination Code: 34

Remark:

Country Code:

Nature Of Address: Unknown

Traffic Type: NONE

Originator Attributes

Optionally, an additional match is required if the originator of the call belongs to the specified Class of Service and Routing Area.

Class Of Service:

Routing Area:

NPA:

Destination

Specify additional parameters to determine how the call will be routed.

Destination Type: Destination

Destination Name: SIPServerGWDEST

DN Office Code:

Configuring a Destination Code

6. Click Save.
7. When the confirmation message box appears, informing you that the Destination Code was created successfully, click `Close`.

End

8. Configure Agent Destinations.

Configuring an Agent Destination for SIP Server

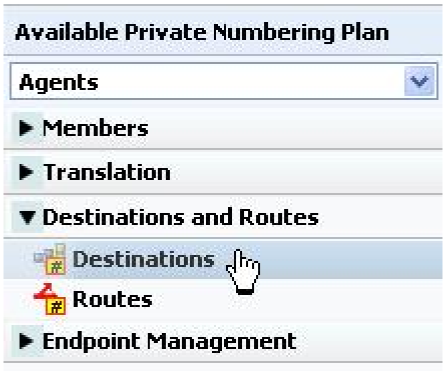
Purpose

To configure a Destination for the Agent Numbering Plan for SIP Server.

Start

1. Click `Private Numbering Plan`, and then click the Agent Numbering plan—for example, `Agents`.

2. Click `Destinations and Routes`, click `Destinations`, and then click Add.

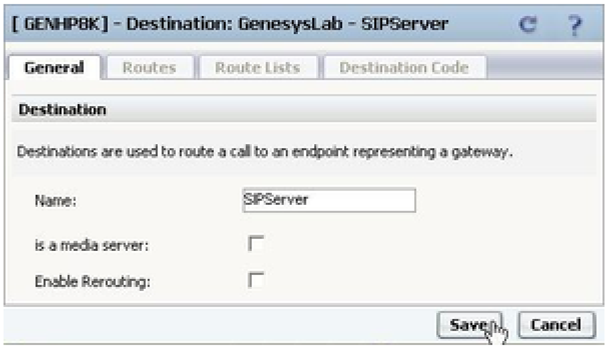


Selecting Destinations

3. In the `Destination - <Agent Numbering Plan>` dialog box, click the `General` tab, and then do the following:
    a. In the Name text box, enter a unique name for the Destination.

       Note: Destinations must be unique within the switch configuration database, not just within the Numbering Plan and Business Group.

    b. Make sure that all check boxes are cleared.
    c. When you are finished, click Save, and then close the dialog box.



Destination

Destinations are used to route a call to an endpoint representing a gateway.

Name: SIPServer

is a media server: □

Enable Rerouting: □

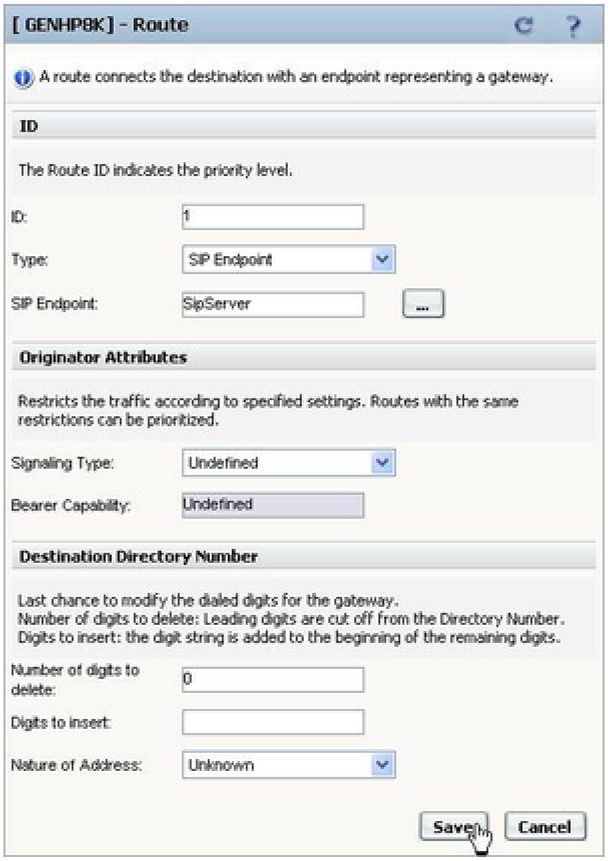
Configuring a SIP Server Destination in the Agent Numbering Plan

4. Click the Destination that you just created—for example, `SIPServer`.
5. Click the `Routes` tab, and then click Add.
6. In the `Route` dialog box, do the following:
    a. In the `ID` text box, enter 1.

       Note: The ID of the first Route must always be 1.

    b. Set the Type text box to `SIP Endpoint`.
    c. Set the `SIP Endpoint` text box to the Endpoint that you created for SIP Server in Configuring a SIP Server Endpoint, by clicking the browse (...) button.
    d. When you are finished, click Save.

Note: Genesys recommends that you not modify the dialed-digit string that is passed on to SIP Server at this point.



[GENHP8K] - Route

A route connects the destination with an endpoint representing a gateway.

ID

The Route ID indicates the priority level.

ID: 1

Type: SIP Endpoint

SIP Endpoint: SipServer

Originator Attributes

Restricts the traffic according to specified settings. Routes with the same restrictions can be prioritized.

Signaling Type: Undefined

Bearer Capability: Undefined

Destination Directory Number

Last chance to modify the dialed digits for the gateway.
Number of digits to delete: Leading digits are cut off from the Directory Number.
Digits to insert: the digit string is added to the beginning of the remaining digits.

Number of digits to delete: 0

Digits to insert:

Nature of Address: Unknown

Configuring a Route for SIP Server in the Agent Numbering Plan

5. When the confirmation message box appears, informing you that the Route was added successfully, click `Close`.

End

9. Configure Agent Access and Destination Codes.

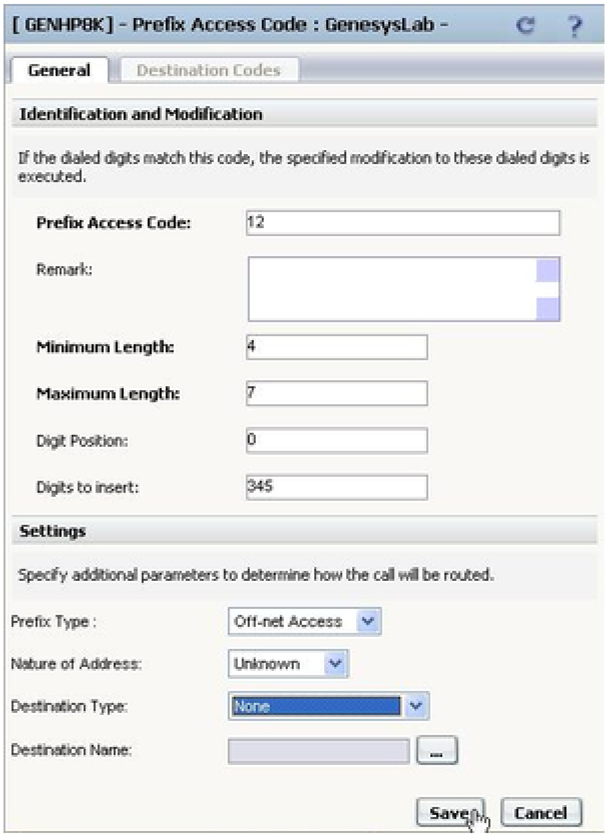
Configuring Agent Prefix Access Codes and Destination Codes

In this section, you configure dialing patterns for the Agents. Every number that the agent dials must be configured. If an agent dials a four-digit extension, the Prefix Access Code should be configured to convert the dialed-digit string to the full E.164 code that OpenScape Voice expects. If the agent dials a number that must to be routed to an external gateway, make sure that the dialed-digit string is correct for that gateway before it reaches SIP Server.

As mentioned earlier, all calls must go to SIP Server first; otherwise, the calls will not be visible to SIP Server. In the Private Numbering Plan for agents, every Prefix Access Code must route the call to a Destination Code that points the call to SIP Server. It is best to copy the nonagent Prefix Access Codes from the General Numbering Plan; however, make sure that the destination is always SIP Server.

Start

1. Click `Private Numbering Plan`, and then click the Agent Numbering Plan—for example, `Agents`.
2. Click `Translation`, click `Prefix Access Codes`, and then click `Add`.
3. In the `Prefix Access Code` dialog box, do the following:
   a. In the `Prefix Access Code` text box, enter the digits you that want to use for routing, and any modifications that OpenScape Voice will need to make in order to route the call properly.
   b. Set the `Prefix Type` text box to `Off-net Access`.
   c. Set the `Nature of Address` text box to `Unknown`.
   d. Set the `Destination Type` text box to `None`.
   e. Click `Save`, and close the dialog box.



[GENHP8K] - Prefix Access Code : GenesysLab -

Identification and Modification

If the dialed digits match this code, the specified modification to these dialed digits is executed.

Prefix Access Code: 12

Remark:

Minimum Length: 4

Maximum Length: 7

Digit Position: 0

Digits to insert: 345

Settings

Specify additional parameters to determine how the call will be routed.

Prefix Type : Off-net Access

Nature of Address: Unknown

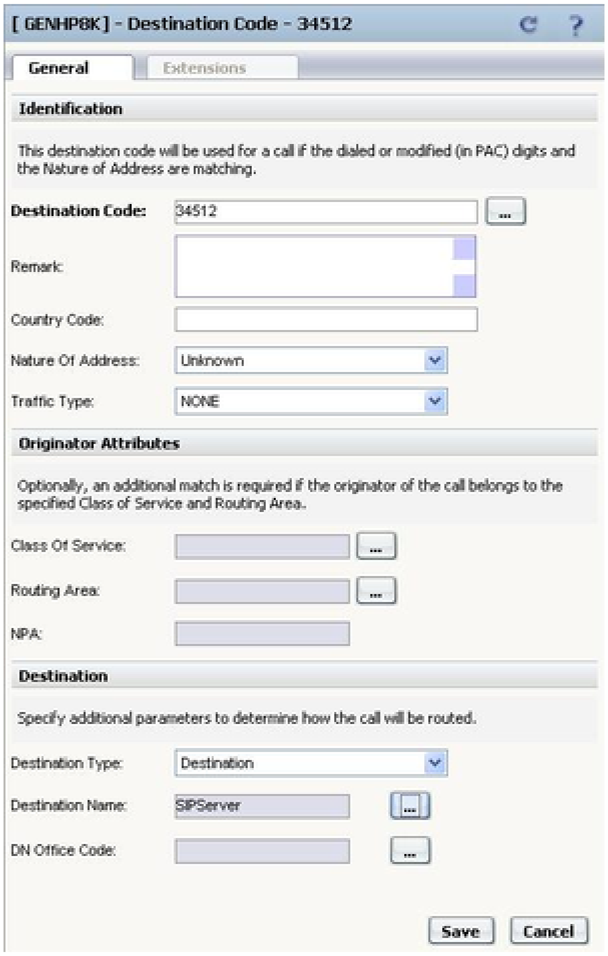
Destination Type: None

Destination Name:

Configuring a Prefix Access Code for the Agent Numbering Plan

   f. In the `Prefix Access Code` dialog box, click the Prefix Access Code that you just created, and then click the `Destination Codes` tab.

7. In the `Destination Code` dialog box, click the `General` tab, and then do the following:
    a. Do not modify the `Destination Code` text box.
    b. Make sure that the `Nature of Address` text box is set to `Unknown`.
    c. Make sure that the `Destination Type` text box is set to `Destination`.
    d. Set the `Destination Name` text box to the Destination that you created for SIP Server in Configuring an Agent Destination for SIP Server—for example, `SIPServer`--by clicking the browse (...) button.
    e. When you are finished, click Save.



[GENHP8K] - Destination Code - 34512

Identification

This destination code will be used for a call if the dialed or modified (in PAC) digits and the Nature of Address are matching.

Destination Code: 34512

Remark:

Country Code:

Nature Of Address: Unknown

Traffic Type: NONE

Originator Attributes

Optionally, an additional match is required if the originator of the call belongs to the specified Class of Service and Routing Area.

Class Of Service:

Routing Area:

NPA:

Destination

Specify additional parameters to determine how the call will be routed.

Destination Type: Destination

Destination Name: SIPServer

DN Office Code:

Configuring a Destination Code for the Agent Destination

6. When the confirmation message box appears, informing you that the Destination Code was created successfully, click `Close`.
7. Repeat Steps 2-6 to create other Prefix Access Codes and Destination Codes, as necessary.

End

10. Configure Click-to-Answer.

Optional Configuration for SIP Server

This configuration is not required for the integration to work, however, some might be required by local laws, or make the solution easier to configure.

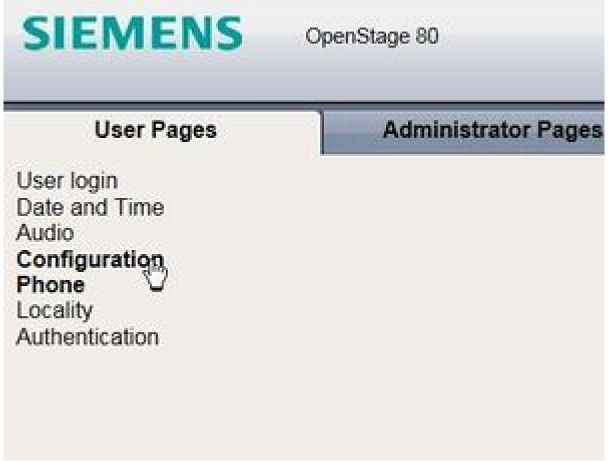
Configuring Click-to-Answer

Purpose

The Click-to-Answer feature enables agents to click within Genesys Agent Desktop to answer the phone. The Click-to-Answer feature requires the referenced Patchset on OpenScape Voice and a device that supports it. The current procedure provides instructions for OpenStage phones.

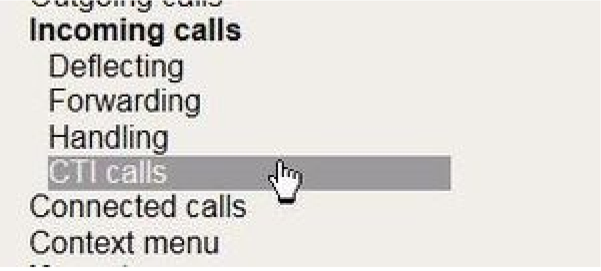
Start

1. On the phone that you have to configure, select `Configuration` (see the following figure).



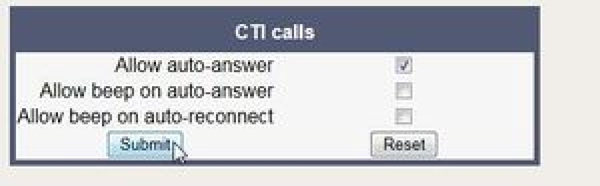
Selecting Configuration on the OpenStage Phone

2. Click `Incoming calls`, and then click `CTI calls` (see the following figure).



Configuring CTI Calls on the OpenStage Phone

3. Select the `Allow auto-answer` check box, and click `Submit` (see the following figure).



CTI calls

Allow auto-answer ☑

Allow beep on auto-answer □

Allow beep on auto-reconnect □

Submitting Allow auto-answer on the OpenStage Phone

4. Repeat Steps 1-3 for every agent phone on the switch.

End

11. Configure emergency call routing.

Optional Configuration for SIP Server

This configuration is not required for the integration to work, however, some might be required by local laws, or make the solution easier to configure.

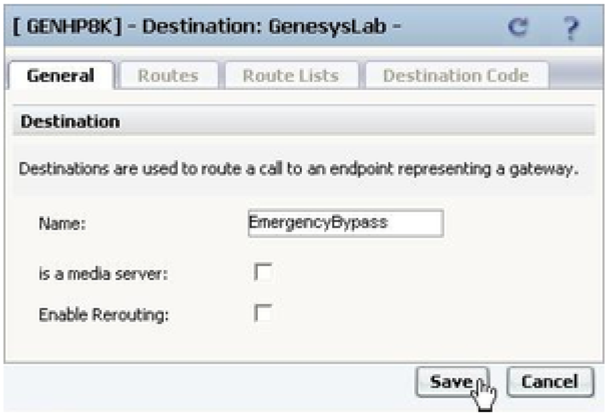
Configuring emergency call routing

The emergency call routing feature provides alternate call routing in cases in which SIP Server is unavailable, if your local emergency (or 911) laws require some form of alternate routing for agents.

During the first 30 seconds after the emergency calling support is activated, calls will fail to route. After that, OpenScape Voice will route calls via the alternate route that you configure and the calls will work.

Start

1. Log in to the HiPath Assistant, and navigate to the Business Group of the contact center that you want to configure—for example, `GenesysLab`.
2. Click `Private Numbering Plan`, and then click the `Agent Numbering Plan`.
3. Click `Destinations and Routes`, click `Destinations`, and then click `Add`.
4. In the `Destination` dialog box, do the following:
    a. In the `Name` text box, enter a new destination for the gateway through which you want emergency calls to go—for example, `EmergencyBypass`.
    b. Make sure that all check boxes are cleared.
    c. Click `Save`.



Destination

Destinations are used to route a call to an endpoint representing a gateway.

Name: EmergencyBypass

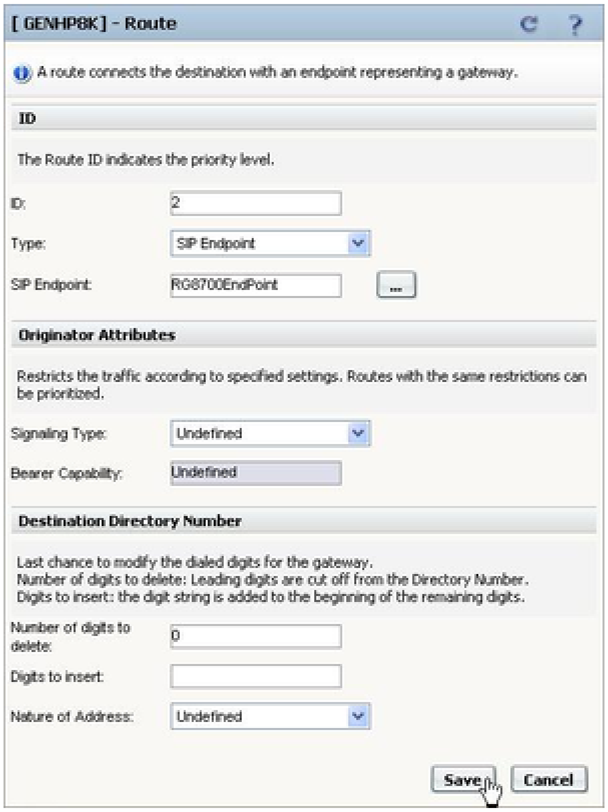
is a media server: □

Enable Rerouting: □

Configuring a Destination for Emergency Call Routing

4. Click the Destination that you just created—for example, `EmergencyBypass`.
5. Click the `Routes` tab, and then click Add. In this step you are adding a route that goes to SIP Server. This is necessary in order to prevent calls from bypassing SIP Server while it is working.
6. In the `Route` dialog box, do the following:
   a. In the `ID` text box, enter 1. This route goes to SIP Server, just like all the others.
   b. Set the Type text box to `SIP Endpoint`.
   c. Set the `SIP Endpoint` text box to the Endpoint that you created in Configuring a SIP Server Endpoint.

4. When you are finished, click Save.
5. Click the Destination that you just created—for example, `EmergencyBypass`.
6. Click the `Routes` tab, and then click `Add` again.
7. In the `Route` dialog box, do the following:
   a. In the `ID` text box, enter 2.
   b. Set the Type text box to `SIP Endpoint`.
   c. Set the `SIP Endpoint` text box to the gateway for emergency calling.
   d. When you are finished, click Save.



[GENHP8K] - Route

A route connects the destination with an endpoint representing a gateway.

ID

The Route ID indicates the priority level.

ID: 2

Type: SIP Endpoint

SIP Endpoint: RG8700EndPoint

Originator Attributes

Restricts the traffic according to specified settings. Routes with the same restrictions can be prioritized.

Signaling Type: Undefined

Bearer Capability: Undefined

Destination Directory Number

Last chance to modify the dialed digits for the gateway.
Number of digits to delete: Leading digits are cut off from the Directory Number.
Digits to insert: the digit string is added to the beginning of the remaining digits.

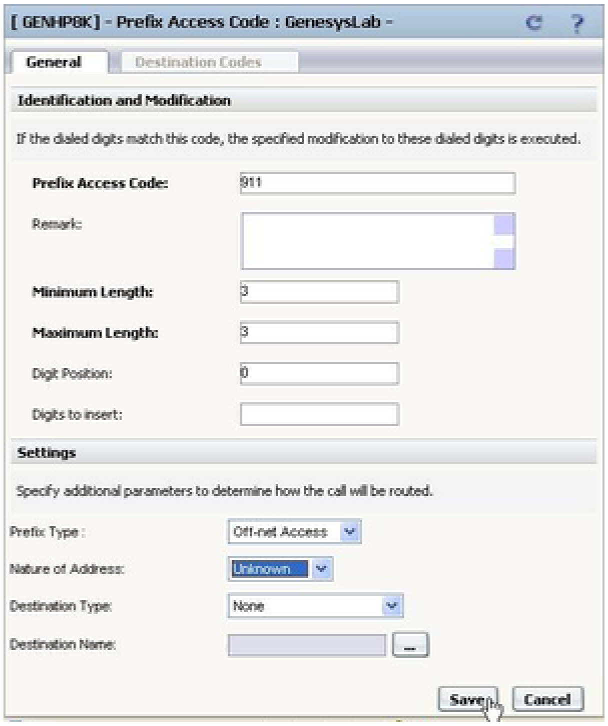
Number of digits to delete: 0

Digits to insert:

Nature of Address: Undefined

Configuring a Route for Emergency Call Routing

5. Click `Prefix Access Codes`, and then click Add.
6. In the `Prefix Access Code` dialog box, do the following:
   a. In the `Prefix Access Code` text box, enter the digits for your emergency number.
   b. Set the `Prefix Type` text box to `Off-net Access`.
   c. Set the `Nature of Address` text box to Unknown.
   d. Set the `Destination Type` text box to None.
   e. Click Save, and close the dialog box.



[GENHP8K] - Prefix Access Code : GenesysLab -

Identification and Modification

If the dialed digits match this code, the specified modification to these dialed digits is executed.

Prefix Access Code: 911

Remark:

Minimum Length: 3

Maximum Length: 3

Digit Position: 0

Digits to insert:

Settings

Specify additional parameters to determine how the call will be routed.

Prefix Type : Off-net Access

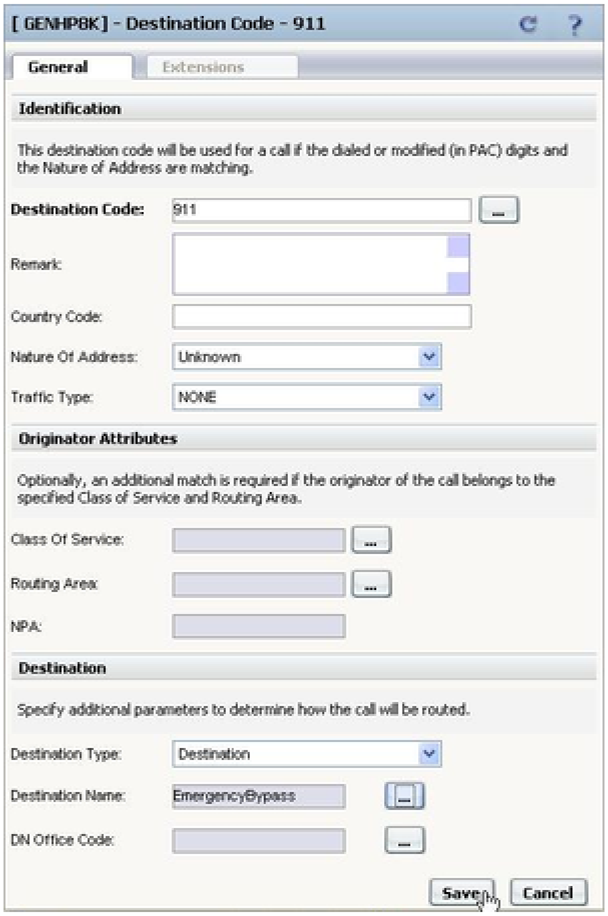
Nature of Address: Unknown

Destination Type: None

Destination Name:

Configuring a Prefix Access Code for Emergency Call Routing

6. In the `Prefix Access Code` dialog box, click the `Destination Codes` tab.
7. On the `General` tab, do the following:
    a. Make sure that the `Destination Type` text box is set to `Destination`.
    b. Set the `Destination Name` text box to the Destination that you created in Step 4—for example, `EmergencyBypass`—by clicking the browse (...) button.
    c. When you are finished, click `OK`.



[GENHP8K] - Destination Code - 911

Identification

This destination code will be used for a call if the dialed or modified (in PAC) digits and the Nature of Address are matching.

Destination Code: 911

Remark:

Country Code:

Nature Of Address: Unknown

Traffic Type: NONE

Originator Attributes

Optionally, an additional match is required if the originator of the call belongs to the specified Class of Service and Routing Area.

Class Of Service:

Routing Area:

NPA:

Destination

Specify additional parameters to determine how the call will be routed.

Destination Type: Destination

Destination Name: EmergencyBypass

DN Office Code:

Configuring a Destination Code for Emergency Call Routing

End

Next Steps

- Configuration of OpenScape Voice is now complete. Proceed with Configuring DN Objects.

Configuring DN Objects

This page provides an overview of the main steps to configure DNs under the OpenScape Voice `Switch` object in the Configuration Layer. The `Switch` object is assigned to the appropriate SIP Server.

Configuring DN Objects

1. Configure a Voice over IP Service DN.

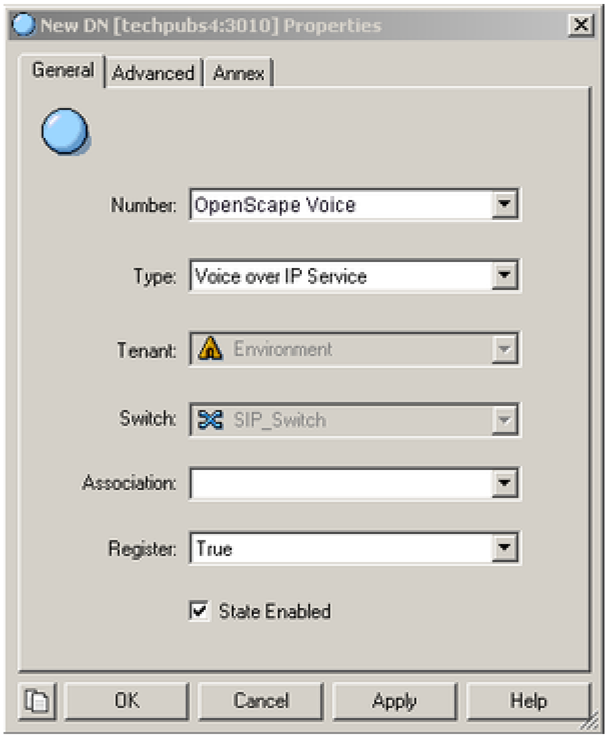
Configuring a Voice over IP Service DN

Purpose

To configure a DN of type `Voice over IP Service` that specifies the connection and options for OpenScape Voice communication with a SIP Server that is running in Application Server (B2BUA) mode.

