Genesys

Voice Platform Solution 8.1

Integration Guide

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Preface

Welcome to the *Voice Platform Solution 8.1 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 8.1 SIP Server Deployment Guide.*

This document applies to the following solution component versions:

- Genesys Media Server 8.1.6
- SIP Server 8.1.0
- Management Framework 8.1.2
- Genesys Administrator / Genesys Administrator Extension 8.1.3
- Composer 8.1.1
- IVR Server 8.1.0
- Genesys Security Pack 8.1.1
- Genesys Voice Platform (GVP) 8.1.7

Note: For versions of this document created for other releases of this product, 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 contains the following sections:

For information about related resources and about the conventions that are used in this document, see the supplementary material starting on page 233.

About Voice Platform Solution 8.1

The Voice Platform Solution 8.1 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) 8.1, SIP Server 8.1, and Management Framework 8.1—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.

Intended Audience

This document is primarily intended for system administrators and system integrators. It has been written with the assumption 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 8.1 and Genesys Voice Platform 8.1.

- Have already installed and are familiar with SIP Server and related components, as well as GVP and its related components.

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Document Change History

This is the first release of the *Voice Platform Solution 8.1 Integration Guide* that contains this section. This section lists content that is new or that has changed significantly since the first release of this document. The most recent changes appear first.

New in This Document

8.1.7 Chapter 13, "Integrating with SSG," on page 195:

- Added a warning about Legacy GVPi applications that use attached data to the section "Task Summary: SSG Integration, Routing Point Call Flow" on page 201.

Appendix B, "Configuration Options," on page 225:

- Added a note to the option Appendix B, "RPDN," on page 232.

8.1.6 Chapter 1, "About the Voice Platform Solution," on page 17:

- Modified a Note about REFER limitations on page 20.

Chapter 4, “Supported Call Flow Scenarios,” on page 47:

- Modified a paragraph about REFER limitations on page 55.

Chapter 6, “High Availability,” on page 77:

- Added to the section “HA In Scaled Deployment” on page 81:
  - “Solution Level Components and Interfaces” on page 78
  - “Configuration, Reporting and Development Tools” on page 78

Chapter 13, “Integrating with SSG,” on page 195:

- Added a note about Paraxip media gateway testing to Table Task Summary:, “Integrating the SSG for Outbound Calls,” on page 195.
- Added a note about Paraxip media gateway testing to Table 9, “SIP Server Options—TServer Section,” on page 206.
- Added a note about Paraxip media gateway testing to page 209.

Appendix B, “Configuration Options,” on page 225

- Removed section describing SIP Server configuration options. These options are described in SIP Server documentation.

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Part

1 Solution Overview

For an overview of the Voice Platform Solution and its various components, as well as the kinds of architecture configurations and call flow scenarios that the solution supports, see the following chapters:

Chapter

1 About the Voice Platform Solution

This chapter provides an overview of the Voice Platform Solution (VPS) 8.1 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 8.1 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) 8.1, SIP Server 8.1, and Management Framework 8.1—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.

Functional Overview

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

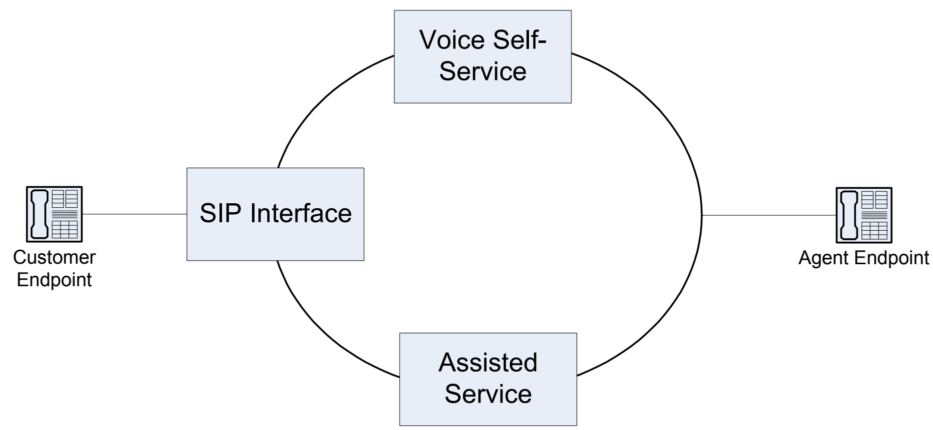


Figure 1: General Functioning of the VPS

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.
- **Outbound Calls**—GVP can be used as the media server to provide services such as call progress detection and media bridging to outbound VoiceXML applications. The Supplementary Services Gateway (SSG) provides the HTTP interface used to make the outbound call to the customer, and connect the customer to the VoiceXML application.

Features and Benefits

The following lists include the new features for the latest release, as well as the main features and benefits of the solution.

What's New In VPS 8.1.2

The following list includes the specific high-level features and benefits that are new to VPS in release 8.1.2.

- PSTN Connector—For VPS integration with traditional telephony environments using Dialog hardware and software. For a sample deployment, see "PSTN Connector" on page 101.
- Hierarchical Multi-Tenancy (HMT)—Supports HMT configurations for service providers, enabling them to apportion a select number of inbound ports for each customer, which provides greater flexibility when enforcing policies during service selection. See "Hierarchical Multi-Tenant Environments" on page 63.
- Genesys Media Server—Supports the same set of features that were previously provided by Genesys Stream Manager (7.x), along with several new codec formats for voice delivery associated with outbound calling, call parking, call recording, conferencing, and IVR prompting.

  Note: For more information, consult the *Genesys Media Server 8.1 Deployment Guide*.

- Resource Manager in Active Cluster Configuration—Support for high availability (HA) deployments of Resource Manager instances as a pair of active servers, for use in HA networks with certain restrictions. See "High Availability" on page 77.

Main Features and Benefits

This following list includes the specific high-level features and benefits offered by an integrated Voice Platform Solution 8.1.

- Flexible CTI through SIP Server integration—For CTI through SIP Server, a single set of integration procedures supports a variety of inbound call flow scenarios.
- Flexible CTI through IVR Server integration—For VPS integration with IVR Server, an additional set of integration procedures supports inbound call flows for IVR-centric voice applications. The CTI Connector provides the integration point between the solution and the IVR Server.
- Two methods for launching VoiceXML applications:
  - Play Application treatments in the routing strategy (CTI through SIP Server only).

  - IVR Profile mapping on the Resource Manager
- Outbound calling through the Supplementary Services Gateway (SSG)—Independent from the inbound call flow configuration, the VPS with SSG provides an HTTP interface for initiating applications with functions outside of what is available in VoiceXML and CCXML applications. For example, placing a batch of outbound calls using an IVR Profile.
- 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
  - Media redirect transfers
  - Consultation transfers using the SIP `REFER with replaces` method

Notes: Consultation transfers using the SIP `REFER with replaces` method are not completely supported.

REFER is not supported if the inbound leg is originally pinned on a route point (e.g., GVP is configured as a VOIP service DN).

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 125.

- In CTI through IVR Server configurations, CTI Connector with IVR Server provides support for the following:
  - Legacy 7.6 GVP voice applications (also called GVPi applications)
  - Next Generation Interpreter (NGI) applications with proprietary VoiceXML `<send>` and `<receive>` tags for direct CTI functionality from the voice application.
- 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 Chapter 6, "High Availability," on page 77.

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 8.1.

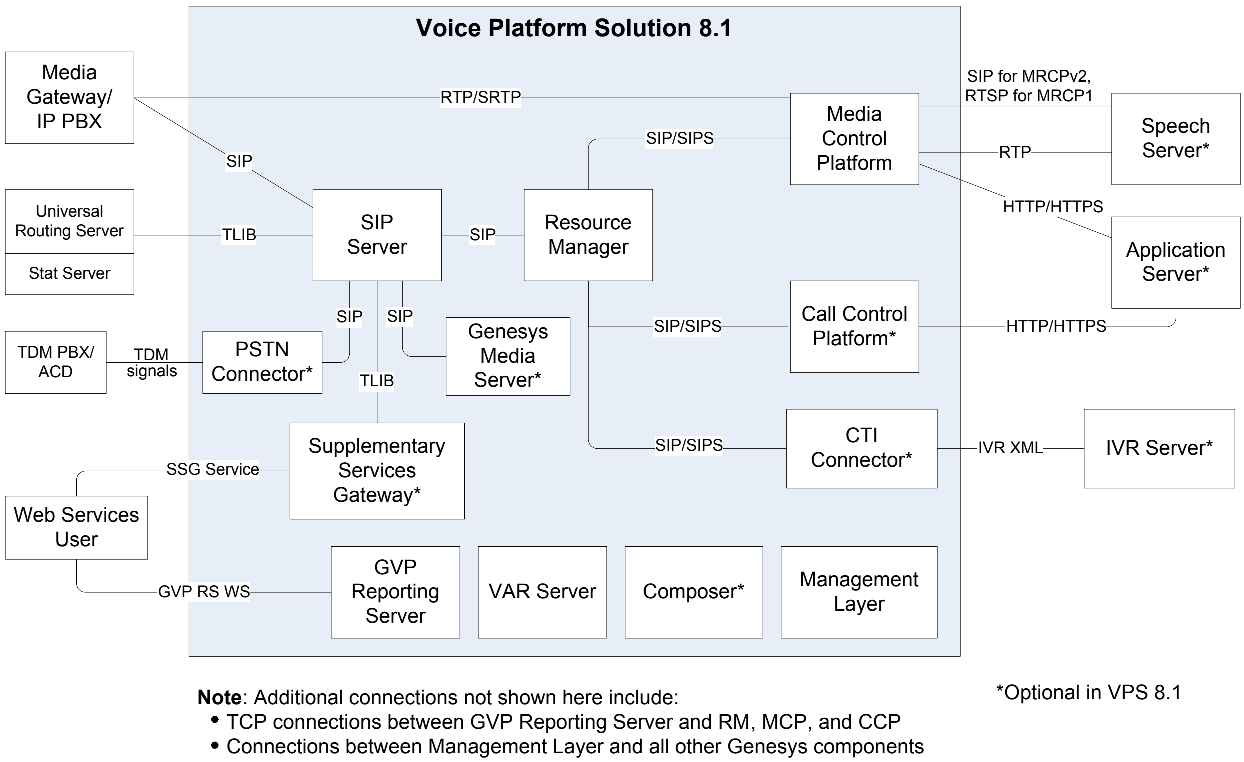


Figure 2: Component Architecture

As shown in Figure 2, the Voice Platform Solution 8.1 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. It includes a legacy VoiceXML interpreter (GVPi) and a Next Generation Interpreter (NGI). MCP can also be used as the Genesys Media Server, replacing Stream Manager to support media streaming through the media server markup language (MSML) interface. For more information, see the *Genesys Media Server 8.1 Deployment Guide*.
- **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.
- **Supplementary Services Gateway (SSG)**—The SSG provides an HTTP interface for initiating outbound calls to customers, and connecting the customer to an outbound VoiceXML application.
- **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**—Composer is an application development tool that developers can use to author the VoiceXML applications (design and edit) or edit the CCXML applications (edit only, design for CCXML applications is unavailable) used by the solution. Although not mandatory to the deployment, it is recommended.
- **CTI Connector**—The CTI Connector provides VPS integration with IVR Server in order to support IVR-centric voice applications—including legacy GVPi applications and NGI applications with CTI extensions—as well as certain switch configurations that require a CTI through IVR Server architecture.
- **PSTN Connector**—The PSTN Connector enables the VPS to integrate with Dialogic hardware and software for legacy TDM environments.
- **Genesys Media Server—**The Genesys Media Server, provided through GVP, is used to stream media files in order to provide announcements and music to callers queued on Routing Points or ACD queues. This is an optional component, required for SIP Server control over the playing of announcements or music. For details, see the *Genesys Media Server 8.1 Deployment Guide*.
- **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 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 Genesys Administrator. For more information, see “About Genesys Administrator” on page 29.

Note: The Voice Platform Solution 8.1 does not require either the Speech Server or the Application Server in the deployment. The MCP and CCP can execute simple applications that use locally stored pre-recorded audio files, instead of the third-party speech or application servers.

How It Works—The Basic Inbound Call Flow

In a typical pure-IP deployment, contact center agents are registered on the 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. Voice self-service is provided to the customer through one of three kinds of VoiceXML applications:

- Standard VoiceXML applications—These applications provide voice self-service using standard VoiceXML tags. Transfers to the assisted-service part of the call can be initiated from the application, using the `<transfer>` tag. Both GVPi and NGI applications are supported.
- URS-centric applications—In these applications, an inbound call arrives at a routing strategy first, where the strategy initiates the VoiceXML application. After the voice interaction, call control returns to the routing strategy for delivery to an agent or other DN. Only NGI applications are supported.
- IVR-centric applications—In these applications, the application itself controls CTI actions through VPS integration with the IVR Server. These mid-call CTI actions can include routing, getting statistics, or attaching data. Both GVPi and NGI applications are supported.

Call Delivery to GVP

Depending on how your deployment selects the voice application, the caller is connected to the voice self-service application in different ways:

- GVP configured as a `Trunk` DN—The call arrives on the `Trunk` DN, where GVP is immediately given call control in order to start a standard VoiceXML application.
- GVP configured as a `Voice Over IP Service` DN—The call arrives at a `Routing Point` DN, where a routing strategy uses a Play Application treatment to launch a URS-centric voice application, in CTI through SIP Server configurations.
- GVP is configured as a series of `Voice Treatment Port` DNs—The call arrives at a `Routing Point` DN, where a strategy selects a `Voice Treatment Port` DN that the solution uses to launch an IVR-centric voice application. This option is only available in CTI through IVR Server configurations.
- GVP is configured as a `Trunk` DN with port number—In station-side connected configurations, the call arrives on the `Trunk` DN with the port number from the PBX that GVP requires to perform DNIS lookup, in CTI through IVR Server configurations.

Call Flow for Standard VoiceXML Applications

For Standard VoiceXML applications, the call reaches GVP through a SIP Server `Trunk` DN, and the MCP launches a VoiceXML application as mapped on the Resource Manager through IVR Profiles. Several 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.

Figure 3 shows the basic call flow for a standard VoiceXML application.

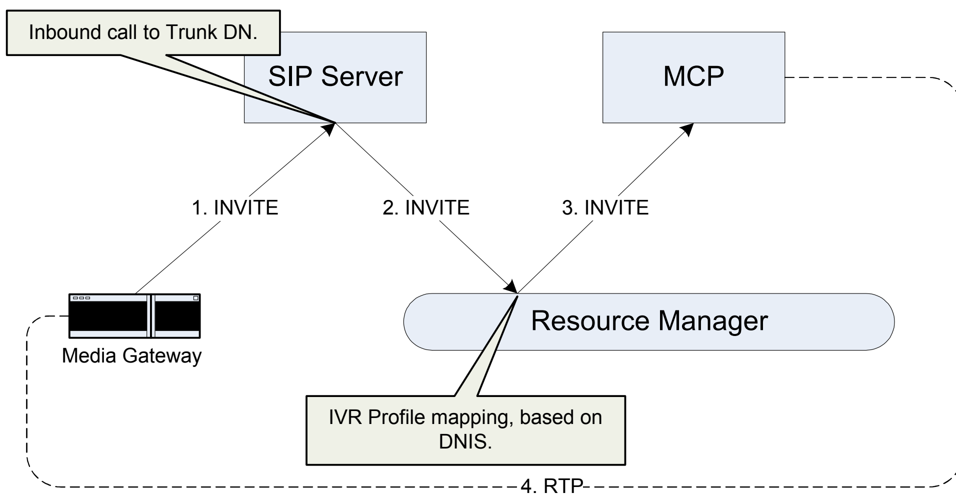


Figure 3: Basic Standard VoiceXML Application Call Flow

Call Flow for URS-centric Applications

In CTI through SIP Server configurations, you can design URS-centric voice applications, where the call reaches the routing point first, and the URS (according to the routing strategy) can initiate a simple 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.

In this case, the VoiceXML application does not initiate call transfers. After the voice self-service interaction is completed, control of the call returns to the URS, where the routing strategy can then determine where to route the call.

Figure 4 on page 25 shows the basic call flow for a URS-centric application.

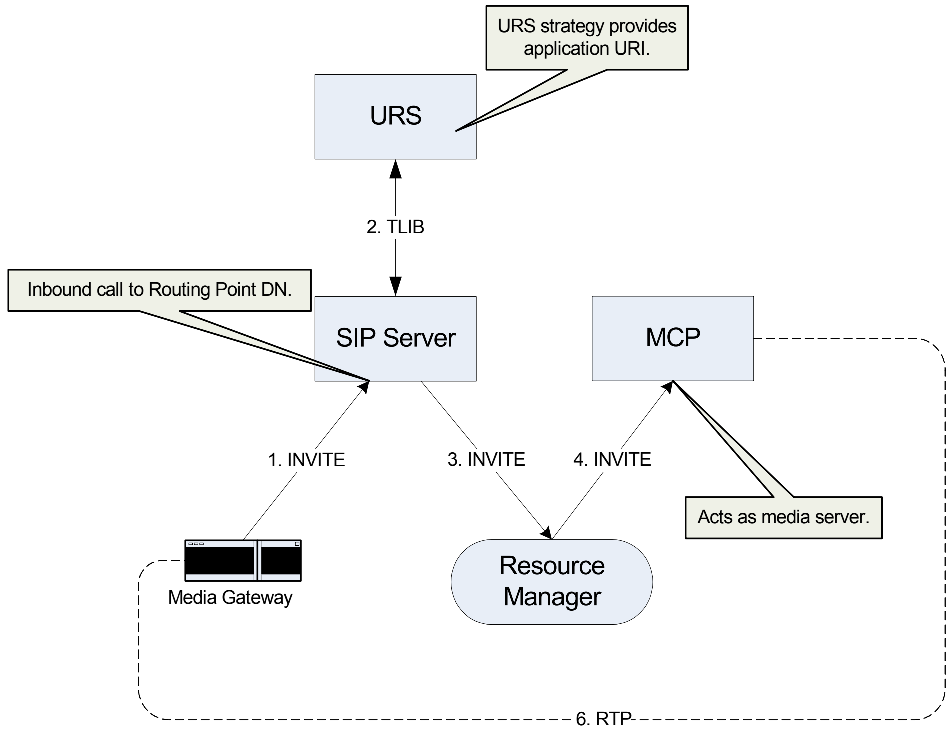


Figure 4: Basic URS-centric Application Call Flow

Note: This method for initiating applications is only available for CTI through SIP Server configurations. This method is not available for VPS integrations with CTI Connector and IVR Server.

Call Flow for IVR-centric Applications

In CTI through IVR Server configurations, the incoming call arrives at a `Routing Point` DN, where the strategy selects an IVR port configured as a `Voice Treatment Port` DN. SIP Server sends this port number to the Resource Manager, so that it can then send this info to CTI Connector and IVR Server to obtain call information (for example, ANI, DNIS, and UUID). After the CTI Connector receives the call details, it sends an `INVITE` to the Resource Manager with the DNIS required to map the IVR Profile and launch the voice application.

An exception is in a station side-connected configuration, where the PBX sends the port number through the Media Gateway as the DN (requires configuration on the Media Gateway). The call arrives on a `Trunk` DN configured for the port number provided by the PBX. This port number is forwarded to the CTI Connector for DNIS lookup before IVR Profile mapping can take place. Figure 5 on page 26 shows the basic call flow for a typical IVR-centric application.

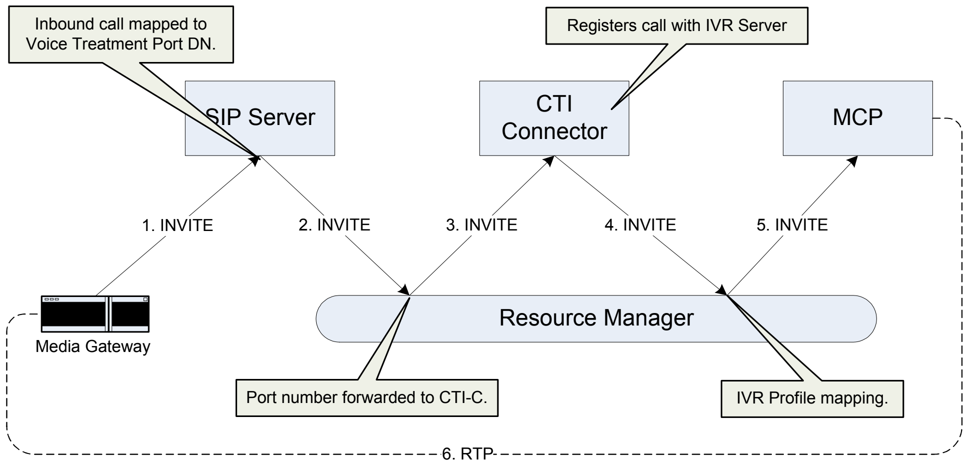


Figure 5: Basic IVR-Centric Application Call Flow

How It Works—The Basic Outbound Call Flow

To support outbound dialing, the VPS includes the Supplementary Services Gateway (SSG), an HTTP interface that can manage the initiation of outbound calls, either singly or in bulk, and connect the called number to a Standard VoiceXML application.

To initiate a call, a third-party "trigger" application (TA) sends CreateCall attributes in an `HTTP POST` request to the SSG. The SSG translates certain required parameters into `Extensions` attributes that it then includes in the `TMakePredictiveCall` request that it sends to SIP Server. These attributes can include Call Progress Detection (CPD) control parameters, used to determine the status of the called number—answering machine, fax, or voice—before connecting the called number with a particular voice application, as identified in the IVR Profile included in the `HTTP POST`.

Figure 6 on page 27 shows the basic call flow for a TA-initiated outbound call.

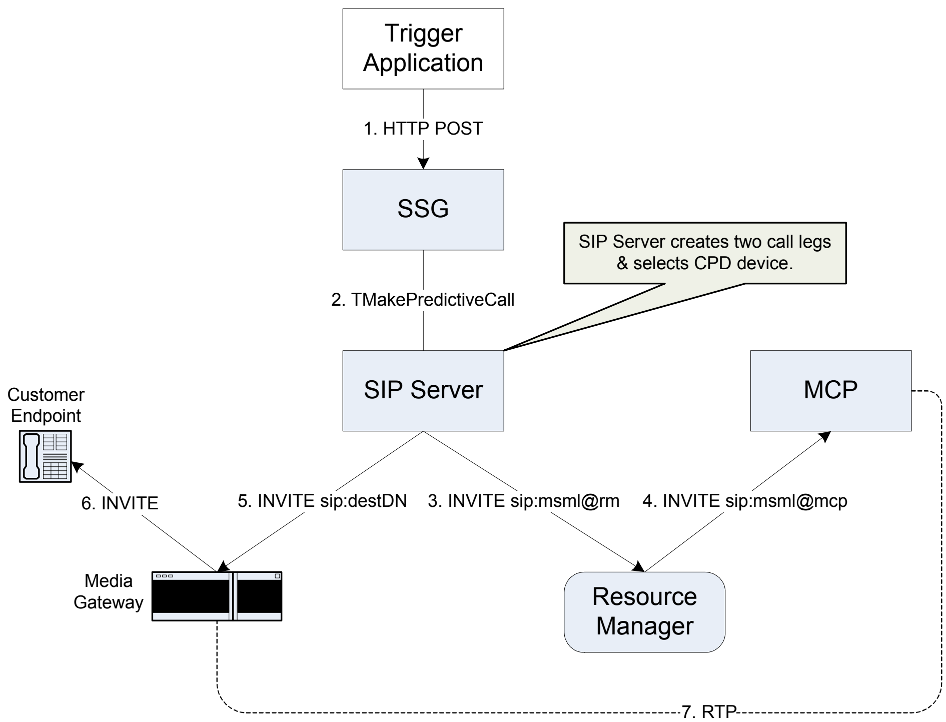


Figure 6: Basic Outbound Call Flow

Some key considerations about this call flow:

- For outbound calls, GVP is configured as a `Trunk Group` DN. SIP Server sends the `INVITE` to the outbound customer from this DN.

  Starting in 812, outbound calls can also be configured to go through a Routing Point DN (as an alternative to the Trunk Group configuration), for legacy deployments where GVP is configured as a series of Voice Treatment Ports.
- If CPD parameters are not specified in the `HTTP POST` request, the default parameters in the SIP Server application are used instead.
- If CPD is configured on both the media gateway and the MCP, then the media gateway will be used.

How It Works—CTI in the Solution

Depending on the needs of your deployment, the Voice Platform Solution provides computer-telephony integration (CTI) with the larger Genesys suite either through the SIP Server or through the IVR Server.

The three basic configurations as they relate to CTI in the solution are:

- Voice Platform Only (no CTI), page 28
- CTI Through SIP Server, page 28
- CTI Through IVR Server, page 28

Voice Platform Only (no CTI)

Suitable for small or medium Enterprise deployments that do not have any agents to service, and so do not require a full computer-telephony integration with the larger Genesys suite. SIP Server is still included in the deployment, though its CTI capability is not in use. In this case, you do not need to include CTI Connector or IVR Server in the deployment.

Voice platform-only configurations support only standard VoiceXML applications.

Note: For most switch configurations, SIP Server is able to provide CTI functionality should your deployment later require the addition of, for example, Universal Routing Server for VPS transfers to agents or workstations within the enterprise.

CTI Through SIP Server

Suitable for most IP-only contact center deployments, CTI through SIP Server configurations support the following types of applications:

- Standard VoiceXML applications
- URS-controlled applications

In deployments that use SIP Server for the CTI link to the larger Genesys suite, Genesys recommends that you develop your applications for voice self-service only, with all CTI functionality provided by the URS routing strategy. IVR-centric applications are not available in this configuration.

For detailed descriptions and call flow diagrams, see the following chapter:

- Chapter 2, “Supported CTI Through SIP Server Configurations,” on page 31

CTI Through IVR Server

Suitable for deployments that require backwards compatibility for legacy voice applications, or in switch configurations where SIP Server cannot provide the CTI connection to the larger suite, CTI through IVR Server configurations support the following types of applications:

- Standard VoiceXML applications
- IVR-centric applications

URS-controlled applications are not available in this configuration.

For detailed descriptions and call flow diagrams, see the following chapter:

- Chapter 3, “Supported CTI Through IVR Server Configurations,” on page 39.

About Genesys Administrator

Genesys Administrator is a Web-based user interface for the management and configuration of Genesys components.

Use Genesys Administrator to deploy, configure, provision, and monitor the solution.

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

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

For more information about deploying or using Genesys Administrator, consult the following:

- *Framework 8.1 Genesys Administrator Deployment Guide*
- *Framework 8.1 Genesys Administrator Help*

Chapter

2 Supported CTI Through SIP Server Configurations

The Voice Platform Solution (VPS) 8.1 supports a number of configurations where CTI is provided by SIP Server only (CTI Connector is not used). The diagrams and descriptions in this chapter demonstrate some of the more common CTI through SIP Server configurations, describing the kinds of voice applications each configuration supports, as well as the methods used to deliver CTI.

This chapter includes the following configurations:

PBX Trunk-Side Connection

Figure 7 on page 32 shows a PBX architecture configuration, where CTI is provided through SIP Server.

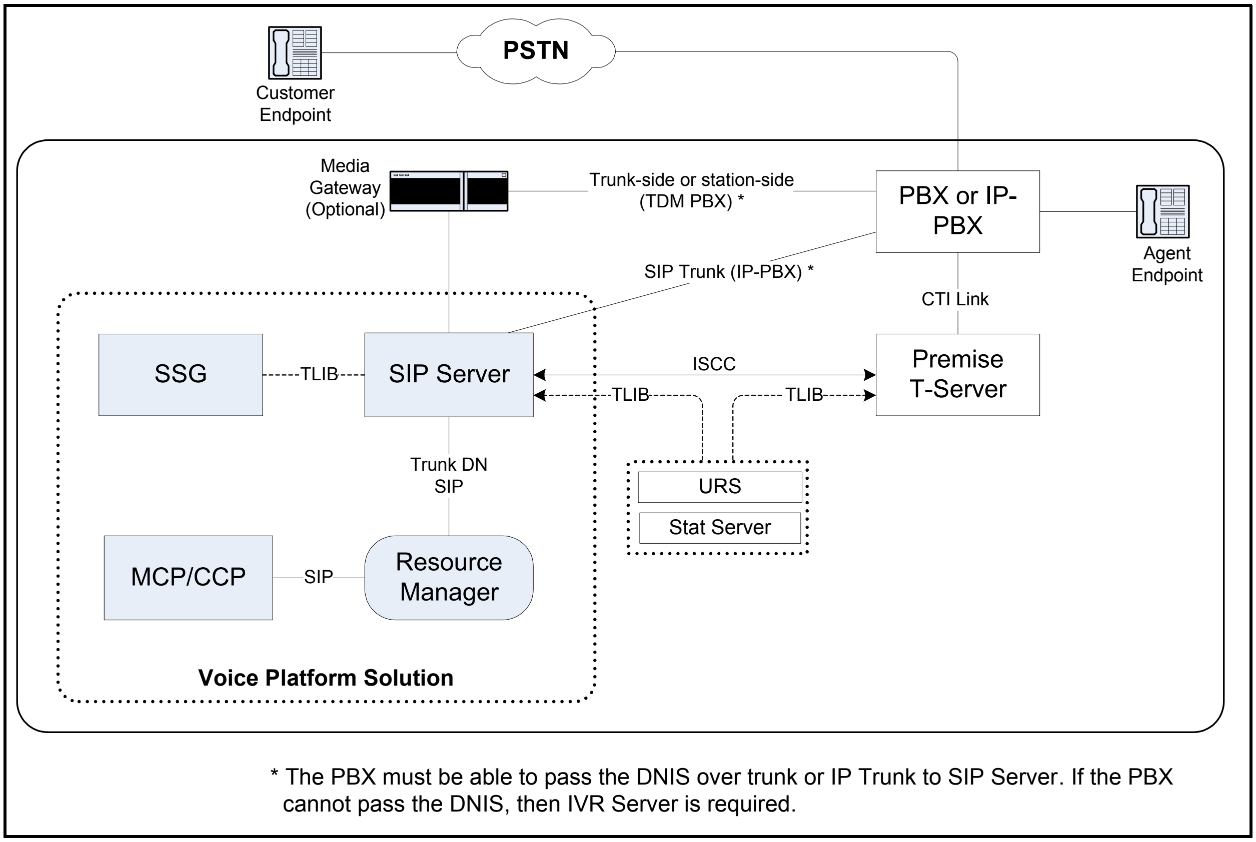


Figure 7: PBX Trunk-Side Connection, CTI Through SIP Server

In this architecture, the VPS is integrated with either a Time-Division Multiplexing (TDM) or Internet Protocol (IP) Private Branch Exchange (PBX). With an IP PBX, a direct trunk connection to the SIP Server is used. For more detail about the difference between TDM and IP PBX configurations, see “REFER Transfers to Agents on a PBX” on page 53.

For this architecture, you can build the following kinds of applications:

- URS-centric applications
- Standard VoiceXML applications

Incoming Calls

Incoming calls arrive at the VPS differently, depending on the kind of initial application invoked.

URS-centric Applications

For a URS-centric application, the incoming call arrives at a `Routing Point` DN configured in the SIP Server switch. A routing strategy loading on the Routing Point executes a Play Application treatment to collect customer input. SIP Server sends an `INVITE` specifying the URI for the voice application. MCP executes the application—customer data is collected, then returned to SIP Server in the BYE message. The routing strategy receives the attached data and determines the next action for the call.

Standard VoiceXML Applications

For a Standard VoiceXML application, the inbound call is forwarded to GVP directly through the Trunk DN to execute the self-service application. When the VoiceXML application determines that the call should be transferred to an agent, it executes the <transfer> tag (using SIP REFER) in order to transfer the call to a Routing Point DN on the SIP Server switch. After the transfer to SIP Server, the URS routing strategy takes control of the call (the call is considered parked on URS). The strategy initiates call treatments before transferring the call to an agent. To make the transfer, SIP Server communicates with the premise T-Server through Inter Server Call Control (ISCC) and sends a REFER request to the media gateway (or directly to the trunk for an IP PBX). This completes the transfer.

