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Integration Reference Manual

SIP Server 8.1.1

12/29/2021

Table of Contents

SIP Server 8.1 Integration Reference

Welcome to the *SIP Server 8.1 Integration Reference*. This document introduces you to the concepts, terminology, and procedures related to integrating SIP Server with SIP endpoints, 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 *SIP Server Deployment Guide*.

SIP Endpoint Application Notes

- Polycom SIP Phones
- CounterPath Bria SIP Phones
- Yealink SIP Phones
- AudioCodes/Genesys SIP Phones
- Grandstream GXP SIP Phones
- Grandstream GXP1625/GCC1700 SIP

SBC Application Notes

- Sonus 1000/2000 SBC
- Sonus 5000 SBC
- AudioCodes Mediant SBC
- Oracle Enterprise SBC
- Oracle Acme Packet E-SBC Version 8.3.x
- CUBE SBC

Switch Integrations

- Siemens OpenScape Voice
- Cisco Unified Communications Manager
- Alcatel OXE
- BroadSoft BroadWorks
- RedSky E911 Manager

SIP Trunks Application Notes

- AT&T IP Toll Free with AudioCodes Mediant SBC
- AT&T IP Flexible Reach with AudioCodes Mediant SBC
- AT&T IPTF and IPXC with Sonus SBC GSX/PSX
- Colt SIP Trunk with AudioCodes Mediant SBC
- Telenor SIP Trunk with AudioCodes Mediant SBC
- Windstream SIP Trunk with AudioCodes Mediant SBC
- CenturyLink SIP Trunk with AudioCodes

Media Gateway Application Notes

AudioCodes Mediant Gateway

Cisco Media Gateway

Network Load Balancer Integrations

F5 Networks BIG-IP LTM

SIP Endpoint SDK

SIP Endpoint SDK

Siemens OpenScape Voice

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

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.

SIP Server and OpenScape Voice Integration 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, Media Server is signaled only by SIP Server. No direct SIP signaling occurs between OpenScape Voice and Media Server. For information about configuring SIP Server to use Media Server, 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 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.

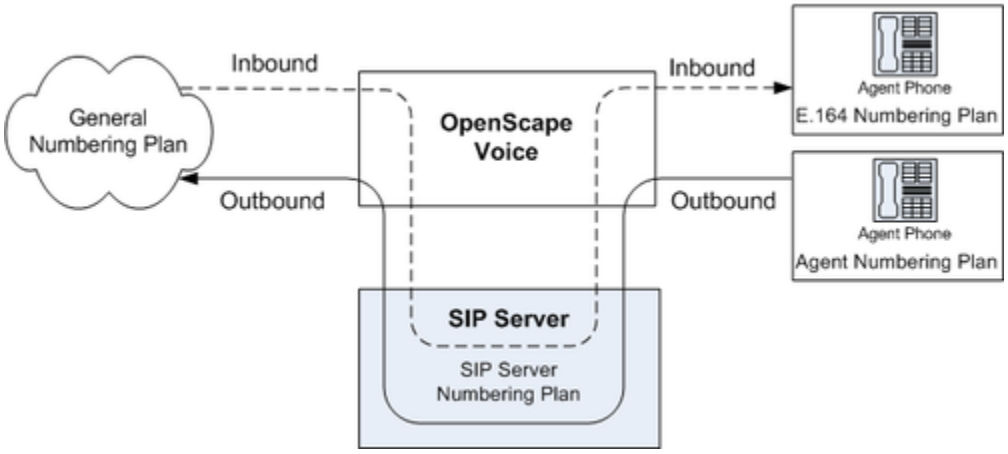
For GVP integration with SIP Server, the configuration must be performed on the SIP Server side, not on the OpenScape Voice side.

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

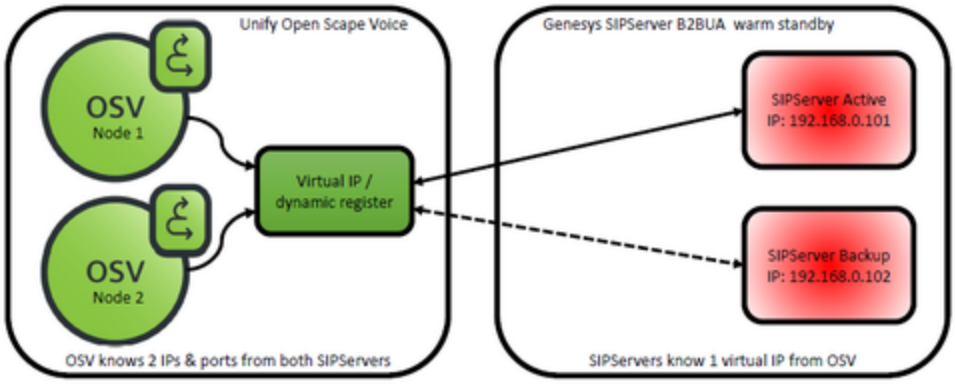
In the General Numbering Plan (the Numbering Plan that contains the gateways), the contact center is given a range of numbers for agents (assuming that the agents have direct lines) and Routing Points. Those numbers route directly to SIP Server, which then routes the calls accordingly.

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.

Other HA Deployment Architecture

Genesys supports an alternative HA deployment architecture for OpenScape Voice. The OpenScape Voice softswitch uses an internal mechanism to detect the active SIP Server. This architecture does not require Windows Network Load Balancing (NLB) or IP Address Takeover HA methods.



Alternative HA Deployment Architecture

Main configuration points for this deployment in OpenScape Voice are:

- Configure a SIP Endpoint for SIP Server.
- For the Endpoint Type, select **SIP Trunking** and type **Dynamic**.
- Associate primary and backup SIP Server IP addresses as aliases for a SIP Endpoint. For example:

Main configuration points for this deployment on the Genesys side are:

- Configure a Trunk DN to point to the SIP Endpoint configured in the OpenScape Voice softswitch. It must contain these configuration options:
  - **contact**=`sip:<OpenScape Voice IP address>:<SIP port>;transport=tcp`—The contact URI that SIP Server uses for communication with OpenScape Voice, where `<IP address>` is the IP address of the softswitch and `<SIP port>` is the SIP port number of the softswitch.
  - **force-register**=`sip:<SIP Endpoint name>@<OpenScape Voice IP address>:<SIP port>`—Enable trunk registration on the OpenScape SIP Endpoint.
  - **oos-check**— Specify how often (in seconds) SIP Server checks a device for out-of-service status.
  - **oos-force**—Specify when SIP Server places an unresponsive device into out-of-service state when the **oos-check** option is enabled.
- Configure two **sip-address** SIP Server application-level options to specify both primary and backup SIP Server IP addresses.

Deployment Limitation

After a SIP Server switchover, first-party call control (1pcc) operations with established calls might

cause the calls to be dropped. This limitation does not affect third-party call control (3pcc) operations.

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.

1. Check that OpenScape Voice is working
2. Configure Numbering Plans
3. Configure the Endpoint Profile
4. Configure the Endpoint
5. Configure Gateway Destinations
6. Configure Prefix Access Codes
7. Configure Destination Codes
8. Configure Agent Destinations
9. Configure Agent Access and Destination Codes
10. (Optional) Configure Click-to-Answer
11. (Optional) Configure emergency call routing

1. 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.

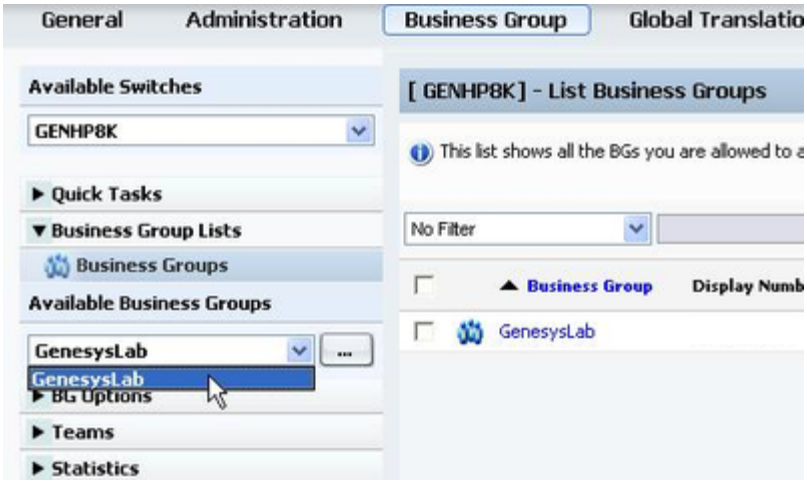
2. Configuring Numbering Plans

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.

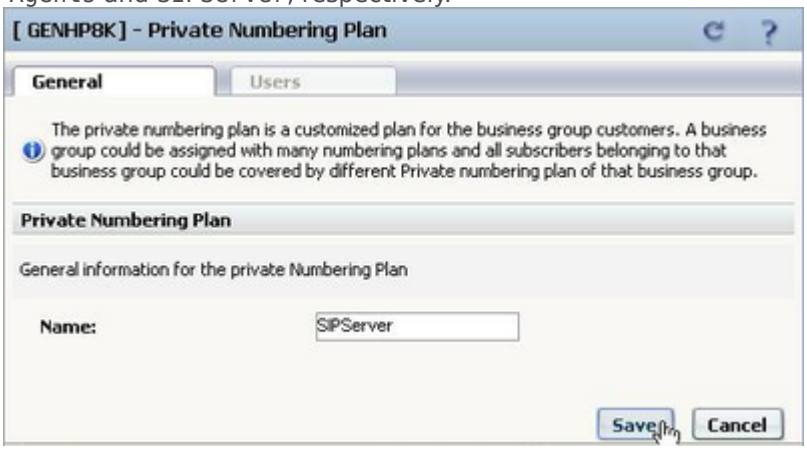


Selecting the Business Group

2. Click `Private Numbering Plans`.

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.



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. Configuring a SIP Server Endpoint Profile

Start

1. Click `Private Numbering Plan`, and then click the SIP Server Numbering Plan—for example, `SIPServer`.

Selecting the Numbering Plan

2. Click `Endpoint Management`, and then click `Endpoint Profiles`.

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.

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.

Enabling the Call Transfer Service

End

4. Configuring a SIP Server Endpoint

Start

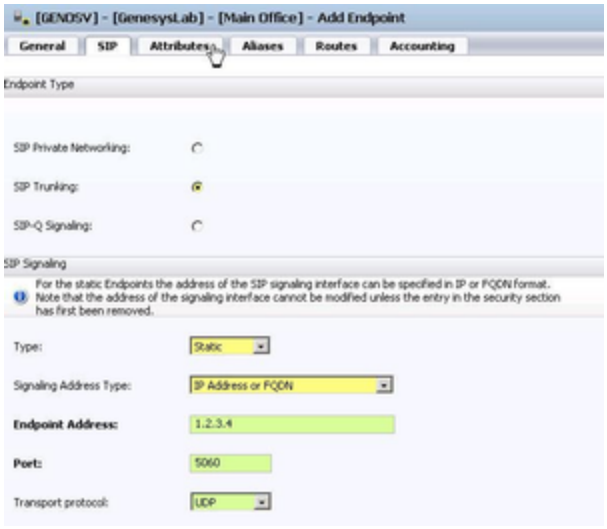
1. Click `Private Numbering Plan`, and then click the SIP Server Numbering Plan—for example, `SIPServer`.
2. Click `Endpoints`, and then click `Add`.

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.

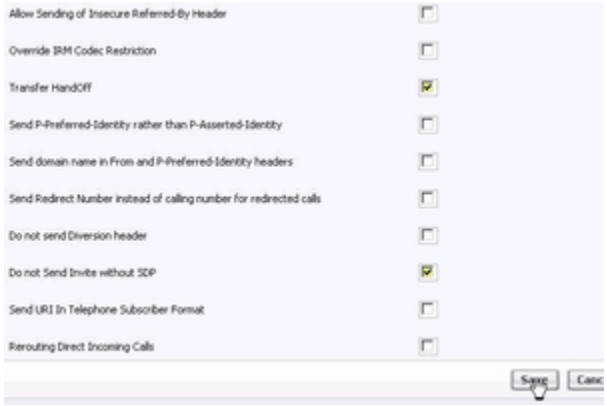
Configuring Endpoints: General Tab

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.



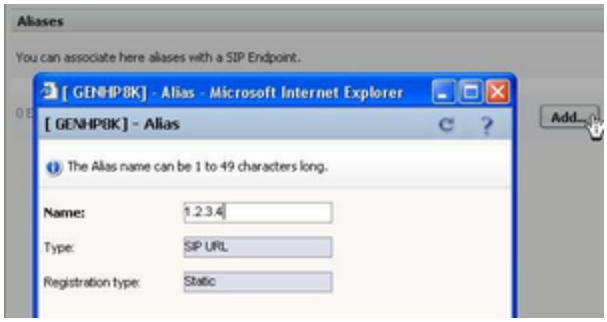
Configuring Endpoints: SIP Tab

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.



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.



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. 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`.

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.

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.

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. 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.

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:

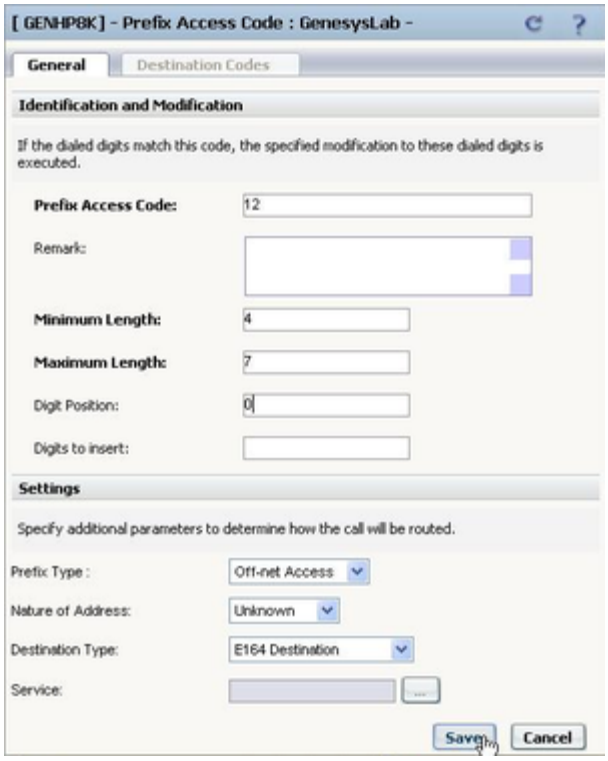
   a. In the `Prefix Access Code` text box, enter the digits you want to use to route calls to Subscribers.
   Note: For the SIP Server Numbering Plan, minimal modifications should be required. Dialed numbers should be modified before they reach SIP Server. This convention should be followed at all sites, 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`.