Start

1. In Configuration Manager, under a configured `Switch` object, select the `DNs` folder. From the `File` menu, select `New` > `DN` to create a new `DN` object.
2. In the `New DN Properties` dialog box, click the `General` tab, and then specify the following properties:
    a. `Number`: Enter the softswitch name—for example, `OpenScape Voice`. Although this name is currently not used for any messaging, it must still be unique.
    b. `Type`: Select `Voice over IP Service` from the drop-down box.



Number: OpenScape Voice

Type: Voice over IP Service

Tenant: Environment

Switch: SIP_Switch

Association:

Register: True

☑ State Enabled

Creating a Voice over IP Service DN for OpenScape Voice: Sample Configuration

c. Click the Annex tab.

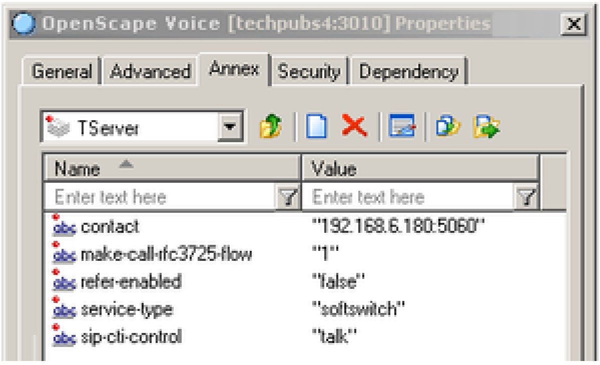
d. Create a section that is named `TServer`. In the `TServer` section, create options as specified in the following table.

Option Name	Option Value	Description
contact	`<ipaddress>:<SIP port>`	The contact URI that SIP Server uses for communication with the OpenScape Voice softswitch, where `<ipaddress>` is the IP address of the softswitch and `<SIP port>` is the SIP port number of the softswitch.
dual-dialog-enabled	`false`	Set this option to `false` if Siemens phones are used in `re-INVITE` mode for third-party call-control (3pcc) operations.
makecall-subst-uname	1, or none	For OpenScape Voice version 2.1, set this option to 1.For OpenScape Voice version 2.2 and later, do not configure this option. When this option

Option Name	Option Value	Description
		is set to 1, SIP Server sets the From header to the same value as the To header in the INVITE request, to work around issues with pre-2.2 versions of OpenScape Voice.
make-call-rfc3725-flow	1	Set this option to 1.When this option is set to 1, SIP Server selects the SIP call flow number 1 (described in RFC 3725) for a call that is initiated by a TMakeCall request.
refer-enabled	false	Set this option to false for SIP Server to use a re-INVITE request method when contacting the softswitch. This is the only method that is supported in the OpenScape Voice configuration.
ring-tone-on-make-call	true	When this option is set to true, SIP Server connects the caller with an audio ringtone from Stream Manager when the destination endpoint responds with a 180 Ringing message.
service-type	softswitch	Set this option to softswitch.
sip-cti-control	talk	When this option is set to talk, SIP Server instructs the endpoint to go off-hook by sending a SIP NOTIFY message with the Event: talk header. This enables a TAnswerCall request to be sent to SIP Server. SIP Server then sends the NOTIFY message to the switch. Setting this option to talk sets the default for all endpoints that are configured with this softswitch.The talk value is supported only on OpenScape Voice version 2.2 Patchset 14 or later. **Note:** You must also configure OpenScape

Option Name	Option Value	Description
		Voice to support this functionality. See Configuring Click-to-Answer.
sip-ring-tone-mode	1	When this option is set to 1, SIP Server waits for a response from the called device, and connects Stream Manager to a call to play an audio ring tone only when the returned response cannot be used as the offer to a calling device.

e. When you are finished, click `Apply`.



Name	Value
contact	"192.168.6.180:5060"
make-call-rfc3725-flow	"1"
refer-enabled	"false"
service-type	"softswitch"
sip-cti-control	"talk"

Setting Options for a Voice over IP Service DN: Sample Configuration

End

2. Configure a Trunk DN.

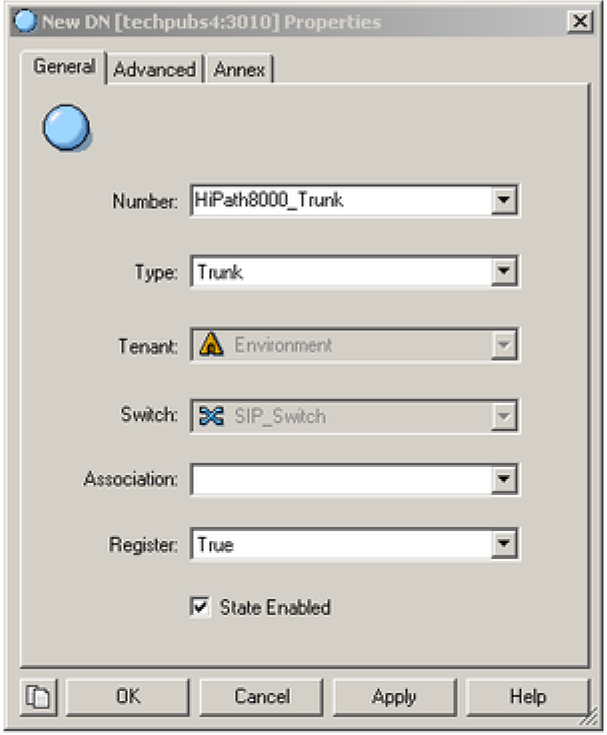
Configuring a Trunk DN

Purpose

To configure a DN of type Trunk that specifies how SIP Server handles outbound calls. It is also used for configuration of gateways, SIP proxies (including connections to other instances of SIP Server), and other SIP-based applications. From the SIP Server perspective, OpenScape Voice in Application Server (B2BUA) mode is considered a gateway or SIP proxy.

Start

1. Under a configured `Switch` object, select the DNs folder. From the `File` menu, select New > DN to create a new DN object.
2. In the `New DN Properties` dialog box, click the `General` tab, and then specify the following properties:
   a. `Number:` Enter a name for the Trunk DN. This name can be any unique value, and it can be a combination of letters and numbers.
   b. `Type:` Select Trunk from the drop-down box.



New DN [techpubs4:3010] Properties

Number: HiPath8000_Trunk

Type: Trunk

Tenant: Environment

Switch: SIP_Switch

Association:

Register: True

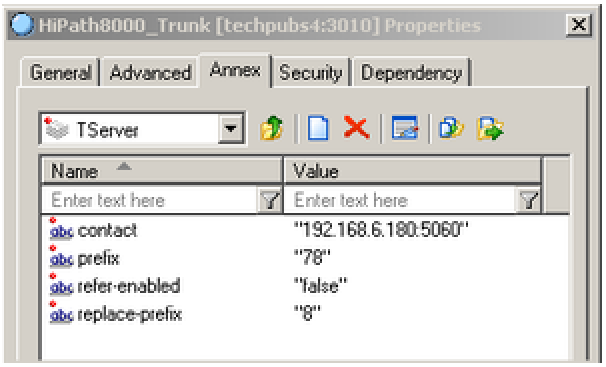
☑ State Enabled

Creating a Trunk DN for OpenScape Voice: Sample Configuration

3. Click the Annex tab.
4. Create a section that is named `TServer`. In the `TServer` section, create options as specified in the following table.

Option Name	Option Value	Description
contact	`<ipaddress>:<SIP port>`	The contact URI that SIP Server uses for communication with the OpenScape Voice softswitch, where `<ipaddress>` is the IP address of the softswitch and `<SIP port>` is the SIP port number of the

Option Name	Option Value	Description
		softswitch.
prefix	Any numerical string	The initial digits of the number that SIP Server matches to determine whether this trunk should be used for outbound calls. For example, if `prefix` is set to 78, dialing a number that starts with 78 will cause SIP Server to consider this trunk a gateway or SIP proxy. If multiple `Trunk` objects match the prefix, SIP Server will select the one that has the longest prefix that matches.
refer-enabled	`false`	Set this option to `false` for SIP Server to use a `re-INVITE` request method when contacting the softswitch. This is the only method that is supported in the OpenScape Voice configuration.
replace-prefix	Any numerical string	The digits (if necessary) that replace the prefix in the DN. For example, if `prefix` is set to 78, and `replace-prefix` is set to 8, the number 786505551212 will be replaced with 86505551212 before it is sent to the gateway or SIP proxy (in this case, OpenScape Voice).



Name	Value
contact	"192.168.6.180:5060"
prefix	"78"
refer-enabled	"false"
replace-prefix	"8"

Setting Options for a Trunk DN: Sample Configuration

5. When you are finished, click `Apply`.

End

3. Configure Extension DNs.

Configuring Extension DNs

Purpose

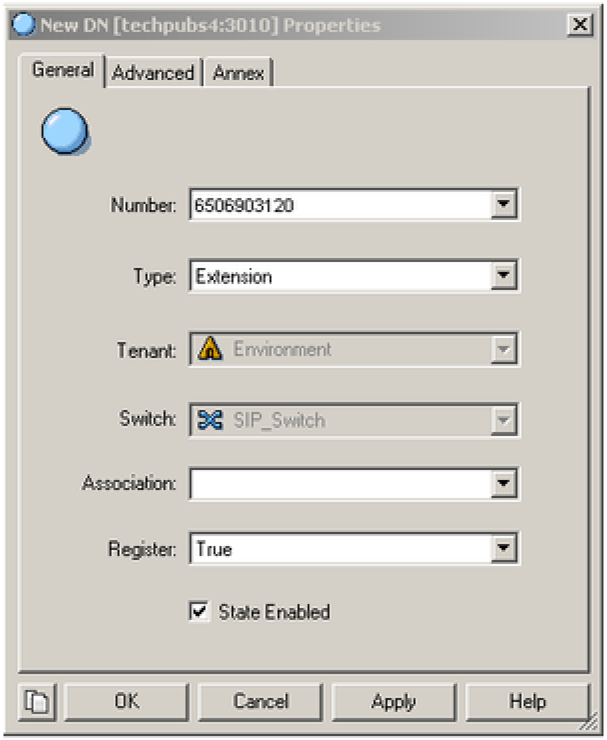
To configure DNs of type `Extension` that represent agent phone extensions and register directly with the softswitch.

When you configure an extension where the phone registers directly with SIP Server, you must configure options in the `TServer` section on the `Annex` tab. However, if you are using a softswitch in Application Server (B2BUA) mode, SIP Server takes the `Extension` DN name together with the value of the `contact` option in the softswitch object configuration (not the `Extension` object) to access the phone. This procedure describes the configuration for phones that are registered directly with OpenScape Voice and not with SIP Server. As a result, SIP Server sends the request to OpenScape Voice to communicate with the phone.

Start

1. Under a configured `Switch` object, select the `DNs` folder. From the `File` menu, select `New > DN` to create a new `DN` object.
2. In the `New DN Properties` dialog box, click the `General` tab, and then specify the following properties:
    a. `Number:` Enter a name for the `Extension` DN. In general, this should be the 10-digit phone number of the extension. You must not use the @ symbol or a computer name. The name of this DN must map to the SIP user name of the extension in OpenScape Voice.

    b. `Type:` Select `Extension` from the drop-down box.



Number: 6506903120

Type: Extension

Tenant: Environment

Switch: SIP_Switch

Association:

Register: True

☑ State Enabled

Creating an Extension DN for OpenScape Voice: Sample Configuration

c. When you are finished, click `Apply`.

Note: No configuration options are required for the `Extension` DN. Adding configuration options—such as `contact`, `password`, `refer-enabled`, and others—might cause unexpected results.

End

4. Configure Routing Point DNs.

Configuring Routing Point DNs

Purpose

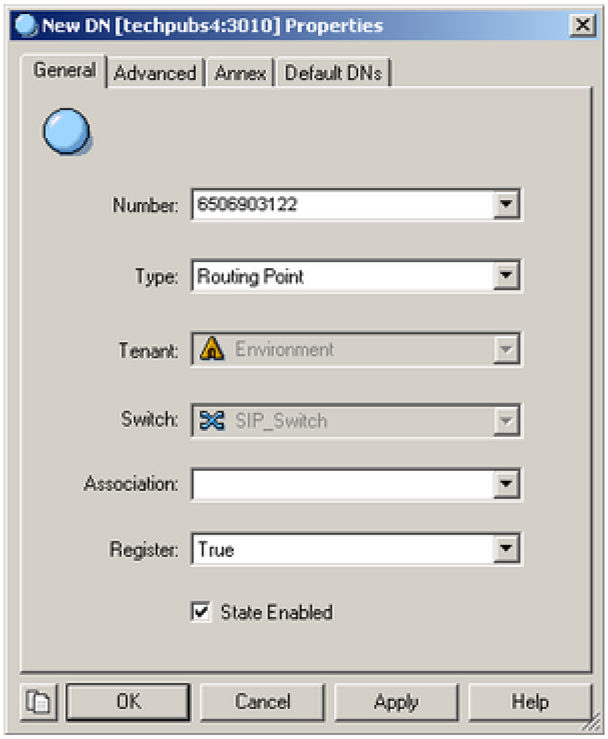
To configure a DN of type `Routing Point` that is used to execute a routing strategy with Genesys URS. When SIP Server receives an `INVITE` request on a DN that is configured as a Routing Point, it sends an `EventRouteRequest` message to URS.

Start

1. Under a configured `Switch` object, select the `DNs` folder. From the `File` menu, select `New > DN` to create a new `DN` object.
2. In the `New DN Properties` dialog box, click the `General` tab, and then specify the following properties:

a. `Number:` Enter a number for the `Routing Point` DN. This number must be configured on OpenScape Voice.

b. `Type:` Select `Routing Point` from the drop-down box.



New DN [techpubs4:3010] Properties

Number: 6506903122

Type: Routing Point

Tenant: Environment

Switch: SIP_Switch

Association:

Register: True

☑ State Enabled

Creating a Routing Point for OpenScape Voice: Sample Configuration

c. When you are finished, click `Apply`.

Although no configuration options are required for the Routing Point, URS does look at options to determine how to handle the Routing Point and what strategy is currently loaded. For details about these options, see the *Genesys 8.x Universal Routing Server Reference Guide*.

End

Support for First-Party Call-Control Operations

Beginning with the Siemens OpenScape Voice switch release V5, SIP Server provides support for first-party call-control (1pcc) operations, including a transfer that uses the REFER method when it is integrated with the OpenScape Voice softswitch.

Feature Configuration

To support 1pcc operations, you must configure a DN of type `Voice over IP Service` DN and `Extension` DNs.

To configure a DN of type `Voice over IP Service` DN, see Procedure: Configuring a Voice over IP Service DN for OpenScape Voice.

To configure `Extension` DNs, see Procedure: Configuring Extension DNs for OpenScape Voice. No configuration options are required for `Extension` DNs.

To enable a blind transfer, set the `blind-transfer-enabled` configuration option to `true`, at the SIP Server Application level, or at the `Voice over IP Service` DN level.

Feature Limitations

There are several known limitations that result from the Siemens OpenScape Voice release V5 integration:

- Mix of 1pcc and 3 pcc with a call is not supported.
- For 3 pcc calls, the re-`INVITE`--based call control method is used.

Support for Split-Node Deployments

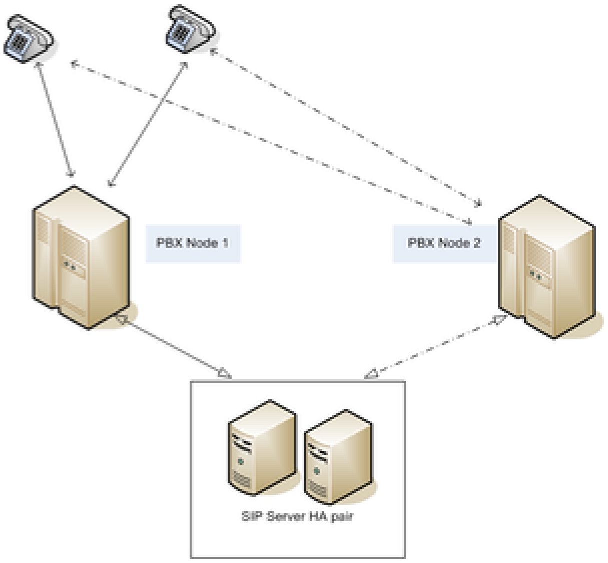
The Siemens OpenScape PBX can be configured to operate in a SIP Business Continuity configuration. There are two supported modes:

- High-availability pair configuration, in which two OpenScape Voice nodes are physically located in the same area and share the same IP address for initiating and receiving calls.
- Split-node configuration, in which each OpenScape Voice node is geographically separated from the other. In this configuration, each PBX node has its own IP address on different subnets. Each node can be active for certain DNs; so, when a failure occurs, the remaining node will handle all calls, without taking over the IP address of the failed node.

Previous deployments of SIP Server with OpenScape Voice utilized only the first mode. Beginning with release 8.1, SIP Server supports a split-node configuration.

In a split-node configuration of the OpenScape Voice (with the same SIP Server), each OpenScape Voice node has a different IP address on different subnets. When both nodes are active, calls from each node arrive at SIP Server (typically, each node handles a subset of DNs). SIP Server recognizes all calls as coming from the same switch, as both nodes are part of the same OpenScape Voice switch.

When one of the OpenScape Voice nodes fails, the remaining node takes over all existing and future calls. SIP Server will handle existing and future calls to and from the remaining node, which has a different IP address on a different subnet. This take-over process will be transparent to endpoints (which are registered at the OpenScape Voice switch and will be re-registered at the remaining node in case of failure), to agents, and to Genesys T-Library client applications. See the **Split-Node Deployment** figure.



Split-Node Deployment

Feature Configuration

To support the split-node configuration, all OpenScape Voice (or PBX) nodes are represented in the configuration environment as a single `Voice over IP Service` object with the `service-type` option set to `softswitch`.

All PBX nodes share the same FQDN, which could be resolved through the DNS SRV records. DNS SRV records must be administered in such a way that the IP address of the node, in which endpoints are registered by default, has the highest priority. SIP Server tests the availability of all resolved addresses by using `OPTION` requests. The available address with the highest priority is used for SIP communication. If the original node fails, endpoints are re-registered at an alternative node. SIP Server starts using the alternative node when it discovers that the original node is not available.

The Configuring Split-Node deployment table lists the tasks that are required to configure the SIP Server and DN objects to support SIP Business Continuity with the Siemens OpenScape PBX.

Configuring Split-Node deployment

Objective	Related procedures and actions
1. Configure each SIP Server.	In the SIP Server `Application` object, in the `TServer` section, configure the following options: • `sip-enable-gdns`—Set this option to `true.` This enables the internal DNS client. • `sip-address`—Set this option to the IP address of the SIP Server host computer (not the URI). • `sip-address-srv`—Set this option to the FQDN of the SIP Server host computer. SIP Server will send this address as its own contact inside SIP requests to the PBX.
2. Configure the `Voice over IP Service` DN.	Configure the DN of type `Voice over IP Service` with `service-type` set to `softswitch` with the following options: • `contact=<FQDN of Siemens PBX>`—The FQDN must be resolvable by DNS SRV records. • `oos-check=`—Specify the time interval, in seconds, in which SIP Server will send `OPTION` requests to transport addresses returned by DNS SRV resolution. SIP Server will send an `OPTION` request by transport for those addresses at which active SIP communication is not present. • `oos-force=`—Specify the time interval, in seconds, in which SIP Server will mark the transport address as unavailable when there is no response to the `OPTION` request. This configuration option applies only if the configuration option `oos-check` is set to a non-

Objective	Related procedures and actions
	zero value. See also the following table for additional configuration options for this softswitch DN.
3. Configure `Extension` DNs.	Complete the following procedure: - Procedure: Configuring Extension DNs for OpenScape Voice
4. (Optional) Configure a `Trunk` DN.	For SIP Server to handle outbound calls, configure a DN of type `Trunk` with the following option:* `contact=<FQDN of Siemens PBX>`—This is the same value as configured on the softswitch DN.

Feature Limitations

Verification of split-node functionality was done with geographically-separated nodes that were configured without RG8700 as a SIP Proxy Server.

Asterisk

This topic describes how to integrate SIP Server with the Asterisk switch. It contains the following sections:

- Overview
- Asterisk for Business Calls Routing
- Asterisk as a Voicemail Server
- Asterisk as a Media Server

Note: The instructions in this topic assume that both Asterisk and SIP Server are fully functional as stand-alone products. The instructions only highlight modifications to the existing configuration to make these products work as an integrated solution.

Overview

Asterisk integrated with SIP Server can function in three different roles:

- As a PBX with a business call routing capability.

Asterisk is configured to send business calls to SIP Server to engage a Genesys routing solution. SIP Server uses the routing results to forward the call to the selected agent.

- As a voicemail server.

SIP Server uses Asterisk as a voicemail server. Unanswered calls are forwarded to Asterisk to record the voice messages. Contact center agents receive indication on their T-Library agent desktops about new voice messages waiting in their voicemail box. Agents can access and manage their voicemail boxes hosted on Asterisk.

- As a media server.

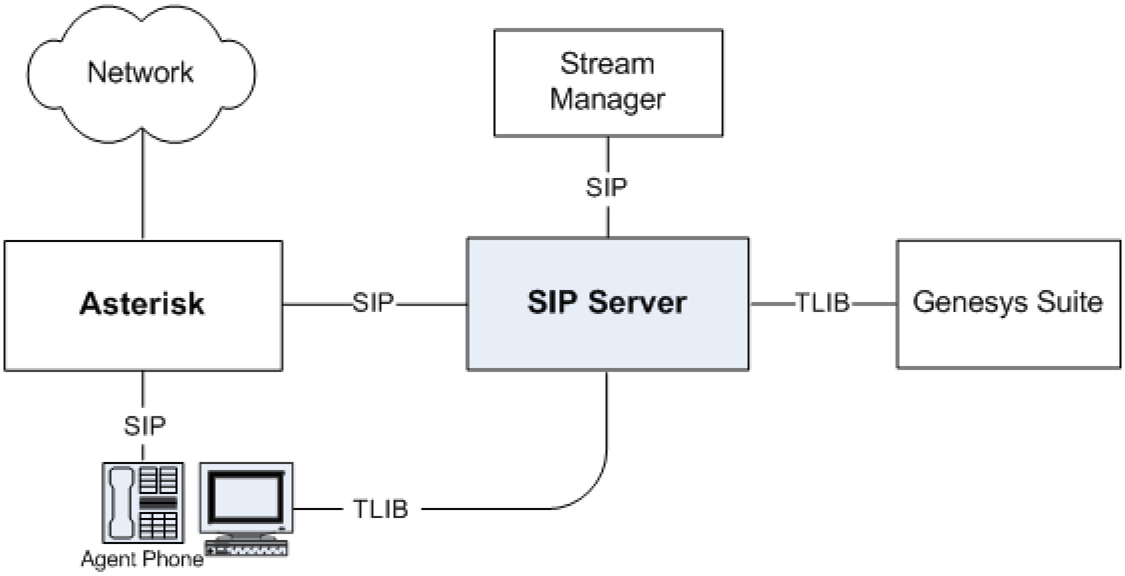
SIP Server uses Asterisk as a Media Server. Asterisk is engaged in the call to perform one of the following functions:

- Call recording
- Announcement or music playing
- DTMF digits collection
- Conferences

Asterisk with a Business Call Routing Capability

The **SIP Server - Asterisk Deployment Architecture** figure depicts a sample deployment architecture of SIP Server with Asterisk, in which:

- Asterisk is connected to the network via a SIP gateway.
- The agent endpoint is registered on Asterisk.
- The agent endpoint is associated with a T-Library desktop application.



SIP Server - Asterisk Deployment Architecture

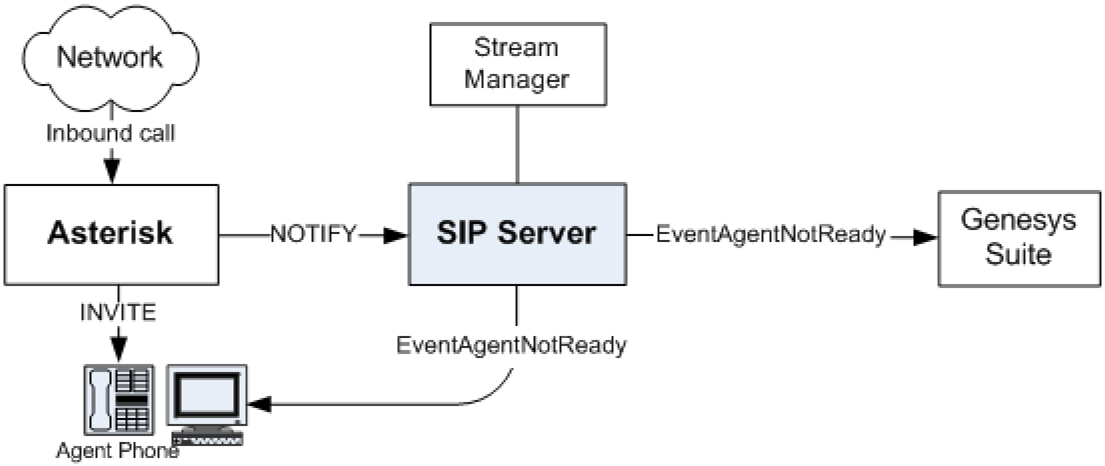
Integration with the Asterisk switch relies on the SIP presence subscription from SIP Server. For any call handled by the agent endpoint, Asterisk is requested to provide a notification about the status change for that endpoint. SIP Server uses those notifications to synchronize an agent state visible to all Genesys T-Library clients with the actual state of this agent. The business call routing solution that is built on these integration principles involves SIP Server to handle the business calls only. Private calls are processed locally on Asterisk. Agent statuses are reported to SIP Server for all call types, because they are used to identify the agents' availability for the Genesys Routing Solution.

All figures in this topic depicting Stream Manager refer to the Genesys Stream Manager. This component, when working together with SIP Server, provides different kinds of media services, such as ring-back, music-on-hold, DTMF digit collection, and others. You can also configure Asterisk to work as a media server for SIP Server. For information about architectural and configuration details of this solution, see Asterisk as a Media Server.

Private Calls

An Asterisk dialing plan can be set up in such a way that private calls (direct calls to an agent, for example) are not forwarded to SIP Server. Instead, only the notification about the busy status of the endpoint is passed to SIP Server. SIP Server uses this status change notification to set the endpoint DN to a busy state (`EventAgentNotReady`), so that the rest of the Genesys suite will not consider that DN available for the routing of contact center calls.

The following figure illustrates the processing of private calls.



Private Call Processing

Contact Center Calls

In the same way that you can set up an Asterisk dialing plan to bypass SIP Server for private calls, you can write rules so that Asterisk connects contact center calls (typically, calls to the service number of the company) to SIP Server. After that, SIP Server triggers a strategy for Universal Routing Server (URS) to process this type of call. Eventually, an agent DN is selected to handle the customer call and SIP Server initiates a new dialog to Asterisk for the selected endpoint. Finally, Asterisk delivers the call to the agent endpoint.

