Genesys 7.5

GVP - SIP Server

Integration Guide

About Genesys

Genesys Telecommunications Laboratories, Inc., a subsidiary of Alcatel-Lucent, is 100% focused on software for call centers. Genesys recognizes that better interactions drive better business and build company reputations. Customer service solutions from Genesys deliver on this promise for Global 2000 enterprises, government organizations, and telecommunications service providers across 80 countries, directing more than 100 million customer interactions every day. Sophisticated routing and reporting across voice, e-mail, and Web channels ensure that customers are quickly connected to the best available resource—the first time. Genesys offers solutions for customer service, help desks, order desks, collections, outbound telesales and service, and workforce management. Visit www.genesyslab.com for more information.

Each product has its own documentation for online viewing at the Genesys Technical Support website or on the Documentation Library DVD, which is available from Genesys upon request. For more information, contact your sales representative.

Notice

Although reasonable effort is made to ensure that the information in this document is complete and accurate at the time of release, Genesys Telecommunications Laboratories, Inc., cannot assume responsibility for any existing errors. Changes and/or corrections to the information contained in this document may be incorporated in future versions.

Your Responsibility for Your System's Security

You are responsible for the security of your system. Product administration to prevent unauthorized use is your responsibility. Your system administrator should read all documents provided with this product to fully understand the features available that reduce your risk of incurring charges for unlicensed use of Genesys products.

Trademarks

Genesys, the Genesys logo, and T-Server are registered trademarks of Genesys Telecommunications Laboratories, Inc. All other trademarks and trade names referred to in this document are the property of other companies. The Crystal monospace font is used by permission of Software Renovation Corporation, www.SoftwareRenovation.com.

Technical Support from VARs

If you have purchased support from a value-added reseller (VAR), please contact the VAR for technical support.

Technical Support from Genesys

If you have purchased support directly from Genesys, please contact Genesys Technical Support at the following regional numbers:

Region	Telephone	E-Mail
North and Latin America	+888-369-5555 or +506-674-6767	support@genesyslab.com
Europe, Middle East, and Africa	+44-(0)-1276-45-7002	support@genesyslab.co.uk
Asia Pacific	+61-7-3368-6868	support@genesyslab.com.au
Japan	+81-3-6361-8950	support@genesyslab.co.jp

Prior to contacting technical support, please refer to the *Genesys Technical Support Guide* for complete contact information and procedures.

Ordering and Licensing Information

Complete information on ordering and licensing Genesys products can be found in the ***Genesys 7 Licensing Guide***.

Released by

Genesys Telecommunications Laboratories, Inc. www.genesyslab.com

Document Version: 75gvp-sip_ig_06-2008_v7.5.001.00

Table of Contents

List of Procedures

Preface

Welcome to the *Genesys 7.5 GVP–SIP Server Integration Guide*. This guide provides an overview of the Genesys Voice Platform (GVP)–SIP Server integration in its various modes—In-Front, Behind, and Stand-Alone—as well as the relevant procedures for completing the integration.

Integrating SIP Server with the GVP requires the configuration of various Genesys components. To reduce the number of documents that you need to consult to perform the integration, this document provides all the necessary integration-specific procedures. It starts with an overview of the three integration modes, and then continues with step-by-step instructions for configuring the applications and components required for each mode.

This document applies to the 7.5 release of SIP Server, and the 7.5 and 7.6 releases of GVP.

Note: For releases of this document created for other releases of this product, please visit the Genesys Technical Support website, or request the Documentation Library DVD, which you can order by e-mail from Genesys Order Management at orderman@genesyslab.com.

This preface includes the following sections:

Intended Audience

This guide is primarily intended for system engineers and other members of an implementation team who will complete the integration of existing SIP Server and GVP deployments. This guide assumes that you have a basic understanding of:

- Computer-telephony integration (CTI) concepts, processes, terminology, and applications.

- The Session Initiation Protocol generally, as well as how SIP messaging is used within the Genesys environment—through the SIP Server and related components.
- Network design and operation.
- Your own network configurations.

This guide also assumes that:

- You are familiar with the Genesys Framework architecture and functions that support SIP Server 7.5 and the Genesys Voice Platform.
- You have already installed, and are familiar with, SIP Server and its related components, as well as GVP and its related components.

Chapter Summaries

In addition to this preface, this document contains the following chapters:

- Chapter 1, "Overview," on page 13, describes the three possible modes for a GVP–SIP Server integration: In-Front, Behind, and Stand-Alone. It also includes sample architectural drawings for each of the modes, as well as details about the basic inbound call flows through the integrated environment.
- Chapter 2, "GVP-SIP Server Integration—Behind Mode," on page 25 provides a task flow of the main steps required to integrate GVP and SIP Server in the Behind mode, along with key actions and, if you need them, links to more detailed procedures found later in the chapter.
- Chapter 3, "GVP-SIP Server Integration—In-Front Mode," on page 45, provides a task flow of the main steps required to integrate GVP and SIP Server in the In-Front mode, along with key actions and, if you need them, links to more detailed procedures found later in the chapter.
- Chapter 4, "GVP–SIP Server Integration—Stand-Alone Mode," on page 75, provides a task flow of the main steps required to integrate GVP and SIP Server in the in Stand-Alone mode, along with key actions and, if you need them, links to more detailed procedures found later in the chapter.
- Appendix A, "Integration Worksheets," on page 83, provides worksheets to help organize the configuration values required for each integration mode.

Document Conventions

This document uses some stylistic and typographical conventions with which you might want to familiarize yourself.

Document Version Number

A version number appears at the bottom of the inside front cover of this document. Version numbers change as new information is added to this document. Here is a sample version number:

76mm_dep_03-2007_v7.6.001.00

You will need this number when you are talking with Genesys Technical Support about this product.

Type Styles

Italic

In this document, italic is used for the titles of documents, when a term is being defined, for emphasis, and for mathematical variables.

Examples

- Please consult the *Genesys 7 Migration Guide* for more information.
- *A customary and usual practice* is one that is widely accepted and used within a particular industry or profession.
- Do *not* use this value for this option.
- The formula, $x + 1 = 7$ where x stands for . . .

Monospace Font

A monospace font, which is shown in the following examples, is used for all programming identifiers and GUI elements.

This convention includes the *names* of directories, files, folders, configuration objects, paths, scripts, dialog boxes, options, fields, text and list boxes, operational modes, all buttons (including radio buttons), check boxes, commands, tabs, CTI events, and error messages; the values of options; logical arguments and command syntax; and code samples.

Examples

- Select the `Show variables on screen` check box.
- Click the `Summation` button.
- In the `Properties` dialog box, enter the value for the host server in your environment.
- In the `Operand` text box, enter your formula.
- Click `OK` to exit the `Properties` dialog box.

- The following table presents the complete set of error messages T-Server® distributes in `EventError` events.
- If you select `true` for the `inbound-bsns-calls` option, all established inbound calls on a local agent are considered business calls.

Monospace font is also used for any text that users must manually enter during a configuration or installation procedure, or on a command line:

Example
- Enter `exit` at the command line.

Screen Captures Used in This Document

Screen captures from the product GUI (graphical user interface), as used in this document, may sometimes contain a minor spelling, capitalization, or grammatical error. The text accompanying and explaining the screen captures corrects such errors *except* when such a correction would prevent you from installing, configuring, or successfully using the product. For example, if the name of an option contains a usage error, the name would be presented exactly as it appears in the product GUI; the error would not be corrected in any accompanying text.

Square Brackets

Square brackets indicate that a particular parameter or value is optional within a logical argument, a command, or some programming syntax. That is, the parameter's or value's presence is not required to resolve the argument, command, or block of code. The user decides whether to include this optional information. Here is a sample:

```
smcp_server -host [/flags]
```

Angle Brackets

Angle brackets indicate a placeholder for a value that the user must specify. This might be a DN or port number specific to your enterprise. Here is a sample:

```
smcp_server -host <confighost>
```

Related Resources

This guide assumes that you have already installed and configured the component products listed in this section.

SIP Server

Consult the following additional resource as necessary:

- *Framework 7.5 SIP Server Deployment Guide*

Genesys Voice Platform

Consult the following additional resources as necessary:

- *Genesys Voice Platform 7.5 Deployment Guide*
- *Genesys Voice Platform 7.5 Reference Manual*
- *Genesys Voice Platform 7.5 Troubleshooting Guide*
- *Genesys Voice Platform 7.5 Studio Deployment Guide*
- *Genesys Studio Help 7.5*
- *Genesys Voice Platform 7.5 VoiceXML 2.1 Reference Manual*

IVR Server

Consult the following additional resource, if necessary:

IVR Interface Option 7.5 IVR Server—System Administrator's Guide

Universal Routing

Consult the following resources as necessary:

- *Universal Routing 7.5Reference Manual,* which contains descriptions of all routing strategy objects.
- *Universal Routing 7.5 Strategy Samples,* which describes the sample strategies supplied with Universal Routing.
- *Universal Routing 7.5 Business Process User's Guide,* which contains step-by-step instructions for using Interaction Routing Designer to design interaction workflows.
- *Universal Routing 7.5 Interaction Routing Designer Help,* which is a guide to Interaction Routing Designer.

Others

Consult the following resources as necessary:

- *Framework 7.5 Configuration Options Reference Manual*
- *Framework 7.5 Configuration Manager Help*
- *Genesys Technical Publications Glossary,* which ships on the Genesys Documentation Library DVD and which provides a comprehensive list of the Genesys and CTI terminology and acronyms used in this document.
- *Genesys 7 Migration Guide*, also on the Genesys Documentation Library DVD, which contains a documented migration strategy for Genesys product releases 5.x and later. Contact Genesys Technical Support for additional information.
- Release Notes and Product Advisories for these products, which are available on the Genesys Technical Support website at `http://genesyslab.com/support`.
- Documentation on the other three members of the Genesys Customer Interaction Platform: Universal Routing, Reporting, and Management Framework.

Information about supported hardware and third-party software is available on the Genesys Technical Support website in the following documents:

- *Genesys 7 Supported Operating Systems and Databases*
- *Genesys 7 Supported Media Interfaces*

Genesys product documentation is available on the:

- Genesys Technical Support website at `http://genesyslab.com/support`.
- Genesys Documentation Library DVD, which you can order by e-mail from Genesys Order Management at `orderman@genesyslab.com`.

Making Comments on This Document

If you especially like or dislike anything about this document, please feel free to e-mail your comments to `Techpubs.webadmin@genesyslab.com`.

You can comment on what you regard as specific errors or omissions, and on the accuracy, organization, subject matter, or completeness of this document. Please limit your comments to the information in this document only and to the way in which the information is presented. Speak to Genesys Technical Support if you have suggestions about the product itself.

When you send us comments, you grant Genesys a nonexclusive right to use or distribute your comments in any way it believes appropriate, without incurring any obligation to you.

Chapter

1 Overview

This chapter provides an overview of three possible modes for integrating the Genesys Voice Platform (GVP) with the SIP Server: In-Front, Behind, and Stand-Alone. Use this chapter to gain an understanding of how these integration modes differ, as well as their basic high-level architectures and call flows.

This chapter includes the following sections:

Note: For the In-Front and Behind integration modes, the terms "in-front" and "behind" refer to the logical or configured relationship of the IVR Server to the switch. Previously, in Time-division Multiplexing (TDM) environments, the terms "in-front" and "behind" as modes for the IVR Server implied a more concrete physical relationship to the switch than they do in pure IP environments. For example, in a pure IP configuration, the IVR Server might be placed physically behind the switch, but configured for the In-Front mode. Keep this distinction in mind when reviewing the descriptions and architecture drawings in this chapter.

What Is the GVP–SIP Server Integration?

An integrated GVP–SIP Server configuration combines the voice self-service functionality of GVP with the IP-based routing capability of the Genesys SIP Server. Depending on your configuration, SIP Server can provide customer access to live agents, in addition to the self-service features offered by GVP, like speech recognition and text-to-speech technologies.

For an integration that includes Framework routing to agents, choose either the In-Front or Behind integration modes. The Stand-Alone mode is a pure self-service configuration, in which the SIP Server provides mostly pre-call routing to the self-service applications, but without Framework routing to agents as in other modes.

Configuration for each of these modes differs primarily because of the different ways that GVP receives call information—typically the Dialed Number Identification Service (DNIS)—that it needs in order to start the self-service VXML application.

The various available GVP–SIP Server integration modes include:

- Stand-Alone
- In-Front (IVR Server configured for In-Front mode, and GVP also configured for In-Front mode)
- Behind (IVR Server configured for Behind, and GVP usually configured for In-Front, though sometimes, in blended TDM/IP environments, configured for Behind)

IVR Behind–GVP In-Front (pure IP environment)

In a pure IP configuration, where the SIP Server acts as the soft switch, the most common integration includes an IVR Server configured for the Behind mode, with GVP configured for In-Front. In this configuration, as part of the initial call, GVP receives the call-related information that it needs to start the self-service application: primarily the Automatic Number Identification (ANI) and the DNIS. The IVR Server is connected directly to the SIP Server as a client, and is not involved in passing call-related data to GVP.

For pure IP environments, Genesys recommends this IVR Behind/GVP In-Front approach whenever possible. On balance, this configuration offers more connection, feature, and performance advantages than the other IVR/GVP mode combinations.

IVR In Front–GVP In Front (blended TDM/IP or pure IP environment)

In blended or pure environments, where SIP Server acts as a SIP proxy for GVP (but does not function as the soft switch), and GVP is placed behind a TDM or hybrid switch, the IVR Server might not be connected directly to the SIP Server. In this case, you must configure the IVR Server for the In-Front mode. In this scenario, the call lands first on GVP, which then informs the IVR Server of the ANI/DNIS for that call.

Less common than the Behind-mode scenario, the IVR In Front/GVP In Front combination is used for pure self-service deployments or in configurations where T-Servers are not accessible. In this mode, GVP provides the self-service application. In addition, the IVR Server can launch voice treatments on the call and also check if agents are available. However, transferring the call to agents for assisted-service is not provided by the Genesys framework, but by GVP itself.

IVR Behind–GVP Behind (blended TDM/IP environment)

Another configuration that supports the blended environment (TDM or hybrid switch) is one where the IVR Server is configured for Behind mode, so that it can communicate directly with the SIP Server, while GVP is also configured for Behind mode. In this configuration, the call arrives at the switch first and call information is passed to the IVR Server. GVP then queries the IVR Server to get the call-related information that it needs (ANI and DNIS) to establish the call and start the self-service application.

GVP in Stand-Alone Mode (pure self-service)

In this mode, as in the other modes, GVP uses SIP Server as its SIP proxy instead of the GVP SIP Session Manager component. In the Stand-Alone mode, however, the IVR Server is not included in the configuration. GVP is integrated into the Framework to the degree that the SIP Server, as a TServer application, requires Genesys Framework. However, Framework routing to agents is not available in this configuration.

About the Components

Integrating SIP Server with GVP requires both Genesys applications and third-party networking components. Depending on the integration mode, these applications and components can include:

- SIP Server
- Genesys Voice Platform:
  - Voice Platform Common

    - Resource Manager
    - IP Communication Server (IPCS)
- Soft switch
- Media Gateway
- IVR Server
- Stream Manager
- Stat Server
- Universal Routing Server (URS)

SIP Server

Genesys SIP Server is a combined T-Server and switching component. In a GVP–SIP Server integration, SIP Server acts as a proxy for GVP, providing an interface for SIP communication with external components. Depending on the integration mode, SIP Server can also act as a T-Server or a soft switch for GVP.

GVP Resource Manager

The Genesys Voice Platform can integrate with the SIP Server through the GVP Resource Manager component. The Resource Manager is optional for the integration, but required for certain features. For example, IPCS section based on specific resources, such as Automatic Speech Recognition.

In a typical GVP configuration, the Resource Manager maintains the resource states for the IPCS and the Media Gateway—it monitors whether these resources are currently in use or available to accept calls. In the GVP–SIP Server integration, the Resource Manager also acts as a SIP Redirect Server. In this scenario, SIP Server sends a new call to the Resource Manager, and then the Resource Manager finds and provides an available resource on the IPCS based on specific feature requirements.

IP Communication Server (IPCS)

The IPCS handles calls through Voice over Internet Protocol (VoIP). To send or receive a call using a Media Gateway, the IPCS sets up a SIP session, handles security, generates events, retrieves customer applications as necessary, maintains the media stream, and closes the SIP session at the end of the call.

For the GVP–SIP Server integration, with GVP in the Behind mode, the IPCS requires certain specific configuration settings. For information about these required settings, see “Configure the IPCS for GVP Behind mode (optional).” on page 26.

Behind Mode—Architecture and Call Flows

Figure 1 shows a sample architecture for a GVP–SIP Server integration in the Behind mode, where agents are registered on the SIP Server.

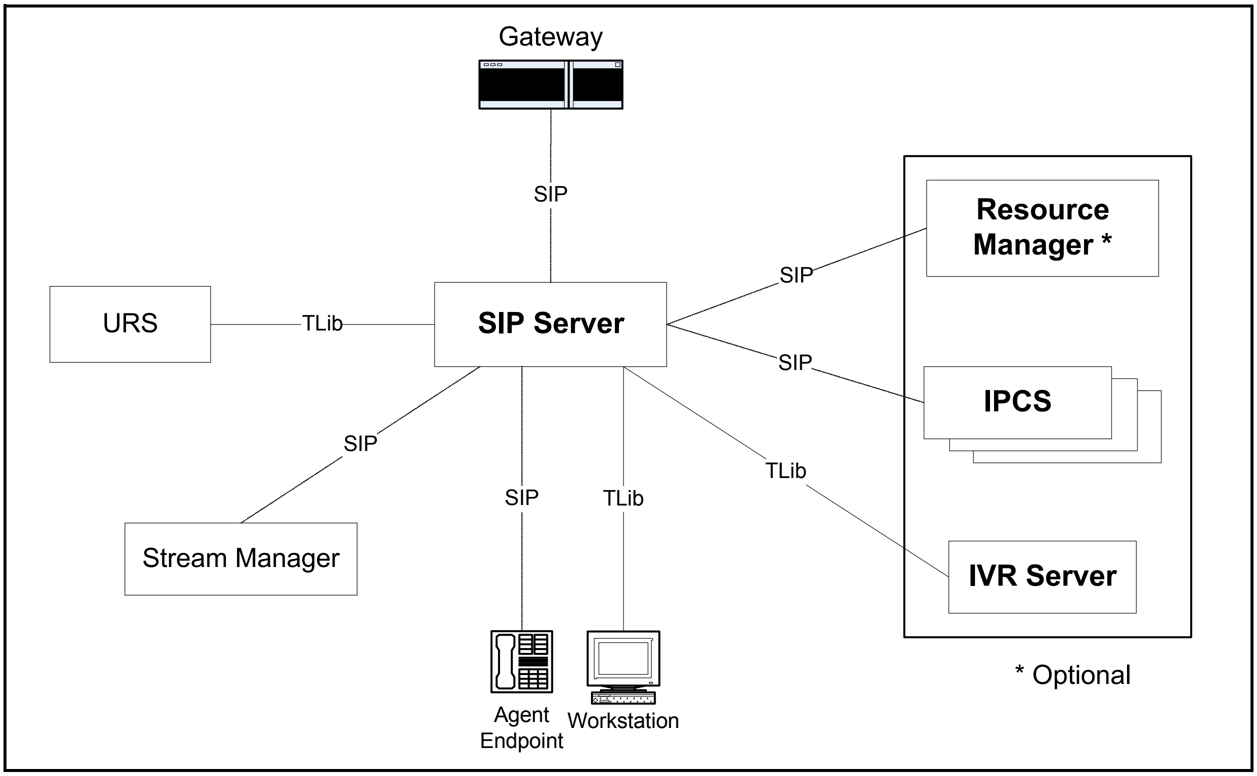

Figure 1: Behind Mode—Sample Architecture

Inbound Call Flows—Behind Mode

A typical inbound call flow can be separated into two phases: call delivery to GVP, and call delivery from GVP to an agent.