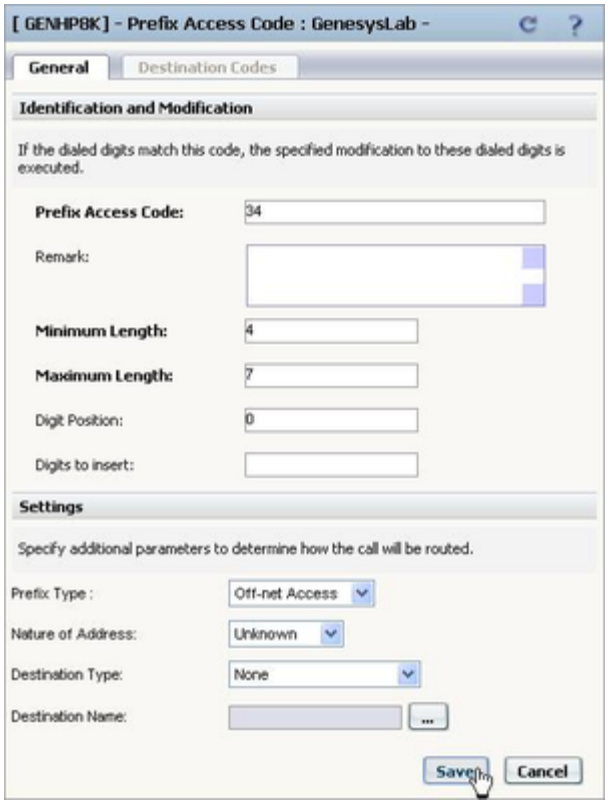
   d. Set the `Destination Type` text box to `E164 Destination`.

   e. Click Save.



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.



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:

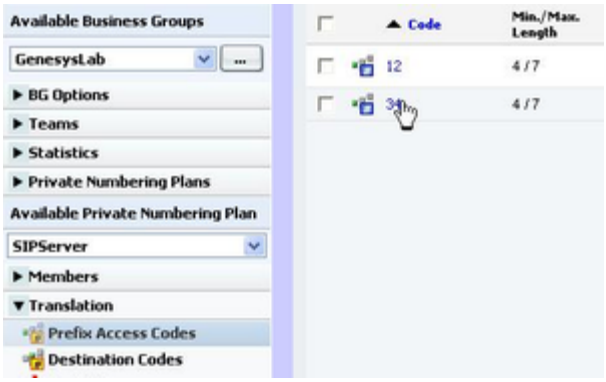
- Configuring SIP Server Destination Codes

7. Configuring SIP Server Destination Codes

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

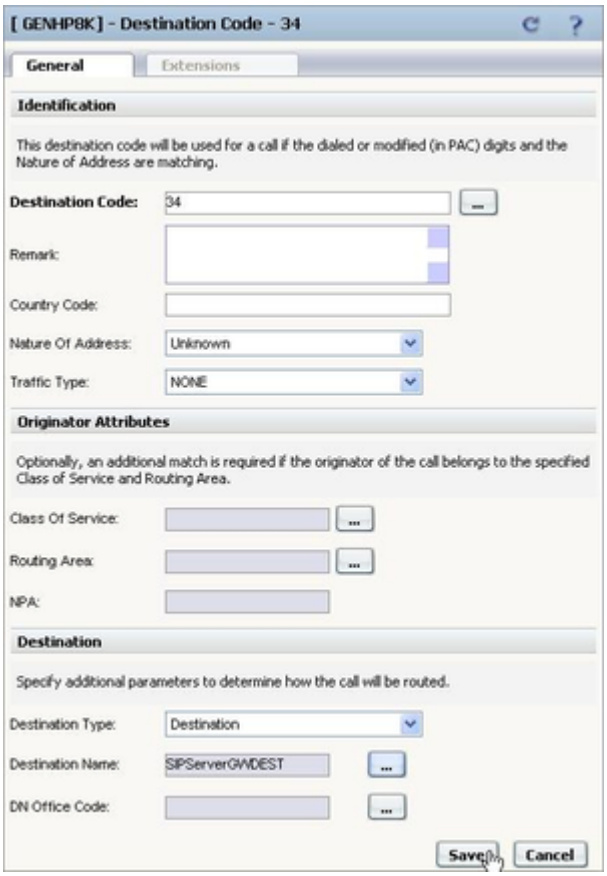
Start

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.



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.



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. 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.

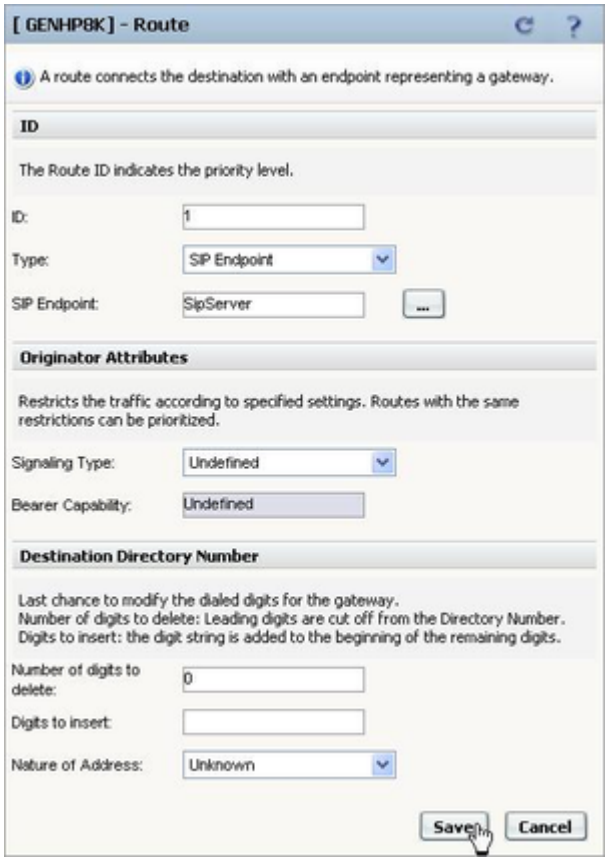
Configuring a SIP Server Destination in the Agent Numbering Plan

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.



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

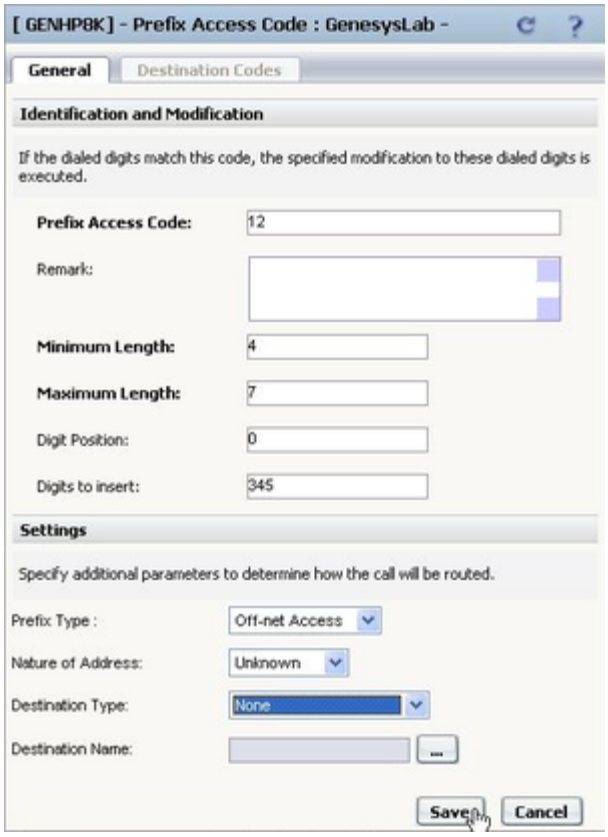
9. Configuring Agent Prefix Access Codes and Destination Codes

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

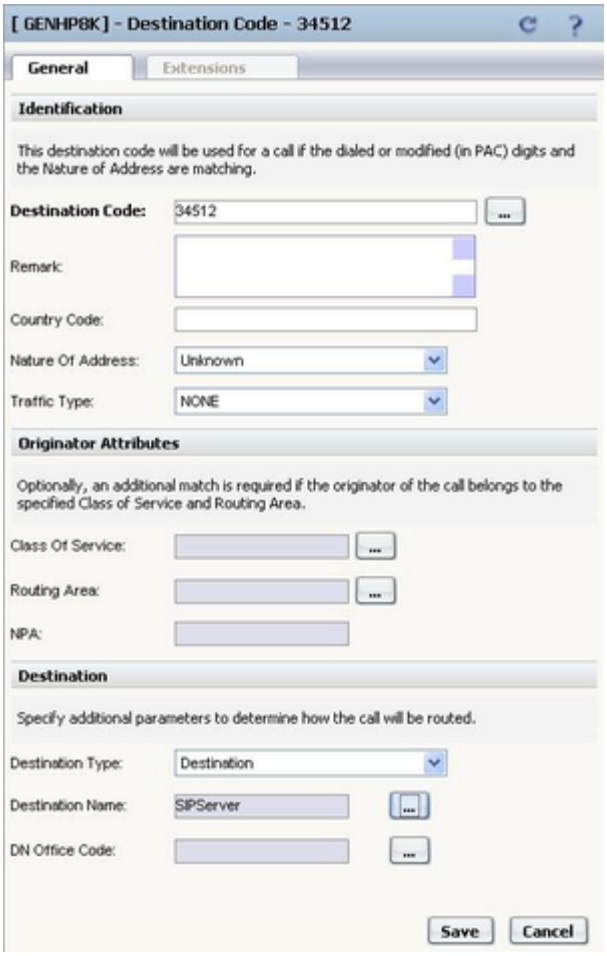
1. Click `Private Numbering Plan`, and then click the Agent Numbering Plan—for example, `Agents`.
2. Click `Translation`, click `Prefix Access Codes`, and then click `Add`.
3. In the `Prefix Access Code` dialog box, do the following:
    a. In the `Prefix Access Code` text box, enter the digits you that 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.



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.



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

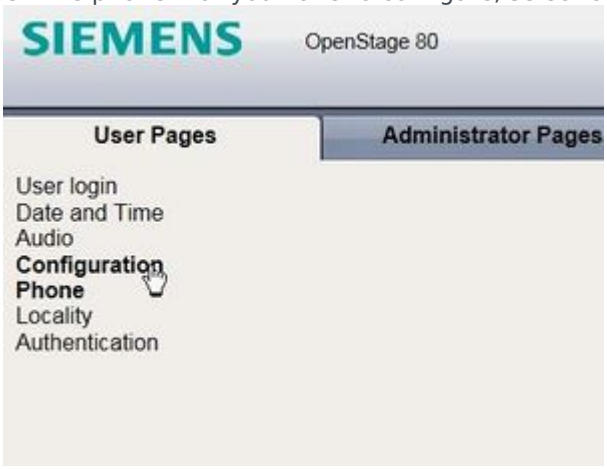
10. (Optional) Configure Click-to-Answer

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.

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`.

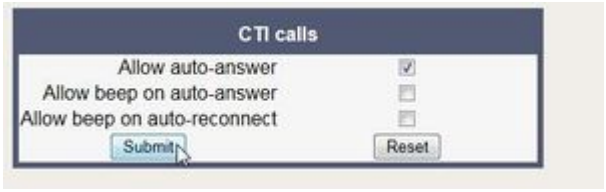


Selecting Configuration on the OpenStage Phone

2. Click `Incoming calls`, and then click `CTI calls`.

Configuring CTI Calls on the OpenStage Phone

3. Select the `Allow auto-answer` check box, and click `Submit`.



Submitting Allow auto-answer on the OpenStage Phone

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

End

11. (Optional) Configure emergency call routing

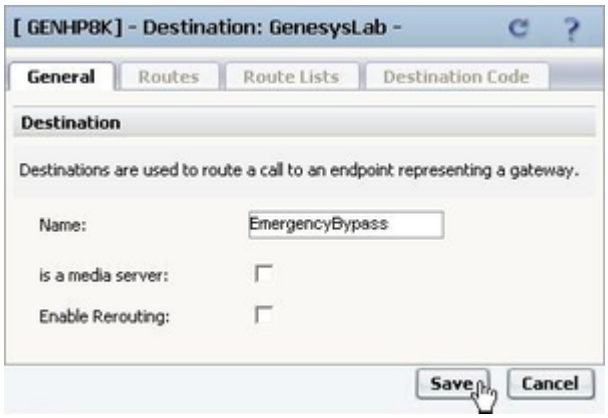
This configuration is not required for the integration to work, however, some might be required by local laws, or make the solution easier to configure.

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

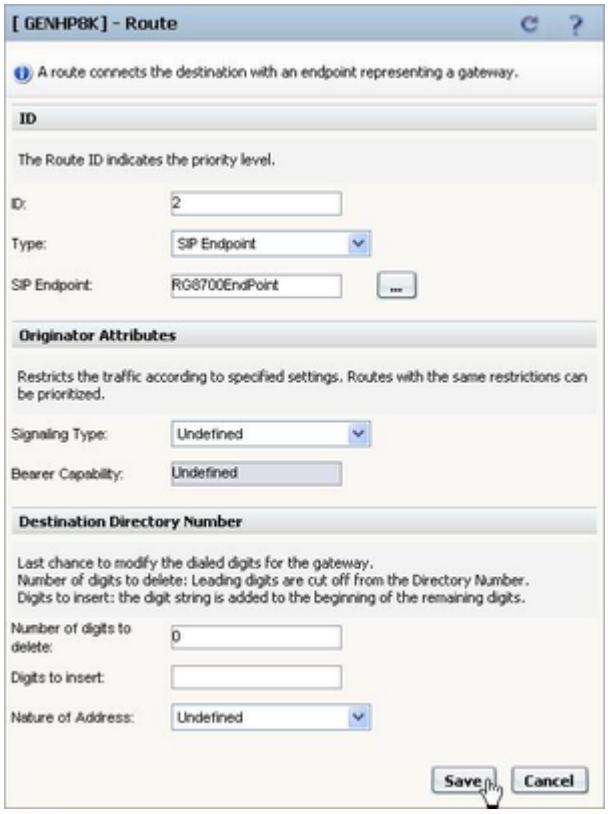
1. Log in to the HiPath Assistant, and navigate to the Business Group of the contact center that you want to configure—for example, GenesysLab.
2. Click Private Numbering Plan, and then click the Agent Numbering Plan.
3. Click Destinations and Routes, click Destinations, and then click Add.
4. In the 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.



Configuring a Destination for Emergency Call Routing

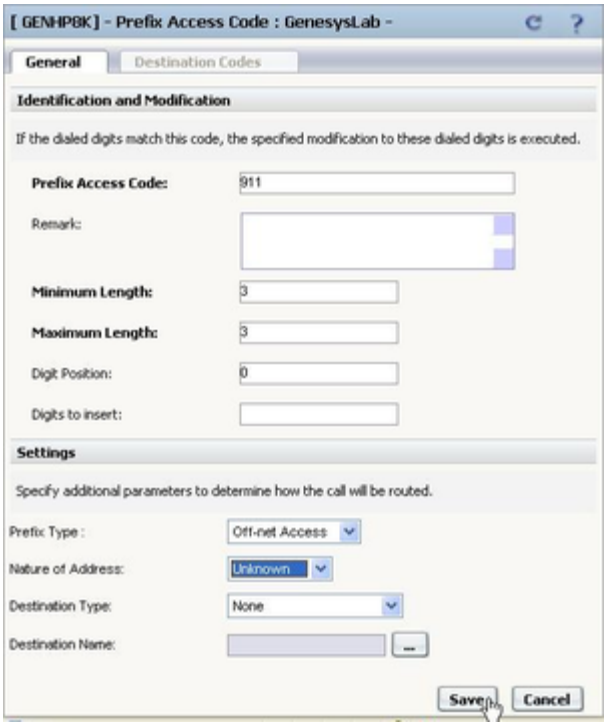
4. Click the Destination that you just created—for example, EmergencyBypass.
5. Click the Routes tab, and then click Add. In this step you are adding a route that goes to SIP Server. This is necessary in order to prevent calls from bypassing SIP Server while it is working.
6. In the Route dialog box, 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.