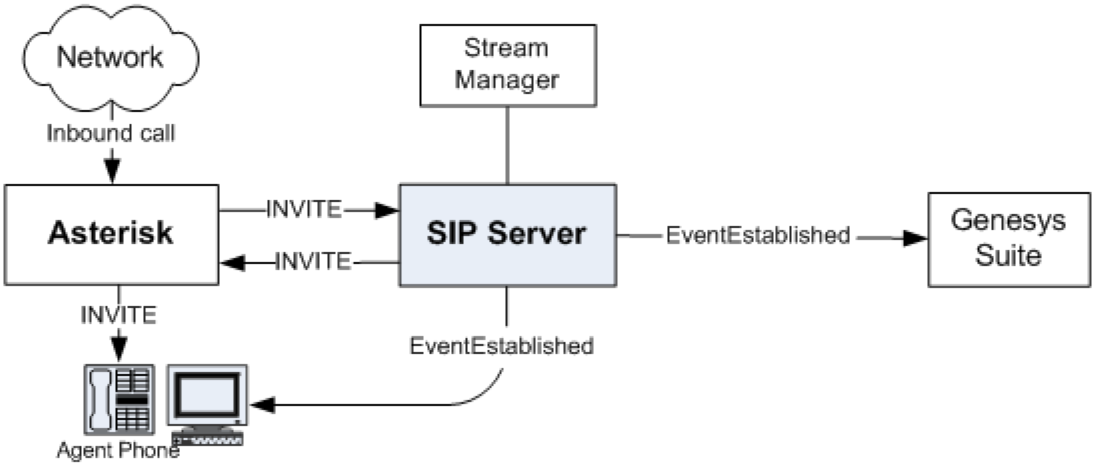
This mechanism creates a signaling loop inside SIP Server, which is then in charge of maintaining the inbound leg from Asterisk (customer leg) with the outbound leg to Asterisk (agent leg).

Note: From the Asterisk perspective, the two legs are two completely separate calls. Correlation is performed at the SIP Server level.

By staying in the signaling path, SIP Server detects any change in call status, and can therefore produce call-related events (`EventRinging`, `EventEstablished`, `EventReleased`, and so on).

Any call control operation from the agent must be performed using a third-party call control (3pcc) procedure. In other words, the agent desktop must be used for any call control operation (besides the answer call operation). This includes, but is not limited to, hold, transfer, and conference requests.

The following figure illustrates the processing of contact center calls.

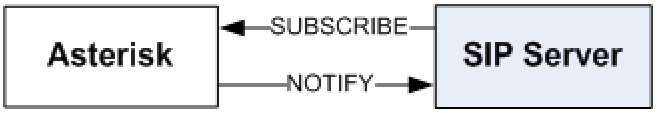


Contact Center Call Processing

Call Flows

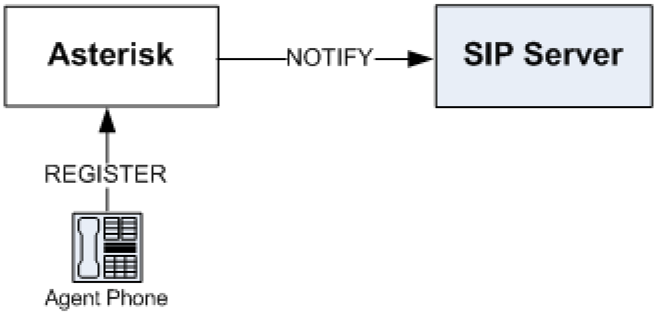
Subscription

At startup, SIP Server sends `SUBSCRIBE` messages to the Asterisk switch, which notifies about changes in the endpoints' status. The Asterisk switch sends `NOTIFY` messages to SIP Server to report the endpoints' status. See the following figure.



Presence Subscription from SIP Server

If an endpoint is not yet registered, the Asterisk switch reports its status as `closed.` As soon as the endpoint registers, Asterisk sends a `NOTIFY` message to SIP Server, reporting the status open. See [[Integration With Asterisk#.



Presence Notification to SIP Server

Private Calls

For private calls, the Asterisk dialing plan is set up in such a way that the call is sent directly to the endpoint. Asterisk notifies SIP Server about the call activity on that particular endpoint. In this case, SIP Server generates `EventAgentNotReady`, which reports the overall agent status as unavailable for contact center calls. (See the **Private Call Processing** figure.)

SIP Server generates only agent-related TEvents for the private Asterisk calls"for example, EventAgentReady and EventAgentNotReady. Call-related events"such as EventRinging, EventEstablished, and so on--are not generated for private calls, because SIP Server is not involved in the processing of private calls.

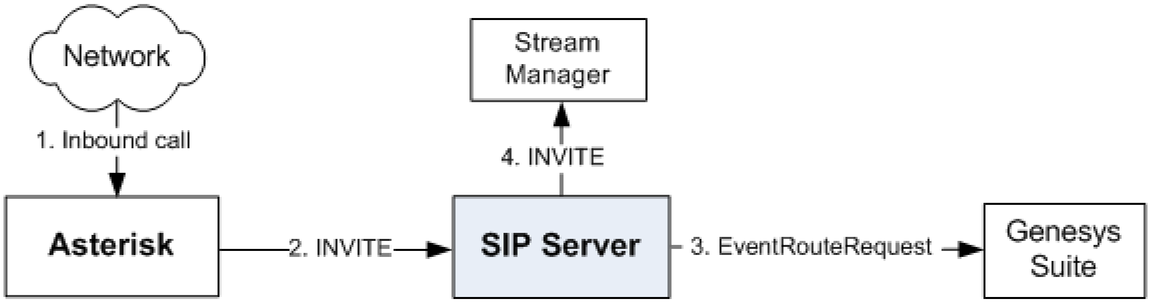
As soon as the call is released at the endpoint, Asterisk notifies SIP Server, which then generates an EventAgentReady message. The agent is then considered available for contact center calls.

	Note: The mechanism for private outbound call processing is exactly the same. SIP Server receives the NOTIFY messages sent by Asterisk.

Contact Center Calls

Inbound Calls to SIP Server

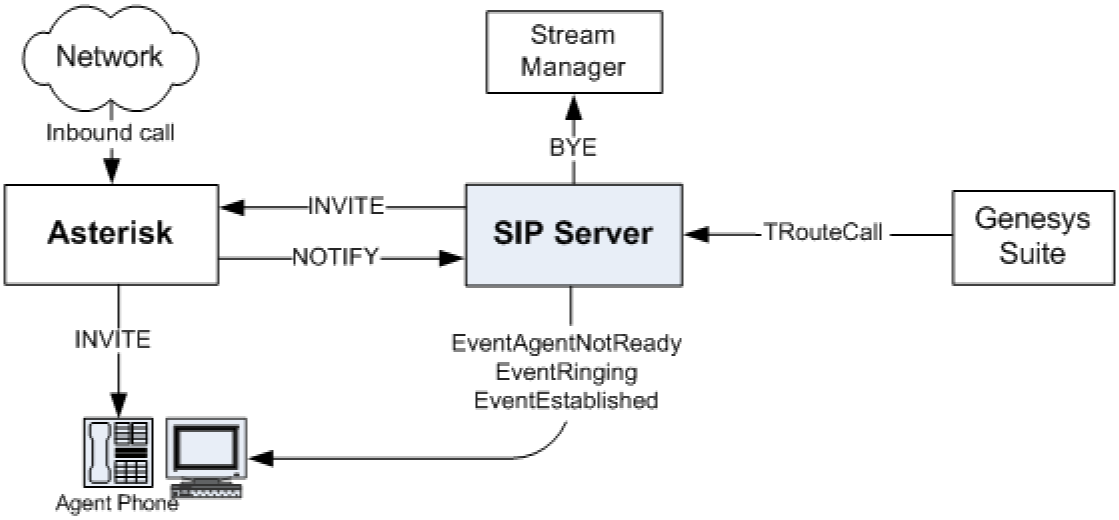
Inbound contact center calls are programmed within the Asterisk dialing plan to be directed to SIP Server. In this case, the call arrives at a Routing Point, and URS is triggered. You can request a call treatment (using the TApplyTreatment request) to play announcement or music. If Stream Manager is configured to provide a treatment functionality, SIP Server connects a caller to Stream Manager to listen to the treatment while waiting for an agent to become available. See the following figure.



Handling Contact Center Calls

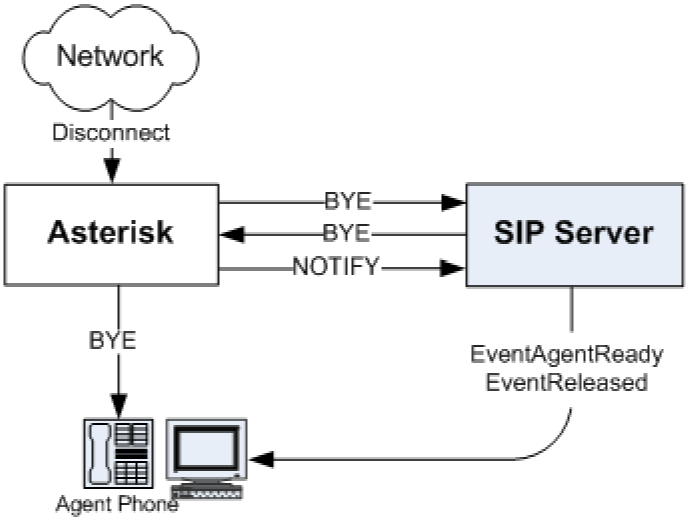
Whenever the agent becomes ready, SIP Server receives a TRouteCall request to the targeted agent endpoint. Because this endpoint is configured to point to Asterisk, SIP Server then initiates a new dialog with Asterisk to engage the agent. Asterisk forwards the call to the specified endpoint and reports to SIP Server the call activity on that endpoint with a NOTIFY message (EventAgentNotReady). When the call is answered, Stream Manager is disconnected, and the original SIP dialog is renegotiated between SIP Sever and Asterisk.

Because SIP Server is in the signaling path for contact center calls, it generates all call-related events (EventRinging, EventEstablished, and so on) for the agent's DN. See the following figure.



Delivering the Call to the Agent

Furthermore, when the call is released, SIP Server also generates `EventReleased`, and Asterisk notifies SIP Server with a `NOTIFY` message (`EventAgentReady`). See the following figure.



Contact Center Call Disconnection

Inbound Calls to Extensions

Inbound contact center calls, and manual internal first-party call control (1pcc) calls that are directed to extensions, are not visible to SIP Server; as a result, you cannot make third-party call control (3pcc) calls for them. Only inbound calls that are directed to Routing Points on SIP Server, and manual internal calls, which go via Routing Points can be seen by SIP Server; as a result, 3pcc calls can be made for them.

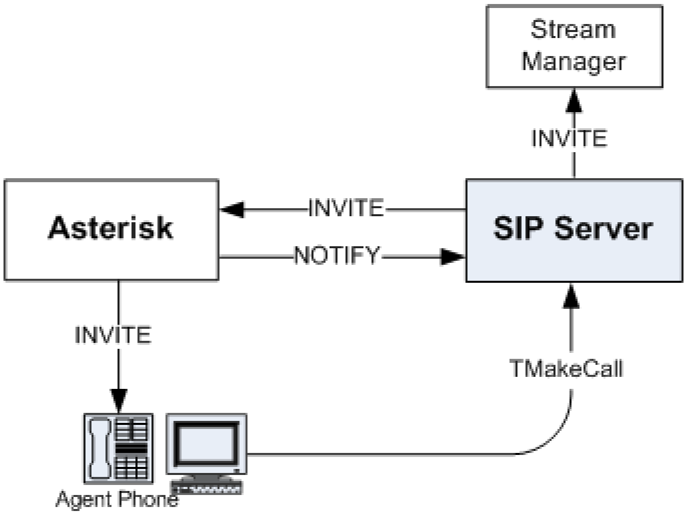
Outbound Calls

An outbound call that is contact-center-related (for example, a call back to a customer) must be performed using 3pcc operations. This ensures that SIP Server creates and controls the SIP dialogs on behalf of the agent endpoint. SIP Server uses the call flow 1 described in RFC 3725 to create a call initiated from the agent's T-Library client using the `TMakeCall` request.

An agent initiates the outbound call by sending the `TMakeCall` request from the T-Library client to SIP Server. SIP Server attempts to engage the agent by sending the `INVITE` message to this agent endpoint (via Asterisk).

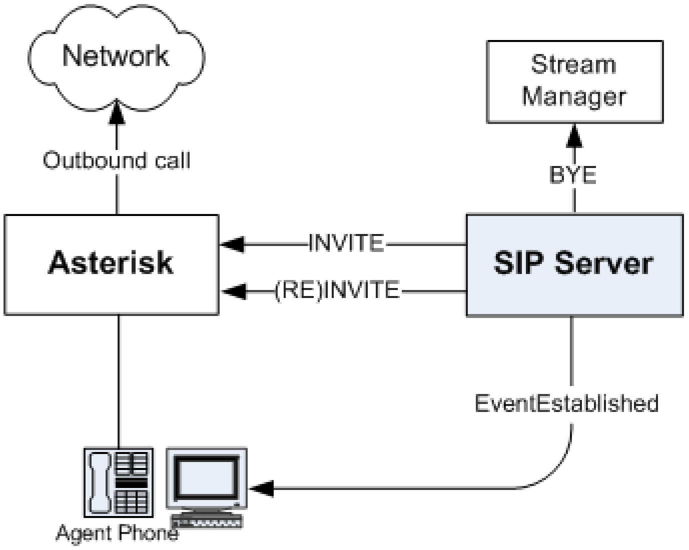
Note: If the phone is not configured with auto-answer, the agent must manually answer the call. This is the only manual action that is required for contact center calls.

If Stream Manager is configured to provide treatments, then SIP Server connects the agent to Stream Manager to listen to a ringback tone while establishing a connection to the outbound call destination. See the following figure.



Engaging the Agent Endpoint for an Outbound Call

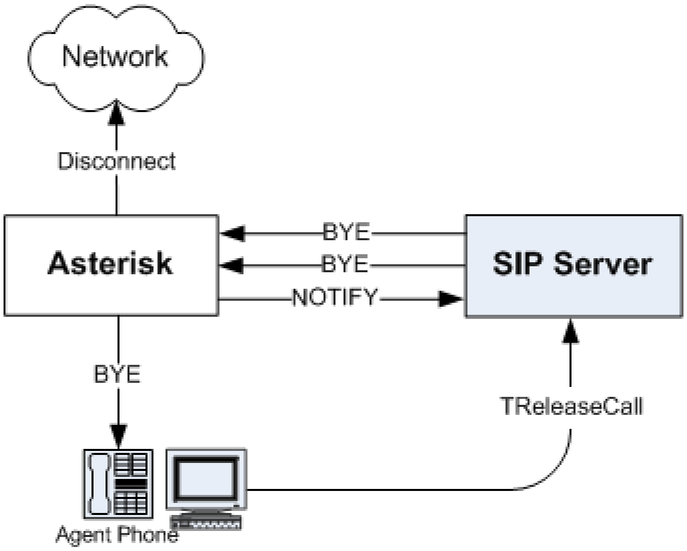
SIP Server contacts the requested destination number. After the destination answers the call, SIP Server discontinues the ringback tone (by sending the BYE message to Stream Manager) and renegotiates with the agent endpoint (via Asterisk), so that the media stream is connected between the agent and the customer. See the following figure.



Connecting to the Customer

Although disconnection would work if it were initiated directly from the agent endpoint, it is good practice to always use a desktop application to perform any actions related to contact center calls. Therefore, the disconnection is requested by sending the `TReleaseCall` request to SIP Server.

SIP Server manages two dialogs: one for the agent and another for the customer. It sends the BYE message to both of them, and the call is eventually disconnected. See the following figure.



Outbound Call Disconnection

Asterisk as a Voicemail Server

Asterisk can provide the voicemail server functionality. A stand-alone Asterisk solution allows all agents registered on Asterisk to use multiple voicemail boxes. SIP Server integration with Asterisk adds several new voice-mail-related features to the standard Asterisk set:

1. Agents registered on SIP Server (an agent VOIP phone sends the SIP `REGISTER` message to SIP Server) can use voicemail boxes hosted on Asterisk.
2. All agents (registered on Asterisk or on SIP Server) can receive voicemail notifications on their T-Library client desktops.
3. Voicemail boxes can be associated with extensions, agent logins, and agent groups.

Voicemail Boxes For Agents Registered on SIP Server

One or multiple voicemail boxes can be created on Asterisk for the agents registered on SIP Server. All voicemail features configured on Asterisk become available for SIP Server agents. Unanswered calls can be forwarded to the corresponding voicemail box allowing callers to leave a voice message. SIP Server agents can call their voicemail boxes from their VOIP phones to listen to the voice messages and to manage the voicemail box.

Voicemail Notifications Sent to SIP Server T-Library Clients

Genesys contact center agents use T-Library client desktops. If Asterisk is configured as a voicemail server for SIP Server, agents can receive notifications about the new voice messages left in their voicemail boxes on their T-Library client desktops. These notifications also provide information about the number of old and new messages stored in the voicemail box.

Voicemail Boxes Associated with Extensions, Agent Logins, or Agent Groups

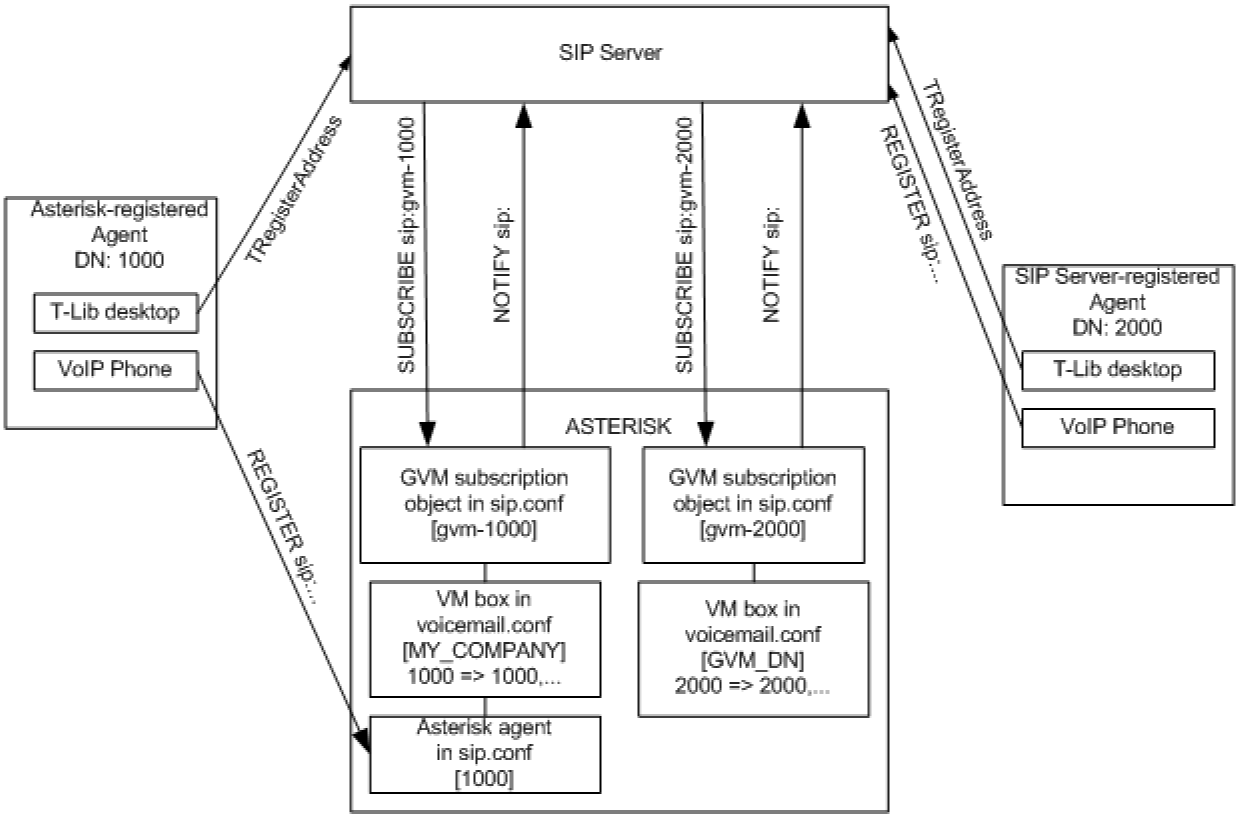
SIP Server associates each voicemail box it controls on Asterisk with one of the following configuration objects in the Configuration Layer: `Extension`, `Agent Login`, or `Agent Group`. The voicemail box associated with a corresponding object defines a group of SIP Server T-Library clients to receive voicemail status notifications for a particular voicemail box. Voicemail notifications described in this section are transmitted using the T-Library interface. SIP Server sends messages to its T-Library clients.

If the voicemail box is associated with an extension, then notifications are sent to an agent whose T-Library client is registered to this extension. If the voicemail box is associated with the agent login, then SIP Server sends voicemail notifications to this agent T-Library client. In this case, it does not matter what DN this agent used to log in.

It is also possible to associate a voicemail box with the agent group. If a new voice message is left in such a voicemail box, then all logged in agents associated with this agent group will receive a notification about this message.

Call flows

The following figure illustrates a general integration schema representing Asterisk configured as a voicemail server for SIP Server.


SIP Server
TRegisterAddress
SUBSCRIBE sip:gvm-1000
NOTIFY sip:
SUBSCRIBE sip:gvm-2000
NOTIFY sip:
TRegisterAddress
REGISTER sip:...
Asterisk-registered Agent DN: 1000
T-Lib desktop
VoIP Phone
REGISTER sip:...
ASTERISK
GVM subscription object in sip.conf [gvm-1000]
GVM subscription object in sip.conf [gvm-2000]
VM box in voicemail.conf [MY_COMPANY] 1000 => 1000,...
VM box in voicemail.conf [GVM_DN] 2000 => 2000,...
Asterisk agent in sip.conf [1000]
SIP Server-registered Agent DN: 2000
T-Lib desktop
VoIP Phone

Asterisk Configuration as a Voicemail Server

The figure also shows how voicemail services can be provided for two agents: Agent DN 1000 and Agent DN 2000. Both agents use T-Library desktops connected to SIP Server via the T-Library protocol. Agent DN 1000 has the VOIP phone that is registered on Asterisk. Agent DN 2000 has the VOIP phone that is registered on SIP Server.

Asterisk is configured to fully support all calls made from and to DN 1000. For this purpose, it has a SIP entity `[1000]` configured in the `sip.conf` file to represent the agent's phone. It also has a voicemail box configured in some private context `[MY_COMPANY]` in the Asterisk `voicemail.conf` configuration file.

SIP Server integration with Asterisk requires adding a new object to the Asterisk configuration to provide the voicemail functionality for the SIP Server agent at DN 2000. A new voicemail box for this agent is created in the `[GVM_DN]` context of the Asterisk `voicemail.conf` configuration file.

The Asterisk Message Waiting Indicator (MWI) interface is used to integrate Asterisk as a voicemail server with SIP Server. The MWI interface utilizes the SIP subscription schema. SIP Server subscribes to the `message-summary` event at Asterisk using the SIP `SUBSCRIBE` request method:

```
SUBSCRIBE sip:gvm-1000@192.168.0.300 SIP/2.0

From: sip:gvm-1000@192.168.0.300;tag=7C217D88

To: sip:gvm-1000@192.168.0.300;tag=as050e992c

Call-ID: 1CD815F7-1@192.168.0.300
```

```
CSeq: 1103 SUBSCRIBE
Content-Length: 0
Via: SIP/2.0/UDP 192.168.0.200:5060;branch=z9hG4bK3B
Event: message-summary
Accept: application/simple-message-summary
Contact: <sip:gsipmwi@192.168.0.200:5060;mb=1000;dn=1000;tp=1>
Expires: 600
```

Asterisk sends notifications to SIP Server about the voicemail box status using the SIP `NOTIFY` message:

```
NOTIFY sip:gsipmwi@192.168.0.200:5060;mb=1000;dn=1000;tp=2 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.200:5070;branch=z9hG4bK219f391e
From: "asterisk" <sip:asterisk@192.168.0.200:5070>;tag=as13d3077a
To: <sip:gsipmwi@192.168.0.200:5060;mb=1000;dn=1000;tp=2>
Contact: <sip:asterisk@192.168.0.200:5070>
Call-ID: 1CD815F7-1@192.168.0.300
CSeq: 102 NOTIFY
User-Agent: Asterisk PBX
Event: message-summary
Content-Type: application/simple-message-summary
Content-Length: 43
Messages-Waiting: yes
Voice-Message: 1/0
```

SIP Server generates the `EventUserEvent` message based on this notification and sends it to the T-Library client registered on a DN associated with a particular voicemail box. This is an example of such a T-Library event:

```
EventUserEvent
AttributeUserData [120] 00 01 03 00..
'gsipmwi'(list) 'Mailbox 1000'
```

'Messages-Waiting' 'true'

'Voice-Message' '1/0'

'NewMessages " 1

"OldMessages " 0

```
AttributeUserEvent [1001]
```

```
AttributeThisDN '1000'
```

Dedicated SIP objects are created in the `sip.conf` Asterisk configuration file to support the MWI subscription. These objects are `gvm-1000` and `gvm-2000` in the **Asterisk Configuration as a Voicemail Server** figure. The GVM acronym in the object name stands for Genesys Voicemail. These objects are created in Asterisk for MWI subscription purposes only, and no SIP clients are registered on these objects. Both objects have a parameter pointing to a specific Asterisk voicemail box:

```
[gvm-1000]

mailbox=1000@MY_COMPANY

[gvm-2000]

mailbox=2000@GVM_DN
```

SIP Server activates one SIP subscription per voicemail box it needs to monitor. The above configuration guarantees that SIP Server will receive notification on a correct voicemail box when it subscribes to a corresponding GVM object.

MWI Subscription Scope

SIP Server activates one or multiple MWI subscriptions for each voicemail box it needs to monitor. Individual voicemail boxes created for Extensions or Agent Logins are monitored by a single MWI subscription per box. The number of MWI subscriptions activated per Agent Group voicemail box is equal to the number of agents currently logged in to this Agent Group.

SIP Server is designed in the assumption that all extensions have voicemail boxes. So, if MWI monitoring is enabled for the extensions (`mwi-extension-enable` is set to `true`), SIP Server at start up attempts to activate MWI subscriptions for all extensions configured in the Configuration Layer. Subscriptions for the Extension-related voicemail boxes are deactivated when SIP Server shuts down.

MWI subscription for Agent Login is when an agent with the corresponding agent ID logs in to SIP Server. SIP Server keeps this subscription active while the agent is logged in and stops it when the agent logs out.

The same MWI subscription logic is applied to the monitoring of voicemail boxes created for the Agent Groups. SIP Server activates MWI subscription for the group when the first agent associated with this group logs in. SIP Server stops the subscription when the last agent of this group logs out.

If, for some reason, a subscription request for any voicemail box type is rejected or times out, SIP Server attempts to activate this subscription again in one minute.

Building a Voicemail Solution

The Voicemail functionality in SIP Server and Asterisk allows you to build multiple Voicemail solutions with different complexity to address different business needs. This section provides examples that show how to build Voicemail solutions. It outlines general architectural ideas that refer to some configuration options only for clarification purposes. For configuration procedures, see the Framework 8.1 *SIP Server Deployment Guide*.

The easiest approach to a Voicemail solution is to have calls, which are not answered on a DN during a specified timeout, forwarded to the voicemail box associated with this DN (extension). This solution requires that you associate an Asterisk-hosted voicemail box with the DN. A `DN` object in Configuration Manager should be configured with the following options:

- `no-answer-overflow`
- `no-answer-timeout`

The `no-answer-timeout` option specifies the time during which the call must be answered. When the `no-answer-timeout` timer expires and the call is not answered, SIP Server uses the value of option `no-answer-overflow` to decide how to process the call. If this option contains the name of the voicemail box associated with this DN, then SIP Server sends the call to this voicemail box.

