Voice Platform Solution 8.0

Integration Guide

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Document Version: 80gvp_ig-vps_11-2008_v8.0.001.06

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List of Procedures

Preface

Welcome to the *Voice Platform Solution 8.0 Integration Guide*. This document provides an overview of the Voice Platform Solution (VPS), with an aim to integrating the various components that make up the solution—in other words, to get the components working together.

What this guide does not cover:

- Deployment procedures—This guide provides step-by-step instructions of the changes that need to be made to make the solution work, but not how to install or initially configure the individual components. For deployment information, consult the respective product Deployment Guides.
- Network integration—Although the call flow scenarios in this guide may include third-party components external to the solution, such as gateways or a PBX (Private Branch Exchange), this guide does not explain in detail the integration of those components with the solution. SIP Server is mostly responsible for these network connections, and so for more information you should consult the *Framework 7.6 SIP Server Deployment Guide*.

This document applies to the 7.6 release of SIP Server, and the 8.0 release of the Genesys Management Framework and Genesys Voice Platform.

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

- Computer-telephony integration (CTI) concepts, processes, terminology, and applications.
- The Session Initiation Protocol (SIP) generally, as well as the integration of SIP messaging into the Genesys environment: SIP Server and related components.
- Network design and operation.
- Your own network configurations.

This guide also assumes that you:

- Are familiar with the Genesys Management Framework architecture and functions that support SIP Server 7.6 and Genesys Voice Platform 8.0.
- Have already installed and are familiar with SIP Server and 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 15, provides a general description of the solution and its various components, as well as details about support call flow scenarios. You can also find information about using the Genesys Administrator and an overview of how high availability works within the solution.
- Chapter 2, "VPS 8.0 Integration Procedures," on page 37, provides a task flow of the main steps required to integrate GVP and SIP Server, along with key actions and, if you need them, links to more detailed procedures found later in the chapter.
- Appendix A, "Sample User Data Mapping," on page 69, provides sample user data exchanges between SIP Server and the T-Library event, in both directions
- Appendix B, "Configuration Options," on page 75, describes all configuration options modified during any of the procedures included in this guide.

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:

80gvp_ig-vps_06-2008_v8.0.001.00

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 below.

SIP Server

Consult the following additional resource as necessary:

- *Framework 7.6 SIP Server Deployment Guide*, which provides information to configure and install SIP Server.

Genesys Voice Platform

Consult these additional resources as necessary:

- *Genesys Voice Platform 8.0 Deployment Guide*, which provides information to install and configure GVP.
- *Genesys Voice Platform 8.0 VoiceXML 2.1 Help*, which provides information about developing VoiceXML applications. It presents VoiceXML concepts and provides examples that focus on the GVP implementation of VoiceXML.
- *Genesys Voice Platform 8.0 CCXML Reference Manual*, which provides information about developing CCXML applications for GVP.
- *Genesys Voice Platform 8.0 Troubleshooting Guide*, which provides information about SNMP MIBs and traps for GVP, as well as troubleshooting methodology.
- *Composer Voice 8.0 Deployment Guide*, which provides installation and configuration instructions for Composer Voice.
- *Composer Voice 8.0 Help*, which provides online information about using Composer Voice, a GUI for the development of applications based on VoiceXML and CCXML.
- *Genesys Voice Platform 8.0 Configuration Options Reference*, which replicates the metadata available in the Genesys provisioning GUI to provide information about all the GVP configuration options, including descriptions, syntax, valid values, and default values.
- *Genesys Voice Platform 8.0 Metrics Reference*, which provides information about all the GVP metrics (VoiceXML and CCXML application event logs), including descriptions, format, logging level, source component, and metric ID.
- *W3C Voice Extensible Markup Language (VoiceXML) 2.1, W3C Recommendation 19 June 2007*, which is the W3C VoiceXML specification that GVP supports.

- *W3C Speech Synthesis Markup Language (SSML) Version 1.0, Recommendation 7 September 2004*, which is the W3C SSML specification that GVP supports.
- *W3C Voice Browser Call Control: CCXML Version 1.0, W3C Working Draft 29 June*, which is the W3C CCXML specification that GVP supports.

Universal Routing

Consult these additional resources as necessary:

- *Universal Routing 7.6 Reference Manual,* which contains descriptions of all routing strategy objects.
- *Universal Routing 7.6 Strategy Samples,* which describes the sample strategies supplied with Universal Routing.
- *Universal Routing 7.6 Business Process User's Guide,* which contains step-by-step instructions for using Interaction Routing Designer to design interaction workflows. It also describes the sample business processes.
- *Universal Routing 7.6 Interaction Routing Designer Help,* which is a guide to Interaction Routing Designer, including the portion of it that designs interaction workflows and business processes.

Management Framework

Consult these additional resources as necessary:

- *Framework 8.0 Deployment Guide*, which provides information to configure, install, start, and stop Framework components.
- *Framework 8.0 Configuration Options Reference Manual*, which provides descriptions of configuration options for Framework components.
- *Framework 8.0 Genesys Administrator Help*, which provides instructions for configuring and provisioning contact center objects using Genesys Administrator.

Genesys

Consult these additional resources as necessary:

- The *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.
- The *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.

- The Release Notes and Product Advisories for these products, which are available on the Genesys Technical Support website at http://genesyslab.com/support.
- The documentation on the other three members of the Genesys Customer Interaction Platform: Universal Routing, Reporting, and Management Framework.

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

- *Genesys Supported Operating Systems and Databases*
- *Genesys 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.

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Chapter

1 Overview

This chapter provides an overview of the Voice Platform Solution (VPS) 8.0 components, basic component and system architecture, supported call scenarios, as well as the steps required to integrate the various components into a functioning solution.

This chapter includes the following sections:

What Is the Voice Platform Solution?

The Voice Platform Solution (VPS) 8.0 combines voice self-service, agent-assisted service, and application management functions into a single, IP-based contact center solution.

Using Voice over Internet Protocol (VoIP) technology, the VPS can process incoming IP calls and decide with a high degree of flexibility where and when in the call flow to launch voice self-service applications, and when to transfer calls to an available agent for customer assistance, using several available transfer methods.

The solution combines components from three main Genesys products—Genesys Voice Platform (GVP) 8.0, SIP Server 7.6, and Management Framework 8.0—into one integrated product that supports a variety of call flow scenarios. The procedures in this guide include the basic configuration steps required to get the various components working together. After the components have been integrated, application developers can design the routing strategies, voice dialog applications, and call control applications for the various call flow scenarios.

Note: Depending on the design of your VoiceXML application, media redirect transfers may require some additional configuration on the Media Control Platform. For more information, see "Configuring the solution for media redirect transfers" on page 62.

Functional Overview

Figure 1 shows the overall VPS functionality. This figure shows functions only, not components.

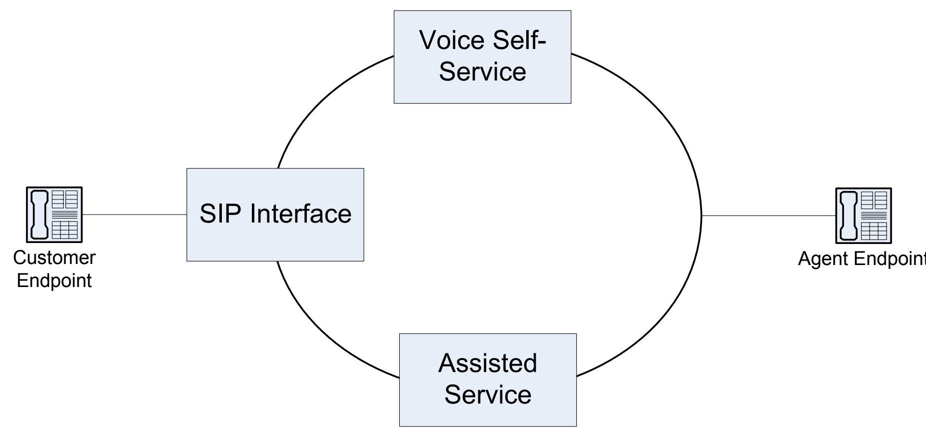


Figure 1: General Functioning of the VPS

The three major functions shown in Figure 1 are:

- **SIP Interface**—SIP Server provides this function, connecting the solution to the external network, and providing call setup and tear down between customer and agent endpoints, as well as between the solution components themselves.
- **Voice Self-Service**—The GVP components provide this function, which can include VoiceXML applications, CCXML applications, Speech Recognition, Text-to-Speech conversion, and other features during the voice dialog portion of the interaction between the calling customer and the contact center.
- **Assisted-Service**—Although not a mandatory part of the solution, Universal Routing Server (URS) is used in most supported call flow scenarios to provide this function. URS controls the routing strategies that deliver the call to an available agent for the assisted-service portion of the call, after the voice dialog portion is completed. URS can also launch self-service applications on GVP directly from the routing strategy.

Features and Benefits

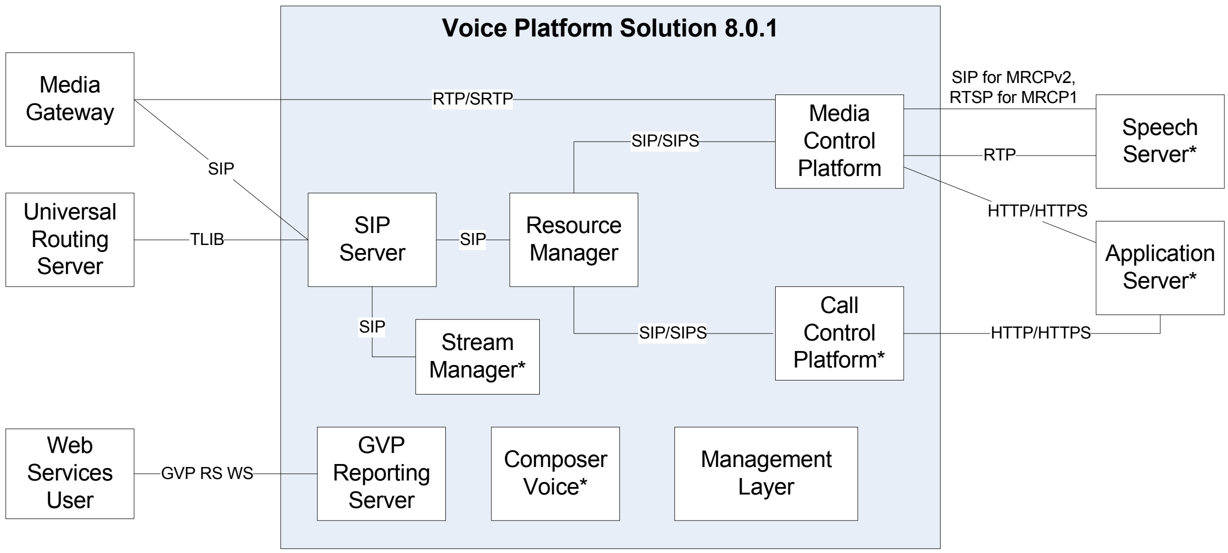
This following list includes the specific high-level features and benefits offered by an integrated Voice Platform Solution 8.0.

- Flexible integration—A single set of integration procedures supports a variety of call flow scenarios.
- Two methods for launching VoiceXML applications:
  - Play Application treatments in the routing strategy.
  - IVR Profile mapping on the Resource Manager
- Multiple methods for transferring calls between the self-service portion of the call to the assisted-service portion. These transfer methods include:
  - SIP `REFER` requests
  - Bridged transfers
  - Consultation transfers using the SIP `REFER with replaces` method
  - Media redirect transfers
- CCXML Conferencing
- Speech Recognition—The VPS supports Media Resource Control Protocol (MRCP) sessions for speech recognition. This feature requires a third-party speech server.
- Text-To-Speech (TTS) technology
- Real-Time Debugging
- High Availability to ensure that services are not interrupted in the event of a failure or process restart. For a basic overview of this feature, see "High Availability—Capability and Limitations" on page 35.

For features and benefits offered by individual components, see the respective product Deployment Guide.

About the Components

Figure 2 shows the component architecture of the Voice Platform Solution 8.0.



Note: Additional connections not shown here include:
- TCP connections between GVP Reporting Server and RM, MCP, and CCP
- Connections between Management Layer and all other Genesys components

*Optional in VPS 8.0.1

Figure 2: Component Architecture

As shown in Figure 2, the Voice Platform Solution 8.0 includes the following components:

- **SIP Server**—SIP Server provides the network interface for the solution. It also provides the CTI link to the T-Library applications used by the solution, such as URS and Agent Desktop.
- **Media Control Platform (MCP)**—This is the core component used to deliver the VoiceXML applications that control the voice self-service portion of the call.
- **Call Control Platform (CCP)**—This SIP-based call controller is used to deliver CCXML applications. It is an optional component, required only if you intend to use CCXML applications in your deployment.
- **Resource Manager (RM)**—The Resource Manager controls access and routing between the various GVP components. It also acts as a proxy for SIP messaging between the GVP components.
- **Voice Platform Reporting Server**—The Reporting Server collects and provides access to data and statistics submitted by VPS components. You can also use Reporting Web Services to make the raw data available for third-party report generation.

- **Genesys Composer Voice**—Composer Voice is an application development tool that developers can use to author the VoiceXML applications or edit the CCXML applications used by the solution. Although not mandatory to the deployment, it is recommended.
- **Stream Manager**—Stream Manager is a Genesys client application that streams media files in order to provide announcements and music to callers queued on Routing Points or ACD queues. It can also serve as a music server or as a Multipoint Conference Unit (MCU). This is an optional component, required for SIP Server control over the playing of announcement or music.
- **Management Layer**—A number of service components are used to provide management capability. You can access the Management Layer using either the Solution Control Interface, or the Genesys Administrator, a web-based interface that lets you start and stop components in the solution, monitor their activity, or make configuration changes as required. Where possible, the integration procedures in this guide use the Genesys Administrator. For more information, see "About Genesys Administrator" on page 31.

Note: The Voice Platform Solution 8.0 does not require either the Speech Server or the Application Server in the deployment. The MCP and CCP can execute simple applications that use pre-recorded audio files stored locally, instead of using the third-party speech or application servers.

How It Works—The Basic Call Flow

In a typical pure-IP deployment, contact center agents are registered on SIP Server (or on a separate T-Server for a hybrid switch). Incoming calls come in to SIP Server from the Public Switched Telephony Network (PSTN) through a third-party media gateway. Depending on how the developer designs the call flow, either the call reaches a routing point on SIP Server, or SIP Server passes the call directly to GVP.

If the call reaches the routing point first, the URS (according to the routing strategy) can initiate the VoiceXML application on GVP as follows:

1. URS sends a `TApplyTreatment` request of the type `TreatmentPlayApplication` to SIP Server.
2. SIP Server sends an `INVITE` to GVP—specifically to the Resource Manager.
3. MCP launches the actual VoiceXML application.

If the routing strategy is not involved in requesting the VoiceXML application, MCP launches the application as mapped on the Resource Manager through IVR Profiles. If the call came to GVP via a SIP Server trunk, a variety of transfer methods are available to transfer the call from GVP to the routing point, where the routing strategy can instruct URS to launch additional VoiceXML applications, or deliver the call to an available agent.

Supported Scenarios

The Voice Platform Solution 8.0 supports a variety of call flow scenarios. Application developers can design the URS routing strategies and VoiceXML/CCXML applications to accommodate these scenarios.

Supported call flow scenarios include:

- URS Launches the Voice Self-Service Application, page 21
- REFER Transfers to Agents on SIP Server, page 22
- Bridged Transfers to Agents on SIP Server, page 23
- Media Redirect Transfers to Agents on SIP Server, page 24
- REFER Transfers to Agents on Private Branch Exchange, page 25
- GVP Launches the REFER with Replaces Transfer Method, page 26
- CCXML Conferencing, page 27
- Speech Recognition, page 28
- Integration With Cisco Call Manager, page 29
- Call Parking on Stream Manager, page 30

URS Launches the Voice Self-Service Application

Figure 3 shows a basic configuration that supports the URS method for launching the voice self-service application.