Call Delivery to GVP

1. A call is delivered from the gateway to the SIP Server.
2. The user name portion of the `INVITE` request Uniform Resource Identifier (URI) provides the destination for the call. This destination must exist as a Routing Point in Configuration Manager, and there must be a routing strategy loaded on the Routing Point.
3. The Universal Routing Server strategy requests (via SIP Server) a Resource Manager inquiry into available resources.
4. The SIP Server delivers the `INVITE` request to the Resource Manager.

5. The Resource Manager replies with a redirection response (`302 Moved`) to one of the IPCS instances.
6. The SIP Server returns information to the strategy, specifically about the IPCS to which the call should be redirected.
7. Based on the information provided by the SIP Server, the routing strategy selects one of the IPCS ports.
8. The SIP Server delivers the `INVITE` request to the selected IPCS instance.
9. If GVP is configured for Behind mode, then the following events also take place as part of establishing the call:
   a. GVP communicates with Genesys Framework, via the IVR Server, to obtain necessary call details—primarily the DNIS.
   b. GVP uses the DNIS provided by the IVR Server to establish the call and start the Voice XML application.

Note: Steps 3 to 6 are optional. Because the IPCS ports are configured in Configuration Manager, you can define the routing strategy so that it selects a port directly, without needing to query the Resource Manager. However, in this case you lose the benefits provided by the Resource Manager. For example, IPCS selection based on specific resources, such as Automatic Speech Recognition.

Call Delivery to an Agent

Call queuing can be performed by using either of the following methods:

- Call Queuing on GVP
- Call Queuing on SIP Server Using Stream Manager

Call Queuing on GVP

1. The GVP script invokes the routing block on behalf of a Virtual Routing Point.
2. The routing strategy loaded on the Virtual Routing Point selects the appropriate voice treatment.
3. The IVR Server returns the selected voice treatment to the GVP script.
4. When an agent becomes ready, the routing strategy instructs the IVR Server to route the call.
5. The IVR Server instructs the SIP Server to transfer the call to the selected agent’s phone. For more information about transfer methods, see the *note* at the end of this section.
6. The SIP Server sends an `INVITE` request to the agent’s phone, after which the SIP dialog with the IPCS is terminated.

Call Queuing on SIP Server Using Stream Manager

1. After the IVR portion of the call is completed, the GVP script requests a call transfer to a SIP Server Routing Point. For more information about transfer methods, see the *note* at the end of this section.
2. The SIP dialog with the IPCS is terminated.
3. The URS routing strategy returns the voice treatment that is to be applied to the call.
4. The SIP Server invokes Stream Manager.
5. When an agent becomes ready, the routing strategy instructs the SIP Server to route the call.
6. The SIP Server sends an `INVITE` request to the agent's phone, after which the SIP dialog with Stream Manager is terminated.

Note: The Behind-mode configuration supports three possible call transfer methods: IPCS using requests (`Transfer on Platform`), IPCS using a re-`INVITE` request (also called a *bridged transfer*), and IVR Server using `TSingleStepTransfer` requests.

In-Front Mode—Architecture and Call Flows

Figure 2 shows a sample architecture for a GVP–SIP Server integration in the In-Front mode. The sample includes the Alcatel Softswitch 5020 in the configuration, though a third-party soft switch is not required. Note that the IVR Server, though physically behind the switch, is configured for the In-Front mode.

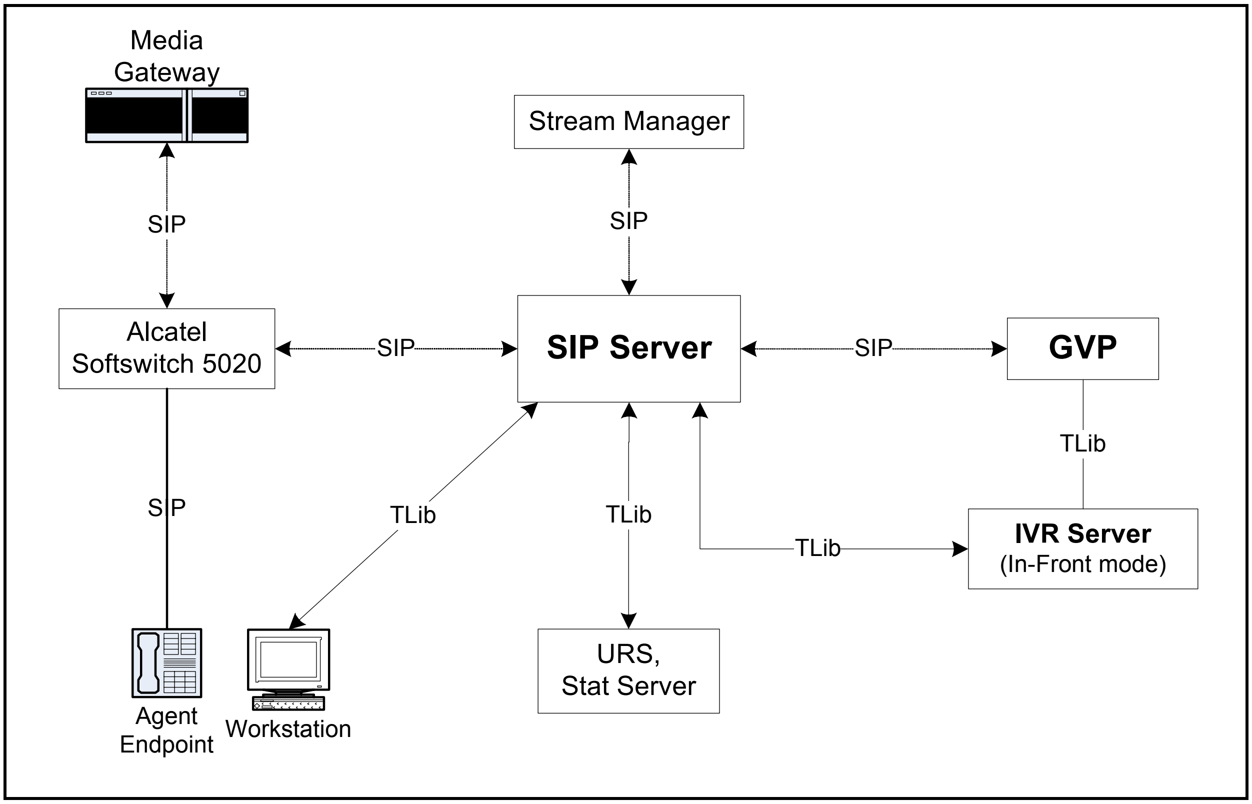

Figure 2: In-Front Mode—Sample Architecture

Inbound Call Flows—In-Front Mode

A typical inbound call flow can be separated into two phases: A) call delivery to GVP, and B) call delivery from GVP to the agent, and any transfers that take place after that. Delivery to GVP is the common start for all call scenarios.

Call Delivery to GVP

1. A call is delivered from the gateway to the SIP Server.
2. The user name portion of the `INVITE` request URI provides the destination for the call. This destination must exist as a Routing Point in Configuration Manager, and there must be a routing strategy loaded on the Routing Point.

3. The Universal Routing Server strategy requests (via SIP Server) a Resource Manager inquiry into available resources.
4. The SIP Server delivers the `INVITE` request to the Resource Manager.
5. The Resource Manager replies with a redirection response (`302 Moved`) to one of the IPCS instances.
6. The SIP Server returns information to the strategy, specifically about the IPCS to which the call should be redirected.
7. Based on the information provided by the SIP Server, the routing strategy selects one of the IPCS ports.
8. The SIP Server delivers the `INVITE` request to the selected IPCS instance.

Call Delivery from GVP to an Agent

If agents are registered on the SIP Server, call delivery to an agent can be accomplished using any of the following methods:

- GVP routes the call directly to SIP Server through the Internet Protocol Contact Center (IPCC) platform. The destination for the call can be either an agent or a routing point.
- GVP initiates a bridge transfer, in which case GVP remains in the call path.
- GVP uses IVR Server to invoke a routing strategy that finds an agents on the SIP Server. A platform based transfer (through IPCC) routes the call to the destination agent.

If agents are registered to another switch, managed by a separate TServer, then GVP uses the IVR Server to trigger a routing strategy that then finds an available agent on the other switch. In this case, Inter Server Call Control (ISCC) must be configured between the IVR Server and the other switch.

As in the Behind-mode integration, call queuing can take place on either GVP or on SIP Server using Stream Manager. The call flows for these options are almost identical to the Behind mode (see “Call Delivery to an Agent” on page 18), except that for call queuing on GVP, a Routing Point is used instead of a Virtual Routing Point.

Stand-Alone Mode—Architecture and Call Flows

Figure 3 shows a sample architecture for a GVP–SIP Server integration in the Stand-Alone mode.

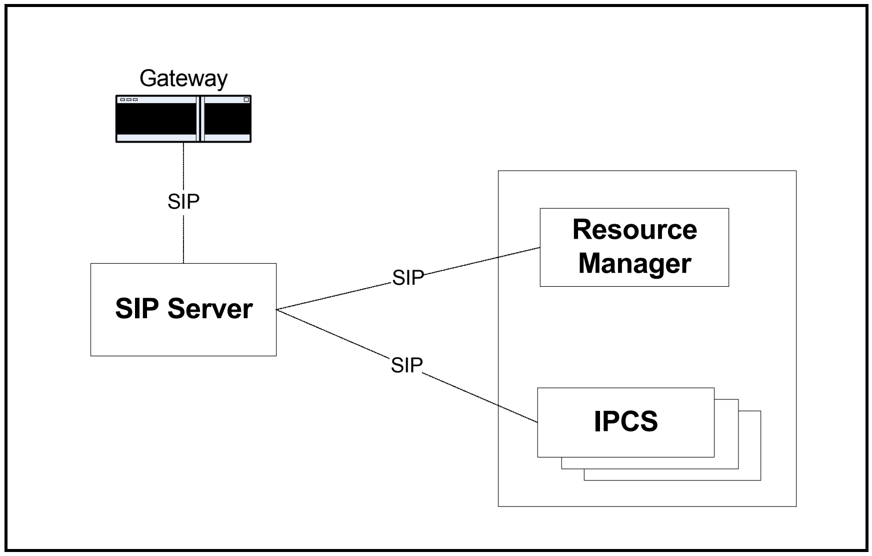

Figure 3: Stand-Alone Mode—Sample Architecture

Note: This sample architecture does not support outbound calls from GVP: neither regular outbound calls, nor transfers in bridged mode are supported.

Inbound Call Flows—Stand-Alone Mode

In Stand-Alone mode, a typical call flow consists of call delivery to GVP. Stand-Alone mode does not support call transfers out of GVP.

Call Delivery to GVP

1. A call is delivered from the gateway to the SIP Server.
2. The user name portion of the `INVITE` request URI provides the destination for the call. This destination is matched to a `Trunk` directory number (DN) that points to the Resource Manager.
3. The SIP Server sends an `INVITE` request to the Resource Manager.
4. The Resource Manager replies with a redirection (`302 Moved`) response to one of the IPCS instances.
5. The SIP Server sends the `INVITE` request to the selected IPCS instance.

High Availability

High Availability ensures that a service is not interrupted in the event of a failure or a process restart. Genesys SIP Server supports High Availability through Network Load Balancing Services (NLBS) and the GVP Resource Manager currently supports High Availability through the Microsoft Windows Cluster Service.

The High Availability feature for GVP Resource Manager is available when both primary and backup Resource Manager hosts are deployed with Microsoft Windows Cluster Service.

The High Availability feature for SIP Server is available when both primary and backup SIP Server hosts are deployed with NLBS.

It is important to ensure that, for the combined integrated solution to utilize the High Availability feature, the GVP Resource Manager and the SIP Server must be installed on separate hosts, with both the GVP Resource Manager and the SIP Server independently deployed in a manner to support the High Availability feature.

When deployed in a High Availability configuration, the virtual IP address is provided by the associated Cluster Service for the GVP Resource Manager.

Limitations

When using SIP Server, GVP, and Outbound Contact Server to perform outbound calls, the following limitations occur:

- GVP attempts to use a busy DN.
- SIP Server reports that an outbound call is not valid, but GVP reports that the same call is valid.

These limitations occur because SIP Server physically maps available DNs to a GVP virtual ports group. GVP can assign the DNs to these virtual ports, but cannot send any DN information back to SIP Server in an outbound call scenario.

Limitations for GVP 7.5 Only

- GVP 7.5 does not support an incoming SIP re-INVITE message in the following scenario:
  - SIP Server performs a mute transfer call to an agent using the REFER method.
  - The agent attempts to bridge GVP with the call.
  - SIP Server uses Stream Manager to initiate a conference call with GVP.
- GVP 7.5 will release a call when an agent places a call on hold, or attempts to perform a bridged transfer call.

Chapter

2 GVP-SIP Server Integration—Behind Mode

This chapter describes how to integrate the Genesys Voice Platform (GVP) with the SIP Server, in the Behind mode. It includes the following sections:

Task Flow—Behind Mode

Table 1 provides an overview of the main steps required to integrate SIP Server 7.5 with GVP in the Behind Mode. To help coordinate values across multiple components, fill in the Integration Worksheet (see "Integration Worksheet—Behind Mode" on page 85) before you begin.

Table 1: Taskflow to integrate the SIP Server with GVP—Behind Mode

Objective	Related Procedures and Actions
1. Configure the SIP Server for Behind mode integration.	Configure the SIP Server options as follows: • Set `override-to-on-divert` to `false`. • Set `event-ringing-on-100trying` to `true`. • Set `handle-vsp` to `all`. For the detailed procedure, see Configuring the SIP Server for Behind mode, page 27
2. Set the IVR Server application to Behind.	Create a `Virtual Switch for IVR In-Front`. No DNs or Agents required. For the detailed procedure, see Setting the IVR Server to Behind mode, page 29.

Table 1: Taskflow to integrate the SIP Server with GVP—Behind Mode (Continued)

Objective	Related Procedures and Actions
3. Enable call queuing on GVP (optional).	If you want to use call queuing on GVP, on the SIP Server switch, create at least one `Virtual Routing Point` DN for the `IVR` object. Then, in the `IVR` object, add an option pointing to this new DN. For the detailed procedure, see Enabling call queuing on GVP (optional), page 30.
4. Ensure that the gateway uses `INVITE` requests for transferring calls out of IPCS.	On the `Trunk` DN for the gateway, set the `refer-enabled` option to `false`. • Disabling REFER requests for call transfers, page 31
5. Configure IPCS ports.	To configure IPCS ports as `Voice Treatment Port` type DNs, complete the following procedures: 1. Creating Voice Treatment Port DNs, page 32 2. Configuring a Place object for each IPCS port, page 34 3. Configuring a Place Group for each IPCS, page 35
6. Create an `IVR` object for the GVP instance.	Create the `IVR` object, adding a port for every `Voice Treatment Port` DN that you created for IPCS. • Creating an IVR object for the GVP instance, page 35
7. Configure the IPCS for GVP Behind mode (optional).	The IPCS defaults to the GVP In-Front mode; no further configuration is needed for that mode. However, to configure the IPCS to run in GVP Behind mode, you must: • Set the Call Flow Assistant to use the CTI client for ANI and DNIS. • Set the database URL to the `GenericDID.xml` file. For the full procedure, see Configuring the IPCS for GVP Behind mode (optional), page 37
8. Set the IVR Server mode in GVP.	In the EMPS interface for GVP, set the IVR Server mode to `Behind`, and configure the mode to `UseDNIS` for new call messages. • Configuring the IVR Server Client for the GVP Behind mode, page 38

Table 1: Taskflow to integrate the SIP Server with GVP—Behind Mode (Continued)

Objective	Related Procedures and Actions
(Optional) Enable the Auto-Login feature.	To use the Auto-Login feature offered by the IVR Server, you must: • Create Agent Logins and Persons in addition to the existing `VTP` DNs, as well as Places for these new Persons. • Create an `Agent Group` object and use this as the target in the routing strategy (replacing the Place Groups in the sample routing strategy (see Figure 14 on page 41)). For the detailed procedure, see Enabling the Auto-Login feature (optional), page 39.
(Optional) Configure the Resource Manager interaction.	If you want to route calls based on specific resources—for example, Automatic Speech Recognition—you must include the Resource Manager in your routing strategy. • Create the inbound routing strategy, targeting one of the IPCS Place Groups. • Configure the `Play application` block with the following parameters: `GSIP_APP_ID` (with a value of `501`), and `GSIP_RM_URI` (with the SIP host and port as the value) For the detailed procedure, see Configuring the Resource Manager interaction (optional), page 40.

Procedures—Behind Mode

This section provides detailed procedures for configuring the various components and applications required for the GVP–SIP Server integration in the Behind mode.

Procedure:
Configuring the SIP Server for Behind mode

Purpose: To perform additional configuration of the SIP Server application in preparation for the GVP–SIP Server integration.

Start of procedure

1. In Configuration Manager, open the SIP Server `Application` object.
2. On the `Options` tab, in the `TServer` section, configure the options as described in Table 2.