A similar solution can be configured for agents. SIP Server can apply the same algorithm that is used for process unanswered calls for an agent who ignores the DN where the agent logs in. In this case, the Asterisk-hosted voicemail box should be associated with the Agent Login (and not the extension). Also, the `no-answer-timeout` and `no-answer-overflow` options should be specified in the `Agent Login` configuration object.

SIP Server also allows you to use voicemail boxes in business call routing. Usually in those scenarios, calls are controlled by the URS strategy, which attempts to find an appropriate agent to forward the call to. There are many ways to write a URS strategy to utilize a Voicemail solution. For example, if a call is routed to an agent group that does not have any currently available agents, URS can send a call to the voicemail box associated with the Agent Group. In this case, all logged in members of this group will receive a notification about the new message left in the group voicemail box.

SIP Server can also redirect unanswered calls to the voicemail box based on the options configured for the SIP Server `Application` configuration object. There are two groups of options, which define how SIP Server processes unanswered calls for extensions and for agents:

- `extn-no-answer-XXX`
- `agent-no-answer-XXX`

See the Framework 8.1 *SIP Server Deployment Guide* for more information about the options.

Asterisk as a Media Server

You can configure Asterisk as a media server for SIP Server. SIP Server can utilize the following services provided by Asterisk:

- Play announcements.

- Collect DTMF digits.
- Organize conferences.
- Recording calls.

Communication between two servers is mainly based on RFC 4240--an exception is the recording service, which is not described in this RFC.

Asterisk for Business Calls Routing

Integration Task Summary

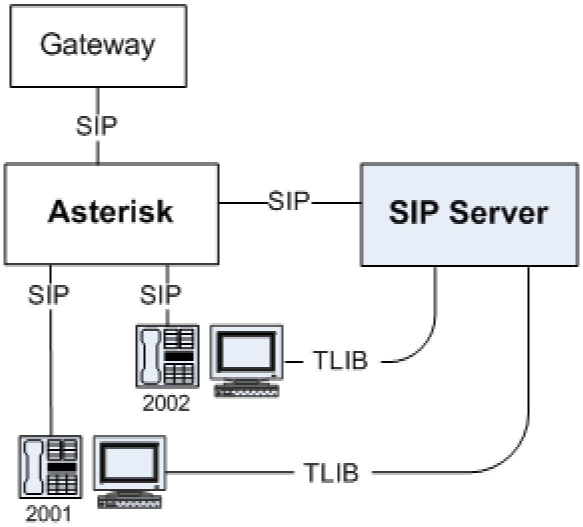
The following table summarizes the steps to integrate SIP Server with Asterisk to support business calls routing.

Objective	Related Procedures and Actions
1. Configure Asterisk to support business call routing.	See Configuring Asterisk.
2. Configure DNs for the Asterisk `Switch` object in the Configuration Layer.	See Configuring DN Objects.

Configuring Asterisk

This section describes the procedures for configuring Asterisk in the following environment:

- Asterisk is connected to the network via a SIP gateway.
- Two SIP endpoints, 2001 and 2002, are registered on Asterisk.
- Each endpoint is associated with a T-Library desktop application.



Asterisk Sample Configuration

The following table provides an overview of the main steps to integrate SIP Server with Asterisk.

Objective	Related Procedures and Actions
1. Confirm that Asterisk is functional and handling calls appropriately.	The procedures in this topic assume that Asterisk is functional and handling calls appropriately. For more information, see Asterisk documentation.
2. Configure the `sip.conf` file.	See Configuring the sip.conf file.
3. Configure the `extensions.conf` file.	See Configuring the extensions.conf file.

Configuring the sip.conf file

Purpose

To configure the `sip.conf` file for Asterisk.

Start

1. Configure two peers, one describing the gateway access, and the other describing SIP Server access—for example: `[gwsim]` `type=peer` `host=10.0.0.1` `port=5066` `context=default` `canreinvite=no` `[gsip]` `type=peer` `username=gsip` `host=10.0.0.1` `context=default` `canreinvite=no`
2. Configure the endpoints. The user name of the endpoint must match the `Extension` DN configured on the SIP Server side"for example: `[2001]` `type=friend` `username=2001` `host=dynamic` `context=default` `notifyringing=yes` `canreinvite=no` `[2002]` `type=friend` `username=2002` `host=dynamic` `context=default` `notifyringing=yes` `canreinvite=no`
   Note: SIP Server does not support receiving authentication challenges. For this reason, Asterisk users must not be configured with the `secret` option; otherwise, Asterisk would challenge INVITE messages that SIP Server issues on behalf of the user, and SIP Server would fail to respond to the challenge.
3. When you are finished, save your configuration.

End

Configuring the extensions.conf file

Purpose

To configure the `extensions.conf` file for Asterisk.

Start

1. For each endpoint that SIP Server monitors, configure a *hint* entry to ensure that Asterisk will accept a presence subscription (from SIP Server, in this case) for those endpoints—for example:
```
exten => 2001,hint,SIP/2001
exten => 2001,1,Dial(SIP/2001,60)
exten => 2002,hint,SIP/2002
exten => 2002,1,Dial(SIP/2002,60)
```

2. Configure a basic dialing plan for contact center calls.
   In this example, extension 2400 is used as a company's service number, so all business calls should arrive to this extension. Those calls are routed to SIP Server. If a call is not answered within 30 seconds, it will be dropped. The "r" flag tells Asterisk to generate a ringback tone for the caller while the call is being routed.
   ```
   ; Inbound call to routing point 2400 -> contact SIP Server
   exten => 2400,1,Dial(SIP/${EXTEN}@gsip,30,r)
   exten => 2400,2,Hangup()
   ```
3. Configure a basic dialing plan for calls to external numbers"for example:
   ```
   ; Any number with prefix "0' -> contact gateway (with remaining digits only)
   exten => _0.,1,Dial(SIP/${EXTEN:1}@gwsim,60)
   ```
4. When you are finished, save your configuration.

End

Configuring DN Objects

The following table provides an overview of the main steps to configure different DNs under the Asterisk Switch object in the Configuration Layer.

Objective	Related Procedures and Actions
1. Configure a Trunk DN.	See Configuring a Trunk DN.
2. Configure an Extension DN.	See Configuring Extension DNs.

If you integrate SIP Server with Asterisk in order to support the business routing capability, you do not need to set any configuration options in the SIP Server Application object. Instead, you configure DNs for the Asterisk Switch object that is assigned to the appropriate SIP Server.

Configuring a Trunk DN

Purpose

To configure a DN of type Trunk to support the presence SUBSCRIBE/NOTIFY functionality and to configure external access through Asterisk.

Start

1. Under a configured Switch object, select the DNs folder. From the File menu, select New > DN to create a new DN object.
2. In the New DN Properties dialog box, click the General tab, and then specify the following properties (see the following figure):
   a. Number: Enter a name for the Trunk DN. This name can be any unique value, and it can be a combination of letters and numbers.

   b. Type: Select Trunk from the drop-down box.

3. Click the Annex tab.
4. Create a section named TServer. In the TServer section, create options as specified in the following table (see also the following figure):

Option Name	Option Value	Description
contact	SIP URI	The contact URI to which SIP Server sends the SUBSCRIBE message.
subscribe-presence-domain	A string	The subscription domain information for the Trunk DN. This option value will be used with the DN name to form the SUBSCRIBE request URI and the To: header.
subscribe-presence-expire	Any positive integer	The subscription renewal interval (in seconds).
subscribe-presence-from	SIP URI	The subscription endpoint information. This option value will be used to form the From: header in the SUBSCRIBE request.
prefix	Any positive integer	The initial digits of the number used to direct to Asterisk any call that SIP Server does not recognize as an internal DN.
refer-enabled	false	Set this option to false for SIP Server to use a re-INVITE request method when contacting Asterisk.

5. When you are finished, click Apply.

End

Configuring Extension DNs

Purpose

To configure Asterisk endpoints that SIP Server will monitor and control.

Start

1. Under a configured Switch object, select the DNs folder. From the File menu, select New > DN to create a new DN object.
2. In the New DN Properties dialog box, click the General tab, and then specify the following properties:
   a. Number: Enter a name for the Extension DN. In general, this should be the phone number of the extension. You must not use the @ symbol or a computer name.

b. `Type:` Select `Extension` from the drop-down box.

3. Click the `Annex` tab.

4. Create a section named `TServer`. In the `TServer` section, create options as specified in the following table:

Option Name	Option Value	Description
contact	SIP URI	The contact URI to which SIP Server sends the `SUBSCRIBE` message.
dual-dialog-enabled	`false`	Set this option to `false` so that consultation calls are handled using the same SIP dialog that is sent to Asterisk.
make-call-rfc3725-flow	1	Set this option to `1`, so that 3pcc call flow will be used according to RFC3725.
refer-enabled	`false`	Set this option to `false` if you are using the RFC3725 flow.
sip-hold-rfc3264	`false`	Set this option to `false` so that RTP stream hold is performed in a manner compliant with RFC2543.
subscribe-presence	A string	The name of the `Trunk` DN that is configured for the presence subscription messages to be sent to Asterisk.

5. When you are finished, click `Apply`.

End

Configuring Asterisk

The Genesys Voicemail Adapter (GVMA) utility is provided by Genesys to propagate the Voicemail configuration from the Configuration Layer to the Asterisk configuration files. GVMA performs the following steps:

1. GVMA starts.
2. GVMA connects to Configuration Server using the SOAP protocol.
3. GVMA makes a backup copy of the Asterisk configuration.
4. GVMA loads the Voicemail configuration from the following configuration objects:
    - `DNs`
    - `Agent Logins`
    - `Agent Groups`
5. GVMA updates Asterisk configuration files with the information retrieved from the Configuration Layer during Step 4.
6. GVMA instructs Asterisk to reload configuration files.
7. GVMA exits.

GVMA can be run manually or scheduled for periodic execution using the OS scheduling tools, such as cron on Linux systems.

The following table provides an overview of the main steps to integrate SIP Server with Asterisk to support the Voicemail solution.

Configuring Asterisk

Objective	Related Procedures and Actions
1. Define all required parameters in the GVMA configuration file.	See the following sections: • Prerequisites • GVMA Location • Configure the GVMA Configuration File
2.Run the GVMA utility on the Asterisk host to configure Asterisk.	Run the GVMA utility by executing the `gvma_asterisk76.pl` script.

Prerequisites

Back Up the Asterisk Configuration

The GVMA utility modifies the following Asterisk configuration files: `extensions.conf, sip.conf`, and `voicemail.conf`. To save the original Asterisk configuration, create backup copies of all Asterisk configuration files before using the GVMA utility.

Perl Interpreter

You must install the Perl interpreter on the Asterisk host to run the GVMA utility, which is written as a perl script. Install these additional perl packages that are required to run GVMA:

- SOAP-Lite
- Net-Telnet

Enable the Asterisk Manager Interface

Enable the Asterisk Manager Interface (AMI) by setting the following parameters in the `manager.conf` Asterisk configuration file:

```
[general]

enabled = yes

port = 5038

bindaddr = 0.0.0.0
```

Enable the GVMA Utility to Change the Asterisk Configuration

Enable the GVMA utility to change the Asterisk configuration by adding the following section in the `manager.conf` Asterisk configuration file:

```
[gvma]

secret = genesys1

deny=0.0.0.0/0.0.0.0

permit=127.0.0.1/255.255.255.0

read = system,call,log,verbose,command,agent,user

write = system,call,log,verbose,command,agent,user
```

GVMA Location

The GVMA utility is located in the `tools` folder of the SIP Server installation utility. Files in the tools directory include:

- `gvma_asterisk76.cfg`--The GVMA utility for 7.6 SIP Server.
- `gvma_asterisk76.pl`--The GVMA utility configuration file for 7.6 SIP Server.
- `gvma_asterisk.cfg`--The GVMA utility for 7.5 SIP Server.
- `gvma_asterisk.pl`--The GVMA utility configuration file for 7.5 SIP Server.

Depending on the `mwi-mode` option value set in the SIP Server `Application` object, you choose which configuration file and script to run. If the `mwi-mode` option is set to `SUBSCRIBE`, use the following files:

- `gvma_asterisk76.cfg`
- `gvma_asterisk76.pl`

If the `mwi-mode` option is set to `REGISTER`, use the following files:

- `gvma_asterisk.cfg`
- `gvma_asterisk.pl`

The `REGISTER` value of the `mwi-mode` option is for backward compatibility with 7.5 releases of SIP Server.

Configure the GVMA Configuration File

Configure the following sections in the GVMA configuration file before using the utility:

- `cfgserver`
- `gvma_settings`

Section *cfgserver*

Parameters in the `cfgserver` section define how GVMA connects to Configuration Manager and what information GVMA reads from it.

Note that option `port` refers to the SOAP port of Configuration Server and not to the port where Configuration Manager is connected. The Configuration Server SOAP port is specified in the Configuration Server configuration file as a `port` option in the `[soap]` section.

```
[cfgserver]

host=<config server hostname or IP>

port=<config server SOAP port>
```

```
username = <config server username>

password = <config server password>
```

The second part of the `cfgserver` section provides several examples about how to define a query to allow for the GVMA utility to collect information about `DNs`, `Agent Logins`, and `Agent Groups` from the Configuration Layer. One query should be chosen for each of these three object types. The following placeholders in the selected queries should be replaced with the information from the Configuration Layer:

- `<Switch DBID>`
- `<tenant DBID>`
- `<tenant name>`
- `<Switch Name>`

```
#Query examples using DBIDs:
#dnquery = CfgDN[(@ownerDBID=<Switch DBID>) and (@type=1)]
#agentquery = CfgAgentLogin[@ownerDBID=<Switch DBID>]
#agentgroupquery = CfgAgentGroup[@tenantDBID=<tenant DBID>]

#Query examples using switch and tenant names:
dnquery = CfgTenant[@name='<tenant Name>']/switches/CfgSwitch[@name='<swith
name>']/DNs/CfgDN[@type='1']
agentquery = CfgSwitch[@name='<Switch name>']/agentLogins/CfgAgentLogin
agentgroupquery = CfgTenant[@name='<tenant name>']/agentGroups/CfgAgentGroup
```

Section *gvma_settings*

The first group of parameters in the `gvma_settings` section specifies the location of Asterisk configuration files and what files you have to change:

- `asterisk_cfg_path=/etc/asterisk`
- `asterisk_cfg_file_sip=sip.conf`
- `asterisk_cfg_file_vm=voicemail.conf`
- `asterisk_cfg_file_exten=extensions.conf`

The following parameters define the comments, which GVMA puts as a boundaries around the parts it inserts into the Asterisk configuration files.

- `asterisk_cfg_gvma_begin=;$—-GVMA-BEGIN-GVMA—-$`
- `asterisk_cfg_gvma_end=;$—-GVMA-END-GVMA—-$`

GVMA creates backup copies of the configuration files to be modified in the location defined by the `backup_path` parameter:

- `backup_path=./gvma_backup`

GVMA uses the Asterisk Manager Interface port to connect to Asterisk:

- asterisk_cm_port=5038

On the Asterisk side, this port is defined in the manager.conf file.

Use the siptserver_host and siptserver_port parameters to specify the host and port, respectively, in the GVM subscription objects created in the sip.conf file.

- siptserver_host=<SIP Server hostname or IP>
- siptserver_port=<SIP Server Port>

Finally, the gvma_settings section has a group of parameters specifying how to access different types of voicemail boxes from the agent VOIP phones:

- vm_dn_ext_prefix=37
- vm_agt_ext_prefix=38
- vm_grp_ext_prefix=39
- vm_voicemail_main_ext=9500

GVMA Modifications to Asterisk Configuration Files

You can easily find all modifications the GVMA utility makes to the Asterisk configuration files by searching for the beginning and end key specified in the GVMA configuration file in the parameters asterisk_cfg_gvma_begin and asterisk_cfg_gvma_end.

File extensions.conf

GVMA creates a new context called [GVMA] in the Asterisk dialing plan. This context includes six wildcards. The following wildcard is created to provide access to the agent voicemail boxes from the agent VOIP phones:

```
exten => _37X.,1,Wait(1)

exten => _37X.,2,Set(GVM_DEST=${EXTEN:2})

exten => _37X.,3,GotoIf($["${CALLERID(num)}" = "${GVM_DEST}"]?4:6)

exten => _37X.,4,VoicemailMain(${GVM_DEST}@GVMA_DN)

exten => _37X.,5,Hangup

exten => _37X.,6,GotoIf($["${GVM_DEST}" = "9500"]?7:9)

exten => _37X.,7,VoicemailMain(@GVMA_DN)

exten => _37X.,8,Hangup

exten => _37X.,9,Voicemail(${GVM_DEST}@GVMA_DN,u)
exten => _37X.,10,Hangup
```

Three wildcards of this type are created to provide access to three different types of voicemail boxes: `Extensions`, Agent Logins, and `Agent Groups`. Prefixes used in these wildcards are taken from the following GVMA configuration file parameters:

- `vm_dn_ext_prefix`
- `vm_agt_ext_prefix`
- `vm_grp_ext_prefix`

Another three wildcards that are created in the GVMA context are:

- `_gvm-X`
- `_gvm-a-X`
- `_gvm-g-X`

These wildcards are not supposed to be dialed directly, but they are required for the MWI subscription to function properly.

Note: You must manually include a new GVMA context into the existing dialing plan context that is used to process agent calls on Asterisk. If there is no special context created for this purpose, you must include the GVMA context into the default dialing plan context. Include the following parameters: `[default]include => GVMA`

File *sip.conf*

The GVMA utility creates a block of new GVM SIP entities in the `sip.conf` file. Each SIP entity is associated with one voicemail box. SIP Server activates one MWI subscription for each GVM SIP entity.

```
;$—-GVMA-BEGIN-GVMA—-$

; Generated by Genesys VoiceMail Configuration Adapter for Asterisk.

; Content generated at Tue Jan 15 20:36:50 2008

[gvm-1111]

type=friend

host=192.168.0.200

port=5060

mailbox=1111@GVMA_DN

vmexten=1111

...
```

```
;$—-GVMA-END-GVMA—-$
```

The GVMA utility creates multiple `gvm-*` objects in the `sip.conf` configuration file. If Asterisk is also integrated with SIP Server to perform a business call routing, then the `sip.conf` file also contains an object representing a SIP Server. The `host` and `port` parameters specified for the SIP Server object are the same as the ones defined for the `gvm-*` entities in the `sip.conf` file. This configuration can cause a problem if the Asterisk dialing plan uses the `host:port` format in the `Dial()` function to send calls to SIP Server. For example:

```
SIP-SERVER_HOST = 10.10.10.1

SIP-SERVER_PORT = 5060

exten => 2400,1,Dial(SIP/${EXTEN}@${SIP-SERVER_HOST}:${SIP-SERVER_PORT},30,r)
```

Asterisk can select any `gvm-*` object to send calls, instead of the SIP Server object. In this case, a call is delivered to the correct destination but the call processing depends on the `sip.conf` object parameters, which are different for SIP Server and `gvm-*` objects.

To avoid this problem, Genesys recommends using the dial plan `Dial()` function with reference to the object name defined in the `sip.conf` file instead of using the `host:port` format. For example:

```
extensions.conf:

exten => 2400,1,Dial(SIP/${EXTEN}@genesys-sip-server,30,r)

sip.conf:

[genesys-sip-server]

host=10.10.10.1

port=10.10.10.1
```

File *voicemail.conf*

The GVMA utility creates three new Voicemail contexts in the `voicemail.conf` Asterisk configuration file: `GVMA_DN`, `GVMA_AGENT` and `GVMA_AGENTGROUP`. Those contexts contain voicemail boxes created for `Extensions`, `Agent Logins`, and `Agent Groups`, respectively. GVMA takes all parameters that are specified for the GVM voicemail boxes from the configuration of the corresponding the Configuration Layer objects.

```
;$—-GVMA-BEGIN-GVMA—-$

; Generated by Genesys VoiceMail Configuration Adapter for Asterisk.

; Content generated at Tue Jan 15 20:36:50 2008

; ######## Voicemail Boxes for the Extensions #######

[GVMA_DN]

1111 => 1111,1111,,
; ######## Voicemail Boxes for the Agents #######
[GVMA_AGENT]
2222 => 2222, 2222, 2222@192.168.0.200, 2222@192.168.0.200,operator=yes
; ######## Voicemail Boxes for the Agent Groups #######
[GVMA_AGENTGROUP]
3333 => 3333, 3333, 3333@192.168.0.200, 3333@192.168.0.200,operator=yes
;$—-GVMA-END-GVMA—-$
```

Asterisk as a Media Server

In order for Asterisk to work as a media server integrated with SIP Server, you must enhanced the Asterisk dialing plan with several Genesys macros and global variables as described in this section.

Configuring Asterisk

Dialing Plan Global Variables

You must add the following list of global variables to the `[globals]` section of the Asterisk dialing plan.

```
SIP_PREFIX=.*sip:.*@.*:[0-9]+.*
DIG_PRMT_REGEX=silence/1?[0-9]
FIND_CLT_REGEX=${SIP_PREFIX}play=[ ]*(music/collect).*
FIND_PLY_REGEX=${SIP_PREFIX}play=[ ]*([^>\;]*)[>\;].*
FIND_REP_REGEX=${SIP_PREFIX}repeat=[ ]*([^>\;]*)[>\;].*
FIND_REC_REGEX=${SIP_PREFIX}record=[ ]*([^>\;]*)[>\;].*
FIND_COF_REGEX=.*sip:conf=(.*)@.*:[0-9]+.*
DEFAULT_FILE_TO_PLAY= /var/lib/asterisk/moh/fpm-calm-river
```

Variable `DEFAULT_FILE_TO_PLAY` points to the default music file that is played for the Genesys treatments. In the example, above it refers to the voice file, which comes with Asterisk (if Asterisk is installed in the standard directory). You can change this reference to any other file in the actual deployment.

Dialing Plan Macro to Perform Genesys Treatments

You must add this treatment to the Asterisk dialing plan to perform Genesys treatments.

```
[macro-treatment]
;
; ${ARG1} - SIP_HEADER(To)
;
; IF treatment == CollectDigits
;
exten => s, 1, Answer
exten => s, 2, Set(collect=$["${ARG1}":"${FIND_PLY_REGEX}"])
exten => s, 3, GotoIf($[$["${collect}"="music/collect"] | $["${collect}"="music/
silence"]] ? 15 : 20)
exten => s, 15, macro(get-digits,${collect})
exten => s, 16, Goto(s,99)
;
; ELSE IF treatment == record
;
exten => s, 20, Set(rec_file=$["${ARG1}":"${FIND_REC_REGEX}"])
exten => s, 21, Set(ply_file=$["${ARG1}":"${FIND_PLY_REGEX}"])
exten => s, 22, GotoIf($[${LEN(${rec_file})} != 0] ? 30 : 40)
;
; Recording Treatment
exten => s, 30, GotoIf($[${LEN(${ply_file})} = 0] ? 32 : 31)
exten => s, 31, Playback(${ply_file}) ;
exten => s, 32, Record(genesys-rec-${rec_file}.wav) ;can't detect|report dtmf
exten => s, 33, Goto(s,98)
;
; ELSE
; Play treatment
exten => s, 40, GotoIf($[${LEN(${ply_file})} = 0] ? 41 : 43)
exten => s, 41, Set(ply_file=${DEFAULT_FILE_TO_PLAY})
exten => s, 42, Goto(s,44)
exten => s, 43, Set(ply_count=$["${ARG1}":"${FIND_REP_REGEX}"])
exten => s, 44, GotoIf($[$[${LEN(${ply_count})} = 0] | $["$ply_count" = "forever"]]?
50 : 60)
; Playback forever
exten => s, 50, Playback(${ply_file})
exten => s, 51, GotoIf($[${PLAYBACKSTATUS}=FAILED] ? 52 : 50) ;Goto(s, 50)
exten => s, 52, Goto(s, 99)
; Counted playback
; here probably possible to use background()
exten => s, 60, Playback(${ply_file}) ; Playback
exten => s, 61, Set(ply_count=$[${ply_count} - 1])
exten => s, 62, GotoIf($[$[${ply_count} > 0] & $[${PLAYBACKSTATUS} = SUCCESS]] ? 61 :
98)

exten => s, 98, Hangup
exten => s, 99, NoOp(end-withot-hagup)
```

Dialing Plan Macro to Collect DTMF Digits

You must add this treatment to the Asterisk dialing plan to collect DTMF digits. Replace <COLLECT-MESSAGE-PLACEHOLDER> in the macros below with the name of the file to play to announce digit collection.

```
[macro-get-digits]
exten => s,1, GotoIf($[$[${ARG1}=music/collect] | $[${ARG1}=music/silence]] ? 2 : 3)
exten => s,2, Set(ARG1=silence/2)
exten => s,3,Read(dncdigits,<COLLECT-MESSAGE-PLACEHOLDER>,1,s)
exten => s,4,SendText(Signal=${dncdigits})
exten => s,5, Goto(macro-get-digits,s,3)
```

Dialing Plan Macro to Create a Conference

You must add this treatment to the Asterisk dialing plan to organize a conference using the Asterisk MeetMe application.

```
[macro-conf]
exten => s, 1, Set(conf_id=$["${ARG1}":"${FIND_COF_REGEX}"])
exten => s, 2, NoOp(${ARG1})
exten => s, 3, GotoIf($[${LEN(${conf_id})} != 0] ? 4 : 20)
exten => s, 4, Set(rec_file=$["${ARG1}":"${FIND_REC_REGEX}"])
exten => s, 5, GotoIf($[${LEN(${rec_file})} != 0] ? 6 : 8)
exten => s, 6, MeetMe(${conf_id},drq)
exten => s, 7, Goto(s,20)
exten => s, 8, MeetMe(${conf_id},dq)
exten => s, 20, NoOp()
```

Integrating Genesys Macros into the Dialing Plan

The Asterisk dialing plan all macros provided above. This section suggests one possible way to do that. Add the following macro in the dialing plan:

```
[moh_conf_treatment]
include => macro-treatment
exten => annc, 1, macro(treatment,${SIP_HEADER(To)})
exten => _co[n]f=., 1, macro(conf,${SIP_HEADER(To)})
```

You must include this macro into the context used to process agent calls. If there is no special context created for this purpose, you must include macro into the default dialing plan context.

```
[default]
include => moh_conf_treatment
```

Media Files

Media files used for the Genesys treatments should be placed into the standard Asterisk sounds directory. The default location of this directory is:

`/var/lib/asterisk/sounds`

Call recordings created by Asterisk are also stored in this directory. There are two types of recordings, which can be activated by SIP Server:

- Regular (proxy mode)
- Emergency

By default, names of the recordings made in `regular` mode are prefixed with `genesys-rec.` Names of the emergency recordings start with the `meetme-conf-rec` prefix. In both cases, the name prefix is followed by a conference ID.

Configuring DN Objects

SIP Server utilizes media services through the DNs of type `Voice over IP Service` configured under the `Switch` object. The `Voice over IP Service` DNs have a `service-type` configuration option, which defines the kind of service this DN can provide. SIP Server selects an appropriate DN when the client application requests a media service.

When you use Asterisk as a media server for SIP Server, you should configure the `Voice over IP Service` DNs with the following `service-type` values in the SIP Server `Switch` object:

- `mcu`
- `treatment`

- `recorder`
- `music`

For information about configuring DNs for different types of services, see the "SIP Device Configuration" topic of the Framework 8.1 SIP Server Deployment Guide.

Cisco Media Gateway

This section describes how to integrate SIP Server with the Cisco Media Gateway Controller (MGC). It contains the following sections:

- Overview
- Configuring Cisco Media Gateway
- Configuring Cisco Media Gateway DN Objects

Overview

The SIP Server and Cisco Media Gateway integration solution described in this topic is not the only method that will work. Although there are other methods, this is the only one that has been tested and approved by Genesys, and that is supported by Genesys Customer Support.