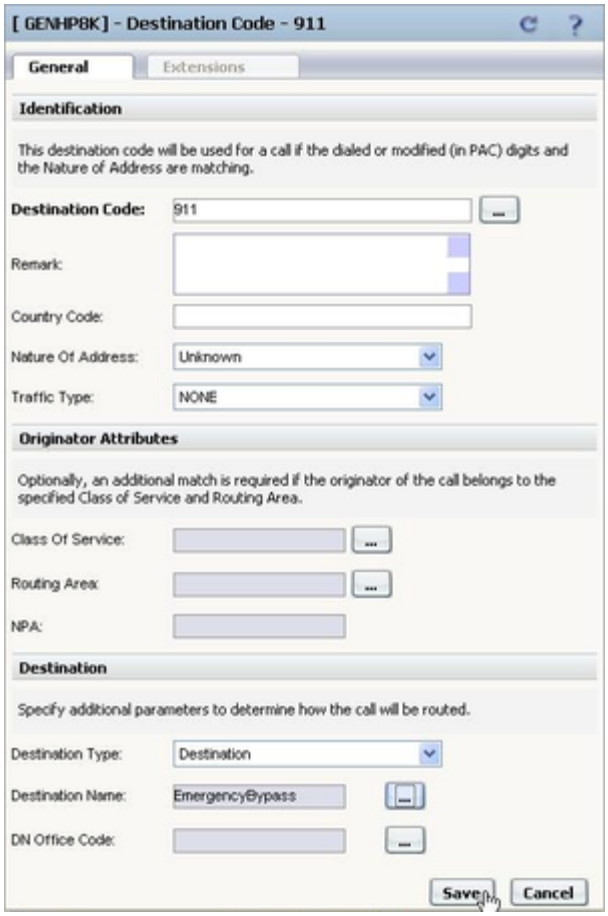
Configuring a Route for Emergency Call Routing

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.



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.



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

You configure DN objects for the OpenScape Voice in the Configuration Layer under the Switch object that is assigned to the appropriate SIP Server.

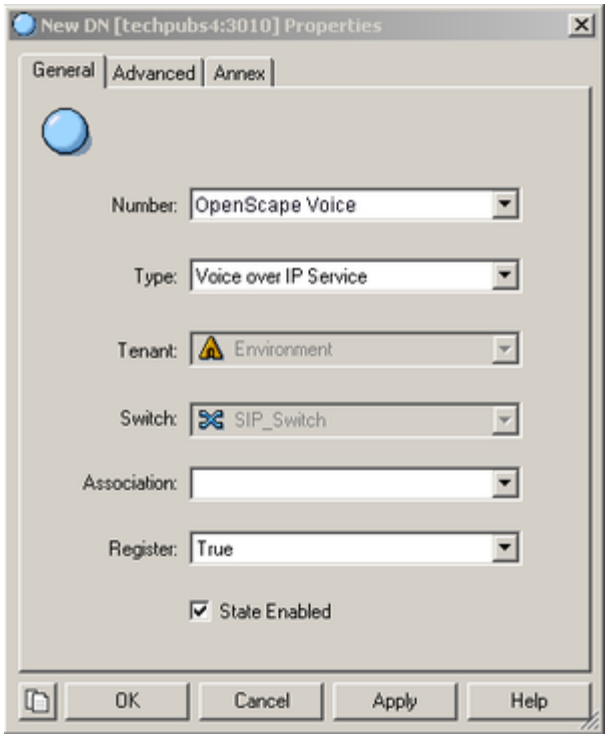
1. Configure a Voice over IP Service DN
2. Configure a Trunk DN
3. Configure Extension DNs
4. Configure Routing Point DNs

1. 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.



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

c. Click the Annex tab.

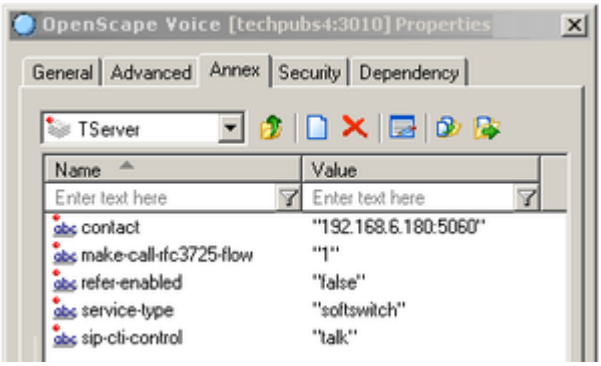
d. Create a section that is named `TServer`. In the `TServer` section, create options as specified in the following table.

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. **TLS configuration:** use the following format: `contact =<ipaddress>:<SIP port>;transport=tls` For more information about how to configure SIP Server, see the Transport Layer Security for SIP Traffic section in the *SIP Server Deployment Guide*.
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.

Option Name	Option Value	Description
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 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 Voice to support this

Option Name	Option Value	Description
		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.



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

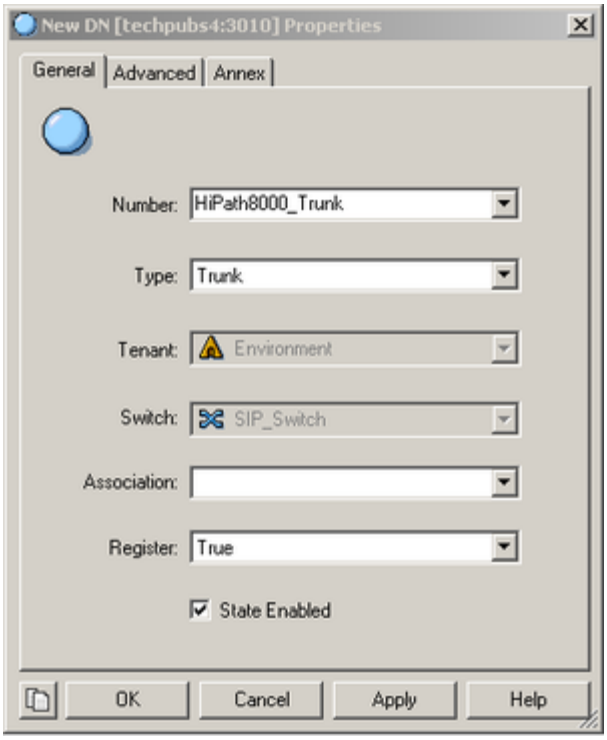
End

2. 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.

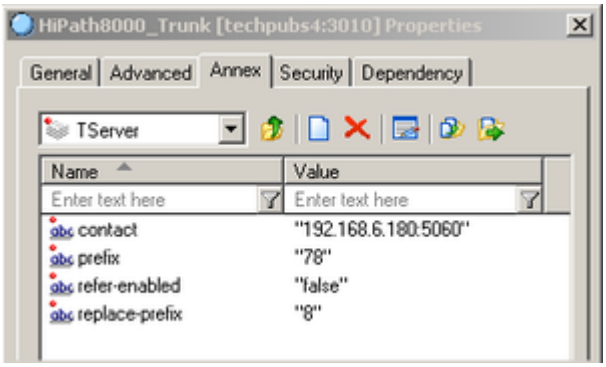


Creating a Trunk DN for OpenScape Voice: Sample Configuration

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 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

Option Name	Option Value	Description
		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).



Setting Options for a Trunk DN: Sample Configuration

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

End

3. 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.

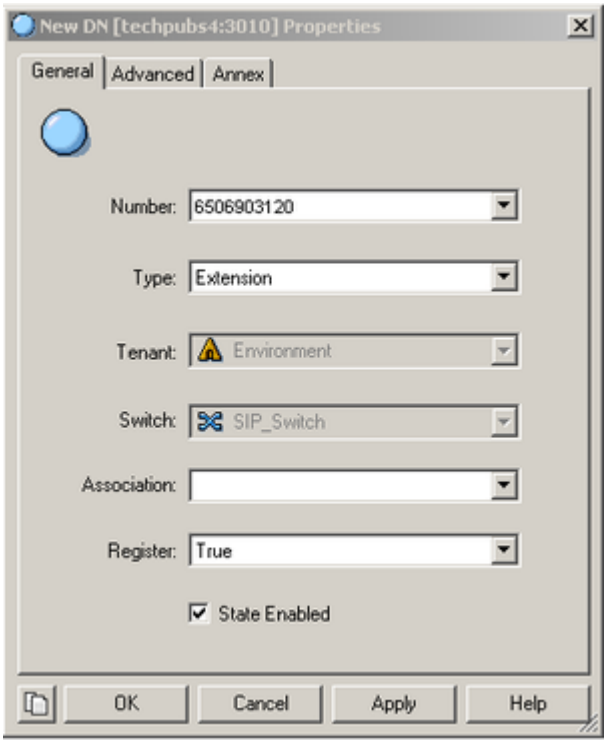
Start

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.



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. Configuring Routing Point DNs

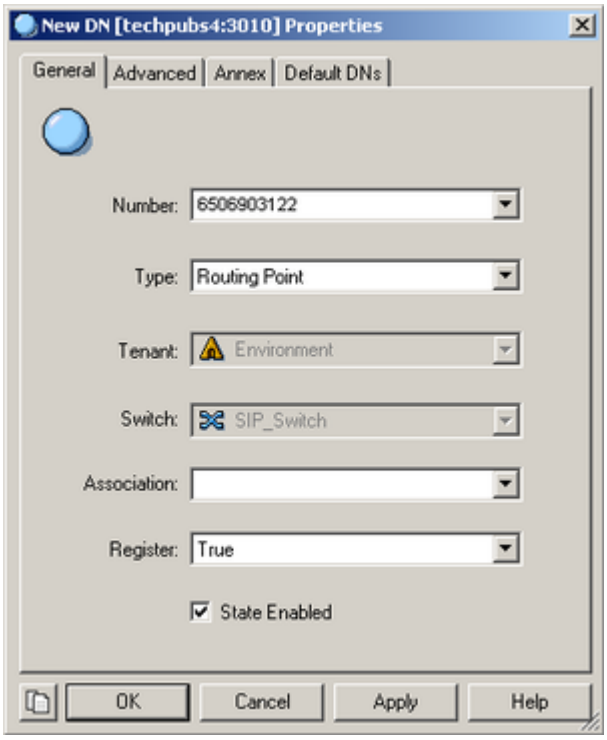
Purpose: To configure a DN of type Routing Point that is used to execute a routing strategy with Genesys URS. When SIP Server receives an INVITE request on a DN that is configured as a Routing Point, it sends an EventRouteRequest message to URS.

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.



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. See Configuring DN Objects for details.

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

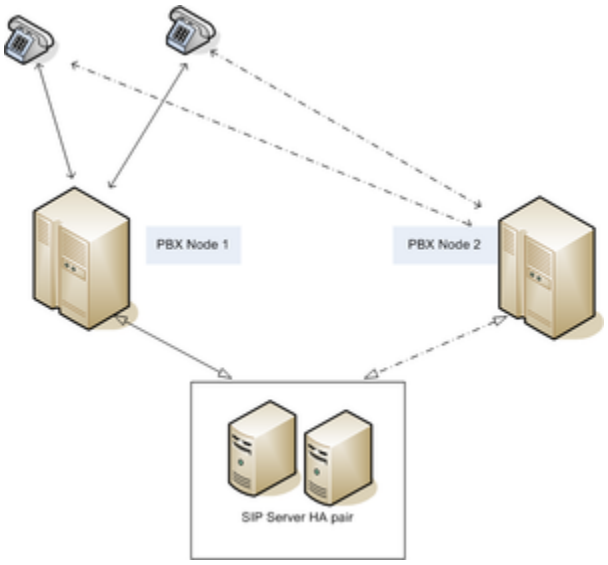
The Siemens OpenScape PBX can be configured to operate in a SIP Business Continuity configuration. There are two supported modes:

- 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

Configuring Split-Node deployment

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.

Configuration steps:

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:
   - `service-type`—Set this to `softswitch`.
   - `contact`—Specify the 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-zero value.

     See also the following configuration options for this softswitch DN.
3. Configure `Extension` DNs by completing the following procedure:
   - Procedure: Configuring Extension DNs
4. (Optional) Configure a `Trunk` DN for SIP Server to handle outbound calls 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.

Handling Call Forwarding Loop

In a SIP Server deployment with the Unified OpenScape Voice platform, you can configure SIP Server to support call forwarding from a back office phone to an agent phone, without creating an extra T-Library call in SIP Server.

Feature Configuration

Set the **sip-enhance-diversion** option to `true`.

sip-enhance-diversion

Setting: **TServer** section, Application level
Default Value: `false`
Valid Values: `false, true`
Changes take effect: For the next call

Specifies how SIP Server processes an INVITE message based on the value of the Diversion header. If the option is set to `true` and the Diversion header references the call forwarding party, SIP Server rejects that INVITE, waits until that rejection is propagated by the PBX back to the SIP Server in the original SIP dialog, and sends a new INVITE message to a forwarding destination.

Feature Limitations

This feature does not apply to the scenario in which an external number is forwarding multiple calls back to SIP Server simultaneously.

Cisco Unified Communications Manager

Genesys SIP Server is a key integration point between the Genesys Customer Experience Platform and Cisco Unified Communications Manager (CUCM). The integration supports several capabilities:

- **In-Front Qualification & Parking** provides centralized queuing & qualification (basic IVR), before routing calls to an agent. (See the *T-Server for Cisco Unified Communications Manager Deployment Guide* for details.)
- **Advanced Self Service** integrates Genesys Voice Platform with advanced VXML, ASR, TTS (and more) to CUCM.
- **Manage Agent-related Calls** allows agents to utilize CUCM as the underlying telephony platform, while SIP Server manages calls & agent state. This is an effective alternative to the standard JTAPI-based T-Server integration.
- **Recording Integration** integrates with Genesys Interaction Recording or the Genesys Recording Connector.

You can deploy any combination of these capabilities. For instance, In-Front Qualification & Parking could be deployed in conjunction with the JTAPI-based T-Server. Or a pure SIP deployment could utilize In-Front Qualification & Parking, and SIP-based agent management.

The configuration for Genesys SIP Server, as it relates to the CUCM integration, is largely the same for most of these capabilities.

It contains the following sections:

Note: The instructions on the following pages in this topic assume that both CUCM 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.

See supported CUCM versions in the *Supported Media Interfaces Guide*.