Table 2: SIP Server Options—TServer Section

Option	Value	Description
override-to-on-divert	false	Set this option to `false` to ensure that the `INVITE` request sent to GVP contains the same user name in the `To` header as it did in the original `INVITE` request. **Note**: The originally dialled number (Routing Point) must exist on GVP as a DID number.
event-ringing-on-100trying	true	Set this option to `true` to force the SIP Server to generate an `EventRinging` message without waiting for the `180 Ringing` message from the IPCS. **Note**: For proper synchronization with the IVR Server application, this option must also be set at the DN level. See Table 5.

3. In the `extrouter` section, configure the `handle-vsp` option as described in Table 3.

Table 3: SIP Server Options—Extrouter Section

Option	Value	Description
handle-vsp	all	If agents are located on a T-Server other than the SIP Server, and call queuing takes place on the GVP side, then setting this option to `all` ensures that ISCC messages flow properly between the SIP Server and the remote T-Server.

End of procedure

Next Steps

- Setting the IVR Server to Behind mode

Procedure: Setting the IVR Server to Behind mode

Purpose: To configure the IVR Server and related `Application` objects for the `Behind` mode integration.

This procedure uses sample IVR Server and related `Application` objects for demonstration purposes. It provides key configuration details related to the `Behind` mode integration of the objects listed in Table 4. For more details about how to configure the IVR Server, see the *IVR Interface Option 7.5 IVR Server—System Administrator's Guide*.

Table 4: IVR Server and Related Application Objects

Application Object	Object Name in Sample Configuration
IVR Server	IServer-Behind-SIP-Config
TServer_IVR	IServer-Behind-SIP
SIP Server	TServer-SIP

Start of procedure

1. In the IVR Server `Application` object, on the `Connections` tab, add connections to TServer_IVR and the SIP Server (see Figure 4).

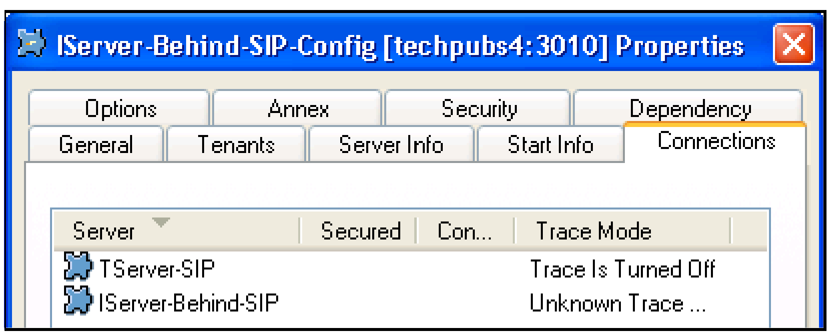

Figure 4: Adding Connections to TServer_IVR and the SIP Server

2. In the TServer_IVR `Application` object, on the `Switches` tab, click `Add` and create the `Virtual Switch for IVR In-Front` (see Figure 5).

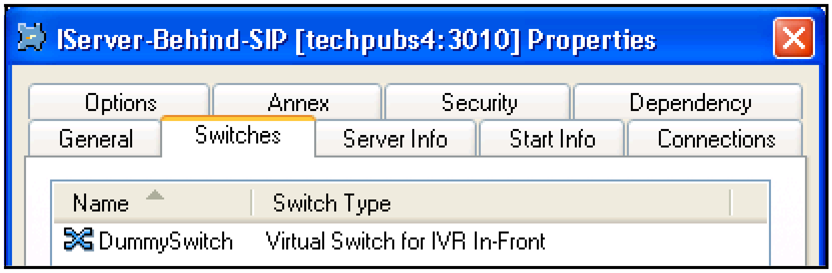

Figure 5: Creating a Virtual Switch for IVR In-Front

You do not need to define any DNs or Agent Logins for this switch.

Tip: Although you can use any type for this switch, Genesys recommends you use Virtual Switch for IVR In-Front, even though we are configuring the Behind mode.

End of procedure

Next Steps

- If you want call queuing to take place on GVP (instead of on Stream Manager), then continue at "Enabling call queuing on GVP (optional)" on page 30.
- If you are not configuring for call queuing on GVP, proceed to "Disabling REFER requests for call transfers" on page 31.

Procedure: Enabling call queuing on GVP (optional)

Purpose: To create the `Virtual Routing Point` DN on the SIP Server switch that the GVP voice application can use in the call routing phase. This is an optional configuration required for call queuing on GVP (instead of Stream Manager).

Start of procedure

1. Under the SIP Server `switch` object in the navigation tree, right-click the `DNs` folder, and create a new DN of the type `Virtual Routing Point`. The sample configuration uses the number `2700`.
2. In the IVR Server `Application` object, on the `Options` tab, create a new section called `VirtualRoutingPoints`.

3. In the `VirtualRoutingPoints` section, create a new option with a name that matches the name of your SIP Server. The option value must match the number that you gave your `Virtual Routing Point` DN in Step 1.

   The sample configuration uses an option called `TSserver-SIP`, with a value of `2700` (see Figure 6).

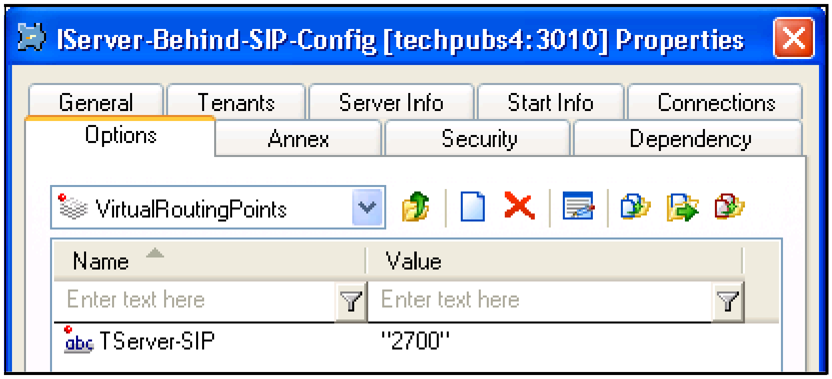

Figure 6: Adding a Virtual Routing Point

End of procedure

Next Steps

- Disabling REFER requests for call transfers

Procedure: Disabling REFER requests for call transfers

Purpose: To ensure that only the recommended re-`INVITE` request method is used to transfer calls out from the IPCS.

Start of procedure

1. In the `Trunk` DN created for the gateway, click the `Annex` tab.
2. If the `TServer` section does not already exist, create it.
3. In the `TServer` section, create a new option called `refer-enabled`, and set the value to `false`.

   If your gateway supports `REFER` messages, setting this option to `false` prevents the use of `REFER`, ensuring that only re-`INVITE` requests are used to transfer calls.

End of procedure

Next Steps

- Creating Voice Treatment Port DNs

Procedure:
Creating Voice Treatment Port DNs

Purpose: To create and configure DNs of the type `Voice Treatment Port` on the SIP Server switch. Configure as many Voice Treatment Ports as the number of declared ports on each IPCS.

Prerequisites

- To identify the number of ports available on GVP, check the `Max Channels` value in the Element Manager Provisioning System (EMPS). For the detailed procedure, see "Checking available ports on GVP" on page 48.

Start of procedure

1. Under your SIP Server `switch` object in the navigation tree, right-click the `DNs` folder and select `New > DN`.
2. Select `Voice Treatment Port` as the type.
3. Assign the new DN a number that matches the value of the PopGateway `ChannelIDStart` option.

   Tip: After numbering the first DN (to match the `ChannelIDStart` option), use continuous numbering (increments of one) when naming all subsequent DNs that you create.

4. In the `Annex` tab, create a `TServer` section and add new options as shown in Figure 7. Table 5 provides detailed descriptions of these options.

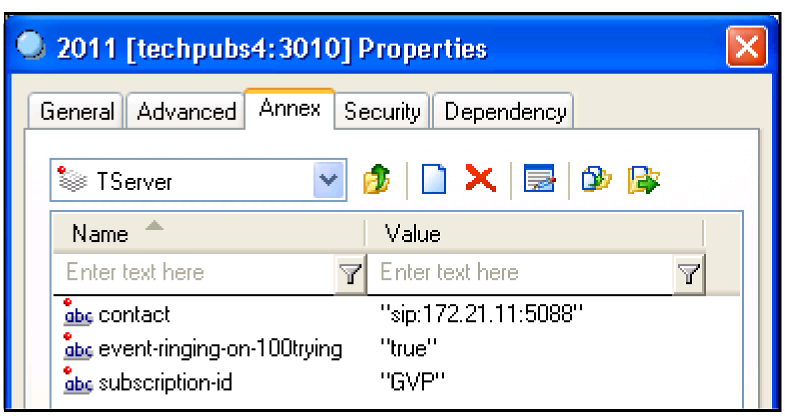

Figure 7: Voice Treatment Port DN—Annex Tab

Table 5: Voice Treatment Port DN Options—TServer Section

Option	Value
contact	Enter the SIP address and port for the IPCS.
event-ringing-on-100trying	Set this option to `true` to force the SIP Server to generate an `EventRinging` message without waiting for the `180 Ringing` message from the IPCS. **Note**: For proper synchronization with the IVR Server application, this option must also be set on the SIP Server. See Table 2.
subscription-id	If the configuration includes the Resource Manager, enter `GVP` as the identifier that the Resource Manager uses to subscribe the port with the SIP Server. **Note**: You must set the same value (`GVP`) for every IPCS port.

5. Repeat Steps 1 to 4, creating one `Voice Treatment Port` DN for every declared port on each IPCS. Use continuous numbering when naming the DNs—assign each DN a new number by an increment of one (see Figure 8).

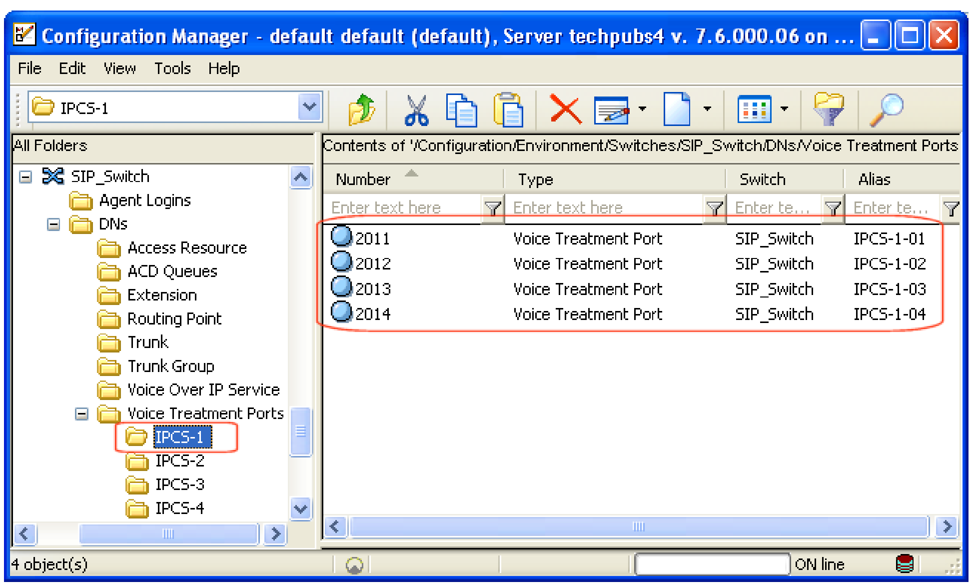

Figure 8: Voice Treatment Port DNs

Tip: To organize your `Voice Treatment Port` DNs according to their corresponding IPCS, consider creating a folder for each IPCS under the main `DNs` folder.

End of procedure

Next Steps

- Configuring a Place object for each IPCS port

Procedure:
Configuring a Place object for each IPCS port

Purpose: To create a `Place` object for every corresponding `Voice Treatment Port` DN in the `DNs` folder of the SIP Server switch.

Start of procedure

1. Under `Resources/Tenant`, right-click the `Places` folder, and select `New > Place`.
2. In the `New Place` dialog box, enter a name for your new Place.

   Tip: Use the same numbering scheme for your Places as you did for the `Voice Treatment Port` DNs.

3. Right-click the newly created Place and select `New > Shortcut to DN`.
4. Browse to the `DNs` folder of the SIP Server switch and select the corresponding `Voice Treatment Port` DN (see Figure 9).

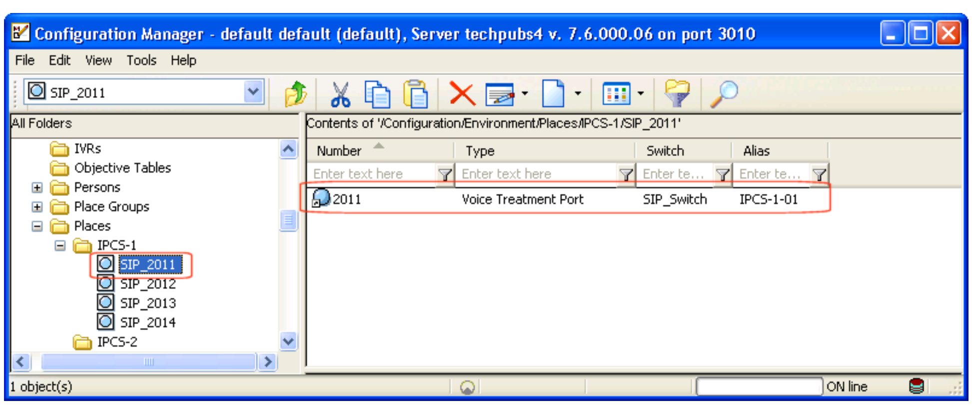

Figure 9: Places Associated with IPCS Ports

5. Repeat Steps 1 to 4 for every newly created `Voice Treatment Port` DN in the `DNs` folder of the SIP Server switch.

End of procedure

Next Steps

- Configuring a Place Group for each IPCS

Procedure: Configuring a Place Group for each IPCS

Purpose: To create the `Place Group` object that the routing strategy will use as the target for sending calls from the SIP Server to GVP.

Start of procedure

1. Under `Resources`, right-click the `Place Groups` folder and select `New > Place Group`.
2. Enter a name for the `Place Group`, and then click `OK`.

   > **Tip:** Although there are no required naming rules for the Place Group, the routing strategy must select one of the groups based on the reply from the Resource Manager. As a naming convention, Genesys suggests that you use the IP address of the targeted IPCS, with the dot (.) separators removed.
   >
   > For example, if IPCS-1 is located at 172.21.11.15, name the related Place Group `172211115`.

3. Right-click the newly created `Place Group` object, and select `New > Shortcut to Place`.
4. Browse to the `Places` folder, select all of your newly created `Place` objects, and then click `OK`.

End of procedure

Next Steps

- Creating an IVR object for the GVP instance

Procedure: Creating an IVR object for the GVP instance

Purpose: To create an `IVR` object that represents the whole GVP instance.

Summary

1. Create the IVR object (Step 1).
2. Create IVR ports associated with the Voice Treatment Port DNs that you created on the SIP Server switch (Step 2).

Start of procedure

1. Create the IVR object for the whole GVP instance:
   a. In Configuration Manager, right-click the IVRs folder and select New > IVR.
   b. In the Name text box, enter a name that represents the whole GVP instance. The sample configuration uses the name GVP.
   c. From the Type drop-down list, select Genesys Voice Platform.
   d. Click the folder icon next to the IVR Interface Server text box, browse for the IVR Server, and then click OK (see Figure 10).

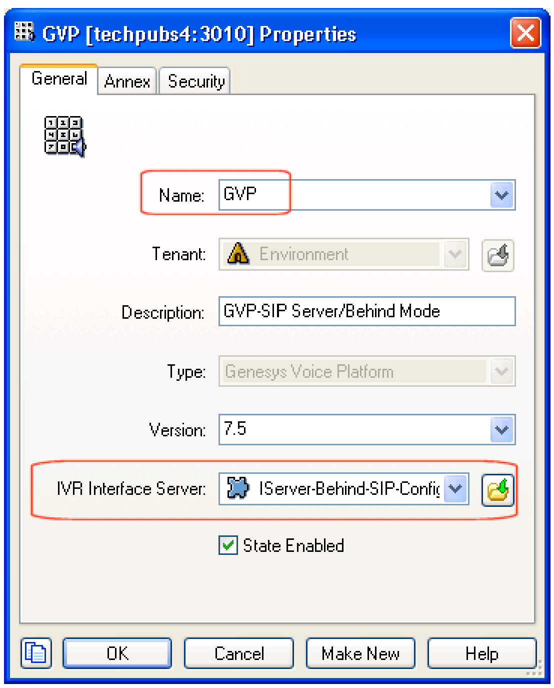

Figure 10: IVR Object—General Tab

> **Tip:** Select the IVR Server object that you configured in "Setting the IVR Server to Behind mode" on page 29.
>
> The sample configuration uses IServer-Behind-SIP-Config.

2. Create ports in the IVR object that correspond to the Voice Treatment Port DNs in the SIP Server switch:
   a. Under Resources/Tenant, right-click your IVR object and select New > IVR Port.

b. In the `New IVR Port Properties` dialog box, enter a port number that matches the corresponding `Voice Treatment Port` DN on the SIP Server switch.

c. In the `Associated DN` drop-down list, select the corresponding `Voice Treatment Port` DN from the SIP Server switch, and then click `OK`.

d. Click `OK`.

e. Repeat Steps a to d until you have created one IVR port for every `Voice Treatment Port` DN in the `DNs` folder of the virtual switch (see Figure 11).

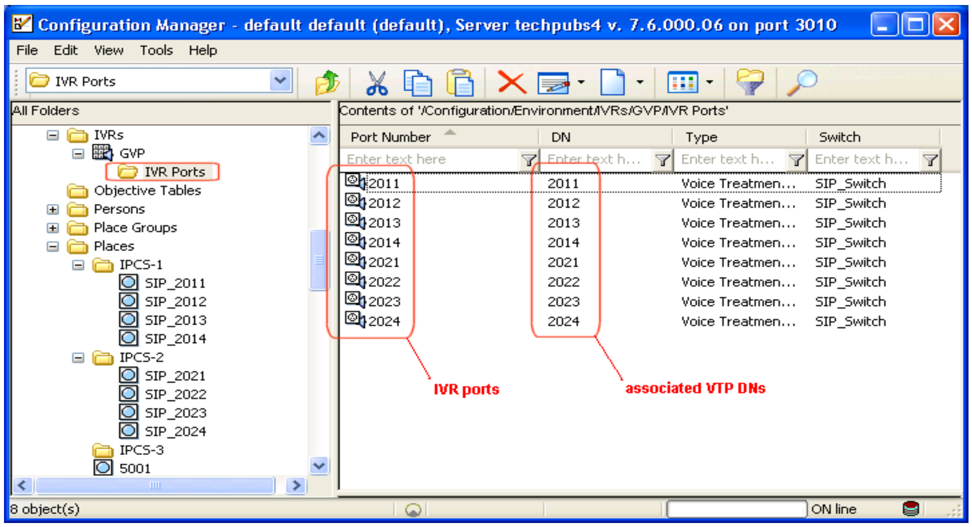

Figure 11: IVR Ports with Corresponding VTP DNs

End of procedure

Next Steps

- If you are using a GVP IPCS configured for the Behind mode, some additional configuration steps are required. Continue at "Configuring the IPCS for GVP Behind mode (optional)".
- For the GVP In-Front mode (recommended), no additional steps are required, as GVP defaults to this mode. Continue at "Configuring the IVR Server Client for the GVP Behind mode" on page 38.