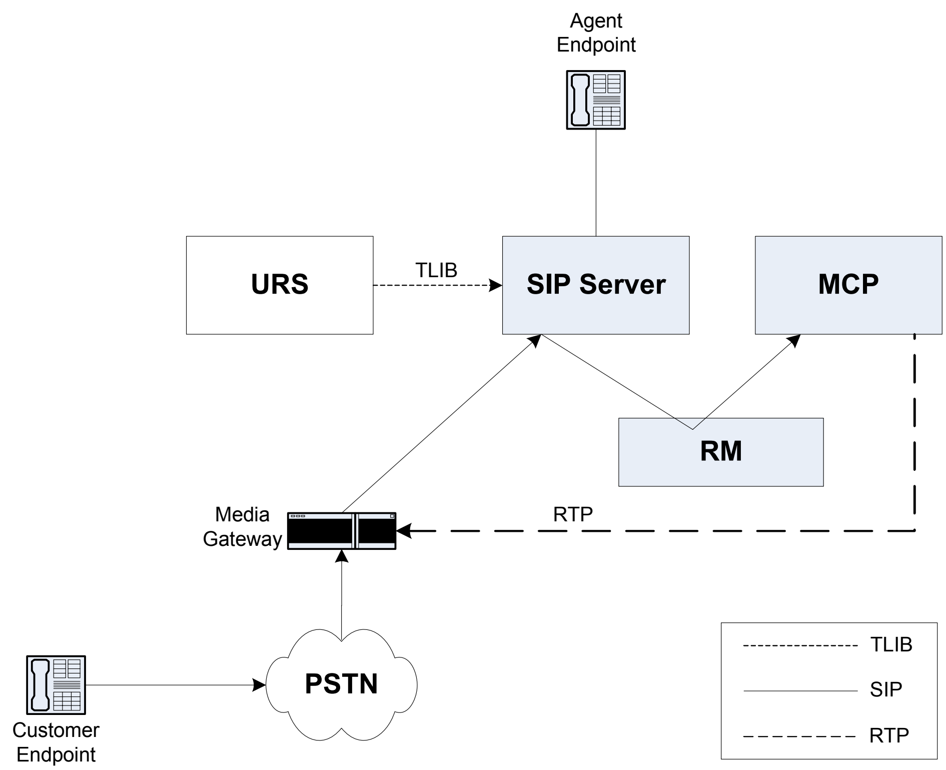


Figure 3: URS Starts the VoiceXML Application—Basic Configuration

In this pure IP-based scenario, the incoming call reaches a `Routing Point` DN on the SIP Server. In the URS strategy loaded on the routing point, a Play Application treatment launches the self-service application on GVP. For each treatment, GVP collects prompts from the user and attaches application data along with the `BYE` request. After the treatment is finished, URS retakes control of the call, at which point it can either execute more Play Application treatments, or route the call to an available agent. The MCP treats any additional treatment from URS as a new SIP call.

Note: When you design a self-service application that is intended for launch from a URS routing strategy, do not include any `<transfer>` tags in the application. GVP cannot execute call control operations in a Play Application treatment.

REFER Transfers to Agents on SIP Server

Figure 4 shows a basic configuration where GVP initiates a blind transfer to an agent registered on the SIP Server.

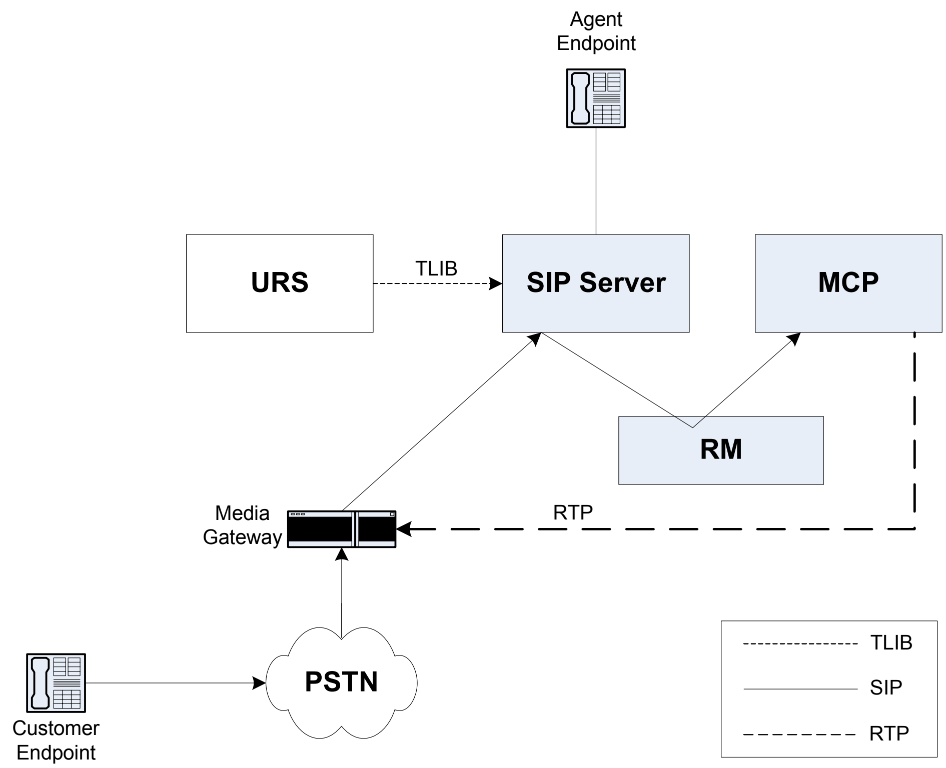


Figure 4: Blind Transfer—Basic Configuration

In this pure IP-based scenario, the incoming call is forwarded directly to GVP, which then executes the initial greeting prompts in a VoiceXML application. When the application decides to transfer the call to an agent, the application executes the `<transfer>` tag, which uses a `REFER` request to execute a blind transfer of the call to a route point on the SIP Server. After SIP Server receives the call, the URS strategy takes control and can at that point initiate any Play Application treatments before routing the call to an agent registered on the SIP Server or other T-Server. After the `REFER` transfer is accepted, the call is considered parked at the URS strategy.

Bridged Transfers to Agents on SIP Server

Figure 5 shows a basic configuration where GVP initiates a bridged transfer to an agent registered on the SIP switch.

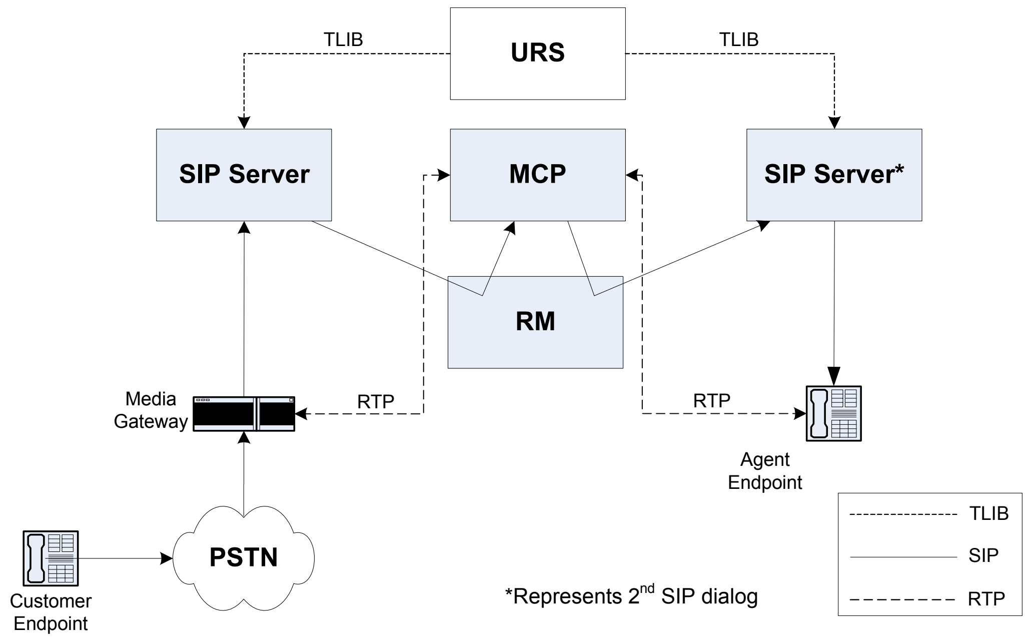


Figure 5: Bridged Transfer—Basic Configuration

In this scenario, the VoiceXML application initiates a bridged transfer to an agent by transferring the call to a `Routing Point` DN on the SIP switch. While the call is being transferred to the agent, SIP Server parks the call on GVP, at which point the URS routing strategy launches a Play Application treatment. After the treatment is finished, SIP Server receives the DN number for the agent and sends an `INVITE` request to both the agent and the bridged call leg simultaneously. This connects the call to the agent. After the call with the agent is completed, the VoiceXML application can continue the voice dialog to complete the call.

Note: Figure 5 shows two instances of the SIP Server, to demonstrate the call path when the caller is connected to the agent; it shows the second SIP dialog established between the MCP and SIP Server during the bridged transfer.

Media Redirect Transfers to Agents on SIP Server

Figure 6 shows a basic configuration where GVP initiates a media redirect transfer to an agent registered on the SIP switch.

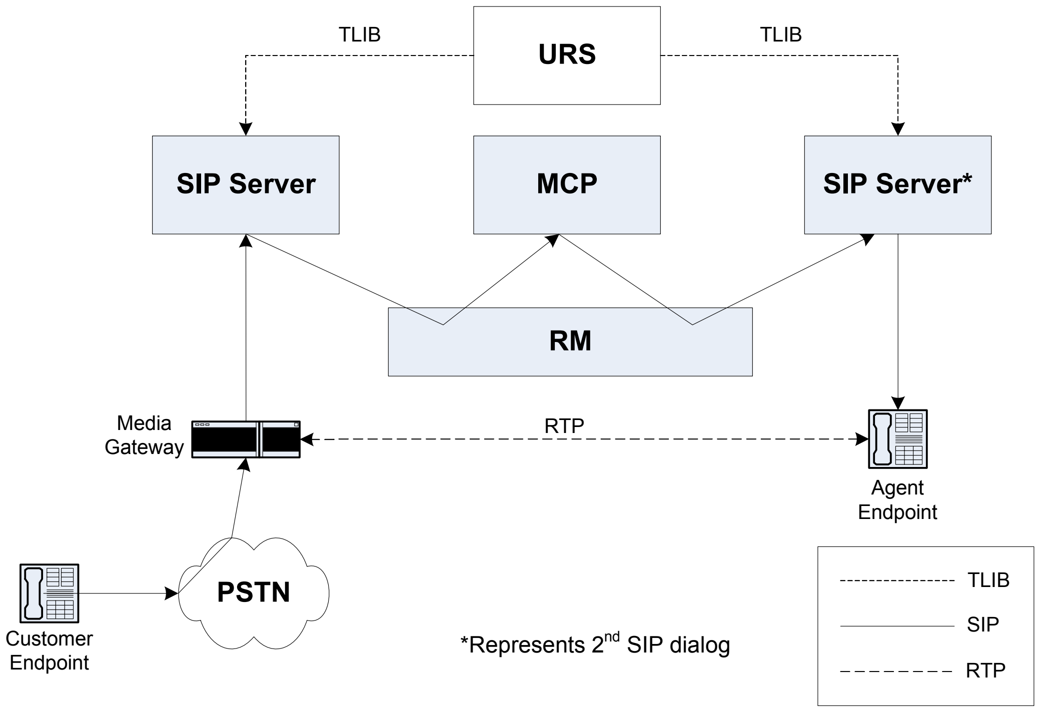


Figure 6: Media Redirect Transfer—Basic Configuration

In this scenario, the VoiceXML application initiates a media redirect transfer to an agent on the SIP Server. Similarly to the bridged transfer, the call is considered parked on the GVP until the transfer is completed. If the agent hangs up the call, the VoiceXML application can continue the voice dialog with the customer—to play further prompts, for example.

The main difference between a media redirect transfer and a bridged transfer is that during a media redirect transfer, the RTP media path is established directly between the customer endpoint and the agent endpoint—the MCP does not bridge the media path. After the transfer is completed, the MCP sends a re-`INVITE` to the customer endpoint to get the media stream back to the MCP. If the transfer fails, the MCP can continue the current VoiceXML dialog to complete the call.

Note: Figure 6 shows two instances of the SIP Server, to demonstrate the second SIP dialog established between the MCP and SIP Server during the media redirect transfer.

REFER Transfers to Agents on Private Branch Exchange

Figure 7 shows a configuration that includes a Private Branch Exchange (PBX) in an enterprise deployment.

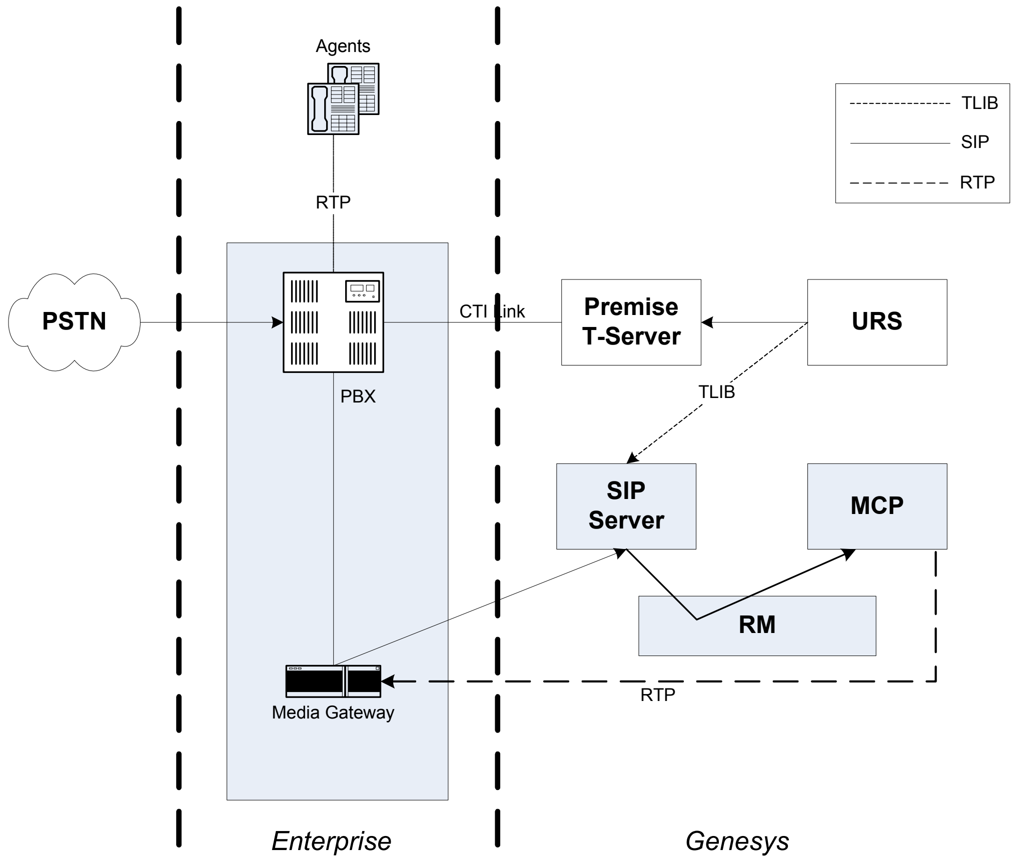


Figure 7: Enterprise PBX—Basic Configuration

In this scenario, GVP sits behind the PBX, and the PBX can direct calls to the VPS through the media gateway. Agents are registered as telephone extensions on the PBX.

Time-Division Multiplexing PBX

For a Time-Division Multiplexing (TDM) PBX, you can configure the PBX for Direct Inward Dialing (DID) by using either of two methods:

- Configure the PBX to accept a DN assigned to a DID number, for delivery to SIP Server through the media gateway.
- Configure the DID number as a `Routing Point` DN on the SIP switch.

In the first method, the PBX sends the call for the DID number through the media gateway to the SIP switch. Upon receiving the call, SIP Server directs the call to the MCP, which then launches the self-service application. The application can transfer the call to an agent by invoking a blind transfer with `REFER` on the MCP, at which point the MCP also attaches application data to the `REFER` request.

The URS strategy can then use Play Application treatments to launch further prompts, or it can transfer the call to the routing point on the PBX. The Premise T-Server assumes responsibility for transferring the call to an available agent and completing the call.

In the second method, SIP Server receives the call from the PBX and then recognizes the DID number as matching the routing point, at which point the URS strategy takes control of the call. The strategy can use Play Application treatments to launch VoiceXML applications. The application can collect customer information, and the MCP attaches that data to the `BYE` message.

IP-Enabled PBX

If your deployment uses an IP-based PBX, a simpler configuration that uses SIP trunks is possible. In this case, the media gateway is no longer involved in passing calls for the DID number to the SIP Server. The IP PBX sends these calls directly to the SIP Server using the SIP trunk.

GVP Launches the REFER with Replaces Transfer Method

Figure 8 shows a configuration where the MCP performs a consultation transfer using the `REFER with replaces` method.