Managing Agents over SIP

Overview of SIP-based Integration for Managing Agents

Genesys SIP Server supports operation as a type of “Application Server.” Effectively, agents will have their telephone service (dial tone) provide by a phone which is part of a Cisco UCM deployment. SIP Server will serve as the application server which manages contact center calls to and from the agents. This integration utilizes a SIP Trunk between the two components, and typically involves an exchange of presence information for each DN. The presence information informs SIP Server when an agent is involved in a non-contact center call, and this status is taken into account when the Genesys URS/ORS routing components select an agent (such as not routing a contact center call to an agent who is already busy).

This SIP-based integration offers several key benefits:

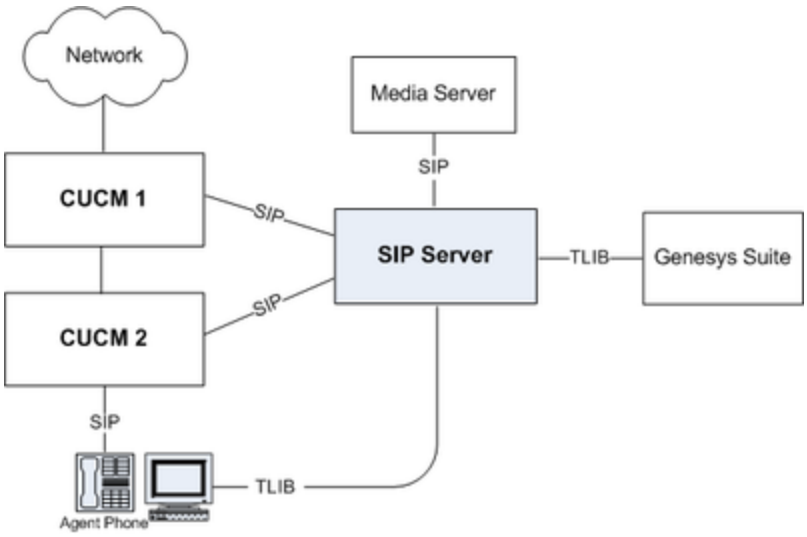
- Standards based integration using the SIP protocol.
- Fewer components—SIP Server is often deployed for purposes of recording or qualification & parking, and extending it to manage agents requires nothing more than configuration updates. This benefit is also valuable in case it is not practical to deploy a dedicated JTAPI-based T-Server (such as when Cisco UCM is geographically distant from the Genesys infrastructure).
- Access to unique features provided by SIP Server—such as full support of Genesys Interaction Recording, Personal Greetings for agents, support of Genesys SIP Voicemail.

The SIP-based integration does have some limitations which will be visible to the agents:

- First-Party Call Control (1PCC) Limitations—Agents are not allowed to use their phone for initiating mid-call changes like hold/retrieve, conference or transfer. These operations must be initiated using the agent desktop application.
- Third-Party Call Control (3PCC) Limitations—SIP Server does not have complete control over the phone, so some 3PCC operations are not supported; the main 3PCC operation which is missing is Click-to-answer (the agent must manually answer each call, or they must set the phone to “auto answer” each incoming call).

Deployment Architecture

SIP Server is integrated with CUCM via SIP trunks.The figure below depicts a sample deployment architecture of SIP Server with CUCM.



This sample call flow describes the steps for an incoming call handled by the CUCM–SIP Server integration solution:

1. Genesys SIP Server is monitoring the status of each agent's phone.
2. An incoming customer call arrives at CUCM.
3. CUCM delivers the call to SIP Server via a configured SIP Trunk.
4. Genesys SIP Server (typically) uses a routing strategy on URS/ORS which selects a qualified agent who is in the Ready state, or places the call in queue until an agent is available. Direct calls to an agent are also possible if the agent has a Direct Inward Dial (DID) phone number.
5. SIP Server delivers the call to CUCM via a SIP trunk (if multiple SIP trunks are defined, the selection is configurable as round-robin or primary/backup).
6. CUCM delivers the call to the agent's phone and the agent answers the call.
   The agent will use their desktop application (connected to SIP Server) for all mid-call operations, such as transfer, conference, supervision, recording control, and hold/retrieve.
7. The call disconnects when the caller or the agent hangs up.

Call Flows

Subscription

Genesys SIP Server integration with CUCM relies on the SUBSCRIBE/NOTIFY feature that is supported by CUCM.

- At startup, SIP Server sends subscription messages for a device or a list of configured devices to be notified about the endpoints status change.
- CUCM provides NOTIFY messages to SIP Server according to the endpoints status. As soon as an endpoint registers with CUCM, CUCM sends a NOTIFY message to SIP Server with the status reported as active.

Private Calls

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

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

Contact Center Calls

In the same way that you can set up a CUCM dialing plan to bypass SIP Server for private calls, you can write rules governing how CUCM connects contact center calls (typically, calls to the service number of the company) to SIP Server.

After connection, 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 CUCM for the selected endpoint.

Finally, CUCM 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 CUCM (customer leg) with the outbound leg to CUCM (agent leg).

By staying in the signaling path, SIP Server detects any change in call status, and generates 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 (except the answer call operation). This includes, but is not limited to, hold, transfer, and conference requests.

If a Network/Media Gateway is directly connected to SIP Server, then contact center calls are first received by SIP Server. The call flow for routing the call is very similar to the flow described above, except that there is only one call leg in CUCM.

Conferences

SIP Server supports conferences for agents on CUCM. A conference is initiated by a T-Library client (for example, Workspace Desktop). SIP Server can be configured to use either Media Server or other third-party MCUs to provide the conferencing feature.

Call Supervision

Supervisor features—such as, Silent-Monitoring and Barge-In—are also supported for this integration with CUCM. Supervisor functionality is supported via a T-Library interface. SIP Server includes a supervisor in a call between a customer and an agent (conference) and signals Media Server to either keep the supervisor media leg open for two-way media (sendrecv - Barge-In) or one way (for Silent Monitoring).

Presence

Genesys needs to be aware of non-contact center calls (and a general phone status), to make the best contact center routing decisions. SIP Server SUBSCRIBEs towards CUCM for presence information per RFC 3856, and CUCM NOTIFYs of the device state. SIP Server maps this presence information to the agent/device state in the Genesys T-Library model.

SIP Server transforms notifications about presence state changes into the Agent State updates, according to the following rules:

- For the initial NOTIFY with a basic status Open:
  - SIP Server uses the initial NOTIFY message to generate EventAgentLogin and EventAgentReady messages.
- For subsequent NOTIFY messages with the basic status Open:
  - SIP Server checks whether an agent is logged in. If not, SIP Server sends an event that the agent has logged in (EventAgentLogin).
  - SIP Server checks whether any activity is indicated in presence notification. If not, and if the agent is in the Not Ready state, SIP Server sends an event that the agent is Ready (EventAgentReady).
  - If an activity is indicated in the presence notification, and if the agent is Ready, SIP Server sends an event that the agent is not Ready (EventAgentNotReady), attaching the activity from the presence notification as a ReasonCode.
- For presence notification with the basic status Closed is received:
  - SIP Server checks whether the agent is logged in. If yes, SIP Server sends an event that the agent has logged out (EventAgentLogout).

In NOTIFY messages from CUCM, when an agent is handling a direct in-dial call, SIP Server expects a SIP NOTIFY message with the basic status Open and activities as "on-the-phone". Note that the Cisco extension must be set to allow for call waiting. If the extension is not configured to allow call waiting, then whenever the agent is handling a direct in-dial call, Cisco will send a SIP NOTIFY message with the basic status Closed and activities as "on-the-phone".

Agent States through SIP Presence and T-Library Interface

While working with CUCM, SIP Server could modify agent states based on two different inputs:

- NOTIFY message within a SUBSCRIBE dialog
- T-Library client requests from the agent desktop

Such dualism could lead to conflict unless some priority rules are applied. SIP Server always considers a message that brings an agent in the Not Ready/Logout state of the higher priority over a message that brings an agent in the Logging/Ready state coming from a different interface.

Endpoint Support

The SIP-based integration between SIP Server and CUCM works for all known endpoints including Cisco IP Communicator and Cisco Hardphones (with a minor disclaimer that only a select few phones were actually tested, but all had positive results). With IP communicator CUCM sends NOTIFY with the Open state when it starts, so SIP Server can set an agent linked to this extension in the Login/Ready

state. At proper exit of IP Communicator, CUCM sends NOTIFY with the terminated state, so SIP Server sets an agent in the Logout state.

Supported Versions

For supported CUCM versions, see the Genesys Supported Media Interfaces Guide Supported Infrastructure page.

Integration Task Summary

To integrate SIP Server with CUCM, complete the following procedures:

1. Configure CUCM.
2. Configure DN objects for CUCM.

Configuring CUCM

This is an overview of the main steps that are required to configure CUCM. Refer to the Cisco Unified Communcations Manager documentation for standard configuration procedures.

1. Configure Cisco Phones to point to CUCM. Genesys SIP Server currently supports this integration with only SIP Phones that are compatible with CUCM.
2. Ensure that the Service Parameter Configuration for the **[server-name]**, the **Clusterwide parameters (System – Presence)** for **Default Inter-Presence Group Subscription** is set to `Allow Subscription`.
3. Ensure that the **SIP Trunk Security Profile > Outgoing Transport Type** is set to `UDP`, and the **Accept Presence Subscription** checkbox is selected.
4. Build a SIP Trunk to point it to the SIP Server host using the SIP Trunk Security Profile configured in the previous step, with an appropriate **Inbound Calling Search Space** and **SUBSCRIBE Calling Search Space**.

Note: If a Presence Group is added to segregate resources, the SIP trunk and all appropriate resources must be in the same Presence Group, and resources must also share the same SUBSCRIBE Calling Search Space.

Configuring DN Objects

You configure Genesys DN objects to represent CUCM in the Configuration Layer under the `Switch` object that is assigned to the appropriate SIP Server.

1. Configure a Voice over IP Service DN.

For each CUCM SIP Trunk present in the cluster, create a DN object of type Voice over IP Service. Configure the following options in the TServer section for such object.

Option Name	Option Value	Description
`contact`	SIP URI	Specify the contact URI that SIP Server uses for communication with CUCM.
`dual-dialog-enabled`	`false`	Set this option to `false` to instruct SIP Server not to create new a SIP dialog when making a consultation call.
`enable-agentlogin-presence`	`true`	Set this option to `true` for SIP Server to provide accurate information about agent states.
`make-call-rfc3725-flow`	1	Set this option to 1, so SIP Server selects the SIP call flow number 1 (described in RFC 3725) for a call that is initiated by a `TMakeCall` request.
`oos-check`		Specify how often (in seconds) SIP Server checks a device for out-of-service status.
`oos-force`		Specify the time interval (in seconds) that SIP Server waits before placing a device that does not respond in out-of-service state when the `oos-check` option is enabled.
`refer-enabled`	`false`	Set this option to `false` for SIP Server to use a `re-INVITE` request method when contacting CUCM.
`service-type`	`softswitch`	Set this option to `softswitch`.

2. Configure a Trunk DN.

Create a DN of type Trunk with the name, for example, `CUCMTrunk`. 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`	SIP URI	Specify the contact URI that SIP Server uses for communication with CUCM. If there is a cluster CUCM deployment, specify the address of any CUCM.
`contact-backup`	SIP URI	(For CUCM cluster only) May contain a comma-separated list of addresses of other members of the CUCM cluster. In case of a single node CUCM deployment, this option must not be configured.
`oos-check`		Specify how often (in seconds) SIP Server checks a device for out-of-service status.
`oos-force`		Specify the time interval (in seconds) that SIP Server waits before placing a device that does not respond in out-of-service state when the `oos-check` option is enabled.
`subscribe-presence-domain`	Any valid domain	Specify the subscription domain information for the Trunk DN. This option value is used with the DN name to form the SUBSCRIBE request URI and the To header.
`subscribe-presence-from`	SIP URI	Specify the subscription endpoint information. This option value is used to form the `From` header in the SUBSCRIBE request to CUCM. This must be the same address as the SIP Server host that is part of the SIP trunk configuration in CUCM.
`subscribe-presence-expire`		Specify the subscription renewal interval (in seconds).

3. Configure Extension DNs.

Enable a presence subscription for the Extension DNs by specifying the option `subscribe-presence`. This option's value must be the name of the Trunk DN that contains subscription parameters—for

example, `CUCMTrunk`.

Extension DN objects...

- ...must not have any other options configured.
- ...must not have the option `contact` configured.
- ...must be created for all DNs used as agent phones.

4. Create Agent IDs.

Create Agent IDs for all Extension DNs with the subscription enabled. The Agent ID must be same as the ID of the DN of type Extension.

Cisco Media Gateway

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

Note: The instructions in this section assume that the Cisco Media Gateway is fully functional.

SIP Server and Cisco Media Gateway Integration 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.

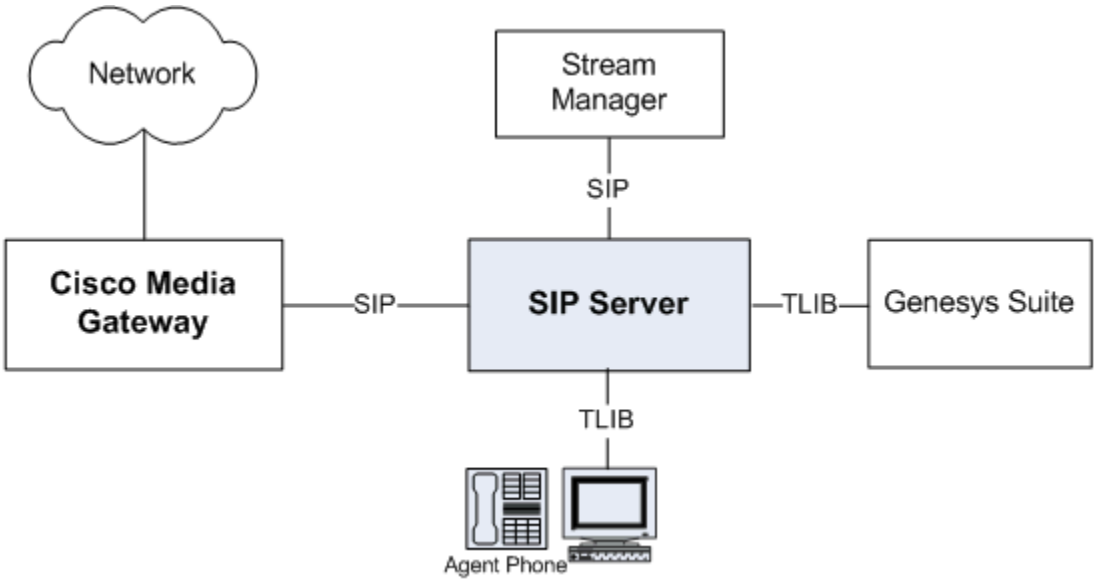
The following Cisco IOS Software versions were tested:

- 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 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

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:

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.

SIP Server and BIG-IP LTM Integration Overview

The SIP Server and BIG-IP LTM integration solution described in this document 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.

Integration Solution Notes

- UDP, TCP, or TLS can be used as the transport for SIP signaling.
- Up-front load balancing via Network SIP Server or other device could be implemented, but is not described in this document.
- Configuration of BIG-IP High Availability is not described in this document and has not been validated with SIP Server.
- BIG-IP LTM can be configured in a more complex load-balancing role. This is beyond the scope of this document.