Outbound Calls

In this architecture, outbound calls are initiated using a trigger application through the SSG. The call is placed to GVP through either a Trunk Group DN or a Routing Point DN, depending on the kind of voice application used to connect with the called party. For more information, see "Outbound Calls Using SSG" on page 38.

Carrier-Connected Architecture

Figure 8 on page 34 shows a carrier-connected architecture configuration, where CTI is provided through SIP Server, and the agents are located off-premises.

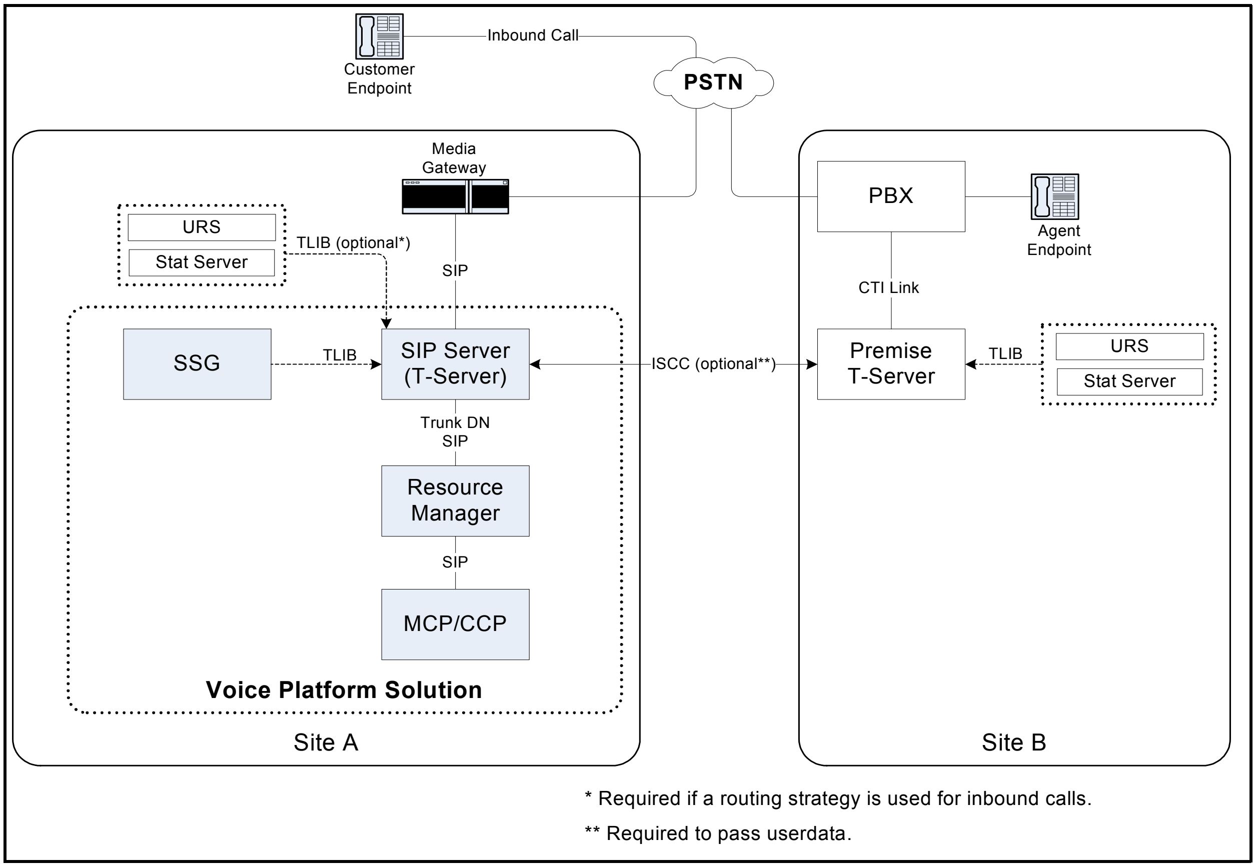


Figure 8: Carrier Connected, CTI Through SIP Server

In a carrier-connected architecture, the VPS and agents may reside on different premises. In this case, transfers from the VPS to an agent are made through ISCC communication with SIP Server on the VPS side, and with the premise T-Server for the site where the agents are located.

This architecture supports the following kinds of applications (as described for PBX trunk-side connected configurations):

- URS-centric applications
- Standard VoiceXML applications

Incoming Calls

Incoming calls arrive at the VPS differently, depending on the kind of initial application invoked:

- For URS-centric applications, see “URS-centric Applications” on page 32
- For Standard VoiceXML applications, see “Standard VoiceXML Applications” on page 33.

Outbound Calls

In this architecture, outbound calls are initiated using a trigger application through the SSG. The call is placed to GVP through either a `Trunk Group` DN or a `Routing Point` DN, depending on the kind of voice application used to connect with the called party. For more information, see “Outbound Calls Using SSG” on page 38.

Carrier-Connected—Voice Treatment Port Configuration

Figure 9 shows a carrier-connected configuration for legacy GVP customers, where GVP is configured as a series of `Voice Treatment Port` DNs.

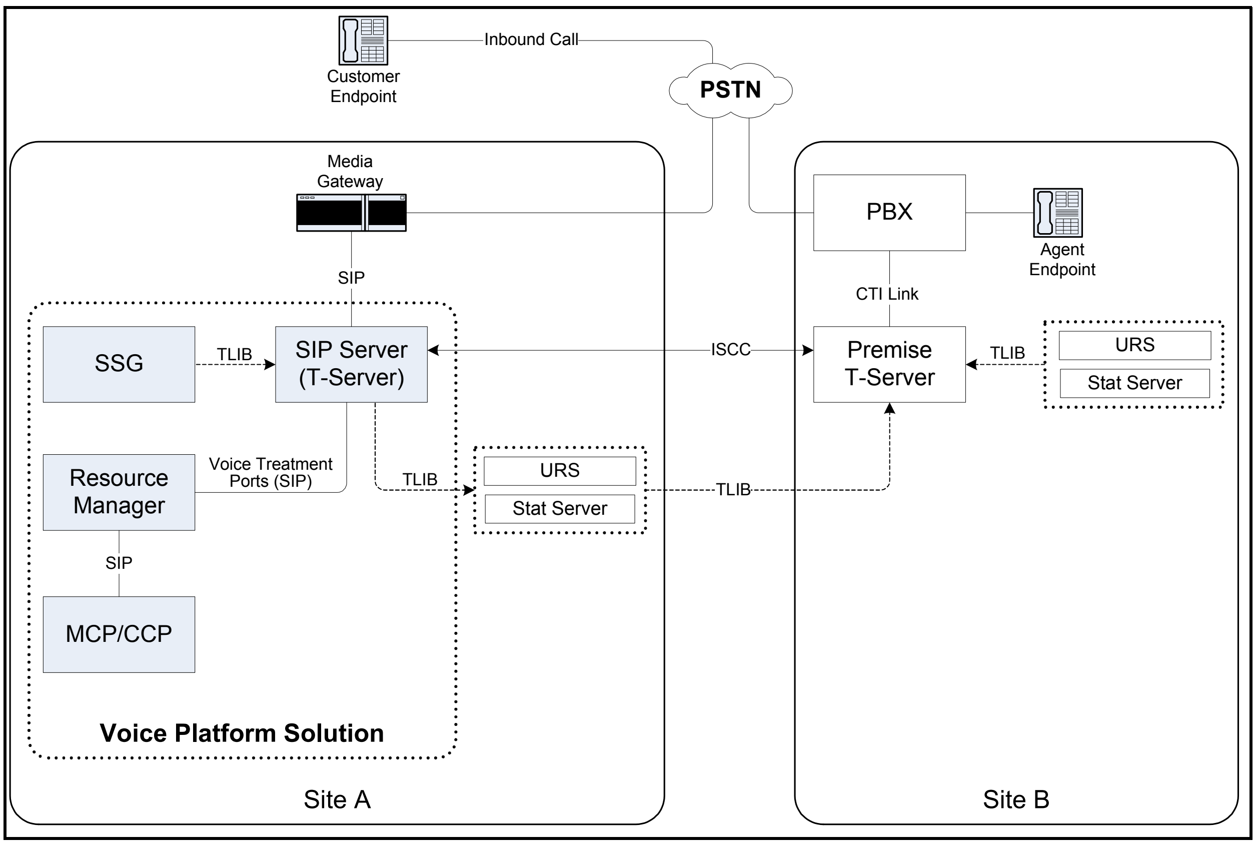


Figure 9: PBX—GVP Configures as Voice Treatment Port DNs

In this configuration, GVP acts only as an IVR when incoming calls are routed to the `Voice Treatment Port` DNs. At the same time, GVP can also act as the media server for all other Voice over IP services configured on SIP Server—SIP Server can use GVP as both an IVR and media server for the same call session.

This architecture supports the following type of application only:

- Standard VoiceXML applications

Incoming Calls

Inbound calls arrive at a Routing Point DN. The strategy routes the call to a Place Group, where the individual Places are configured as a set of Voice Treatment Port DNs. When routed to this Place Group, SIP Server selects a port and forwards it to Resource manager. Resource Manager selects an IVR Profile for a particular VoiceXML application, based on the DNIS provided in the To header of the original INVITE.

Transferring the Inbound Call

When the VoiceXML application determines that the call should be transferred to an agent, it can execute the <transfer> tag (using SIP REFER) in order to transfer the call to a DN on the SIP Server. For example, a Routing Point DN, where a second routing strategy can request that GVP—acting as a media server—supply call treatments while the customer is parked on the Routing Point DN, awaiting transfer to an available contact center agent.

Outbound Calls

In this architecture, outbound calls are initiated using a trigger application through the SSG, and the call is placed to GVP through a Routing Point DN. SIP Server sends an INVITE to GVP as the media server (Voice over IP Service DN with service-type set to msml). If the HTTP request specifies CPD, then CPD can be performed on the media server. After CPD, if the called line is determined to be suitable, the routing strategy can select a Place Group to determine the Voice Treatment Port DN that will be sent in the INVITE to Resource Manager, in order to select the IVR Profile for the voice application to be connected to the customer. After the initial customer interaction, the application can use the VoiceXML <transfer> tag to deliver the call to a destination—for example, a Routing Point DN for agent queuing. You can also use direct CTI from the application to transfer the call.

Pure IP Configuration

Figure 10 on page 37 shows a pure-IP configuration where SIP Server acts as both a T-Server and as a SIP switch to which the contact center agent registers. GVP is configured as a Trunk DN and can act as a media server.

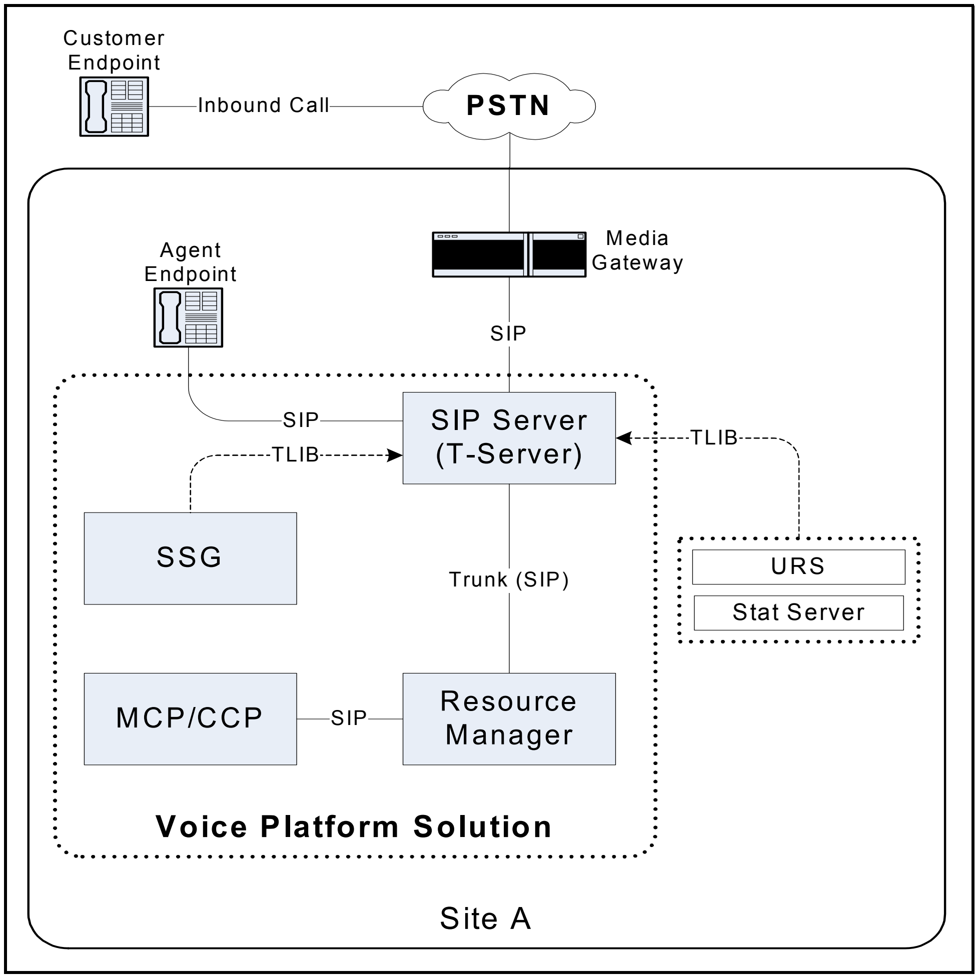


Figure 10: Pure IP Configuration

This architecture supports the following types of applications:

- URS-centric applications
- Standard VoiceXML applications

Incoming Calls

Incoming calls arrive at the VPS differently, depending on the kind of initial application invoked:

- For URS-centric applications, see "URS-centric Applications" on page 32
- For Standard VoiceXML applications, see "Standard VoiceXML Applications" on page 33.

Outbound Calls

In this architecture, outbound calls are initiated using a trigger application through the SSG. The call is placed to GVP through either a `Trunk Group` DN or a `Routing Point` DN, depending on the kind of voice application used to connect with the called party. For more information, see "Outbound Calls Using SSG" on page 38.

Outbound Calls Using SSG

In CTI through SIP Server architectures, outbound calls are made through either a `Trunk Group` DN or a `Routing Point` DN, depending on the kind of voice application used to connect with the called party.

NGI Applications—Trunk Group Call Flow

For NGI applications, SIP Server sends the outbound `INVITE` through a `Trunk Group` DN configured for MSML communication to GVP. SIP Server can apply call progress detection (CPD) on the media gateway. After CPD, if the called line is determined to be suitable, it is connected to the voice application.

GVPi Applications—Routing Point Call Flow

For GVPi applications, SIP Server places the originating call leg on a `Routing Point` DN. SIP Server sends an `INVITE` to GVP as the media server (`Voice over IP Service` DN with `service-type` set to `msml`). If the HTTP request specifies CPD, then CPD can be performed on the media server. After CPD, if the called line is determined to be suitable, the routing strategy can select a Place Group to determine the `Voice Treatment Port` DN that will be sent in the `INVITE` to Resource Manager, in order to select the IVR Profile for the voice application to be connected to the customer.

Transferring the Call

After the initial customer interaction, the application can use the VoiceXML `<transfer>` tag to deliver the call to a destination—for example, a `Routing Point` DN for agent queuing.

Note: For information about how to integrate SSG into the solution, see Chapter 13, "Integrating with SSG," on page 195.

Chapter

3 Supported CTI Through IVR Server Configurations

The Voice Platform Solution (VPS) 8.1 supports a number of configurations where CTI is provided through the IVR Server, with CTI Connector providing the link between GVP and the rest of the Genesys suite. The diagrams and descriptions in this chapter demonstrate some of the more common CTI through IVR Server architectures, describing the kinds of voice applications each architecture supports, as well as the methods used to deliver CTI.

This chapter includes the following sections:

Carrier-Connected, CTI Through IVR Server

Figure 11 on page 40 shows a carrier connected architecture configuration, where CTI is provided through IVR Server, and the agents are located off-premise.

In this architecture, the VPS is integrated with an IVR Server configured for Behind mode operation. To facilitate this integration, an additional VPS component is required—CTI Connector. For a detailed call flow of an integration with IVR Server in Behind mode, see “Integration with IVR Server—Behind Mode” on page 90.

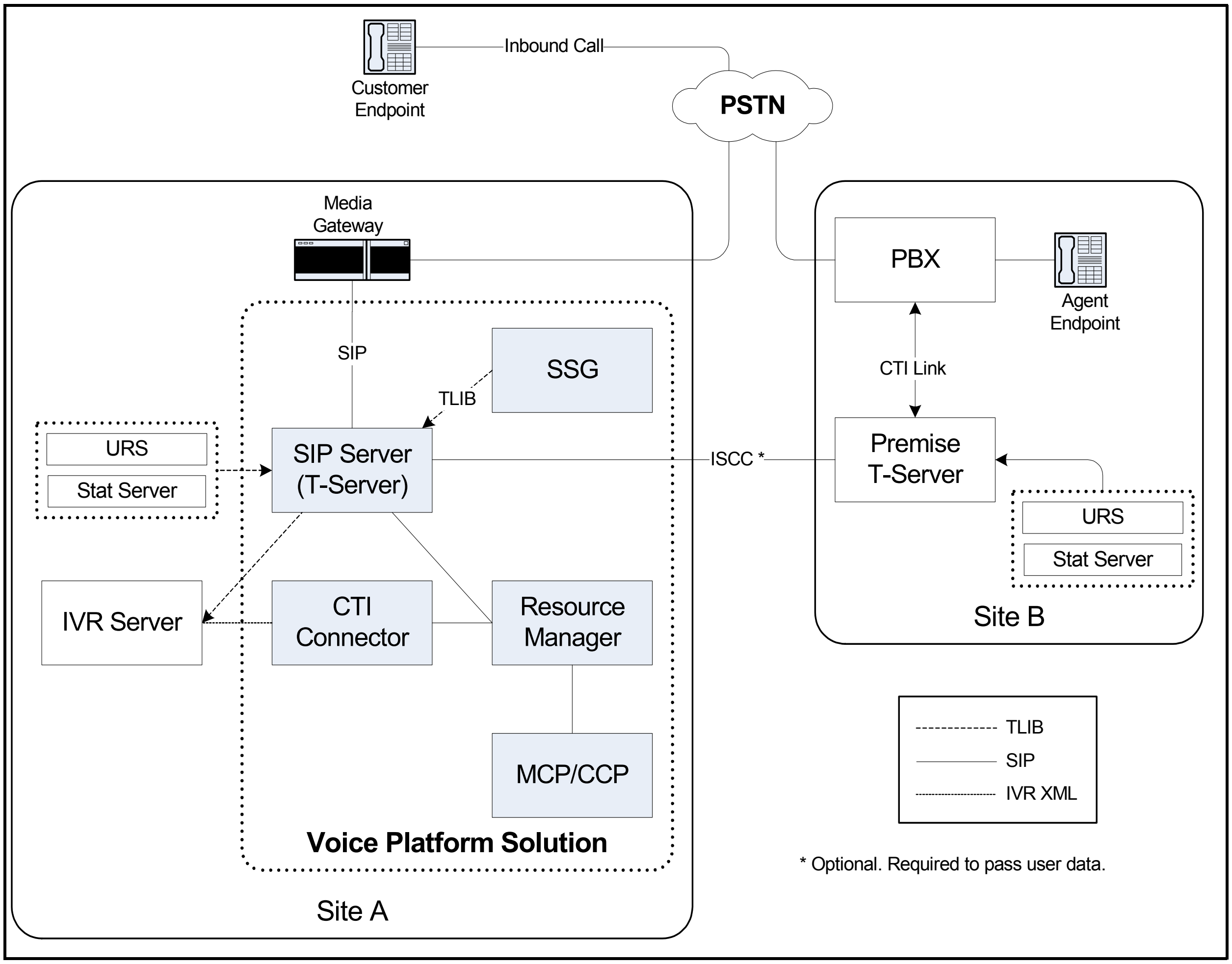


Figure 11: Carrier Connected, CTI Through IVR Server

This architecture is typically required for migrating customers who must use legacy GVPi applications in 8.1 VPS deployments. The architecture supports the following kinds of applications:

- Standard VoiceXML applications—Applies to NGI applications only. The application uses standard VoiceXML tags to perform the voice interaction, and it uses the `<transfer>` tag to place the caller in a queue for routing to an agent. While the call is waiting in the queue, it is considered parked at the CTI Connector—MCP can provide call treatments while the agent waits.
- IVR-centric applications—Applies to both NGI and GVPi applications. The VoiceXML application maintains control of the call while the caller waits in a queue for routing to an agent. GVPi applications can be built using VoiceXML and TXML tags; by contrast, NGI applications use VoiceXML tags, as well as proprietary `<send>` and `<receive>` extensions to perform CTI functions directly from the application.

Incoming Calls

In this architecture, the VPS can be configured to flag the gateway resource for the SIP Server, as well as the IVR Profile for the voice application, so that the call will be sent through IVR Server—for applications that require CTI through IVR Server functionality; it can also be configured to bypass the IVR Server for voice-only applications that do not required direct CTI functionality from the application itself.

Note: To configure this functionality, see “Configuring CTI Flagging for IVR Profiles” on page 188.

Incoming Call for an Application with CTI Functionality

For applications that include direct CTI functionality, the incoming call arrives at a `Routing Point` DN on the SIP Server switch, where a routing strategy selects a `Place` object mapped to a `Voice Treatment Port` DN. This operation selects the IVR port that SIP Server includes as the user part of the request URI in the `INVITE` message that it sends to Resource Manager. CTI Connector, through IVR Server, performs the DNIS lookup and provides the Resource Manager with the DNIS it needs in order to map the IVR Profile for the destination voice application.

Incoming Call for a Self-Service Application (No CTI)

For voice-only applications with no direct CTI, the incoming call arrives at GVP through a `Trunk` DN on the SIP Server switch. SIP Server does not select a `Voice Treatment Port` DN at this time. Based on CTI flagging, the Resource Manager forwards the `INVITE` from SIP Server to MCP directly, instead of going through CTI Connector. The VoiceXML application cannot perform any CTI functionality.

Transfers

For Standard VoiceXML applications, this architecture supports both blind (`REFER`) and bridge (`INVITE`) transfers. For IVR-centric applications, this architecture supports blind transfers through CTI link or through SIP, as well as bridge transfers through SIP.

Standard VoiceXML Applications

Transfers from Standard VoiceXML applications work in this configuration as follows:

- Blind transfer to an agent—The application uses the `<transfer>` tag to initiate a `REFER` transfer from the voice self-service portion of the call to the agent-assisted portion. When the agent is ready to receive the caller, the IVR Server sends a route number to CTI Connector, and CTI Connector sends a `REFER` request to the SIP Server. The media gateway negotiates the transfer from the VPS site, through the PSTN, to the site where the agent is located. For a more detailed description of the setup for these transfers, see "Standard VoiceXML Applications—REFER" on page 97.

- Bridge transfer to an agent—The application also uses the `<transfer>` tag. In this case, CTI Connector receives an `INVITE` request for a bridged transfer. For a more detailed description, see "Standard VoiceXML Applications—Bridge Transfer" on page 98.

IVR-centric Applications

After the IVR phase of the call, the `RouteRequest` block in the IVR-centric application can be used to send a request through CTI Connector to find an agent. CTI Connector acts as a SIP back-to-back user agent (B2BUA), remaining in the call path until the transfer to the agent is complete.

Transfers from IVR-centric applications work in this configuration as follows:

Blind transfer Blind transfers in IVR-centric applications have two distinct steps:

1. CTI—A `RouteRequest` block in the VoiceXML application obtains a route number.
2. `REFER` transfer or `OneStepXfer`—Once the route number is known, the VoiceXML application uses `<transfer>` to perform the actual call transfer. MCP sends a `REFER` message to CTI Connector. CTI Connector can choose, based on the IVR Profile, whether to pass the `REFER` to SIP Server and the gateway to complete the transfer, or to use `OneStepXfer` for a CTI transfer through IVR Server to a ready agent.

   In a carrier-connected deployment, the `REFER` through SIP Server method would typically be used; however, `OneStepXfer` is also supported.

Note: If the connection between SIP Server and the premise T-Server is through ISCC, then transfers through CTI require the use of `AccessNumGet` block in the VoiceXML application. For more information, see "Enabling Midcall CTI Routing (IVR-centric Applications)" on page 191.

Bridge transfer If the IVR-centric application is configured for a bridge transfer, MCP sends an `INVITE` through Resource Manager to CTI Connector, requesting a bridge transfer. When the agent is ready, CTI Connector obtains the agent DN, sends an `INVITE` to the SIP Server, and bridges the call to the ready agent.

Configuring IVR-centric Transfers

For detailed information about configuring IVR Profiles for these various transfers, see "Task Summary: Configuring Transfers" on page 163.

Outbound Calls

In this architecture, outbound calls are initiated using a trigger application through the SSG. The call is placed to GVP through a Routing Point DN. For more information, see "Outbound Calls Using SSG" on page 45.

Station Side-Connected, CTI Through IVR Server

Figure 12 shows a VPS integration with a PBX that does not support trunk-side connections, but that requires a station-side port connection instead.

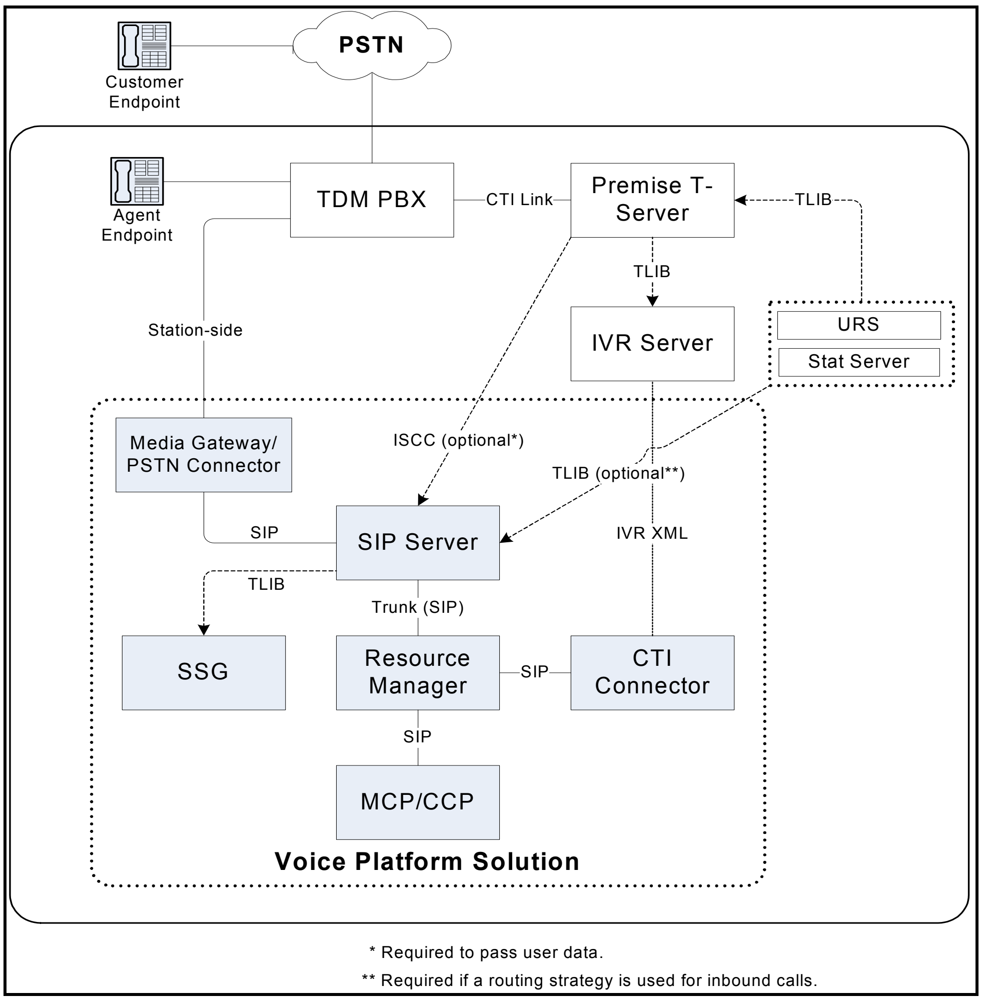


Figure 12: PBX Station Side-Connected, CTI Through IVR Server

In this architecture, the PBX forwards the incoming call to the media gateway (or PSTN Connector) and from there to SIP Server. SIP Server forwards the call to the Resource Manager through a `Trunk` DN. Resource Manager then forwards the call to CTI Connector, so that the call can be registered through the IVR Server.

This architecture supports the following kinds of applications:

- Standard VoiceXML applications
- IVR-centric applications

Note: For the relevant procedures for this configuration, see “Task Summary: IVR Behind, TDM-Connected Integration” on page 151.

Incoming Calls

A `Trunk` DN is required for every incoming port number from the PBX. The media gateway forwards the port number in the `INVITE` to the SIP Server. CTI Connector, through IVR Server, uses this port number to request the DNIS for the call. CTI Connector then sends the DNIS in an `INVITE` to the Resource Manager, for IVR Profile mapping to the voice application.

Note: In this architecture, all calls must go through CTI Connector. Configure the gateway resource so that `CTI usage` is set to `Always On` (`use-cti=1`)

Transfers

For Standard VoiceXML applications, this architecture supports both blind (`REFER`) and bridge (`INVITE`) transfers. For IVR-centric applications, this architecture supports blind transfers through CTI link, as well as bridge transfers through SIP. Blind transfers through SIP Server are not available for this configuration.

Standard VoiceXML Applications

Transfers from Standard VoiceXML applications work in this configuration as follows:

- Blind transfer to an agent—The application uses the `<transfer>` tag to initiate a `REFER` transfer from the voice self-service portion of the call to the agent-assisted portion. When the agent is ready, the IVR Server sends a route number to CTI Connector, CTI Connector initiates a CTI transfer to the premise T-Server through the IVR Server, and the PBX connects the caller to the agent directly. For a more detailed description of the setup for these transfers, see “Standard VoiceXML Applications—REFER” on page 97.
- Bridge transfer to an agent—The application also uses the `<transfer>` tag. In this case, CTI Connector receives an `INVITE` requesting a bridged

transfer. For a more detailed description, see “Standard VoiceXML Applications—Bridge Transfer” on page 98.

IVR-centric Applications

After the IVR phase of the call, an IVR-centric application can send a request through CTI Connector to find an agent. CTI Connector acts as a SIP B2BUA, remaining in the call path until the transfer to the agent is complete.

Transfers from IVR-centric application work in this configuration as follows:

- Blind transfers through CTI—In configurations with IVR Server in Behind mode, the application can initiate `REFER` transfers through IVR Server by using the `OneStepxfer` request. In this case, if the application is configured to use IVR Server (`CTI Transfer` in the IVR Profile is set to `true`), when an agent is ready, the application retrieves the agent number and MCP sends a `REFER` to CTI Connector. Instead of passing the `REFER` to the media gateway, CTI Connector performs a CTI transfer using the IVR XML `OneStepXfer` message, and the PBX connects the caller to the agent directly. For a more detailed description of the setup for these transfers, see “IVR-centric Applications—REFER Transfer” on page 98.
- Bridge transfers—If the IVR-centric application is configured for a bridge transfer, MCP sends an `INVITE` through Resource Manager to CTI Connector, requesting a bridge transfer. CTI Connector sends a `RouteRequest` to the IVR Server. When the agent is ready, IVR Server returns a route number, and CTI Connector sends an `INVITE` to the SIP Server and bridges the call to the ready agent.

Outbound Calls

In this architecture, outbound calls are initiated using a trigger application through the SSG. The call is placed to GVP through a Routing Point DN. For more information, see “Outbound Calls Using SSG”.