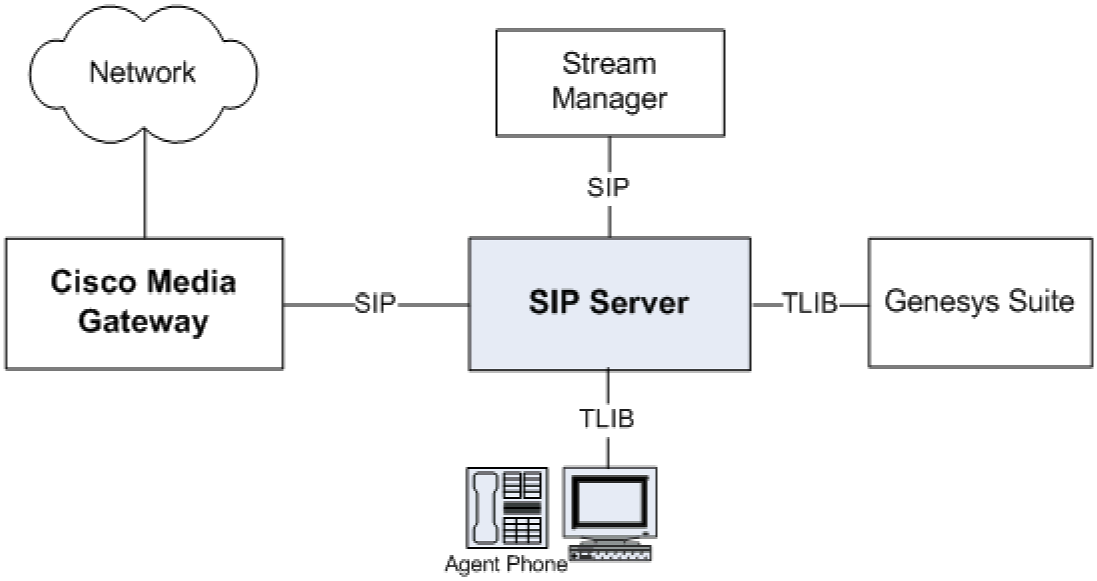
The following Cisco IOS Software versions were tested:

- 2800 Series
- 3700 Series
- 3800 Series
- 5300 Series
- 5400 Series

For confirmation of the supported Cisco IOS Software versions, contact Genesys Technical Support. For more information about Cisco IOS Software, go to the Cisco web site at http://www.cisco.com/.

Deployment Architecture

The following figure depicts a sample deployment architecture of SIP Server with Cisco Media Gateway.



SIP Server - Cisco Media Gateway Deployment Architecture

Integration Task Summary

To integrate SIP Server with Cisco Media Gateway, complete the following procedures:

1. Configure Cisco Media Gateway.

2. Configure a Trunk DN for Cisco Media Gateway.

Configuring Cisco Media Gateway

This page provides an overview of the main steps that are required in order to configure Cisco Media Gateway.

Integrating with Cisco Media Gateway

1. Check Prerequisites.

Verify that Cisco Media Gateway is working

Verify that Cisco Media Gateway is functional and handling calls appropriately.

The procedures in this topic assume that Cisco Media Gateway is functional and handling calls appropriately. For more information, see Cisco Media Gateway-specific documentation.

2. Configure an E1 environment.

Configuring an E1 environment

Purpose

To configure an E1 environment. This section provides an example of an E1 configuration.

Start

1. Configure a controller:
   ```
   controller E1 0/2/0
   framing NO-CRC4
   ds0-group 0 timeslots 1 type fxo-loop-start
   ds0-group 1 timeslots 2 type fxo-loop-start
   ds0-group 2 timeslots 3 type fxo-loop-start
   ```
2. Configure voice ports:
   ```
   voice-port 0/2/0:0
   output attenuation 0
   station-id name 2300090
   voice-port 0/2/0:1
   output attenuation 0
   station-id name 2300091
   ```

```
voice-port 0/2/0:2
output attenuation 0
station-id name 2300092
```

3. Configure dial peers:

```
dial-peer voice 2300090 pots
destination-pattern 6...
supplementary-service pass-through
port 0/2/0:0
forward-digits all
dial-peer voice 2300091 pots
destination-pattern 6...
supplementary-service pass-through
port 0/2/0:1
forward-digits all
dial-peer voice 2300092 pots
destination-pattern 6...
supplementary-service pass-through
port 0/2/0:2
forward-digits all
dial-peer voice 8800 voip
service session
destination-pattern 8800
voice-class codec 4
session protocol sipv2
session target ipv4:192.168.50.137
dtmf-relay rtp-nte
supplementary-service pass-through
```

End

Next Steps

- Configuring a T1 CAS environment

3. Configure a T1 CAS environment.

Configuring a T1 CAS environment

Purpose

To configure a T1 CAS environment. This section provides an example of a T1 CAS configuration.

Start

1. Configure a controller:

```
controller T1 1/0/1
framing sf
clock source internal
```

```
linecode ami
ds0-group 0 timeslots 1 type e&m-immediate-start
ds0-group 1 timeslots 2 type e&m-immediate-start
ds0-group 2 timeslots 3 type e&m-immediate-start
```

2. Configure voice ports:

```
voice-port 0/2/0:0
output attenuation 0
station-id name 2300090
voice-port 0/2/0:1
output attenuation 0
station-id name 2300091
voice-port 0/2/0:2
output attenuation 0
station-id name 2300092
```

3. Configure dial peers:

```
dial-peer voice 2300090 pots
destination-pattern 6...
supplementary-service pass-through
port 0/2/0:0
forward-digits all
dial-peer voice 2300091 pots
destination-pattern 6...
supplementary-service pass-through
port 0/2/0:1
forward-digits all
dial-peer voice 2300092 pots
destination-pattern 6...
supplementary-service pass-through
port 0/2/0:2
forward-digits all
dial-peer voice 8800 voip
service session
destination-pattern 8800
voice-class codec 4
session protocol sipv2
session target ipv4:192.168.50.137
dtmf-relay rtp-nte
supplementary-service pass-through
```

End

Next Steps

- Configuring a T1 PRI environment

4. Configure a T1 PRI environment.

Configuring a T1 PRI environment

Purpose

To configure a T1 PRI environment. This section provides an example of a T1 PRI configuration.

Start

1. Configure a controller:

   ```
   controller T1 0/0/0
   framing esf
   linecode b8zs
   pri-group timeslots 1-24
   ```

2. Configure an interface serial:

   ```
   interface Serial0/0/0:23
   no ip address
   encapsulation hdlc
   isdn switch-type primary-ni
   isdn incoming-voice voice
   no cdp enable
   ```

3. Configure a voice port:

   ```
   voice-port 0/0/0:23
   ```

4. Configuring dial peers:

   ```
   dial-peer voice 9 pots
   destination-pattern 9T
   incoming called-number 9...
   port 0/0/0:23
   dial-peer voice 8800 voip
   service session
   destination-pattern 8800
   voice-class codec 4
   session protocol sipv2
   session target ipv4:192.168.50.137
   dtmf-relay rtp-nte
   supplementary-service pass-through
   ```

End

Next Steps

- Configuring an E1 PRI environment

5. Configure an E1 PRI environment.

Configuring an E1 PRI environment

Purpose

To configure an E1 PRI environment. This section provides an example of an E1 PRI configuration.

Start

1. Configure a controller:
   ```
   controller E1 0/2/1
   framing NO-CRC4
   pri-group timeslots 1-31
   ```
2. Configure an interface serial:
   ```
   interface Serial0/2/1:15
   no ip address
   encapsulation hdlc
   isdn switch-type primary-net5
   isdn protocol-emulate network
   isdn incoming-voice voice
   no cdp enable
   ```
3. Configure a voice port:
   ```
   voice-port 0/2/1:15
   ```
4. Configure dial peers:
   ```
   dial-peer voice 130 pots
   destination-pattern 130T
   direct-inward-dial
   port 0/2/1:15
   dial-peer voice 8800 voip
   service session
   destination-pattern 8800
   voice-class codec 4
   session protocol sipv2
   session target ipv4:192.168.50.137
   dtmf-relay rtp-nte
   supplementary-service pass-through
   ```

End

Next Steps

- Configuring a SIP User Agent

6. Configure a SIP User Agent.

Configuring a SIP User Agent

Purpose

To configure a SIP User Agent. This section provides an example of a SIP User Agent configuration.

Start

Configure a SIP User Agent: enter global configuration "configure terminal":

```
sip-ua
timers notify 400
sip-server dns:host.genesyslab.com
```

End

Configuring DN Objects

Configure a Trunk DN for Cisco Media Gateway under the Switch object associated with SIP Server in the Configuration Layer.

Configuring a Trunk DN for Cisco Media Gateway

Start

1. Under a configured Switch object, select the DNs folder. From the File menu, select New > DN to create a new DN object.
2. In the New DN Properties dialog box, click the General tab, and then specify the following properties (see the following figure):
   a. Number: Enter the gateway name.
   b. Type: Select Trunk from the drop-down box.
3. Click the Annex tab.
4. Create a section named TServer. In the TServer section, create options as specified in the following table.

Option Name	Option Value	Description
contact	<ipaddress>:<SIP port>	The contact URI that SIP Server uses for communication with the gateway, where <ipaddress> is the IP address of the gateway and <SIP port> is the SIP port number of the gateway.
oos-check	0-300	How often (in seconds) SIP Server checks a DN for out-of-service status.
oos-force	0-30	How long (in seconds) SIP Server waits before placing a DN out of service.
prefix	Any numerical string	The initial digits of the number that SIP Server matches to determine whether this trunk should be used for outbound calls. For example, if prefix is set to 78, dialing a number starting with 78 will cause SIP Server to consider this trunk a gateway or SIP proxy. If multiple Trunk objects match the prefix, SIP Server will select the one with the longest prefix that

Option Name	Option Value	Description
		matches.
priority	Any non-negative integer	The gateway priority that SIP Server uses to decide a route. A smaller number designates higher priority. If more than one gateway with the same prefix is selected, the gateway with highest priority is normally selected. This `priority` option is used to control primary-backup gateway switchover, and to provide lowest-cost routing.
refer-enabled	`false`	Set this option to `false` for SIP Server to use a `re-INVITE` request method when contacting the gateway. This is the only method supported in the Cisco Media Gateway configuration.
recovery-timeout	`0-86400`	The length of time that a device is set to out-of-service in case of an error.
replace-prefix	Any numerical string	The digits that replace the prefix in the DN. For example, if `prefix` is set to 78, and `replace-prefix` is set to 8, the number 786505551212 will be replaced with 86505551212 before it is sent to the gateway or SIP proxy (here, Cisco Media Gateway).

5. When you are finished, click `Apply`.

End

AudioCodes Gateway

This topic describes how to integrate SIP Server with the AudioCodes Gateway. It contains the following sections:

- Overview
- Configuring the AudioCodes Gateway
- Configuring AudioCodes Gateway DN Objects

Note: The instructions in this topic assume that the AudioCodes Gateway is fully functional and connected to the corresponding PBX.

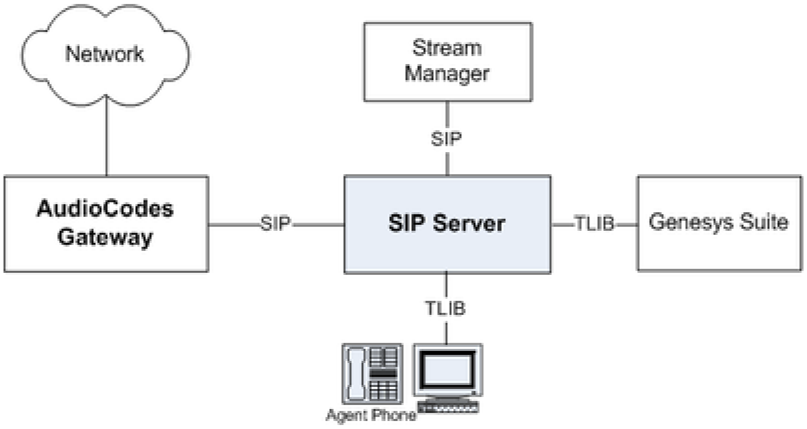
Overview

The SIP Server and AudioCodes integration solution described in this topic is not the only method that will work. Although there are other methods, this is the only one that has been tested and approved by Genesys, and that is supported by Genesys Customer Support.

In the configuration example, the AudioCodes IPMedia 2000 Gateway is used. The same configuration procedures are also applicable to the AudioCodes Mediant 2000 and the TP (or TrunkPack) gateways.

Deployment Architecture

The following figure depicts a sample deployment architecture of SIP Server with the AudioCodes Gateway.



Integration Task Summary

To integrate SIP Server with the AudioCodes Gateway, complete the following procedures:

1. Configure the AudioCodes Gateway.
2. Configure a Trunk DN for the AudioCodes Gateway.

Configuring the AudioCodes Gateway

The following table provides an overview of the main steps that are required in order to configure the AudioCodes Gateway.

Objective	Related Procedures and Actions
1. Confirm that AudioCodes Gateway is functional and handling calls appropriately.	The procedures in this topic assume that AudioCodes Gateway is functional and handling calls appropriately. For more information, see AudioCodes Gateway-specific documentation.
2. Configure the AudioCodes Gateway.	Procedure: Configuring the AudioCodes Gateway

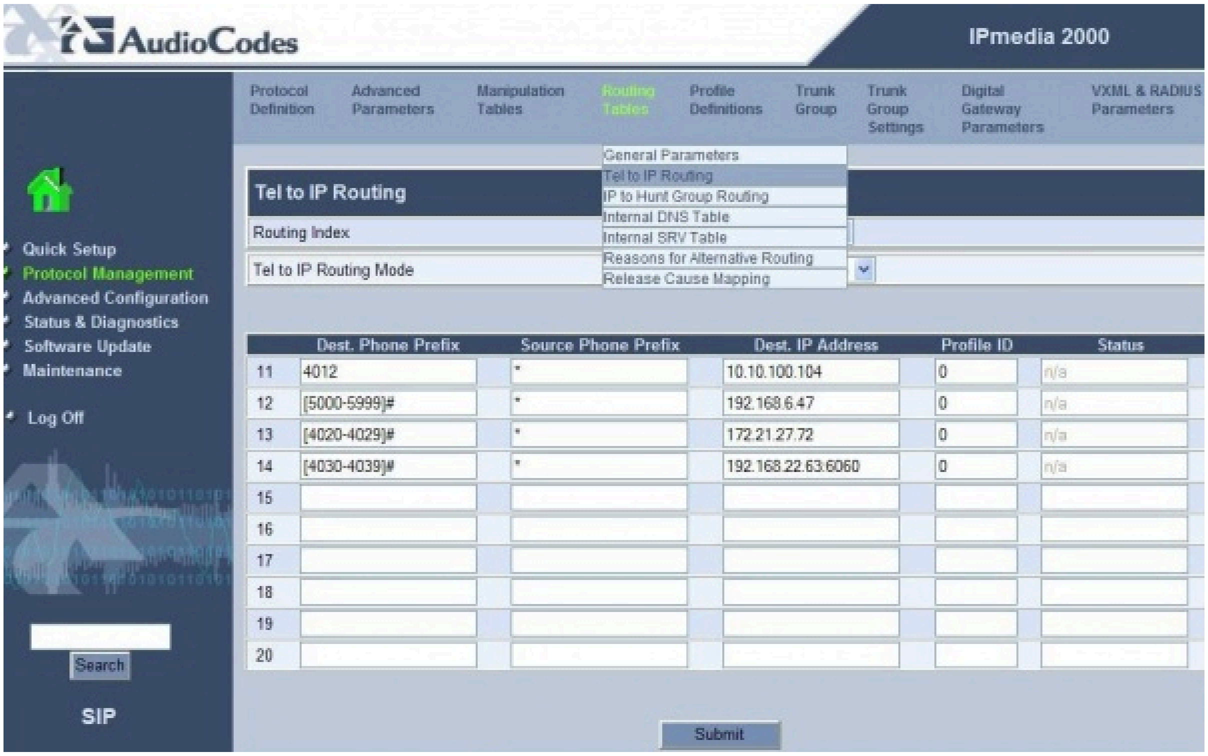
Configuring the AudioCodes Gateway

Purpose

- To configure the AudioCodes Gateway to support integration with SIP Server.

Start

1. Log in to the AudioCodes web administrative interface (see the following figure).



Tel to IP Routing

Routing Index

Tel to IP Routing Mode

	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Profile ID	Status
11	4012	*	10.10.100.104	0	n/a
12	[5000-5999]#	*	192.168.6.47	0	n/a
13	[4020-4029]#	*	172.21.27.72	0	n/a
14	[4030-4039]#	*	192.168.22.63:6060	0	n/a
15					
16					
17					
18					
19					
20					

Configuring the AudioCodes Gateway: Sample Configuration

2. From the left pane menu, select `Protocol Management`.
3. Navigate to the `Routing Tables` tab, and select `Tel to IP Routing` from the drop-down menu.
4. In the `Dest. Phone Prefix` text box, enter the DNs that you will be routing through the gateway.
5. In the `Source Phone Prefix` text box, enter an asterisk (*) to accept any source phone number.
6. In the `Dest. IP Address` text box, enter the SIP Server IP address and port. Note that port is only required if other than default port `5060` is used.
   In the example configuration (see the previous figure), line 14 demonstrates that the range of DNs `4030` through `4039` is passed through the AudioCodes Gateway to SIP Server at the address `192.168.22.63, port 6060`.

End

Configuring DN Objects

Configure a Trunk DN for AudioCodes Gateway under the Switch object associated with SIP Server in the Configuration Layer.

Configuring a Trunk DN for the AudioCodes Gateway

Start

1. Under a configured Switch object, select the DNs folder. From the File menu, select New > DN to create a new DN object.
2. In the New DN Properties dialog box, click the General tab, and then specify the following properties (see the following figure):
    a. Number: Enter the gateway name.
    b. Type: Select Trunk from the drop-down box.

- Click the Annex tab.
- Create a section named TServer. In the TServer section, create options as specified in the following table.

Option Name	Option Value	Description
contact	<ipaddress>:<SIP port>	The contact URI that SIP Server uses for communication with the gateway, where <ipaddress> is the IP address of the gateway and <SIP port> is the SIP port number of the gateway.
oos-check	0-300	How often (in seconds) SIP Server checks a DN for out-of-service status.
oos-force	0-30	The length of time (in seconds) that SIP Server waits before placing a DN out-of-service.
prefix	Any numerical string	The initial digits of the number that SIP Server matches to determine whether this trunk should be used for outbound calls. For example, if prefix is set to 78, dialing a number starting with 78 will cause SIP Server to consider this trunk a gateway or SIP proxy. If multiple Trunk objects match the prefix, SIP Server will select the one with the longest prefix that matches.
priority	Any non-negative integer	The gateway priority that SIP Server uses to decide a route. A smaller number

Option Name	Option Value	Description
		designates higher priority. If more than one gateway with the same prefix is selected, the gateway with highest priority is normally selected. This `priority` option is used to control primary-backup gateway switchover, and to provide lowest-cost routing.
refer-enabled	`true, false`	Specifies whether the REFER method is sent to an endpoint. When set to `false`, SIP Server uses the `re-INVITE` method instead.
recovery-timeout	`0"86400`	The length of time that a device is set to out-of-service in case of an error.
replace-prefix	Any numerical string	The digits that replace the prefix in the DN. For example, if `prefix` is set to 78, and `replace-prefix` is set to 8, the number 786505551212 will be replaced with 86505551212 before it is sent to the gateway or SIP proxy (here, AudioCodes Gateway).

- When you are finished, click `Apply`.

End

F5 Networks BIG-IP LTM

This document describes how to integrate SIP Server with the F5 Networks BIG-IP Local Traffic Manager (hereafter referred to as *BIG-IP LTM*) to support SIP Server `hot standby` high-availability (HA) mode. It contains the following sections:

- Overview
- Configuring SIP Server HA
- Configuring BIG-IP LTM

Note: The instructions in this document assume that BIG-IP LTM is fully functional. They also assume that Genesys SIP Server has already been installed and configured to function properly.

Overview

The SIP Server and BIG-IP LTM integration solution described in this topic enables you to preserve SIP sessions between SIP Server and other SIP-enabled devices that are involved in contact center operations, in switchover scenarios.

In this integration solution, one Virtual Server configured on the BIG-IP LTM is associated with a single IP address (referred to as *Virtual IP address*), and it represents one HA pair of SIP Servers configured as members of one server pool that is associated with the Virtual Server. It is possible to have more than one HA pair running behind a single BIG-IP LTM. This requires configuring additional Virtual Servers and server pools for each HA pair in the way that the one unique Virtual IP address is used for each HA pair.

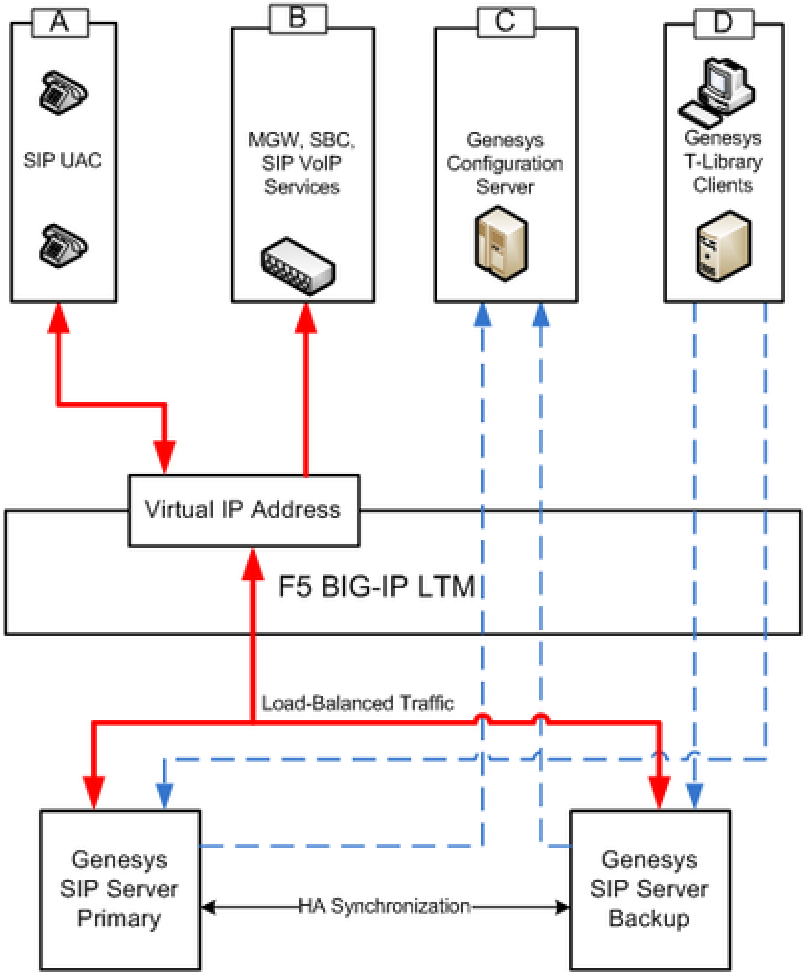
Integration Solution Notes

- Up-front load balancing via Network SIP Server or other device could be implemented, but is not described in this topic.
- BIG-IP LTM supports an active/hot-standby HA mode itself; configuration of the LTM in HA mode is not described in this topic and has not been validated with SIP Server.
- Either UDP or TCP can be used as the transport for SIP signaling. Use of TLS for encrypted SIP signaling has not been validated, and configuration of TLS is not described in this topic.
- BIG-IP LTM can be configured in a more complex load-balancing role. This is beyond the scope of this topic.

Deployment Architecture

The following figure depicts a sample deployment architecture of primary and backup SIP Servers with the BIG-IP LTM, in which:

- BIG-IP LTM is positioned as a network router between a SIP Server HA pair and other network entities.
- Hosts where SIP Servers are running use the BIG-IP LTM as the default gateway.
- BIG-IP LTM is configured to apply SNAT (Secure Network Address Translation) to all outbound packets, with the exception of destinations that are defined in the SNAT exclusion group.



Deployment Requirements

There are four different communication groups of devices that interact with SIP Server (see the preceding figure). Each group has its own requirements that must be considered when configuring the BIG-IP LTM.

SIP Phones Group

The SIP Phones group (group A in the preceding figure) includes SIP phones that are used by agents.

Initially, devices of this group use the `REGISTER` method to notify SIP Server of the current `Contact` URI (IP address). SIP Server uses the `Contact` information for further communication with the device.

By default, SIP Server uses the UDP to communicate with devices of the group. Devices send requests to and receive responses from the BIG-IP LTM Virtual IP address.

This group requires that:

- Any inbound packets received at the BIG-IP LTM Virtual IP address are directed to the primary SIP Server.

- SNAT is applied to any outbound packets that are sent to devices of the group, which means that a source IP address of the outbound packet is translated from a SIP Server physical IP address to the BIG-IP LTM Virtual IP address.

SIP Service Devices Group

The SIP Service Devices group (group B in the preceding figure) includes media gateways, softswitches, Session Border Controllers (SBC), and SIP-based VoIP Service devices such as Genesys Stream Manager. These devices do not register with SIP Server; their contact information is known in advance and it remains consistent.

By default, SIP Server uses the UDP to communicate with devices of the group. Devices receive requests from the BIG-IP LTM Virtual IP address.

This group requires that:

- Any inbound packets received at the BIG-IP LTM Virtual IP address are directed to the primary SIP Server.
- SNAT is applied to any outbound packets that are sent to devices of the group.

Genesys Configuration Server

SIP Server maintains permanent TCP/IP connection with Genesys Configuration Server (group C in the preceding figure). Requests to Configuration Server are sent from a SIP Server physical IP address. Responses from Configuration Server are directed to the SIP Server physical IP address.

This group requires that:

- No SNAT is applied to outbound packets sent to Configuration Server.
- The primary or backup SIP Server is accessible via its physical IP address.

Genesys T-Library Clients Group

All Genesys T-Library clients (group D in the preceding figure) that implement Genesys T-Library functionality maintain permanent TCP/IP connection with SIP Server. Devices send requests to and receive responses from a SIP Server (primary or backup) physical IP address.

This group requires that:

- No SNAT is applied to outbound packets sent to devices of the group.
- The primary or backup SIP Server is accessible via its physical IP address.

Note: In this deployment architecture, the HA synchronization traffic between primary and backup SIP Servers does not pass through the BIG-IP LTM, that is why it is excluded from applying SNAT.

Integration Task Summary

The following table summarizes the steps that are required in order to integrate SIP Server with the BIG-IP LTM.

Integrating SIP Server with BIG-IP LTM

Objective	Related procedures and actions
1. Configure the BIG-IP LTM.	See Configuring the BIG-IP LTM.
2. Configure SIP Server HA.	See Configuring SIP Server Applications.

Configuring the BIG-IP LTM

The following table provides an overview of the main steps that are required in order to configure the BIG-IP LTM. Complete all steps in the order in which they are listed.

Integrating with BIG-IP LTM

1. Check Prerequisites.

Verify that BIG-IP LTM is working

The procedures in this topic assume that the BIG-IP LTM is properly licensed and fully functional, with login and password access configured. For more information, see BIG-IP LTM specific documentation.

2. Configure VLANs.

Configuring VLANs

Purpose

To configure two VLANs (Virtual Local Area Networks): one VLAN for the external interface (physical interface 1.3) and one VLAN for the internal (SIP Server side) interface (physical interface 1.1). VLANs are used to logically associate Self IP interfaces with physical interfaces on the BIG-IP LTM.