Supported Versions

The integration solution described in this document supports the following software versions:

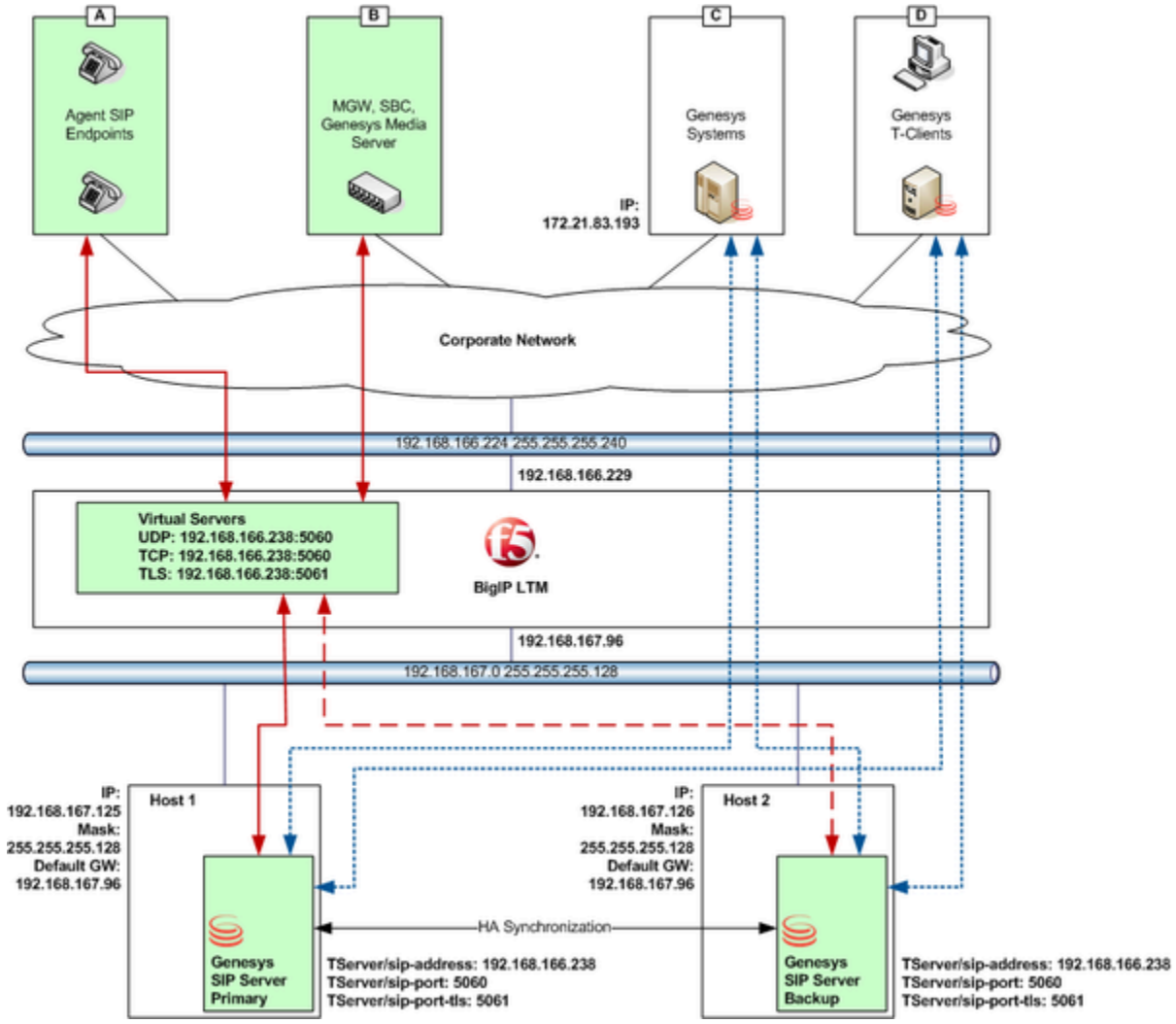
Software	Version
Genesys SIP Server	8.1.100.98 and later 8.1.001.22 and later (does not support TLS/Active-Active RM)
Genesys Voice Platform	8.1.7 and later (required for TLS/Active-Active RM)
F5 BIG-IP Local Traffic Manager	BIG-IP v11.x (v11.1.0 Build 1943.0 Final or later), BIG-IP v12.x

Deployment Architecture

The figure below depicts the architecture of a sample deployment 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.

Agent SIP Endpoints Group

The Agent SIP Endpoints 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 Media Server (Resource Manager and MCP). 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 Systems Group

The Genesys Systems Group includes Genesys Configuration Server, Genesys Message Servers and other SIP Servers in multi-site deployments (group C in the preceding figure). SIP Server maintains permanent TCP/IP connections with members of the Genesys Systems Group. Connections with members of the Genesys Systems Group are initiated from a SIP Server physical IP address. Responses from them are directed to the SIP Server physical IP address. Note that missing definitions of Genesys Systems Group members in the SNAT exclusion group result in SNAT being applied to the outbound packets. In this case, the Genesys Systems are unaware of physical IP addresses of the SIP Servers and, instead, receive packets from the Virtual IP address; this complicates troubleshooting.

IT infrastructure servers, such as DNS, LDAP, log server, or software repository, are typically included into this group for similar reasons.

This group requires that:

- No SNAT is applied to outbound packets sent to the members of the Genesys Systems Group.

Note: In this deployment architecture, HA synchronization traffic between primary and backup SIP Servers does not traverse the BIG-IP LTM, and SNAT control is not applicable to this type of connections (the primary and backup SIP Servers do not belong to the Group).

Genesys T-Client Group

The Genesys T-Client Group includes clients initiating and maintaining connections with the SIP Server (group D in the preceding figure). These are Stat Servers, URSs, ORSs, T-Library desktops like

Interaction Workspace, and other T-Library clients. For this group, connection is initiated by the members of the group toward the SIP Server (primary or backup) physical IP address, and SIP Server accepts the connection. In this case, SNAT is not applied to the outbound packets, and SNAT control does not apply to this type of connections; from the perspective of group members, both SIP Servers send packets from the physical IP address of SIP Server.

IT infrastructure systems, such as monitoring servers querying hosts, or the workstation of the system administrator, belong to this group.

This group requires that:

- The primary or backup SIP Server is accessible via its physical IP address.

Note: SNAT pool must not be activated as the configuration element of Virtual Server. SNAT must not be applied to SIP messages sent by clients to SIP Server. Enabling SNAT for these messages results in a broken call flow, as in some circumstances the SNAT is not aligned with SIP Protocol.

Integration

Network and Host Prerequisites

- BIG-IP LTM and SIP Server hosts must be on the same network (broadcast domain, VLAN).
- The Self-IP (or Floating IP address) of BIG-IP LTM must be configured as the Default Gateway in routing tables of SIP Server hosts.
- All traffic between SIP Server hosts and any other SIP peers in the deployment, represented by groups A and B such as Agent SIP Endpoints, MGW, SBC, and RMs, must traverse BIG-IP LTM.
- Firewall settings of SIP Server hosts must enable sending of the ICMP Port Unreachable messages towards the other SIP peers of the deployment represented by groups A and B, such as Agent SIP Endpoints, MGWs, SBCs, and RMs.
- Firewall settings of SIP Server hosts must enable receiving of the ICMP Echo Requests from the Self-IP (or Floating IP address) of BIG-IP LTM.

Required Provisioning

- The IP address of the Virtual Server configured on BIG-IP LTM for SIP communications with SIP Server over UDP, TCP, or TLS must match the value of the `sip-address` option configured in the TServer section of the SIP Server application.
- The port of the Virtual Server configured on BIG-IP LTM for SIP communications with SIP Server over UDP and TCP must match the value of the `sip-port` option configured in the TServer section of the SIP Server application.
- The port of the Virtual Server configured on BIG-IP LTM for SIP communications with SIP Server over TLS must match the value of the `sip-port-tls` option configured in the TServer section of the SIP Server application.

Integration Steps

To integrate SIP Server with the BIG-IP LTM, complete the following procedures:

1. Configure SIP Server HA.
2. Configure BIG-IP LTM.
3. Configure Resource Manager HA.

Note: Genesys strongly advises to use Resource Manager in Active-Active high-availability. Refer to the *Genesys Voice Platform Integration* section in the ''Framework 8.1 SIP Server Deployment Guide''.

Configuring SIP Server HA

To configure SIP Server HA, complete the following steps:

1. Configure Host objects for primary and backup SIP Server applications.
2. Configure 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:
   a. Enter the name of the host for the primary SIP Server application—for example, `192.168.167.125`.
   b. Enter the IP address of the host—for example, `192.168.167.125`.
   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.
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, `192.168.167.126`.
   b. Enter the IP address of the host—for example, `192.168.167.126`.
   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.
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.

Note: To configure SIP Server to use TLS encryption, refer to the Transport Layer Security for SIP Traffic section in the *Framework 8.1 SIP Server Deployment Guide*.

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—for example, `192.168.167.125`.
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 `192.168.166.238`.
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. The port of the Virtual Server configured on BIG-IP LTM for SIP communications with SIP Server over UDP and TCP must match this value. In our example, this would be `5060`.
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 `192.168.167.125`.
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—for example, 192.168.167.126.
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 192.168.166.238.
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. The port of the Virtual Server configured on BIG-IP LTM for SIP communications with SIP Server over UDP and TCP must match this value. In our example, this would be 5060.
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 192.168.167.126.
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

Configuring BIG-IP LTM

The following page 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 1.6). VLANs are used to logically associate Self IP interfaces with physical interfaces on the BIG-IP LTM.

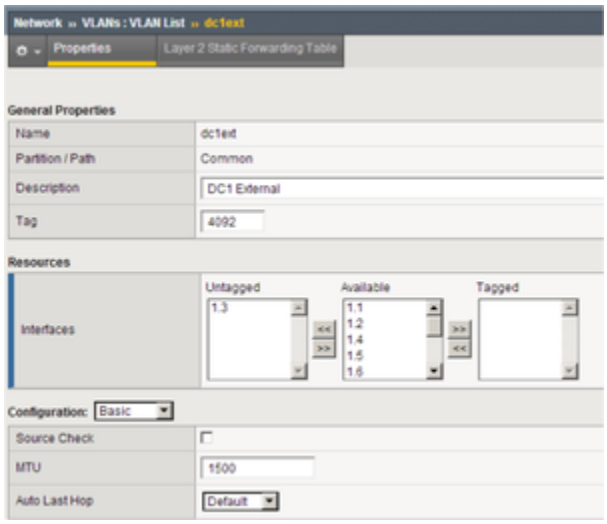
Prerequisites

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

Start

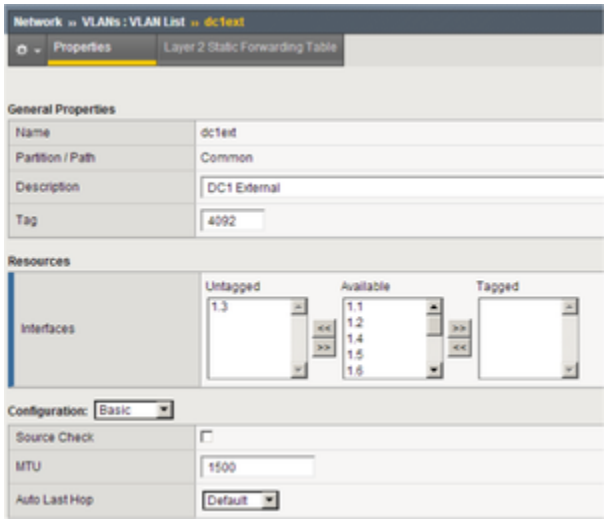
1. Go to `Network > VLANs > VLAN List`.
2. Click `Create`.
3. In the dialog box that appears, specify the following properties:
   a. `Name:` Enter the VLAN name for the external interface—for example, `dc1ext`.
   b. `Tag: 4092` (it is set automatically).

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.



Configuring a VLAN for the External Interface

4. Click `Finished`.
5. Click `Create`.
6. In the dialog box that appears, specify the following properties:
   a. `Name:` Enter the VLAN name for the internal interface—for example, `dc2sip`.
   b. `Tag:` `4090` (it is set automatically).
   c. `Resources > Interfaces > Untagged:` Select `1.6` in the `Available` section and click the left-pointing arrow button to move it into the `Untagged` section.



Configuring a VLAN for the Internal Interface

7. Click `Finished`.

End

3. Configure Self IP addresses.

Configuring Self IP addresses

Purpose

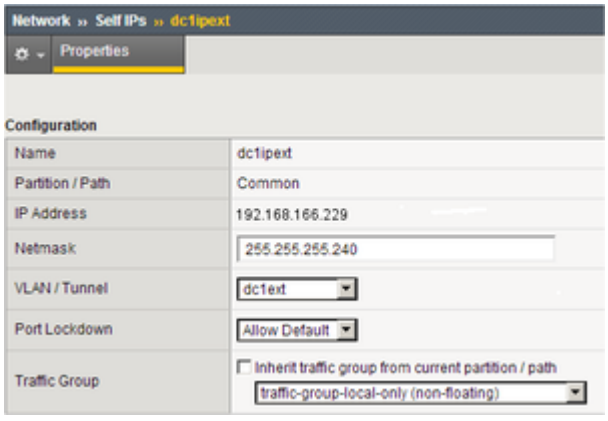
To configure two Self IP addresses—one for the external interface and one for the internal interface—and associate them with the VLANs, to access hosts in those VLANs.

Prerequisites

- Procedure: Configuring VLANs

Start

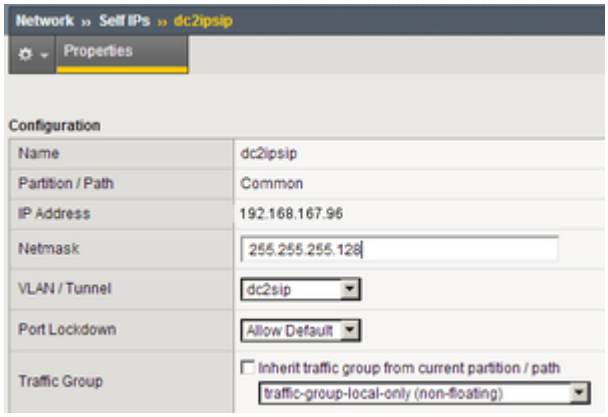
1. Go to `Network > Self IPs`.
2. Click `Create`.
3. In the dialog box that appears, specify the following properties:
   a. `Name:` Enter the name for the Self IP address—for example, `dc1ipext`.
   b. `IP Address:` Enter the IP address for the internal interface—for example, `192.168.166.229`.
   c. `Netmask:` Enter the netmask—for example, `255.255.255.240`.
   d. `VLAN:` Select the name of the VLAN to which you want to assign the Self IP address—for example, `dc1ext`.