Outbound Calls Using SSG

In CTI through IVR Server architectures, SIP Server places the originating call leg on a `Routing Point` DN. SIP Server sends an `INVITE` to GVP as the media server (`Voice over IP Service` DN with `service-type` set to `msml`). If the HTTP request specifies CPD, then CPD can be performed on the media server. After CPD, if the called line is determined to be suitable, the routing strategy can select a Place Group to determine the `Voice Treatment Port` DN that will be sent in the `INVITE` to Resource Manager, in order to select the IVR Profile for the voice application to be connected to the customer.

Transferring the Call

After the initial customer interaction, you can also use direct CTI from the application to transfer the call.

Note: For information about how to integrate SSG into the solution, see Chapter 13, "Integrating with SSG," on page 195.

Chapter

4 Supported Call Flow Scenarios

The Voice Platform Solution 8.1 supports a variety of transfer types, call features, and integrations with different Genesys or third-party components. The diagrams in this chapter demonstrate how the call behaves for some of the more common call flow scenarios. Application developers can design the URS routing strategies and VoiceXML/CCXML applications to accommodate these scenarios.

Some common call flow scenarios used by the solution include:

URS-Centric Voice Applications

Figure 13 shows a basic configuration that supports the URS method for launching the voice self-service application.

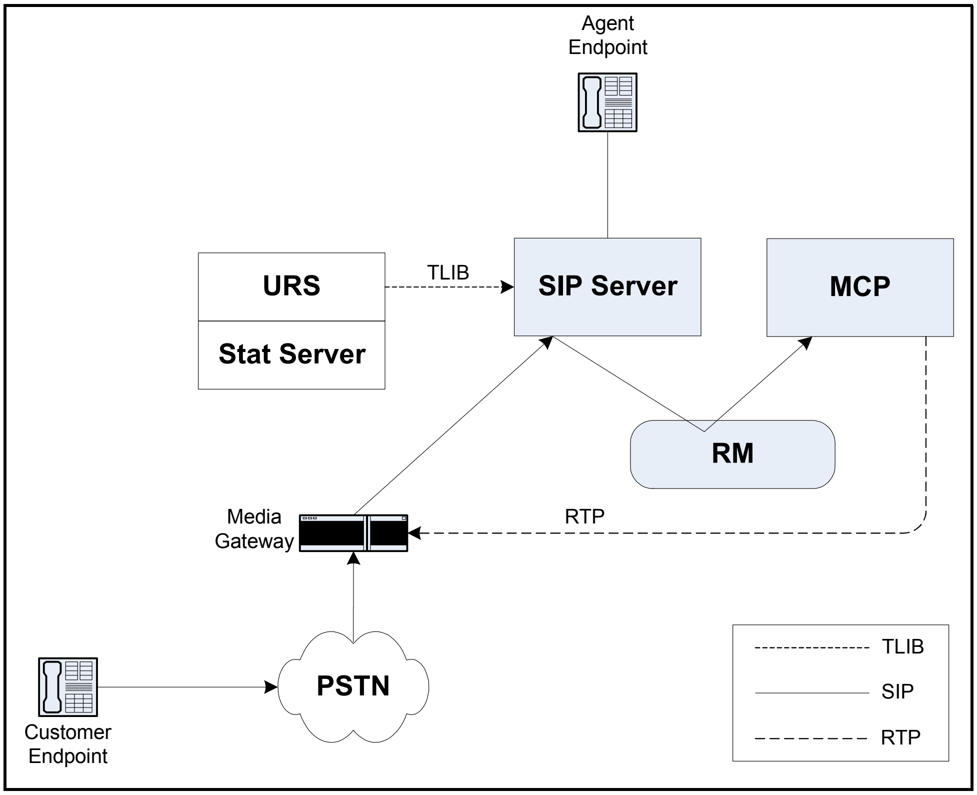


Figure 13: URS Starts the VoiceXML Application—Basic Configuration

In this 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.

Transfer Types

Supported transfer types included:

- "REFER Transfers to Agents on SIP Server"
- "Bridged Transfers to Agents on SIP Server"
- "Media Redirect Transfers to Agents on SIP Server"
- "REFER Transfers to Agents on a PBX"
- "GVP Launches the REFER with Replaces Transfer Method"

Note: For detailed information about SIP Server behavior and configuration for `REFER` transfers, consult the "Call Transfer and Conference" section of the *Framework 8.1 SIP Server Deployment Guide*. That section includes descriptions of how the following options are used:
- `refer-enabled`
- `oosp-transfer-enabled`

REFER Transfers to Agents on SIP Server

Figure 14 shows a basic configuration where GVP initiates a blind transfer to an agent registered on the SIP Server.

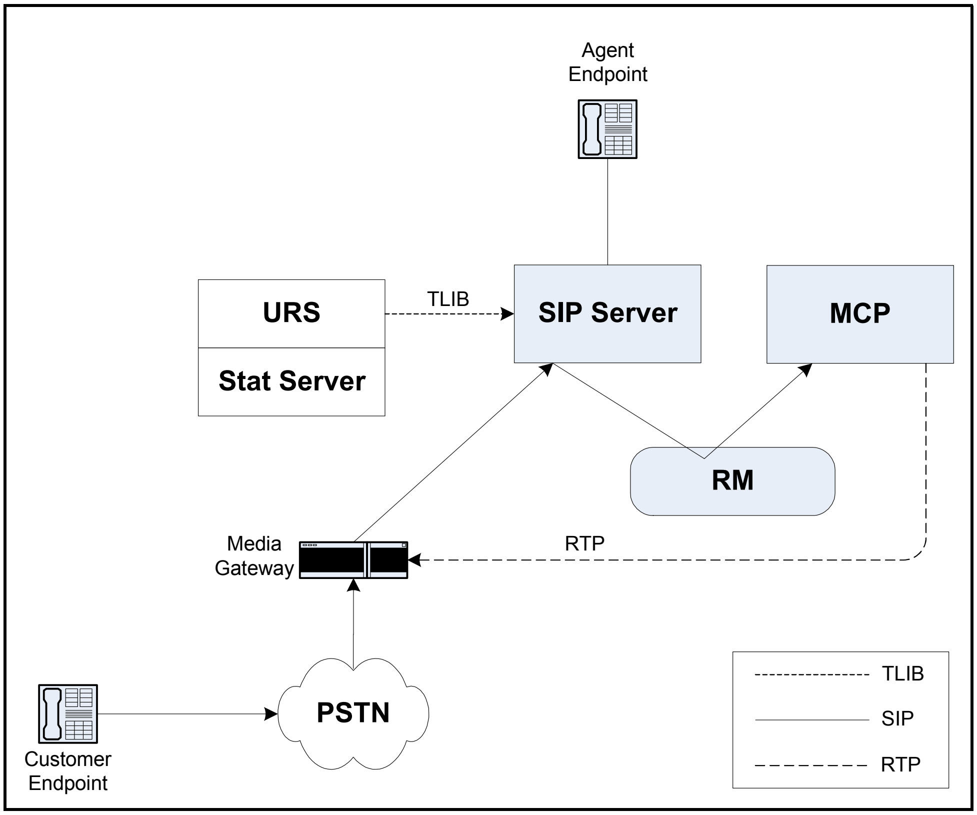


Figure 14: Blind Transfer—Basic Configuration

In this IP-based scenario, the incoming call is forwarded directly to GVP (configured as a `Trunk` DN on the SIP Server switch), 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 `Routing Point` DN (or any other DN) on the SIP Server. After SIP Server receives the call, the URS strategy takes control and can route the call to an agent who is 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 15 on page 51 shows a basic configuration where GVP initiates a bridged transfer to an agent registered on the SIP switch.

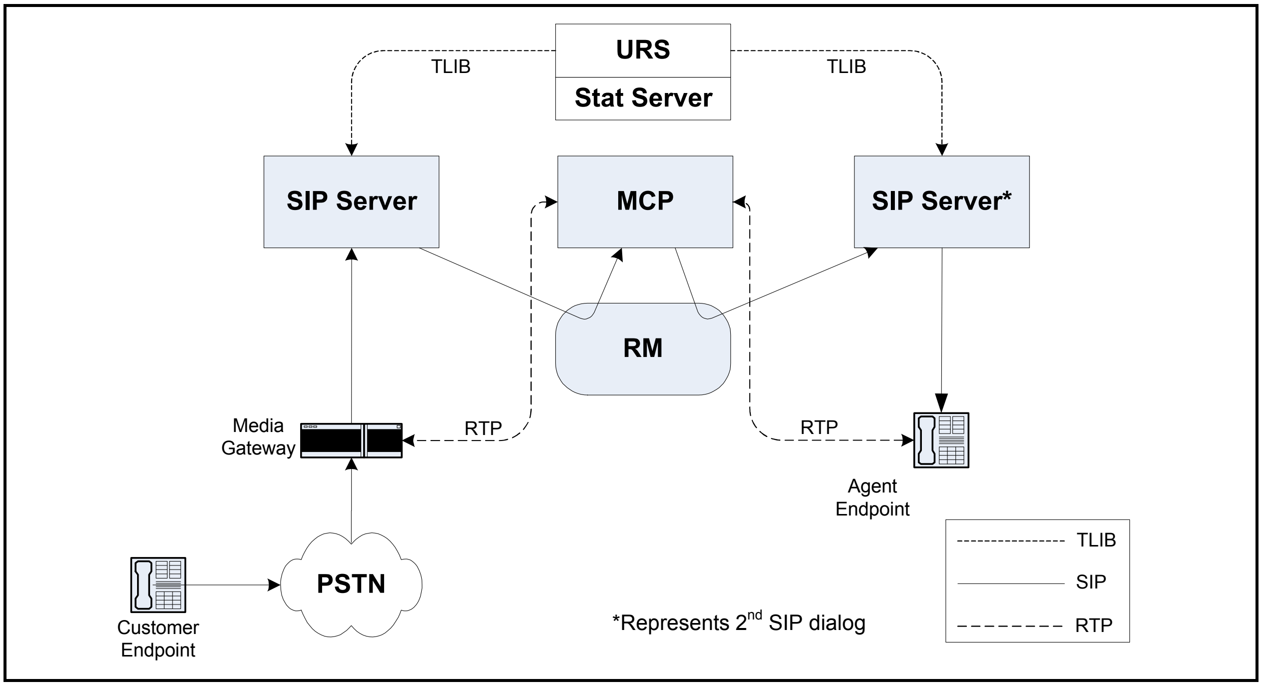


Figure 15: Bridged Transfer—Basic Configuration

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 15 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 16 on page 52 shows a basic configuration where GVP initiates a media redirect transfer to an agent registered on the SIP switch.

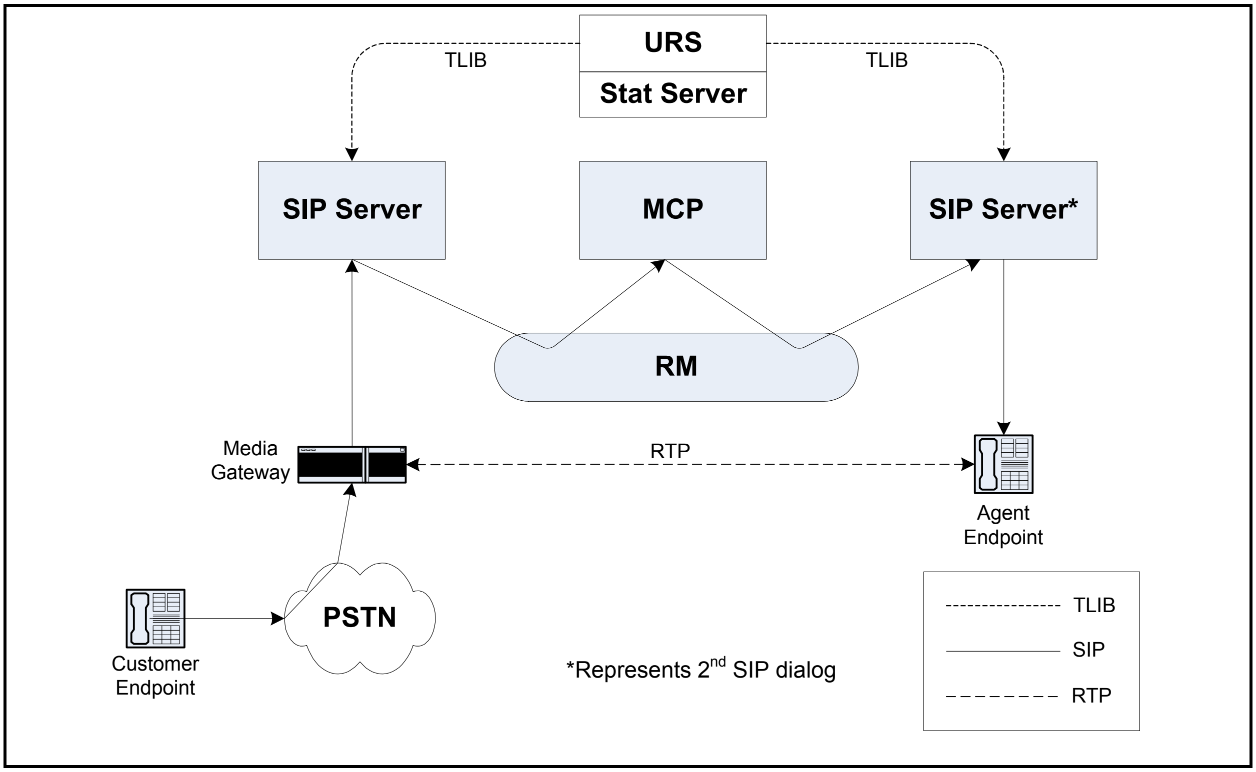


Figure 16: Media Redirect Transfer—Basic Configuration

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 16 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.

Special Configuration

For Media Redirect transfers, you must either change the default transfer type in MCP to `mediaredirect`, or configure the `<transfer>` tag in the VoiceXML application with the appropriate gvp extensions. For configuration details, see “Configuring the Solution for Media Redirect Transfers” on page 125.

REFER Transfers to Agents on a PBX

Figure 17 shows a configuration that includes a Private Branch Exchange (PBX) in an enterprise deployment.

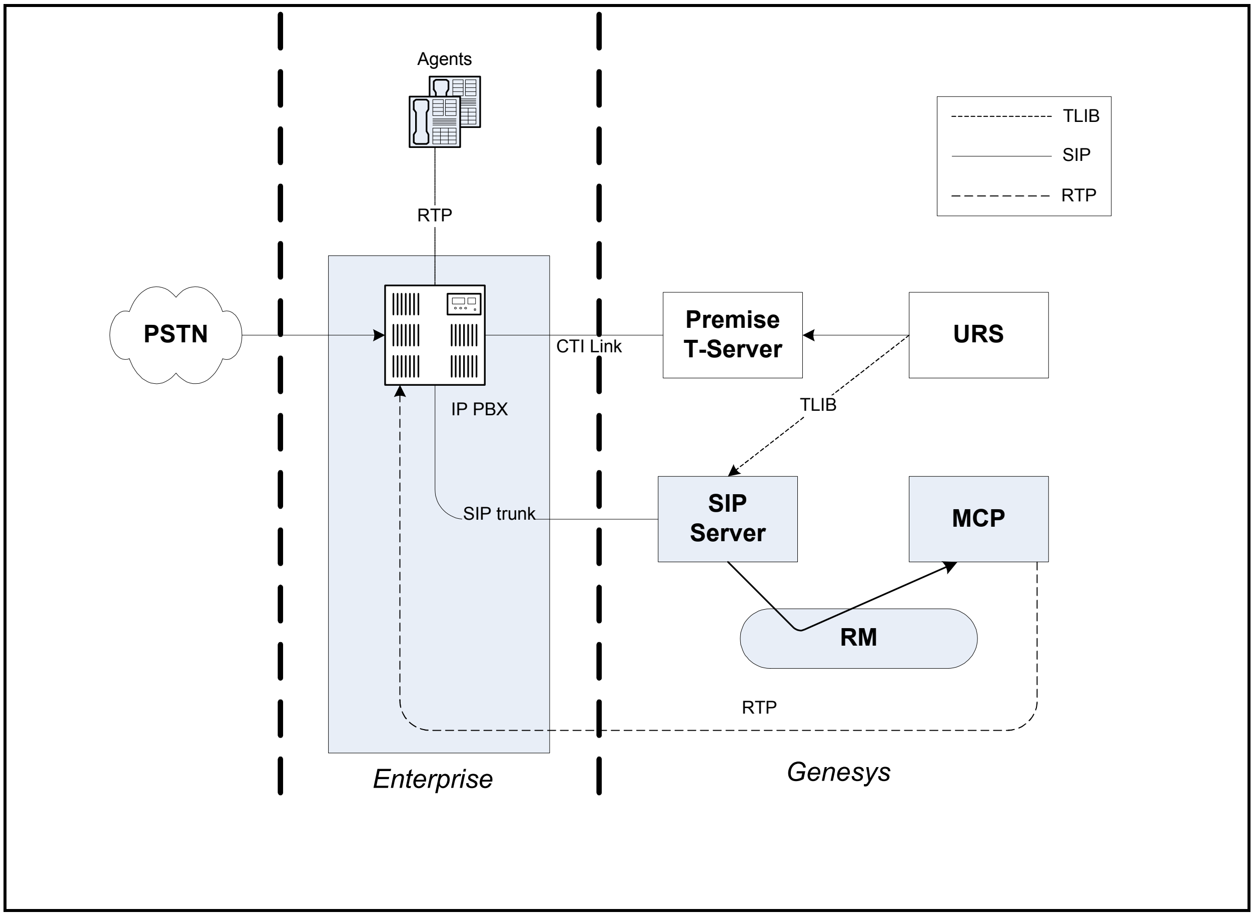


Figure 17: Enterprise PBX—Basic Configuration

In this scenario, GVP sits behind an IP PBX, and the PBX can direct calls to the VPS using a SIP trunk to the SIP Server. Agents are registered as telephone extensions on the PBX.

Time-Division Multiplexing PBX

If using 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 performs an Inter Server Call Control (ISCC) transfer from the SIP Server to an available agent and completes 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.

GVP Launches the REFER with Replaces Transfer Method

Figure 18 shows a configuration where the MCP performs a consultation transfer using the `REFER with replaces` method.

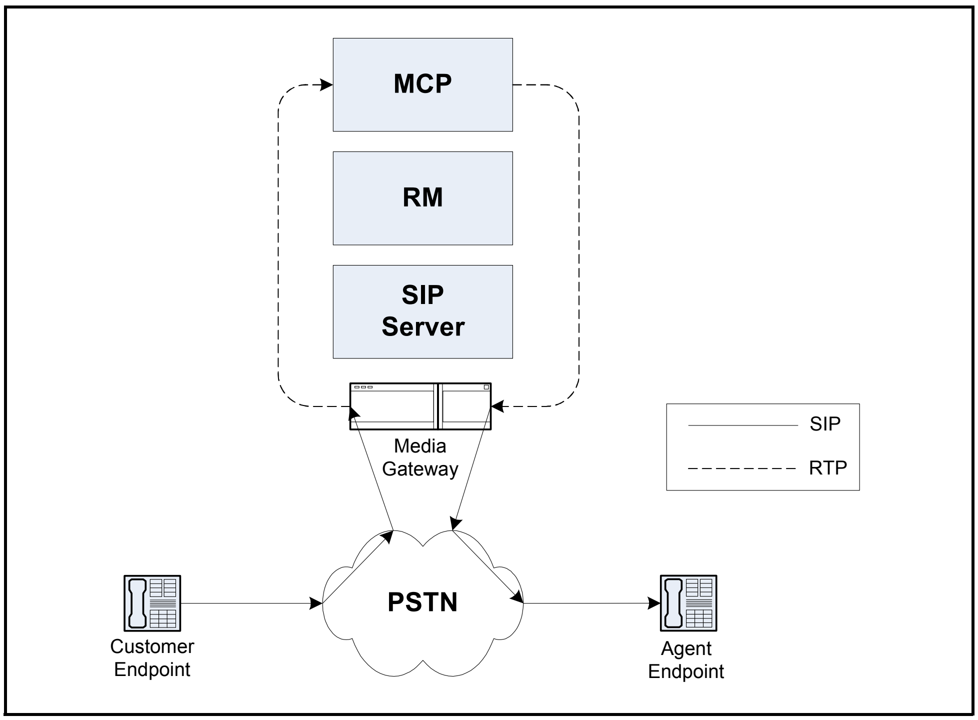


Figure 18: REFER with replaces—Basic Configuration

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 18 on page 54 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 18, 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.

Limitation

Consultation transfers using the SIP `REFER with replaces` method are not completely supported.

REFER is not supported if the inbound leg is originally pinned on a route point (e.g., GVP is configured as a VOIP service DN).

CCXML

Figure 19 shows an example of third party call control involving a CCXML application.

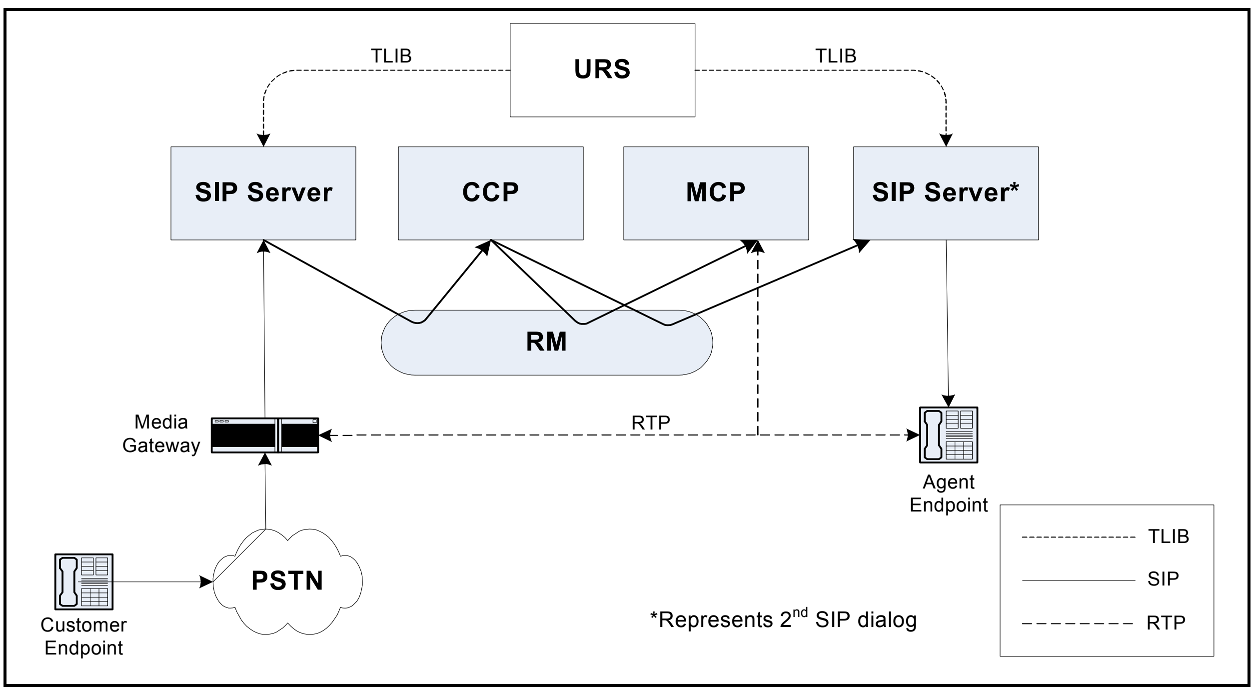


Figure 19: CCXML Conferencing—Basic Configuration

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

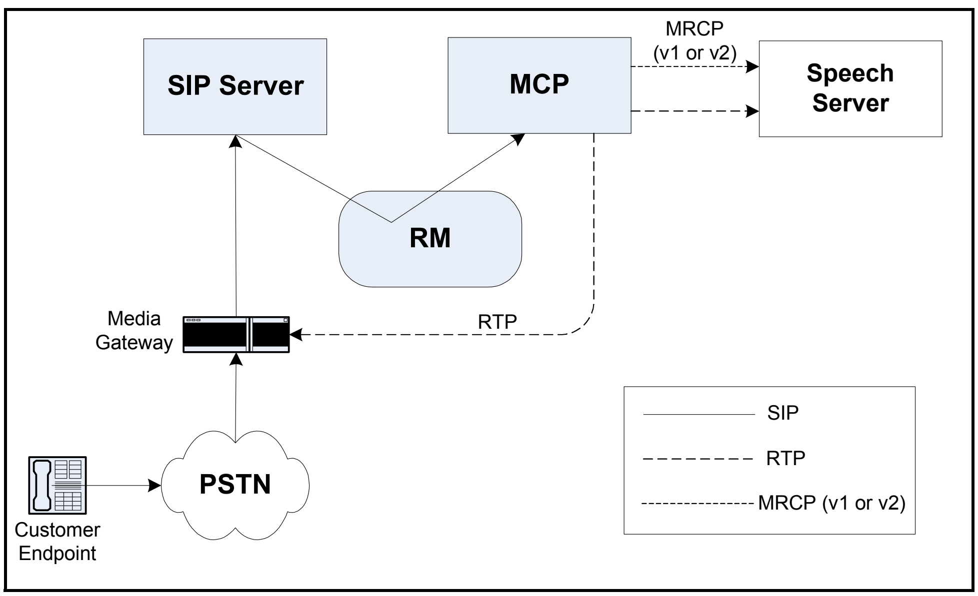


Figure 20: Speech Recognition—Basic Configuration

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.

Cisco CallManager

Figure 21 shows a configuration that includes the Cisco CallManager (CCM) in an enterprise deployment.

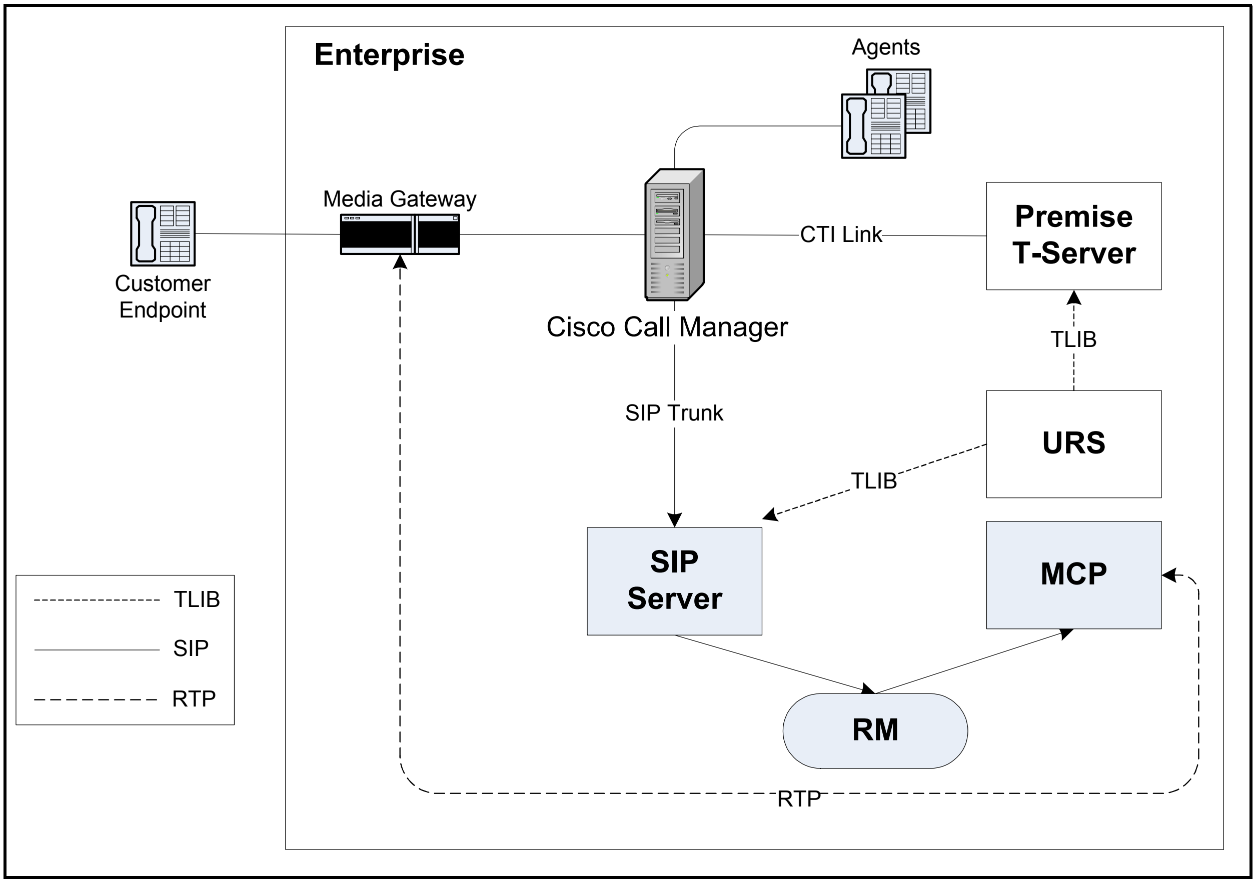


Figure 21: VPS Integration with Cisco CallManager

In this scenario, the Voice Platform Solution is integrated with the Cisco CallManager (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.

Outbound Calls

Supported outbound call flows include:

- "Outbound Calls Using the Supplementary Services Gateway"
- "Outbound Calls Using Remote Dialer Service"
- "Outbound Calls Using CCP"

Outbound Calls Using the Supplementary Services Gateway

Figure 22 shows an outbound call scenario initiated by a trigger application and controlled by the Supplementary Services Gateway (SSG).

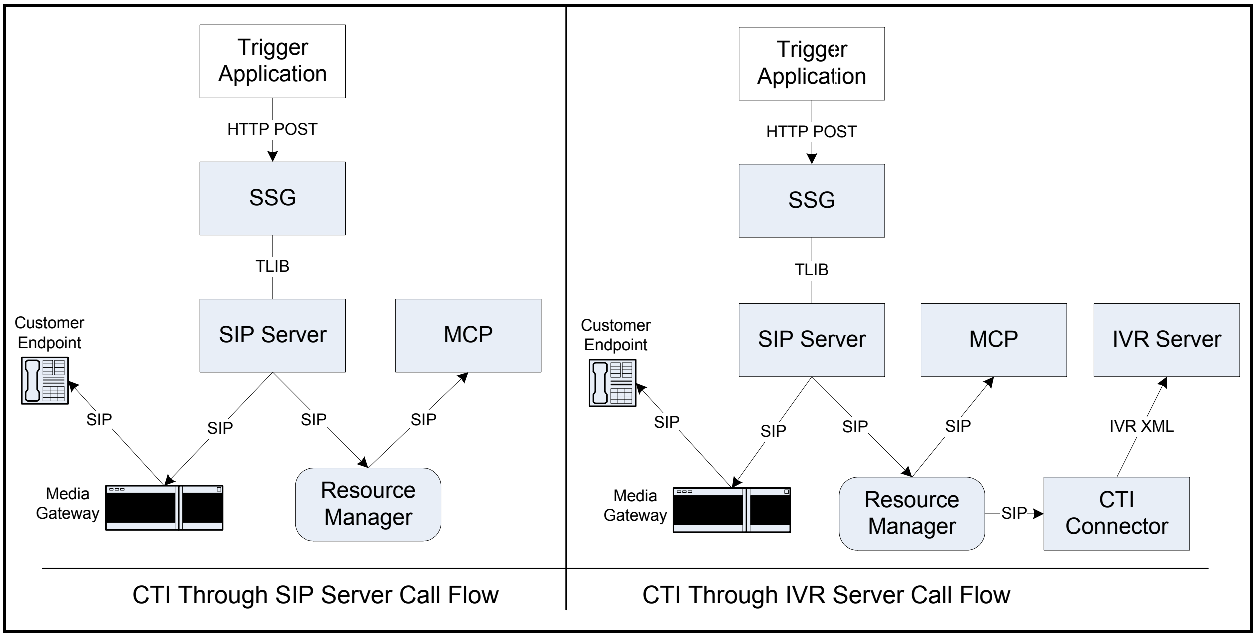


Figure 22: Outbound Call Initiated Using SSG

In this scenario, a trigger application sends an `HTTP POST` request with the information required by the VPS to make an outbound call. SSG translates this request into a T-Library `TMakePredictiveCall` request that it sends to SIP Server. The call parameters from the `HTTP POST` are included as `Extensions` in the T-Library request. SIP Server creates two call legs: one to MCP in order to start the voice application, another to the media gateway in order to establish a connection with the targeted destination for the call. If the trigger application includes instructions to use Call Progress Detection (CPD) before connecting the destination to a voice application, SIP Server will start CPD—on either MCP or the media gateway, depending on the configuration—and then, depending on CPD results, connect the called number with the voice application.