Procedure: Configuring the IPCS for GVP Behind mode (optional)

Purpose: To configure the IPCS for GVP Behind mode. This configuration requires that you A) set the Call Flow Assistant (CFA) to use the computer-telephony integration (CTI) client for ANI and DNIS, and B) set the database URL in the PopGateway to the `GenericDID.xml` file.

Note: “Behind” in this instance applies to GVP only, not to the overall GVP–SIP Server integration. For a detailed explanation of how the GVP modes fit into the more general integration modes, see “What Is the GVP–SIP Server Integration?” on page 14.

Start of procedure

1. In a web browser, enter the following URL:

   `http://<EMPS-hostname>:9810/spm`

2. In the EMPS navigation tree, expand the nodes `Servers > IPCS > <server name> > CFA`, and then set `useclientforanidnis` to the value of `1`.
3. Expand the nodes `Servers > IPCS > <server name> > Popgateway`, and for the `databaseurl` on the `IVR` tab, enter the following address:

   `http://localhost:9810/did_url_mappings/GenericDID.xml`

End of procedure

Next Steps

- Configuring the IVR Server Client for the GVP Behind mode

Procedure: Configuring the IVR Server Client for the GVP Behind mode

Purpose: To set the IVR Server Client mode to `Behind` and configure the mode to include the DNIS in the called number field of new call messages.

Start of procedure

1. In the EMPS navigation tree, expand the `Reseller` object node, and then expand the node for the reseller to which the customer belongs.
2. Right-click the customer that you want to provision, and select `Provision`.

   The customer provisioning property pages open, starting with `Policy`.

3. On the `GenesysCTI` tab, select the `IVR Svr Client Active` check box. This assigns an active status to the IVR Server Client.
4. From the `Primary IVR Svr Client Machine` drop-down list, select the host machine for your primary IVR Server Client (see Figure 12).

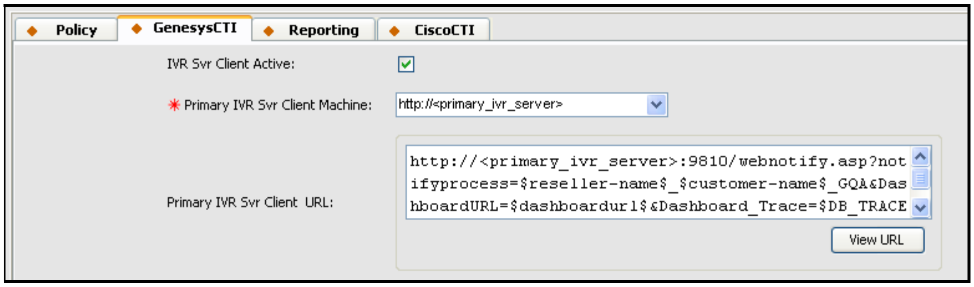

Figure 12: Activating the Primary IVR Server Client

5. From the `IVR Server Mode` drop-down list, select `Behind`, and from the `Called Number` drop-down list, select `UseDNIS` (see Figure 13).

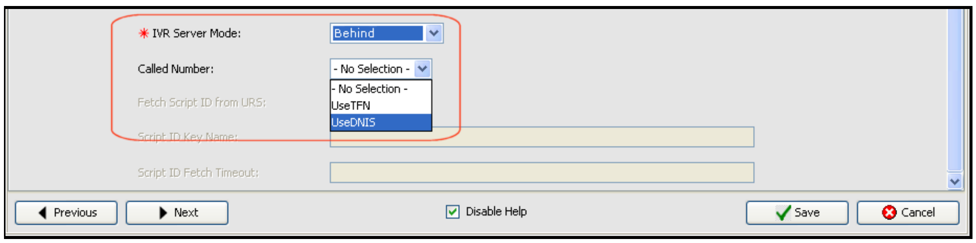

Figure 13: Selecting the IVR Server Mode and Called Number

End of procedure

Next Steps

- If you want to use the Auto-Login feature offered by the IVR Server, continue at "Enabling the Auto-Login feature (optional)".
- If your configuration includes the Resource Manager component, continue at "Configuring the Resource Manager interaction (optional)".
- Otherwise, you have completed all the required steps for the GVP–SIP Server Behind-mode integration.

Procedure: Enabling the Auto-Login feature (optional)

Purpose: To enable the Auto-Login feature offered by the IVR Server. This feature enables your IVR ports to log in as agents. It requires `Person`, `Agent Login`, and `Agent Group` objects.

Start of procedure

1. Create an `Agent Login` object and associated `Person` objects for every IVR port that you want configured for Auto-Login.
2. On the `Annex` tab of the IVR port, create a new section called `AutoLogin`. Then, in this section, create a new option called `Agent Login`, setting the value to the ID of the agent that you want this IVR port to use when logging in.
3. Repeat Step 2 for every IVR port that you intend to configure for Auto-Login.
4. Create an `Agent Group` object and add shortcuts to all of the `Person` objects that you created in Step 1.
5. Modify the routing strategy so that it uses this Agent Group as the target, instead of the Place Group created in earlier steps.

End of procedure

Next Steps

- If your configuration includes the Resource Manager component, continue at Configuring the Resource Manager interaction (optional), page 40.
- Otherwise, you have completed all the required steps for the GVP–SIP Server Behind-mode integration.

Procedure: Configuring the Resource Manager interaction (optional)

Purpose: To integrate the Resource Manager component into the GVP–SIP Behind-mode configuration, via the Universal Routing Server (URS) strategy.

Note: This procedure is optional. If your configuration does not include Resource Manager, you should design the routing strategy to target a specific IPCS port instead of a Place Group. However, in this case you lose the benefits provided by Resource Manager—for example, IPCS selection based on specific resources, such as Automatic Speech Recognition.

This procedure provides basic information about using the Interaction Routing Designer (IRD) in the context of configuring the Resource Manager interaction. For more information about creating routing strategies with IRD, see the *Universal Routing 7.6 Deployment Guide*.

Start of procedure

1. Using IRD, create a routing strategy that targets one of the IPCS Place Groups. Figure 14 shows the minimum strategy required for delivering an inbound call to GVP.

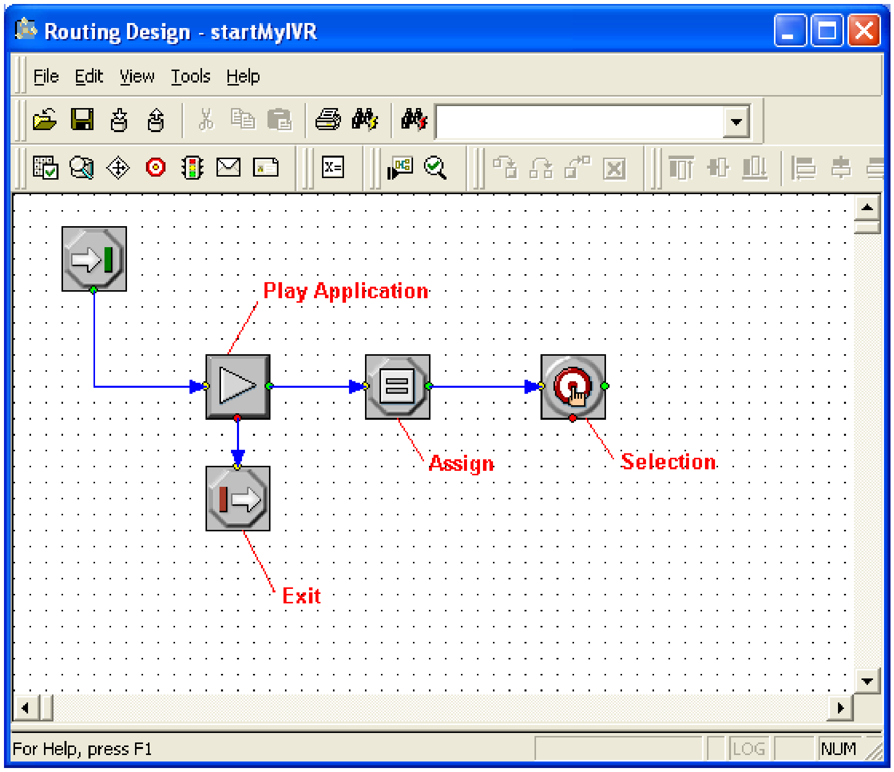

Figure 14: Sample Routing Strategy

> **Tip:** The strategy should be built to accommodate the following flow:
>
> 1. The strategy requests a `PlayApplication` treatment.
> 2. The SIP Server sends an `INVITE` request to the Resource Manager.
> 3. The Resource Manager replies with a `302 Moved` response and IPCS information in the `Contact` header.
> 4. The SIP Server returns an `EventTreatmentEnd` message to the Universal Routing Server (URS).
> 5. The strategy analyzes the response parameters and decides which IPCS to target.

2. Configure the `PlayApplication` block with the `GSIP_APP_ID` and `GSIP_RM_URI` parameters (see Figure 15).

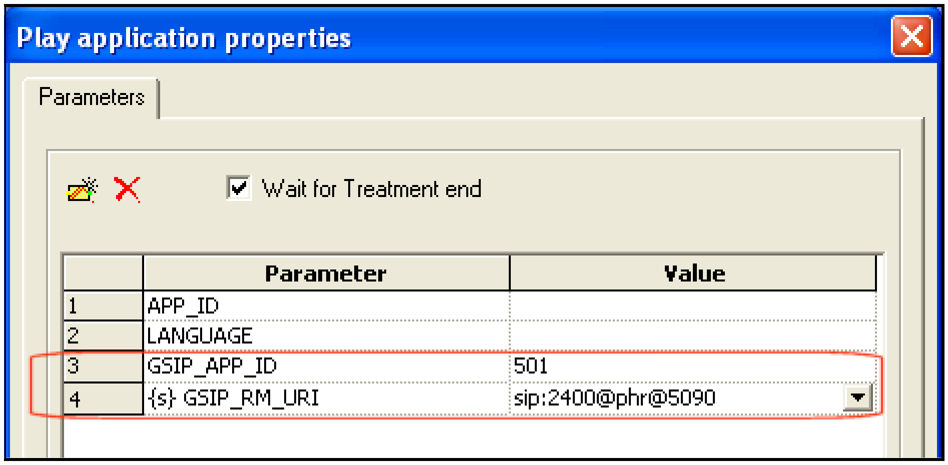

Figure 15: Play Application Block Properties

a. For `GSIP_APP_ID`, enter a value of `501`.

b. For `GSIP_RM_URI`, enter the fully qualified SIP URI attribute that specifies where the Resource Manager can be contacted. Use the following format for this parameter:

`sip:<user name>@<host>:<port>`

The SIP Server uses the value of this parameter to send an `INVITE` request to the Resource Manager

> **Tip:** To force URS to read the value of any parameter as a string (instead of as an integer), add the prefix `{s}` to the parameter name. For example, `{s}GSIP_RM_URI`.

3. Configure the `Assign` block so that when the SIP Server generates the `EventTreatmentEnd` and `AttributeUserData` messages, they contain the following two keys:

- `GSIP_APPDATA_RM_IPCSID`—This key identifies the IPCS selected by the Resource Manager. The value is the IP address of the IPCS, with the dot (.) separators removed. For example, if the IPCS is located at 172.21.11.15, the key-value is `172211115` (see Figure 16).

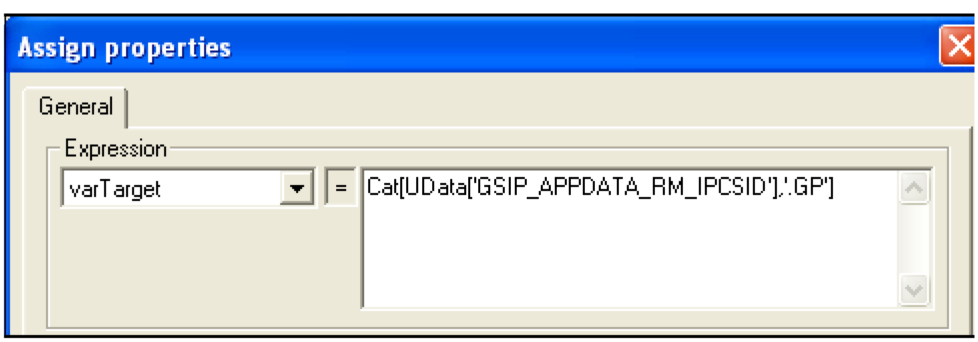

Figure 16: Using the GSIP_APPDATA_RM_IPCSID User Data

- `GSIP_APPDATA_RM_IPCSCONTACT`—This key specifies the full value of the `Contact` header returned by the Resource Manager in the `302 Moved` response.

  Tip: Typically, the strategy can determine which IPCS to target using only the `GSIP_APPDATA_RM_IPCSID` key, provided you named your Place Groups according to the recommended convention—the IP address for the target IPCS, with the dot (.) separators removed.

4. Configure the `Selection` block to target one of the IPCS Place Groups (see Figure 17).

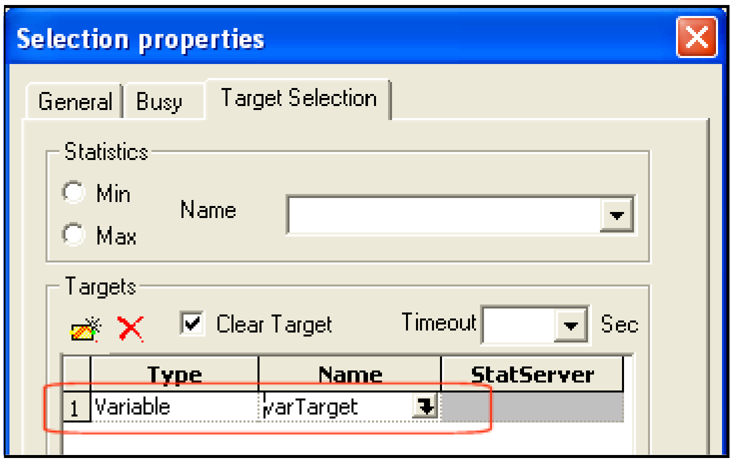

Figure 17: Targeting One of the IPCS Place Groups

End of procedure

Next Steps

- You have completed all the required steps to integrate the SIP Server 7.5 with GVP in the Behind mode.

Chapter

3 GVP-SIP Server Integration—In-Front Mode

This chapter describes how to integrate the Genesys Voice Platform (GVP) with the SIP Server, in the In-Front mode. It includes the following sections:

Task Flow—In-Front Mode

Table 6 provides an overview of the main steps required to integrate SIP Server 7.5 with GVP in the In-Front mode. Complete all steps in the order listed. To help coordinate values across multiple components, fill in the Integration Worksheet (see "Integration Worksheet—In-Front Mode" on page 83) before you begin.

Table 6: Taskflow to Integrate SIP Server with GVP—In-Front Mode

Objective	Related Procedures and Actions
1. Confirm that the IVR Server for GVP is in place.	These procedures assume that the IVR Server for GVP is already installed in the In-Front mode, and the necessary configuration objects have been created. These objects include: • Switching office of the `Virtual Switch for IVR In-Front` type. • IVR `Application` object. • IVR Server `Application` object. • TServer IVR `Application` object. For more information about these objects, see the *IVR Interface Option 7.5 IVR Server System Administrator's Guide*.
2. Configure the Virtual Switch.	In the virtual `Switch` for GVP, create as many `Voice Treatment Port` (VTP) DNs as there are available ports on GVP. IVR Server uses the `Voice Treatment Port` DNs to calculate the maximum number of calls it can send to GVP. For detailed procedures, see: 1. Checking available ports on GVP, page 48 2. Creating Voice Treatment Port DNs, page 50
3. Configure Framework resources.	Under `Tenant/Resources` in Configuration Manager, create the `Place` and `Place Group` objects to be used for routing calls to GVP. 1. Creating Place objects for each VTP DN, page 50 2. Creating a Place Group object and adding Places, page 51
4. Configure the `IVR` object.	In the `IVR` object, create as many ports as there are `Voice Treatment Port` DNs in the virtual switch for GVP. 1. Creating IVR ports, page 52 2. Linking the IVR object to the IVR T-Server, page 53
5. Set the IVR Server mode in GVP.	In the EMPS interface for GVP, set the IVR Server mode to `In Front` and configure the mode to `UseDNIS` for new call messages. • Configuring the IVR Server Client for the In-Front mode, page 54

Table 6: Taskflow to Integrate SIP Server with GVP—In-Front Mode (Continued)

Objective	Related Procedures and Actions
6. Point the GVP IPCS to the SIP Server.	1. In EMPS, enter the SIP Server IP address for `SessionMgrIPAddr1`. 2. If the SIP Server is not on the default port (`5060`), enter the current SIP Server port for `SessionMgrSipPort1`. For the detailed procedure, see "Pointing the GVP IPCS to the SIP Server" on page 55
7. Configure the GVP Resource Manager (optional).	If your configuration includes the Resource Manager, point the IPCS to the Resource Manager IP address and port, and then assign the primary user agent of the Resource Manager to the SIP Server. 1. In EMPS, enter the IP address and port for `ResourceMgr`. 2. For `PrimaryUserAgent`, enter IP address and port for SIP Server. For the detailed procedure, see "Configuring the GVP Resource Manager" on page 56.
8. Configure access from the SIP Server to GVP.	Configure the `Access Code` that allows the SIP switch to communicate with the GVP virtual switch, then create the corresponding `Access Resource` DN in the virtual switch. • Configuring access from the SIP Server to GVP, page 57
9. Configure access from GVP to the SIP Server.	Configure the access code that allows the GVP virtual switch to communicate with the SIP switch, then create the corresponding `External Routing Point` DN on the SIP switch. • Configuring access from GVP to the SIP Server, page 60
10. Configure routing from the SIP Server to GVP.	Create the routing objects and routing strategy that the SIP Server uses to receive incoming calls from the Media Gateway. 1. Creating a Routing Point for incoming calls, page 61 2. Creating a routing strategy for incoming calls, page 62 3. Creating a Trunk Group DN, page 64
11. Configure routing from GVP to agents on the SIP Server.	Create the routing objects and routing strategy for sending incoming calls from GVP to agents on the SIP Server switch. 1. Creating a Routing Point for selecting an agent, page 66 2. Creating the voice self-service application, page 67 3. Creating the routing strategy on the virtual switch, page 68

Table 6: Taskflow to Integrate SIP Server with GVP—In-Front Mode (Continued)

Objective	Related Procedures and Actions
(Optional) Create a trunk on SIP Server for the Alcatel 5020 Softswitch.	If your architecture includes an Alcatel 5020 Softswitch, you must also complete the following additional actions: 1. Configure a trunk for the 5020 on the SIP Server switch. 2. If you want `REFER` requests routed through the 5020 soft switch, set `refer-enabled` on a DN of type `Trunk` to `true`. OR If you want SIP Server to handle the `REFER` requests itself, set the option to `false`. For the detailed procedure, see "(Optional) Creating a trunk on the SIP Server for the Alcatel 5020 Softswitch" on page 71.
(Optional) Remove the `To` header from `INVITE` requests.	Set `override-to-on-divert` in the SIP Server application to `true`. • Removing the "To" header from REFER requests, page 72
(Optional) Enable call matching with multiple transfers to and from GVP.	Set `match-call-once` in the SIP Server application to `false`. • Enabling call matching for multiple transfers in GVP, page 73

Procedures—In-Front Mode

This section provides detailed procedures for configuring the various components and applications required for the GVP–SIP Server integration in the In-Front mode.