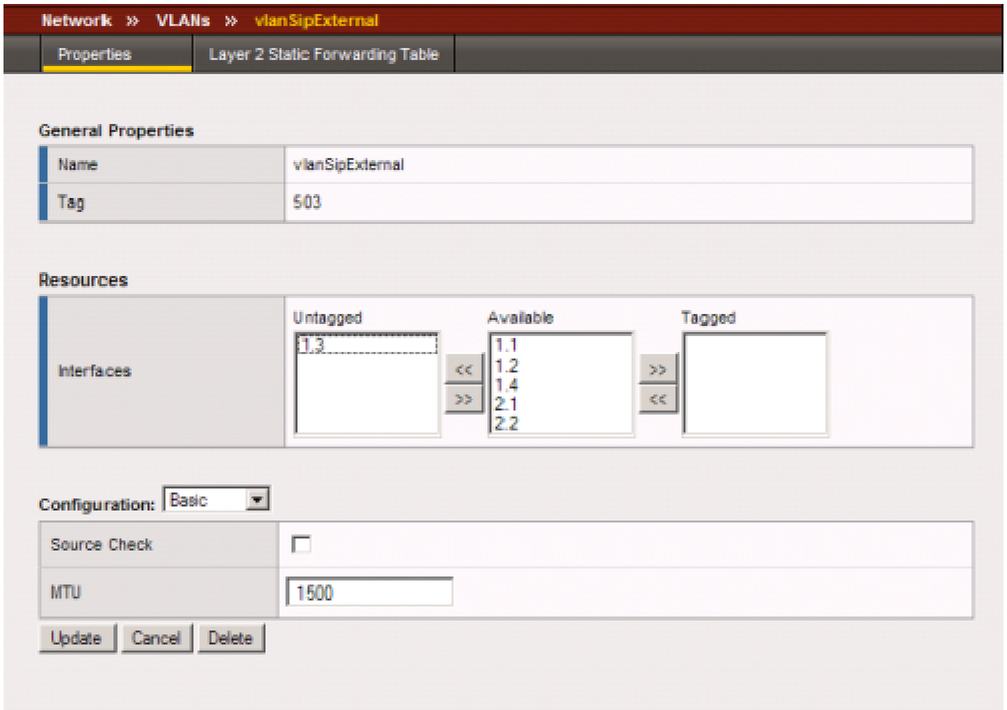
Prerequisites

- You are logged in to the BIG-IP LTM web interface.

Start

1. Go to `Network > VLANs > VLAN List`.
2. Click `Create`.
3. In the dialog box that appears, specify the following properties (see the following figure):
    a. `Name`: Enter the VLAN name for the external interface--for example, `vlanSipExternal`.
    b. `Tag`: `503` (it is set automatically).

c. `Resources > Interfaces > Untagged`: Select `1.3` in the `Available` section and click the left-pointing arrow button to move it into the `Untagged` section.


Network » VLANs » vlanSipExternal
Properties
Layer 2 Static Forwarding Table
<<
>>
>>
<<
Update
Cancel
Delete

General Properties

Name	vlanSipExternal
Tag	503

Resources

	Untagged	Available	Tagged
Interfaces	1.3	1.1 1.2 1.4 2.1 2.2	

Configuration: Basic

Source Check	□
MTU	1500

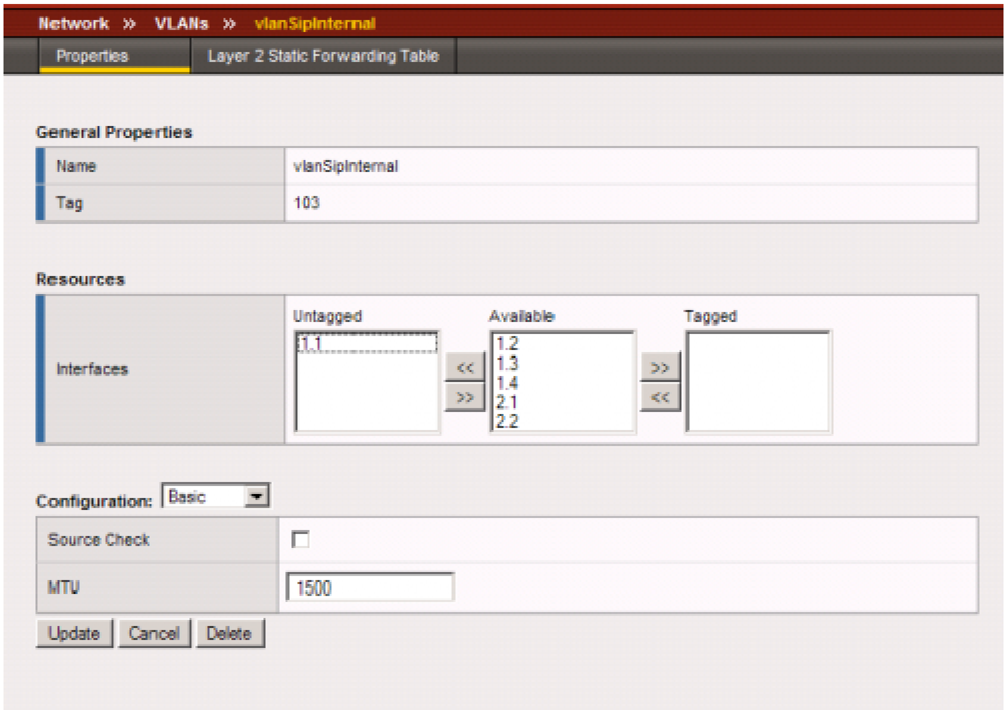
Configuring a VLAN for the External Interface

4. Click `Finished`.
5. Click `Create`.
6. In the dialog box that appears, specify the following properties (see the following figure):

   a. `Name`: Enter the VLAN name for the internal interface--for example, `vlanSipInternal`.

   b. `Tag`: `103` (it is set automatically).

   c. `Resources > Interfaces > Untagged`: Select `1.1` in the `Available` section and click the left-pointing arrow button to move it into the `Untagged` section.


Network » VLANs » vlanSipInternal
Properties
Layer 2 Static Forwarding Table
<<
>>
>>
<<
Update
Cancel
Delete

General Properties

Name	vlanSipInternal
Tag	103

Resources

	Untagged	Available	Tagged
Interfaces	1.1	1.2 1.3 1.4 2.1 2.2	

Configuration: Basic

Source Check	☐
MTU	1500

Configuring a VLAN for the Internal Interface

7. Click `Finished`.

End

Next Steps

- Configuring Self IP addresses

3. Configure Self IP addresses.

Configuring Self IP addresses

Purpose

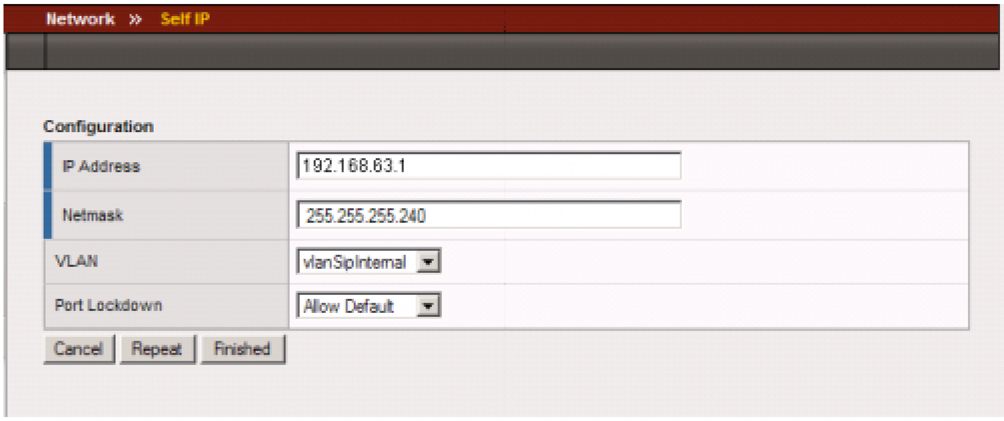
To configure two Self IP addresses--one for the external interface and one for the internal interface--and associate them with the VLANs, to access hosts in those VLANs.

Prerequisites

- Procedure: Configuring VLANs

Start

1. Go to `Network > Self IPs`.
2. Click `Create`.
3. In the dialog box that appears, specify the following properties (see the following figure):
    a. `IP Address`: Enter the IP address for the internal interface--for example, `192.168.63.1`.
    b. `Netmask`: Enter the netmask--for example, `255.255.255.240`.
    c. `VLAN`: Select the name of the VLAN to which you want to assign the self IP address--for example, `vlanSipInternal`.

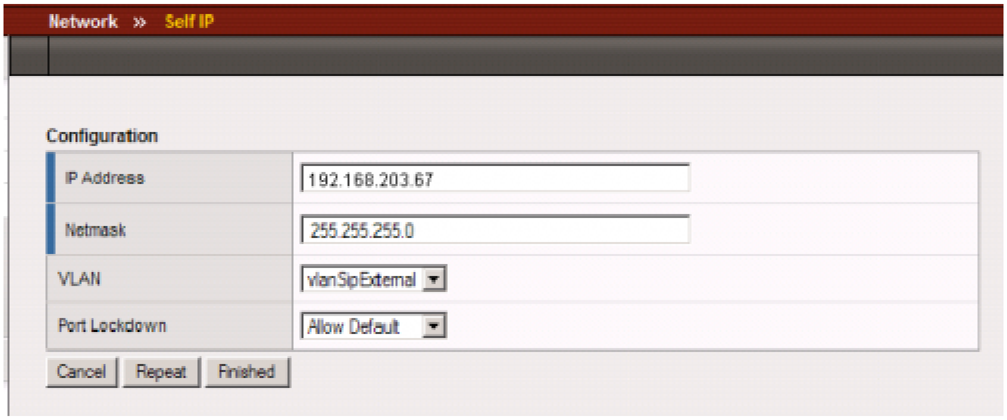


Configuration

IP Address	192.168.63.1
Netmask	255.255.255.240
VLAN	vlanSipInternal
Port Lockdown	Allow Default

Configuring a Self IP Address for the Internal Interface

4. Click `Finished`.
5. Click `Create`.
6. In the dialog box that appears, specify the following properties (see the following figure):
    a. `IP Address`: Enter the IP address for the external interface--for example, `192.168.203.67`.
    b. `Netmask`: Enter the netmask--for example, `255.255.255.0`.
    c. `VLAN`: Select the name of the VLAN to which you want to assign the self IP address--for example, `vlanSipExternal`.
    d. Click `Finished` (see the following figure).



Configuration

IP Address	192.168.203.67
Netmask	255.255.255.0
VLAN	vlanSipExternal
Port Lockdown	Allow Default

Configuring a Self IP Address for the External Interface

End

Next Steps

- Configuring the Default IP route

4. Configure the Default IP route.

Configuring the Default IP route

Purpose

To configure the default IP route.

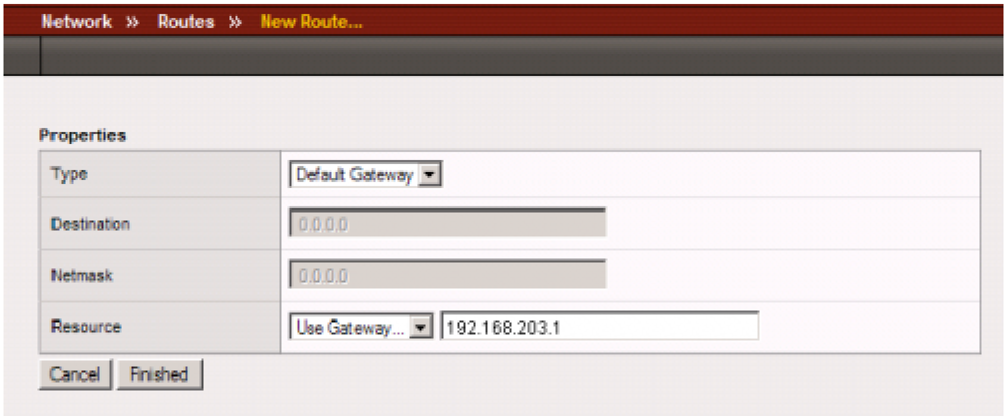
Prerequisites

- Configuring Self IP addresses

Start

1. Go to `Network > Routes`.
2. Click Add.
3. In the dialog box that appears, specify the following properties (see the following figure):
   a. `Type`: Select `Default Gateway`.
   b. `Resource > Use Gateway`: Enter the IP address for this default IP route--for example, `192.168.203.1`.

c. Click `Finished`.



Properties

Type	Default Gateway
Destination	0.0.0.0
Netmask	0.0.0.0
Resource	Use Gateway... 192.168.203.1

Configuring Default IP Route

End

Next Steps

- Configuring SIP Server nodes

5. Configure SIP Server nodes.

Configuring SIP Server nodes

Purpose

To configure two SIP Server nodes, primary and backup.

Prerequisites

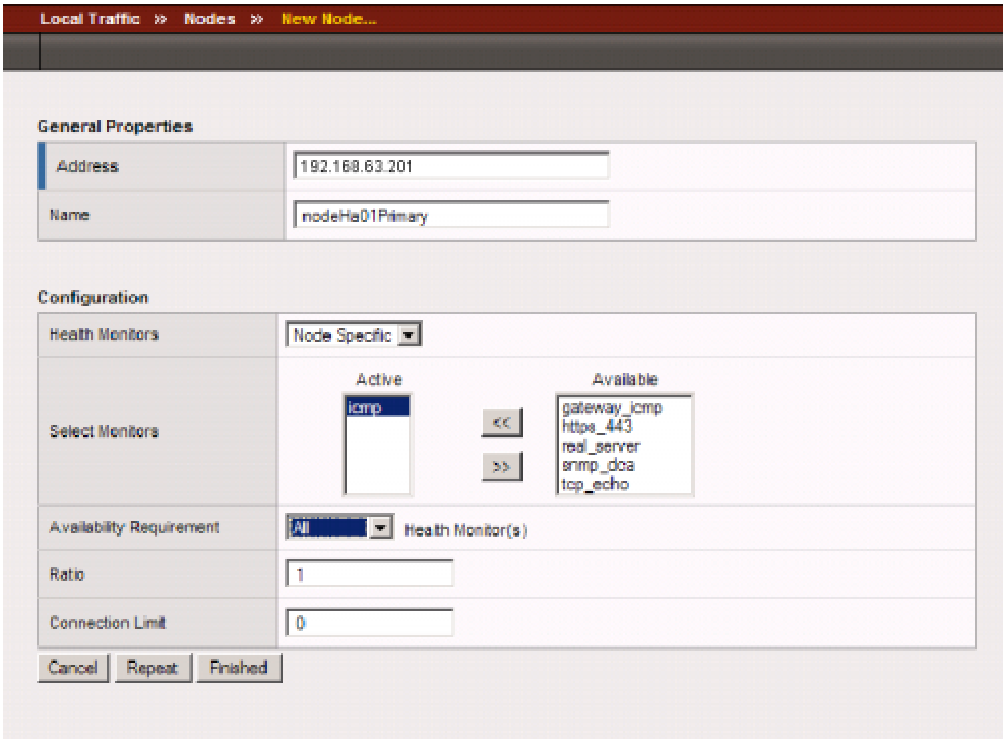
- Configuring the Default IP route

Start

1. Go to `Local Traffic > Nodes`.
2. Click `Create`.
3. In the dialog box that appears, specify the following properties (see the following figure):
   a. `Address:` Enter the IP address for the primary SIP Server node--for example, `192.168.63.201`.
   b. `Name:` Enter the node name--for example, `nodeHa01Primary`.

c. `Health Monitors`: Select `Node Specific`.

d. `Select Monitors > Active`: Select `icmp`.



General Properties

Address	192.168.63.201
Name	nodeHa01Primary

Configuration

Health Monitors	Node Specific
Select Monitors	Active: icmp Available: gateway_icmp, https_443, real_server, snmp_dca, tcp_echo
Availability Requirement	All Health Monitor(s)
Ratio	1
Connection Limit	0

Configuring a Primary SIP Server Node

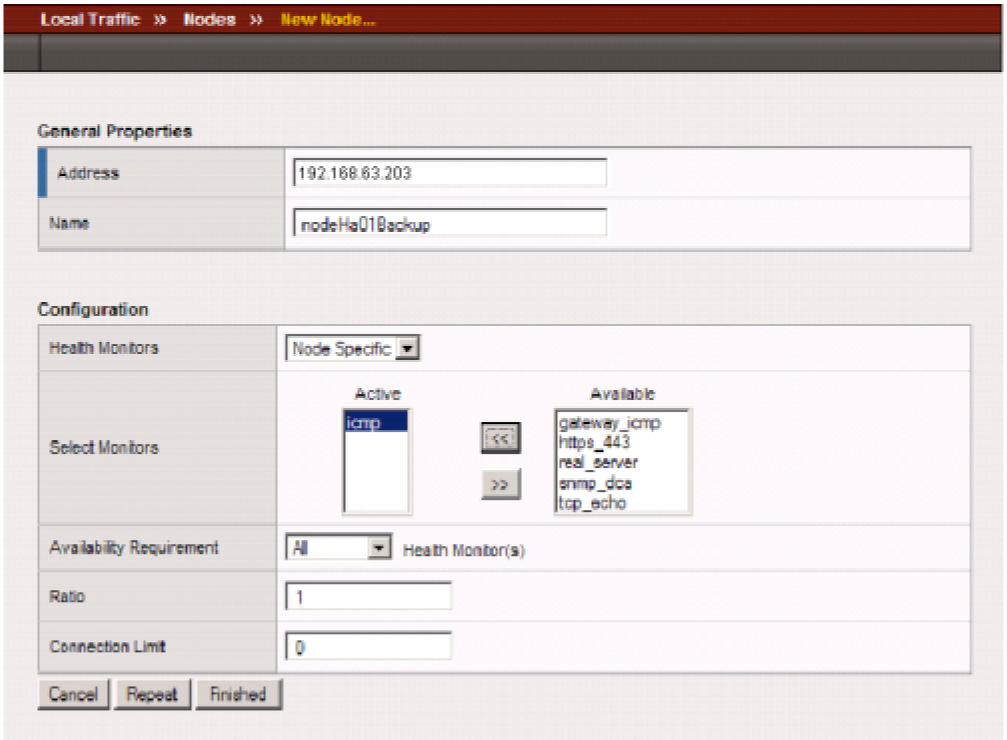
4. Click `Finished`.
5. Click `Create`.
6. In the dialog box that appears, specify the following properties (see the following figure):

a. `Address`: Enter the IP address for the backup SIP Server node--for example, `192.168.63.203`.

b. `Name`: Enter the node name--for example, `nodeHa01Backup`.

c. `Health Monitors`: Select `Node Specific`.

d. `Select Monitors > Active`: Select `icmp`.



General Properties

Address	192.168.63.203
Name	nodeHa01Backup

Configuration

Health Monitors	Node Specific
Select Monitors	Active: icmp Available: gateway_icmp, https_443, real_server, snmp_dca, tcp_echo
Availability Requirement	All Health Monitor(s)
Ratio	1
Connection Limit	0

Configuring a Backup SIP Server Node

7. Click `Finished`.

End

6. Configure a health monitor.

Configuring a health monitor

Overview

In general, the BIG-IP LTM uses health monitors to determine whether a server to which messages can be routed is operational (active). Servers that are flagged as not operational (inactive) will cause the BIG-IP LTM to route messages to another server if one is present in the same server pool. However, primary and backup SIP Servers must be configured as the only members of the same server pool--one member active (primary) and one member inactive (backup).

In this procedure, the BIG-IP LTM is configured to use the health monitor of SIP type in UDP mode. This means that the `OPTIONS` request method will be sent to both primary and backup SIP Servers.

Any response to `OPTIONS` is configured as `Accepted Status Code`.

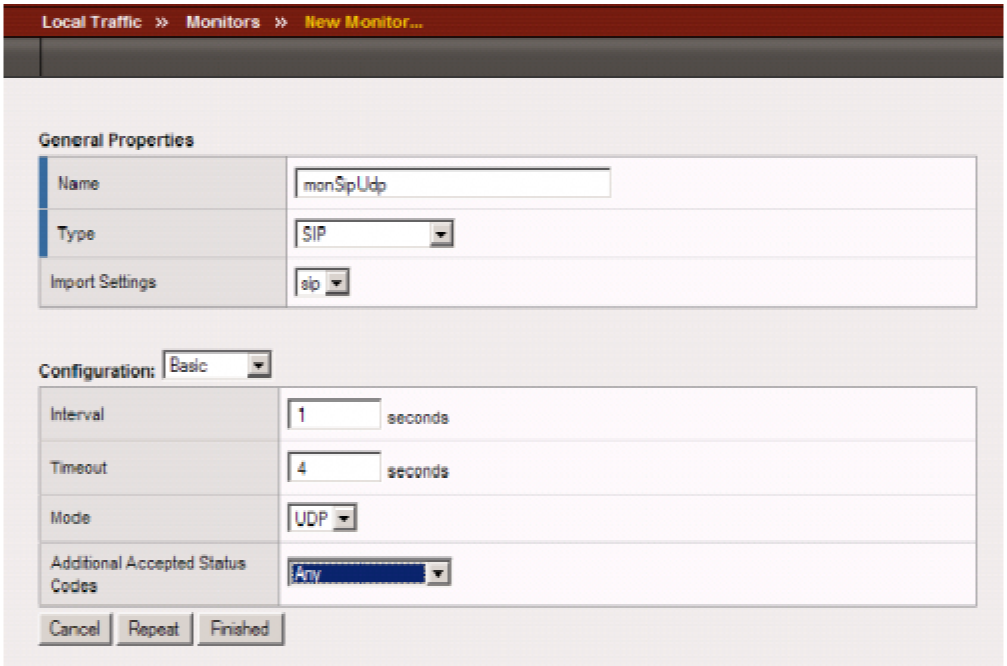
SIP Server always starts in backup mode, establishes a permanent connection with the Genesys Management Layer, and changes its role to primary only if a trigger from the Management Layer is received. Such trigger is only generated if no other primary SIP Server is currently running. After switching to primary mode, SIP Server responds to UDP packets received on the SIP port specified by the `sip-port` configuration option. Therefore, after receiving the `OPTIONS` request from the BIG-IP LTM, SIP Server responds to the health check, and the BIG-IP LTM marks SIP Server as active.

When running in backup mode, SIP Server ignores UDP messages. Since the BIG-IP LTM does not receive any response to the `OPTIONS` request, it marks the backup SIP Server as inactive. If SIP Server does not respond because of network latency or other reasons, the BIG-IP LTM will mark SIP Server as inactive, and continue sending ping messages periodically.

The `Interval` setting defines how often pool members (primary and backup) are checked for presence. The `Timeout` setting defines the waiting time before an unresponsive member of the pool is marked as inactive. Regardless of the member's status (or SIP Server status), the BIG-IP LTM will always check servers for presence. When an inactive member responds to the health check, it is marked as active. In this configuration, the `Interval` parameter is set to one second and `Timeout` to four seconds in order to minimize a possible delay that might result from a switchover.

Start

1. Go to `Local Traffic > Monitors`.
2. Click `Create`.
3. In the dialog box that appears, specify the following properties (see the following figure):
    a. `Name`: Enter the name for this health monitor--for example, `monSipUdp`.
    b. `Type`: Select `SIP`.
    c. `Configuration`: Select `Basic`.
    d. `Interval`: Enter `1`.
    e. `Timeout`: Enter `4`.
    f. `Mode`: Select `UDP`.
    g. `Additional Accepted Status Codes`: Select `Any`.



General Properties

Name	monSipUdp
Type	SIP
Import Settings	sip

Configuration: Basic

Interval	1 seconds
Timeout	4 seconds
Mode	UDP
Additional Accepted Status Codes	Any

Configuring a Health Monitor

4. Click Finished.

End

Next Steps

- Configuring a server pool

7. Configure a server pool.

Configuring a server pool

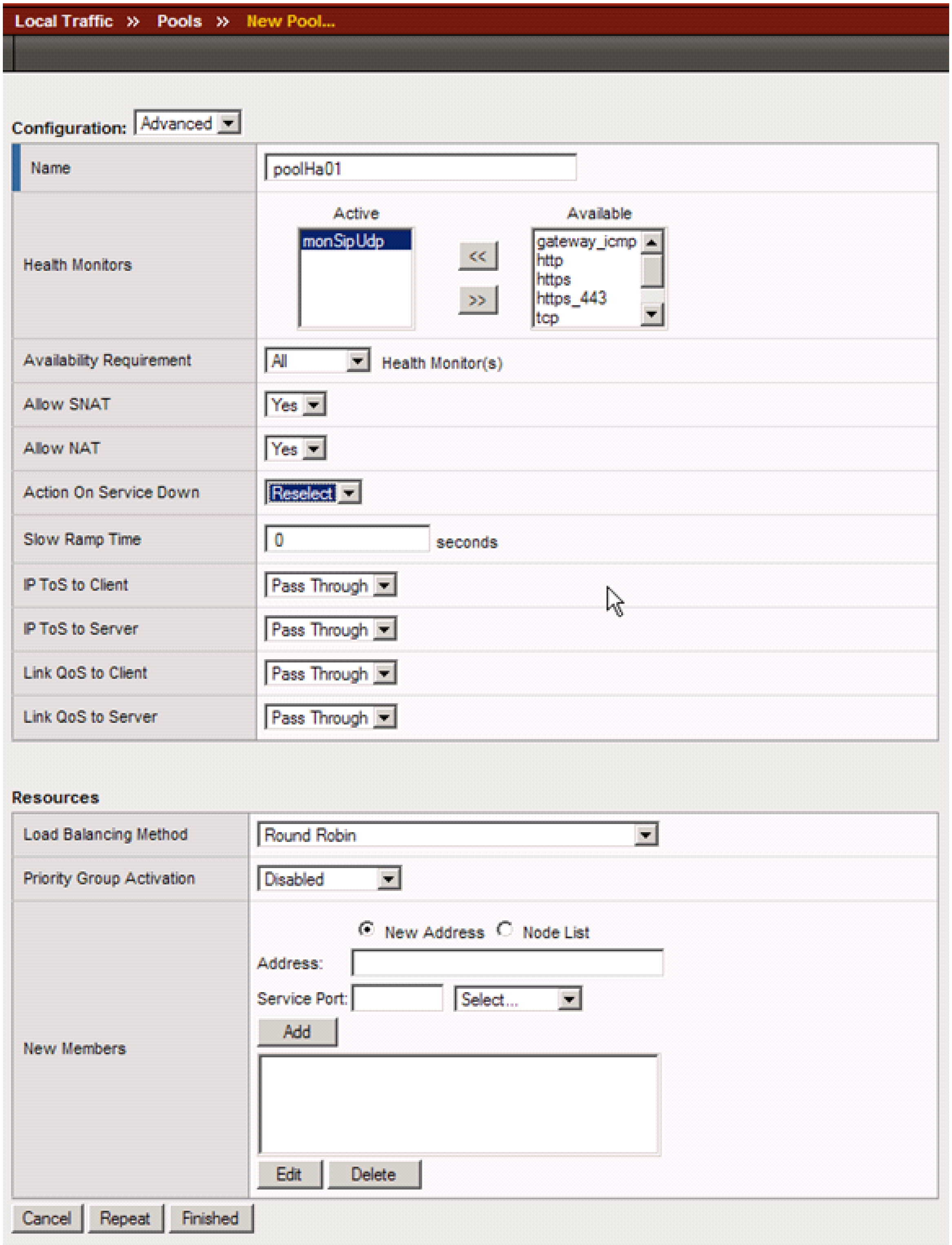
Purpose

To configure a server pool with which the BIG-IP LTM will communicate.