Making Calls Through a Trunk Group DN

Depending on the call scenario, SIP Server will send calls initiated by SSG to GVP through either a `Trunk Group` DN or a `Routing Point` DN.

The `Trunk Group` DN call flow is used for the following scenarios:

- CTI through SIP Server, where NGI applications are used.

Making Calls Through a Routing Point

For other scenarios, the outbound call from SSG must go through a Routing Point, with GVP configured as a series of `Voice Treatment Port` DNs. These scenarios include:

- **CTI through IVR Server integrations.** CTI Connector cannot process MSML INVITE requests. For deployments that include CTI Connector, you must configure VPS to support this call flow.
- **Legacy GVPi applications.** The Media Control Platform cannot process GVPi applications in an MSML scenario. If you want the outbound call to connect customers to legacy a GVPi application, you must configure VPS to support the Routing Point call flow.

Note: To configure VPS for either call flow, see Chapter 13, "Integrating with SSG," on page 195.

Outbound Calls Using Remote Dialer Service

Figure 23 shows an outbound call scenario initiated by the Remote Dialer (remdial) service provided by the Media Control Platform. This scenario is supported by any configuration in which the solution is able to make an outbound call to the network.

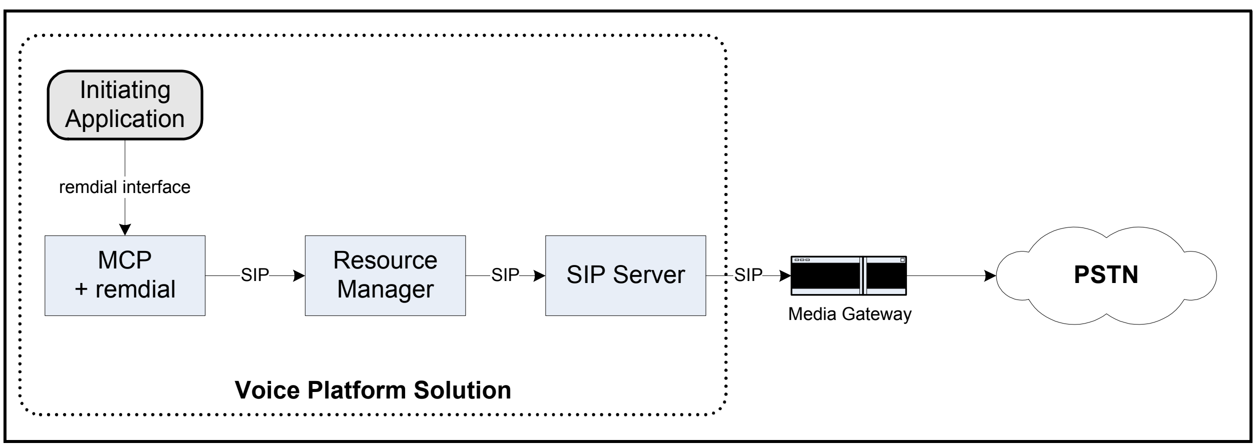


Figure 23: Outbound Call Initiated Using the Remdial Service

With the remote dial service, you can use a Telnet command-line interface to connect to a preconfigured remote dialing port in order to place outbound calls.

Basic Remote Dialing

In basic remote dialing—where no Call Progress Analysis (CPA) is performed—the user sends a remote dial service request from a telnet session. The telnet command takes the following format:

```
call <telno> <ani> <url> <refno> [uuidata] [defaults] [parameter_list]
```

For example:

```
call 6796500 65 file://C:/samples/main.vxml 205
```

Table 1 describes these command parameters in more detail.

Table 1: Remote Dialing Command Parameters

Parameter	Description
<telno>	This can be any request URI, but MCP will generate a call through Resource Manager only.
<ani>	Set this to the local SIP URI for MCP. If you enter a number instead, Reporting may return incorrect information.
<url>	Enter the http:// or file:// path of the VoiceXML application.
<refno>	Enter the unique number by which this call will be identified.
[uuidata], [defaults], [parameter_list]	Optional parameters. For more information, see the *Genesys Voice Platform 8.1 User's Guide*.

On receiving the request, MCP loads the specified VoiceXML page and sends an `INVITE` to the Resource Manager. Resource Manager forwards the `INVITE` to SIP Server, which then routes the call to the media gateway, and from there to the destination specified in the request. An RTP media stream is established between the destination and MCP—the customer begins interaction with the VoiceXML page.

For information about configuring remote dialing—including the required semantics for making the request—see the chapter "Enabling Outbound Dialing" in the *Genesys Voice Platform 8.1 User's Guide*.

Outbound Calls Using CCP

Figure 24 shows an outbound call scenario initiated by the Call Control Platform. This scenario is supported by any configuration in which the solution is able to make an outbound call.

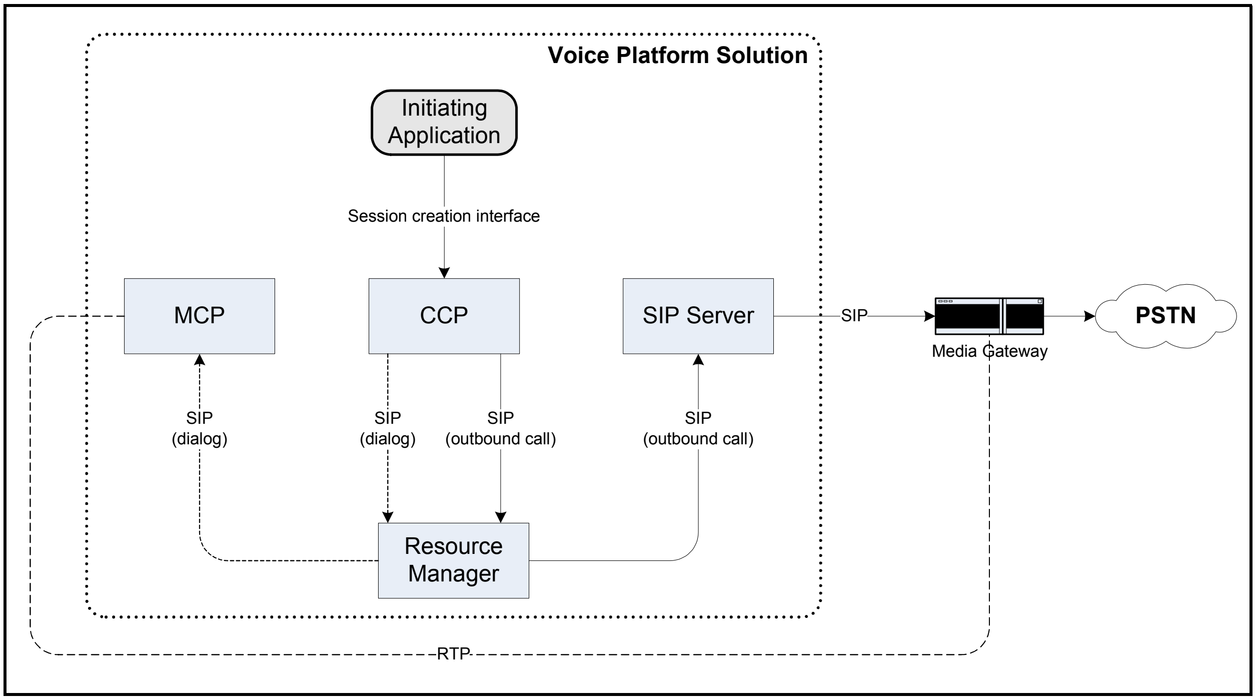


Figure 24: Outbound Call Initiated by the Call Control Platform

In this scenario, an initiating application defines the customer number to be dialed, and then passes this as a parameter through the HTTP interface. The initiating application uses the HTTP session-creation event processor in CCP to start a new CCXML application. This new CCXML application creates the outbound call that connects the destination—the customer—to a VoiceXML application. First, the CCXML application prepares the VoiceXML dialog, and then it starts a new connection to the customer. When the customer answers the call, the CCXML application receives a `connection.connected` event, and can then start the prepared VoiceXML dialog using the `<dialogstart>` tag.

Note: VPS does not support video outbound calls when initiated by a CCXML application; you can only make outbound video calls using the remote dialer service (see See “Outbound Calls Using Remote Dialer Service” on page 59).

Call Progress Analysis

CPA for outbound calls using CCXML is available for Paraxip media gateways only. Using CPA, the media gateway can determine the actual outcome of the call and report that information to the CCXML application. The CCXML application can then determine whether it needs to drop the call—for example, if the call connects to a fax machine—or connect the call to a VoiceXML dialog if a customer voice is detected.

Outbound Call With CPA—Paraxip Media Gateway

For the Paraxip media gateway, a typical CPA call flow is as follows:

1. The session creation event processor starts the CCXML application, which negotiates with the MCP to prepare the VoiceXML dialog.
2. The CCP sends an `INVITE` with the customer destination to the Resource Manager. This `INVITE` includes a private header that the Paraxip media gateway will use to start CPA. In order to insert this header to the `INVITE`, the CCXML must include the following sample code:

```
<var name="hints" expr="new Object()"/>
<assign name="hints.protocol" expr="new Object()"/>
<assign name="hints.protocol.name" expr="'sip'"/>
<assign name="hints.protocol.sip" expr="new Object()"/>
<assign name="hints.protocol.sip.headers" expr="new Array()"/>
<assign name="hints.protocol.sip.headers['X-Detect']"
expr="'Request: cpd=on'"/>
<assign name="hints.deviceprofile" expr="'Paraxip Gateway'"/>
<createcall ... hints="hints"/>
INVITE sip:4161234567@rm SIP/2.0
...
```

   X-Detect: Request: cpd=on
   ...

3. The Resource Manager sends the `INVITE` to SIP Server, which then routes the call to the Paraxip media gateway.
4. The Paraxip media gateway delays sending the `200 OK` message until it analyses the destination.
5. After it gets the CPA result, the Paraxip media gateway sends this information back to CCP (through SIP Server and Resource Manager) in the `200 OK` message.
6. In order to receive the CPA result, the CCXML application must be configured to look for the result in the `connection.connected` event that is sent when the customer answers the call. The following is a sample of the required code:

```
...
<transition event="connection.connected">
  <if cond="event$.protocol.sip.headers['CPD-Result'] == 'Voice'">
    <dialogstart prepareddialogid="dialogid"
connectionid="event$.connectionid"/>
  </if>
<transition>
...
```

Chapter

5 Hierarchical Multi-Tenant Environments

The Voice Platform Solution 8.1 supports hierarchical multi-tenant configurations, which allows you to create levels of tenants, where each Tenant object can be a parent, child, or both. The diagrams and descriptions in this chapter demonstrate some of the more common VPS deployments that make use of Genesys 8.1 hierarchical multi-tenant capability.

About VPS in a Hierarchical Multi-Tenant Deployment

In hierarchical multi-tenant environments, you can create any number of parent/child layers of tenants, where each child tenant inherits the resources (MCP instances, CTI Connector instances, for example) of its parent tenant, while policies (port counting, usage limits, dialing rules, permissions, for example) are enforced top-down in the hierarchy. For example, a service provider, configured as the parent tenant, can create a separate child tenant for each reseller that it services. The parent tenant can enforce specific policies for the child tenants that the parent owns.

One Resource Manager Per Hierarchical Family

In the Voice Platform Solution 8.1, Resource Manager is used to manage the resources and policies required in a hierarchical multi-tenant deployment. To ensure there are no conflicting port counting policies, you can only deploy one Resource Manager within any single hierarchical "family".

For example:

- A Resource Manager instance cannot have any other Resource Manager instance configured on any of its child tenants.
- A Resource Manager instance cannot have any other Resource Manager instance configured on any of its direct parent tenants.

In other words, multiple Resource Managers cannot share the same tenant hierarchy at the same time (one exception is when the Resource Manager instances are deployed separately at different sites, or in separate physical deployment). For examples of correct and incorrect multi-tenant Resource Manager deployments, see Figure 25.

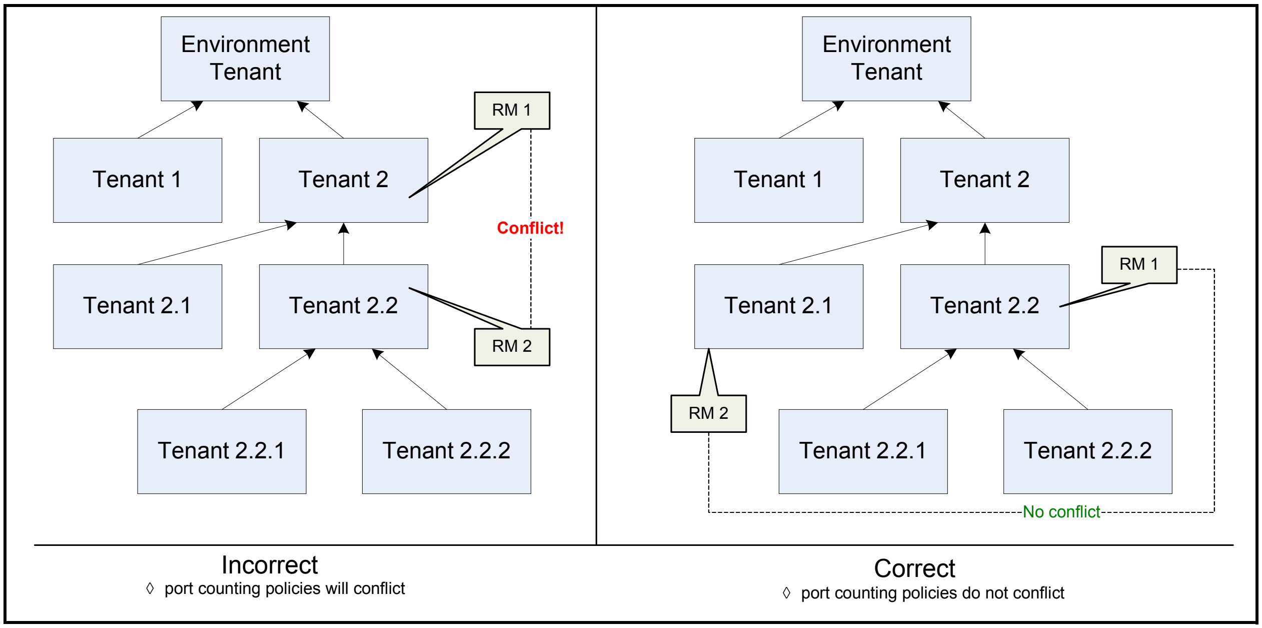


Figure 25: Incorrect and Correct Multi-Tenant Resource Managers

In the incorrect deployment, the parent `Tenant 2` and its child `Tenant 2.2` are both deployed with their own instance of Resource Manager. In this case, the port counting policies for the Resource Manager in the parent tenant will conflict with the policies of the Resource Manager on the child tenant.

In the correct sample deployment, `Tenant 2.1` and `Tenant 2.2` are located at the same level in the hierarchy. In this case, because the two tenants are fully independent of one another, sharing no child tenants, and having no Resource Manager instances located higher up in the hierarchy, they can be deployed with independent instances of Resource Manager.

Multiple Sites Allow For Multiple Resource Managers

You can deploy multiple Resource Managers on a single tenant in order to manage multiple sites. In this case, the port-counting policies are not enforced across the multiple sites—each Resource Manager shares the management of all the child tenants, but maintains independent port counting polices for each site. For an example, see Figure 26.

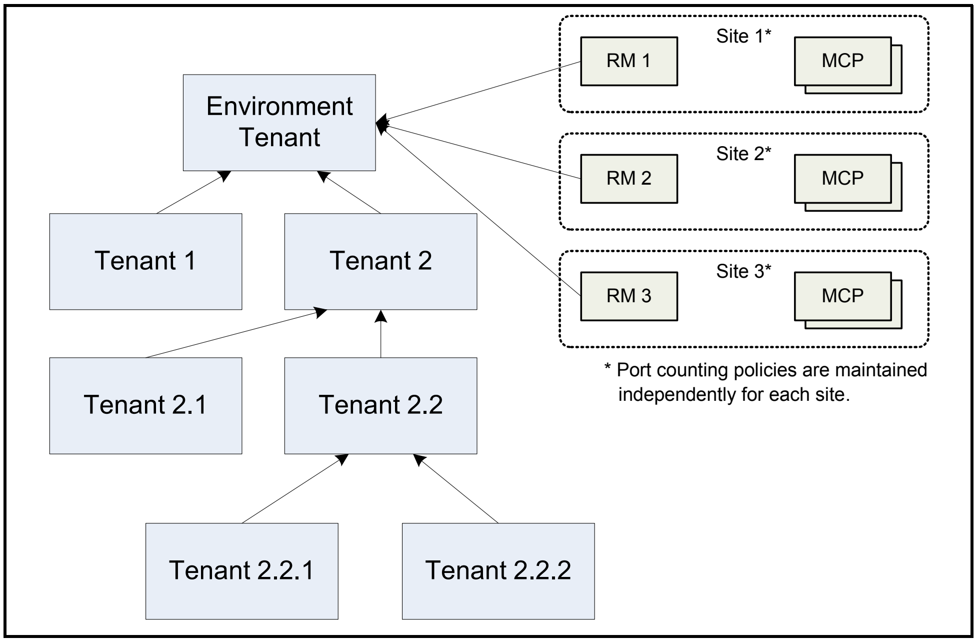


Figure 26: Multi-site on a Single Tenant

In this deployment, all of the Resource Manager instances manage all tenants, since the Resource Manager instances all belong to the `Environment` tenant.

Supported Multi-Tenant Deployments

The following sample architectures describe how hierarchical multi-tenancy can work in a variety of typical supported VPS configurations.

- "Service Provider—Hosted IVR"
- "Service Provider—Hosted IVR With Resellers"
- "Service Provider—Hosted IVR with CTI on a Premise PBX"
- "Enterprise—TDM PBX Configuration with IVR Server"

Service Provider—Hosted IVR

Figure 27 on page 66 shows a CTI through SIP Server deployment, where a service provider maintains two different pools of Media Control Platform (MCP) resources—one common pool available to all child tenants, and an independent pool of resources assigned to a particular child tenant. For example, a preferred customer that needs dedicated resources that are not shared with any other tenants.

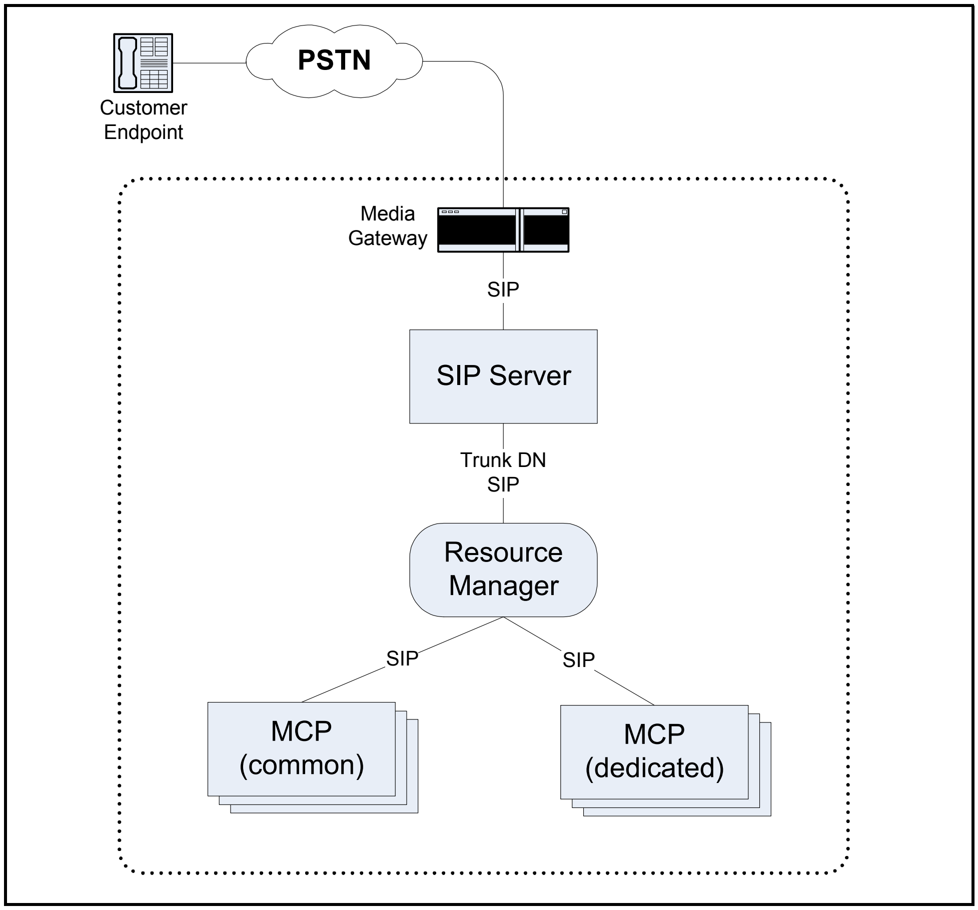


Figure 27: Hosted IVR Deployment

Sample Hierarchy Tree

Figure 28 on page 67 shows a sample hierarchy tree for a hosted IVR deployment.

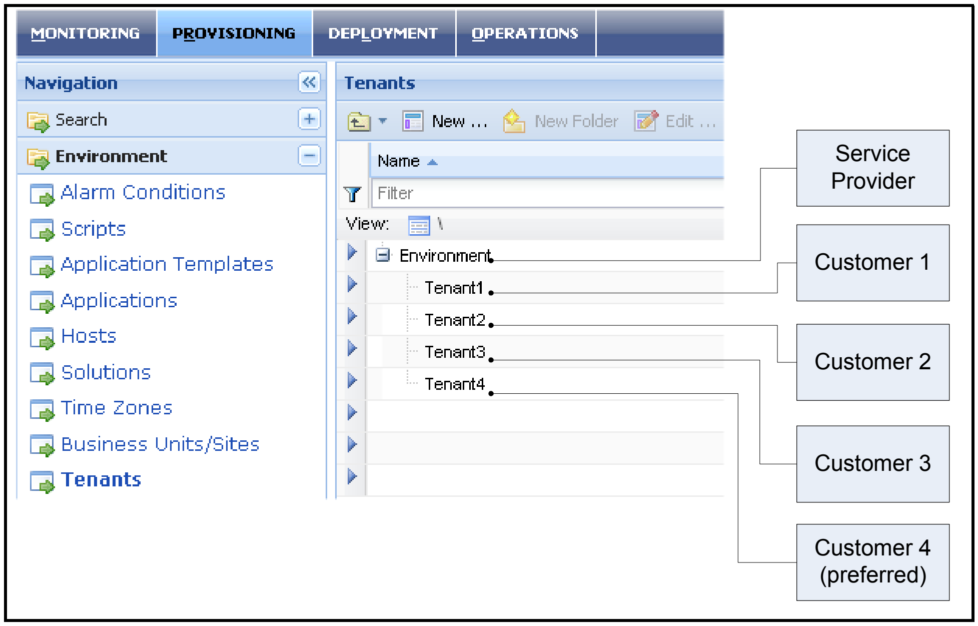


Figure 28: Sample Hierarchy Tree for Hosted IVR Deployment

Sample Configuration

The following table summarizes some sample configuration steps for this kind of deployment, based on the following assumptions:

- The MCP (common) pool of resources hosted on the service provider tenant is shared with all customer tenants.
- The MCP (dedicated) pool of resources is assigned to Customer 4 (preferred) and cannot be shared with any of the other customers.

Task Summary: Sample Hosted IVR Deployment

Objective	Tenant	Recommended Configuration
1. Deploy Resource Manager.	Service Provider	This instance of Resource Manager is used to manage the resources of all child tenants.
2. Configure MCP pools.	Service Provider	Create a logical resource group for each pool of MCPs: • shared MCPs—In the `Resource Group Wizard`, do not assign this resource group to any tenants. • dedicated MCPs—In the wizard, assign this resource group to `Tenant 4`.

Task Summary: Sample Hosted IVR Deployment (Continued)

Objective	Tenant	Recommended Configuration
3. Configure IVR Profiles.	Tenant 1	Create the IVR Profiles for the Voice XML applications required by this customer.
	Tenant 2	Create the IVR Profiles for the Voice XML applications required by this customer.
	Tenant 3	Create the IVR Profiles for the Voice XML applications required by this customer.
	Tenant 4	Create the IVR Profiles for the Voice XML applications required by this customer.

Service Provider—Hosted IVR With Resellers

Figure 29 shows a CTI through SIP Server deployment, where a service provider is the parent tenant for several reseller tenants.

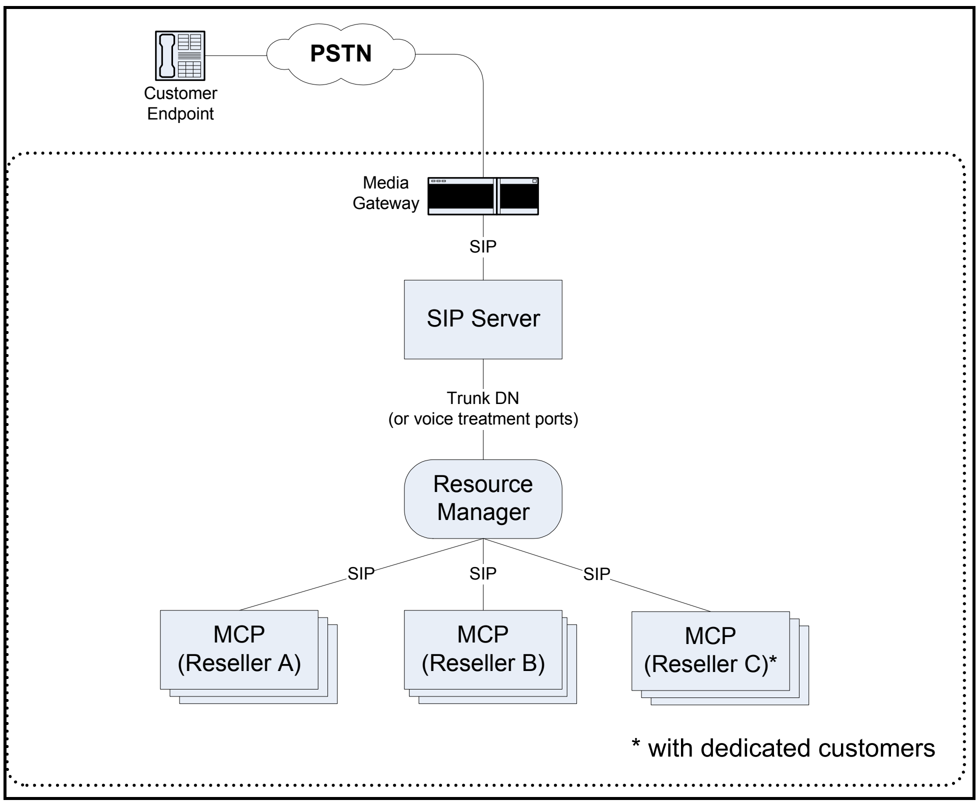


Figure 29: Hosted IVR With Resellers

In this deployment, the service provider physically hosts the Media Control Platform (MCP) applications for all resellers and customers.

Shared Resources For MCP resources that are shared across child tenants—the dedicated reseller tenants, or any other customers in the environment—the service provider tenant creates a logical resource group for those MCP applications to be shared.

Owned Resources Each reseller can also own a dedicated group of MCP resources hosted by the service provider. In this case, the reseller tenant creates a logical resource group, selecting those MCP applications that it will then own. The reseller has administrative rights to these resources, and can assign the resources to customer tenants.

Assigned Resources The reseller can also assign its owned resources to any child tenant underneath it. In this case, you specify which tenant the resource group will be assigned to on the `Tenant Assignment` page of the `Resource Group Wizard`. Only assigned tenants will be able to use these MCP resources.

Sample Hierarchy Tree

Figure 30 shows a sample hierarchy tree for a hosted IVR with resellers deployment.

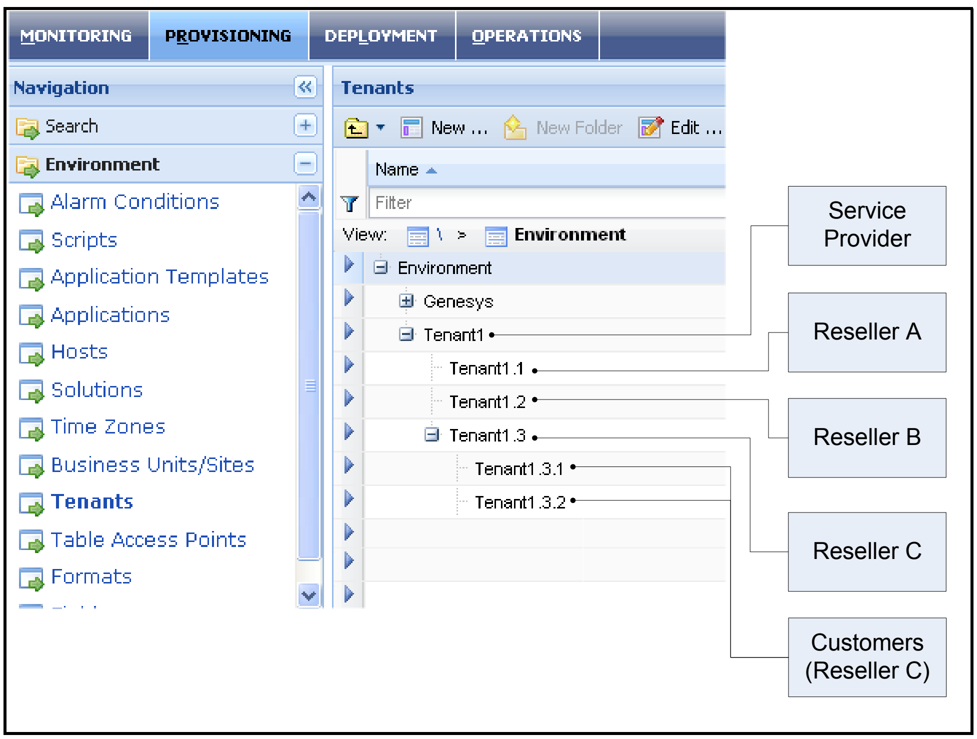


Figure 30: Sample Hierarchy Tree for Hosted IVR With Reseller Deployment

Sample Configuration

The following table summarizes some sample configuration steps for this kind of deployment, based on the following assumptions:

- Each reseller owns its own pool of MCP resources hosted by the service provider.
- Reseller C assigns its MCP resources to specific customer tenants. These resources are not shared across customers, but remain dedicated to the assigned customer. **Note:** reseller cannot dedicate resources to itself.

Task Summary: Sample Hosted IVR with Resellers Deployment

Objective	Tenant	Recommended Configuration
1. Deploy Resource Manager.	Tenant 1	This instance of Resource Manager is used to manage the resources of all child tenants.
2. Deploy MCPs.	Tenant 1	Deploy the MCP applications to be used by the resellers.
3. Group MCPs per ownership.	Tenant 1.1	Create an MCP resource group for the MCP instances to be used by Reseller A.
	Tenant 1.2	Create an MCP resource group for the MCP instances to be used by Reseller B.
4. Assign MCPs.	Tenant 1.3	1. Create two MCP resource groups, one for each customer tenant that Reseller C will service. For example: • RG_1 • RG_2 2. Assign each MCP resource group to the customer tenant: • In the `Resource Group Wizard` for RG_1, assign the group to `Tenant 1.3.1`. • In the wizard for RG_2, assign the group to `Tenant 1.3.2`.
5. Configure IVR Profiles.	Tenant 1.3.1	Create the IVR Profiles for the Voice XML applications required by this customer.
	Tenant 1.3.2	Create the IVR Profiles for the Voice XML applications required by this customer.