Procedure:
Checking available ports on GVP

Purpose: To identify the number of ports available on GVP. This value lets you know how many Voice Treatment Ports you need to create on the virtual switch (or on the SIP Server switch in the Behind integration mode).

Start of procedure

1. To determine the number of available ports in your GVP application, check the `Max Channels` value in the Element Management Provisioning System (EMPS) interface. In a web browser, enter the following URL:

   `http://<EMPS-hostname>:9810/spm`

2. Log in as `Admin`.

3. In the navigation tree, expand the nodes `Servers > IP Communication Server > <server name>` > `Popgateway`, and then right-click `Route`.

4. On the `General` tab, note the value in the `Max Channels` text box (see Figure 18).

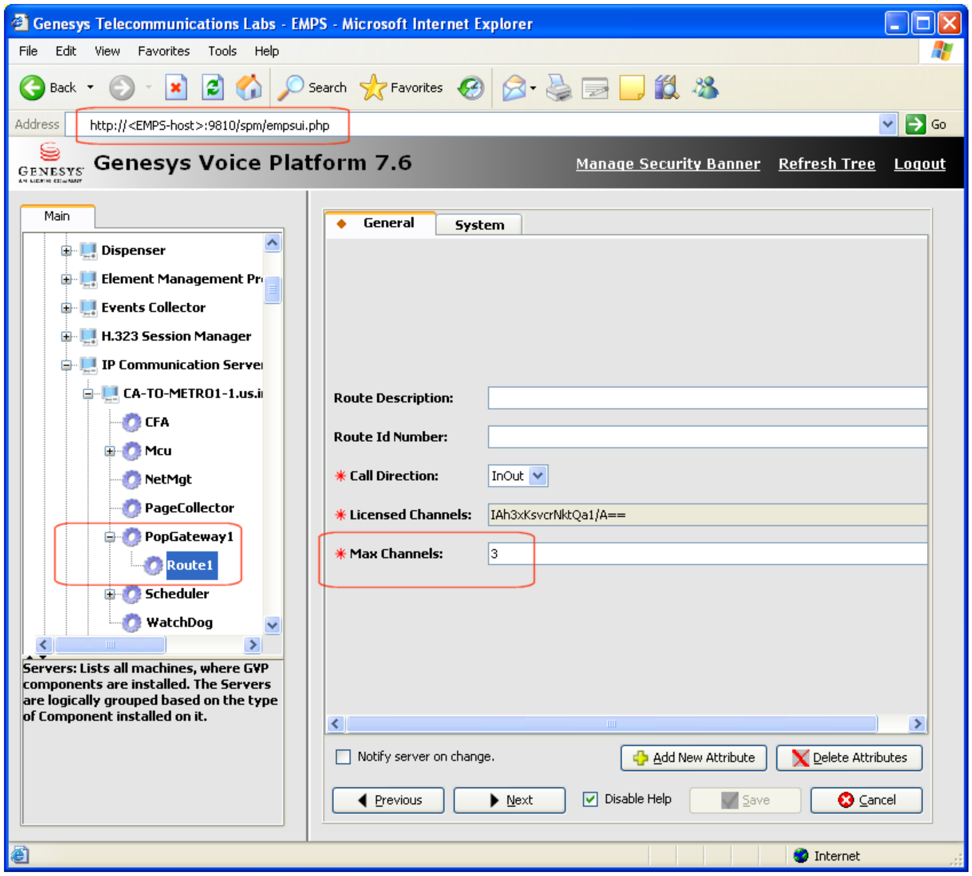

Figure 18: EMPS—Max Channels

The `Max Channels` value specifies the maximum number of ports that GVP can accept.

Tip: To help organize your integration, copy the `Max Channels` value that you find here onto your Integration Worksheet (see “Integration Worksheet—In-Front Mode” on page 83.

End of procedure

Next Steps

- Create as many DNs of the `Voice Treatment Port` type as you identified in the `Max Channels` text box. See “Creating Voice Treatment Port DNs” (or if you are implementing the Behind mode integration, see “Creating Voice Treatment Port DNs” on page 32).

Procedure: Creating Voice Treatment Port DNs

Purpose: To create and configure DNs of the `Voice Treatment Port` type on the virtual switch for GVP.

Start of procedure

1. In Configuration Manager, right-click the `DNs` folder for the virtual switch and select `New > DN`.
2. Select `Voice Treatment Port` as the type.
3. Create as many `Voice Treatment Port` DNs as there are available ports on GVP—the `Max Channels` value that you noted in Step 4 on page 49 of Checking available ports on GVP (see Figure 19).

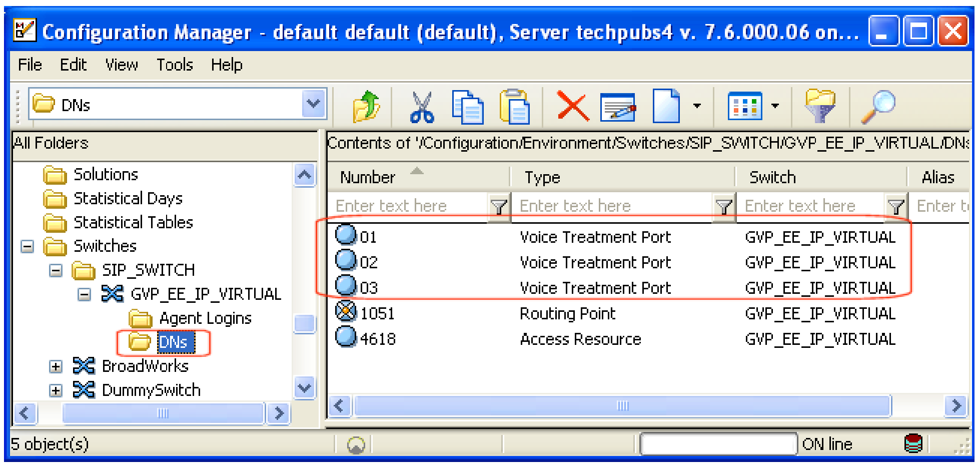

Figure 19: Voice Treatment Port DNs on the Virtual Switch

End of procedure

Next Steps

- Creating Place objects for each VTP DN

Procedure: Creating Place objects for each VTP DN

Purpose: To create a `Place` object for every corresponding `Voice Treatment Port` DN in the `DNs` folder of the virtual switch.

Start of procedure

1. Under `Resources`, right-click the `Places` folder and select `New > Place`.
2. In the `New Place` dialog box, enter a name for your new Place.

   Tip: Use the same numbering scheme for your Places as you did for the Voice Treatment Ports.

3. Right-click the newly created Place and select `New > Shortcut to DN`.
4. Browse to the `DNs` folder of the virtual switch and select the corresponding `Voice Treatment Port` DN.
5. Repeat Steps 1to 4 for every newly created `Voice Treatment Port` DN in the `DNs` folder of the virtual switch.

End of procedure

Next Steps

- Creating a Place Group object and adding Places

Procedure: Creating a Place Group object and adding Places

Purpose: To create a `Place Group` object that routing will use as the target for sending calls from the SIP Server to GVP.

Start of procedure

1. Under `Resources`, right-click the `Places` folder and select `New > Place Group`.
2. Enter a name for the `Place Group` object, and then click `OK`.

   The sample configuration uses the name `GVP_EE_IP_VIRTUAL`.
3. Right-click the newly created `Place Group` object, and then select `New > Shortcut to Place`.
4. Browse to the `Places` folder, select all of your newly created `Place` objects, and then click `OK`.

End of procedure

Next Steps

- Creating IVR ports

Procedure: Creating IVR ports

Purpose: To create ports in the `IVR` object that correspond to the `Voice Treatment Port` DNs in the virtual switch.

Start of procedure

1. Under `Resources`, right-click your `IVR` object, and select `New > IVR Port`.
2. In the `New IVR Port Properties` dialog box, enter a port number that matches the corresponding `Voice Treatment Port` DN in the virtual switch.
3. In the `Associated DN` text box, select the corresponding `Voice Treatment Port` DN from the virtual switch for GVP, and then click `OK` (see Figure 20).

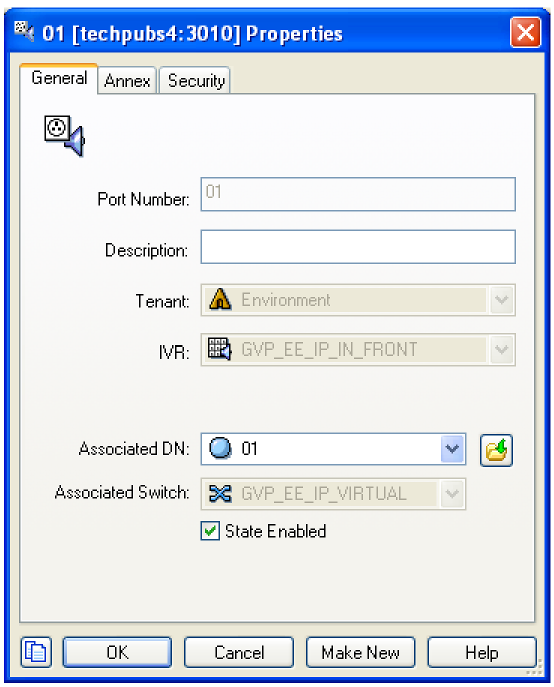

Figure 20: Adding IVR Ports—Properties Dialog Box

4. Click `OK`.
5. Repeat Steps 1 to 4 until you have created one IVR port for every `Voice Treatment Port` DN in the `DNs` folder of the virtual switch (see Figure 21).

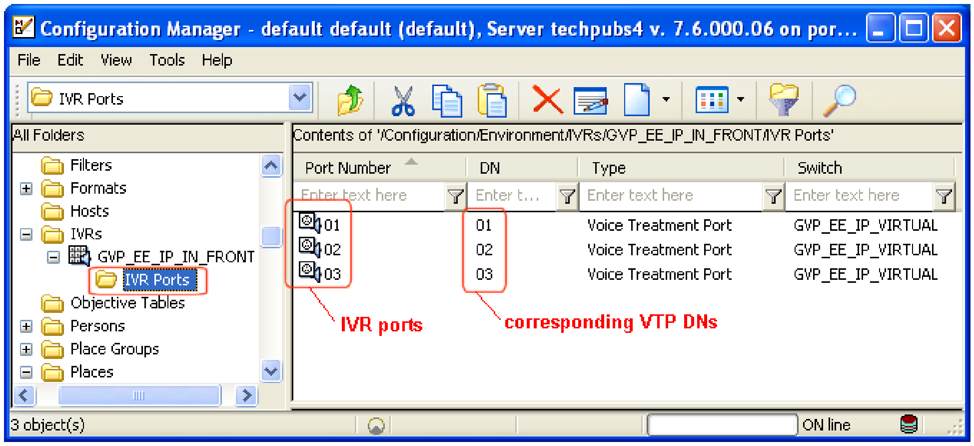

Figure 21: IVR Ports with Corresponding VTP DNs

End of procedure

Next Steps

- Linking the IVR object to the IVR T-Server

Procedure:
Linking the IVR object to the IVR T-Server

Purpose: To configure the link between the IVR object and the IVR T-Server.

Start of procedure

1. In the IVR Interface Server text box of the IVR Application object, select your IVR Server and click OK (see Figure 22).

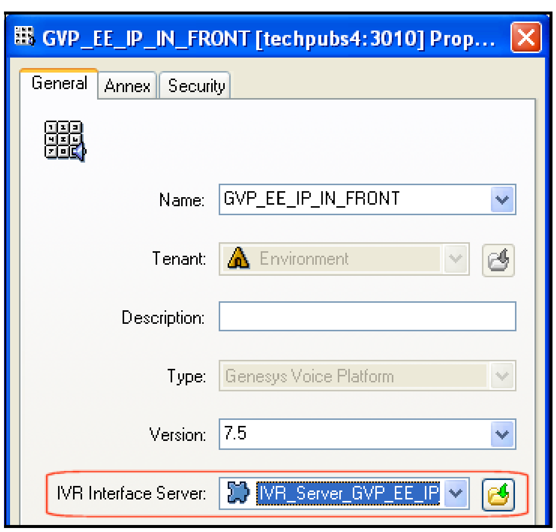

Figure 22: IVR Application Object—Selecting the IVR Server

2. In the `Options` tab of the IVR Server `Application` object, configure the link to the `TServer_IVR` application—create a section with the same name as your `TServer_IVR` (`TServer-IVR-SIP-Proxy` in the sample configuration), and then add the following options (see Figure 23):
   - `hostname`—Enter the Fully Qualified Domain Name (FQDN) for the SIP Server.
   - `port`—Enter the SIP Server port number.

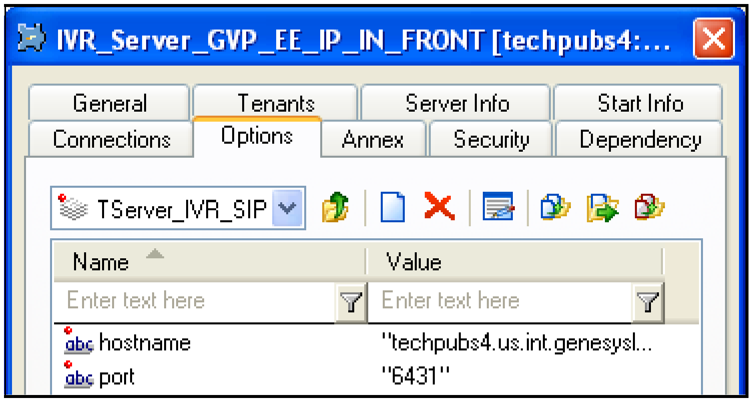

Figure 23: Linking the IVR Object to the IVR T-Server

End of procedure

Next Steps

- Configuring the IVR Server Client for the In-Front mode

Procedure:
Configuring the IVR Server Client for the In-Front mode

Purpose: To set the IVR Server Client to the `In Front` mode and configure the mode to include the DNIS in the called number field of new call messages.

Start of procedure

1. In a web browser, enter the following URL:

   `http://<EMPS-hostname>:9810/spm`

2. In the navigation tree, expand the `Reseller` object, and then expand the reseller to which the customer belongs.
3. Right-click the customer that you want to provision, and select `Provision`.

   The customer provisioning property pages open, starting with Policy.

4. On the `GenesysCTI` tab, select the `IVR Svr Client Active` check box. This assigns an active status to the IVR Server Client.
5. From the `Primary IVR Svr Client Machine` drop-down list, select the host machine for your primary IVR Server Client (see Figure 24).

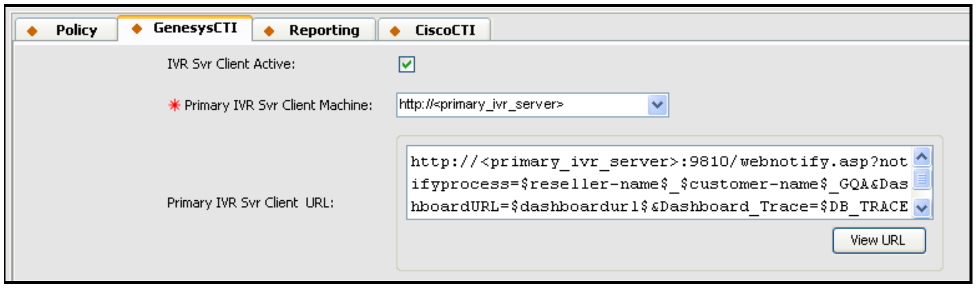

Figure 24: Activating the Primary IVR Server Client

6. From the `IVR Server Mode` drop-down list, select `In Front`, and from the `Called Number` drop-down list, select `UseDNIS` (see Figure 25).

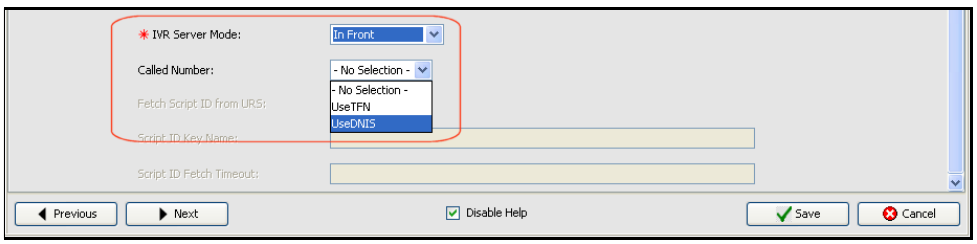

Figure 25: Selecting the IVR Server Mode and Called Number

End of procedure

Next Steps

- Pointing the GVP IPCS to the SIP Server

Procedure:
Pointing the GVP IPCS to the SIP Server

Purpose: To configure the GVP IPCS so that it uses the SIP Server as its proxy for relaying SIP messages. To do this, you add the IP address for the SIP Server in the field for SIP Session Manager (the GVP component that SIP Server essentially replaces).