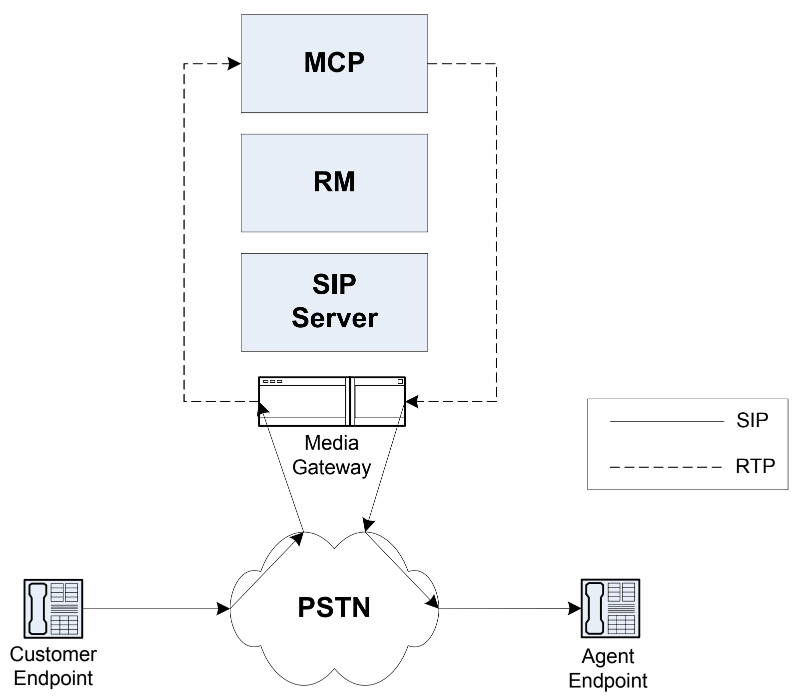


Figure 8: REFER with replaces—Basic Configuration

In this scenario, the VoiceXML application initiates a transfer, at which point the MCP sends a consultation call to the destination DN. After the SIP dialog is established, the MCP sends a `REFER with replaces` request to the originating DN to merge the original and consultation calls. If the originating DN is connected through a media gateway, the outbound call to the destination DN must land on the same media gateway as well. Resource Manager ensures that this happens.

Figure 8 shows the call state where the MCP has reached the destination DN, but just before it sends the `REFER with replaces` request to the originating party. After call establishment is completed, MCP is no longer involved in either call, as both calls then reside on the media gateway.

Note: In Figure 8, SIP Server is involved in passing the `REFER with replaces` messages from MCP to merge these calls; however, to simplify the call flow, those parts of the flow are not included in the diagram.

CCXML Conferencing

Figure 9 shows a more complex example of third party call control involving a CCXML application.

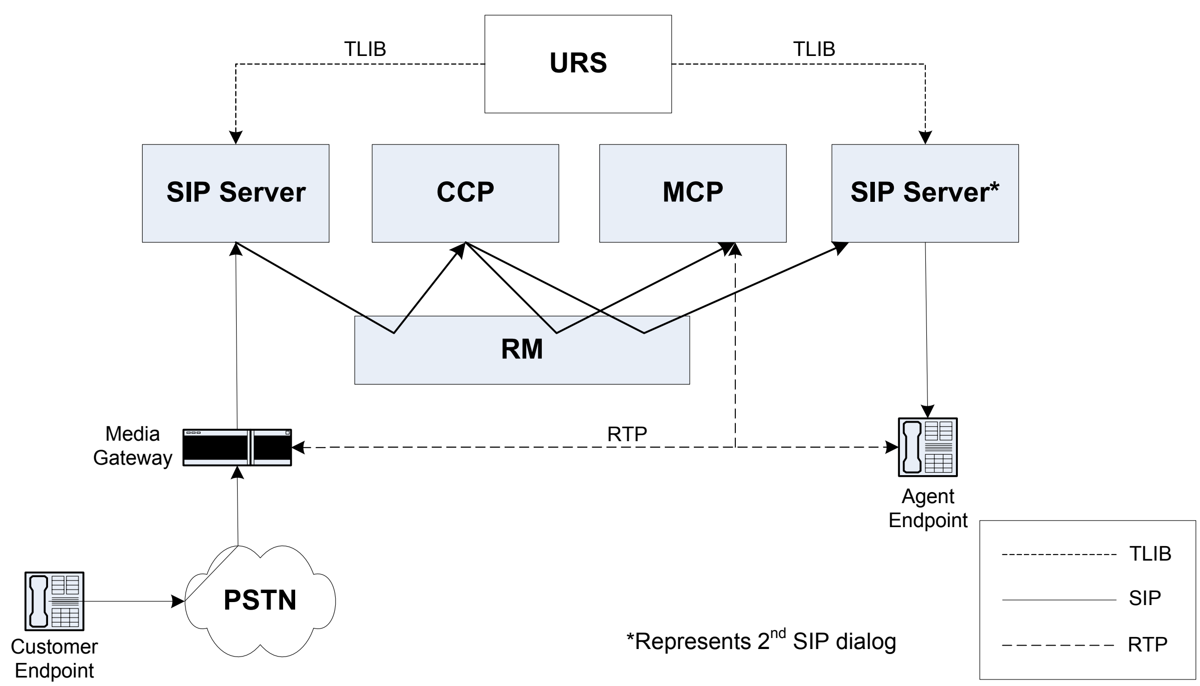


Figure 9: CCXML Conferencing—Basic Configuration

In this scenario, SIP Server forwards the incoming call to GVP, at which point the Resource Manager maps the DID number to a CCXML application. The RM then forwards the call to the Call Control Platform (CCP), which executes the CCXML application. The application starts a conference call with the MCP. CCP establishes the media session between the originating DN and the conference call. At the same time, the CCXML application starts a new voice dialog with the MCP, which allows the application to join the media output from the voice dialog to the conference call. The CCP makes a call to a routing point on the SIP Server, at which point the URS strategy takes control and routes the call to the agent. The CCP adds the media path with the agent to the conference. After the call is established, the originating DN, the VoiceXML dialog, and the agent are all joined in the conference.

Note: Only GVP can initiate conferences from the CCXML application. In cases where the URS routing strategy Play Application treatment calls a CCXML application, that application cannot include any requests for conferences.

Speech Recognition

Figure 10 shows a scenario where MCP establishes an Media Resource Control Protocol (MRCP) session for speech recognition.

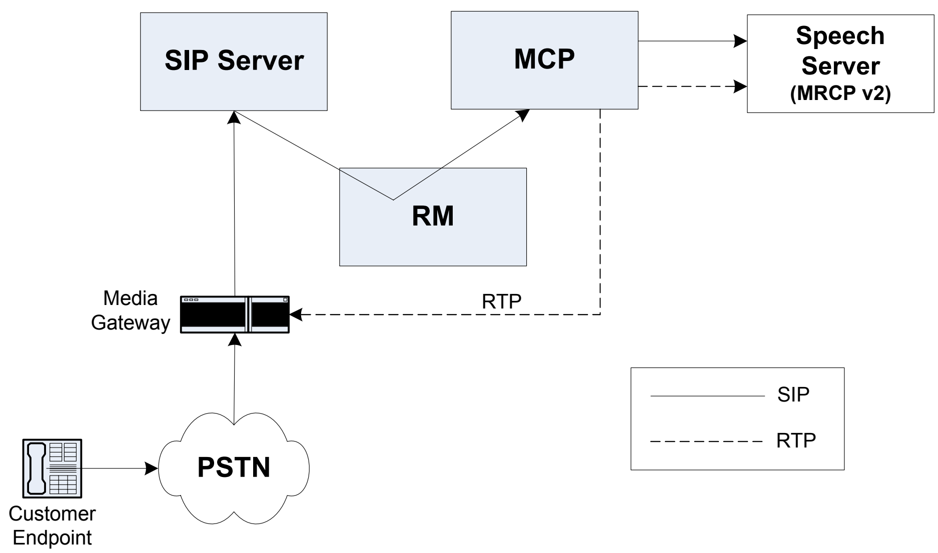


Figure 10: Speech Recognition—Basic Configuration

In this scenario, the MCP establishes a SIP dialog directly with the third-party speech server to establish an MRCP session for speech recognition. The SIP dialog establishes the RTP media path between the MCP and the speech server so that speech can be recognized, and between the MCP and the media gateway so that it can play prompts for the caller.

Integration With Cisco Call Manager

Figure 11 shows a configuration that includes the Cisco Call Manager (CCM) in an enterprise deployment.

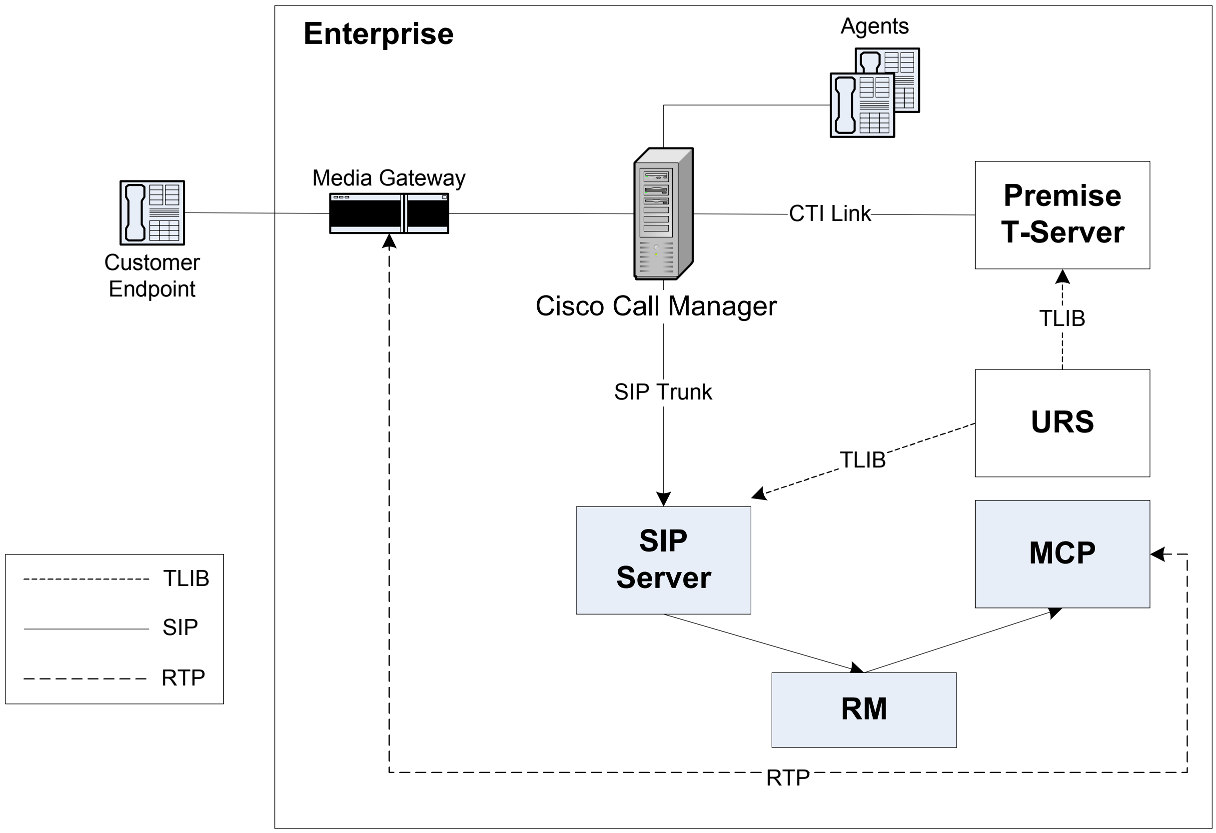


Figure 11: VPS Integration with Cisco Call Manager

In this scenario, the Voice Platform Solution is integrated with the Cisco Call Manager (CCM), an IP telephony call-processing system that acts as both a softswitch, with control over the media gateway for routing incoming calls, and as a PBX, with enterprise agents registering with the CCM.

Note: The CCM does not support line-side connections. You must configure a SIP trunk connection for the VPS on the CCM. For information about configuring CCM, consult the Cisco-specific documentation Cisco CallManager System Guide and the Cisco CallManager Administration Guide.

Call Parking on Stream Manager

Figure 12 shows a sample deployment that includes Stream Manager, which can provide music to callers while they wait in queue for an agent to become available.

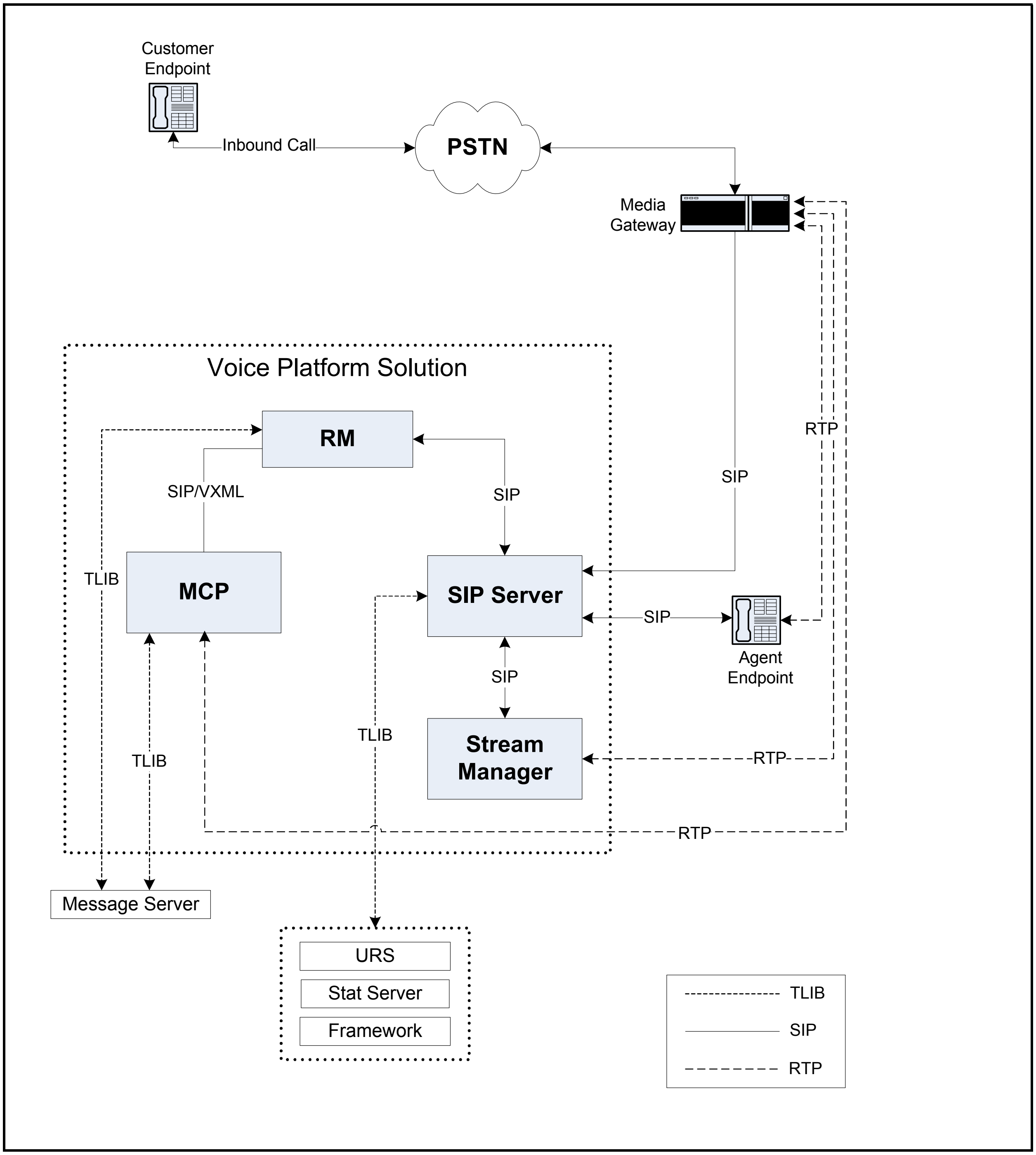


Figure 12: Sample Stream Manager Integration in the VPS

In this scenario, Stream Manager is integrated into the VPS as a client application of SIP Server. A typical incoming call reaches the VPS through a `Trunk Group` DN, initiating a VoiceXML application on MCP. At the end of this initial self-service portion of the call, the VoiceXML application transfers the call to a `Routing Point` DN, where the URS routing strategy tries to route the call to an agent. If no agent is available, the call is then parked on Stream Manager, providing music to the caller while they wait. After an agent becomes available, the call completes its transfer—the call is negotiated and established between the caller and the agent.

About Genesys Administrator

The Genesys Administrator is a GUI that provides a web-based interface to the Genesys Configuration and Management Layers.

Use the Genesys Administrator to deploy, configure, provision, and monitor 8.0 components. For more detailed information about the Genesys Administrator, see the *Framework 8.0 Deployment Guide* and the *Framework 8.0 Genesys Administrator Help*.