Service Provider—Hosted IVR with CTI on a Premise PBX

Figure 31 shows a hosted IVR environment where a single pool of MCPs deployed on the service provider site are then shared across several different customer sites. In this case, the service provider and customer sites can be deployed on separate instances of Management Framework. CTI is provided to the customer premise PBX through communication between the CTI Connectors on the service provider site and the IVR Servers on the customer sites.

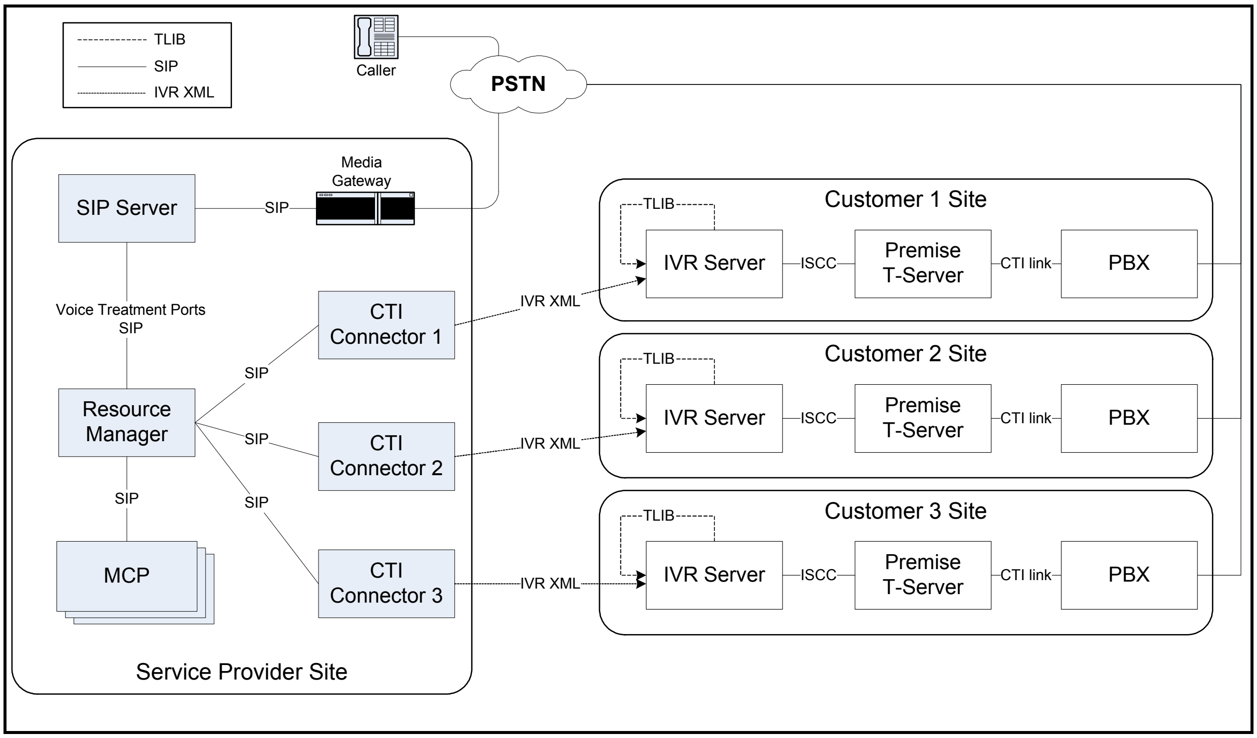


Figure 31: Hosted IVR with CTI on a Premise PBX

Sample Hierarchy Tree

Figure 32 on page 72 shows a sample hierarchy tree for a hosted IVR with CTI on premise PBX deployment.

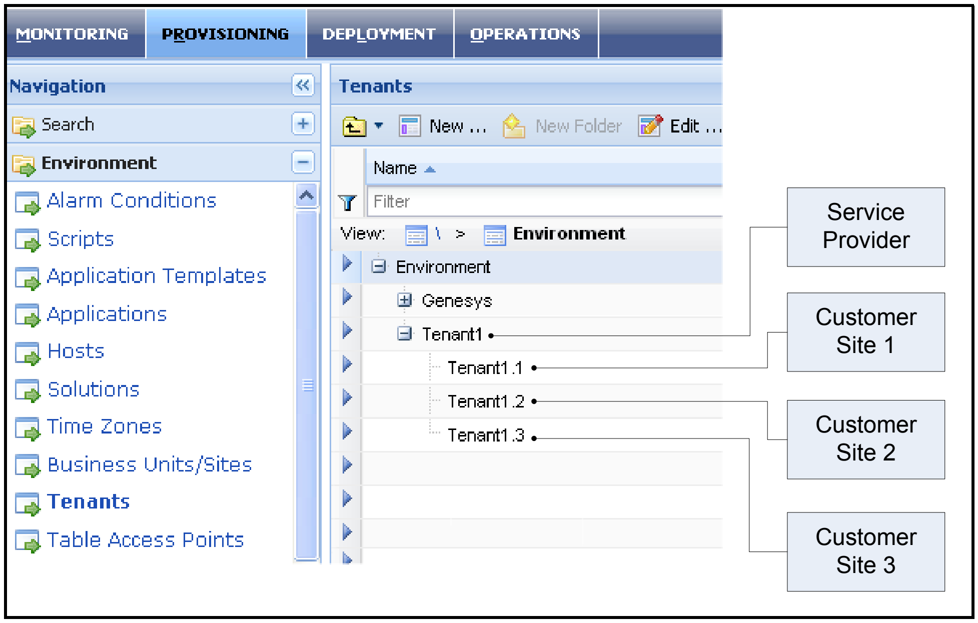


Figure 32: Sample Hierarchy for Hosted IVR with CTI on Premise PBX

Sample Configuration

The following table summarizes some sample configuration steps for this kind of deployment, based on the following assumptions:

- One group of MCP resources will be shared by all customer sites.
- No customer site will own any MCP resources, or be assigned any resources.

Task Summary: Hosted IVR with CTI on PremisePBX

Objective	Tenant	Recommended Configuration
1. Deploy Resource Manager.	Tenant 1	This instance of Resource Manager is used to manage the resources of all child tenants.
2. Configure gateway	Tenant 1	1. Create a gateway resource group for SIP Server. 2. Set `CTI Usage` to `Based on DN Lookup` (`use-cti=2`).
3. Configure MCPs.	Tenant 1	1. Deploy the MCP applications to be used by the customers. 2. Create an MCP resource group (Voice XML) to be shared by all customers.

Task Summary: Hosted IVR with CTI on PremisePBX (Continued)

Objective	Tenant	Recommended Configuration
4. Configure CTI Connectors.	Tenant 1	1. Create a CTI Connector resource group for each child tenant: • Tenant 1.1 • Tenant 1.2 • Tenant 1.3 2. Deploy the CTI Connector instance for each of these resource groups.
5. Configure IVR Profiles.	Tenant 1.1 Tenant 1.2 Tenant 1.3	1. On each customer tenant, create the IVR Profiles for the Voice XML applications required by this customer. 2. Set `CTI Allowed` in the IVR Profile to `true`.

Enterprise—TDM PBX Configuration with IVR Server

Figure 33 shows an enterprise environment where different business groups are configured as separate tenants, for the purposes of self-service applications.

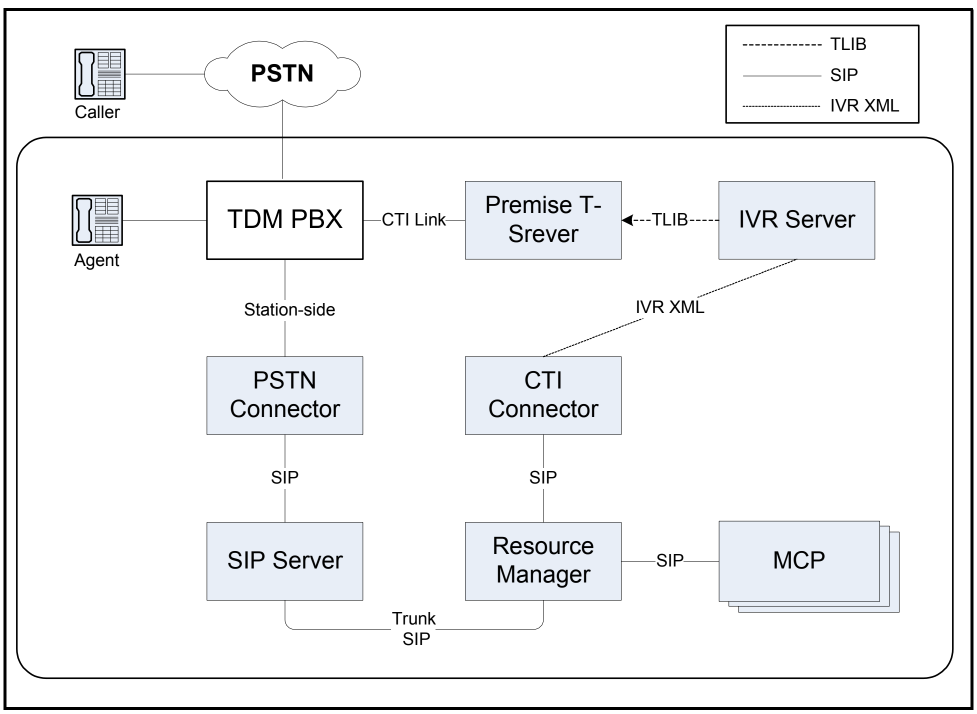


Figure 33: TDM PBX Configuration with IVR Server

In this deployment, all tenants share the same TDM PBX (the premise T-Server itself only has a single tenant). The VPS connects to the PBX—

through the PSTN Connector—using a station-side connection, where only the port number is provided in the incoming INVITE request, with no DNIS available. To get the DNIS, Resource Manager sends the call to CTI Connector, deferring the decision about tenant hierarchy until the IVR Server returns the DNIS information. CTI Connector forwards the request and DNIS back to Resource Manager, which can then determine the tenant hierarchy from the IVR profile.

Note: Hierarchical multi-tenancy can be created within GVP and is separate from T-Server tenancy. In this scenario, for example, the premise T-Server itself has only a single tenant. For GVP, tenant selection is deferred until after initial call processing by CTI Connector and IVR Server (register the call and obtain the actual DNIS) is finished.

Sample Hierarchy Tree

Figure 34 shows a sample hierarchy tree for an Enterprise TDM PBX deployment with multiple business group tenants.

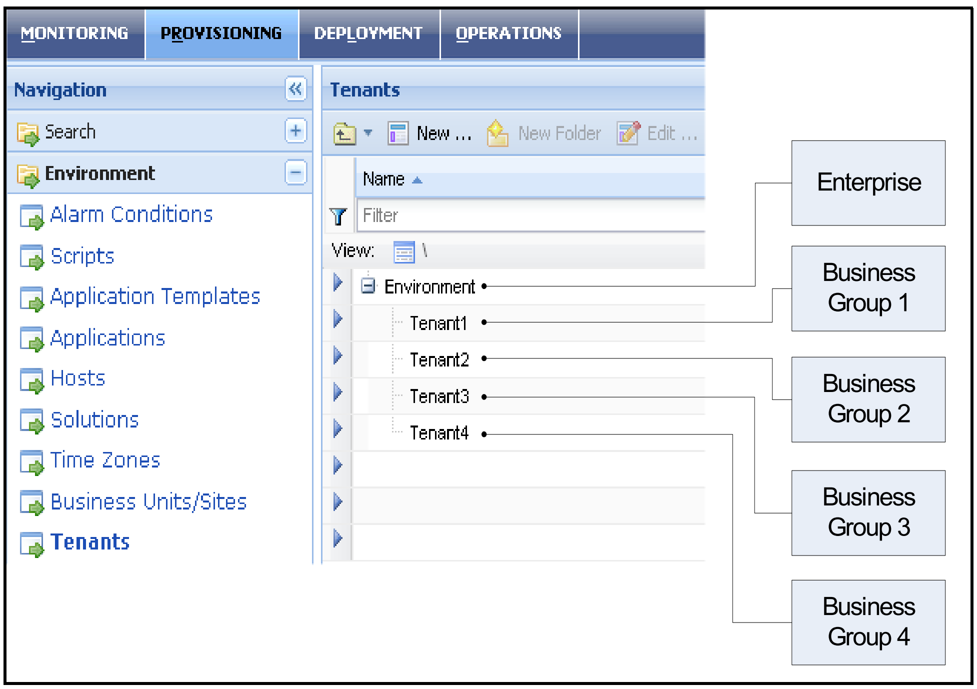


Figure 34: Sample Hierarchy for Enterprise TDM PBX With Business Groups

Sample Configuration

The following table summarizes some sample configuration steps for this kind of deployment, based on the following assumption:

- All Business Groups share a common pool of MCP resources hosted by the Environment tenant.

Task Summary: Sample Hosted IVR with CTI on Premise PBX

Objective	Tenant	Recommended Configuration
1. Deploy Resource Manager.	Environment	This instance of Resource Manager is used to manage the resources of all child tenants.
2. Configure gateway	Environment	1. Create a gateway resource group for SIP Server. 2. Set `CTI Usage` to `Based on DN Lookup` (`use-cti=2`).
3. Configure CTI Connectors.	Environment	1. Create a CTI Connector resource group. 2. Deploy the CTI Connector instance for this resource group.
4. Configure MCPs.	Environment	1. Deploy the MCP applications to be used by the business groups. 2. Create an MCP resource group to be shared by all business groups.
5. Configure IVR Profiles.	Tenant 1	1. On each business group tenant, create the IVR Profiles for the Voice XML applications required by this group. 2. If CTI is required by the application, set `CTI Allowed` in the IVR Profile to `true`.
	Tenant 2	
	Tenant 3	
	Tenant 4	

Chapter

6 High Availability

High availability (HA) ensures that a service is not interrupted in the event of a failure or a process restart. High availability is configured at the component level. This chapter provides an overview of supported HA methods, a description of how HA works for the different solution interfaces, as well as details about how a failover takes places for variously scaled solutions.

Note: This chapter describes HA configuration requirements at a high-level. For detailed configuration procedures, or for more information about how HA works at the component level, consult the respective product *Deployment Guides*.

This chapter contains the following sections:

About HA in the Voice Platform Solution

At the solution level, HA for the various components can help ensure that the following communication interfaces remain viable in the event of component failure:

- SIP/RTP communication—Active/standby SIP Server pairs deployed using Windows Network Load Balancing (NLB) clusters, virtual IPs, DNS-based distribution, or client-side intelligence can ensure availability of the external SIP interface.

GVP 8.1.x components and interfaces use different methods of availability and failure recovery, and some examples are:

- Load balancing is used to distribute interactions across multiple resources, providing a means of scaling capacity for a given type of resource, and for providing access to more resources (one kind of resource grouping) than might be needed in case one resource fails—for example, (N+1) designs.

- CTIC ‘load balancing’ can use a round-robin selection of IVR Servers if more than one is available in a primary group, and failure spillover to a backup group if the primary group fails to answer.
- Some HA solutions use purely a primary and backup pair; if one fails the other can be manually brought up (standby) or automatically brought up (hot standby) to provide high availability. The MRCP Proxy (GVP 8.1.6) is an example of a hot standby solution.
- The Resource Manager has its own load balancing and backup failover mechanism for a group of Media Servers (MCP). Resource Manager is deployed in pairs. It can be configured as a standalone Primary, an active Primary/standby Backup, or uniquely within GVP/GMS, an active-active pair where both RMs are available, communicate with each other (redundant), and are used in a round-robin manner (through a load balancer).

Solution Level Components and Interfaces

At the solution level, HA and failure recovery for the various components can help ensure that the following platform elements and their communication interfaces remain viable in the event of component failure.

Configuration, Reporting and Development Tools

Genesys Administrator (GA) is a web-based interface that allows the solution owner to configure the various resources and application objects associated with solution components, addressing, and some quality reporting. The configuration data is stored in Genesys Management Framework Configuration Server. Please see the *Framework Deployment Guide* for descriptions of how Configuration Server is deployed in HA modes. GA comes with the Genesys Voice Platform.

Genesys Administrator Extended (GAX) is a web-based interface, separate from GA, allowing solution owners and their customers access to reports using data available from the GMS/GVP Reporting Server. GAX fetches this information as needed via a reporting interface request from the Reporting Server (RS) that is active at the time. GAX comes with Genesys Voice Platform. GAX will try the primary RS first and then backup RS. GAX will remember the last RS that was tried and try that again for the next request, i.e., if the primary is down and GAX does use the backup RS, GAX will try the backup RS to get a report until the backup RS fails.

Composer is a Genesys provided VoiceXML development and visual design tool which comes with Genesys Voice Platform (GVP). Composer operates as a thick client usually on a workstation which is not a part of the production GVP call processing solution environment. Composer can communicate with the production system however. Please see the *Composer Deployment Guide* for configuration options where Composer can permit testing of VoiceXML applications with web services, Media Control Platform (MCP) components.

Core Components

SIP Server

SIP Server can be deployed in HA mode and several groups of SIP Servers can be front-ended by load balancers to divide incoming voice traffic across SIP Servers at the same or different sites. Active/standby SIP Server pairs deployed using Windows Network Load Balancing (NLB) clusters, F5, virtual IP solutions, DNS-based distribution, or client-side intelligence can ensure availability of the external SIP interface. See the *SIP Server Deployment Guide* for details.

Resource Manager

Resource Manager (RM) communicates bi-directionally with SIP Server, the Media Control Platform (MCP), the Call Control Platform (CCP if used), CTI-C clients. RM reads from Policy Server, and writes to the Reporting Server.

Resource Manager supports both active/standby and active/active clustering of RM pairs, depending on the HA solution selected. SIP Server communication with RM may require either a third party load balancing solution (NLB, F5) or a Virtual IP takeover solution. See the *GVP Deployment Guide* Appendices for RM HA configuration details.

- RM can access both primary and backup MCP resource groups, if configured. RM selects the next MCP in a group, round robin, upon each MSML or SIP Invite request.
- RM can access both Primary and Backup CTI-C clients and/or CTI-C groups if configured.
- RM can access both Primary and Backup Reporting Servers. If a single RS is used, and is down, RM can use either a memory-based or external (optional) DB to buffer logs, and update RS when RS is restored.
- RM generally reads from Policy Server only once at start-up unless alerted to refresh.

Media Control Platform and Call Control Platforms

Multiple independent instances of MCP and CCP (N+1) are used to ensure load support.

MCP/CCP communicates bi-directionally with Web Applications Servers and Resource Manager, and writes to the Reporting Server. The MCP communicates bi-directionally with the MRCP Proxy server if configured or

MRCP speech servers otherwise if speech is used, to the PSTN Connector if deployed, and to T-Library via RM/SIP Server SIP messaging.

- MCP/CCP will work in an N+1 Web Application Server configuration, and a third party, customer selected, load-balancer may be used between the MCP and web servers.
- MCP/CCP will buffer logs if the Reporting Server is down similar to the methods used by RM as discussed above.
- If a primary speech server is down, MCP can look for a secondary speech server. However, it is unlikely that a deployment would have two MRCP speech servers (e.g. Nuance NSS) deployed.
- If a speech resource is not available, GVP will return an event message to the application for application recovery and control, as well as send an alarm to Genesys Management Framework SCS monitoring.

MRCP Proxy Server

The MRCP Proxy /Server communications with the MCP and writes to the Reporting Server. The MRCP Proxy can be configured as primary/backup in warm-standby (GVP 8.1.5 or later) or hot-standby (GVP 8.1.6 or later) modes for MRCP Proxy HA. MRCP proxy is required in deployments where statistics on peak ASR and TTS use are required including breakdown of use by tenant and application.

If RS is down, the MRCP Proxy Server will use the same methods as RM to buffer logs as explained above.

MRCP Speech Services

Nuance depends on GVP to provide MRCP HA. If GVP is directly communicating with the MRCP Speech Servers (rare now) it can place each server in a primary and back-up group. If GVP is communicating to the Nuance NSS MRCP Server, then the NSS might deploy an N+1 strategy for speech servers to achieve HA. To support the MRCP Proxy in HA mode, the latest versions of Management Framework and LCA must be installed and the Solution Control Server (SCS) application configured to support HA licenses.

Reporting Server

Active/standby Reporting Server can be configured in HA. In GVP 8.1.6 and later, RS can be configured in Hot Standby. For more information on Reporting Server HA, please see the *GVP Deployment Guide* Appendix for Reporting Server HA.

CTI Support

GVP has two major methods of CTI support: SIP messaging via T-Server Library or CTI messaging via Genesys IVR Server using a CTI-C client. A

T-Library interface is used by Genesys Universal Routing Server for route requests and sharing call-related 'attached' data.

- For T-Library communication involving IVR Server (in CTI through IVR Server configurations), IVR Server can also be configured as primary and secondary pairs to provide a highly available connection to the T-Server. GVP does NOT support IVR Server 8 HA, but rather can address multiple instances of IVR Server as primary and secondary components in a failover solution. Effectively GVP CTI-C clients load balance across IVR Server instances within a CTI group in round robin fashion.
- When using SIP CTI mode, GVP is assured HA access to T-Server via SIP Server pairs that maintain the HA solution.

SSG Service Interface

The Supplementary Services Gateway (SSG), which manages this interface, can not be deployed in high availability configurations. However, each instance of the SSG manages an independent queue for outbound calls, providing persistence for this queue that can survive application failures. In case of failure, the queue continues after the application restarts.

Third Party Access to the Reporting Server {non Genesys}

If a customer wants to read CDR and VAR information from the Reporting Server, Genesys provides a Web Services API for this purpose. Direct SQL access, while possible, is not supported by Genesys with rare exception.

- If RS fails to respond, the third party entity must have its own recovery plan.

RTSP Streaming Server (Non-Genesys)

The GVP/GMS media services will support access and use of third party RTSP Streaming media for audio and video. However, this is not HA, and if the RTSP Streaming server fails to respond, the MCP will send a failure message to the requesting VoiceXML application to handle.

Depending on the particular configuration or scale of your solution, high availability has different capabilities, or in some instances, limitations as to what services can survive a component failure. The priority is on ensuring that all new calls are processed.

HA In Scaled Deployment

For an overview of how different solution configurations at various scales can respond to failures of any of the individual components, see the following sections:

- "All-In-One Configuration"

- “Small Scale Enterprise”
- “Medium Scale Enterprise”

All-In-One Configuration

The All-In-One Platform provides a complete voice self-service solution on a single physical server. All software components that make up the Voice Platform Solution are installed on the same physical server, including any speech engines required for speech services—Text to Speech (TTS) and Automatic Speech Recognition (ASR). The VoiceXML applications, and the web server that services them, can be deployed on a separate server. In CTI configurations, URS and Stat Server (and any other routing/CTI components) can be deployed on a separate server.

This configuration does not provide any high availability—the failure of a single component essentially causes all calls to fail.

Figure 35 shows the distribution of components in an All-In-One Voice Platform Solution configuration.

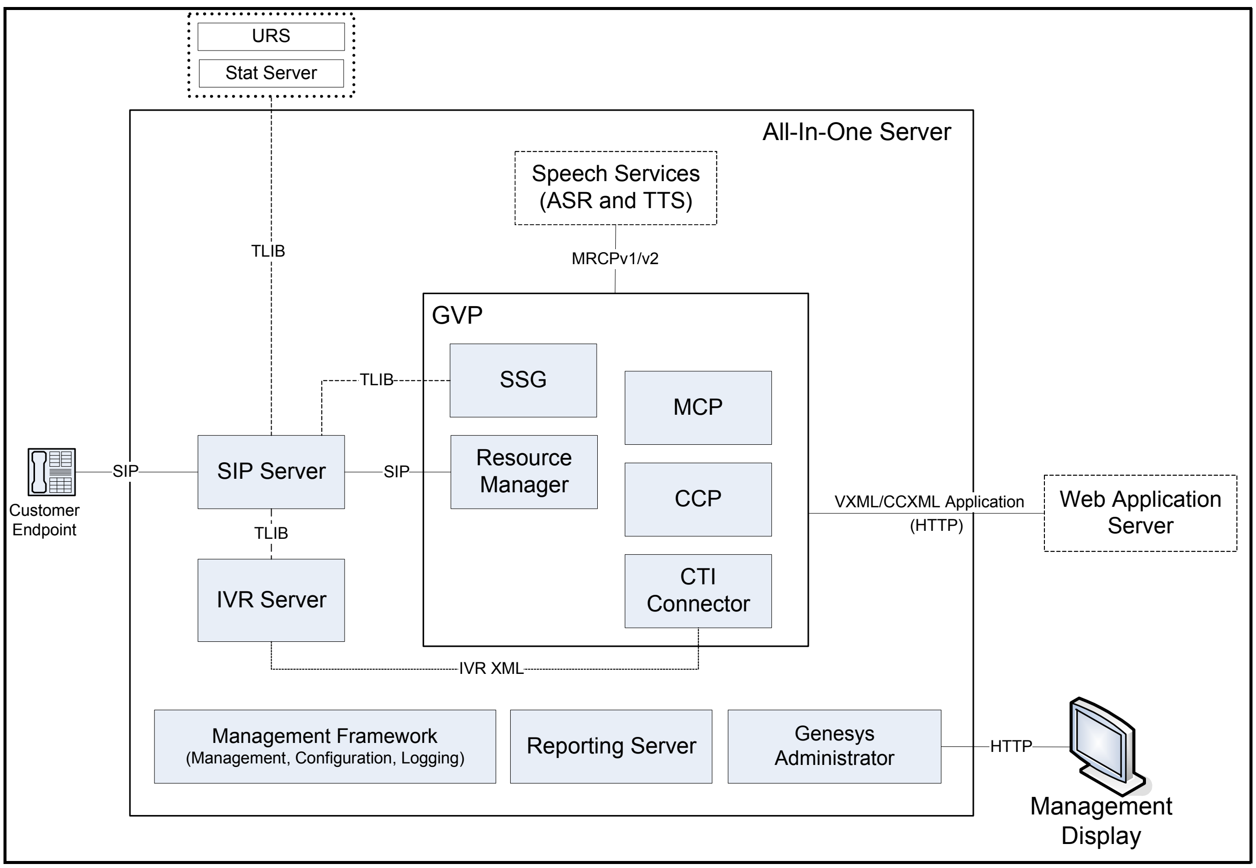


Figure 35: All-In-One Server Configuration

Table 2 on page 83 describes how the Voice Platform Solution responds to different failure scenarios.

Table 2: HA Response to Component Failure—All-In-One Scenario

Failure	Solution Response
Machine fails.	All calls are lost. The solution cannot accept new incoming calls until the machine restarts.
SIP Server fails.	Because SIP Server is a third-party call control (3pcc) component, its failure does not immediately affect all outstanding voice calls until a call control request is made (for example, a transfer). The remote SIP endpoint will eventually terminate the call after the session timeout expires. The solution cannot accept new incoming calls or make outbound calls until SIP Server is restarted.
Resource Manager fails.	Because Resource Manager is a SIP proxy, its failure does not affect ongoing voice calls—the voice dialog can continue, and call control requests can be initiated after the Resource Manager is restarted. After the restart, resource tracking is reset, and so the Resource Manager will not have an accurate view of current resource usage until all calls started before the failure have been terminated.
MCP fails.	All voice calls are terminated. No new calls are processed until after MCP restarts.
CCP fails.	Because CCP is a 3pcc component, its failure does not affect ongoing voice calls until a call control request is required. As in a SIP Server failure, the other SIP endpoint will eventually terminate the call after the session timeout expires.
IVR Server fails.	The failure of IVR Server does not affect ongoing voice calls until a CTI function is requested. CTI functions in this case will fail—however, the voice application may recover from the failure, keeping the caller on the line for continued voice self-service. New calls will not have CTI functionality.
CTI Connector fails.	Because CTI-C is a 3pcc component, its failure does not affect ongoing voice calls until the application requests a CTI function, or attempts a call transfer. At this point, SIP transactions for the call will fail and the call will be terminated. New calls will not have CTI functionality.
SSG fails	Outbound call requests already queued in the SSG can survive a failure of the application—after restart, SSG will continue with the same queue. Calls placed in queue but not reported to the client application (and whose TimeToLive (TTL) timer has not expired) are still considered to be in the queue, and will also be retried after process restart.

Small Scale Enterprise

The Small Scale Enterprise configuration provides an off-board speech server in order to offload the most resource intensive process in the solution to a separate server from the one used for the rest of the Voice Platform Solution components. This configuration is similar to the All-In-One configuration, except for the off-board speech server. Like the All-In-One configuration, the Small Scale Enterprise configuration does not offer high availability to protect against component failures. In this case, the solution responds as it does in an All-In-One configuration (see Table 2 on page 83).

Figure 36 shows the distribution of components in a Small Scale Enterprise Voice Platform Solution configuration.

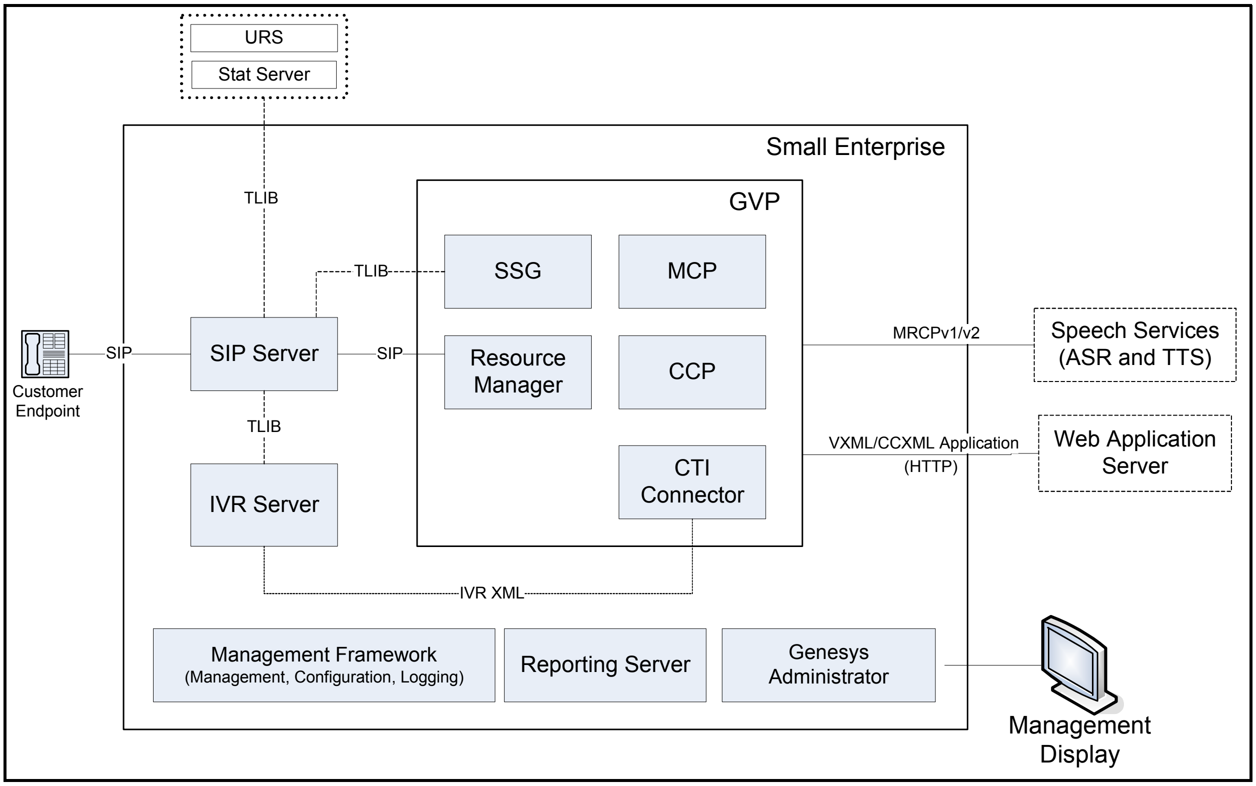


Figure 36: Small Scale Enterprise Configuration

Medium Scale Enterprise

In a Medium Scale Enterprise configuration, the Voice Platform Solution components are distributed across multiple servers, providing load-balancing for resource intensive services, as well as high availability for communication interfaces. Figure 37 on page 85 shows a typical distribution of components for a sample Medium Scale Enterprise configuration.