Start of procedure

1. On the GVP IPCS in the EMPS navigation tree, expand the nodes `Server > IPCS > <server name> > Popgateway`, and then click `Edit`.
2. In the `SessionMgrIpAddr1` text box, enter the IP address for the SIP Server.

3. If SIP Server is not on the default port (`5060`), set the `SessionMgrSipPort1` text box to the currently configured SIP Server port.

End of procedure

Next Steps

- If your configuration includes the GVP Resource Manager, continue at "Configuring the GVP Resource Manager".
- If your configuration does not include the Resource Manager, then continue at "Configuring access from the SIP Server to GVP" on page 57.

Procedure: Configuring the GVP Resource Manager

Purpose: To configure an existing GVP Resource Manager for integration with the Genesys SIP Server. This requires you to point the IPCS to the Resource Manager IP address and port, and then assign the Resource Manager's primary user agent to the SIP Server

Summary

1. In the `ResourceMgr` field for the IPCS Pop Gateway, enter the IP address and port for the GVP Resource Manager (Step 1).
2. In the `PrimaryUserAgent` text box for the Resource Manager, enter the IP address and port for the SIP Server (Step 2).

Start of procedure

1. Point the IPCS to the GVP Resource Manager:
   a. On the GVP IPCS in the EMPS navigation tree, expand the nodes `Server > IPCS > <server name> > Popgateway`, and then select `Edit`.
   b. In the `ResourceMgr` text box on the `SIP` tab, enter the IP address and port for the GVP Resource Manager.
2. Assign the primary user agent to the SIP Server.
   a. In the EMPS navigation tree, expand the nodes `Servers > Resource Manager > <ServerName>`, right-click `Resource Manager`, and select `Edit`.
   b. On the `SIP Config` tab, enter the SIP Server IP address and port in the `Primary UA Address and Port` text box (see Figure 26).

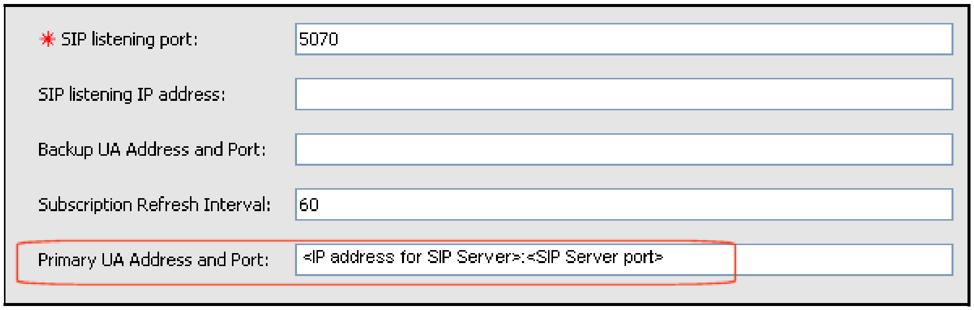

Figure 26: Pointing the Primary User Agent to the SIP Server

End of procedure

Next Steps

- Configuring access from the SIP Server to GVP

Procedure: Configuring access from the SIP Server to GVP

Purpose: To create the Access Code and the corresponding `Access Resource` that the SIP Server switch uses to reach a DN on the GVP virtual switch.

Summary

1. In the SIP Server `Switch` object, create the Access Code that allows the SIP Server to connect to the `TServer_IVR` application through ISCC (Step 1).
2. In the virtual switch for GVP, create a new DN of the type `Access Resource`, which corresponds to the Access Code you created in the SIP switch (Step 2).

Start of procedure

1. Create the Access Code that allows SIP Server to connect with the virtual switch for GVP:
   a. In the SIP Server `Switch` object, click the `Access Codes` tab.
   b. Click `Add`, browse for the GVP virtual switch (`GVP_EE_IP_VIRTUAL` in the sample configuration), and then click `OK`.

      The `Switch Access Code Properties` dialog box opens (see Figure 27).

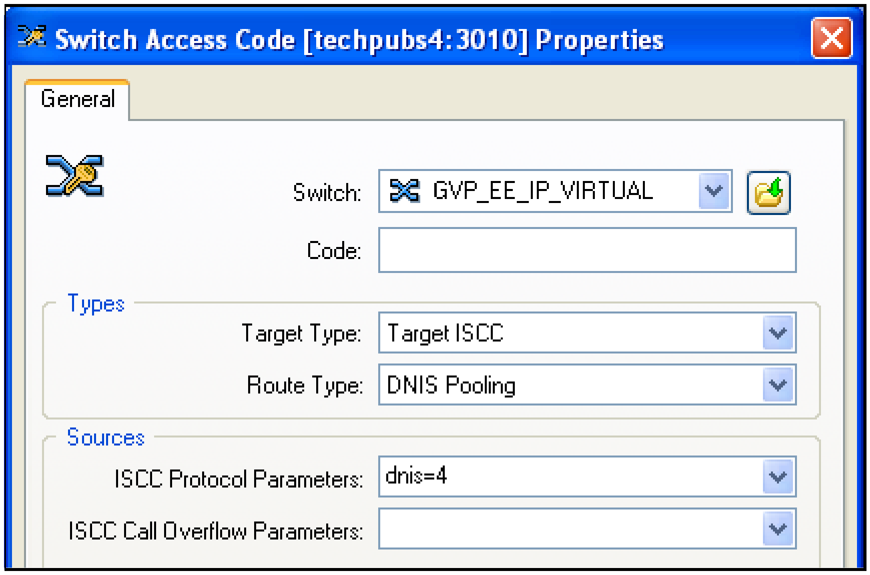

Figure 27: Configuring the Switch Access Code

c. From the `Target Type` drop-down list, select `Target ISCC`.

d. From the `Route Type` drop-down list, select `DNIS Pooling`.

e. In the `ISCC Protocol Parameters` text box, enter the following value:

`dnis-tall=4`

f. Click `OK`.

2. Create a new DN of the type `Access Resource:`

a. In the virtual `Switch` object for GVP, right-click the `DNs` folder, and then select `New > DN`.

b. Enter a number for your `Access Resource` DN—the sample configuration uses the number `4618` (see Figure 28).

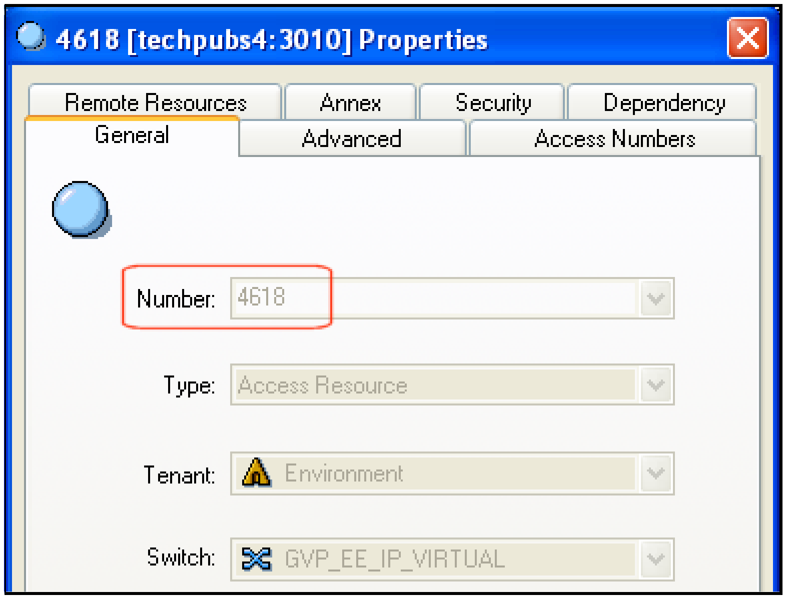

Figure 28: Configuring the Access Resource DN

c. From the `Type` drop-down list, choose `Access Resource`.

d. On the `Advanced` tab, select `dnis` from the `Resource Type` drop-down list (see Figure 29).

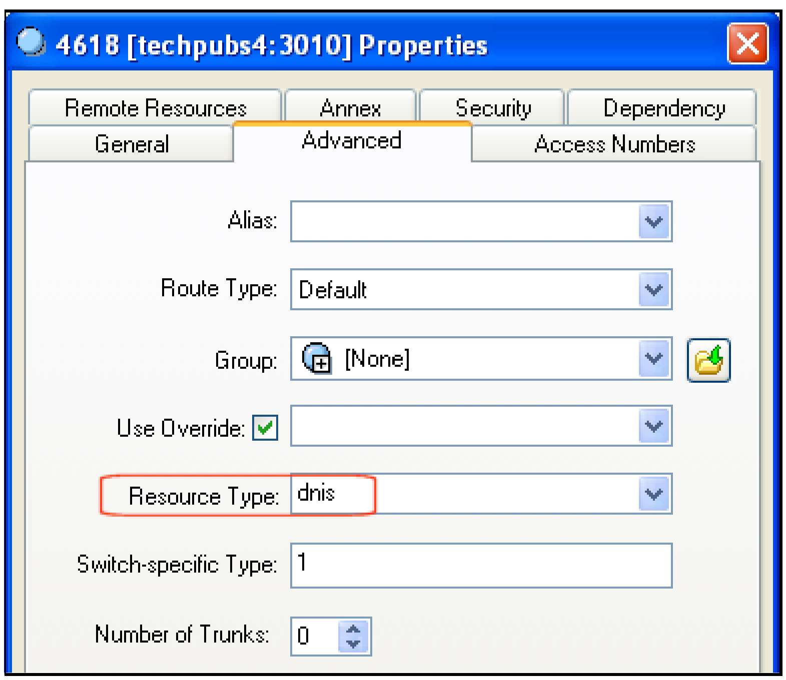

Figure 29: Configuring the Access Resource DN—Advanced Tab

End of procedure

Next Steps

- Configuring access from GVP to the SIP Server

Procedure:
Configuring access from GVP to the SIP Server

Purpose: To create the Access Code and the corresponding `External Routing Point` DN that the GVP virtual switch uses to reach a DN on the SIP Server switch.

Summary

1. In the GVP virtual switch, create the Access Code that allows GVP to connect to the SIP Server switch (Step 1).
2. In the SIP Server switch, create a corresponding `External Routing Point` DN (Step 2).

Start of procedure

1. Create the Access Code that the GVP virtual switch uses to reach a DN on the SIP Server switch:
   a. In the GVP virtual `Switch` object, click the `Access Codes` tab.
   b. Click `Add`, browse for your SIP Server `Switch` object, and then click `OK`.
      The `Switch Access Code Properties` dialog box opens (Figure 30).

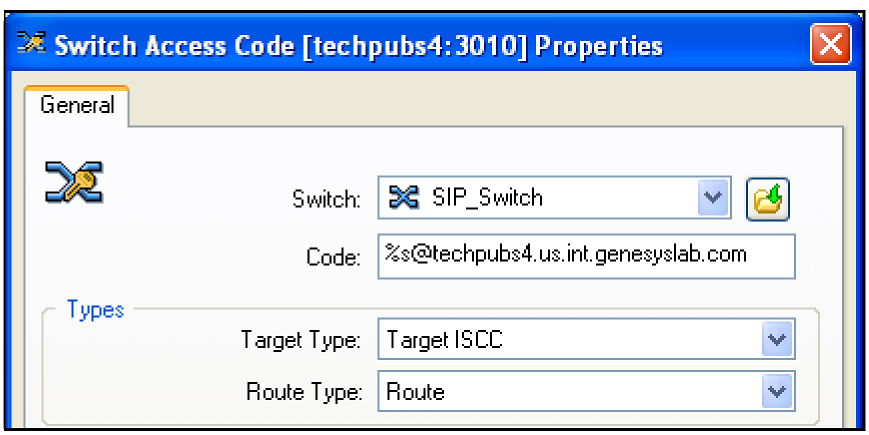

Figure 30: Configuring the Switch Access Code

   c. From the `Switch` drop-down list, select the SIP Server `switch` object.
   d. In the `Code` text box, enter the Access Code that this switch uses to reach the SIP Server switch.
   e. From the `Target Type` drop-down list, select `Target ISCC`.

f. From the `Route Type` drop-down list, select `Route`.

g. Click `OK`.

2. Create a new DN of the `External Routing Point` type:

   a. In the SIP Server `Switch` object, right-click the `DNs` folder, and then select `New > DN`.

   b. Enter a number for your External Routing Point—the sample configuration uses the number `4612` (see Figure 31).

Figure 31: Configuring the External Routing Point DN

   c. From the `Type` drop-down list, select `External Routing Point`.

End of procedure

Next Steps

- Creating a Routing Point for incoming calls

Procedure:
Creating a Routing Point for incoming calls

Purpose: To create a Routing Point in the SIP Server switch that receives inbound calls from the Media Gateway.

Start of procedure

1. In the SIP Server `Switch` object, right-click the `DNs` folder and select `New > DN`.
2. Select `Routing Point` as the `type` of DN.
3. Assign the Routing Point a number equal to the number that the external caller will dial—the DNIS.

   Tip: In the sample configuration, the external caller dials the number `4617`; therefore, create the Routing Point `4617`, so that the Media Gateway will forward inbound calls to this Routing Point in the SIP Server.

4. Click `OK`.

End of procedure

Next Steps

- Creating a routing strategy for incoming calls

Procedure: Creating a routing strategy for incoming calls

Purpose: To create a routing strategy and load it on the Routing Point for incoming calls.

Prerequisites

- A Place Group containing all the Voice Treatment Ports on the virtual switch for GVP. See Creating a Place Group object and adding Places, page 51.

Start of procedure

1. Start Interaction Routing Designer (IRD) and enter your login information.

   Tip: For more information about logging in to IRD, see the *Universal Routing 7.6 Deployment Guide*.

2. In the `Routing Design` window, create the routing strategy.

   The sample configuration uses a simple strategy consisting of the following routing objects: `Entry`, `Selection`, and `Exit` (see Figure 32).

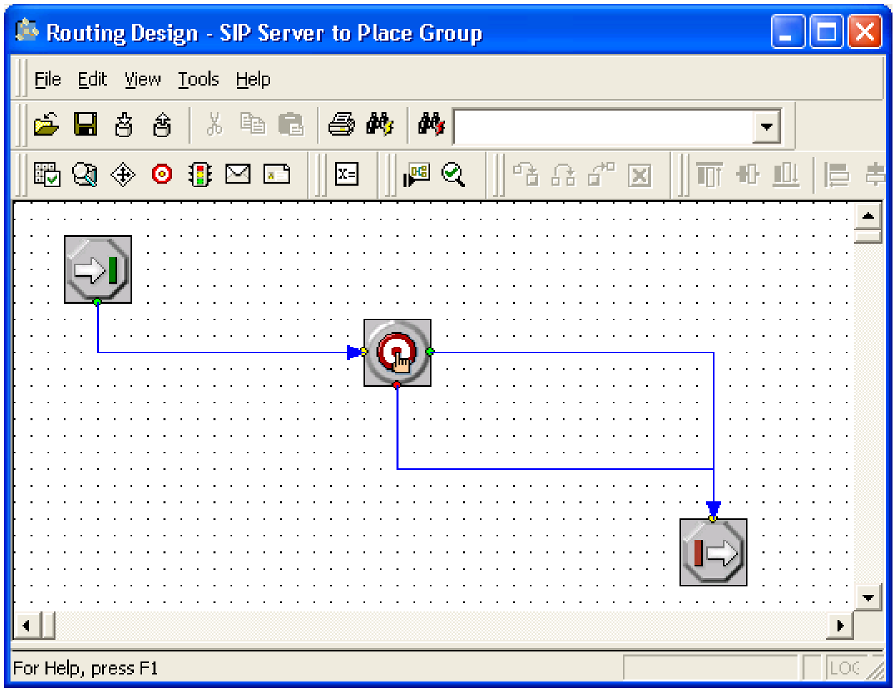

Figure 32: Simple Routing Strategy for the Sample Configuration

3. Configure the `Selection` object in this strategy so that it uses as its target the Place Group that you created in "Creating a Place Group object and adding Places" on page 51.
   a. Double-click the `Selection` object.
   b. On the `Target Selection` tab, click the `Add item` icon.
   c. Configure the following fields for the required Place Group:
      - `Type`—Select `Place Group`.
      - `Name`—Select the Place Group that you created on the virtual switch.
      - `StatServer`—Select the Stat Server for your SIP environment.

   Figure 33 shows the completed `Selection properties` dialog box.

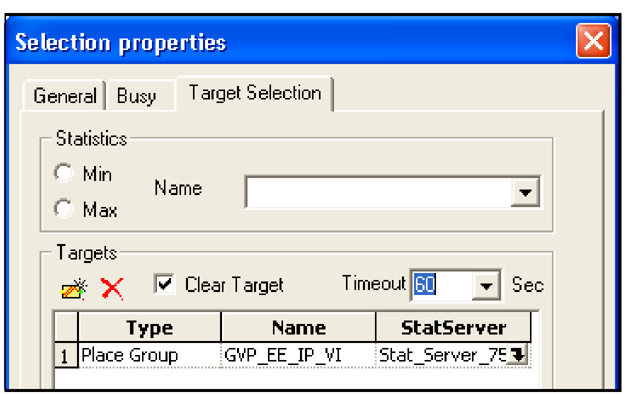

Figure 33: Creating the Place Group as the Target

4. In the Monitoring window of IRD, load your new strategy on the Routing Point that you created in "Creating a Routing Point for incoming calls" on page 61.
   a. In the Shortcut bar, click the Loading icon.
   b. In the Loading window, expand your SIP Server switch.
   c. Right-click the Routing Point and select Load strategy.
   d. Select the strategy that you created in Steps 2 and 3, and then click OK.

End of procedure

Next Steps

- Creating a Trunk Group DN

Procedure: Creating a Trunk Group DN

Purpose: To create and configure a DN of Trunk Group type in the SIP Server switch. This Trunk Group DN associates the SIP Server with the GVP instance.

Start of procedure

1. In Configuration Manager, right-click the DNs folder under your SIP Server Switch object, and select New > DN.
2. Select Trunk Group as the type.
3. Assign the new DN a number. This number should combine two values using the following format:

   ```
   <Access Code from SIP Server to GVP> + <Access Resource DN
   configured in the GVP virtual switch>
   ```

   Tip: For details about access codes, see Step 1 on page 57. For details about the access resources, see Step 2 on page 58.