Figure 13 shows a typical Genesys Administrator page.

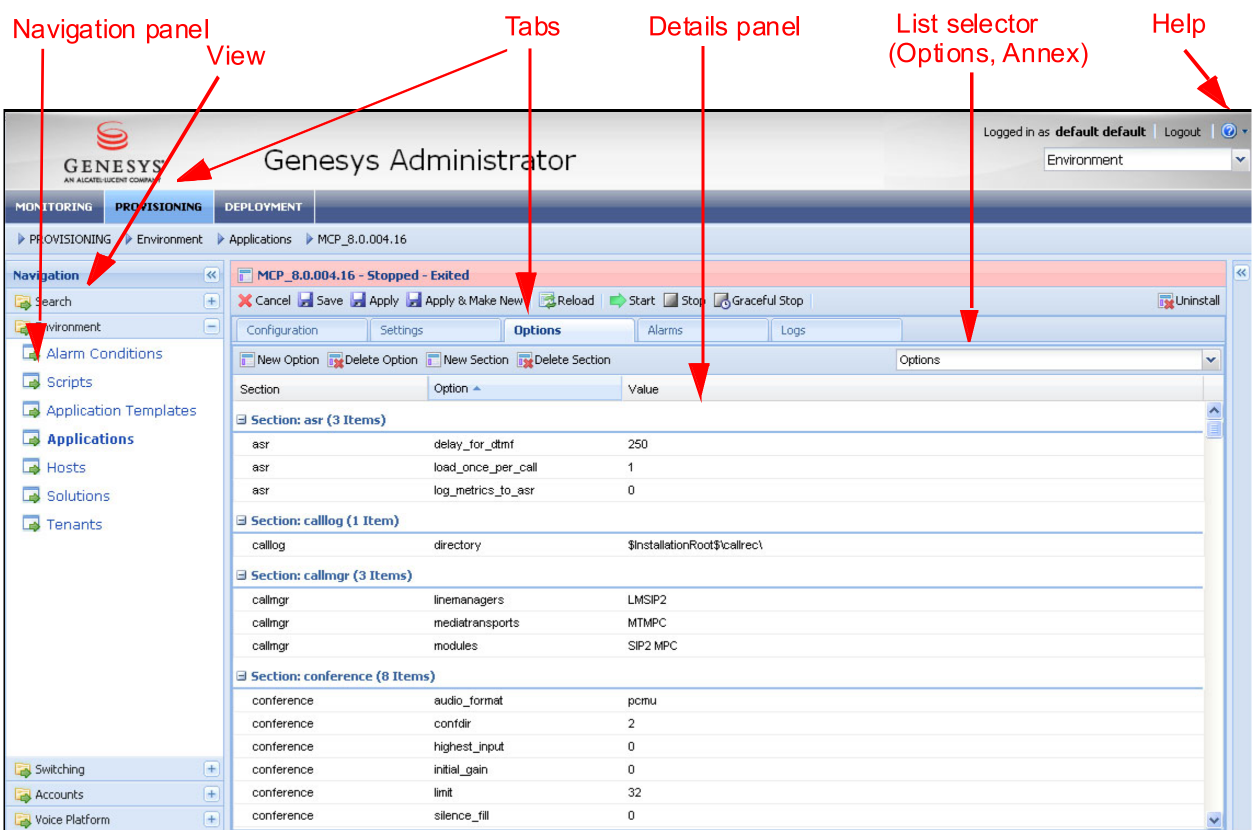


Figure 13: Genesys Administrator

To access the Genesys Administrator for your Genesys deployment, go to the following URL:

`http://<Genesys Administrator host>/wcm`

Configuring GVP Processes in the Genesys Administrator

The following procedure describes how to configure GVP `Application` objects in the Genesys Administrator. For more information about using the Genesys Administrator, see the online *Framework 8.0 Genesys Administrator Help.*

Procedure:
Viewing or modifying GVP configuration parameters

Purpose: To describe the general method to use the Genesys Administrator to view or modify configuration options in GVP `Application` objects.

Summary

This procedure describes how to use the `Settings` tab to view information about or to modify configuration parameters.

You can also use the `Options` tab to modify parameters. However, Genesys recommends using the `Settings` tab because it gives you access to the metadata descriptions, and also validates settings. Furthermore, for ease of use, the `Settings` tab only exposes those parameters that are needed by the customer, hiding the rest.

Prerequisites

- The `Application` object has been created as described in the *Genesys Voice Platform 8.0 Deployment Guide.* In particular, for GVP `Application` objects, the `Application` was created from an `Application Template` into which metadata has been imported.
- You are logged in to the Genesys Administrator. To access the Genesys Administrator, go to the following URL:
  `http://<Genesys Administrator host>/wcm`

Start of procedure

1. In the Genesys Administrator, go to `Provisioning` > `Environment` > `Applications` > `<Component Application>` > `Settings` tab (Figure 14).

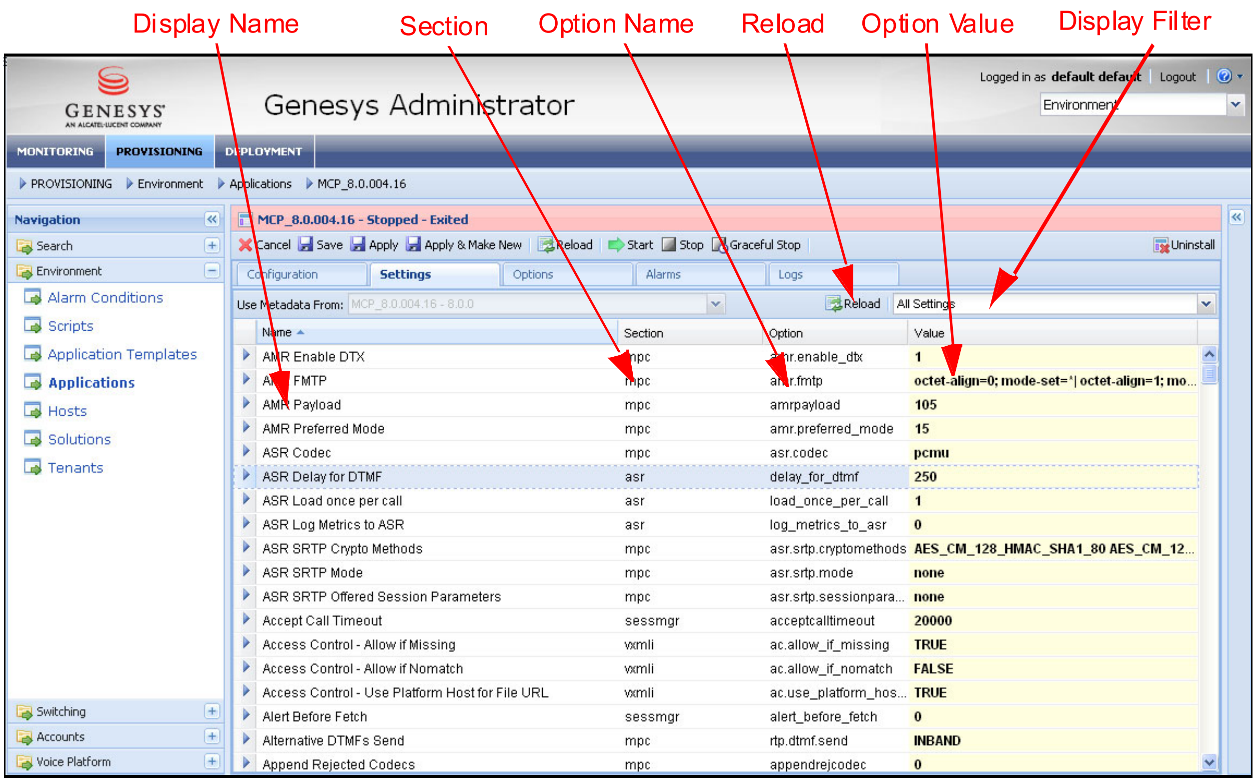


Figure 14: Settings Tab

For each configurable parameter, the `Settings` tab displays the following information:

- A plain-language display name.
- The configuration section that contains the option.
- The configuration option name, as it would appear on the `Options` tab.
- The current option value, either user-defined or default. User-defined values display in bold.

  User-defined values also appear on the `Options` tab.

2. You can change the display in a number of ways:

- To sort the information in ascending or descending order by column, click the column header to activate a drop-down list, and select the desired sort order option.
- To show or hide a column, click any column header to activate a drop-down list, select the `Columns` submenu, and select or clear check boxes in the `Columns` list to show or hide columns.

- To filter the options that are displayed, select a different grouping from the `Display Filter` drop-down list.

  For example, the Media Control Platform `Application` provides the following option groupings:
  - Media Control Platform Main Settings
  - Logging
  - GVP Logging and Reporting
  - Conference Application Module Settings
  - Media Processing Settings
  - Session Manager Settings
  - SIP Settings
  - Speech Resource Management Settings
  - SNMP Settings
  - NGI Settings
  - All Settings

3. To change an option setting:
   a. Click the `Name` or `Value` of the option that you want to change.

      A dialog box appears, which displays a full description of the option, its current value, valid values, default value (if applicable), and when changes takes effect.

      You can view the same information for all configuration options in the *Genesys Voice Platform 8.0 Configuration Options Reference*.
   b. Enter the new value in the `Value` field or, if applicable, select the option from the `Value` drop-down list. In cases where multiple values can be selected, hold down `Ctrl` to select multiple values.
   c. Click `OK`.
4. To save your changes, click `Save` and then `Apply`.

   Option values that you change on the `Settings` tab get updated on the `Options` tab as well.
5. To update the metadata descriptions (from an updated Templates XML file), click `Reload`. This reloads the metadata file without affecting configured option values.

End of procedure

More Information

- For information about installing the Genesys Administrator, see the *Framework 8.0 Deployment Guide.*
- For general information about using the Genesys Administrator, see the online *Framework 8.0 Genesys Administrator Help.*

- For information about using the Genesys Administrator to configure and provision GVP `Application` objects, or for monitoring GVP and viewing reports, see the *Genesys Voice Platform 8.0 User's Guide*.

High Availability—Capability and Limitations

High Availability (HA) ensures that a service is not interrupted in the event of a failure or a process restart. For 8.0, the Voice Platform Solution supports high availability for Resource Manager failures only, with the priority on making sure that all new calls are processed. However, SIP messages sent after the initial `INIVTE`, as well as usage limits may not survive the failover from active to standby Resource Manager.

For information about configuring active/standby Resource Manager pairs for high availability, see the Appendix "NLB Clustering for Resource Manager" in the *Genesys Voice Platform 8.0 Deployment Guide*.

What Happens If the Resource Manager Fails?

High availability protects all established calls. If the active Resource Manager in a high availability pair fails, the standby Resource Manager takes over all existing calls without impact—provided the caller sends no new SIP messages during the active session.

Limitations for existing calls

If the caller sends further SIP messages for an established call after the failover, the following limitations apply:

- `BYE` messages for calls initiated before the failover cannot be processed. The request times out, and the failover Resource Manger responds with a `481` response, indicating the call/transaction does not exist.
- `INFO` requests for calls initiated before the failover cannot be processed, because the failover Resource Manager has no access to the transaction information. As a result, any further call processing that requires the information from the failed `INFO` request will also fail.
- The session timer used to manage Resource Manager sessions cannot survive the failover. If the caller sends a re-`INVITE` to refresh the session, the failover Resource Manager considers it a new `INVITE`, and is unable to refresh the session.

Limitations for call transfers

For calls flows that include a blind or bridged transfer out of GVP, `REFER` messages sent after the failover will only succeed if they are sent to the same gateway as the one that sent the original inbound `INVITE` request. Limitations for these transfers include:

- If your deployment includes multiple gateways, `REFER` message for blind transfers may not arrive at the correct gateway, even if the `use-same-gateway` option is enabled, because the failover Resource Manager has no access to information about the initial `INVITE` request.
- If the `REFER` message was sent out before the failover, the failover Resource Manager cannot sent the `NOTIFY` message in response.
- After the failover, if the MCP sends an outbound `INVITE` message for a bridged transfer, there is no guarantee that the message will route to the same gateway that initiated the call. Only requests to the original gateway will be successfully transferred.

Limitations for usage limits

None of the port capacity settings or policies related to session counts or usage limits can survive the failover. These include CCXML, VoiceXML, and * usage limits. Other settings remain unaffected by the failover.

Chapter

2 VPS 8.0 Integration Procedures

This chapter describes how to integrate the various components of the Voice Platform Solution (VPS)—primarily the configuration steps that are required to integrate SIP Server with the Genesys Voice Platform (GVP) components included in the solution. These procedures assume that all of the components involved in the solution have already been installed and their initial configuration completed, according to the procedures in their respective product Deployment Guides.

These procedures support the minimum configuration required to integrate the solution. This basic infrastructure supports multiple architecture configurations and transfer modes, as described in "How It Works—The Basic Call Flow" on page 19 and "Supported Scenarios" on page 20.

This chapter includes the following sections:

Task Flow

Table 1 provides an overview of the main steps that you must complete in order to integrate SIP Server with the other VPS components.

Table 1: Task Flow to Integrate SIP Server and GVP

Objective	Related Procedures and Actions
1. Check that prerequisite components are successfully deployed.	Make sure all required VPS components are deployed before you begin the integration procedures. For a list of required components, their respective deployment guides, as well as any key actions or information to ready the component for the integration, see "Integration Prerequisites" on page 43.
2. Check default ports.	Check default port settings for the various components: 1. All GVP components are assigned different default ports, allowing you to run GVP on a single host—typically for lab or testing purposes. If you change any of these default port settings, be sure they do not conflict with the settings of any other component running on that computer. 2. The Resource Manager uses a default port setting of `5060`, which is also the default for SIP Server. If your deployment includes Resource Manager and SIP Server on the same host, Genesys recommends that you change the Resource Manager default port to avoid conflicts with SIP Server, or any other GVP components.
3. Configure MCP for integration with SIP Server.	To prepare MCP for integration with SIP Server, in the `SIP` section of the MCP `Application` object, configure the following options: • Set the `outcalluseoriggw` option to `1`. • For the `routeset` option, enter the IP address and port for the Resource Manager, in the following format (outer angle brackets included): `<sip:<RM IP address>:<RM SIP port>, lr>` For the detailed procedure, see "Configuring MCP for integration with SIP Server" on page 46.

Table 1: Task Flow to Integrate SIP Server and GVP (Continued)

Objective	Related Procedures and Actions
4. Configure Resource Manager for integration with SIP Server.	To communicate with SIP Server, Resource Manager requires a gateway resource that represents SIP Server. 1. Create a logical resource group on the RM. 2. Configure the connection between the RM and the gateway resource access point to include the SIP Server. For the detailed procedure, see "Configuring a Gateway Resource for SIP Server on the Resource Manager" on page 48.
5. Create SIP Server application objects.	If your configuration does not already include the following prerequisite SIP Server-related objects, create them now: 1. Create a SIP `Switching Office` object. 2. Create a SIP `Switch` object. For detailed procedures, see the *Framework 7.6 SIP Server Deployment Guide*.
6. Create the link between SIP Server and GVP.	To enable SIP Server to identify GVP in the solution, create the following DNs: • Create a GVP `TrunkGroup` DN on the SIP switch, pointing the `TServer > contact` option to the IP address and port for Resource Manager. • Create a `Voice over IP Service` DN on the SIP switch, pointing the `TServer > contact` option to the IP address and port for Resource Manager. For the detailed procedure, see "Linking SIP Server with GVP" on page 50.
7. Configure user data exchange between SIP Server and GVP.	If your call flow design requires the exchange of customer data between SIP Server and the VoiceXML application, you must configure SIP Server so that it maps data in both of the following directions: • SIP Server takes data from GVP SIP messages and attaches it to the call (that is, maps the data to the T-Library `EventAttachedDataChanged` message). • SIP Server attaches data from the T-Library message and adds it as headers in the SIP message sent to GVP. (*continued on next page*)

Table 1: Task Flow to Integrate SIP Server and GVP (Continued)

Objective	Related Procedures and Actions
7. (continued) Configure user data exchange between SIP Server and GVP.	There are two options available to configure this mapping: • `userdata-map-trans-prefix`—SIP Server maps user data from all custom headers with the prefix specified in this option. Configure this option in the `TServer` section of the SIP Server `Application` object. • `userdata-map-filter`—Use this option to specify which headers need to be mapped for user data required by GVP. If GVP does not need the user data, then you can leave this option undefined. Configure this option in the `TServer` section of the GVP `TrunkGroup` and `Voice over IP Service` DNs. For a more detailed procedure, see "Enabling user data exchange between SIP Server and GVP" on page 53. For examples of user data mapping, see Appendix A, "Sample User Data Mapping" on page 69.
8. Create SIP extensions for your agent endpoints.	If your SIP Server configuration does not already include SIP agent endpoints (SIP phones connected to the switch), then you must create at least one in order to test the sample routing strategy that you will create in later steps. 1. On the SIP Switch, click `DNs` and create an extension for each SIP phone you want to connect to the switch. 2. On the `Options` tab of the DN, create a `TServer` section with a `contact` option whose value points to the IP address of the SIP or agent endpoint. **Note**: This is the minimum configuration required for a SIP endpoint, suitable for testing the integrated solution. For more information about configuring endpoints, see the *SIP Sever 7.6 Deployment Guide*.
9. Create a Routing Point DN on the SIP Server.	For URS routing to agents, as well as to start the VoiceXML application, you need to create a `Routing Point` DN on the SIP Server switch. For the detailed procedure, see "Creating a Routing Point on the SIP Server" on page 57.
10. Create a routing strategy and load it on the Routing Point DN.	Use the Interaction Routing Designer to: 1. Create the routing strategies that URS uses to route calls to agents and launch VoiceXML applications. 2. Load the strategy on the `Routing Point` DN that you created in Step 9. For the detailed procedure, see "Creating and loading a routing strategy" on page 57.