Configuring a Self IP Address for the External Interface

4. Click `Finished`.
5. Click `Create`.

6. In the dialog box that appears, specify the following properties:
   a. `Name:` Enter the name for the Self IP address—for example, `dc2ipsip`.
   b. `IP Address:` Enter the IP address for the internal interface—for example, `192.168.167.96`.
   c. `Netmask:` Enter the netmask—for example, `255.255.255.128`.
   d. `VLAN:` Select the name of the VLAN to which you want to assign the self IP address—for example, `dc2sip`.



Configuring a Self IP Address for the Internal Interface

7. Click `Finished`.

End

4. Configure the Default IP route.

Configuring the Default IP route

Purpose

To configure the default IP route.

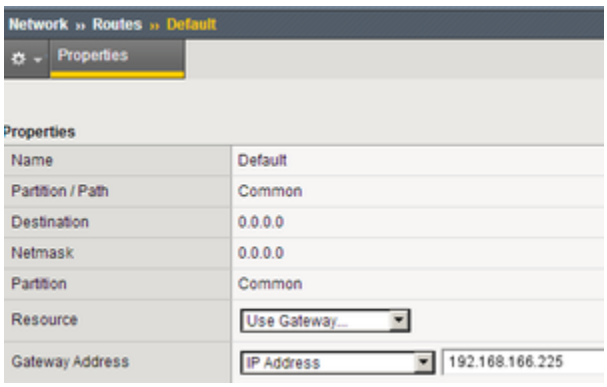
Prerequisites

- Procedure: Configuring Self IP addresses

Start

1. Go to `Network > Routes`.
2. Click `Add`.
3. In the dialog box that appears, specify the following properties:

a. Name: Enter Default.
b. Resource: Select Use Gateway.
c. Gateway Address: Enter the IP address for this default IP route—for example, 192.168.166.225.



Configuring Default IP Route

4. Click Finished.

End

5. Configure SIP Server nodes.

Configuring SIP Server nodes

Purpose

To configure two SIP Server nodes, primary and backup.

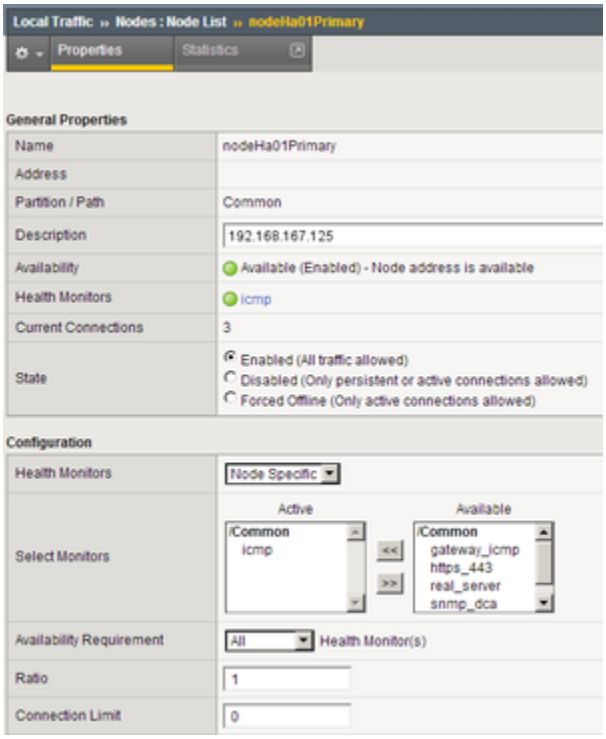
Prerequisites

- Procedure: Configuring the Default IP route

Start

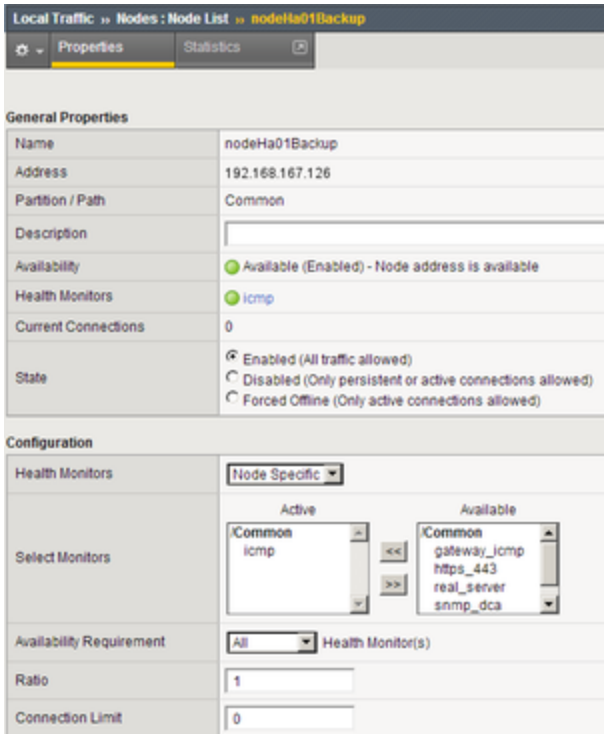
1. Go to Local Traffic > Nodes.
2. Click Create.
3. In the dialog box that appears, specify the following properties:
   a. Name: Enter the node name—for example, nodeHa01Primary.
   b. Address: Enter the IP address for the primary SIP Server node—for example, 192.168.167.125.
   c. Health Monitors: Select Node Specific.

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



Configuring a Primary SIP Server Node

4. Click `Finished`.
5. Click `Create`.
6. In the dialog box that appears, specify the following properties:
   a. `Name:` Enter the node name—for example, `nodeHa01Backup`.
   b. `Address:` Enter the IP address for the backup SIP Server node—for example, `192.168.167.126`.
   c. `Health Monitors:` Select `Node Specific`.
   d. `Select Monitors > Active:` Select `icmp`.



Configuring a Backup SIP Server Node

7. Click `Finished`.

6. Configure a health monitor.

Configuring a health monitor

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

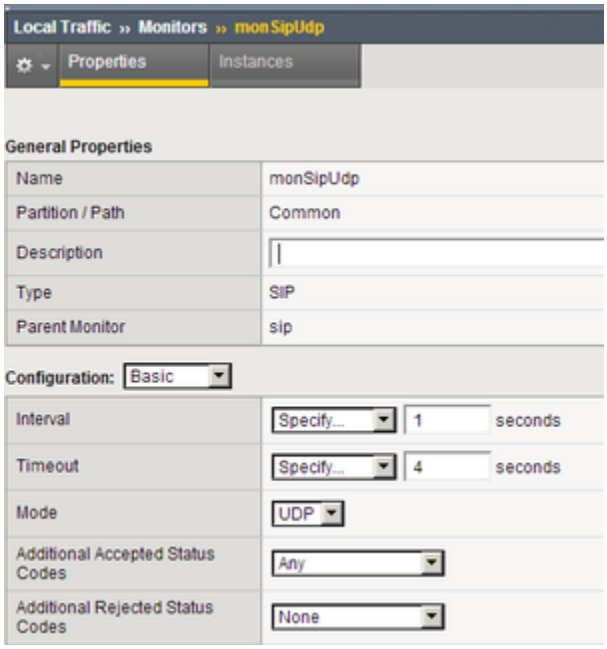
LTM, SIP Server responds to the health check, and the BIG-IP LTM marks SIP Server as active.

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 1 second and `Timeout` to 4 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:
   a. `Name:` Enter the name for this health monitor—for example, `monSipUdp`.
   b. `Type:` Select `SIP`.
   c. `Configuration:` Select `Basic`.
   d. `Interval:` Enter 1 (seconds).
   e. `Timeout:` Enter 4 (seconds).
   f. `Mode:` Select `UDP`.
   g. `Additional Accepted Status Codes:` Select `Any`.



Configuring a Health Monitor

4. Click `Finished`.

End

7. Configure a server pool.

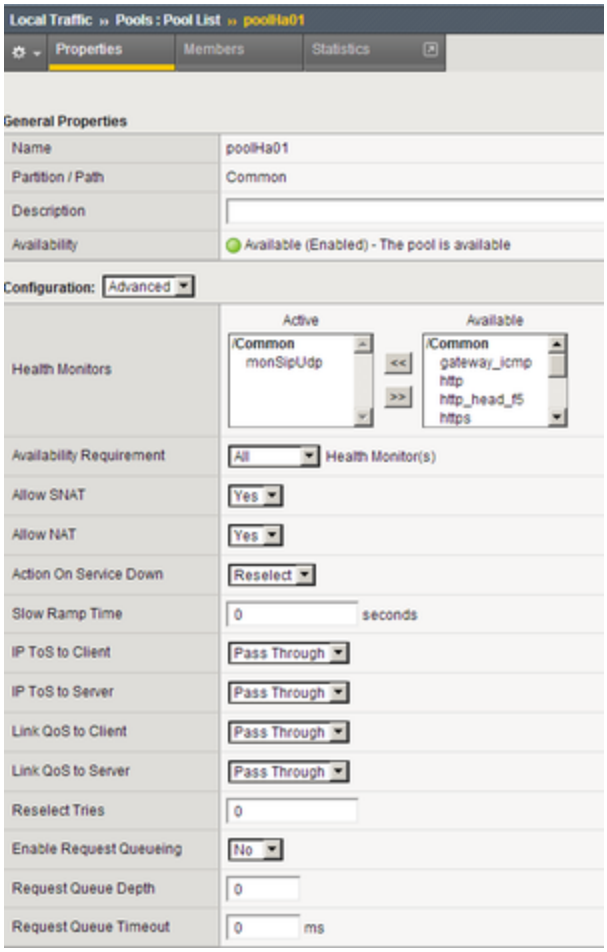
Configuring a server pool

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:
   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. `Priority Group Activation:` Select `Disabled`.



Configuring a Server Pool

4. Click Finished.

End

8. Add server pool members.

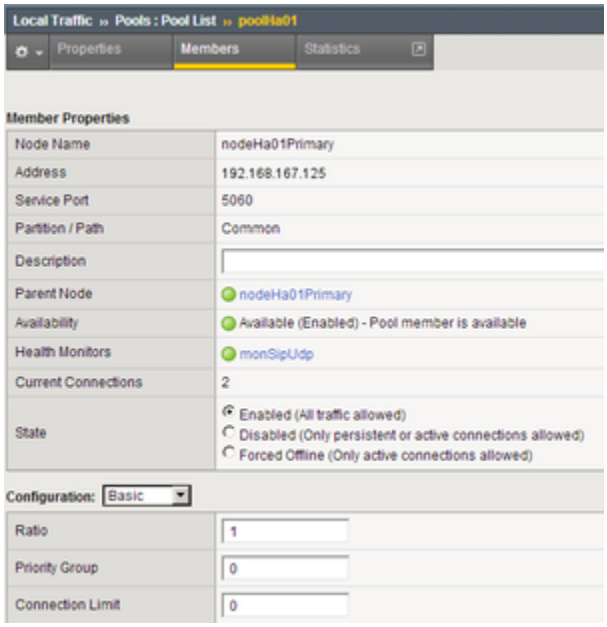
Adding server pool members

Purpose

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

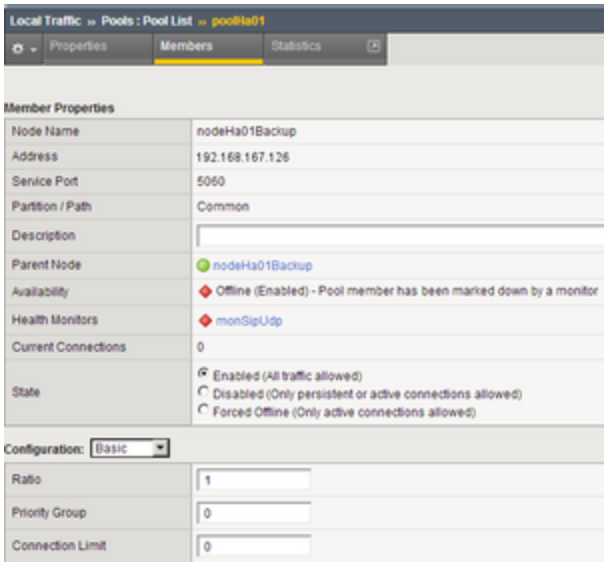
Start

1. Go to `Local Traffic > Pools > poolHa01 > Members`.
2. Click Add.
3. In the dialog box that appears, specify the following properties:
   a. `Node Name:` Select the primary server node you created in Configuring SIP Server nodes. In our example, it would be `nodeHa01Primary`.
   b. `Address:` Specify the IP address of the primary server node. In our example, it would be `192.168.167.125`.
   c. `Service Port:` Enter `5060`.



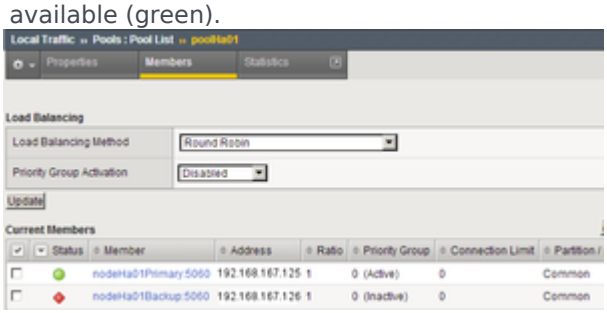
Adding the Primary SIP Server to the Server Pool

4. Click `Finished`.
5. Click Add.
6. In the dialog box that appears, specify the following properties:
   a. `Node Name:` Select the backup server node you created in the Configuring SIP Server nodes. In our example, it would be `nodeHa01Backup`.
   b. `Address:` Specify the IP address of the backup server node. In our example, it would be `192.168.167.126`.
   c. `Service Port:` Enter `5060`.



Adding the Backup SIP Server to the Server Pool

7. Click `Finished`.
8. Set the `Load Balancing Method` to `Round Robin`.
9. Go to `Local Traffic > Pools`. The status of the `nodeHa01Primary` server pool member displays as available (green).



The Server Pool of Two Members

End

9. Configure data groups.

Configuring data groups

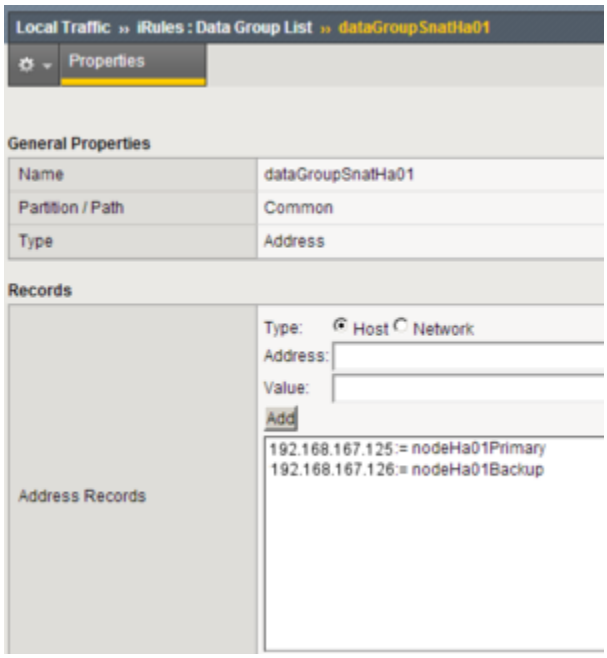
Purpose

To configure data groups that will be used by the iRule. One data group (`dataGroupSnatHa01`)

contains physical IP addresses of primary and backup SIP Server nodes. The second data group (dataGroupSnatExcludedHa01) contains IP addresses of the systems that will be excluded from applying SNAT, such as Genesys Configuration Server, Genesys Message Servers and other SIP Servers in multi-site deployments (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:
   a. Name: Enter the name for this data group—for example, dataGroupSnatExcludedHa01.
   b. Type: Select Address.
   c. Address Records > Type Host > Address: Enter the host IP address of the primary server node—for example, 192.168.167.125.
   d. Click Add.
   e. Address Records > Type Host > Address: Enter the host IP address of the backup server node—for example, 192.168.167.126.
   f. Click Add.