   In the sample configuration, the access code is left empty, and the Access Resource DN is 4618. So in this case, the Trunk Group DN uses the number 4618 (see Figure 35).

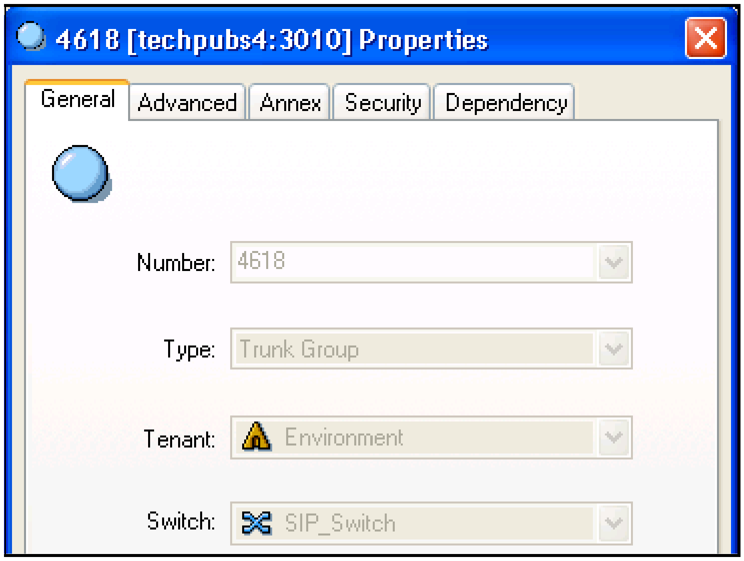

Figure 34: Trunk Group DN Properties

4. On the `Annex` tab, create a new section called `TServer`.
5. In the `TServer` section, create a new option called `contact` and enter the GVP IP address (if Resource Manager is included in your configuration, use its IP address here; otherwise, use the IP address for the IPCS).

   SIP Server uses the IP address in this DN to send calls to GVP after routing.

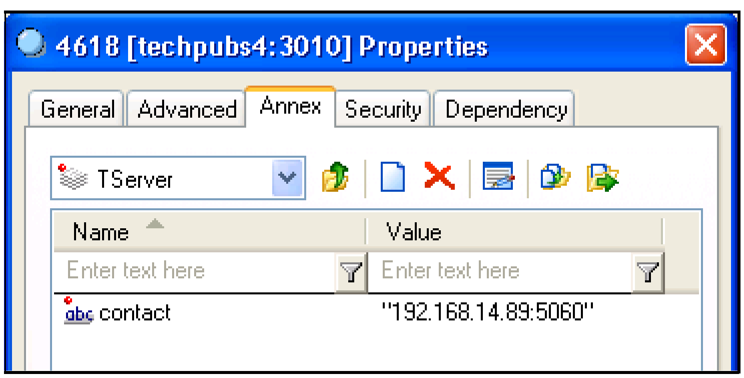

Figure 35: GVP IP Address

6. Restart the SIP Server.

End of procedure

Next Steps

- Creating a Routing Point for selecting an agent

Procedure:
Creating a Routing Point for selecting an agent

Purpose: To create a Routing Point in the virtual switch that GVP uses to transfer calls to an agent.

Start of procedure

1. In the GVP virtual `Switch` object, right-click the `DNs` folder, and select `New > DN`.
2. From the `Type` drop-down list, select `Routing Point`.
3. Enter a number for the `Routing Point` DN. The sample configuration uses the number `1051` (see Figure 36).

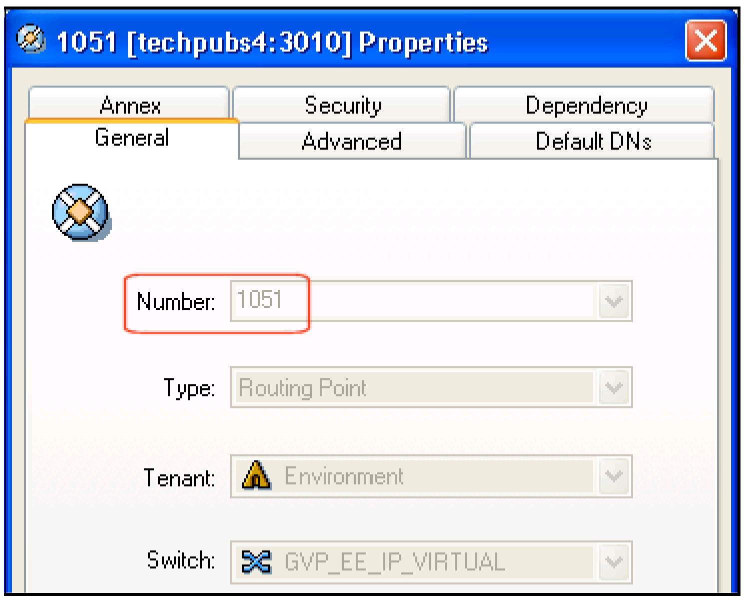

Figure 36: Routing Point DN for Selecting an Agent

4. Click `OK`.

End of procedure

Next Steps

- Creating the voice self-service application

Procedure: Creating the voice self-service application

Purpose: To create a sample voice self-service application (using Genesys Studio URS blocks) for sending calls to an agent.

Start of procedure

1. Using Genesys Studio URS blocks, create a sample voice self-service application.

   Tip: For information about using URS blocks to build an application, see *Genesys Studio Help*.

   Figure 37 shows the simple voice self-service application used in the sample configuration.

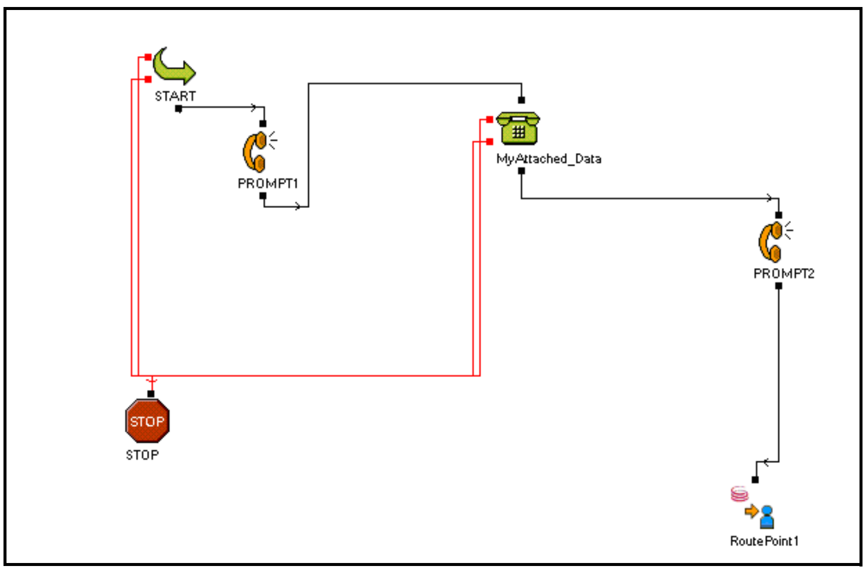

Figure 37: Sample Voice Self-Service Application

2. For `RoutePoint1` in this application, in the `Route DN` text box, enter the same number that you used for the Routing Point in the virtual switch (see "Creating a Routing Point for selecting an agent" on page 66).

   The sample configuration uses the number `1051` (see Figure 38).

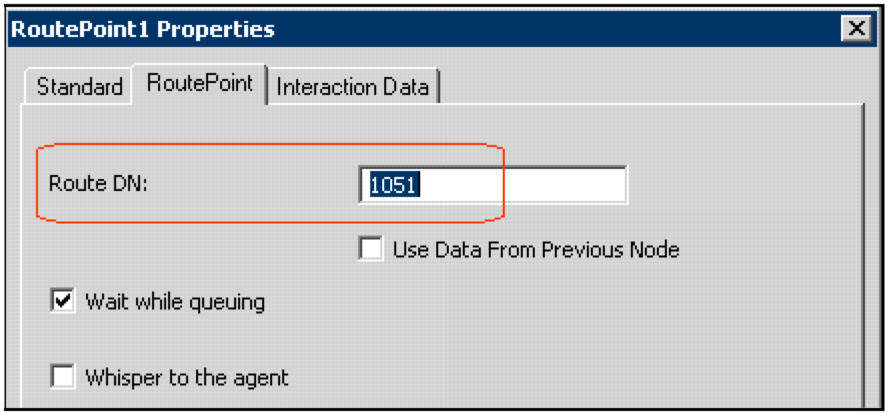

Figure 38: Routing Point Properties

3. In the GVP application settings of the EMPS interface, assign the same DN number for the `Phone Number` as you did for `RoutePoint1` in the URS block.

   The sample configuration uses the number `1051`.

4. Set the `Transfer Option` to `SIPRefer`.

End of procedure

Next Steps

- Creating the routing strategy on the virtual switch

Procedure: Creating the routing strategy on the virtual switch

Purpose: To create a routing strategy and load it on the Routing Point that GVP uses to select an agent.

Start of procedure

1. Start IRD, and enter your login information.

   Tip: For more information about logging in to IRD, see the *Universal Routing 7.5 Deployment Guide*.

2. In the `Routing Design` window of IRD, create the routing strategy.

   The sample configuration uses a simple strategy consisting of the following routing objects: `Entry`, `Selection`, and `Exit` (see Figure 39).

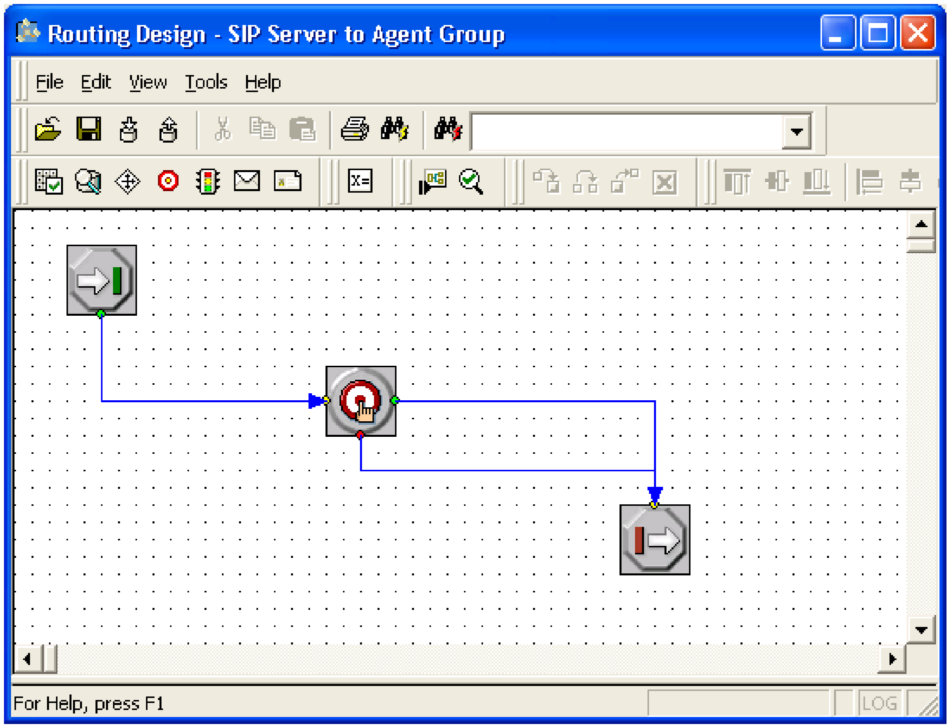

Figure 39: Simple Routing Strategy for the Sample Configuration

3. Configure the `Selection` object in this strategy so that it points to a target on the SIP Server switch. The sample configuration selects an `Agent Group` object as its target. The Agent Group includes agents who are logged in to the SIP Server.
   a. Double-click the `Selection` object.
   b. On the `Target Selection` tab, click the `Add item` icon.
   c. Configure the following fields, for the required Agent Group:
      - `Type`—Select `Agent Group`.
      - `Name`—Select an Agent Group from the SIP Server switch.
      - `StatServer`—Select the Stat Server for your SIP environment.

   Figure 40 shows the completed `Selection properties` dialog box.

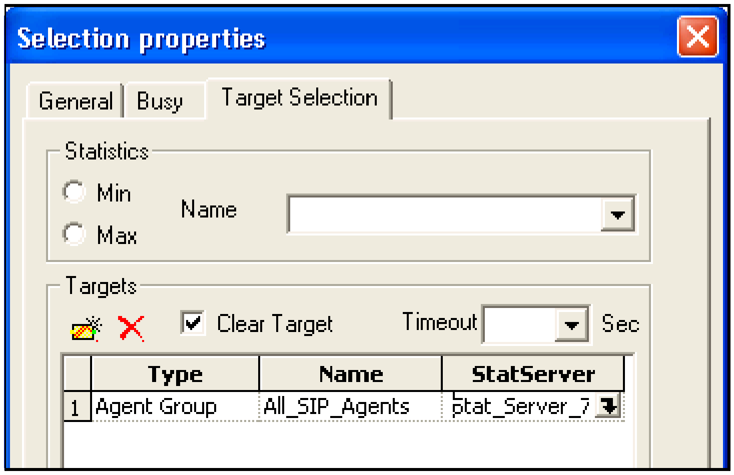

Figure 40: Creating the Agent Group as the Target

4. In the `Monitoring` window of IRD, load your new strategy on the Routing Point that you created in "Creating a Routing Point for selecting an agent" on page 66.
   a. In the `Shortcut` bar, click the `Loading` icon.
   b. In the `Loading` window, expand the GVP virtual switch.
   c. Right-click the Routing Point, and select `Load strategy`.
   d. Select the strategy that you created in Steps 2 and 3, and then click `OK`.

End of procedure

Next Steps

- If you are not using the Alcatel 5020 Softswitch in your integration, you have now completed all required steps for the GVP–SIP In-Front mode integration.
- If your environment includes the Alcatel 5020 Softswitch, some additional configuration is required. See (Optional) Creating a trunk on the SIP Server for the Alcatel 5020 Softswitch.
- If you want to remove the `To` header in `REFER` requests, see Removing the "To" header from REFER requests, page 72.
- If you want to enable call matching for multiple transfers to and from GVP, see Enabling call matching for multiple transfers in GVP, page 73

Procedure: (Optional) Creating a trunk on the SIP Server for the Alcatel 5020 Softswitch

Purpose: To create and configure a trunk on the SIP Server switch for use with the Alcatel 5020 Softswitch.

Start of procedure

1. In the SIP Server `Switch` object, create a `Trunk` type DN (see Figure 41).

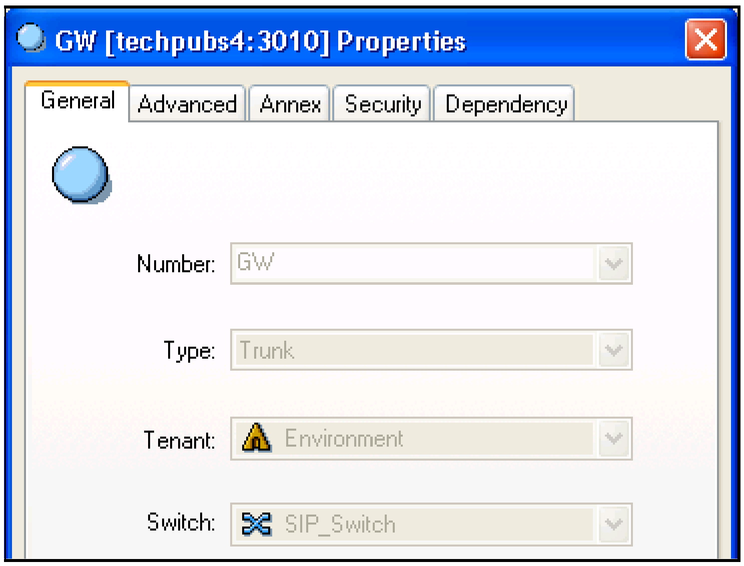

Figure 41: Trunk Type DN for the Alcatel 5020 Softswitch

2. On the `Annex` tab, create a new section called `TServer`.
3. In the `TServer` section, create a new option called `Contact`, and enter the IP address for the Alcatel 5020 Softswitch.
4. If you want the SIP Server to send `REFER` requests from GVP to the Alcatel 5020 Softswitch, create a new option called `refer-enabled` in the `TServer` section, and set the value to `true`.

   OR

   If you want the SIP Server to process the `REFER` requests without involving the Alcatel 5020, set the `refer-enabled` option to `false` (see Figure 42).

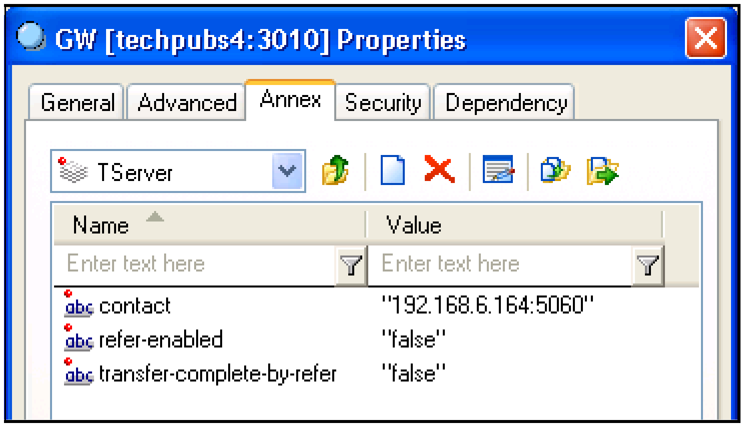

Figure 42: Annex Tab for the Alcatel 5020 Softswitch Trunk

End of procedure

Procedure: Removing the “To” header from REFER requests

Purpose: To replace the `To` header in SIP `REFER` requests with the `Trunk Group` DN.

Start of procedure

- In the `TServer` section of the SIP Server application `Options` tab, set the `override-to-on-divert` option to `true`.

  When this option is set to `true`, SIP Server overrides the value of the `To` header in the `INVITE` request that it sends to GVP. In its place, SIP Server uses the `Trunk Group` DN number that you specified in “Creating a Trunk Group DN” on page 64.

  > **Tip:** The sample configuration uses `4618` for the `Trunk Group` DN. Because GVP must treat this value as the DNIS, you must configure GVP to enable a corresponding voice application for this DNIS.

End of procedure

Procedure:
Enabling call matching for multiple transfers in GVP

Start of procedure

- In the `extrouter` section of the `SIP Server` application, set the `match-call-once` option to `false`.