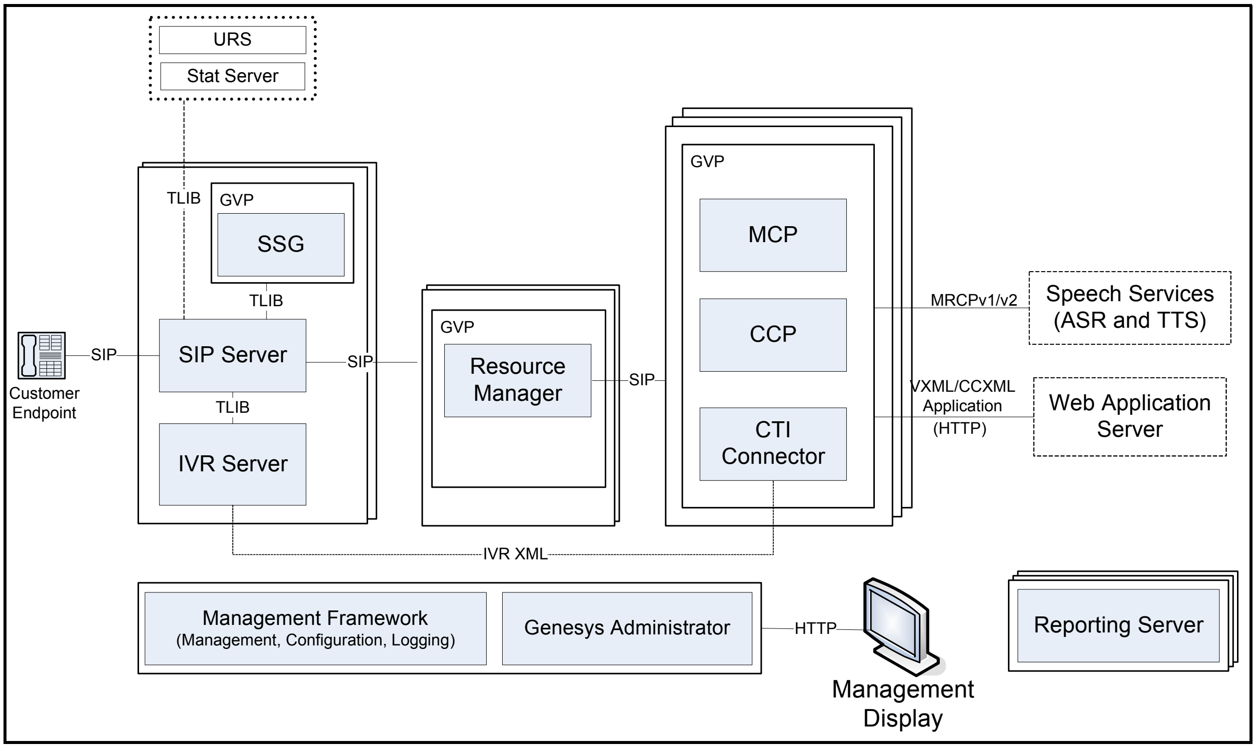


Figure 37: Medium Scale Enterprise Configuration

High Availability for SIP Server

In the Medium Scale Enterprise deployment shown in Figure 37, high availability is configured using Windows Network Load Balancing clusters of primary/backup SIP Server pairs, deployed on separate physical servers. These SIP Server pairs will synchronize call states to ensure that the backup SIP Server can take over if the primary instance fails. NLB provides a single IP address for both the primary and backup SIP Servers, so that if the primary SIP Server fails, the backup can resume service using the same IP address that was used before the failover. The Universal Routing Server, if deployed, connects to both SIP Server instances as a T-Library client, which enables the URS to switch over to the backup SIP Server if the primary instance fails.

For more information about configuring high availability for SIP Server, consult the *Framework 8.1 SIP Server Deployment Guide*.

High Availability for Resource Manager

High availability for Resource Manger, as for SIP Server, is configured using Windows Network Load Balancing clusters of primary/backup Resource Manager pairs, deployed on separate physical servers. NLB provides a single IP address for both primary and backup Resource Managers, so that the other SIP endpoints—including SIP Server—can continue communicating with the same IP address after a failover. Because Resource Manager is a SIP proxy, call state synchronization between primary and backup instances is not required. However, the failover from primary to backup Resource Manager resets resource tracking, and so Resource Manager will not have an accurate view of current resource usage until all calls started before the failure have been terminated.

For information about configuring active/standby Resource Manager pairs for high availability, see "NLB Clustering for Resource Manager" in the *Genesys Voice Platform 8.1 Deployment Guide*.

High Availability for MCP/CCP

For a Medium Scale Enterprise configuration, the number of ports typically required by the solution outnumbers the port capacity for a single MCP/CCP, and so multiple instances of these components are deployed across several servers. The Resource Manager performs load balancing of the incoming calls for even distribution across the multiple servers. Because of how VoiceXML and CCXML applications work, neither MCP or CCP is fault tolerant. If a failure occurs at any one instance of MCP/CCP, all outstanding calls at that instance will fail. However, because Resource Manager monitors the status of each resource continuously, it will forward any new incoming call to one of the remaining instances; it will not attempt to forward an incoming call to a failed server.

High Availability for Reporting Server

High availability for the Reporting Server is also configured using primary/backup pairs on separate servers. If the primary Reporting Server fails, the backup can take over, enabling the continuous reporting of data to the database.

Solution Response to a Component Failure

Table 3 describes how the Voice Platform Solution responds to different failure scenarios.

Table 3: HA Response to Component Failure—Medium Scale Enterprise

Failure	Solution Response
SIP Server (or its machine) fails.	The backup SIP Server takes over, using the shared IP address. Calls in an unstable state will fail, but stable calls—those connected to the VoiceXML application or an agent—will continue to operate, with no impact on the customer experience.

Table 3: HA Response to Component Failure—Medium Scale Enterprise (Continued)

Failure	Solution Response
Resource Manager (or its machine) fails.	The backup Resource Manager takes over, using the shared IP address. Ongoing SIP transactions will fail, but this affects only incoming calls currently in the ringing state. All other calls remain unaffected, with no impact on the customer experience. Because resource tracking is reset after a failover, Resource Manager will not have an accurate view current resource usage until all calls started before the failure have been terminated
MCP fails.	All voice calls on the failed server will be terminated. The customer will hear silence from the IVR and will presumably hang up the call. If the customer does not hang up, SIP Server will eventually timeout and terminate the call. Resource Manager tracks this resource as unavailable, and does not forward any new calls to this instance—resulting in a temporary reduction in solution capacity. The LCA on the host machine should automatically restart the MCP, at which point solution capacity returns to normal levels.
CCP fails.	Because CCP is a 3pcc component, its failure does not affect ongoing voice calls until a call control request is required. As in a SIP Server failure, the other SIP endpoint—MCP or SIP Server—will eventually terminate the call after its session timeout expires. As in an MCP failure, the Resource Manager detects the failure and forwards new calls to the remaining instances. Any temporary reduction in solution capacity is remedied after the LCA automatically restarts the CCP.
The GVP server (machine with MCP and CCP) fails.	The response is identical to the response described for MCP and CCP failures, except both failures and responses occur at the same time.
IVR Server fails.	IVR Server failures affect only the calls processed by that particular instance. Voice calls remain unaffected until a CTI function is requested. Even if the CTI function fails, the voice application may be able to recover from the failure, keeping the caller connected for continued voice self-service. New incoming calls are forwarded to the backup IVR Server instance, where CTI functions are still available.
CTI Connector fails.	CTI Connector failures affect only the calls processed by that particular instance. Because CTI-C is a 3pcc component, its failure does not affect ongoing voice calls until the application requests a CTI function or attempts a call transfer. At this point, SIP transactions for the call will fail and the call will be terminated. New incoming calls are forwarded to a secondary CTI Connector instance, where CTI functions are still available.
Reporting Server fails.	The backup Reporting Server takes over as the active instance in the pair; all reporting data will be processed by this instance.

Table 3: HA Response to Component Failure—Medium Scale Enterprise (Continued)

Failure	Solution Response
SSG fails	Outbound call requests already queued in the SSG can survive a failure of the application—after restart, SSG will continue with the same queue. Calls placed in queue but not reported to the client application (and whose TimeToLive (TTL) timer has not expired) are still considered to be in the queue, and will also be retried after process restart.

Genesys

Chapter

7 Support for IVR Server

For deployments that require the use of IVR Server, Genesys offers the CTI Connector (CTI-C), a GVP 8.1 component that, together with the IVR Server, provides the CTI required to integrate the Voice Platform Solution (VPS) with the larger Genesys suite, in cases where the standard SIP interface cannot provide CTI itself.

This chapter includes the following sections:

About the CTI Connector

CTI Connector is used for the integration of legacy CTI-enabled VoiceXML/TXML applications. It is also used in some configurations where IVR Server is required to integrate CTI with the voice application. CTI Connector is compatible with both NGI and GVPi applications—though Genesys recommends building new applications using NGI.

Using CTI Connector, the Voice Platform Solution is able to integrate with the IVR Server in different ways, depending on the role that the IVR Server plays in the deployment. The VPS supports integration with IVR Server configured for the following modes:

- Behind mode (see page 90)
- In-Front mode (see page 94)
- Network mode (see page 96)

Integration with IVR Server—Behind Mode

In this deployment, IVR Server is configured for Behind mode. In a TDM PBX integration, this mode refers to the physical location of the IVR Server behind the switch. In an IP-based integration, the mode refers to the configured or logical relationship of the IVR Server to the switch.

Depending on the deployment, the inbound call reaches either a `Routing Point` DN (IP-based deployment) or a `Trunk` DN (TDM deployments) on the SIP Server.

Behind-mode integrations are supported in the following deployments:

- "Carrier-Connected Deployments"
- "PBX-Connected Deployments"

Carrier-Connected Deployments

Figure 38 illustrates how the VPS, in a carrier-connected integration with IVR Server in Behind mode, handles call setup for a typical inbound call, where a legacy IVR application (GVPi) is triggered.

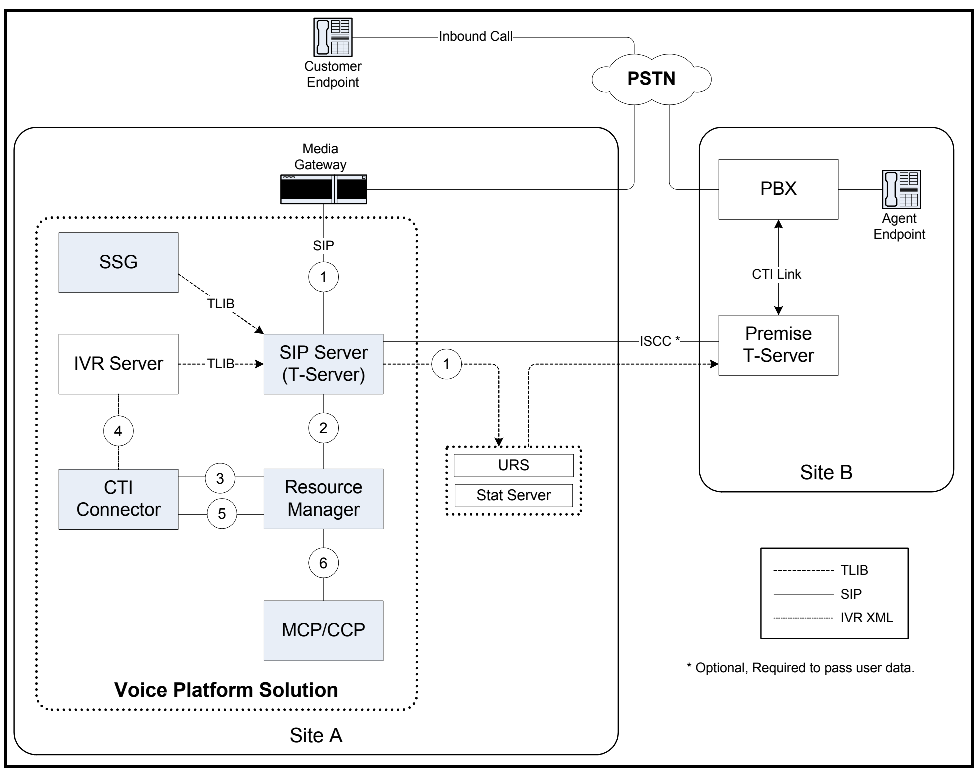


Figure 38: Call Setup for Typical Inbound Call—Behind Mode

1. A call comes in from the PSTN, through a media gateway, to a `Routing Point` DN on the SIP Server. The routing strategy selects a `Place Group`, and SIP Server picks an available `Voice Treatment Port` DN from a `Place` in the `Place Group`.

2. The SIP Server forwards the port number (as the user part of the `Request-URI` and in the `To` header) in the `INVITE` it sends to Resource Manager.
3. The Resource Manager creates a session ID, and then forwards this ID and the IVR port in an `INVITE` to CTI Connector.
4. The CTI Connector sends the IVR Server an XML request to register the new call with SIP Server. This anchors the call on the CTI Connector, so that CTI functions can be executed directly from the application.
5. After obtaining the response from the IVR Server, the CTI Connector sends an `INVITE` to Resource Manager with the DNIS in the `To` header.
6. The Resource Manager matches the DNIS with an IVR Profile, and then sends an `INVITE` to the Media Control Platform with the URL to launch the VoiceXML application.
7. The Real-time Transport Protocol (RTP) media path is established between the Media Control Platform and the gateway and through to the caller.

The caller is now interacting with the voice application. For information about call transfers from the voice self-service application to an available agent for assisted-service, see "Call Transfers Using CTI Connector" on page 97.

Configuration Overview

The relevant configuration steps required for a carrier-connected deployment include the following:

1. Create `Place` objects for each `Voice Treatment Port` DN in the SIP Server switch, then add all the Places to a Place Group. The routing strategy needs to target this Place Group.
2. Create `Voice Treatment Port` DNs in the dummy switch for `TServer_IVR` to match the DNs in the SIP Server switch. These DNs do not require any configuration.
3. Configure IVR Ports. The numbers for these ports must match the `Voice Treatment Port` DNs in the dummy switch, and also follow the same range.
4. Go to the SIP Server `Application` object, add connections to:
   - Message Server
5. In the IVR Server `Application` object, add a connection to `TServer_IVR`.
6. Go to `Provisioning > Voice Platform > Resource Groups`, and use the wizard to configure resource groups for the following:
   - Media Control Platform
   - Call Control Platform
   - CTI Connector
   - Gateway (SIP Server, with `CTI Usage` set to `Based on DN Lookup` (`use-cti` set to `2`)).

Note: This configuration uses CTI flagging to allow Standard VoiceXML applications to bypass CTI Connector. If your deployment uses only IVR-centric applications, you can set `CTI Usage` to `Always On` (`use-cti` to `1`). In this case, you do not need to create matching DNs in the dummy switch.

For detailed step-by-step procedures, see "Task Summary: IVR Behind, Carrier-Connected Integration" on page 147.

For more information about CTI flagging, see "Configuring CTI Flagging for IVR Profiles" on page 188.

PBX-Connected Deployments

In a TDM PBX deployment, the IVR Server is a T-Library client of the premise T-Server and not the SIP Server. Figure 39 illustrates how the VPS, in a TDM deployment, handles call setup for a typical inbound call.

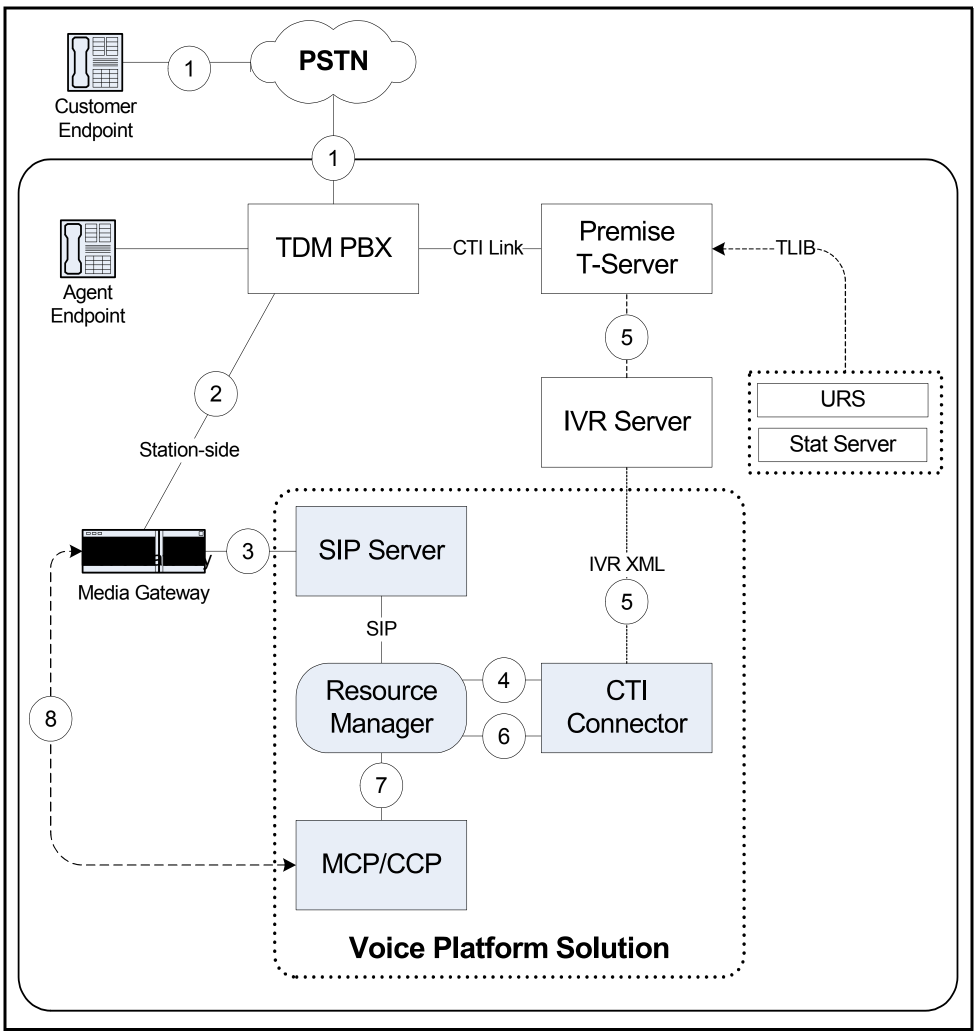


Figure 39: Call Setup for Typical Inbound Call—Behind Mode

1. Caller dials a number on the PSTN network and the call is routed to the customer premise, where it reaches the PBX.
2. From the PBX, the call arrives at the media gateway—the PBX sends the port number to the gateway, not the DNIS as dialed by the customer. At the same time, the premise T-Server (connected to the PBX) sends an `EventRinging` message to the IVR Server.
3. The media gateway sends the call to the SIP Server, where it reaches a `Trunk` DN. This `Trunk` DN has the `contact` option configured to the sip address for Resource Manager.
4. As per the configuration for the gateway resource group on Resource Manager (`CTI Usage` set to `Always On` (`use-cti` set to `1`), Resource Manager sends the call to CTI Connector.
5. CTI Connector retrieves the DNIS (in this case, the `Trunk` DN number) from the `Request URI` it receives from Resource Manager, then sends this number to the IVR Server in a `<NewCall>` message (Genesys proprietary IVR XML). IVR Server returns the DNIS for the IVR Profile to CTI Connector.
6. CTI Connector generates a new call leg with Resource Manager.
7. The Resource Manager matches the DNIS returned from the IVR Server with an IVR Profile, then sends an `INVITE` to the Media Control Platform with the URL to launch the VoiceXML application.
8. The RTP media path is established between the Media Control Platform and the media gateway—and through the PBX to the caller.

Configuration Overview

The relevant configuration steps required for a TDM deployment include the following:

1. Create `Trunk` DNs with the `prefix` option set to match the initial digits of the range to match the number of ports configured on the PBX for GVP.
2. Configure `Voice Treatment Port` DNs on the premise T-Server switch. The number and name of `Voice Treatment Port` DNs depends on the type of switch. No configuration of options for these DNs is required.
3. Configure IVR Ports:
   - The number of IVR ports must match the number of ports configured on the PBX for GVP.
   - The names for the IVR ports must match the port numbers that the media gateway will send to SIP Server as the DNIS.
   - Map each IVR port to the appropriate `Voice Treatment Port` DN on the premise switch.
4. In the IVR Server `Application` object, add a connection to the premise T-Server.

5. In the premise T-Server `Application` object, add connections to the following:
   - Message Server
6. In the SIP Server Application object, add connections to the following:
   - Message Server
7. Go to `Provisioning > Voice Platform > Resource Groups`, and use the wizard to configure resource groups for the following:
   - Media Control Platform
   - Call Control Platform
   - CTI Connector
   - Gateway (SIP Server, with `CTI Usage` set to `Always On` (`use-cti` set to `1`)

For detailed step-by-step procedures, see "Task Summary: IVR Behind, TDM-Connected Integration" on page 151.

Integration with IVR Server—In-Front Mode

In this deployment, IVR Server is configured for In-Front mode—used in premise deployments where the VPS is running on a different instance of Genesys Framework than the premise switch where the agents are registered. In this case, the IVR Server acts as the link between the two instances of Genesys Framework, allowing GVP to transfer a call from the service provider to the agent.

With this premise-based IVR Server In-Front configuration, the call typically arrives at the VPS through a TDM PBX. In this case, the station-side connection cannot provide the DNIS, so the CTI Connector sends a request to IVR Server for information about the call (including the DNIS). IVR Server, running as a T-Server, assigns its own connection ID (`ConnID`) to the call—required for reporting purposes.

This deployment is suitable for Managed Service Provider (MSP) configurations, where the host site and customer site are not linked together in the same Management Framework. All IVR Server functionality resides at the MSP site, and the customer connects to the MSP site only through the IVR Server, configured for In-Front mode.

Figure 40 on page 95 illustrates how the VPS, integrated with IVR Server in the In-Front mode, handles call setup for a typical inbound call.

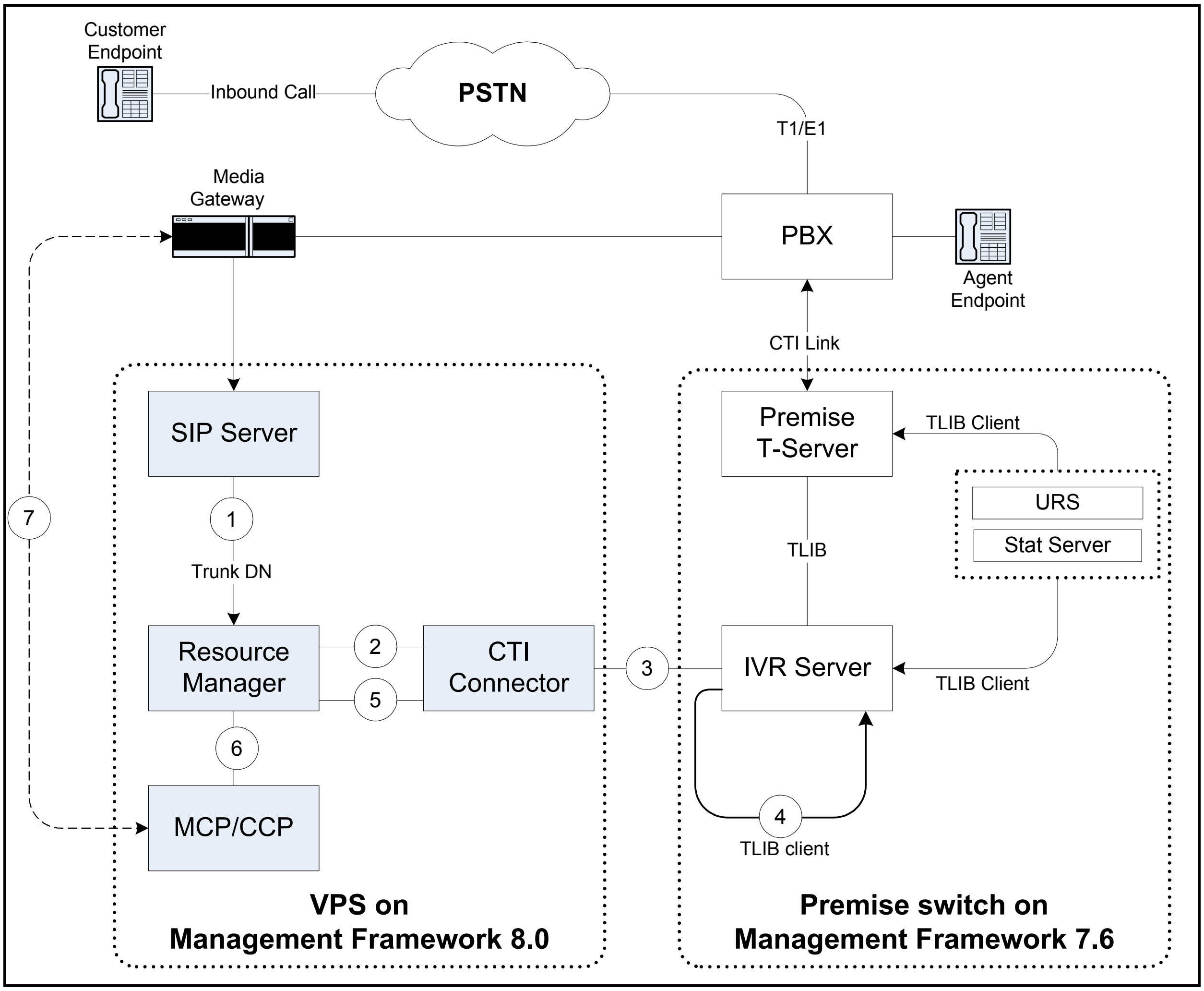


Figure 40: Call Setup for a Typical Inbound Call—In-Front Mode

1. A call comes in from the PSTN through a PBX, then to the third-party media gateway and on to SIP Server. The CTI Connector uses the port number provided by the PBX to send a request for call info to IVR Server, in order to obtain the DNIS for the call.
2. The Resource Manager generates a session ID and forwards it to CTI Connector for call registration.
3. The CTI Connector generates an IVR port number, and then sends this number in an XML request to the IVR Server to register the call.
4. In the In-Front mode, the IVR Server is running as a T-Server and so it assigns the `ConnID` itself—it maps the port number sent by CTI Connector to one of the ports configured in the `IVR` object in the Configuration Layer, linking the `ConnID` with the GVP session ID for the call.
5. The CTI Connector returns the `ConnID` in an `INVITE` to the Resource Manager, along with required call information (DNIS).

6. The Resource Manager selects an IVR Profile based on the DNIS, and then sends an `INVITE` to the Media Control Platform with the URL to launch the VoiceXML application.
7. The voice path is established between the Media Control Platform and the caller.

Integration with Network IVR Server

In this deployment, IVR Server is configured for Network mode in order to integrate with a Network T-Server. The VPS is located at the carrier environment where it is connected to a Service Switching Point (SSP).

Figure 41 illustrates how the VPS, integrated with IVR Server in Network mode, handles a typical inbound call.

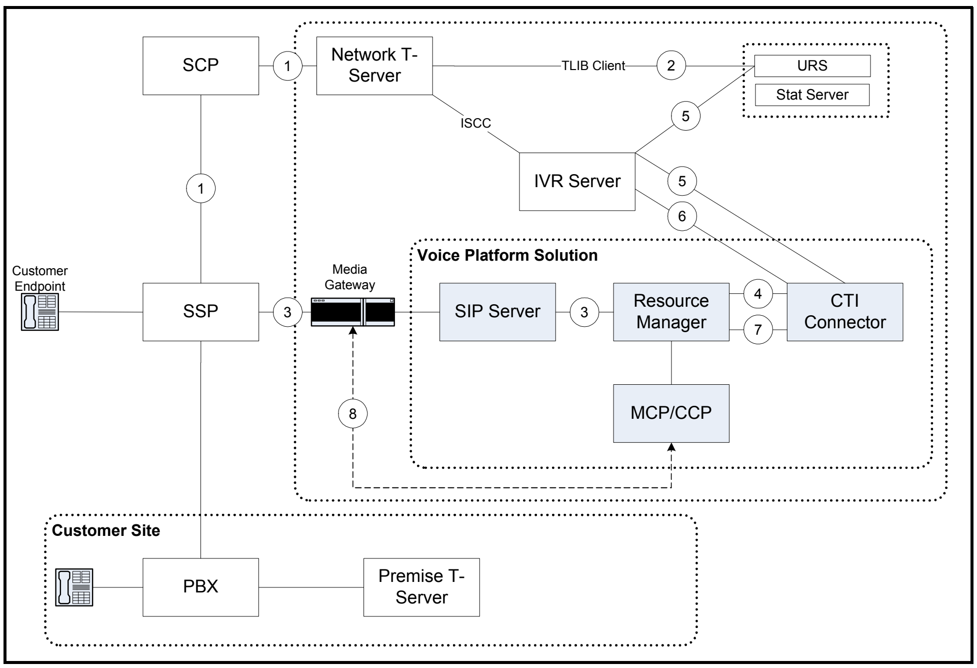


Figure 41: Typical Inbound Call Flow—IVR Server Integration

1. The incoming call reaches the SSP, which sends a route request through the Service Control Point (SCP) to a `Routing Point` DN on the Network T-Server switch.
2. The routing strategy selects the `Trunk` DN that represents GVP as the target, and the call is routed through Inter Server Call Control (ISCC) transfer back to the SSP.
3. The SSP forwards the call to the `Trunk` DN on the SIP Server, which then sends an `INVITE` to the Resource Manager with either the DNIS or a toll free number in the header.

4. The Resource Manager generates a session ID. It forwards this ID, along with the DNIS or toll free number, to CTI Connector
5. The CTI Connector sends an XML request to the IVR Server for call registration with the Network T-Server and URS.
6. The IVR Server obtains the `ConnID`, IVR port, and other call-related data and sends it by XML to CTI Connector.
7. The CTI Connector sends this call-related data in an `INVITE` to the Resource Manager.
8. The Resource Manager selects an IVR Profile based on the DNIS or toll free number, and then sends an `INVITE` to the Media Control Platform with the URL to launch the VoiceXML application.
9. The RTP media path is established between the Media Control Platform and the caller.

Call Transfers Using CTI Connector

For integrations that provide CTI through IVR Server, CTI Connector is the essential component for negotiating and completing call transfers to agents. Depending on your architecture configuration and the kind of voice application used, different methods may or may not be supported for transferring the caller from the self-service portion of the call to the agent-assisted portion.

Supported call transfers include:

- `REFER` transfer for Standard VoiceXML applications
- Bridge transfer for Standard VoiceXML applications
- `REFER` transfer for IVR-centric applications

Standard VoiceXML Applications—REFER

In Standard VoiceXML applications, the application uses the `<transfer>` tag to place the call in queue when the caller wants to transfer to an agent. CTI Connector receives a blind transfer request from the `REFER` message. In response, CTI Connector sends `RouteRequest` to the IVR Server. Call treatments can be applied while the caller waits in the queue. CTI Connector initiates a new call leg with the MCP for each call treatment request. If the treatment collects any customer data, MCP can provide a treatment result.