Configuring a Data Group for SNAT

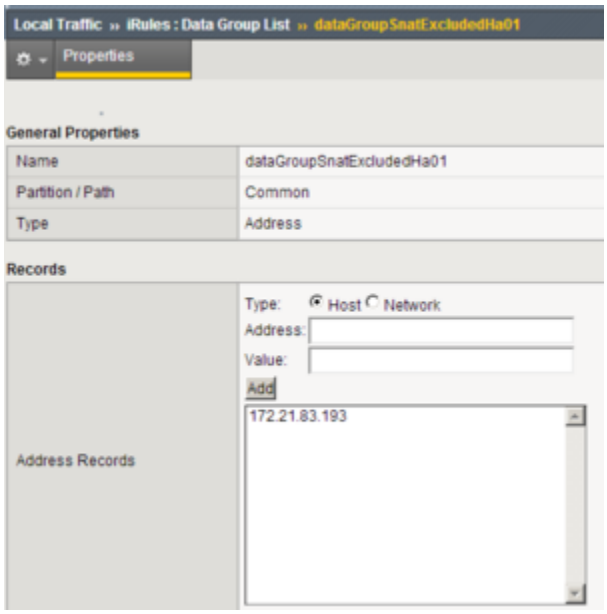
4. Click Finished.
5. Click Create.
6. In the dialog box that appears, specify the following properties:

a. `Name:` Enter the name for this data group—for example, `dataGroupSnatExcluded01`.

b. `Type:` Select `Address`.

c. `Address Records > Type Host > Address:` Enter the host IP address of Genesys Configuration Server—for example, `172.21.83.193`.

d. Click `Add`.



Configuring a Data Group for SNAT Exclusions

7. Click `Finished`.

End

10. Configure a SNAT pool.

Configuring a SNAT pool

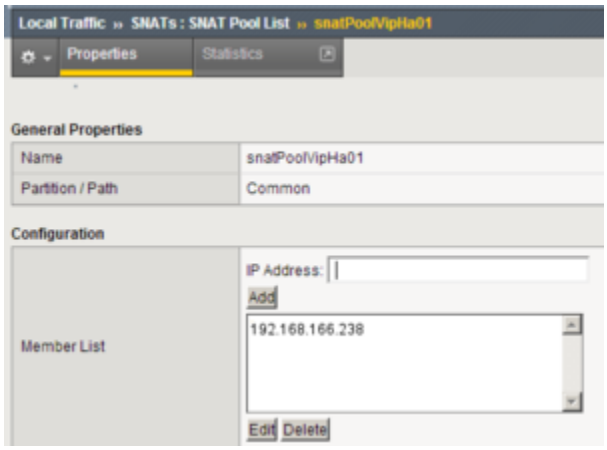
Purpose

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 `dataGroupSnatExcluded01` data group). SNAT is the mapping of one or more original IP addresses to a translation address.

Start

1. Go to `Local Traffic > SNATs`.

2. Click Create.
3. In the dialog box that appears, specify the following properties:
   a. Name: Enter the name for this SNAT pool—for example, snatPoolVipHa01.
   b. Configuration > Members List > IP Address: Enter the IP address to be used as a source IP address—for example, 192.168.166.238.
   c. Click Add.



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:
   a. Name: Enter the name for this iRule—for example, iRuleSnatOutboundHa01.
   b. Definition: Enter the following text:

```
#=====================================================#
# Apply SNAT as specified in snatPoolVipHa01 for all
# packets originated from dataGroupSnatHa01 members.
# Exclude packets addressed to members of
# dataGroupSnatExcludedHa01.
#=====================================================#
when CLIENT_ACCEPTED {
  if { [class match [IP::remote_addr] equals dataGroupSnatHa01] }
  {
    if { [class match [IP::local_addr] equals dataGroupSnatExcludedHa01] }
    {
    }
    else
    {
      snatpool snatPoolVipHa01
    }
  }
}
```

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 Procedure: Configuring VLANs) and it has iRule assigned to Resources, which applies SNAT to all packets (except for packets addressed to Configuration Server).

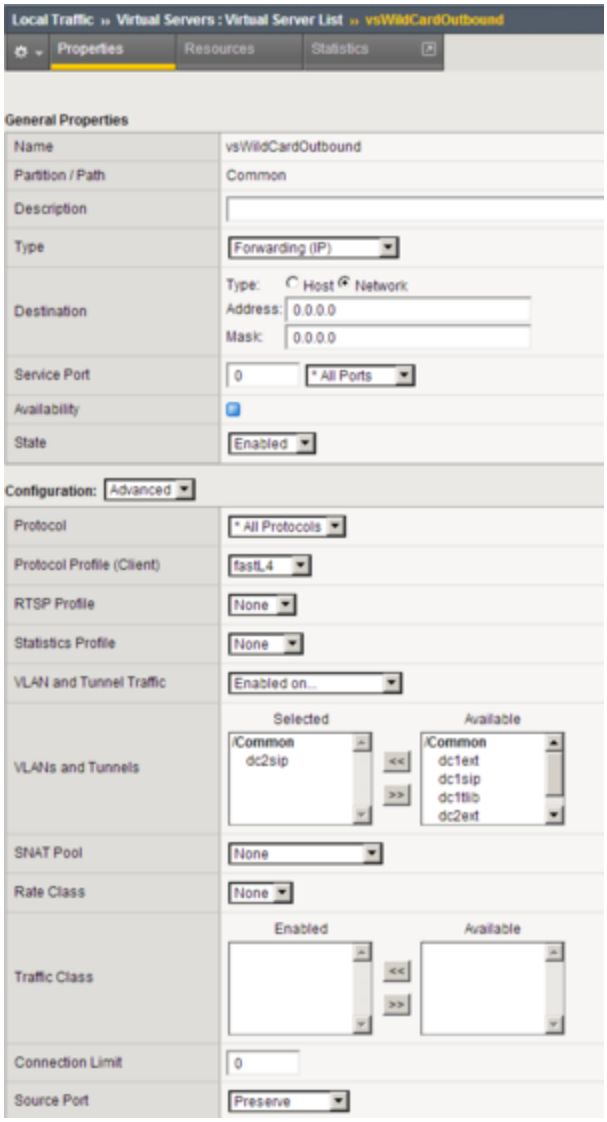
Prerequisites

- Procedure: 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:
   a. Name: Enter the name for this Virtual Server—for example, vsWildCardOutbound.
   b. Type: Select Forwarding (IP).
   c. Destination > Type: Select Network.
   d. Destination > Address: Enter 0.0.0.0.
   e. Destination > Mask: Enter 0.0.0.0.
   f. Service Port: Enter * All Ports.
   g. Configuration: Select Advanced.
   h. Protocol: Select * All Protocols.
   i. VLAN Traffic: Select Enabled on...
   j. VLAN List Selected: Select dc2sip.
   k. SNAT Pool: Select None.
   l. Source Port: Select Preserve.



Configuring a Wildcard Virtual Server for Outbound Traffic

4. Click Finished.
5. Go to Local Traffic > Virtual Server List > vsWildcardOutbound > Resources.
6. Add iRule as follows:

```
Resources > iRule Enabled > iRuleSnatOutboundHa01
```

End

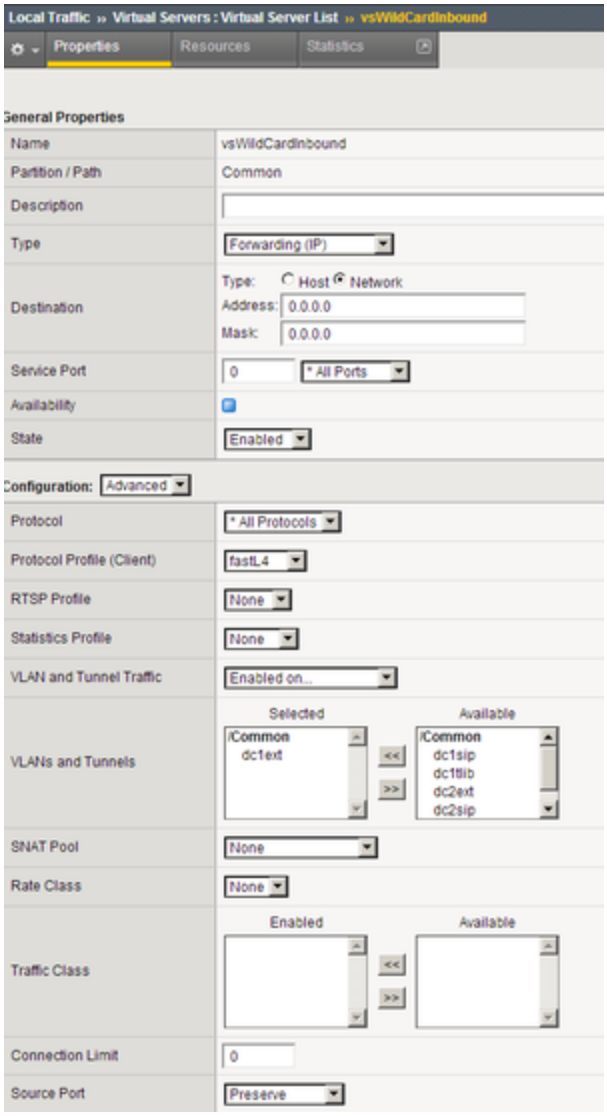
[+] Configuring a Virtual Server for inbound traffic

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:
   a. `Name:` Enter the name for this Virtual Server—for example, `vsWildCardInbound`.
   b. `Type:` Select `Forwarding (IP)`.
   c. `Destination > Type:` Select `Network`.
   d. `Destination > Address:` Enter `0.0.0.0`.
   e. `Destination > Mask:` Enter `0.0.0.0`.
   f. `Service Port:` Enter `* All Ports`.
   g. `Configuration:` Select `Advanced`.
   h. `Protocol:` Select `* All Protocols`.
   i. `VLAN Traffic:` Select `Enabled on...`.
   j. `VLAN List Selected:` Select `dc1ext`.



Configuring a Wildcard Virtual Server for Inbound Traffic

4. Click Finished.

End

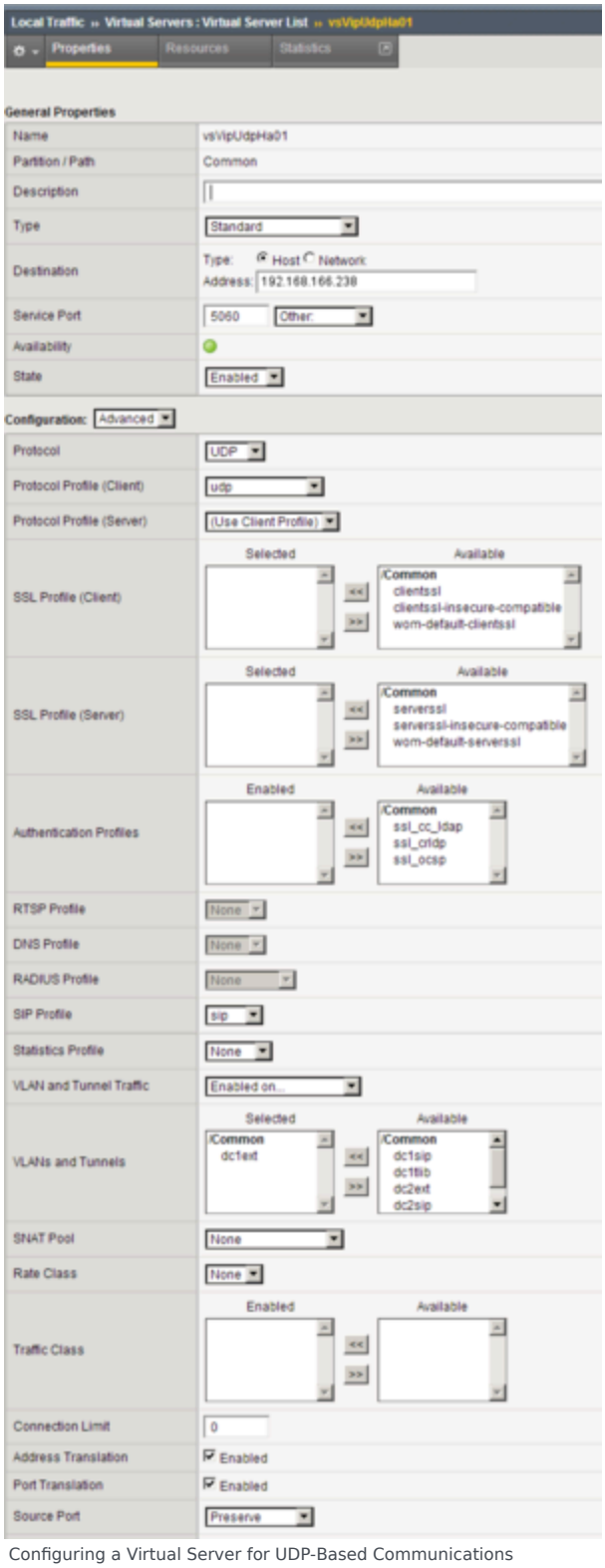
[+] Configuring Virtual Servers for UDP and TCP SIP communications

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.

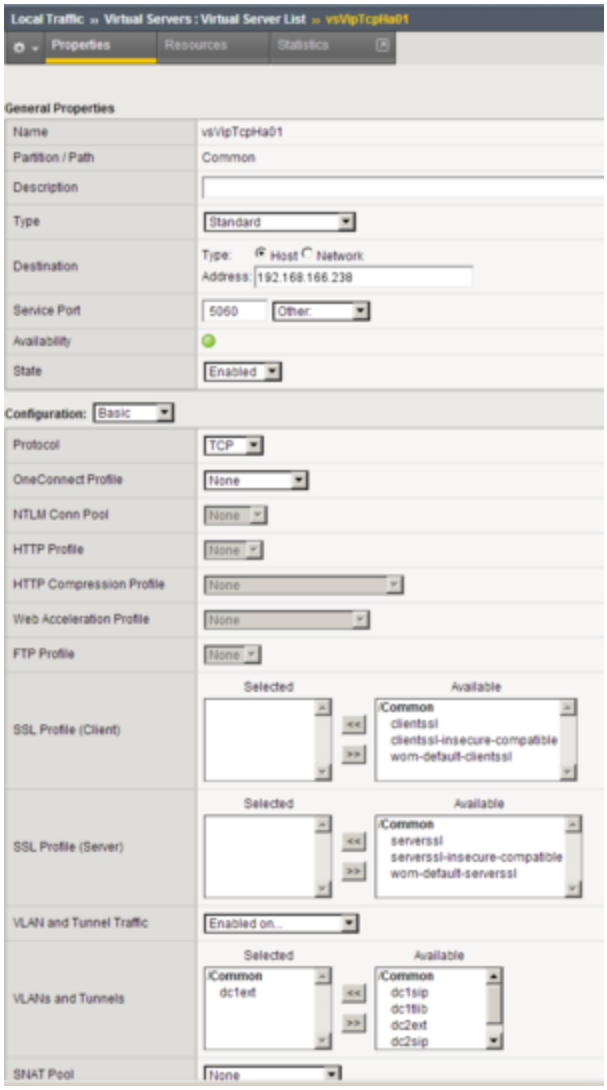
Start

1. Go to `Local Traffic > Virtual Servers`.
2. Click `Create`.
3. In the dialog box that appears, specify the following properties:
   a. `Name:` Enter the name for this Virtual Server—for example, `vsVipUdpHa01`.
   b. `Destination > Type:` Select `Host`.
   c. `Destination > Address:` Enter the IP address for this Virtual Server—for example, `192.168.166.238`.
   d. `Service Port:` Enter `5060` and select `Other`.
   e. `State:` Select `Enabled`.
   f. `Configuration:` Select `Advanced`.
   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 `dc1ext`.