Table 1: Task Flow to Integrate SIP Server and GVP (Continued)

<table>
<tr><th>Objective</th><th>Related Procedures and Actions</th></tr>
<tr><td colspan="2">Additional Special Configuration</td></tr>
<tr><td>If your gateway does not support <code>REFER</code> transfers...</td><td>Create a <code>Trunk</code> DN for the gateway, and a <code>TServer</code> section with the following options:
• Set the <code>refer-enabled</code> option to <code>false</code>.
• For the <code>prefix</code> option, enter the prefix of the ANI for the incoming call.
• For the <code>contact</code> option, enter the IP and port of the media gateway.
For the detailed procedure, see “Creating a Trunk DN for gateways that do not support REFER” on page 61.</td></tr>
<tr><td>If you design your call flow to process all incoming calls first in the URS routing strategy...</td><td>1. Instead of creating a separate GVP <code>TrunkGroup</code> DN for every contact number—as you did in Step 6—create a generic DN that the strategy can use when routing the call. Instead of numbering the DN for the customer dialed number, use a generic identifier. For example, <code>GVP_TrunkGroup</code>.
2. Modify the URS routing strategy in IRD, so that the <code>Function</code> block targets the generic <code>TrunkGroup</code> DN (that is, <code>GVP_TrunkGroup</code>), instead of the DNIS prefix.</td></tr>
<tr><td>To configure MCP for media redirect transfers.</td><td>The VPS supports two types of bridged transfers: bridge and media redirect. You can configure these methods by defining the default transfer method in the MCP <code>Application</code> object:
• For bridge transfers, no special configuration is required. The solution defaults to this method for bridged transfers.
• For media redirect transfers, you may have to configure the MCP as follows:
On the <code>Settings</code> tab, in the <code>sip</code> section, set the <code>defaultbridgexfer</code> option to <code>MEDIAREDIRECT</code>.
Note: If you want to switch back to the bridge transfer method, then you must set the <code>defaultbridgexfer</code> option back to the default value of <code>BRIDGE</code>.
For a detailed procedure, see “Configuring the solution for media redirect transfers” on page 62.</td></tr>
</table>

Table 1: Task Flow to Integrate SIP Server and GVP (Continued)

Objective	Related Procedures and Actions
To integrate a PBX into the VPS.	For deployments that include a TDM or IP-based PBX, additional integration steps include: 1. Install and configure the premise T-Server for your switch, including the following switch-related configuration objects: • `Switching Office` object • `Switch` object • Premise T-Server `Application` object 2. Configure Access Codes for ISCC communication between the premise switch and the SIP switch. 3. Configure coordinated telephony objects (DNs) on the premise switch and in the Configuration Layer. For more information, refer to the T-Server Deployment Guide for your particular switch. For an overview of a PBX scenario, see “REFER Transfers to Agents on Private Branch Exchange” on page 25.
To integrate Stream Manager into the VPS.	To support call parking on Stream Manager in the VPS, complete the following steps: 1. In the Stream Manager application, set the `sip-port` on the `Options` tab to the SIP messaging port used by the solution—typically `5060`. 2. Create a `Voice over IP Service` DN, setting the `service-type` option in the `Annex` tab to the SIP Service that you want Stream Manager to provide. For music or announcements, set the value to one of the following: `music`, `moh`, or `treatment`. 3. Add an announcement or music treatment block to a new or existing URS routing strategy, provisioning the block to point to a media file in the Steam Manager installation directory. For a more detailed procedure, see “Integrating Stream Manager into the solution” on page 63.

Integration Prerequisites

Before you begin the integration, all VPS components must be installed and configured according to the procedures in their respective product Deployment Guides. If properly deployed, you can run these applications from the Genesys Administrator or the Solution Control Interface (SCI).

Table 2 lists all of the required prerequisite components, their respective Deployment Guides, as well as any key actions that you must complete before starting the integration procedures.

Table 2: Prerequisite Components for the VPS Integration

Component	Key Actions or Info	Documentation
VPS Components		
SIP Server 7.6	Requires the following SIP Server-related configuration objects: • SIP Server `Application` object. • SIP `Switching Office` object. • SIP `Switch` object. • SIP or agent endpoints, configured as `Extension` DNs on the SIP Switch.	*Framework 7.6 SIP Server Deployment Guide*
Genesys Voice Platform 8.0	Key GVP components required for the VPS include: • Resource Manager • Media Control Platform • Call Control Platform (optional for CCXML) • Fetching Module (one per host) • Squid Caching Proxy (one per Fetching Module) Some components require additional related Configuration Layer objects or third-party servers. Consult the Deployment Guide for details.	*Genesys Voice Platform 8.0 Deployment Guide*

Table 2: Prerequisite Components for the VPS Integration (Continued)

Component	Key Actions or Info	Documentation
Management Framework 8.0	A centralized Genesys Management Framework, with all required components, must be installed. To start the Management Layer (required for Genesys Administrator): 1. Start the LCA. 2. Start the DB Server that provides access to the Configuration Database. 3. Start Configuration Server. 4. Start Message Server. 5. Start Solution Control Server.	*Management Framework 8.0 Deployment Guide*
Genesys Administrator 8.0	Consult the *Management Framework 8.0 Deployment Guide* to install and configure Genesys Administrator. Before logging on to Genesys Administrator, make sure the following are started: • Configuration Server • Solution Control Server • Microsoft Internet Information Services (IIS) To log on to Genesys Administrator: 1. Enter the application URL in a web browser, using the following format: http://<ga_host>/wcm/LoginEJS.aspx **Note**: Make sure you enter the host where you installed Genesys Administrator (do not confuse with Configuration Manager host). 2. Login to the tool by entering the following info: • User name • Password • Application • Host Name (Configuration Server host) • Port (Configuration Server port)	*Management Framework 8.0 Deployment Guide* *Framework 8.0 Administrator Help*

Table 2: Prerequisite Components for the VPS Integration (Continued)

Component	Key Actions or Info	Documentation
Genesys Composer Voice 8.0	This is an optional component. Genesys recommends that you use Composer Voice for authoring VoiceXML and editing CCXML applications. You can start Composer Voice from the Windows `Start` menu. For information about using the tool, press `F1` in the application to bring up the Help system.	*Genesys Composer Voice 8.0 Deployment Guide* *Genesys Composer Voice 8.0 Help* *Genesys Voice Platform 8.0 VoiceXML 2.1 Help*
Other Genesys Components **Note**: These components are not part of the Voice Platform Solution, although they may be included in your deployment.		
Universal Routing Server 7.6	Required for routing to agents and for launching Play Application treatments. Required connections: • Message Server • SIP Server `Application` object • Stat Server	*Universal Routing Server 7.6 Deployment Guide* *Universal Routing Server 7.6 Reference Manual*
Interaction Routing Designer 7.6	Required for building URS routing strategies, and for defining the UserData attributes that SIP Server collects from the headers and passes to GVP.	*Universal Routing 7.6 Business Process User's Guide* *Universal Routing 7.6 Interaction Routing Designer Help*

Table 2: Prerequisite Components for the VPS Integration (Continued)

Component	Key Actions or Info	Documentation
Stat Server 7.6	Required for monitoring the availability of agents targeted in the routing strategies. Required connections: • Message Server • SIP Server `Application` object	*Framework 7.6 Stat Server Deployment Guide*
Stream Manager 7.6	This is an optional component. With Stream Manager, the VPS can give SIP Server control over the playing of announcements or music, providing more flexibility for the call flow designer. For detailed deployment information, see the *Framework 7.6 Stream Manager Deployment Guide* For information about additional Steam Manager functionality, see the Framework *7.6 SIP Server Deployment Guide.*	*Framework 7.6 Stream Manager Deployment Guide* *Framework 7.6 SIP Server Deployment Guide*

Integration Procedures

This section provides detailed procedures for integrating SIP Server with the other VPS components.

Procedure: Configuring MCP for integration with SIP Server

Purpose: To configure the Media Control Platform (MCP) for integration with SIP Server.

Start of procedure

1. In Genesys Administrator, select `Provisioning > Environment > Applications`, and click the GVP Media Control Platform `Application` object.
2. On the `Settings` tab, set the `Use Original Gateway in Outbound Call` parameter (`sip` section, `outcalluseoriggw` option) to `Enable`.

   Tip: For more information about this parameter, and other parameters used in this procedure, see Table 3.

3. On the `Settings` tab, set the `Route Set` parameter (`sip` section, `routeset` option) to the IP address and port for the Resource Manager, in the format (outer angle brackets included):

   `<sip:<ip_address:port>;lr>`

   For example:

   `<sip:192.168.50.169:5070;lr>`

4. Click `Save` to save the changes.

End of procedure

Additional Info

Table 3 describes in greater details the options that are configured in this procedure.

Table 3: Configuring MCP Options for SIP

Name	Section	Option	Value
Use Original Gateway in Outbound Call	sip	`outcalluseoriggw`	Set this value to 1 so that MCP is able to resolve hosts in cases where the VoiceXML `<transfer>` request does not specify the destination attribute. See the option description on page 77.
Route Set	sip	`routeset`	To point MCP to the routeset on the Resource Manager, enter the IP address and port number for the sip proxy (Resource Manager), using the following format (outer angle brackets included): `<sip:<RM IP address>:<RM SIP port>;lr>` **Note**: If including multiple proxies in the string, separate each by a comma. See the option description on page 78.

Next Steps

- Configuring a Gateway Resource for SIP Server on the Resource Manager

Procedure: Configuring a Gateway Resource for SIP Server on the Resource Manager

Purpose: To create a gateway resource that the Resource Manager uses to communicate with SIP Server.

Summary

Configure a logical resource group of the `gateway` type, then configure the connection between the Resource Manger and the gateway resource access point.

Prerequisites

- To configure SIP Server as a gateway resource, GVP requires a gateway `Application` object of the type `Resource Access Point`, and a connection between the RM and this gateway.

  If your configuration does not include the gateway `Application` object or the RM-to-gateway connection, you must create them. For detailed procedures, see the chapter "Post-Installation Activities on the Hosts" in the *Genesys Voice Platform 8.0 Deployment Guide*.

Start of procedure

1. In the Genesys Administrator, go to the `Provisioning > Voice Platform > Resource Management` panel (see Figure 15).

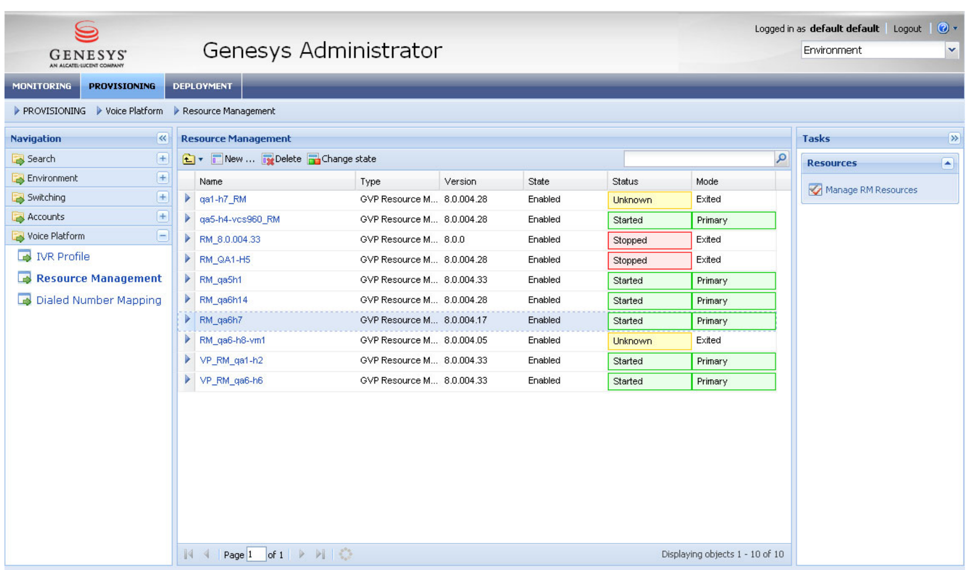


Figure 15: The Resource Management Panel

2. To select the Resource Manager for which you want to configure the logical resource group, highlight anywhere in the applicable row but the `Name`.

   Tip: If you click the `Name`, you will open the configuration tab for the application itself, not the GVP Manage Resources wizard.

3. In the `Tasks` panel, click `Manage RM Resources`.

   The `GVP - Manage <Resource Manager> Resources` wizard displays.

4. Create a new gateway group for the SIP Server:
   a. On the `Step 1: Group List` page, click `New` to create a new group.
   b. In the `New Group` dialog box, enter the following mandatory parameters:
      - `Name`—Enter a name for your gateway group.
      - `Services Type(s)`—Select `gateway`.
      - `Port Usage Type`—Select `in-and-out`.

      Tip: For a more detailed description of these options, see Table 4.

   c. Click `OK`.

      The wizard returns to the `Step 1: Group List` page.
   d. Click `Next`.

5. Add the prerequisite gateway `Application` object to the gateway group:
   a. Select the gateway group you created in Step 4, and click `Next`.

      The `Step 2: Group Resources` page opens.
   b. In the `Assigned` column, select the check box next to the prerequisite gateway `Application` object.

      The `Address-of-record` and `Max.Ports` fields become active.
   c. In the `Address-of-record` field, enter the IP address and port of the SIP Server that receives the requests from the RM. Use the following format:

      `sip:<IP_address>:<port>`
   d. In the `Max.Ports` field, enter the port capacity for SIP Server. The value can be any unsigned integer.

      These parameters will be applied to the prerequisite connection between the Resource Manager and the gateway object.

6. Click `Next`.

7. To confirm and save the changes, click `Finish`.

End of procedure

Additional Info

Table 4 describes in greater detail the parameters as configured in this procedure. To modify these parameters, you can do so from the `Resource`

Management panel, or on the Options tab in the Resource Manager Application object (where you can see the group represented as a section, with the name that you specified in the wizard).

Table 4: Configuring a Gateway Resource for the SIP Server

Section	Option	Value
<gateway_resource_section>	monitor-method	Enter a value of none. Resource Manager will not monitor the health of this resource. It assumes that resources in this group are always alive. See the option description on page 79.
<gateway_resource_section>	port-usage-type	Set the value to in-and-out. The Resource Manager considers SIP dialogs originating from and directed to the gateway resource when calculating usage, for resource management purposes. See the option description on page 80.
<gateway_resource_section>	service-types	Enter the value gateway. See the option description on page 80.

Next Steps

- Next, you will create the DNs that SIP Server uses to identify GVP in the solution. DNs are created on a configured switch—a prerequisite to these procedures. If your solution does not yet include the required SIP Server-related objects, create the following before moving on to the next step:
  - SIP Switching Office
  - SIP Switch
- If your configuration already includes these objects, continue at "Linking SIP Server with GVP".

Procedure:
Linking SIP Server with GVP

Purpose: To create the DNs that SIP Server uses to identify GVP in the solution: a TrunkGroup DN and a Voice over IP Service DN. Both DNs are required to support the different methods that the solution uses to launch VoiceXML applications.

Summary

1. On the SIP Server switch, create a `Trunk Group` DN, pointing the `contact` option to the Resource Manager (Step 3).

   SIP Server uses this `TrunkGroup` DN to access VoiceXML applications that are mapped as IVR Profiles on the Resource Manager. For more information about mapping applications on the Resource Manager, see the *Genesys Voice Platform 8.0 Deployment Guide*.