For Carrier-Connected Architectures

When the agent is ready, the IVR Server sends a route number to CTI Connector, and CTI Connector sends a `REFER` request to the SIP Server. The media gateway negotiates the transfer from the VPS site, through the PSTN, to the site where the agent is located.

For Station Side-Connected Architectures

When the agent is ready, the IVR Server sends a route number to CTI Connector, and CTI Connector initiates a CTI transfer to the premise T-Server through the IVR Server; the PBX patches the caller to the agent directly.

Standard VoiceXML Applications—Bridge Transfer

For bridged transfers from Standard VoiceXML applications, the application also uses the `<transfer>` tag. In this case, CTI Connector receives an `INVITE` requesting a bridged transfer. In response, CTI Connector sends a `RouteRequest` to the IVR Server. Call treatments can be applied while the caller waits in the queue. CTI Connector initiates a new call leg with the MCP for each call treatment request. If the treatment collects any customer data, MCP can provide a treatment result. When the agent is ready, the IVR Server sends a route number to CTI Connector, and CTI Connector sends an `INVITE` request to the SIP Server and bridges the call to the agent.

IVR-centric Applications—REFER Transfer

IVR-centric applications can use TXML (in legacy GVPi applications) or proprietary `<send>` and `<receive>` tags (in NGI applications) to access CTI functionality directly from the application, through CTI Connector and IVR Server. For these functions, MCP sends and receives SIP `INFO` messages. CTI Connector provides a tunnel for communication with the IVR Server. CTI Connector acts as a SIP back-to-back user agent (B2BUA), remaining in the call path until the transfer to the agent is complete. While the caller is waiting in the queue, the call is considered parked at MCP, and any treatments must be provided by the application.

For Station Side-Connected Architectures (IVR Behind Mode)

When the agent is ready, the application retrieves the agent number and MCP sends a `REFER` to CTI Connector. Instead of passing the `REFER` to the media gateway, CTI Connector performs a CTI transfer using the IVR XML `OneStepXfer` message, and the PBX patches the caller to the agent directly. This functionality requires that you set the `CTI Transfer` option to `true` in the IVR Profile.

Configuring the IVR Profile for Transfer Type

For a description of the service parameters that you must configure on the IVR Profile for the various types of supported transfers, see the following:

- Procedure: Task Summary: Configuring Voice Treatments, on page 166

Call Treatments Using CTI Connector

When queuing call transfers using either the REFER or INVITE transfer methods, CTI Connector can be used to request a call treatment. When CTI Connector receives a request for a treatment—for example, a Play Application treatment from a URS routing strategy—it translates this request into a new terminating call leg to MCP, with the required treatment parameters provided in NETANN format.

Using CTI Connector, MCP supports the following treatments for both GVPi and NGI applications:

- Play Application—Invokes a particular VoiceXML application as a separate call leg.
- Play Announcement (TTS)—Plays an announcement for the caller.
- Play Announcement (TTS) and Collect Digits—Plays an announcement and connects digits from the caller.
- Music—Plays a `.vox/.wav` file.

Basic Configuration Using Interaction Routing Designer

These treatments are configured using Voice Treatment objects in the Interaction Routing Designer (IRD), where you can design your routing strategy and then load it on a particular `Routing Point` DN.

For information about configuring Voice Treatment blocks, see the "Voice Treatment Options" section of the *Universal Routing 8.1 Reference Manual*.

Additional Solution-Level Treatment Configuration

When integrated with CTI Connector and IVR Server, VPS supports the standard methods for configuring Voice Treatment blocks in IRD. CTI Connector also provides some additional configuration options to give designers more flexibility when building their applications.

For a task summary of these special configuration steps, see the following:

- Procedure: Task Summary: Configuring Voice Treatments, on page 166

Chapter

8 PSTN Connector

For deployments that require the use of Dialogic hardware, Genesys offers the PSTN Connector, a GVP 8.1 component that provides connectivity to traditional telephony environments (PSTN switches, TDM PBXs or ACDs, and so on) for GVP deployments.

This chapter includes the following sections:

Sample Deployment

PSTN Connector can be used as an equivalent to the media gateway for most of the supported call flow scenarios described in this guide. However, it is beyond the scope of this document to detail all deployments where PSTN Connector could be included. To provide an overview of how PSTN Connector works, this section presents a sample of one of the more common supported scenarios that might include the PSTN Connector.

Station-side Connected, With PSTN Connector

The sample configuration described here is similar to “Station Side-Connected, CTI Through IVR Server” on page 43, except that instead of a media gateway, the solution uses the GVP component PSTN Connector, which integrates with Dialogic hardware and software to provide the TDM signaling interface.

Note: PSTN Connector is not intended as a media gateway replacement. Certain media gateway functions—for example, conferencing—are not supported in PSTN Connector integrations.

PSTN Connector is intended for legacy TDM deployments with Dialogic cards.

Figure 42 shows a VPS integration with PSTN Connector.

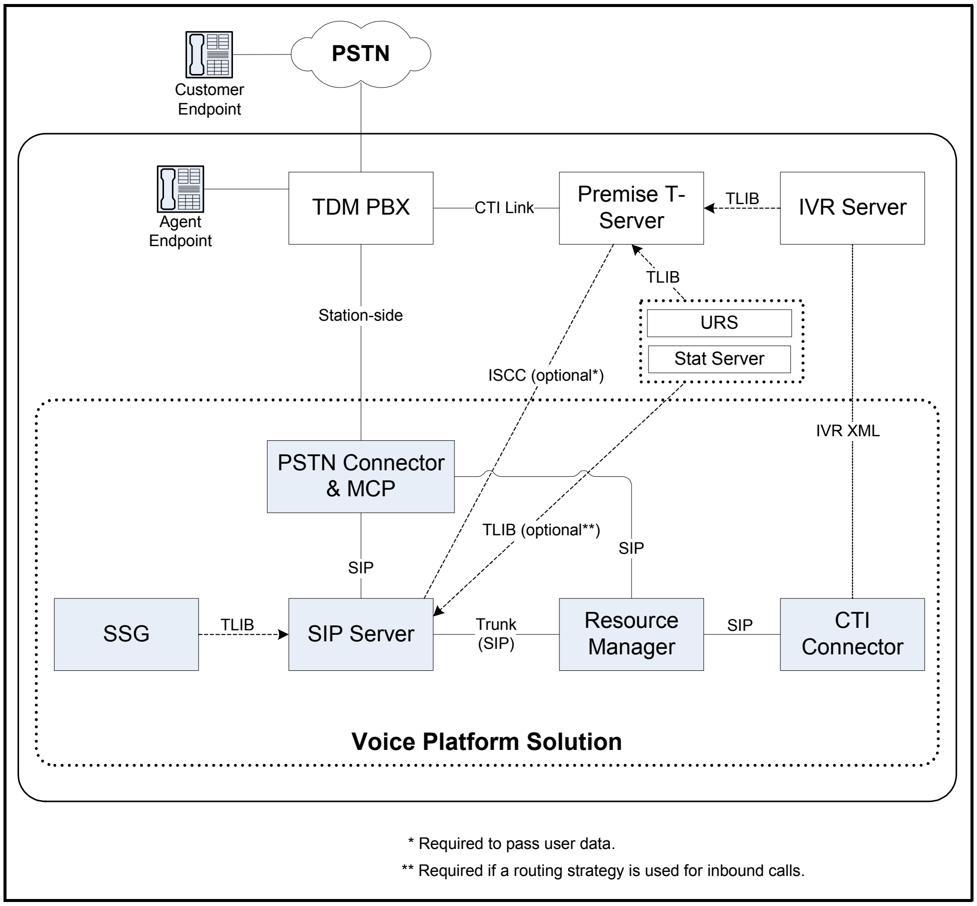


Figure 42: TDM PBX with PSTN Connector

In this architecture, the PBX forwards the incoming call to the PSTN Connector. At the same time it sends info related to the call (ANI, DNIS) through the CTI interface. PSTN Connector sends the call to SIP Server, which in turn forwards the call to the Resource Manager through a `Trunk` DN. Resource Manager then forwards the call to CTI Connector, so that the call can be registered through the IVR Server.

Note: To match legacy deployments, PSTN Connector and MCP can be deployed on the same host machine (optional). On Windows 2008 host machines, some additional configuration for this deployment may be required. For details, see the *Genesys Voice Platform 8.1 Deployment Guide*.

This architecture supports the following kinds of applications:

- Standard VoiceXML applications
- IVR-centric applications

Note: For the relevant procedures for this sample configuration, see Integrating with PSTN Connector, on page 211.

Inbound Call Flows and Transfers

The inbound call flows, including the different kinds of available transfer types, are identical to the call flows described in “Station Side-Connected, CTI Through IVR Server” on page 43—except with PSTN Connector taking the role of the media gateway. For details, please see the previous section.

Outbound Calls

In this architecture, outbound calls can be initiated using a trigger application through the SSG. SIP Server sends the outbound `INVITE` through a `Trunk Group` DN configured for MSML communication to GVP. SIP Server can apply call progress detection (CPD) on the PSTN Connector. After CPD, if the called line is determined to be suitable, the line is connected to the voice application. The application can use the VoiceXML `<transfer>` tag to deliver the call to a destination. PSTN Connector is able to synchronize ports with the IVR Server, allowing for CTI call control from the application (unavailable for outbound calls through a Media Gateway).

PSTN Connector Limitations

While PSTN Connector can be used in most supported call flow scenarios, there are certain limitations. PSTN Connector cannot be used for the following:

- CCXML applications.
- Outbound call scenarios involving the Outbound Contact Solution (OCS). Note that outbound through SSG is supported.
- Media services provided by the following `Voice over IP Service` DN types: `mcu`, `recorder`.

Genesys

Part

2 Integration Procedures

For the procedures that you need to complete to integrate the various components that make up the Voice Platform Solution, to integrate any additional components, or perform any special configuration required by your deployment, see the Task Summary tables and related procedures in the following chapters:

Chapter

9 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 Genesys Administrator or the Solution Control Interface (SCI).

This chapter contains the following section:

Integration Prerequisites

Table 4 lists all of the required prerequisite components, their respective Deployment Guides, and any key actions that you must complete before starting the integration procedures.

Table 4: Prerequisite Components for VPS Integration

Component	Key Actions or Info	Documentation
VPS Components		
SIP Server version 8.0.3 or later	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 8.1 SIP Server Deployment Guide*

Table 4: Prerequisite Components for VPS Integration (Continued)

Component	Key Actions or Info	Documentation
Genesys Voice Platform 8.1	Key GVP components required for the VPS include: • Resource Manager • Media Control Platform • Call Control Platform (optional for CCXML) • Squid Caching Proxy (one per MCP/CCP host) • Reporting Server (optional, required for solution reporting) At a minimum, go to the "List of Procedures" section at the start of the *Genesys Voice Platform 8.1 Deployment Guide*, and complete the following procedures: 1. Configuring system performance settings 2. Adding a host in Genesys Administrator 3. Installing the Local Control Agent 4. Importing the installation packages into the Deployment Repository 5. Using the Installation Wizard to install GVP 6. Creating a Resource Solution object 7. Integrating Application objects with Resource Manager	*Genesys Voice Platform 8.1 Deployment Guide*
Supplementary Services Gateway 8.1	This is an optional GVP 8.1 component—used for initiating outbound calls and connecting the called number to a Standard Voice XML application. For details about integrating the VPS with SSG, see Chapter 13 on page 195	*Genesys Voice Platform 8.1 Deployment Guide*
CTI Connector 8.1	This is an optional GVP 8.1 component—however, it is required for CTI through IVR Server configurations. For details about integrating the VPS with CTI Connector, see Chapter 12 on page 145.	*Genesys Voice Platform 8.1 Deployment Guide*
PSTN Connector 8.1	This is an optional GVP 8.1 component—used for TDM integrations with Dialogic hardware and software. For deployment procedures, see the *Genesys Voice Platform 8.1 Deployment Guide* For details about integrating the PSTN Connector into the solution, see Chapter 14 on page 211.	*Genesys Voice Platform 8.1 Deployment Guide*

Table 4: Prerequisite Components for VPS Integration (Continued)

Component	Key Actions or Info	Documentation
Management Framework 8.0.3 or later	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 8.1 Deployment Guide*
Genesys Administrator 8.1	Genesys Administrator must be installed. 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://<genesys_administrator_host>/wcm` **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)	*Framework 8.1 Genesys Administrator Deployment Guide* *Framework 8.1 Genesys Administrator Help*
Genesys Composer 8.0.2 or later	This is an optional component. Genesys recommends that you use Composer for authoring VoiceXML and editing CCXML applications. You can start Composer from the Windows `Start` menu. For information about using the tool, press `F1` in the application to bring up the Help system.	*Genesys Composer 8.1 Deployment Guide* *Genesys Composer 8.1 Help* *Genesys Voice Platform 8.1 VoiceXML 2.1 Help*

Table 4: Prerequisite Components for VPS Integration (Continued)

Component	Key Actions or Info	Documentation
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 or later	Required for routing to agents and for launching Play Application treatments. Required connections: • Message Server • SIP Server `Application` object • Any other TServer Application included your deployment. • Stat Server	*Universal Routing Server 8.1 Deployment Guide* *Universal Routing Server 8.1 Reference Manual*
Interaction Routing Designer 7.6 or later	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 8.1 Business Process User's Guide* *Universal Routing 8.1 Interaction Routing Designer Help*
Stat Server 7.6 or later	Required for monitoring the availability of agents targeted in the routing strategies. Required connections: • Message Server • SIP Server `Application` object	*Framework 8.1 Stat Server Deployment Guide*
IVR Server 7.5 or later	This is an optional component—however, it is required for CTI through IVR Server configurations. For details, see Chapter 12 on page 145.	*IVR Interface Option 8.1 IVR Server System Administrator's Guide*

Chapter

10 Configuration Tasks for All Deployments

This chapter describes how to integrate the main components of the Voice Platform Solution (VPS)—primarily the configuration steps that are required to integrate SIP Server with 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 common configuration tasks that must be completed for all VPS deployments. After completing these steps, further configuration for CTI through SIP Server or CTI through IVR Server is also required, depending on the needs of your deployment.

This chapter includes the following sections:

Task Summary: Baseline Configuration Tasks

Task Summary: Common Configuration Tasks for All Deployments, on page 112 provides an overview of the main steps that you must complete to integrate SIP Server with the other VPS components.

Task Summary: Common Configuration Tasks for All Deployments

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 107.
2. Check default ports.	Go to: `Provisioning > Environment > Applications` 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.	Go to: `Provisioning > Environment > Applications` 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 118.

Task Summary: Common Configuration Tasks for All Deployments (Continued)

Objective	Related Procedures and Actions
4. Create a resource group for MCP.	Go to: `Provisioning > Voice Platform > Resource Groups` • Click `New` and follow the wizard to create a new resource group of the type `Media Control Platform`. **Key Parameters** • `Monitoring Method`—Accept the default. • `Load Balancing Scheme`—Select `least percent`. • Leave the maximum conference size and count parameters blank for a basic deployment. For the detailed procedure, see "Configuring a resource group for MCP" on page 119
5. Create SIP Server application objects.	Go to: `Provisioning > Switching` 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 8.1 SIP Server Deployment Guide*.
6. Create SIP extensions for your agent endpoints.	Go to: `Provisioning > Switching > Switches` 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 `DNs` tab of the SIP Switch, click `Add` 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 *Framework 8.1 SIP Server Deployment Guide*.

Task Summary: Common Configuration Tasks for All Deployments (Continued)

Objective	Related Procedures and Actions
7. Configure a GVP DN for Standard VoiceXML applications.	Go to: `Provisioning > Switching > Switches` To enable SIP Server to identify GVP in the solution, create the following DN: • Create a GVP `Trunk` DN on the SIP switch, pointing the `TServer > contact` option to the IP address and port for Resource Manager. • `prefix`—Set this option to the first three or four digits of the incoming DNIS For a detailed procedure, see Procedure: Configuring a GVP DN for Standard VoiceXML applications, on page 116.
8. *(Optional)* Integrate SSG into the deployment.	Optional: Required for SSG-initiated outbound call functionality. For information about configuring the VPS with SSG for outbound call initiation, see Chapter 13, "Integrating with SSG," on page 195.
9. *(Optional)* Configure management and monitoring tools.	*Optional: These procedures are only required if you are capturing alarm and trap information.*| 1. Install and configure an SNMP Master Agent an all GVP hosts. 2. Install MIBs on the machine where your SNMP management console (for example HP Open View) is running. 3. Connect GVP components to the SNMP Master Agent on each host. For a detailed procedure, see Procedure: Configuring SNMP Monitoring, on page 123.
Additional Special Configuration	
If your gateway does not support `REFER` transfers...	Go to: `Provisioning > Switching > Switches` On the `Trunk` DN for the gateway, configure the `TServer` section with the following options: • Set the `refer-enabled` option to `false`. • For the `prefix` option, enter the prefix of the ANI for the incoming call. • For the `contact` option, enter the IP address and port of the media gateway. For the detailed procedure, see "Creating a Trunk DN for gateways that do not support REFER" on page 124.

Task Summary: Common Configuration Tasks for All Deployments (Continued)

Objective	Related Procedures and Actions
If SIP Server needs to be taken out of the signaling path after a REFER transfer...	Go to: `Provisioning > Switching > Switches` • In the `Trunk` DN that represents the SIP device, set `oosp-transfer-enabled` to `true`. This takes SIP Server out of the signaling path. **When to Use** In some scenarios, SIP Server needs to be taken out of the signaling path after single-step transfer is completed. For example, if the SIP device cannot receive a different IP address in the `Refer-to` header of the `REFER` message other than its own.
To configure MCP for media redirect transfers.	Go to: `Provisioning > Environment > Applications` 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 `Application` object: • For bridge transfers, no special configuration is required. The solution defaults to this method for bridged transfers. • For media redirect transfers, do one of the following: • On the `Options` tab in the MCP object, in the `sip` section, set the `defaultbridgexfer` option to `MEDIAREDIRECT`. • OR For NGI applications, if you want to leave the default as `BRIDGE`, you can override the default by setting the `genesys:method` parameter in the `<transfer>` tag of the voice application to `mediaredirect`. For a detailed procedure, see "Configuring the solution for media redirect transfers" on page 126.

Task Summary: Common Configuration Tasks for All Deployments (Continued)

Objective	Related Procedures and Actions
To integrate a PBX into the VPS.	For deployments that include a TDM-based 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: • `Switch` object • `Switching Office` 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 example, in a Cisco CallManager (CCM) integration, an `ACD Queue` DN in the Configuration Layer must be configured as a `Routing Point` DN in the CCM. 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 a PBX" on page 53.

Integrating SIP Server with GVP

To integrate the SIP Server and GVP components, complete the following procedures:

1. Procedure: Configuring a GVP DN for Standard VoiceXML applications
2. Procedure: Configuring MCP for integration with SIP Server
3. Procedure: Configuring a resource group for MCP
4. Procedure: Configuring GVP for URS-controlled applications

Procedure: Configuring a GVP DN for Standard VoiceXML applications

Purpose: To create a `Trunk` DN that SIP Server uses to identify GVP in the solution. SIP Server uses this `Trunk` 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 8.1 Deployment Guide*.

Prerequisites

- For the Trunk 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 8.1 User's Guide*.

Start of procedure

1. Go to Provisioning > Switching > Switches, and double-click your SIP Server Switch object.
2. On the DNs tab, click Add.

   A New DN window opens.
3. On the Configuration tab, enter the following information:
   - Number—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 DN.
   - Type—Select Trunk from the drop-down list.
4. On the Options tab, click New and 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, as shown in Figure 43.

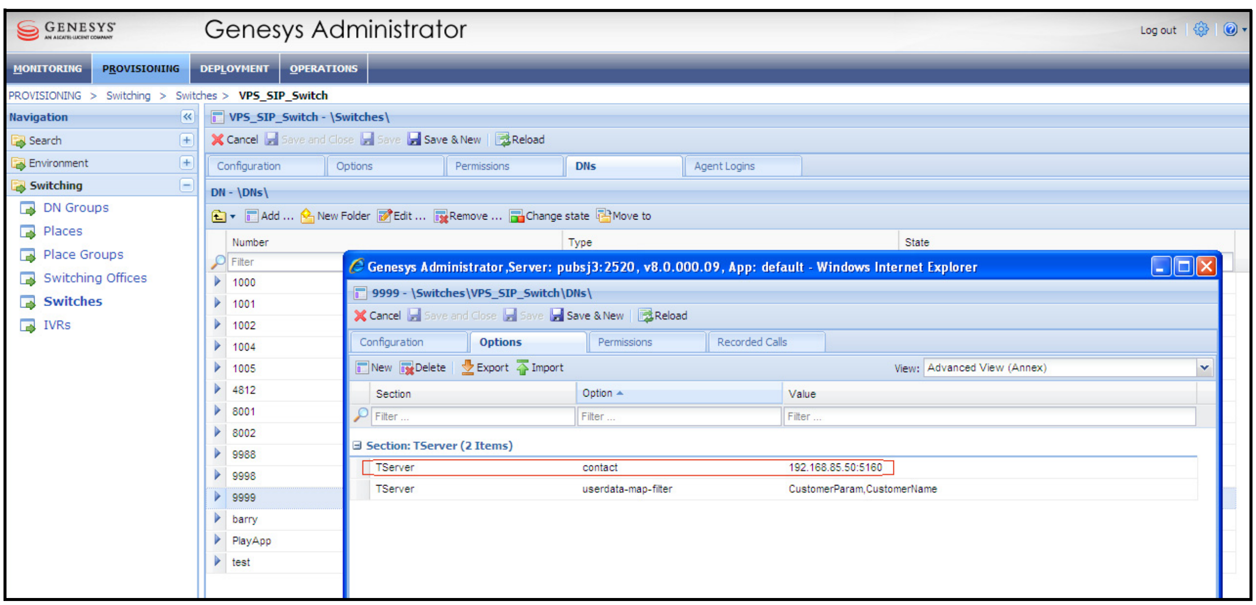


Figure 43: Trunk DN—Contact Option

When an incoming call arrives at the SIP Server, 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.

5. To ensure that SIP Server selects the GVP `Trunk` DN, set the `prefix` option to the first three or four digits of the incoming DNIS (correspondingly, you must provision the IVR Profile for this DNIS range, which is handled by GVP RM).
6. Click `Save and Close` to save all changes.
7. Repeat Steps 2 to 6 for every unique contact number that customers can use to dial into the contact center.

End of procedure

Next Steps

- Procedure: Configuring MCP for integration with SIP Server

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 `Options` 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 5.

3. On the `Options` 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 5 describes in greater details the options that are configured in this procedure.

Table 5: 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 225.
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 226.

Next Steps

- If your Resource Manager already includes a resource group for MCP, continue at:
  - Procedure: Configuring a gateway resource group for SIP Server
- Otherwise, continue at:
  - Procedure: Configuring a gateway resource group for SIP Server

Procedure: Configuring a resource group for MCP

Purpose: To create the resource group on Resource Manager used to define the connection to the MCP.

Start of procedure

1. In Genesys Administrator, go to the `Provisioning > Voice Platform > Resource Groups` panel (see Figure 44 on page 120).

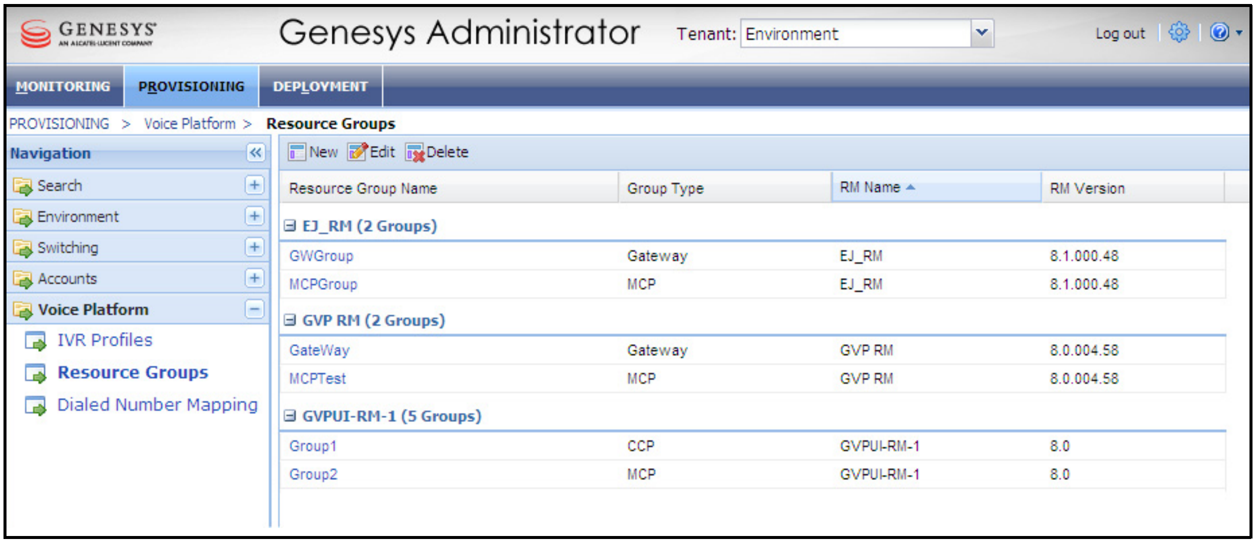


Figure 44: The Resource Groups Panel

2. Click `New` to start the `Resource Group` Wizard.
3. When the wizard opens, click `Next`.
4. On the `Resource Manager Selection` page, select the Resource Manager for which you want to configure the resource group, then click `Next`.
5. On the `Group Name and Type` page, give your resource group a name, and select `Media Control Platform` as the type. Then click `Next`.
6. On the `Tenant Assignment` page, select the child tenant (if available) that you want to dedicate these resources to. Leave unassigned to share the resources across all child tenants. Click `Next`.
7. On the `Group Properties` page, configure the parameters as follows:
   - `Monitoring Method`—Accept the default of `SIP-OPTIONS`.
   - `Load Balancing Scheme`—Select `least percent`.

   Tip: The parameters `Maximum Conference Size` and `Maximum Conference Count` are optional. For information about configuring conferences, consult the *Genesys Voice Platform 8.1 User's Guide*.

8. On the Resource Assignment page, do the following:
   a. Select your Media Control Platform from the list.
   b. Select the `Scheme`—either SIP or SIPS (secure SIP).
   c. Select the `Port` number—typically 5070.
   d. Click `Next`.
9. On the `Confirmation` page, click `Finish`.

End of procedure

- Procedure: Configuring a gateway resource group for SIP Server, on page 121

Procedure: Configuring GVP for URS-controlled applications

Purpose: To configure GVP by creating a default IVR Profile to enable the connection to the Media Control Platform.

Summary

The URS-controlled application that you specify in the `APP_URI` parameter will override the application that is specified in the default IVR Profile.

Start of procedure

1. Check that a default IVR Profile has been created. For more information about creating default IVR Profiles, see Chapter 7 in the *Genesys Voice Platform 8.1 Deployment Guide*.
2. You can also create a separate IVR Profile for this particular application, if there are special requirements (which you can specify in the profile) to be applied to the URS-controlled application.

   For example, after creating the IVR Profile for this application, append the `gvp-tenant-id` parameter to the end of the `APP_URI` parameter in the Play application. For more information about creating Play applications, see Procedure: Creating and loading a routing strategy, on page 141.

End of procedure

Next Steps

- No further steps are required.

Procedure: Configuring a gateway resource group for SIP Server

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

Prerequisites

- To configure SIP Server as a gateway resource, and GVP requires a connection between the RM and the SIP Server.

Start of procedure

1. In Genesys Administrator, go to `Provisioning > Voice Platform > Resource Groups` panel.

2. Click `New` to start the `Resource Group` Wizard.
3. When the wizard opens, click `Next`.
4. On the `Resource Manager Selection` page, select the Resource Manager for which you want to configure the resource group, then click `Next`.
5. On the `Group Name and Type` page, give your resource group a name, and select `Gateway` as the type. Then click `next`.
6. On the Group Properties page, configure the parameters as follows:
   - `Monitoring Method`—Accept the default of `SIP-OPTIONS`.
   - `Load Balancing Scheme`—Accept the default of `Round Robin`.
   - `CTI Usage`—Select `Always Off`.

   Tip: For CTI through IVR Server deployments, you may want to select a different CTI usage. For more information, see "Configuring CTI Flagging for IVR Profiles" on page 188.

7. On the Resource Assignment page, do the following:
   a. Select your SIP Server from the list.
   b. Select the `Scheme`—either SIP or SIPS (secure SIP).
   c. Select the SIP `Port` number—typically 5060.
   d. Select the `Max Ports` for this SIP Server gateway group.
   e. Click `Next`.
8. On the `Confirmation` page, click `Finish`.

End of procedure

Next Steps

- 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
- This completes the baseline integration of the VPS. Additional configuration is also required, depending on how CTI is provided in the solution. Continue at one of the following:
  - Chapter 11, "Configuration Tasks for CTI Through SIP Server," on page 129
  - Chapter 12, "Configuration Tasks for CTI Through IVR Server," on page 145.

Configuring SNMP Monitoring

VPS supports Simple Network Management Protocol (SNMP) monitoring for the Resource Manager, Media Control Platform, Call Control Platform, and

Supplementary Services Gateway components. If configured for it, these components can maintain status information and statistics in SNMP Management Information Base (MIB) tables. You can query these MIBs with an SNMP Management Console (for example, HP Open View).

Note: For more information about the about SNMP traps, as well as basic troubleshooting information for the Genesys Voice Platform, see the *Genesys Voice Platform 8.1 Troubleshooting Guide*.

Procedure: Configuring SNMP Monitoring

Purpose: To install and configure the Management Framework and GVP components required to capture alarm and trap information to a third-party SNMP management console.

Prerequisites

- Management Framework 8.1 CD
- Genesys Voice Platform 8.1 CD

Start of procedure

1. Verify that an MIB management console (for example, HP Open View) is up and running in your environment.
2. From the Management Framework 8.1 CD, install and configure an SNMP Master Agent (SNMP_MA) application on every machine that hosts a GVP component.
   a. Import the `SNMP_Master_Agent_811` template to the `Application Templates` folder.
   b. Create an SNMP_MA `Application` object on each GVP host.
   c. Install the SNMP_MA application on each GVP host.
   d. On the `Options` tab of each SNMP_MA application, in the `snmp` section, set the `trap_target` option to the IP address and port number of the machine where your MIB browser is running.

   Note: For more information about installing SNMP Master Agents, consult the *Framework 8.1 Deployment Guide*.

3. From the Genesys Voice Platform 8.1 CD, load the MIBs Installation Package on your SNMP management console (for example, HP Open View).
   a. Run the setup.exe file for the MIB Installation Package.
   b. When prompted, select the default installation path:

   Windows `C:\Program Files\GCTI\VP MIB 8.1`

Linux `sh install.sh /opt/genesys/gvp/VP_MIB_8.1`

Note: For more information about installing MIBs, consult the *Genesys Voice Platform 8.1 Deployment Guide*.

4. For every GVP component that you want to monitor, on the `Configuration` tab, add a connection to the SNMP_MA installed on the same host.

End of procedure

Integrating With Gateways That Do Not Support REFER

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.

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.

Start of procedure

1. Go to `Provisioning > Switching > Switches`, and double-click the SIP Server `Switch`.
2. On the DNs tab, click `Add`.
3. Enter a `Number` and select `Trunk` as the type.
4. On the `Options` tab, create a `TServer` section, and add new options as follows:
   - Set `refer-enabled` to `false`.
   - Set `prefix` to the initial digits of the ANI for 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 6.

5. Click Save to save all changes.

End of procedure

Additional Info

Table 6 describes in greater detail the options configured in this procedure.

Table 6: Configuring DN Options

Section	Option	Value
TServer	refer-enabled	Set refer-enabled to false so that SIP Server will use a re-INVITE message instead of a REFER message for single-step call transfers.
TServer	prefix	Set this option to the initial digits of the ANI for the incoming call. For example, if the caller number is 9051234567, then set this option to 905.
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.