Start

1. Go to `Local Traffic > Pools`.

2. Click `Create`.
3. In the dialog box that appears, specify the following properties (see the following figure):
   a. `Name`: Enter the name for this server pool--for example, the `poolHa01`.
   b. `Health Monitors > Active`: Select `monSipUdp`.
   c. `Action On Service Down`: Select `Reselect`.
   d. `Load Balancing Method`: Select `Round Robin`.
   e. `Priority Group Activation`: Select `Disabled`.



Local Traffic » Pools » New Pool...

Configuration: Advanced

Name	poolHa01
Health Monitors	Active: monSipUdp Available: gateway_icmp, http, https, https_443, tcp
Availability Requirement	All Health Monitor(s)
Allow SNAT	Yes
Allow NAT	Yes
Action On Service Down	Reselect
Slow Ramp Time	0 seconds
IP ToS to Client	Pass Through
IP ToS to Server	Pass Through
Link QoS to Client	Pass Through
Link QoS to Server	Pass Through

Resources

Load Balancing Method	Round Robin
Priority Group Activation	Disabled
New Members	● New Address ○ Node List Address: Service Port: Select...

Configuring a Server Pool

4. Click Finished.

   End

8. Add server pool members.

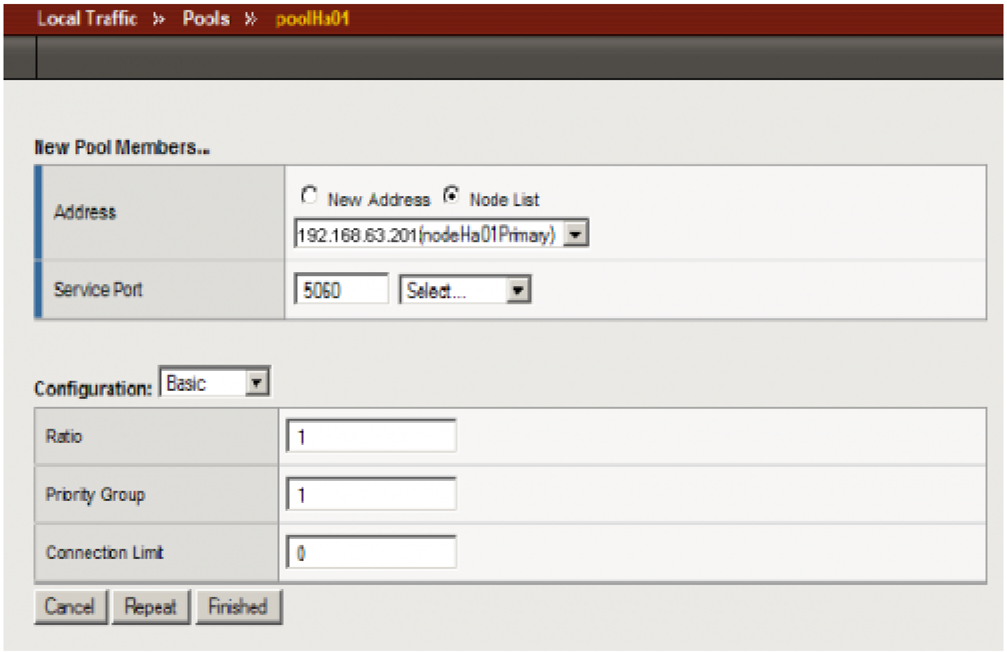
Adding server pool members

Purpose

To add primary and backup SIP Servers to the server pool. Note that they must be the only members of this server pool.

Start

1. Go to `Local Traffic > Pools > poolHa01 > Members`.
2. Click Add.
3. In the dialog box that appears, specify the following properties (see the following figure):
   a. `Address > Node List`: Select the primary server node you created in Configuring SIP Server nodes. In our example, it would be `192.168.63.201` (nodeHa01Primary).
   b. `Service Port`: Enter `5060`.



New Pool Members...

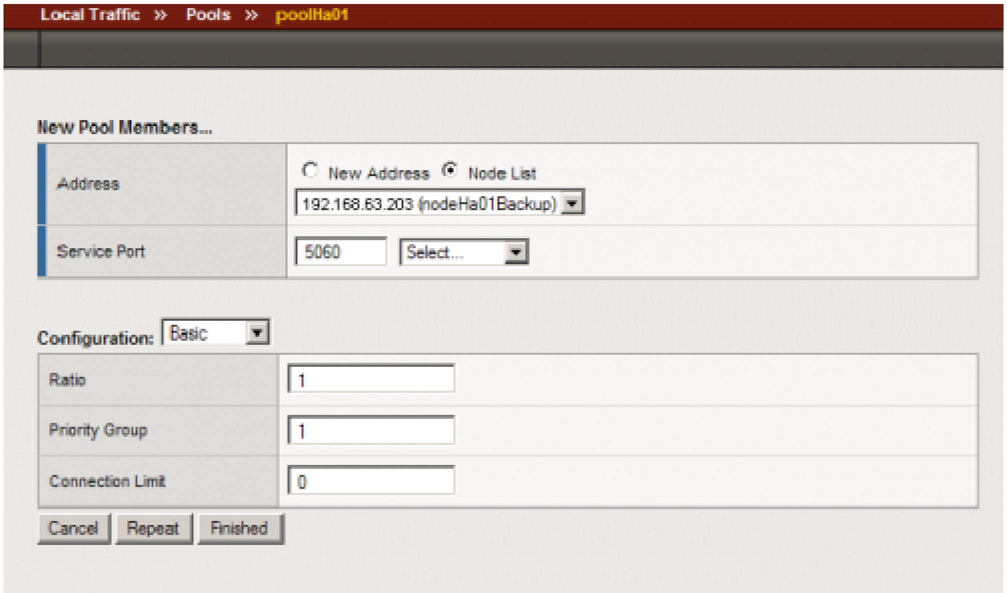
Address	○ New Address ● Node List 192.168.63.201 (nodeHa01Primary)
Service Port	5060 Select...

Configuration: Basic

Ratio	1
Priority Group	1
Connection Limit	0

Adding the Primary SIP Server to the Server Pool

4. Click Finished.
5. Click Add.
6. In the dialog box that appears, specify the following properties (see the following figure):
   1. Address > Node List: Select the backup server node you created in the Configuring SIP Server nodes. In our example, it would be 192.168.63.203 (nodeHa01Backup).
   2. Service Port: Enter 5060.



New Pool Members...

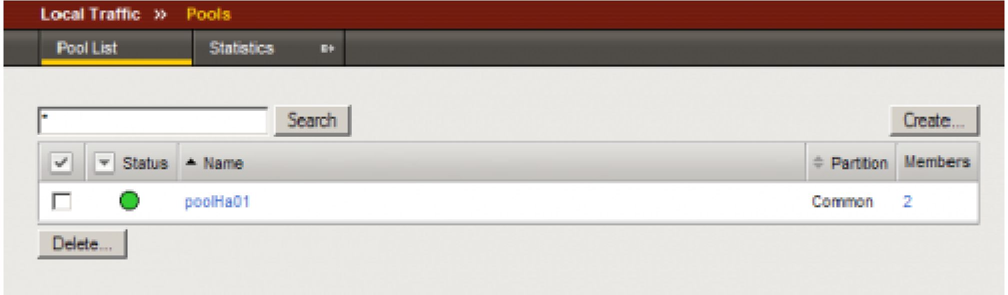
Address	○ New Address ● Node List 192.168.63.203 (nodeHa01Backup)
Service Port	5060 Select...

Configuration: Basic

Ratio	1
Priority Group	1
Connection Limit	0

Adding the Backup SIP Server to the Server Pool

7. Click Finished.
8. Go to Local Traffic > Pools. The status of the poolHa01 server pool displays as available (green) (see the following figure).



☑	Status	Name	Partition	Members
□	🟢	poolHa01	Common	2

The Server Pool of Two Members

End

9. Configure data groups.

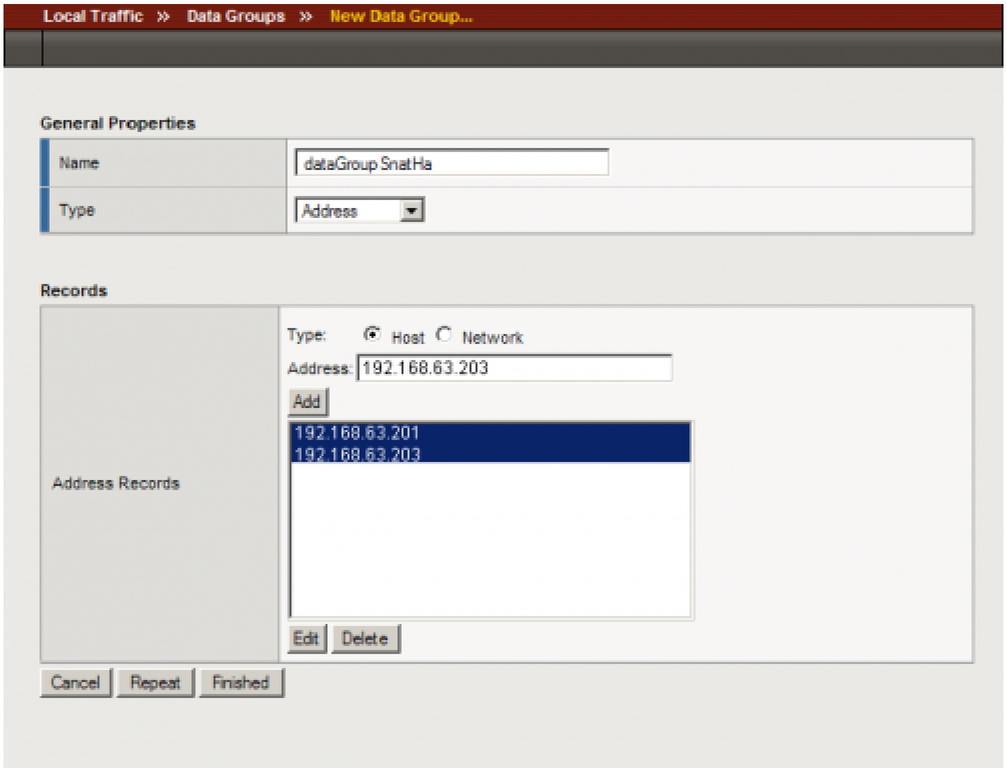
Configuring data groups

Purpose

To configure data groups that will be used by the iRule. One data group (`dataGroupHa`) contains physical IP addresses of primary and backup SIP Server nodes. The second data group (`dataGroupSnatExcluded`) contains IP addresses of the groups that will be excluded from applying SNAT, such as the Genesys Configuration Server group and Genesys T-Library Clients group (see the Device Communication Groups figure).

Start

1. Go to `Local Traffic > iRules > Data Group List`.
2. Click `Create`.
3. In the dialog box that appears, specify the following properties (see the following figure):
   a. `Name`: Enter the name for this data group--for example, `dataGroupSnatHa`.
   b. `Type`: Select `Address`.
   c. `Address Records > Type Host > Address`: Enter the host IP address of the primary server node--for example, `192.168.63.201`.
   d. Click Add.
   e. `Address Records > Type Host >Address`: Enter the host IP address of the backup server node--for example, `192.168.63.203`.
   f. Click Add.



General Properties

Name	dataGroup SnatHa
Type	Address

Records

Address Records

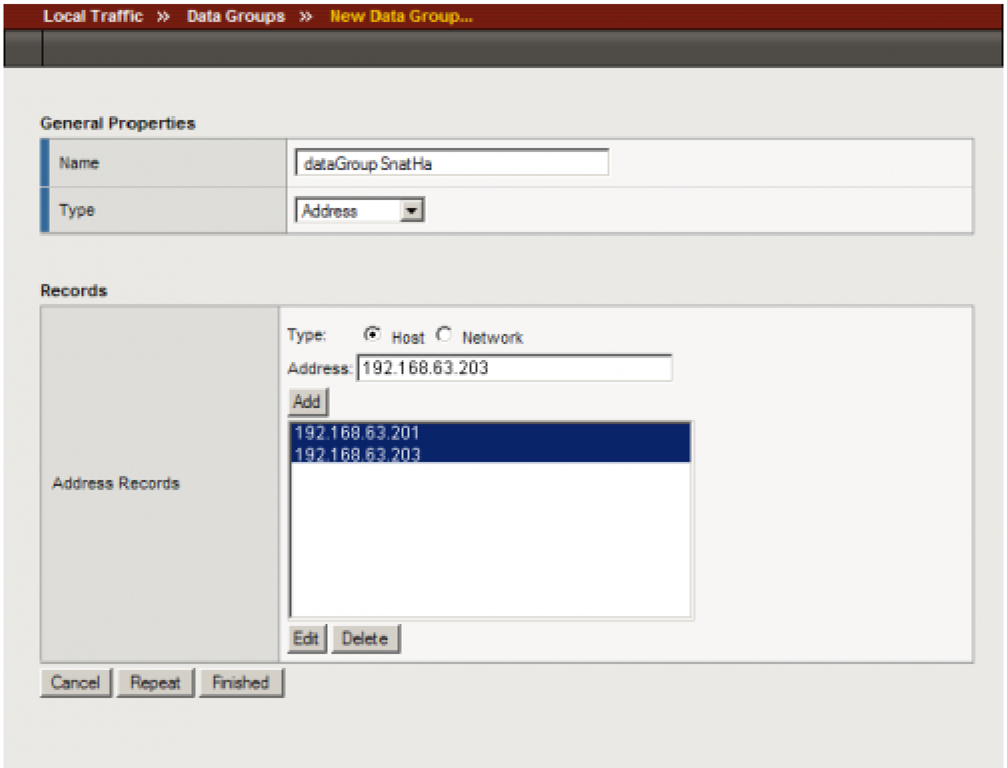
Type: ● Host ○ Network

Address: 192.168.63.203

192.168.63.201
192.168.63.203

Configuring a Data Group for SNAT

4. Click Finished.
5. Click Create.
6. In the dialog box that appears, specify the following properties (see the following figure):
   a. Name: Enter the name for this data group--for example, dataGroupSnatExcluded.
   b. Type: Select Address.
   c. Address Records > Type Host > Address: Enter the host IP address of Genesys Configuration Server--for example, 172.21.226.73.
   d. Click Add.
   e. Address Records > Type Network > Address: Enter the IP address and net mask--for example, 192.168.89.0/255.255.255.0.
   f. Click Add.



Name: dataGroupSnatHa

Type: Address

Address Records

Type: ● Host ○ Network

Address: 192.168.63.203

192.168.63.201
192.168.63.203

Configuring a Data Group for SNAT Exclusions

7. Click `Finished`.

End

10. Configure a SNAT pool.

Configuring a SNAT pool

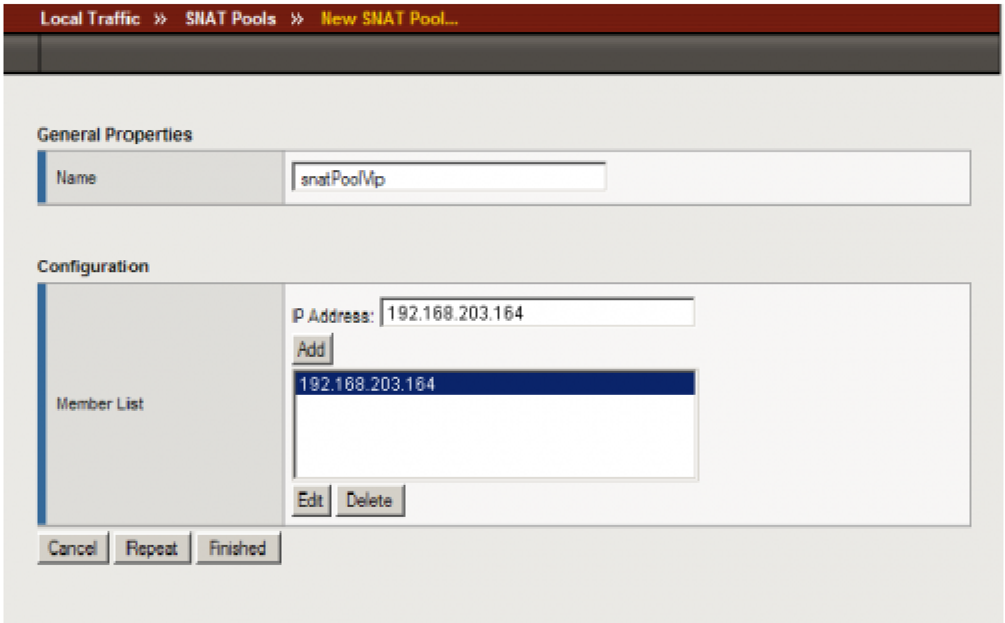
Purpose

To configure a SNAT pool that specifies the Virtual IP address to be used as a source IP address for any packet that originates from the primary or backup SIP Server to which SNAT is applied (with the exception of the devices specified in the `dataGroupSnatExcluded` data group). SNAT is the mapping of one or more original IP addresses to a translation address.

Start

1. Go to `Local Traffic > SNAT Pools`.

2. Click `Create`.
3. In the dialog box that appears, specify the following properties (see the following figure):
   a. `Name`: Enter the name for this SNAT pool--for example, `snatPoolVip`.
   b. `Configuration > Members List > IP Address`: Enter the IP address to be used as a source IP address--for example, `192.168.203.164`.



Name: snatPoolVip

IP Address: 192.168.203.164

192.168.203.164

Configuring a SNAT Pool

4. Click `Finished`.

End

11. Configure an iRule.

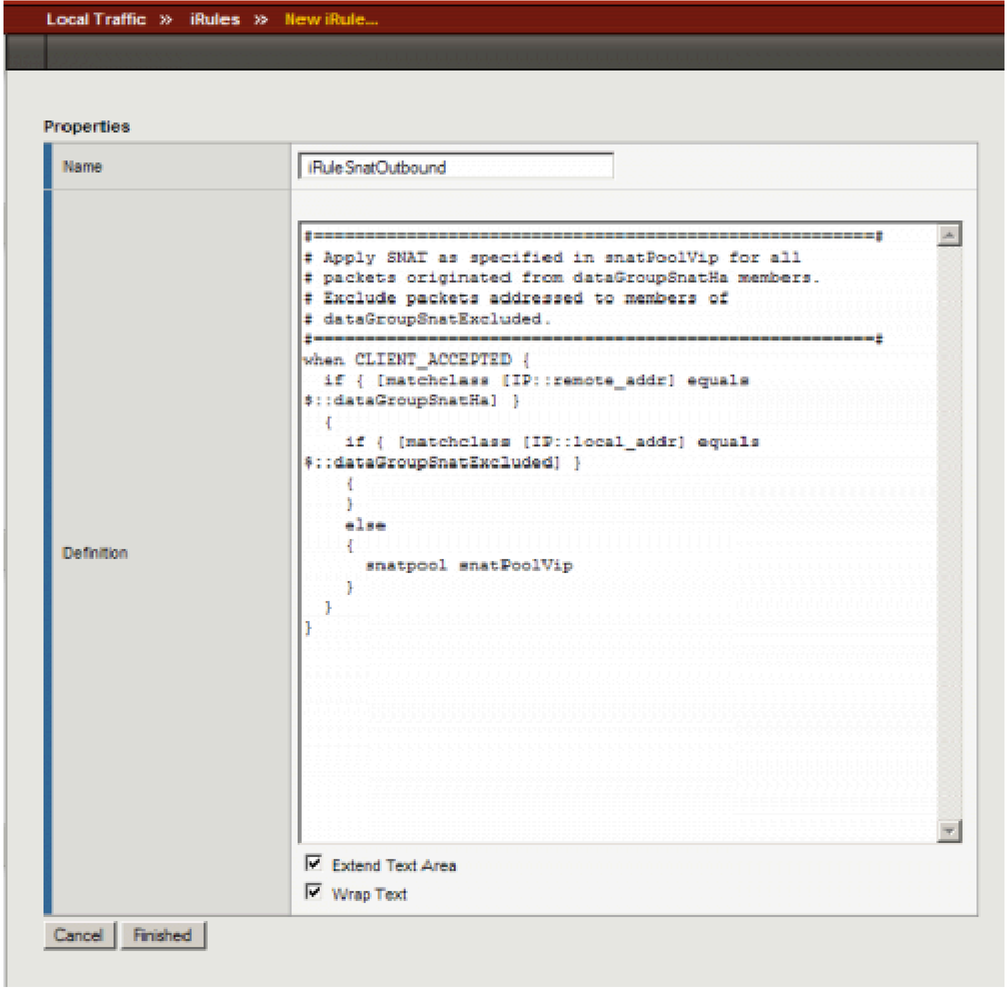
Configuring an iRule

Purpose

To configure an iRule that is used to perform SNAT to the Virtual IP address to any packets that originate from the primary or backup SIP Server (with the exception of the packets addressed to Configuration Server and the Genesys T-Library Clients group). This iRule will then be associated with a Virtual Server for the outbound traffic, `vsWildCardOutbound`. In this deployment architecture, the HA synchronization traffic between primary and backup SIP Servers does not pass through the BIG-IP LTM, that is why it is excluded from applying SNAT.

Start

1. Go to `Local Traffic > iRules`.
2. Click `Create`.
3. In the dialog box that appears, specify the following properties (see the following figure):



Name: iRuleSnatOutbound

Definition:

```
#======================================================#
# Apply SNAT as specified in snatPoolVip for all
# packets originated from dataGroupSnatHa members.
# Exclude packets addressed to members of
# dataGroupSnatExcluded.
#======================================================#
when CLIENT_ACCEPTED {
  if { [matchclass [IP::remote_addr] equals
$::dataGroupSnatHa] }
  {
    if { [matchclass [IP::local_addr] equals
$::dataGroupSnatExcluded] }
    {
    }
    else
    {
      snatpool snatPoolVip
    }
  }
}
```

☑ Extend Text Area

☑ Wrap Text

Configuring an iRule

a. `Name`: Enter the name for this iRule--for example, `iRuleSnatOutbound`.

b. `Definition`: Enter the following text:

```
#======================================================#
# Apply SNAT as specified in snatPoolVip for all
# packets originated from dataGroupSnatHa members.
# Exclude packets addressed to members of
# dataGroupSnatExcluded.
#======================================================#
when CLIENT_ACCEPTED {
  if { [matchclass [IP::remote_addr] equals $::dataGroupSnatHa] }
  {
    if { [matchclass [IP::local_addr] equals $::dataGroupSnatExcluded] }
    {
    }
    else
    {
      snatpool snatPoolVip
    }
  }
}
```

4. Click Finished.

End

12. Configure a Virtual Server.

Configuring a Virtual Server

Complete the following steps:

[+] Configuring a Virtual Server for outbound traffic

Purpose

To configure a Virtual Server to be used for outbound traffic. It is associated with a VLAN that is configured for the internal interface (see Configuring VLANs) and it has iRule assigned to Resources, which applies SNAT to all packets (except for packets addressed to Configuration Server).

Prerequisites

- Configuring an iRule

Start

1. Go to Local Traffic > Virtual Servers.
2. Click Create.
3. In the dialog box that appears, specify the following properties (see the following figure):
    a. Name: Enter the name for this Virtual Server--for example, vsWildCardOutbound.
    b. Destination > Type: Select Network.
    c. Destination > Address: Enter 0.0.0.0.
    d. Destination > Mask: Enter 0.0.0.0.
    e. Service Port: Enter * (asterisk).
    f. Configuration: Select Basic.

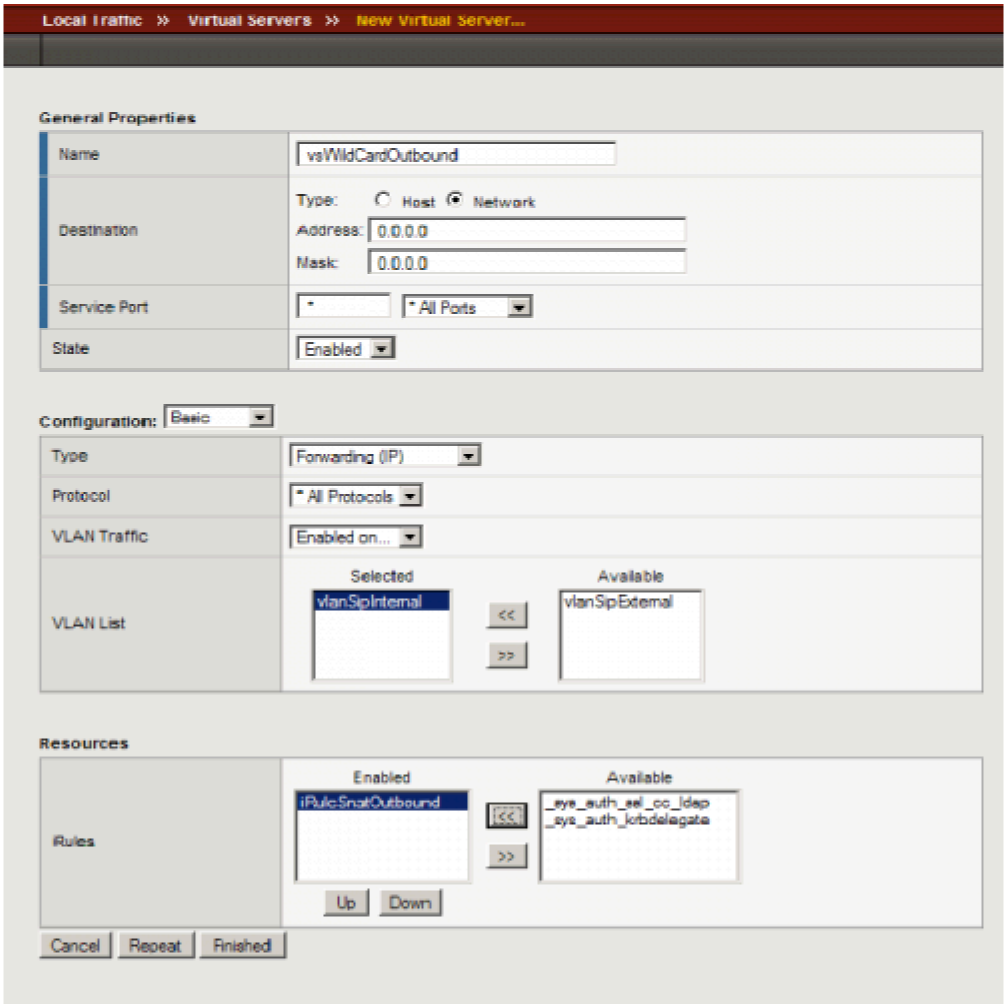
g. `Type`: Select `Forwarding (IP)`.

h. `Protocol`: Select `All Protocols`.

i. `VLAN Traffic`: Select `Enabled on...`.

j. `VLAN List Selected`: Select `vlanSipInternal`.

k. `Resources > iRules > Enabled`: Select `iRuleSnatOutbound`.