2. On the SIP Server switch, create a `Voice over IP Service` DN, pointing the `contact` option to the Resource Manager (Step 4).

   SIP Server uses this `Voice over IP Service` DN to process Play Application requests that arrive on SIP Server from a URS routing strategy. The routing strategy can access the VoiceXML application directly by using the URL specified in the Play Application treatment—no mapping on the Resource Manager is required for this method.

Prerequisites

- For the `TrunkGroup` DN, the VoiceXML application must be mapped on the Resource Manager, using the IVR Profile method—which you can configure in Genesys Administrator, under `Provisioning > Voice Platform > IVR Profile`. For detailed procedures, see the *Genesys Voice Platform 8.0 User's Guide*.

Start of procedure

1. In Genesys Administrator, open the `Provisioning` tab.
2. In the navigation tree, select `Switching > Switches`, and then click the SIP Switch.
3. Create a `Trunk Group` DN:
   a. In the navigation tree, click `DNs`, then in the `DNs` window, click the `New` icon, and select `Trunk Group` as the type.
   b. Number the DN according to the phone number that customers use to dial the contact center. For example, if the contact number is (123) 456-7890, then enter `1234567890` as the number for this `Trunk Group` DN.
   c. On the `Options` tab, create a new section called `TServer`, then add a new option called `contact`. For the value of this option, enter the IP address and SIP port for the Resource Manager. For example:

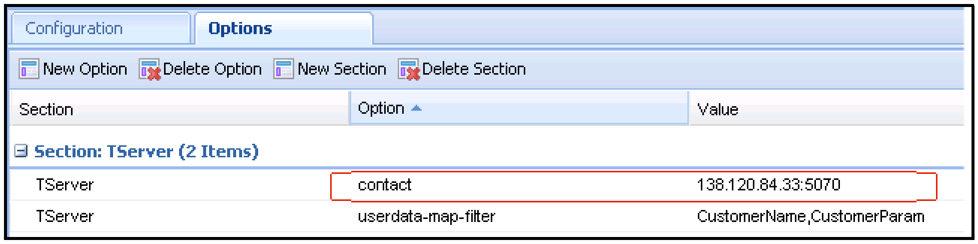


Figure 16: Trunk Group DN—Contact Option

When an incoming call arrives at the SIP Sever, and the user part of the `Request-URI` matches the newly created DN, SIP Server forwards the request to the configured contact—the Resource Manager, as identified by this IP address and port.

d. Click `Save` to save all changes.

e. Repeat Steps a to d for every unique contact number that customers can use to dial into the contact center.

4. Create a `Voice over IP Service` DN:
   a. In the navigation tree, click `DNs`, then in the DNs window, click the `New` icon.
   b. In the `Number` field, enter a name or number, then select `Voice over IP Service` as the type.
   c. On the `Options` tab, create a new section called `TServer`, then add a new option called `contact`. For the value of this option, enter the IP address and SIP port for the Resource Manager.
   d. Create a new option called `service-type` and for the value enter `application`.
   e. Click `Save` to save all changes.

End of procedure

Next Steps

- If your call flow design requires the exchange of customer user data between SIP Server and GVP, you must configure SIP Server so that it maps data in both of the following directions:
  - SIP Server attaches data from GVP SIP messages to the call.
  - SIP Server attaches data from the T-Library message and adds it as headers in the SIP message sent to GVP.

  Continue at "Enabling user data exchange between SIP Server and GVP".
- If your call flow design does not require any user data mapping, continue at "Creating a Routing Point on the SIP Server" on page 57.

Procedure: Enabling user data exchange between SIP Server and GVP

Purpose: To enable the exchange of user-defined customer information between SIP Server and the VoiceXML application.

Data Flows in Two Directions

This data exchange takes place in two directions:

- From SIP Server to GVP—In this case, `UserData` data that is part of a call is mapped to custom headers in the `INVITE` message that SIP Server sends to GVP, making the data available to the VoiceXML application.
- From GVP to Management Framework—In this case, customer info sent in the body of `INFO` and `BYE` requests, or in the headers of `REFER`, or `re-INVITE` messages, is mapped to the T-Library event, making the data available to the URS routing strategy, or any other Genesys application that needs it.
  - For info sent in the body of `BYE` or `INFO` requests, no special configuration is required.
  - For info sent in the headers or `REFER` messages or re-`INVITE` transfers, mapping will occur so long as the prefix in the name of the custom header matches the prefix specified in the `userdata-map-trans-prefix` option.

Mapping Samples

For examples of user data mapping from SIP Server to GVP and GVP to SIP Server, see Appendix A, "Sample User Data Mapping" on page 69.

Prerequisites

For mapping user data from SIP Server to GVP, you need to modify the GVP `Trunk Group` DN and `Voice over IP Service` DN that you created in "Linking SIP Server with GVP" on page 50.

Start of procedure

1. If any user data mapping is required, in the `TServer` section of the SIP Server `Application` object, use the `userdata-map-trans-prefix` option to specify the prefix for the custom headers that SIP Server will map to the T-Library `UserData` attributes. All headers that start with this prefix will be mapped.

   For a more detailed procedure, see "Configuring user data mapping on SIP Server".

2. If user data mapping is required from SIP Server to GVP, in the GVP `TrunkGroup` and `Voice over IP Service` DNs, use the `userdata-map-filter` option to specify which user data key-value pairs will be mapped to the custom headers in the `INVITE` request. Separate the values with a comma. If GVP does not need to receive any user data, then you can leave this option undefined.

   For a more detailed procedure, see "Configuring user data mapping on the GVP DNs".

End of procedure

Next Steps

- If you have completed the mapping procedures required for your CTI operations, continue at "Creating a Routing Point on the SIP Server" on page 57.

Procedure: Configuring user data mapping on SIP Server

Purpose: To configure SIP Server so that it maps custom headers in the `INVITE` request to `UserData` attributes in the T-Library event, according to a defined custom header prefix. Any header that matches this prefix will be mapped.

Mapping takes place as follows:

- For data mapping from SIP Server to GVP, this prefix is added to the user data sent out by SIP Server.
- For data mapping from GVP to SIP Server, user data in the body of `INFO` or `BYE` messages are automatically mapped. For user data configured to pass in the headers of `REFER` messages or `re-INVITE` transfers, any data in headers starting with this prefix are mapped to the T-Library event.

Start of procedure

1. In Genesys Administrator, select `Provisioning > Environment > Applications`, and click the SIP Server `Application` object.
2. On the `Options` tab, in the `TServer` section, select the option `userdata-map-trans-prefix`.
3. For the value, enter the prefix used by the custom headers that carry the user data. Use a single value for this option.

   Figure 17 shows the SIP Server `Options` tab with the `userdata-map-trans-prefix` option set for the `X-Genesys-` prefix. Genesys recommends using this prefix to identify custom headers in the SIP request.

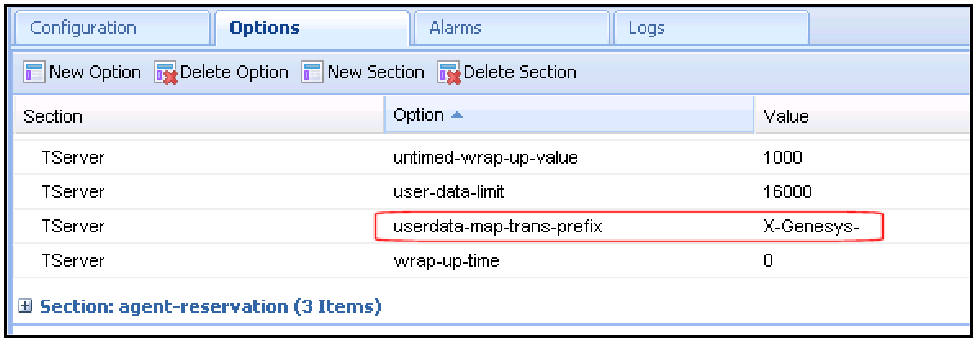


Figure 17: Sample Mapping—Prefix Method

4. Click `Save` to save all changes.

> **Tip:** SIP Server can handle 16K bytes of user data by default. If your routing strategy and VoiceXML application require a larger amount of data, adjust the `user-data-limit` option to meet your needs. This option defaults to a value of `16000`.

End of procedure

Next Steps

- If GVP needs to get user data from SIP Server, you must also define the `userdata-map-filter` option. See "Configuring user data mapping on the GVP DNs".
- If GVP does not need any user data, then continue at "Creating a Routing Point on the SIP Server" on page 57.

Procedure:
Configuring user data mapping on the GVP DNs

Purpose: To enable user data mapping from SIP Server to GVP. Use this procedure to specify only those headers that need to be sent to GVP. If GVP does not need any user data, then skip this procedure.

Prerequisites

- Linking SIP Server with GVP, page 50
- Configuring user data mapping on SIP Server

Start of procedure

1. In Genesys Administrator, select `Provisioning > Switching > Switches`, and from the display of switches, click your SIP Switch configuration object.
2. In the left navigation tree, select `DNs`.
3. Select the `TrunkGroup` DN that you created in Step 3 of “Linking SIP Server with GVP” on page 50.
4. On the `Options` tab, in the `TServer` section, create a new option called `userdata-map-filter`.
5. For the value, enter the prefix for any `UserData` attributes that you want to be mapped from the T-Library message to the `INVITE` request. Separate the values with a comma. Enter `*` to map all user data.

   Tip: All user data filtered from the T-Library message will show up in the custom header of the `INVITE` with an additional custom prefix, as defined in the `userdata-map-trans-prefix` option.

   Figure 18 shows the GVP `TrunkGroup` DN configured with the sample `UserData` attributes `CustomerParam` and `CustomerName`.

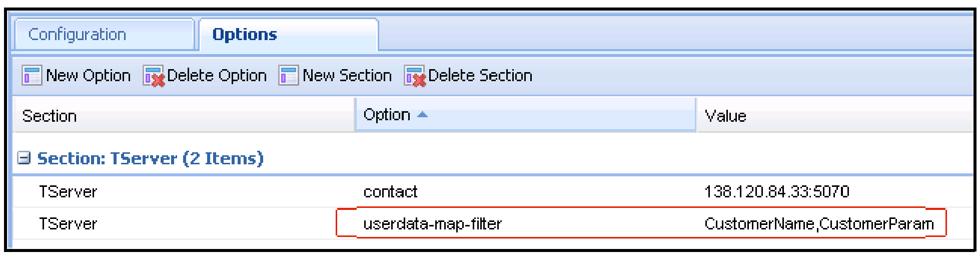


Figure 18: Sample Mapping—Filter Method

   In this example, any user data in the T-Library message starting with these attributes (for example, `CustomerParam1`, `CustomerParam2`, and so on) will be added to the `INVITE` request.

6. Click `Save` to save all changes.
7. Repeat Steps 1 to 6 for the `Voice over IP Service` DN that you created in Step 4 of “Linking SIP Server with GVP” on page 50.

End of procedure

Next Steps

- Creating a Routing Point on the SIP Server

Procedure: Creating a Routing Point on the SIP Server

Purpose: To create the `Routing Point` DN on the SIP Switch that will be used to invoke the routing strategy that you will create later in these procedures.

Calls to SIP Server can arrive on this routing point after coming in from the PSTN, or as a result of a transfer after the initial self-service portion of the call is finished.

Start of procedure

1. In Genesys Administrator, open the `Provisioning` tab.
2. From the left navigation tree, select `Switching > Switches`.
3. From the displayed switches, click your SIP Switch `Application` object.
4. In the left navigation tree, select `DNs`.
5. Click the `Create new object` icon, and select `Routing Point` from the `Type` drop-down list.
6. Assign a name or number to the routing point.
7. Click `Save` to save all changes.

End of procedure

Next Steps

- Creating and loading a routing strategy

Procedure: Creating and loading a routing strategy

Purpose: To create a simple routing strategy that demonstrates the minimum requirements for an integrated solution (suitable for lab or integration testing purposes).

This strategy serves two purposes: to launch a simple VoiceXML application directly from URS, and to route the call to an agent after the treatment is finished. It supports the configuration that is described in "How It Works—The Basic Call Flow" on page 19.

Prerequisites

- A `Routing Point` DN on which to load the strategy. See "Creating a Routing Point on the SIP Server".

- A simple VoiceXML application on GVP to which you can point the routing strategy's Play Application. Creating VoiceXML applications is outside the scope of this guide. For more information, see the following:
  - For information about VoiceXML, see the *Genesys Voice Platform 8.0 VoiceXML 2.1 Help*.
  - For information about creating VoiceXML applications using Genesys Composer Voice, press `F1` from the Composer Voice application to access its help system.
- A SIP agent endpoint on the SIP switch.

Summary

1. In the Interaction Routing Designer (IRD), create the simple routing strategy.
2. Configure the `PlayApplication` block so that the `{s}APP_URI` parameter targets the URI of the prerequisite VoiceXML application.
3. Configure the `Function` block so that the strategy routes the call to an agent on the SIP switch, after the Play Application treatment is finished.
4. Load the strategy on the prerequisite `Routing Point` DN.

Start of procedure

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

   Tip: For more information about using IRD, see the *Universal Routing 7.6 Deployment Guide*. You can also refer to *Interaction Routing Designer Help*, which you can access by pressing `F1` in the application.

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

   The sample strategy consists of the following routing objects: `Entry`, `PlayApplication`, `Function`, and two `Exit` blocks (see Figure 19).

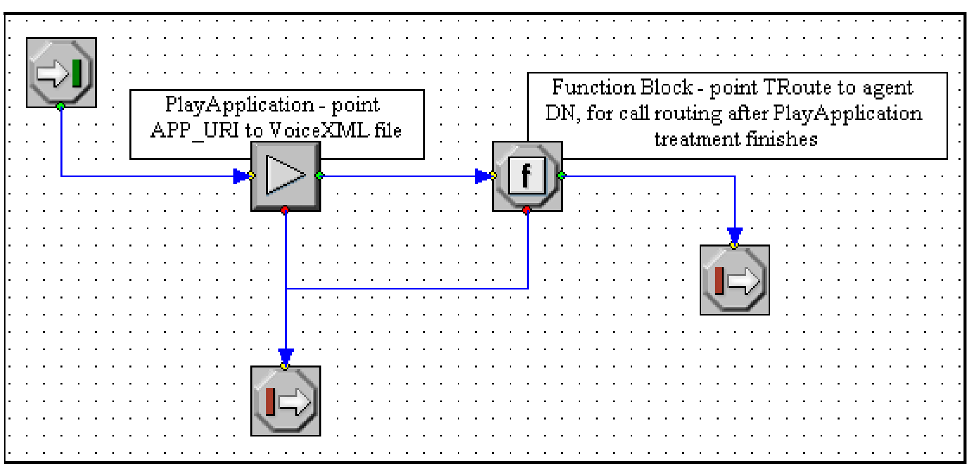


Figure 19: Sample VPS Routing Strategy

3. Configure the PlayApplication block in this strategy, so that the {s}App_URI parameter points to the VoiceXML application:
   a. Double-click the PlayApplication block.
   b. For the APP_ID parameter, enter a value of 1.
   c. For the Language parameter, select English (US).
   d. Click the Add item icon and create a new parameter called {s}APP_URI, with a value that specifies the fully qualified URI of the prerequisite VoiceXML application. Enter the value by using the following format:
      - Add the prefix {s}, so that the parameter is read as a string.
      - Use percent-encoding (%20) for any spaces in the URI path.

   Figure 20 shows a sample Play application properties window, as configured to launch the GVP VoiceXML application.

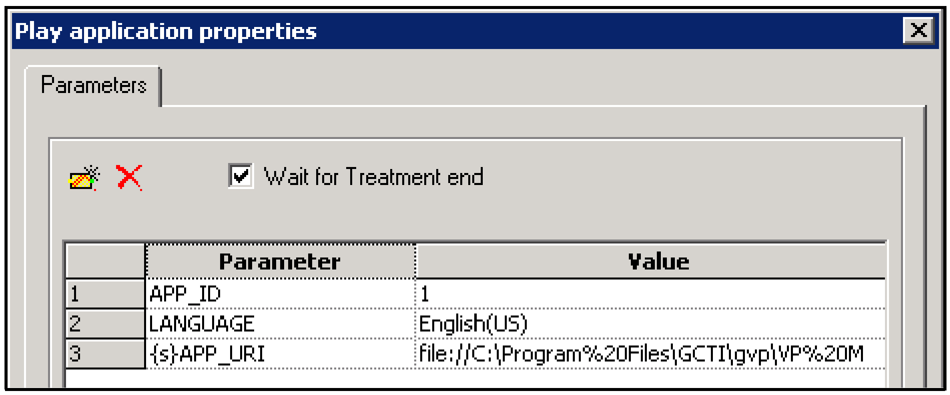


Figure 20: Sample Play Application Properties Window

4. Configure the Function block so that URS will route the call to a specified agent on the SIP Server, after the Play Application treatment is finished:
   a. Double-click the Function block.
   b. Select the TRoute function.
   c. For the Destination parameter, enter the DN number for one of the prerequisite SIP endpoints on the SIP switch.
   d. For the Route Type parameter, select RouteTypeUknown from the value drop-down list.
   e. Click Add, then click OK.