  Setting this option to `false` enables proper call matching with multiple transfers to and from GVP.

End of procedure

Chapter

4 GVP–SIP Server Integration—Stand-Alone Mode

This chapter describes how to integrate the Genesys Voice Platform (GVP) with the SIP Server, in the Stand-Alone mode. It includes the following sections:

Task Flow—Stand-Alone Mode

Table 7 provides an overview of the main steps required to integrate SIP Server 7.5 with GVP in the Stand-Alone mode. To help coordinate values across multiple components, fill in the Integration Worksheet (see “Integration Worksheet—Stand-Alone Mode” on page 86) before you begin.

Table 7: Taskflow to Integrate the SIP Server with GVP—Stand-Alone Mode

Objective	Related Procedures and Actions
1. Configure the SIP Server for Stand-Alone integration.	The integration requires that you configure the SIP Server options as follows: • Set `override-to-on-divert` to `false`. • Set `sip-refer-to-sst-enabled` to `true`. For the detailed procedure, see Configuring the SIP Server for Stand-Alone mode, page 77.
2. Disable `REFER` requests on the gateway (recommended).	On the `Trunk` DN for the gateway, Genesys recommends that you set the `refer-enabled` option to `false`. • Disabling REFER requests on the gateway (recommended), page 77
3. Configure the Resource Manager.	Create a `Trunk` DN that points to the GVP Resource Manager component. • Add a `contact` option specifying the SIP address and port of the Resource Manager. • Add a `prefix` option, specifying the DID prefix of the dialed number. For the detailed procedure, see Configuring the GVP Resource Manager, page 78.
4. Configure IPCS instances.	Create a `Trunk` DN for each IPCS instance in your configuration. • For the `Trunk` DN name, use the IP address of the IPCS, with the dot (.) separators removed. • Add a `contact` option specifying the SIP address and port of the IPCS instance. • Set the `straight-forward` option to `true`. • Set the `subscription-id` option to `GVP`. For the detailed procedure, see Configuring IPCS instances, page 79.

Procedures—Stand-Alone Mode

This section provides detailed procedures for configuring the various components and applications that are required for the GVP–SIP Server integration in the Stand-Alone mode.

Procedure:
Configuring the SIP Server for Stand-Alone mode

Purpose: To perform additional configuration of the SIP Server application in preparation for the GVP–SIP Server integration.

Start of procedure

1. In Configuration Manager, open the SIP Server `Application` object.
2. On the `Options` tab, in the `TServer` section, configure the options as described in Table 8.

Table 8: SIP Server Options—TServer Section

Option	Value	Description
override-to-on-divert	false	Set this option to `false` to ensure that the `INVITE` request sent to GVP contains the same user name in the `To` header as it did in the original `INVITE` request. **Note**: The originally dialed number (Routing Point) must exist on GVP as a DID number.
sip-refer-to-sst-enabled	true	If you set this option to `true`, the SIP Server processes the `REFER` request from the IPCS as if a `TSingleStepTransfer` was requested instead. In cases where the gateway is configured with `refer-enabled=false`, enabling this option allows the conversion of the `REFER` request into the re-`INVITE` request.

End of procedure

Next Steps

- Disabling REFER requests on the gateway (recommended)

Procedure:
Disabling REFER requests on the gateway (recommended)

Purpose: To ensures that only the recommended re-`INVITE` requests are used to transfer calls from the IPCS.

Start of procedure

1. In the `Trunk` DN for the gateway, click the `Annex` tab.
2. If the `TServer` section does not already exist, create one.
3. Create a new option called `refer-enabled`, and assign it a value of `false` (see Table 43).

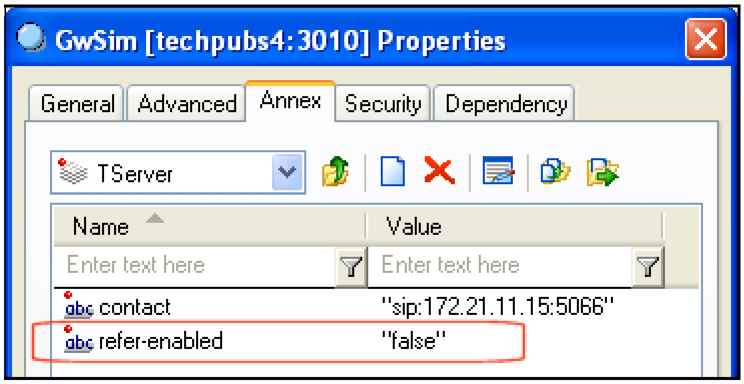

Figure 43: Gateway Configuration

If your gateway supports `REFER` requests, setting this option to `false` prevents the use of `REFER` requests, ensuring that only the recommended re-`INVITE` request method is used to transfer calls.

End of procedure

Next Steps

- Configuring the GVP Resource Manager

Procedure: Configuring the GVP Resource Manager

Purpose: To create and configure the `Trunk` DN that integrates the Resource Manager into the GVP–SIP Stand-Alone configuration.

Start of procedure

1. Under the SIP Server `Switch` in the navigation tree, right-click the `DNs` folder and select `New > DN`.
2. Select `Trunk` as the type.
3. Enter a name for the trunk.

Tip: You can use an arbitrary name for this DN. No required conventions apply. The sample configuration uses the name `Resource Manager`.

4. On the `Annex` tab, in the `TServer` section, configure the options as shown in Figure 44. For a description of these options, see Table 9.

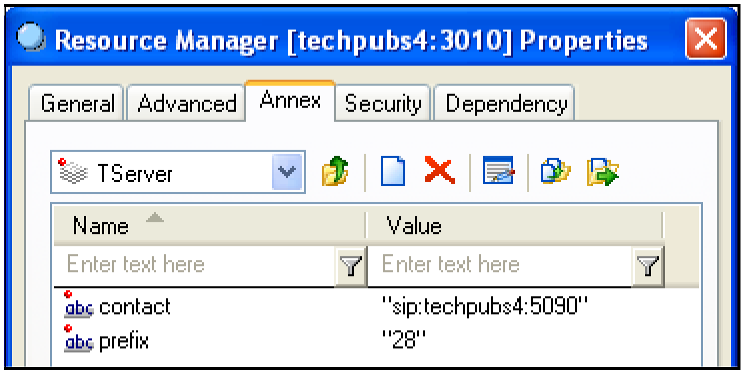

Figure 44: Trunk DN for the Resource Manager

Table 9: Resource Manager Trunk Options—TServer Section

Option	Description
contact	Enter the SIP address and port for the GVP Resource Manager component. The sample configuration uses the value: `sip:techpubs4:5090`
prefix	Enter the DID prefix for incoming calls. The SIP Server uses the dialed number to contact the Resource Manager. For contact to be established, this prefix must match that dialed number. The sample configuration uses the prefix `28`. **Note**: If multiple DID numbers (with no common prefixes) are used, you can create several `Trunk` type DNs to point to the same address and port for the one Resource Manager.

End of procedure

Next Steps

- Configuring IPCS instances

Procedure: Configuring IPCS instances

Purpose: To create and configure a `Trunk` DN for each IPCS instance in the GVP–SIP Server deployment.

Start of procedure

1. Under the SIP Server `Switch` object, right-click the folder containing your IPCS DNs, and select `New > DN`.
2. Select `Trunk` as the type.
3. For the DN number, enter the IP address of the IPCS instance, with the dot (.) separators removed. For example, if the IPCS is located at IP address 172.21.11.15, give the corresponding `Trunk` DN the number `172211115` (see Figure 45).

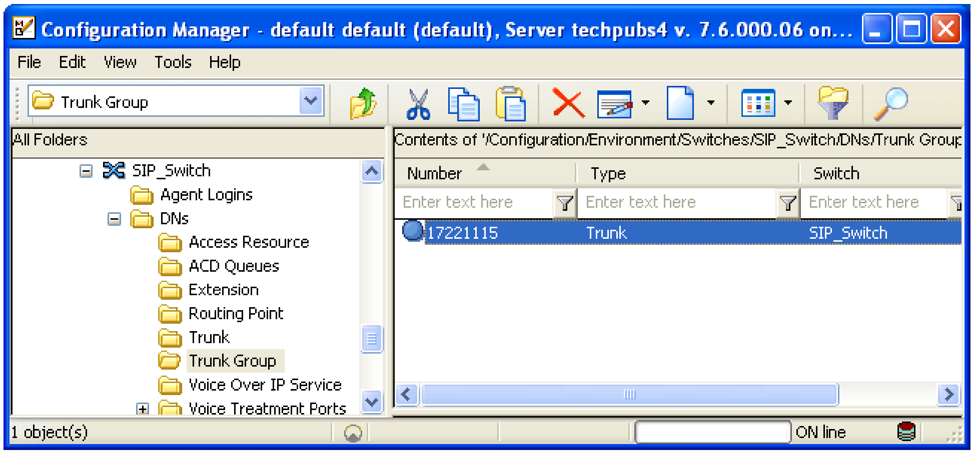

Figure 45: IPCS Configured as a Trunk DN

4. Repeat Steps 1 to 3 for each IPCS instance in your configuration.

   For each new `Trunk` DN that you create, create a section called `TServer`, and add new options as shown in Figure 46. Table 10 provides detailed descriptions of these options.

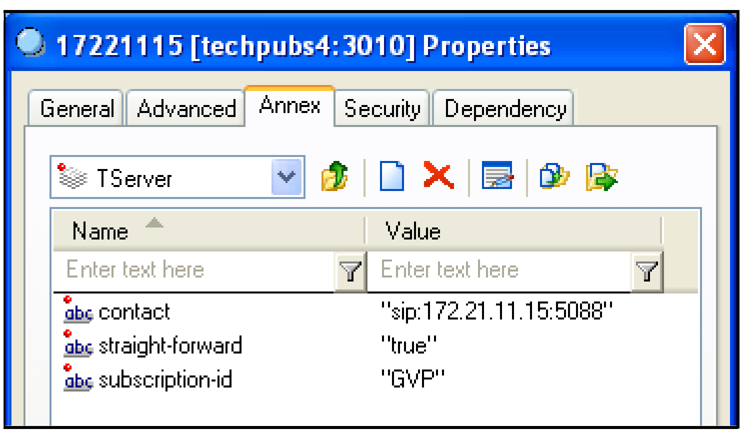

Figure 46: Trunk DN—Annex Tab

Table 10: Trunk DN for IPCS Options—TServer Section

Option	Value	Description
contact	...	Enter the SIP address and port of the corresponding IPCS instance. See the Tip that follows.
straight-forward	true	Set this option to `true` to force the SIP Server to use the SIP URI provided in the `Contact` header of the `302 Moved` response. See the Tip that follows.
subscription-id	GVP	Enter `GVP` as the identifier that the Resource Manager uses when subscribing to the SIP Server.

Tip: If you set the `contact` value to the correct IPCS address and port, you do not need to enable the `straight-forward` option. Similarly, if you set the `straight-forward` option, you do not need to set any particular value for the `contact` option (you can use a dummy value instead).

End of procedure

Next Steps

- You have completed all the required steps to integrate the SIP Server 7.5 with GVP.

Appendix

A Integration Worksheets

This appendix provides worksheets that you can use to help coordinate your integration. Available worksheets include:

Integration Worksheet—In-Front Mode

Table 11 lists the values for the sample In Front configuration used in the procedure sections of this guide. Fill in the appropriate values for your configuration, as an aid to coordinating numbering and naming conventions across the many components involved in the integration.

Table 11: Integration Worksheet—In-Front Mode

Field	Your Values	Sample Values	Description
Virtual Switch for GVP			
Number of available ports on GVP		3	Check the `Max Channels` value in GVP EMPS to determine the number of available ports.
Number of Voice Treatment Port (VTP) DNs		3	The number of VTP DNs must equal the number of ports available on GVP.
VTP DNs		01 to 03	Number range for the Voice Treatment Port DNs.

Table 11: Integration Worksheet—In-Front Mode (Continued)

Field	Your Values	Sample Values	Description
`Access Resource` DN		4618	Create an `Access Resource` DN in the virtual switch that corresponds to the access code configured in the SIP Server switch.
`External Routing Point` DN		4612	Create the `External Routing Point` DN corresponding to the Access Code created in the GVP virtual switch.
Framework Resources			
`Place` objects		01 to 03	Use the same numbering scheme for your `Place` objects as you did for the VTP DNs.
`Place Group` object		GVP_EE_IP_VIRTUAL	Create a Place Group that routing uses as the target when sending calls from SIP Sever to GVP.
`Routing Point` DN		1501	GVP transfers calls to an agent using this Routing Point.
IVR Object			
IVR ports		01 to 03	Use the same numbering scheme for your IVR ports as you did for the VTP DNs.
SIP Server Switch Object			
Routing Point		4617	The Media Gateway forwards inbound calls to this Routing Point in the SIP Server switch.
Trunk Group DN		4618	SIP Server uses this `Trunk Group` DN to direct inbound calls to GVP.
Trunk DN (optional for the Alcatel 5020 Softswitch)		GW	The name of the trunk used to link SIP Server with the Alcatel 5020 Softswitch.

Integration Worksheet—Behind Mode

Table 12 lists the values for the sample Behind mode configuration used in the procedure sections of this guide. Fill in appropriate values for your configuration, as an aid to coordinating numbering and naming conventions across the many components involved in the integration.

Table 12: Integration Worksheet—Behind Mode

Field	Your Values	Sample Values	Description
IVR Server			
Virtual Routing Point DN		2700	For call queuing on GVP, create a Virtual Routing Point in the SIP Server switch, and then point the IVR Server to this Virtual Routing Point in its options.
SIP Server Switch			
Voice Treatment Port DNs		2011 to 2014	Create as many `Voice Treatment Port` DNs as the number of declared ports on each IPCS.
Framework Resources			
Places		2011 to 2014	Create a `Place` object for every corresponding Voice Treatment Port.
Place Group		172211115	Create a `Place Group` object for each IPCS. Use the IP address of the targeted IPCS, with dot (.) separators removed.
GVP Instance			
IVR object for GVP		GVP	Name of the `IVR` object that represents the whole GVP instance.
IVR ports		2011 to 2014 2021 to 2024*	Create an IVR port for each corresponding Voice Treatment Port DN in the SIP Server switch. *represents DNs for second IPCS, for sample purposes

Integration Worksheet—Stand-Alone Mode

Table 13 lists the values for the sample Stand-Alone mode configuration used in the procedures sections of this guide. Fill in appropriate values for your configuration, as an aid to coordinating numbering and naming conventions across the many components involved in the integration.

Table 13: Integration Worksheet—Stand-alone Mode

Field	Your Values	Sample Values	Description
SIP Server Switch			
Trunk DN		Resource Manager	Use an arbitrary name for the trunk that points the SIP Server switch to the GVP Resource Manager.
Trunk Group DN		172211115	Create a `Trunk Group` DN for every IPCS in your configuration. Use the IP address of the targeted IPCS, with dot (.) separators removed.

Appendix

B SIP Server Configuration Options

This appendix describes the configuration options included in this guide that are unique to SIP Server.

event-ringing-on-100trying

Default Value: false
Valid Values:

`true`	SIP Server generates `EventRinging`.
`false`	SIP Server does not generate `EventRinging`.

Changes Take Effect: Immediately

Specifies whether SIP Server generates an EventRinging message for a DN when it receives a 100 Trying SIP message. Normally, the EventRinging message is generated on 180 Ringing SIP message, but this option allows for GVP integration when the IVR Server is configured in Behind-the-Switch mode.

Note: This option must be set at both the Application and at the Switch/DN level because it is used for proper synchronization with the I-Server Application.

override-to-on-divert

Default Value: false
Valid Values:

`true`	The username is equal to the destination DN.
`false`	The username is equal to the Routing Point or ACD Queue number.

Changes Take Effect: Immediately

Controls the username part of the To header URI for outgoing INVITE messages when a call is diverted from a Routing Point or an ACD Queue.

sip-refer-to-sst-enabled

Default Value: true
Valid Values:

`true`	The `re-INVITE` message is used instead of the `REFER` message for single-step transfer functionality.
`false`	The `REFER` message from the endpoint is not converted to a re-`INVITE` message and also is rejected by SIP Server if the endpoint has the `refer-enabled` option set to `false`.

Changes Take Effect: Immediately

Specifies whether a re-INVITE message is used instead of a REFER message for single-step transfer functionality. Use this option when the switch does not support the REFER message when performing a single-step transfer.

refer-enabled

Default Value: true
Valid Values: true, false
Changes Take Effect: With the next new call on this DN

Specifies whether the REFER method is sent to an endpoint. The REFER method is used for:

- The originating DN during a TMakeCall request.
- The receiving DN during a consultation call or a single-step call transfer.
- The DN that is transferred to another destination during a single-step call transfer.

When set to false, SIP Server uses the re-INVITE method instead. In this case, single-step transfers are unavailable.

Note: When integrating GVP with SIP Server, and when you need to use a re-INVITE message instead of a REFER message for a single-step call transfer, set this value to false, and ensure that you have the proper value for the sip-refer-to-sst-enabled option (page 88).

straight-forward

Default Value: false
Valid Values:

`true`	SIP Server sends a new `INVITE` message directly to the location specified in the `Contact` header.
`false`	SIP Server resolves the `username` parameter for the `Contact` header using information from the DN configuration.

Changes Take Effect: Immediately

Describes how SIP Server responds to the 300 or 302 SIP messages received from an endpoint or a server.

Note: This option is required for GVP integration in Stand-Alone mode. Trunk DNs configured for use with Resource Manager and IP Communication Server must contain the value true.

subscription-id

Default Value: NULL
Valid Values: Any valid string
Changes Take Effect: Immediately

Enables the subscription for multiple DNs with one Subscribe message. The value must be same as the user part of the Subscribe URI.

Note: For integration with GVP, this value must be set to GVP.

handle-vsp

Default Value: no
Valid Values:

`requests`	The ISCC component of SIP Server will attempt to translate requests related to this DN before submitting them to the service provider.
`events`	The ISCC component of SIP Server will attempt to process events received from the service provider before distributing them to SIP Server clients.
`all`	The ISCC component of SIP Server will handle both the events and requests.
`no`	No processing will take place.

Changes Take Effect: Immediately

Specifies the way SIP Server will handle events from, and requests to, an external service provider registered for a DN using the AddressType attribute set to VSP.

Configure this option in the extrouter section on the Options tab for the SIP Server Application object in Configuration Manager.

Index

H

I

M

O

P

R

S

T

V