Configuring a Virtual Server for UDP-Based Communications

4. Click Finished.
5. Select Resources > Load Balancing > Default Pool: Select poolHa01.
6. Click Update.
7. Go to Local Traffic > Virtual Servers.
8. Click Create.
9. In the dialog box that appears, specify the following properties:
    a. Name: Enter the name for this Virtual Server—for example, vsVipTcpHa01.
    b. Destination > Type: Select Host.
    c. Destination > Address: Enter the IP address for this Virtual Server—for example, 192.168.166.238.
    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 dc1ext.



Creating a Virtual Server for TCP-Based Communications

10. Click `Finished`.
11. Select `Resources > Load Balancing > Default Pool:` Select `poolHa01`.
12. Click `Update`.

End

Note: SNAT pool must not be activated as the configuration element of Virtual Server. SNAT must not be applied to SIP messages sent by clients to SIP Server. Enabling SNAT for these messages results in a broken call flow, as in some circumstances the SNAT is not aligned with SIP Protocol.

<verttabber>

Configuring TLS

Secure data transfer using TLS is now supported between SIP Server and Active-Active Resource Managers in a deployment where SIP Server high-availability is configured using the F5 Networks BIG-IP LTM. TLS is also supported between SIP Server and all SIP devices in this deployment, including SBCs, Media Gateways, and SIP phones. BIG-IP LTM is not a TLS peer as are other elements in the environment; there is no TLS negotiation between BIG-IP LTM and other components.

Integration Solution Assumptions

The integration solution described in this section makes the following assumptions:

- BIG-IP LTM is deployed as a single instance
- TLS transport is used for SIP signaling
- SIP Server performs load balancing between an Active-Active Resource Manager pair

See Deployment Architecture Example.

Deploying the TLS Solution

To support TLS data transfer in a SIP Server deployment with an Active-Active RM pair and a BIG-IP LTM used for the SIP Server HA, complete the following procedures:

1. Configure BIG-IP LTM for TLS.
2. Provision SSL certificates for workstations hosting SIP Servers, RM, and MCP applications. Refer to the ''Genesys 8.1 Security Deployment Guide''.
3. Configure SIP Server to use TLS data transfer. Refer to the *Transport Layer Security for SIP Traffic* section in the *Framework 8.1 SIP Server Deployment Guide*.
4. Configure Resource Managers in an Active-Active high-availability cluster. Refer to the *Genesys Voice Platform Integration* section in the ''Framework 8.1 SIP Server Deployment Guide''.

To configure TLS data transfer between Genesys Media Server components, refer to the ''Genesys Media Server 8.1 Deployment Guide''.

Configuring BIG-IP LTM for TLS

1. Complete general configuration procedures.

Before starting the TLS-specific configuration of BIG-IP LTM, complete the configuration procedures.

2. Configure a health monitor.

To configure a health monitor:

1. Go to `Local Traffic > Monitors`.
2. Click `Create`.
3. In the dialog box that appears, specify the following properties:
   - `Name:` Enter the name for this health monitor—for example, `monSipTls`.
   - `Type:` Select `TCP`.
   - `Parent Monitor:` Select `TCP`.
   - `Configuration:` Select `Basic`.
   - `Interval:` Enter 1 (seconds).
4. Click `Finished`.

3. Configure a server pool.

To configure a server pool:

1. Go to `Local Traffic > Pools`.
2. Click `Create`.
3. In the dialog box that appears, specify the following properties:
   - `Name:` Enter the name for this server pool—for example, the `poolHa01tls`.
   - `Configuration:` Select `Advanced`.
   - `Health Monitors > Active:` Select `monSipTls`.
   - `Action On Service Down:` Select `Reselect`.
   - `Slow Ramp Time:` Enter`0` (seconds).
4. Click `Finished`.

4. Add server pool members.

To add server pool members:

1. Go to `Local Traffic > Pools > poolHa01tls > Members`.
2. Click `Add`.

3. In the dialog box that appears, specify the following properties:
   - Node Name: Select the primary server node. In our example, it would be nodeHa01Primary.
   - Address: Specify the IP address of the primary server node. In our example, it would be 192.168.167.125.
   - Service Port: Enter 5061.
4. Click Finished.
5. Click Add.
6. In the dialog box that appears, specify the following properties:
   - Node Name: Select the backup server node. In our example, it would be nodeHa01Backup.
   - Address: Specify the IP address of the backup server node. In our example, it would be 192.168.167.126.
   - Service Port: Enter 5061.
7. Click Finished.
8. Set the Load Balancing Method to Round Robin.

5. Configure a Virtual Server.

To configure a Virtual Server:

1. Go to Local Traffic > Virtual Servers.
2. Click Create.
3. In the dialog box that appears, specify the following properties:
   - Name: Enter the name for this Virtual Server—for example, vsVipTlsHa01.
   - Destination > Type: Select Host.
   - Destination > Address: Enter the IP address for this Virtual Server—for example, 192.168.166.238.
   - Service Port: Enter 5061 (Other).
   - State: Select Enabled.
   - Configuration: Select Basic.
   - Type: Select Standard.
   - Protocol: Select TCP.
   - VLAN and Tunnel Traffic: Select Enabled on...
   - VLANs and Tunnels Selected: Select dc1ext.
4. Click Finished.

5. Select `Local Traffic > Virtual Server List > vsVipTlsHa01 > Resources`.
6. Add `Default Pool`: `poolHa01tls`
7. Click `Update`.

6. Disable Virtual Servers for UDP and TCP.

At this point, the BIG-IP LTM is configured for handling communications over different protocols: UDP, TCP, and TLS. If TLS is mandatory for security reasons, Genesys strongly recommends disabling virtual servers for insecure protocols, such as UDP and TCP.

To disable Virtual Servers for UDP and TCP:

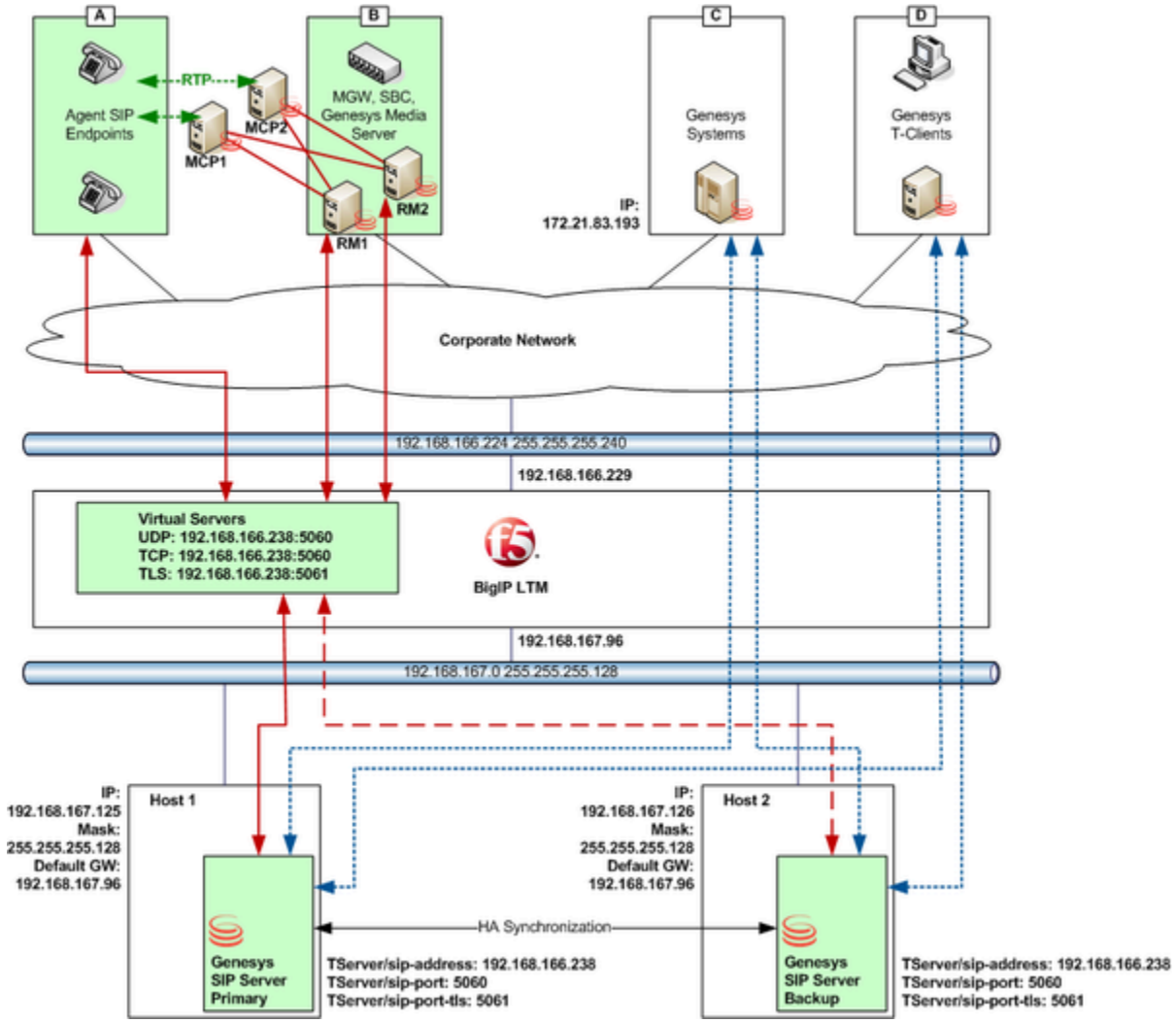
1. Go to `Local Traffic > Virtual Servers`.
2. Select check boxes of the following virtual servers: `vsVipUdpHa01` and `vsVipTcpHa01`.
3. Click `Disable`.

This completes configuring BIG-IP LTM.

Deployment Architecture Example

The figure below depicts the architecture of a sample deployment of primary and backup SIP Servers, Active-Active Resource Manager pair, and the BIG-IP LTM, in which:

- BIG-IP LTM provides a single Virtual IP address for SIP phones and Media Server.
- The pair of SIP Servers configured in Hot-Standby mode is located behind BIG-IP LTM and the actual IP addresses of the primary and backup SIP Servers are not exposed to phones or Media Server components.
- All SIP entities in the environment—phones, SIP Servers, Resource Managers, MCPs—can use TLS for SIP signaling if configured.



RedSky E911 Manager

Starting with release 8.1.102.81, Genesys SIP Server can operate with the RedSky E911 Manager system, the provider of emergency (911) calls in the VoIP environment. Workspace SIP Endpoint version 8.5.113.02 or later must be used for this integration.

SIP Server was tested with the following RedSky components:

- RedSky E911 Manager version 6.7.0
- RedSky Emergency Gateway version 6.5-122

All communication transport protocols are supported: UDP, TCP, or TLS.

For this integration, you configure:

- Genesys SIP Server, DN configuration objects
- RedSky components

Genesys Configuration

Configure the following DN configuration objects under the SIP Server switch:

- The DN object of type **Trunk** for the RedSky E911 Manager must have the following options configured in the **TServer** section:
  - **contact**—Set to the RedSky E911 Manager URI. The URI format is described in the **contact** option description in the Framework 8.1 SIP Server Deployment Guide.
  - **prefix**—Set to a value that matches emergency call starting digits.
  - **contact-list**—Configure this option if there is more than one instance of the Red Sky E911 Manager in the environment.
  - **oos-check**
  - **oos-force**
  - **emergency-device**—Set to `true`.
- The DN object of type **Trunk** for the RedSky Emergency Gateway must have the following options configured in the **TServer** section:
  - **contact**—Set to the RedSky Emergency Gateway URI.
  - **prefix**—Set to a value different from the Trunk DN pointing to Red Sky Server.
  - **contact-list**—Configure this option if there is more than one instance of the Red Sky Server in the environment.
  - **oos-check**

- **oos-force**
- **emergency-device**—Set to `true`.

The SIP Server Application must contain the following configuration options in the **TServer** section:

- **subscription-event-allowed**—Set this option to `reg` or * (asterisk).
- **subscription-max-body-size**—Define the maximum size of the NOTIFY XML body (in bytes) within the SUBSCRIBE dialog. The default value is 14336. The range of valid values is `0`–`500000`. If the option is set to `0` (zero), the message body can be any size. The zero value can be used for TCP transport but is not recommended for UDP. For bulk notification, SIP Server sends more than one NOTIFY, so adjust the size accordingly.
- **sip-elin-timeout**—Define the time interval, in seconds, for SIP Server to keep in memory the association between a 911 caller and the Emergency Location Identification Number (ELIN) assigned to the caller. The default value is `1200`. The range of valid values is `0`–`3600`. If a call arrives at that ELIN before the timeout expires, the call is sent to the associated 911 caller DN. If within this time interval there are several emergency calls with the same ELIN, SIP Server directs the callback to the latest caller.

RedSky Configuration

On the RedSky side, configure the following components:

- RedSky Emergency Gateway
- RedSky E911 Manager—Add "Genesys" as a call server using the Call Servers module. Refer to RedSky documentation for details.

RedSky E911 Manager subscribes to SIP Server for endpoint state notifications. It discovers a physical location of a DN based on its IP address and, if possible, its MAC address, then assigns the Emergency Location Identification Number (ELIN) to each endpoint. When E911 Manager receives an emergency call initial INVITE message, it redirects it providing the ELIN of the calling endpoint. The RedSky Emergency Gateway receives the redirected INVITE and sends the voice call to the public-safety answering point (PSAP) agency.

Known Limitations

For the SIP Server and RedSky solution to work correctly, the following conditions must be met:

- SIP Server is running in the non-cloud environment (no SBC to mask endpoint IP addresses with its own address).
- Agent DNs are self-registered endpoints.
- The MAC address is provided to RedSky only for deployment with Genesys SDK-based endpoints running on dedicated hardware hosts (not virtual machines).