   Figure 21 shows a sample Function properties window, configured for the SIP endpoint DN 9001 on the SIP switch.

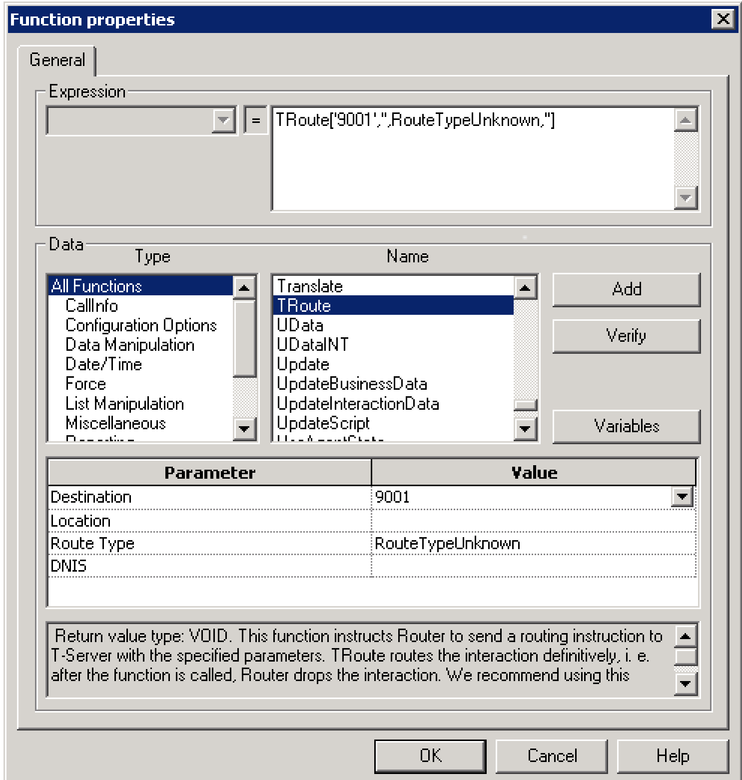


Figure 21: Sample Function Properties Window

5. In the `Monitoring` window of IRD, load the new strategy on the `Routing Point` DN that you created in "Creating a Routing Point on the SIP Server" on page 57:
   a. In the `Shortcut` bar, click the `Loading` icon.
   b. In the `Loading` window, expand the SIP switch.
   c. Right-click the prerequisite Routing Point, and select `Load strategy`.
   d. Select the newly created strategy, and then click `OK`.

End of procedure

Next Steps

- You have completed all of the required steps for the VPS integration.

Procedure: Creating a Trunk DN for gateways that do not support REFER

Purpose: Additional configuration to force the re-INVITE transfer method for media gateways that do not allow REFER transfers.

If your media gateway supports REFER requests, it can respond to a Blind transfer from GVP—forwarded as a REFER request from SIP Server to the gateway. No special configuration is required.

However, if your media gateway does not support REFER requests, it cannot initiate the outbound leg of the call. In this case, you must create a Trunk DN, which represents the gateway when sending the second INVITE request to SIP Server.

Start of procedure

1. In Genesys Administrator, open the Provisioning tab.
2. In the navigation tree, select Switching > Switches, and then click the SIP Switch.
3. In the navigation tree, click DNs, then in the DNs window, click the New icon.
4. Enter a Number and select Trunk as the type.
5. On the Options tab, create a TServer section, and add new options as follows:
   - Set refer-enabled to false.
   - Set prefix to the value of the ANI of the incoming call.
   - Set contact to the IP address and port of the media gateway.

   Tip: For more information about these options as configured in this procedure, see Table 5.

6. Click Save to save all changes.

End of procedure

Additional Info

Table 5 describes in greater detail the options configured in this procedure.

Table 5: Configuring DN Options

Section	Option	Value
TServer	`refer-enabled`	Set `refer-enable` to `false` so that SIP Server will use a re-`INVITE` message instead of a `REFER` message for single-step call transfers. See the option description on page 75.
TServer	`prefix`	Set this option to the value of the ANI of the incoming call. For example, if the caller number is 9051234567, then set this option to `905`. See the option description on page 76.
TServer	`contact`	Enter the IP address and port for the media gateway. Use the following format: `<IP address>:<port>` **Note**: The default port for the media gateway is `5060`. If using the default port, you do not need to include it in this string. See the option description on page 76.

Next Steps

- You have completed all of the required steps to configure the solution for media gateways that do not support `REFER` requests.

Procedure: Configuring the solution for media redirect transfers

Purpose: To configure the MCP so that the solution can support media redirect transfers.

The VPS supports two types of bridged transfers: bridge and media redirect. For bridge transfers, no special configuration is required.

For media redirect transfers, however, you must set the default transfer method for MCP to `MEDIAREDIRECT`, otherwise MCP will try to process the transfer using the bridge method.

Start of procedure

1. In Genesys Administrator, select `Provisioning > Environment > Applications`, and click the GVP Media Control Platform `Application` object.
2. On the `Settings` tab, set the `Default Bridge Transfer` parameter (`sip` section, `defaultbridgexfer` option) to `MEDIAREDIRECT`.

> **Tip:** If you want to switch back to the bridge transfer method, then you must set this `defaultbridgexfer` option back to the default value of `BRIDGE`.

3. Click `Save` to save the changes.

End of procedure

Procedure: Integrating Stream Manager into the solution

Purpose: To perform any configuration steps required to integrate Stream Manager 7.6 with the VPS. Allows SIP Server to control the playing of announcements or music for callers queued on a Routing Point or ACD queue.

If you include Stream Manager in your deployment, the VPS can park calls on Stream Manager when a targeted agent is unavailable, or when an agent places the call on hold. While the caller waits for the agent to become available, Stream Manager plays files using a codec negotiated with the SIP Server switch, as directed by the treatment in the URS strategy loaded on a `Routing Point` DN.

For additional SIP services provided by Stream Manager—for example, conference services, video playback, or dtmf tone generation—see the *Framework 7.6 SIP Server Deployment Guide.*

Prerequisites

- A fully deployed and functioning Voice Platform Solution.
- Stream Manager—Genesys recommends running the Stream Manager Wizard (installed from the CD) from the existing SIP Server Wizard.
- Announcement or music files that SIP Server will access from the Stream Manager installation directory, as directed by the treatment in the URS routing strategy used to initiate the service. These files must be in the appropriate codec format, with the filename suffix corresponding to the codec type.

  For more information about supported codecs and filenames, see the *Framework 7.6 Stream Manager Deployment Guide*.

- A Routing Point DN on which the URS routing strategy that initiates the music or announcement treatment will be loaded. For the VPS, this strategy will typically be designed so that the music or announcement will be played when the targeted agent is unavailable. Refer to the *Universal Routing 7.6 Reference Manual* for more information about the use and configuration of strategies.

Start of procedure

1. Configure the Stream Manager Application object for use with SIP Server.
   a. In Genesys Administrator, open the Provisioning tab.
   b. In the navigation tree, select Environment > Applications, and then click the Stream Manager Application object.
   c. On the Options tab, create a new section called contact, then add a new option called sip-port, with the value set to the SIP messaging port used by the solution—typically 5060.

2. Create a Voice over IP Service DN to specify the connection and options for the desired Stream Manager functionality.
   a. In Genesys Administrator, select the Provisioning tab.
   b. In the navigation tree, select Switching > Switches, and then click the SIP Switch.
   c. In the navigation tree, click DNs, then in the DNs window, click the New icon, and select Voice over IP Service as the type.
   d. On the Options tab, create a new section called TServer, then add a new option called service-type, with the value set to the type of service you want this DN to provide:
      - For music treatments, set this option to music or moh.
      - For announcement treatments, set this option to treatment.

   Tip: Other types of services that Stream Manager can provide SIP Server include: mcu, recorder, conference. For information about configuring DNs for different types of services, see the "SIP Device Configuration" chapter of the *Framework 7.6 Deployment Guide*.

   e. Click Save to save all changes

3. Add prerequisite announcement or music files to the appropriate directory in the Stream Manager installation folder.

   Figure 22 shows a sample treatment audio file—1_gsm.wav, in the format <ID>_<codec_suffix>.wav—placed in the announcement subdirectory of the Stream Manager installation root.

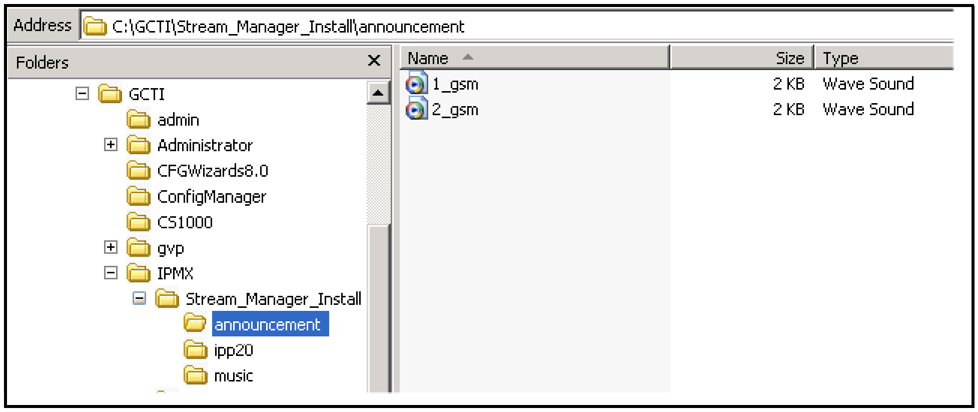


Figure 22: Sample Folder Structure for Audio File in Stream Manager

> **Tip:** If using multiple instances of Stream Manager, make sure the files are replicated to identical folder locations in the installation directory for every Stream Manager instance.

4. In the `Routing Design` window of Interaction Routing Designer (IRD), add an announcement or music treatment to a new or existing routing strategy, to be loaded onto a `Routing Point` DN.

   For example, Figure 23 shows a sample routing strategy that includes a music treatment provided by Stream Manager.

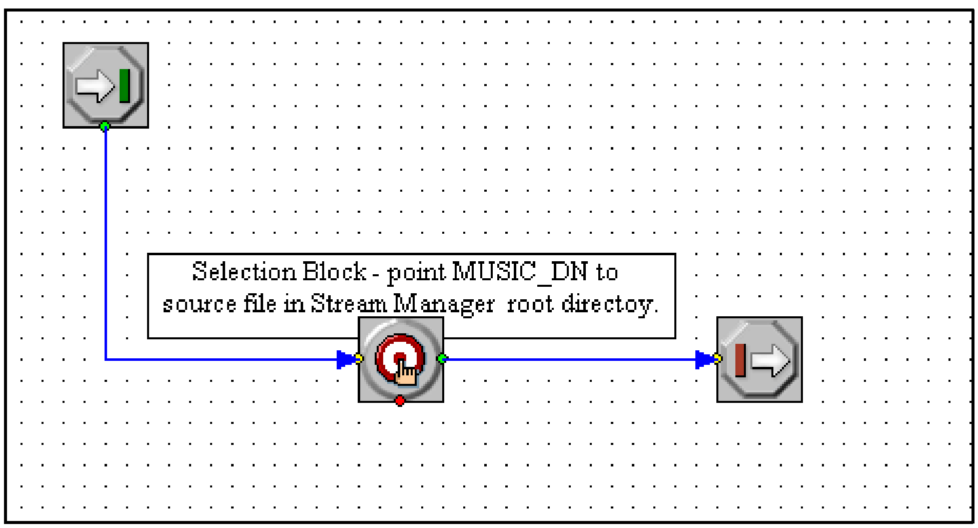


Figure 23: Sample Music Treatment in URS Routing Strategy

In this example, the `Busy` tab in the `Selection` block is configured for a `Music` treatment, using the following parameters:

- `MUISC_DN`—points to a music file in the Stream Manager root directory. This file is played for the caller while they are waiting in the queue for a resource to become available.

  Enter this parameter in the following format:

  `<directory>/<music file name>`

- `Duration`—specifies how long (in seconds) that the music file will play.

Figure 24 shows the `Busy` tab as configured for a music treatment.

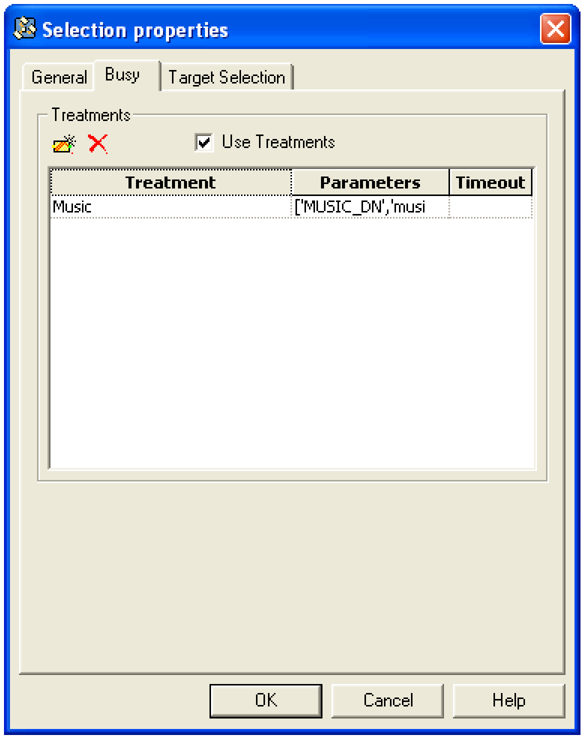


Figure 24: Music Treatment Configured in the Selection Block

> **Tip:** For more information about configuring these music treatment parameters, see Table 6 below.
>
> For information about configuring announcements or other treatments, see the "Music and Announcements" section of the *Framework 7.6 SIP Server Deployment Guide*.

5. In the `Monitoring` window of IRD, load the strategy on the prerequisite `Routing Point` DN—typically this DN is the destination for either the PlayApplication routing strategy or a VoiceXML strategy invoked earlier in the call flow design.
   a. In the `Shortcut` bar, click the `Loading` icon.
   b. In the `Loading` window, expand the SIP switch.
   c. Right-click the Routing Point, and select `Load strategy`.
   d. Select the newly created strategy, and then click `OK`.

End of procedure

Additional Info

Table 6 describes in greater details the music treatment parameters that are configured in this procedure.

Table 6: Music Treatment Parameters

Parameter	Description
`MUSIC_DN`	Specifies the music source that Stream Manager plays. The format is: `<directory>/<music file name>` Where `<directory>` is a sub-directory of the Stream Manager root directory, and <music file name> refers to the name of the file—without the codec extension. For example, music/in_queue refers to the file `music/in_queue_alaw.wav` if the G.711 A-law codec is used. To specify the number of repetitions, the parameter `repeat=<N>` must be used, where `<N>` is any positive integer. If no repetition is specified, the music file is looped forever. The valid formats are: • `<directory>/<music file name>`—The specified file is looped endlessly. • `<directory>/<music file name>;repeat=<N>`—The specified file is repeated `<N>` times. The `default-music` option is used if a value of the `MUSIC_DN` parameter is not specified.
`DURATION`	Specifies the duration of the music (in seconds). **Note**: This parameter is ignored if `MUSIC_DN` is blank. This treatment ends before music is played. To continue playing music after the treatment terminates, consider creating one of the following strategies: • Execute the treatment inside a route-selection treatment block. In this case, the treatment continues until a route target is selected. • Follow the treatment with the `SuspendForTreatmentEnd` function. In this case, the treatment plays music until terminated after the delay specified in option `DURATION`. • Follow the treatment with the delay function. In this case, the treatment plays music for the period specified in option delay. If `DURATION` is less than delay, silence is played for the time difference.

Next Steps

- You have completed the basic steps to integrate Stream Manger into the VPS.

Appendix

A Sample User Data Mapping

Depending on the needs of your call flow design, user data needs to flow from SIP Server to GVP or from GVP to SIP Server. This appendix includes sample data exchanges between SIP Server and the T-Library event, in both directions, as well as VoiceXML code samples used to map user data.

All examples show data mapping for user data that is defined as `CustomerName` and `CustomerParam` with a prefix of `X-Genesys`.

This appendix includes the following examples of mapped user data:

Mapping User Data Received from GVP

The following samples show the mapping of user data from the `INVITE` request to the T-Library event. User data appears in **bold**.