General Properties

Name	vsWildCardOutbound
Destination	Type: ○ Host ● Network Address: 0.0.0.0 Mask: 0.0.0.0
Service Port	* / * All Ports
State	Enabled

Configuration: Basic

Type	Forwarding (IP)
Protocol	* All Protocols
VLAN Traffic	Enabled on...
VLAN List	Selected: vlanSipInternal Available: vlanSipExternal

Resources

Rules	Enabled: iRuleSnatOutbound Available: _sys_auth_ssl_cc_ldap, _sys_auth_krbdelegate

Configuring a Wildcard Virtual Server for Outbound Traffic

4. Click `Finished`.

End

[+] Configuring a Virtual Server for inbound traffic

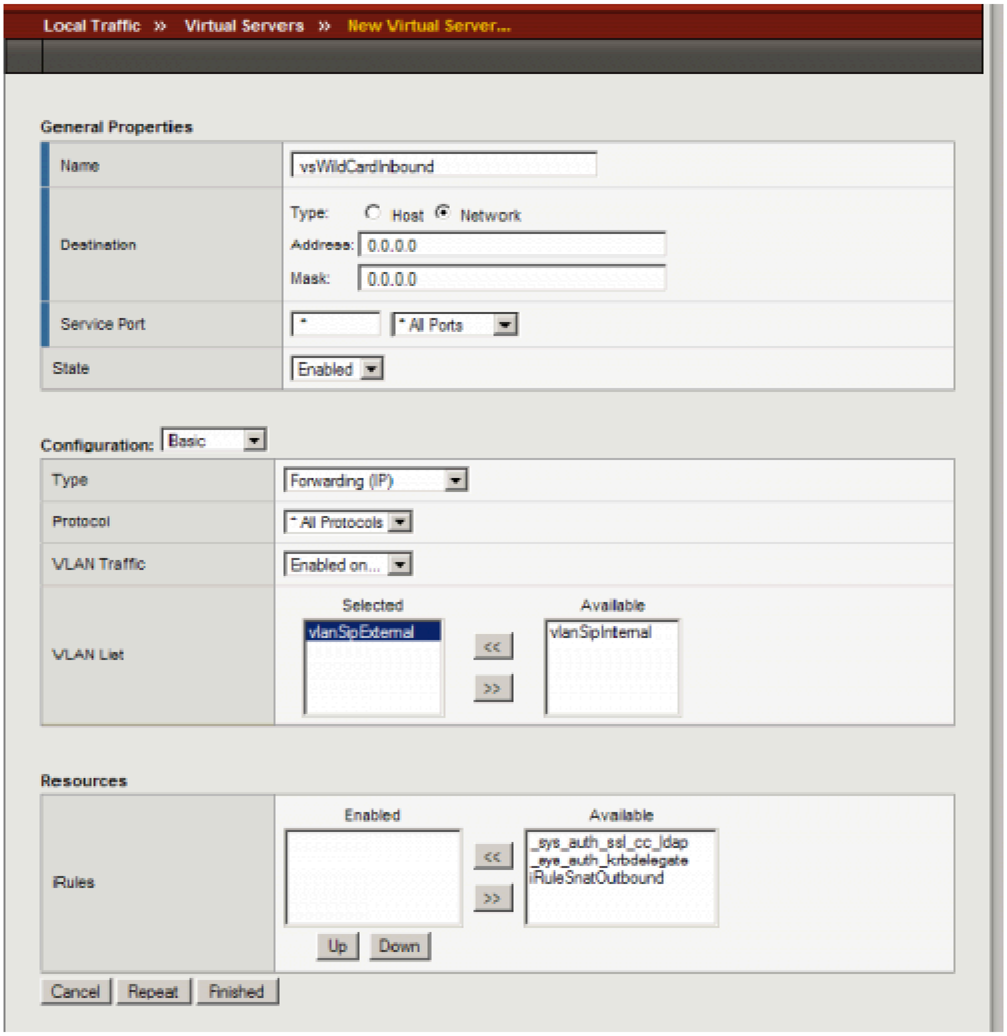
Purpose

To configure a Virtual Server for inbound traffic. In Layer 3/Routing configuration mode, the BIG-IP LTM passes through only those packets that have a destination matching a virtual server. Having the

Virtual Server for inbound traffic allows packets with a destination that matches the physical IP address of the primary or backup SIP Server to pass through.

Start

1. Go to `Local Traffic > Virtual Servers`.
2. Click `Create`.
3. In the dialog box that appears, specify the following properties (see the following figure):
    a. `Name`: Enter the name for this Virtual Server--for example, `vsWildCardInbound`.
    b. `Destination > Type`: Select `Network`.
    c. `Destination > Address`: Enter `0.0.0.0`.
    d. `Destination > Mask`: Enter `0.0.0.0`.
    e. `Service Port`: Enter * (asterisk).
    f. `Configuration`: Select `Basic`.
    g. `Type`: Select `Forwarding (IP)`.
    h. `Protocol`: Select `All Protocols`.
    i. `VLAN Traffic`: Select `Enabled on...`.
    j. `VLAN List Selected`: Select `vlanSipExternal`.



General Properties

Name	vsWildCardInbound
Destination	Type: ○ Host ● Network Address: 0.0.0.0 Mask: 0.0.0.0
Service Port	* * All Ports
State	Enabled

Configuration: Basic

Type	Forwarding (IP)
Protocol	* All Protocols
VLAN Traffic	Enabled on...
VLAN List	Selected: vlanSipExternal Available: vlanSipInternal

Resources

iRules	Enabled: Available: _sys_auth_ssl_cc_ldap _sys_auth_krbdelegate iRuleSnatOutbound

Configuring a Wildcard Virtual Server for Inbound Traffic

4. Click Finished.

End

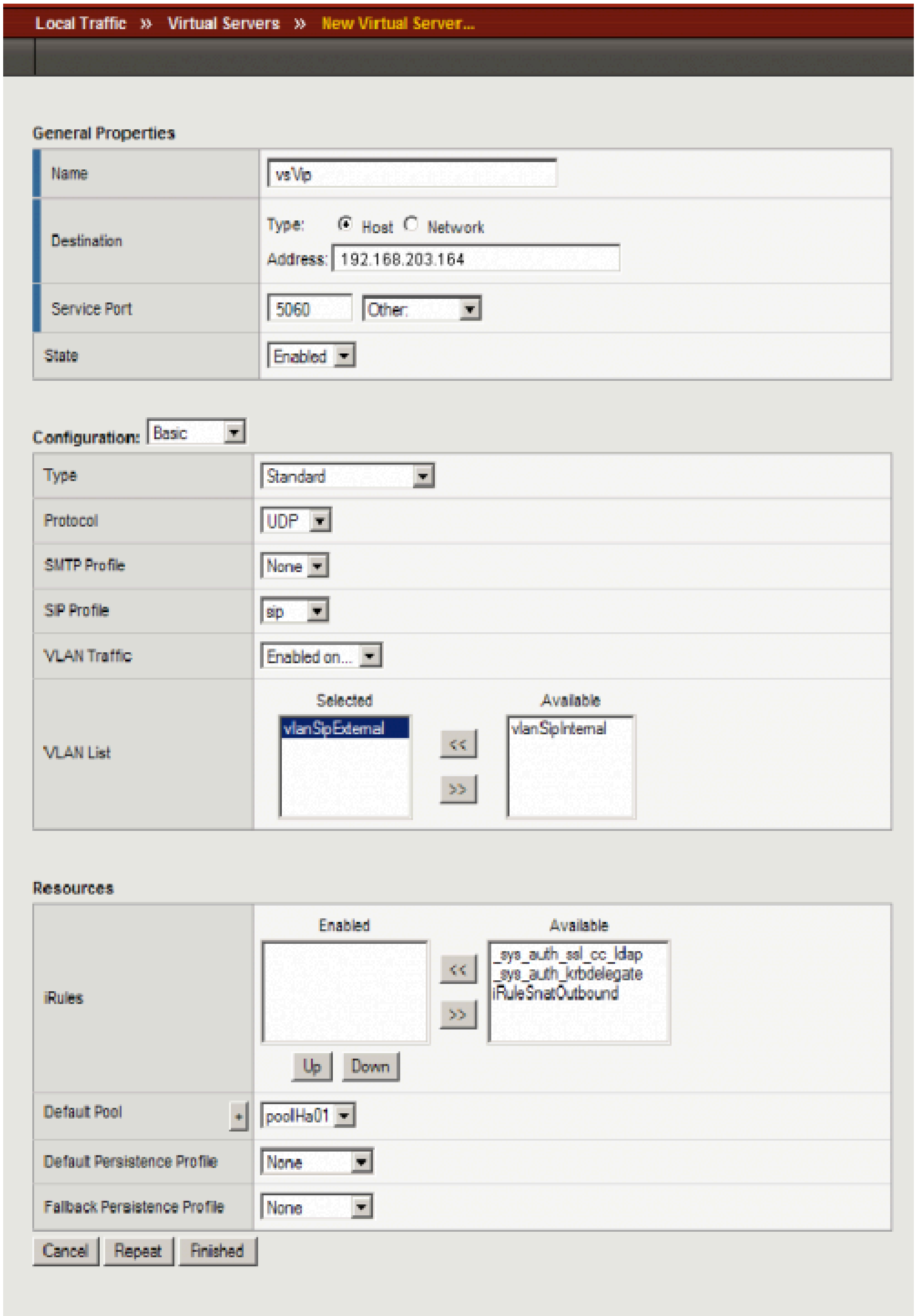
[+] Configuring Virtual Servers for UDP and TCP SIP communications

Purpose

To configure two virtual servers to handle traffic directed to a Virtual IP address: one virtual server for SIP communications using the UDP as a transport protocol and one virtual server for SIP communications using the TCP as a transport protocol. The Virtual IP address is used by SIP clients to contact SIP Server. In other words, the Virtual IP address hides two physical IP addresses (used by the primary and backup servers) and presents the SIP Server HA pair as a single entity for all SIP-based communications.

Start

1. Go to `Local Traffic > Virtual Servers`.
2. Click `Create`.
3. In the dialog box that appears, specify the following properties (see the following figure):
   a. `Name`: Enter the name for this Virtual Server--for example, `vsVip`.
   b. `Destination > Type`: Select `Host`.
   c. `Destination > Address`: Enter the IP address for this Virtual Server--for example, `192.168.203.164`.
   d. `Service Port`: Enter `5060` and select `Other`.
   e. `State`: Select `Enabled`.
   f. `Configuration`: Select `Basic`.
   g. `Type`: Select `Standard`.
   h. `Protocol`: Select `UDP`.
   i. `SMTP Profile`: Select `None`.
   j. `SIP Profile`: Select `sip`.
   k. `VLAN Traffic`: Select `Enabled on...`.
   l. `VLAN List Selected`: Select `vlanSipExternal`.
   m. `Resources > Default Pool` > Select `poolHa01`.



Local Traffic » Virtual Servers » New Virtual Server...

General Properties

Name	vsVip
Destination	Type: ● Host ○ Network Address: 192.168.203.164
Service Port	5060 Other:
State	Enabled

Configuration: Basic

Type	Standard
Protocol	UDP
SMTP Profile	None
SIP Profile	sip
VLAN Traffic	Enabled on...
VLAN List	Selected: vlanSipExternal Available: vlanSipInternal

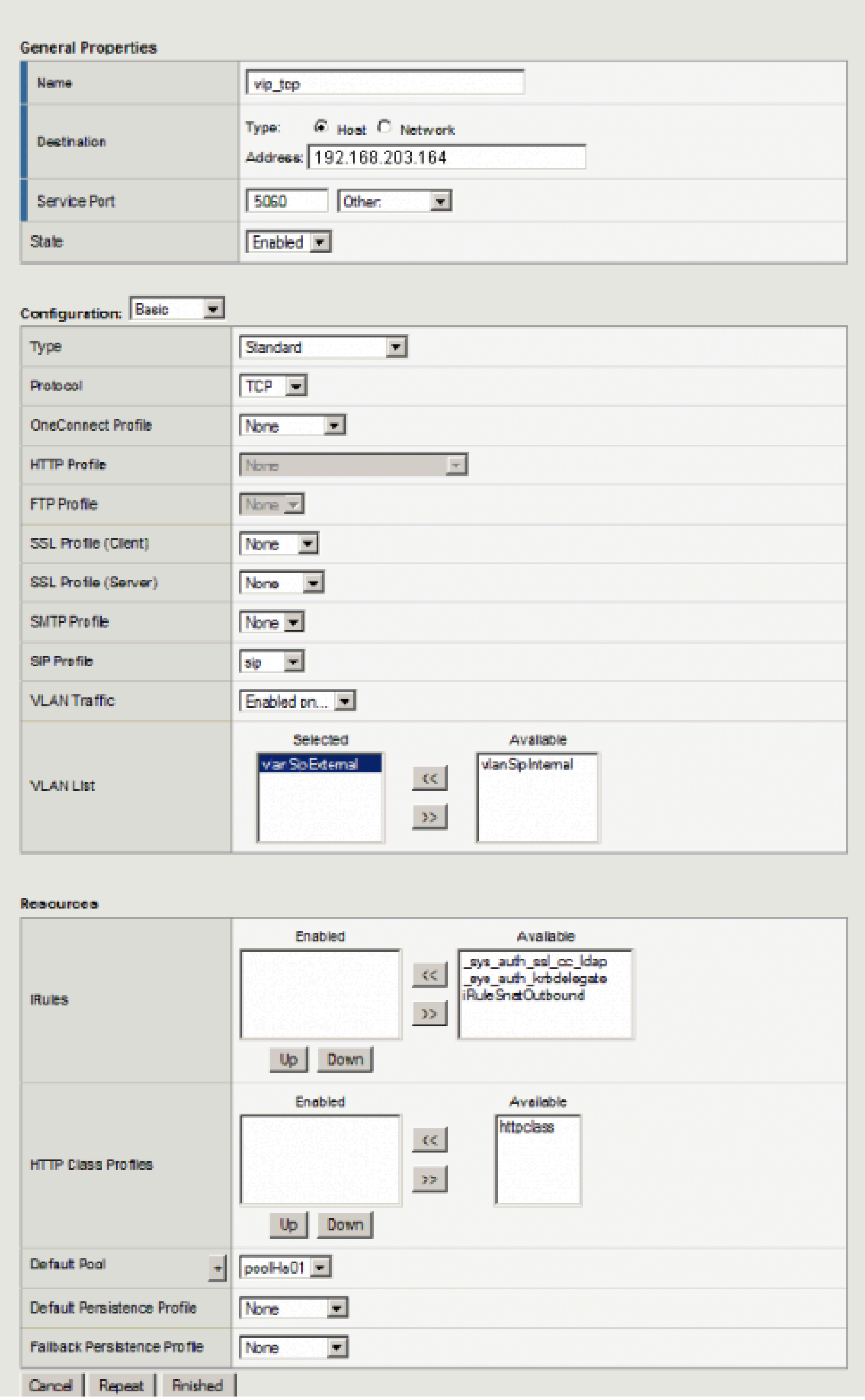
Resources

iRules	Enabled: Available: _sys_auth_ssl_cc_ldap, _sys_auth_krbdelegate, iRuleSnatOutbound
Default Pool	poolHa01
Default Persistence Profile	None
Fallback Persistence Profile	None

Configuring a Virtual Server for UDP-Based Communications

4. Click Finished.
5. Click Create.

6. In the dialog box that appears, specify the following properties (see the following figure):
    a. `Name`: Enter the name for this Virtual Server--for example, `vip_tcp`.
    b. `Destination > Type`: Select `Host`.
    c. `Destination > Address`: Enter the IP address for this Virtual Server--for example, `192.168.203.164`.
    d. `Service Port`: Enter `5060` and select `Other`.
    e. `State`: Select `Enabled`.
    f. `Configuration`: Select `Basic`.
    g. `Type`: Select `Standard`.
    h. `Protocol`: Select `TCP`.
    i. `SMTP Profile`: Select `None`.
    j. `SIP Profile`: Select `sip`.
    k. `VLAN Traffic`: Select `Enabled on....`
    l. `VLAN List Selected`: Select `vlanSipExternal`.
    m. `Resources > Default Pool` > Select `poolHa01`.



General Properties

Name	vip_tcp
Destination	Type: ● Host ○ Network Address: 192.168.203.164
Service Port	5060 Other:
State	Enabled

Configuration: Basic

Type	Standard
Protocol	TCP
OneConnect Profile	None
HTTP Profile	None
FTP Profile	None
SSL Profile (Client)	None
SSL Profile (Server)	None
SMTP Profile	None
SIP Profile	sip
VLAN Traffic	Enabled on...
VLAN List	Selected: vlanSipExternal Available: vlanSipInternal

Resources

iRules	Enabled: Available: _sys_auth_ssl_cc_ldap, _sys_auth_krbdelegate, iRuleSnatOutbound
HTTP Class Profiles	Enabled: Available: httpclass
Default Pool	poolHa01
Default Persistence Profile	None
Fallback Persistence Profile	None

Creating a Virtual Server for TCP-Based Communications

7. Click `Finished`.

End

<verttabber>

Configuring SIP Server HA

The following table provides an overview of the main steps that are required to configure SIP Server HA in the Configuration Layer.

Objective	Related Procedures and Actions
1. Configure Host objects for primary and backup SIP Server applications.	Procedure: Configuring Host objects
2. Configure primary and backup SIP Server applications.	Procedure: Configuring primary and backup SIP Server applications

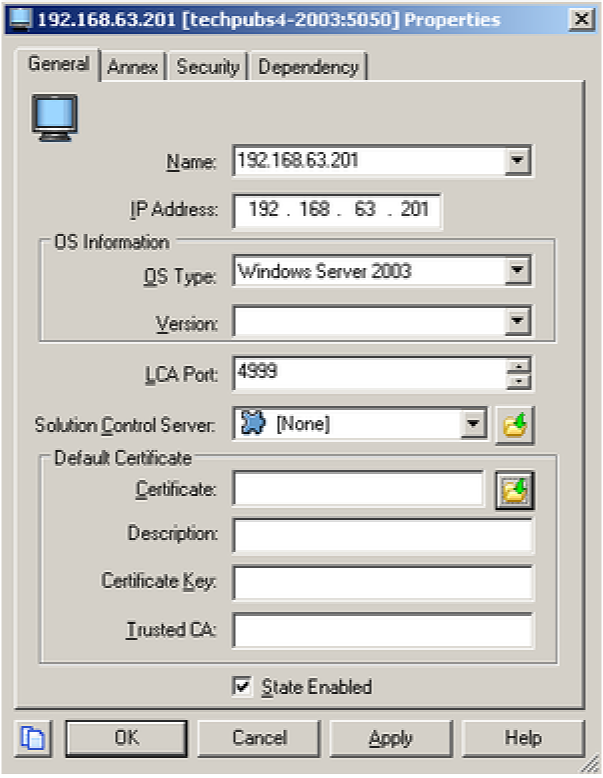
Configuring Host objects

Purpose

To configure a Host object for the computer on which a primary SIP Server application runs and to configure a Host object for the computer on which a backup SIP Server application runs.

Start

1. In Configuration Manager, right-click the Environment > Hosts folder and select New > Host.
2. On the General tab (see the following figure):
    a. Enter the name of the host for the primary SIP Server application—for example, 192.168.63.201.
    b. Enter the IP address of the host—for example, 192.168.63.201.
    c. Select the type of operating system from the OS Type drop-down list, and enter its version, if known.
    d. Enter the LCA port number or accept the default (4999) to be used by the Management Layer to control applications running on this host.



192.168.63.201 [techpubs4-2003:5050] Properties

Name: 192.168.63.201

IP Address: 192 . 168 . 63 . 201

OS Information

OS Type: Windows Server 2003

Version:

LCA Port: 4999

Solution Control Server: [None]

Default Certificate

Certificate:

Description:

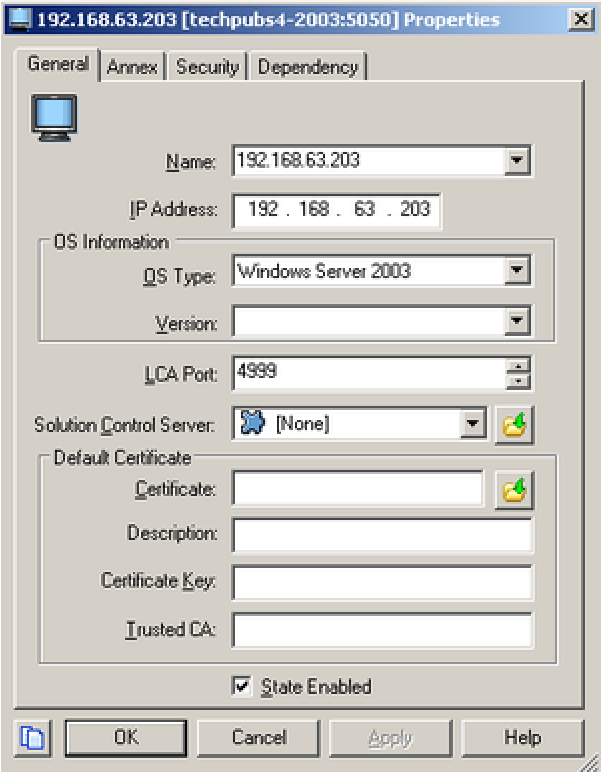
Certificate Key:

Trusted CA:

☑ State Enabled

Configuring a Host Object for a Primary SIP Server Application: Sample Configuration

3. Click `OK`.
4. Right-click the `Environment > Hosts` folder and select `New > Host`.
5. On the `General` tab:
    a. Enter the name of the host for the backup SIP Server application—for example, `192.168.63.203`.
    b. Enter the IP address of the host—for example, `192.168.63.203`.
    c. Select the type of operating system from the `OS Type` drop-down list, and enter its version, if known.
    d. Enter the LCA port number or accept the default (4999) to be used by the Management Layer to control applications running on this host.



192.168.63.203 [techpubs4-2003:5050] Properties

Name: 192.168.63.203

IP Address: 192 . 168 . 63 . 203

OS Type: Windows Server 2003

Version:

LCA Port: 4999

Solution Control Server: [None]

Certificate:

Description:

Certificate Key:

Trusted CA:

☑ State Enabled

Configuring a Host Object for a Backup SIP Server Application: Sample Configuration

6. Click OK.

End

Next Steps

- Configuring primary and backup SIP Server applications

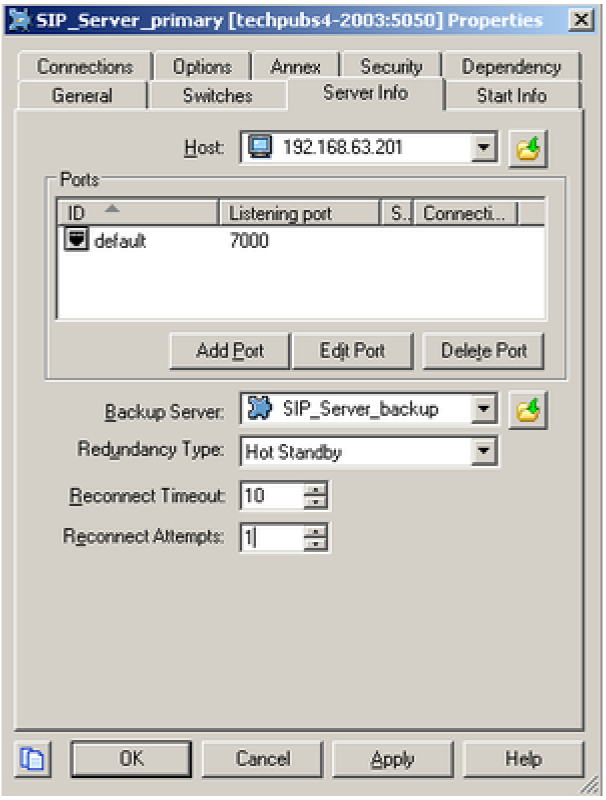
Configuring primary and backup SIP Server applications

Purpose

To configure primary and backup SIP Server applications.

Start

1. Open the primary SIP Server application.
2. Click the `Server Info` tab, and then specify the `Host` you created for the primary SIP Server application.



SIP_Server_primary [techpubs4-2003:5050] Properties

Host: 192.168.63.201

ID	Listening port	S..	Connecti...
default	7000		

Backup Server: SIP_Server_backup

Redundancy Type: Hot Standby

Reconnect Timeout: 10

Reconnect Attempts: 1

Configuring a Primary SIP Server Application: Sample Configuration

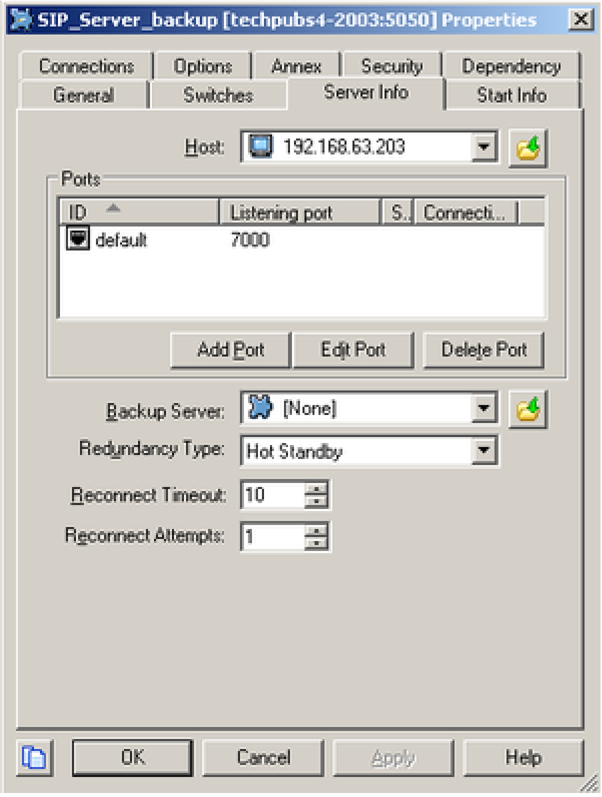
3. Click the `Options` tab. In the `TServer` section, set options as specified in the following table:

Configuration Options for a Primary SIP Server Application

Option Name	Option Value	Description
sip-address	String	Set this option to the value of the BIG-IP LTM Virtual IP address, which is the destination address for all incoming SIP messages. In our example, this would be `192.168.203.164`.
sip-port	<5060>	Set this option to the value of the port on which SIP Server listens to incoming SIP requests. The same port number is used for both TCP and UDP transports.
sip-interface	String	Set this option to the value of the host physical IP address where the primary SIP Server runs. In our example, this would be `192.168.63.201`.

Option Name	Option Value	Description
internal-registrar-enabled	`true, false`	Set this option to `true`.
internal-registrar-persistent	`true, false`	Set this option to `true`.
sip-hold-rfc3264	`true, false`	Set this option to `true`.

4. When you are finished, click `OK`.
5. Open the backup SIP Server application.
6. Click the `Server Info` tab, and then specify the `Host` you created for the backup SIP Server application.



SIP_Server_backup [techpubs4-2003:5050] Properties

Host: 192.168.63.203

Ports

ID	Listening port	S...	Connecti...
default	7000		

Backup Server: [None]

Redundancy Type: Hot Standby

Reconnect Timeout: 10

Reconnect Attempts: 1

Configuring a Backup SIP Server Application: Sample Configuration

7. Click the `Options` tab. In the `TServer` section, set options as specified in the following table:

Configuration Options for a Backup SIP Server Application

Option Name	Option Value	Description
sip-address	String	Set this option to the value of the BIG-IP LTM Virtual IP address, which is the destination address for all incoming SIP messages. In our example, this would be `192.168.203.164`.
sip-port	`<5060>`	Set this option to the value of the port on which SIP Server

Option Name	Option Value	Description
		listens to incoming SIP requests. The same port number is used for both TCP and UDP transports.
sip-interface	String	Set this option to the value of the host physical IP address where the backup SIP Server runs. In our example, this would be `192.168.63.203`.
internal-registrar-enabled	`true, false`	Set this option to `true`.
internal-registrar-persistent	`true, false`	Set this option to `true`.
sip-hold-rfc3264	`true, false`	Set this option to `true`.

8. When you are finished, click OK.

End