Sample SIP INVITE Request

```
INVITE sip:5555@138.120.84.32:5060 SIP/2.0
Via: SIP/2.0/UDP
138.120.84.33:5070;branch=z9hG4bK0167dea01f24e6abcdef09
Via: SIP/2.0/UDP 138.120.84.239:5060;branch=z9hG4bK0aef28e81f24e6
From: sip:9059683348@10.0.0.193;tag=B03F2519-A7ED-4FD0-E5A8-
525C101B5725
To: <sip:5555@138.120.84.33:5070>
Max-Forwards: 69
CSeq: 1 INVITE
Call-ID: 59774435-B48E-45A4-C885-D61B28BC3828-5060@138.120.84.239
Contact: <sip:RM_GVP@138.120.84.239:5060>
Content-Length: 291
Content-Type: application/sdp
```

```
Record-Route: <sip:22687272@138.120.84.33:5070;lr;gvp.rm.datanodes=1>
X-Genesys-CustomerName: John Doe
X-Genesys-CustomerParam: password=1234&amp;zipcode=90210
Min-SE: 90
X-Genesys-CallUUID: DHU3UHQ5TT5DFBQ7OP3BP9P7NC000001
X-Genesys-GVP-Session-ID: 4C9B7DD1-C405-428F-409A-
4C14BA62DE06;gvp.rm.datanodes=1;gvp.rm.tenant-id=GVADS_App_vmdit5
Supported: timer
Session-Expires: 1800
X-Genesys-RM-Application-dbid: 105
```

Sample T-Library Event

```
   AttributeANI     '9059683348'
   AttributeDNIS    '5555'
   AttributeUserData[256] 00 05 00 00..
         'CustomerName'   'John Doe'
         'CustomerParam'  'password=1234&amp;zipcode=90210'
         'CallUUID' 'DHU3UHQ5TT5DFBQ7OP3BP9P7NC000001'
         'GVP-Session-ID' '4C9B7DD1-C405-428F-409A-4C14BA62DE06;
gvp.rm.datanodes=1;gvp.rm.tenant-id=GVADS_App_vmdit5'
         'RM-Application-dbid''105'
   AttributeCallUUID'DHU3UHQ5TT5DFBQ7OP3BP9P7NC000006'
   AttributeConnID008b01918c3dd002
```

Sample VoiceXML Code

The following code sample shows a bridged `<transfer>` in a VoiceXML application, with attached user data:

```
...
<form>
     <script>
     var userdata;
     userdata.CustomerName = "John Doe";
     userdata.CustomerParam = "password=1234&;amp;zipcodde=90210";
     </script>
     <transfer bridge="true" dest="sip:5555" signalvar="userdata"/>
</form>
...
```

Mapping User Data Received from URS

The following samples show the mapping of user data from the T-Library event to the `INVITE` request that is sent to GVP. User data appears in **bold**.

Sample T-Library Event

```
AttributeANI     '9059683348'
   AttributeDNIS    '5555'
   AttributeUserData[704] 00 18 00 00..
      'CustomerName'      'John Doe'
      'CustomerParam'     'password=1234&amp;zipcode=90210'
      'CallUUID'    'DHU3UHQ5TT5DFBQ7OP3BP9P7NC00000B'
      'GVP-Session-ID' 'FCBCBF00-4FBD-418B-6DB3-0B3FE4861960;
gvp.rm.datanodes=1;gvp.rm.tenant-id=GVADS_App_vmdit5'
      'RM-Application-dbid'  '105'
      'RVQID'        ' '
      'RTargetTypeSelected'  '100'
      'RTargetRuleSelected'  ' '
      'RTargetObjectSelected'' '
      'RTargetObjSelDBID'    ' '
      'RTargetAgentSelected' ' '
      'RTargetPlaceSelected' ' '
      'RTenant'          'Environment'
      'RStrategyName'     'Route2DN'
      'RStrategyDBID'     '104'
      'CBR-actual_volume'    ' '
      'CBR-Interaction_cost' ' '
      'CBR-contract_DBIDs'   ' '
      'CBR-IT-path_DBIDs'    ' '
      'RRequestedSkillCombination'' '
      'RRequestedSkills'(list)
      'CustomerSegment'   'default'
      'ServiceType'       'default'
      'ServiceObjective'     ' '
   AttributeCallUUID'DHU3UHQ5TT5DFBQ7OP3BP9P7NC00000G'
   AttributeConnID  008b01918c3dd004
   AttributeCallID  4
   AttributeCallType2
```

Sample INVITE Request (to GVP)

```
INVITE sip:1800@138.120.84.33:5070 SIP/2.0
From: sip:9059683348@10.0.0.193;tag=36A7A329-0740-4AD8-87A1-
AC6AB366EF0B-11
To: <sip:5555@138.120.84.32:5060>
Call-ID: 8B5B60DB-901C-4F20-9AB3-8577E4698254-5@138.120.84.32
CSeq: 1 INVITE
Content-Length: 292
Content-Type: application/sdp
Via: SIP/2.0/UDP 138.120.84.32:5060;branch=z9hG4bKCBA0C951-5D81-
4EB9-904F-99E50A51A323-10
Contact: <sip:1800@138.120.84.32:5060>
Max-Forwards: 70
Allow: INVITE, ACK, PRACK, CANCEL, BYE, REFER, INFO
X-Genesys-CustomerName: John Doe
```

```
X-Genesys-CustomerParam: password=1234&amp;zipcode=90210
X-Genesys-CallUUID: DHU3UHQ5TT5DFBQ70P3BP9P7NC00000G
Session-Expires: 1800;refresher=uac
Min-SE: 90
```

Sample VoiceXML Session Variables

The VoiceXML application receives user data in the following `session.com.genesys.userdata` session variables:

```
session.com.genesys.userdata.CustomerName = 'John Doe';
session.com.genesys.userdata.CustomerParam ='password=1234&amp;zipcode=90210';
```

Mapping User Data Received from GVP in INFO/BYE Body

Mapping for `INFO` and `BYE` requests does not require any special configuration, but takes place automatically in the body of the SIP message. The following samples show the mapping of user data from a `BYE` request to the T-Library event. User data appears in **bold**.

Sample BYE Request

```
BYE sip:PlayApp@138.120.84.32:5060 SIP/2.0
Via: SIP/2.0/UDP
138.120.84.33:5070;branch=z9hG4bK02ba3488d74d19abcdef09
Via: SIP/2.0/UDP 138.120.84.33:5060;branch=z9hG4bK0a9d6548d74d18
From: <sip:PlayApp@138.120.84.32:5060>;tag=E218369F-7B05-4EE7-518A-
D73F8D84417E
To: sip:9059683348@10.0.0.193;tag=2E3CBB9D-3D21-4C6C-ADAE-
239E06E083EA-2
Max-Forwards: 69
   CSeq: 1 BYE
   Call-ID: FBCC203E-7D2E-4E22-9460-94E3625B7379-1@138.120.84.32
   Content-Length: 74
   Content-Type: application/x-www-form-urlencoded;charset=utf-8
   X-Genesys-GVP-Session-ID: 701F7A30-4AE0-458C-75A3-2887610F96FE;
gvp.rm.datanodes=1;gvp.rm.tenant-id=IVRAppDefault
   Min-SE: 90
   Supported: timer

CustomerName=Jane%20Doe&CustomerParam=password%3D1234&__reason=disc
onnect
```

Sample T-Library Event

```
AttributeANI       '9059683348'
   AttributeDNIS    '8000'
   AttributeUserData[65] 00 03 00 00..
      'CustomerName'     'Jane Doe'
      'CustomerParam'    'password=1234'
      '__reason'      'disconnect'
   AttributeCallUUID'F20VBH6HQ54VPAU31P6FT2E79C000001'
   AttributeConnID 006d018d0ec19001
```

Sample VoiceXML Code—Mapping to BYE Body

The following code sample shows the mapping of user data received from GVP to the body of a `BYE` message:

```
...
<form>
...
   <var name="CustomerName" expr="Jane Doe"/>
   <var name="CustomerParam" expr="password=1234"/>
   <exit namelist="CustomerName CustomerParam"/>
</form>
...
```

Sample VoiceXML Code—Mapping to INFO Body

You can design the VoiceXML application to attach user data to SIP Server in the middle of the call. The following VoiceXML code sample shows the mapping of user data received from GVP to the body of an `INFO` message:

```
...
<form>
...
   <var name="CustomerName" expr="Jane Doe"/>
   <var name="CustomerParam" expr="password=1234"/>
   <gvp:send namelist="CustomerName CustomerParam"/>
</form>
...
```

Note: In this case, since this is not a disconnect request but mid-call request, `"_reason=disconnect"` will not appear as user data in either the SIP message or the T-Library event, as it does in the preceding "Sample BYE Request" and "Sample T-Library Event".

Appendix

B Configuration Options

This appendix describes all of the configuration options that are modified during any of the procedures included in this guide. Options are organized according to component type, and include the following:

SIP Server Options

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 instead uses the re-`INVITE` method. 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 77).

prefix

Default Value: None
Valid Values: Any string
Changes Take Effect: Immediately

In MCU configuration: Specifies the starting digits of the number that are used when sending calls to MCU. The full number is built as:

`<prefix><connid>@<ipaddr>:port`.

Typically, MCU servers require a prefix consisting of digits in order to identify a type of conference (for example, voice only, voice and video, and so on). Set the value as `conf=` if Stream Manager is used as the MCU.

In gateway configuration: Contains the initial digits of the number that must match a particular gateway for that gateway to be selected. If multiple gateways match a number, the gateway with the longest prefix is selected.

contact

Default Value: None
Valid Values: Any alphanumerical string
Changes Take Effect: Immediately

Contains the contact URI. This field specifies the device's IP address, if this address is fixed. This option is necessary only for stand-alone configuration, and only if the configured device does not register itself in the SIP Server registrar. It is part of the persistent registrar feature.

For example, if the SIP device sends a `REGISTER` request to a SIP Server and this request is accepted, SIP Server uses the contact information from the `REGISTER` request and updates (or creates) in Configuration Manager the option `contact` in the `TServer` section of the `Annex` tab of the corresponding `DN` object.

The URI format is:

`[sip:][number@]hostport[;transport={tcp/udp}]`

Where:

- `sip:` is an optional prefix.
- `number` is the DN number. This is currently ignored.
- `hostport` is a `host:port` pair, where `host` is either a dotted IP address or a DNS-resolvable `hostname` for the endpoint.
- `transport=tcp` or `transport=udp` is used to select the network transport. The default value is `udp`.

default-music

Default Value: `music/on_hold`
Valid Value: Name and path of any valid audio file
Changes Take Effect: Immediately for all new calls

Specifies the name of the file that is played for the music treatment, if none is specified in `TApplyTreatment`, or if the specified file is missing.

userdata-map-trans-prefix

Default Value: None
Valid Values: Any string
Changes Take Effect: Immediately

Contains a transport prefix to indicate what headers in the SIP message carry the mapped `UserData`. SIP Server adds this prefix to all data mapped to the outgoing `INVITE` message. SIP Server scans incoming `INVITE` or `REFER` messages used to place a call on the Routing Point for headers that start with this prefix, in addition to performing the normal mapping procedure.

If this option is not specified, no prefix is added to the transmitted data.

userdata-map-filter

Default Value: None
Valid Values:

*:	All data is mapped.
A list of prefixes	A comma-separated list of prefixes used to identify the UserData key-value pair to be mapped.

Changes Take Effect: Immediately

Specifies the names of the key-value pairs to be mapped. If this option is not specified, no data will be mapped.

Media Control Platform Options

All of these options are all found in the `SIP` section on the MCP `Options` tab.

defaultbridgexfer

Default Value: `BRIDGE`
Valid Values: `BRIDGE`, `MEDIAREDIRECT`
Changes Take Effect: At restart

Specifies the default transfer method for SIP, for bridge-type transfers. For more information about the transfer types and methods, see the section “Transfers” in the *Genesys Voice Platform 8.0 User’s Guide*.

outcalluseoriggw

Default Value: `1`
Valid Values: `0`, `1`
Changes Take Effect: Immediately

Specifies how the Media Control Platform will determine which gateway to use for an outbound call or transfer, if the destination address does not contain a host name or IP address.

Example:

If `sip.outcalluseoriggw=1` and the inbound call came from a gateway with host name `3000`, the call will be placed to one of the following:

- `tel://3000`
- `sip:3000@`—The ampersand character (`@`) is required to delimit the user part from the host part of the address.

routeset

Default Value: Empty

Valid Values:

```
<sip:<Resource Manager IP address>:<Resource Manager SIP
port>;lr>[,<sip:<Next SIP Proxy or UA IP address>:<Proxy SIP
port>;[lr]>,...]
```

Note: The outer angle brackets are required characters in the string.

Changes Take Effect: Immediately

A comma-separated list of SIP Proxy addresses that defines a route set for non-secure SIP outbound calls. If defined, this route set is inserted as the ROUTE header for all outgoing calls. This forces GVP to use the defined route set for SIP messages.

Using the lr parameter with the URI (see syntax) forces the User Agent Client (UAC) to place the remote target URI into the Request-URI and to include the route set in the ROUTE header.

Example:

```
<sip:RM_host.yourdomain.com:5060;lr>,
<sip:Proxy2.yourdomain.com:5060;lr>.
```

Media Control Platform will send the outgoing request to Resource Manager, which will, in turn, route the request to Proxy 2, which will redirect the message to its intended destination.

Note: The route set does not apply to SIP `REGISTER` messages.

transport.<x>

Default Values:

- `transport.0=transport0 udp:any:5060`
- `transport.1=transport1 tcp:any:5060`
- `transport.2=transport2 tls:any:5061 cert=$InstallationRoot$\config\x509_certificate.pem key=$InstallationRoot$\config\x509_private_key.pem`

Valid Values: `<transport_name> <transport_type>:<ip>:<port> [<parameters>]`

where:

- `<transport_name>` is any alphanumeric string.
- `<transport_type>` is the transport layer protocol: udp|tcp|tls.
- `<ip>` is the IP address of the network interface that accepts incoming SIP messages (the default value of any means all network interfaces).
- `<port>` is the port number where SIP stack accepts incoming SIP messages.
- `[<parameters>]` are any additional, optional SIP transport parameters.

Changes Take Effect: Immediately

The parameters that define the transport layer for SIP stack and the network interfaces that are used to process SIP requests.

<x> is the transport interface index that identifies the transport, so that you can specify different combinations of parameters for different protocols.

For a secure SIP connection, ensure that you specify Transport Layer Security (TLS) parameters:

- cert=<TLS certificate path and file name> (required)
- key=<TLS key path and file name> (required)
- type=<type of secure transport> (optional)
  Valid values: `TLSv1|SSLv2|SSLv3|SSLv23`
  Default value: `SSLv23`
- password=<password associated with the certificate and key pair> (required only if the key file is password protected)

Note: The default transport is the smallest non-empty transport interface index. If all `sip.transport.<x>` values are empty, UDP, TCP, and TLS transports are all enabled, with the default parameter values, and UDP is the default transport.

Resource Manager Options

These options are all found in the `rm` section on the Resource Manager `Options` tab.

monitor-method

Default Value: Empty
Valid Values: `Option, mf, none`
Changes Take Effect: After restart

The method the Resource Manager will use to determine if the physical resources belonging to the logical resource group are alive and healthy.

Available methods are as follows:

- `option`—Resource Manager will use SIP OPTIONS messages.
- `mf`—Resource Manager will use SNMP alarms.
- `none`—Resource Manager will not monitor resource health.

port-usage-type

Default Value: Empty
Valid Values: `in-and-out`, `outbound`
Changes Take Effect: After restart

Determines which SIP dialogs the Resource Manager will consider when calculating the current usage on each resource, for resource management purposes.

Current usage is defined as the outstanding number of established SIP dialogs on a resource plus the current pending requests on the resource. The SIP dialogs that are included in the calculation are:

- `in-and-out`—SIP dialogs originated from and directed to the resource.
- `outbound`—SIP dialogs directed to the resource.

service-types

Default Value: None
Valid Values: `voicexml, ccxml, gateway, conference, external-sip`
Changes Take Effect: After restart

The types of service that are provided by resources in this logical resource group.

To specify multiple types of service (for example, CCXML and Conference), hold down the Ctrl key while selecting additional service types. Resources can be assigned to the group only if they support all the service types that you specify in this parameter.

A more detailed description of the valid values is as follows:

- `voicexml`—Voice application services provided by Media Control Platform resources.
- `ccxml`—Call control application services provided by Call Control
- Platform resources.
- `gateway`—Network gateway services provided by Resource Access Point resources.
- `conference`—Conference services, which can be provided by Media Control Platform and Call Control Platform resources.
- `external`-sip—SIP services provided by an external SIP proxy.

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