Next Steps

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

Configuring the Solution for 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.

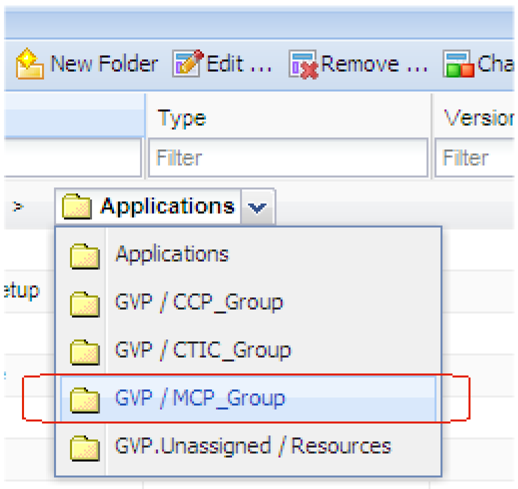
For NGI applications, if you want to use a different bridge transfer method than the default method specified in MCP, you can override the default by setting the gvp.method parameter in the <transfer> tag of the application to the transfer type you want to use: mediaredirect or bridge. Or if building your application in Composer, for the transfermethod, select mediaredirect.

Procedure:
Configuring the solution for media redirect transfers

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

Start of procedure

1. In Genesys Administrator, under `Provisioning > Environment > Applications`, select the GVP/MCP Group from the configuration unit drop-down menu.



2. Open the MCP application, and on the `Options` 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`. Or for NGI applications, you can set `gvp.method` to `bridge` in the voice application, which overrides the default.

3. Click `Save` to save the changes.

End of procedure

Special Configuration for Outbound Calls

Depending on your media gateway, the VPS supports outbound calls using either the remote dialer service feature of MCP, CCXML and the Call Control Platform (CCP), or the Supplementary Services Gateway (SSG). Call Progress

Analysis (CPA) is available for outbound calls, also depending on the media gateway.

Note: This configuration is meant for legacy deployments or applications, where the Supplementary Services Gateway (SSG) is not involved in initiating or controlling the outbound application. However, Genesys recommends using SSG when possible. For details about configuring the solution for SSG, see Chapter 13, "Integrating with SSG," on page 195,

Task Summary: Configuring the Solution for Outbound Calls provides an overview of the special configuration steps required for outbound calls in different scenarios to work.

Task Summary: Configuring the Solution for Outbound Calls

Objective	Related Procedures and Actions

For outbound calls using remote dialer...	1. Check that the `defaultgw` option in the `sip` section of MCP is set to the host name or IP address and SIP port—for example:
	`pstn-gw.voiceplatform.com:5060`
	2. Check that the `routeset` option in MCP is configured to point to the Resource Manager—for example:
	`<sip:rm:5060;lr>`
	MCP needs to have the address of the next hop, which is the Resource Manager, when making outbound calls.
	3. Use the PuTTY terminal emulator to establish a telnet connection to MCP. The default telnet port for MCP is `6999`.
	4. For basic remote dialing, use the following Telnet command:
	`call <telno> <ani> <url> <refno> [uuidata] [defaults] [parameter_list]`
	5. For CPA-enabled remote dialing, use the command in Step 4, appended with an additional parameter—for example (the appended parameter is in **bold**):
	`call 6796500 65 file://C:/samples/main.vxml 205 --` **`Sip.Invite.X-Detect=Request=FAX,AMD`**

Task Summary: Configuring the Solution for Outbound Calls (Continued)

Objective	Related Procedures and Actions
For outbound calls with CCP, using NGI applications...	1. In the IVR Profile, check that the `default-properties-page` parameter points to the `default-ng.vxml` file, not to `default.vxml`. 2. For Call Progress Analysis, the CCXML application must add a private header to the `INVITE` to trigger CPA on the Paraxip gateway. For sample code required to trigger CPA, see Step 2 on page 62. 3. The initiating application must conform to the CCXML specification for creating new sessions. You can find the specification at the following URL: `http://www.w3.org/TR/ccxml/#createsession`

Genesys

Chapter

11 Configuration Tasks for CTI Through SIP Server

This chapter describes the additional configuration tasks required to integrate the solution for a CTI through SIP Server deployment. This basic infrastructure supports multiple architecture configurations and transfer modes, as described in "How It Works—The Basic Inbound Call Flow" on page 23 and "Supported CTI Through SIP Server Configurations" on page 31.

This chapter includes the following sections:

Task Summary: CTI Through SIP Server Configuration

Task Summary: Task Summary: Configuration for CTI Through SIP Server, on page 130 provides an overview of the main steps that you must complete in order to integrate SIP Server with the other VPS components.

Task Summary: Task Summary: Configuration for CTI Through SIP Server

Objective	Related Procedures and Actions
1. Verify the baseline integration tasks are completed.	All baseline tasks in Common Configuration Tasks for All Deployments, page 112 must be completed. To verify that the solution is working: 1. Go to `Provisioning > Environment > Applications`. 2. Check that the green `Started` bar appears under the `Status` column for the following applications: • SIP Server • Resource Manager • Media Control Platform/Call Control Platform • Universal Routing Server • Stat Server If any of these applications are not shown as started, click the application, and then click the green `Start` arrow. If any application does not start, recheck the configuration.
2. Configure Resource Manager for integration with SIP Server.	Go to: `Provisioning > Voice Platform > Resource Groups` To communicate with SIP Server, Resource Manager requires a gateway resource that represents SIP Server. • Click `New` and follow the wizard to create a new resource group of the type `Gateway`. **Key Parameters** • Accept the defaults for `Monitoring Method` and `Load Balancing Scheme`. • Set `CTI Usage` to `Always Off`. For the detailed procedure, see "Configuring a gateway resource group for CTI through SIP Server" on page 132.

Task Summary: Task Summary: Configuration for CTI Through SIP Server (Continued)

Objective	Related Procedures and Actions
3. Configure a GVP DN for URS-centric applications.	Go to: `Provisioning > Switching > Switches` To enable SIP Server to identify GVP for in the solution, create the following DN: • 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. Set the `service-type` option to `application`. For the detailed procedure, see "Configuring a GVP DN for URS-centric applications" on page 133.
4. Configure user data exchange between SIP Server and GVP.	Go to: `Provisioning > Switching > Switches` 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. 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 `Trunk` and `Voice over IP Service` DNs. For a more detailed procedure, see "Enabling user data exchange between SIP Server and GVP" on page 135. For examples of user data mapping, see Appendix A on page 219.
5. Create a Routing Point DN on the SIP Server.	Go to: `Provisioning > Switching > Switches` 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 140.

Task Summary: Task Summary: Configuration for CTI Through SIP Server (Continued)

Objective	Related Procedures and Actions
6. 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 5. For the detailed procedure, see "Creating and loading a routing strategy" on page 141.

Integrating SIP Server with GVP for CTI Through SIP Server

To integrate the SIP Server and GVP components for CTI through SIP Server deployments, complete the following procedures:

1. Procedure: Configuring a gateway resource group for CTI through SIP Server
2. Procedure: Configuring a GVP DN for URS-centric applications, on page 133

Procedure: Configuring a gateway resource group for CTI through SIP Server

Purpose: To create a gateway resource that Resource Manager uses to communicate with SIP Server, in CTI though SIP Server configurations.

Prerequisites

- To configure SIP Server as a gateway resource, and GVP requires a connection between the RM and the SIP Server.

Start of procedure

1. Go to `Provisioning > Voice Platform > Resource Groups` panel.
2. Click `New` to start the `Resource Group` Wizard.
3. When the wizard opens, click `Next`.
4. On the `Resource Manager Selection` page, select the Resource Manager for which you want to configure the resource group, then click `Next`.

5. On the Group Name and Type page, give your resource group a name, and select Gateway as the type. Then click next.
6. On the Group Properties page, configure the parameters as follows:
   - Monitoring Method—Accept the default of SIP-OPTIONS.
   - Load Balancing Scheme—Accept the default of Round Robin.
   - CTI Usage—Select Always Off.
7. On the Resource Assignment page, do the following:
   a. Select your SIP Server from the list.
   b. Select the Scheme—SIP.
   c. Select the SIP Port number—typically 5060.
   d. Select the Max Ports for this SIP Server gateway group.
   e. Click Next.
8. On the Confirmation page, click Finish.

End of procedure

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 with "Configuring a GVP DN for URS-centric applications".

Procedure: Configuring a GVP DN for URS-centric applications

Purpose: To create a Voice over IP Service DN that SIP Server uses to identify GVP in the solution. 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..

Start of procedure

1. Go to Provisioning > Switching > Switches, and double-click your SIP Server Switch object.
2. On the DNs tab, click Add.

   A New DN window opens.
3. On the Configuration tab, enter the following information:
   - Number—Enter a name or number for this DN.
   - Type—Select Voice over IP Service from the drop-down list.
4. On the Options tab, click New and create a new section called TServer, then add the following new options:
   - contact—Enter the IP address and SIP port for the Resource Manager as the value.
   - service-type—Enter application as the value.
5. Click Save and Close 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 with "Enabling user data exchange between SIP Server and GVP" on page 135.
- If your call flow design does not require any user data mapping, continue with "Building a URS-Controlled Application" on page 139.

Mapping User Data

This section provides procedures for enabling the exchange of user data between Management Framework and the GVP voice application.

Data Flows in Two Directions

The exchange of user data 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 to 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 of REFER messages or re-INVITE transfers, mapping will occur, provided that 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 from GVP to SIP Server, see Appendix A on page 219.

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.

Prerequisites

For mapping user data from SIP Server to GVP, you need to modify the GVP Trunk DN and Voice over IP Service DN that you created in "Configuring a GVP DN for URS-centric applications" on page 133.

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" on page 136.

2. If user data mapping is required from SIP Server to GVP, in the GVP Trunk 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" on page 137.

End of procedure

Next Steps

- If you have completed the mapping procedures required for your CTI operations, continue with "Building a URS-Controlled Application" on page 139.

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. Go to `Provisioning > Environment > Applications`, and double-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 45 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.

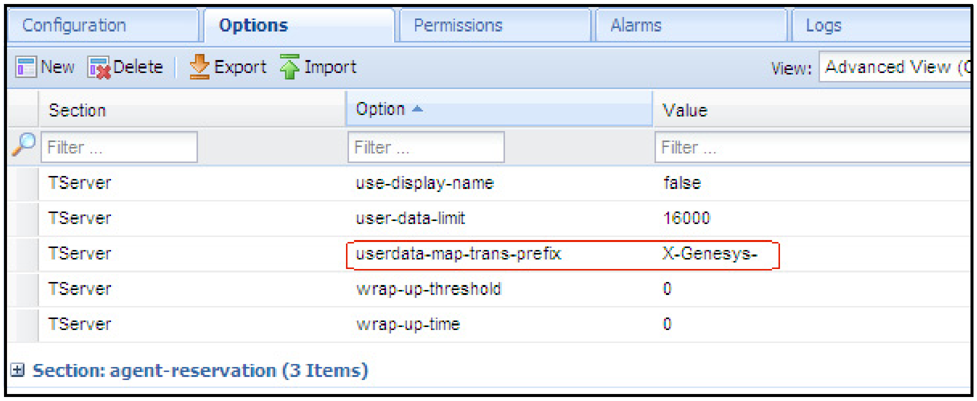


Figure 45: Sample Mapping—Prefix Method

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" on page 137.
- If GVP does not need any user data, then continue with "Building a URS-Controlled Application" on page 139.

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

- Procedure: Configuring a GVP DN for URS-centric applications, on page 133
- Procedure: Configuring user data mapping on SIP Server, on page 136

Start of procedure

1. Go to `Provisioning > Switching > Switches`, and double-click your SIP Server `Switch` object.
2. On the `DNs` tab, select the `Trunk` DN that you created in Step 2 of "Configuring a GVP DN for URS-centric applications" on page 133.

   A new window for this DN opens.
3. On the `Options` tab, in the `TServer` section, create a new option called `userdata-map-filter`.
4. 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 46 shows the GVP `Trunk` DN configured with the sample UserData attributes `CustomerParam` and `CustomerName`.

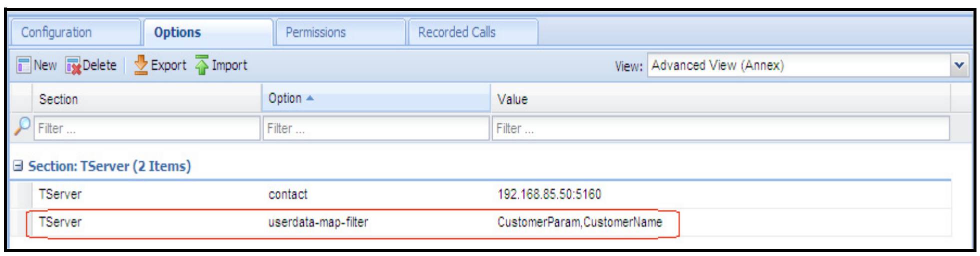


Figure 46: Sample Mapping—Filter Method

   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.
5. Click `Save` to save all changes.
6. Repeat steps 2 to 5 for the `Voice over IP Service` DN that you created in Step 2 of "Configuring a GVP DN for URS-centric applications" on page 133.

End of procedure

Next Steps

- "Building a URS-Controlled Application"

Building a URS-Controlled Application

Genesys recommends that you build your voice call flows using a URS-controlled application paradigm. In this paradigm, simple VoiceXML applications handle the voice dialog portion of the call, while the routing strategy provides all CTI functionality.

Task Summary: Building a URS-Controlled Application

Task Summary: Building a URS-Controlled Voice Application, on page 139 provides an overview of the steps required to build a typical URS-controlled application.

It is important that you create a default IVR Profile before you start building your URS-controlled applications. See Procedure: Configuring GVP for URS-controlled applications, on page 121.

Task Summary: Building a URS-Controlled Voice Application

Call Event	Related Procedures and Actions
1. Call arrives on VPS.	Configure a `Routing Point` DN on the SIP Server switch. Incoming calls for a URS-controlled application should arrive at this DN. For a detailed procedure, see Procedure: Creating a Routing Point on the SIP Server.
2. A routing strategy starts.	Configure a routing strategy to be loaded on the `Routing Point` DN. **Collecting User Data** The strategy can perform actions before the voice dialog starts—for example, it can collect digits or look up customer information in the database, and then pass this info on to the voice application.
3. The strategy launches a voice dialog.	Use Composer (recommended) to build a VoiceXML application, and then add a Play Application treatment block to the routing strategy, configuring the `APP_URI` parameter so that it points to the VoiceXML application. **Collecting User Data** If you want the application to collect data from the customer, include the following key blocks in the application: • An `Input` block to collect the data. • An `InteractionData` block to send the data to the strategy.

Task Summary: Building a URS-Controlled Voice Application (Continued)

Call Event	Related Procedures and Actions
4. Call control returns to the strategy.	The Exit block in the VoiceXML application ends the voice dialog, and call control (as well as any collected data) returns to the routing strategy—no transfer from the application is required. **Using the Attached Data** If the VoiceXML application collected any data, the routing strategy can use a GetInteractionData function to assign the collected information to a variable. The strategy can use this variable in segmentation functions to route the caller—for example, to different agent groups depending on an account number.
5. The call is routed to an appropriate agent.	A TRoute function block in the routing strategy can be used to send the call to a specific agent. For a detailed procedure outlining how to create a simple routing strategy, see Procedure: Creating and loading a routing strategy, on page 141.

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. Go to Provisioning > Switching > Switches, and double-click your SIP Server Switch object.
2. On the DNs tab, click Add.

   A New DN window opens.
3. On the Configuration tab, enter the following information:
   - Number—Enter a name and number for the routing point.
   - Type—Select Routing Point from the drop-down list.
4. Click Save and Close.

End of procedure

Next Steps

- Procedure: 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 Inbound Call Flow" on page 23.

Before you create the URS-controlled applications, ensure that you have an existing default IVR Profile. See Procedure: Configuring GVP for URS-controlled applications, on page 121.

Prerequisites

- A `Routing Point` DN on which to load the strategy. See Procedure: Creating a Routing Point on the SIP Server, on page 140.
- 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 *Genesys Voice Platform 8.1 VoiceXML 2.1 Help*.
  - For information about creating VoiceXML applications using Genesys Composer, press `F1` from the Composer application to access its Help system.

> **Note:** This VoiceXML application must be voice self-service only. The `<transfer>` block is unavailable for URS-controlled applications. Routing to assisted service must be done in the strategy, as no transfer capability exists for this type of voice application.

- 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 `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 8.1 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 47 on page 142).

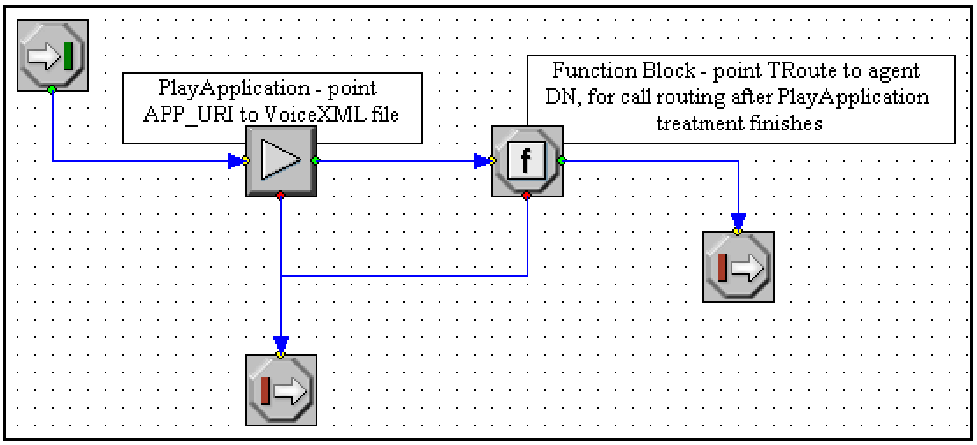


Figure 47: Sample VPS Routing Strategy

3. Configure the `PlayApplication` block in this strategy, so that the `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 `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.

      You can also append the `gvp-tenant-id` parameter to the end of the `APP_URI` parameter in the Play application to specify a particular tenant. For more information about this parameter, see the *Genesys Voice Platform 8.1 User's Guide*.

      Figure 48 on page 143 shows a sample `Play application properties` window, as configured to launch the GVP VoiceXML application.

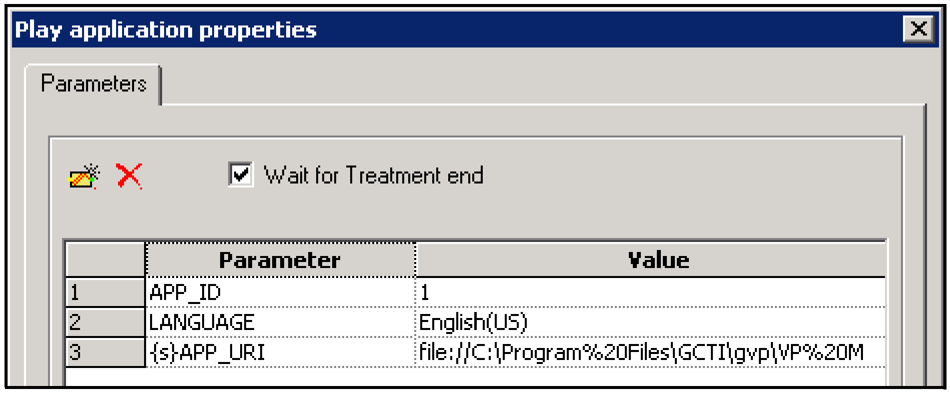


Figure 48: Sample Play Application Properties Window

> **Tip:** In this example, the VoiceXML application is stored locally. If using an application server, the `APP_URI` will point to the HTTP path for the application instead.

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 49 on page 144 shows a sample `Function properties` window, configured for the SIP endpoint DN `9001` on the SIP switch.

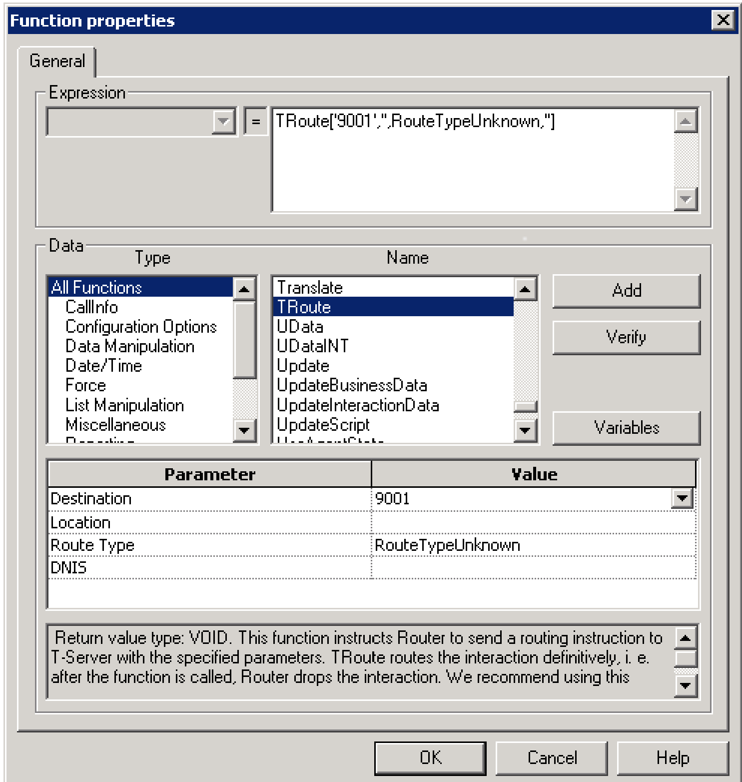


Figure 49: Sample Function Properties Window

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 140:
   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 a CTI through SIP Server VPS integration.

Chapter

12 Configuration Tasks for CTI Through IVR Server

This chapter describes how to integrate the Voice Platform Solution with the IVR Server, using the VPS 8.1 CTI Connector (CTI-C) component. The tasks to integrate IVR Server into the VPS differ, depending on the IVR Server mode your deployment requires.

For a description of how this integration works, or why it might be necessary in your deployment, see Chapter 7, "Support for IVR Server," on page 89.

Complete the following tasks, depending on the IVR Server mode:

Detailed procedures are provided in the following sections:

Task Summary: Verify Prerequisites

Task Summary: Verify Prerequisites, on page 146 lists the component configurations and other actions that must be completed before you begin integrating the VPS with the IVR Server.

Task Summary: Verify Prerequisites

<table>
<tr><th>Requirement</th><th>Related Procedures and Actions</th></tr>
<tr><td>Verify the baseline integration tasks are completed.</td><td>All baseline tasks in Common Configuration Tasks for All Deployments, page 112 must be completed.
To verify that the solution is working:
1. Go to <code>Provisioning > Environment > Applications</code>.
2. Check that the green <code>Started</code> bar appears under the <code>Status</code> column for the following applications:
• SIP Server
• Resource Manager
• Media Control Platform/Call Control Platform
• Universal Routing Server
• Stat Server
If any of these applications are not shown as started, click the application, and then click the green <code>Start</code> arrow. If any application does not start, recheck the configuration.</td></tr>
<tr><td>(Optional) The Supplementary Services Gateway (SSG) has been integrated into the VPS.</td><td>Optional: Required for SSG-initiated outbound call functionality.
If the SSG has not already been integrated into the solution, complete the procedures in Chapter 13, “Integrating with SSG,” on page 195.</td></tr>
<tr><td>The IVR Server and related objects have been created, and the IVR Server is installed.</td><td>If IVR Server is not already included in your deployment, you must install and configure new objects as follows.
Behind or In-Front Mode
• <code>I-Server</code> application
• <code>TServer_IVR</code> application
Genesys recommends that you use the Genesys 8.1 IVR Interface Option Wizard (available on the product CD) to initially configure these objects. For more information, see the “Wizard Configuration” chapter in the IVR Interface Option 8.1 IVR Server System Administrator’s Guide.
Network Mode
• <code>TServer_IVR_Network</code> application
• <code>Network T-Server</code> switch and application
For information about configuring a Network T-Server, refer to the Network T-Server Deployment Guide for your particular switch.</td></tr>
</table>

Task Summary: Verify Prerequisites (Continued)

Requirement	Related Procedures and Actions
The CTI Connector application has been created, and the application is installed.	If CTI Connector is not already included in your baseline GVP deployment, go to the “List of Procedures” section in the *Genesys Voice Platform 8.1 Deployment Guide*, and complete the following procedure: • Installing the CTI Connector (Windows)

Task Summary: IVR Behind, Carrier-Connected Integration

Task Summary: TDM-Connected, IVR Behind provides an overview of the main steps that you must complete in order to integrate the VPS with IVR Server in the Behind mode, in a carrier-connected deployment, with CTI flagging enabled.

Task Summary: Carrier-Connected, IVR Behind

Objective	Related Procedures and Actions
1. Log in to Genesys Administrator.	In your web browser, enter the application URL in the following format: `http://<genesys_administrator_host>/wcm`
2. Verify prerequisites.	Before starting the integration, verify the following prerequisites have been met: • The basic Voice Platform Solution is deployed. • The IVR Server and related objects have been created, and the IVR Server is installed. • The CTI Connector application has been created, and the application is installed. For a more detailed description of this task, see Procedure: Task Summary: Verify Prerequisites, on page 145.
3. Configure the SIP Server for Behind mode integration.	Go to: `Provisioning > Environment > Applications` Configure the SIP Server options as follows: • Set `override-to-on-divert` to `false`. • Set `event-ringing-on-100trying` to `true`. • Set `handle-vsp` to `all`. • Add connections to URS, Stat Server, and Message Server. For the detailed procedure, see Procedure: Configuring the SIP Server for Behind mode, on page 169.

Task Summary: Carrier-Connected, IVR Behind (Continued)

Objective	Related Procedures and Actions
4. Configure the IVR Server.	For step-by-step procedures, see the following: 1. Procedure: Creating the dummy switching office, on page 171 2. Procedure: Creating the dummy switch for TServer_IVR, on page 171 3. Procedure: Configuring the TServer_IVR object, on page 172 4. Procedure: Adding required connections to the I-Server, on page 172 **Key Actions** • Associate `TServer_IVR` with the dummy switch. • In `TServer_IVR`, configure the `Router Timeout` option to a value of `7 s`. • In `TServer_IVR`, configure `active-release` to `false`. • In the `I-Server` object, add a connection to `TServer_IVR`.
5. Create the `IVR` object to represent GVP.	Go to: `Provisioning > Switching > IVRs` 1. Create an `IVR` object to represent GVP: • Select `Genesys Voice Platform` as the IVR Server. 2. Click `Save` to register with the Configuration Layer. For a detailed procedure, see Procedure: Creating an IVR object to represent GVP in the solution, on page 173.

6. Configure the CTI Connector.	Go to: `Provisioning > Environment > Applications`
	1. Open your CTI Connector `Application` object.
	2. On the `Options` tab, configure the following mandatory parameters:
	  • `Resource Manager IP Address`
	  • `IVR Client Name`
	  • `IVR Server Host IP Address`
	  • `IVR Server Communication Port`
	  • `Local Host Name`
	3. On the `Options` tab, configure the following IVR port options:
	  • `Max IVRPorts`
	For a detailed procedure, see Procedure: Configuring the CTI Connector for IVR Behind, on page 174.

Task Summary: Carrier-Connected, IVR Behind (Continued)

Objective	Related Procedures and Actions
7. Configure IVR Ports.	To configure IVR channels as `Voice Treatment Port` DNs, complete the following procedures: 1. Procedure: Creating Voice Treatment Port DNs, on page 175 2. Procedure: Mapping IVR Ports to their matching DNs, on page 177 3. Procedure: Configuring a Place object for each IVR Port, on page 178 4. Procedure: Configuring a Place Group as the target for IVR Ports, on page 178 **Key Rules** • Create the DNs in the SIP Server switch, configuring the `TServer` section for each as follows: • `contact`—Point this option to the Resource Manager, using the format `sip:<RM_ip_address>:<RM_sip_port>`. • `event-ringing-on-100trying`—Set this option to `true`. • Create matching DNs in the dummy switch for `TServer_IVR`—no configuration required. • Map each IVR Ports to appropriate DN in the dummy switch.. • Use identical numbering for the DNs, ports, and places. • Use continuous numbering (in increments of one) when naming the DNs.
8. Configure Resource Manager for integration with CTI Connector.	Go to: `Provisioning > Voice Platform > Resource Groups` For step-by-step procedures, see the following: 1. Procedure: Creating a resource group for the CTI Connector, on page 179 2. Procedure: Configuring the gateway resource for CTI through IVR Server, on page 188 3. Procedure: Flagging the IVR Profile for CTI, on page 190 **Key Actions** • Create a resource group of type `CTI` for the CTI Connector. • To enable CTI flagging: 1. Set `CTI USage` to `Based On DN Lookup` (`use-cti` to `2`). 2. For non-CTI applications, set `CTI Allowed` in the IVR Profile to `false`.

Task Summary: Carrier-Connected, IVR Behind (Continued)

Objective	Related Procedures and Actions
Additional Special Configuration	
To fetch the script ID from URS...	If your switch cannot supply the DNIS for the incoming call, you must configure CTI Connector to fetch the script ID from URS. CTI Connector uses the value of this key as the DNIS in the `INVITE` to Resource Manager for IVR Profile selection: 1. In URS, define the key-value pair for the script ID: • Key—Define the key name that you will match in the CTI Connector application. The default is `scriptidkeyname`. • Value—Define the value that matches the DNIS used to select the desired IVR Profile in Resource Manager. 2. In the CTI Connector `Application` object, set the following options in the `IVRSC` section: • `Fetch Script ID from URS`—Enter `1` to enable CTI Connector to fetch the defined key from URS. • `Script Id Key Name`—Enter the key name as defined in URS. CTI Connector sends this key to retrieve the value for the script ID.
Integrating SSG into CTI-C deployments.	Outbound calls initiated by SSG cannot go through CTI Connector. For deployments that include CTI-Connector, configure as follows: 1. In this behind-mode configuration, CTI Usage is already set to Based on DN Lookup (`use-cti=2`). No further configuration is required. 2. In the IVR Profile, configure `CTI Allowed` as follows: • For outbound calls—Set `CTI Allowed` to `false`. Resource Manager will bypass CTI-C. • For inbound calls, your IVR Profiles should already be configured to allow or disallow CTI. No further configuration is required. For more information, see "Configuring CTI Flagging for IVR Profiles" on page 188.

Task Summary: Carrier-Connected, IVR Behind (Continued)

Objective	Related Procedures and Actions
No CTI flagging required...	If your deployment does not require CTI flagging (all calls will go through CTI Connector), then modify the above procedures as follows: • Set `CTI Usage` in the SIP Gateway to `Always On` (`use-cti=1`). • Set `GetDNISFromIServer` in the CTI Connector to `true`. • Do not configure the `IVRPortBaseIndex` or `MaxIVRPorts` options in the CTI Connector. • Do not create `Voice Treatment Port` DNs in the dummy switch for TServer_IVR. • In the `I-Server` application, add a connection to SIP Server, not `TServer_IVR`. • Map the IVR Ports to the DNs on the SIP Server switch, not the dummy switch for `TServer_IVR`.
To configure the socket layer...	1. In the `TServer_IVR` application, configure the listening port for the socket layer in the GLI sections: ◆ `gli_server > gli-server-mode`—Enter `circuit` to enable the server to open multiple circuits. ◆ `gli_server > gli-n-servers`—Enter the number of circuit groups the server can open. 2. Create a `gli-server-group_n` section for each circuit group: ◆ `gli-server-address`—Enter the IP address and listening port, in the format `<ip_address>:<port>`. 3. In the CTI Connector `Application` object, set the `iserversocket` option to the GLI port as configured in Step 2. (For configuration details, see Procedure: Configuring the CTI Connector for IVR Behind, on page 174.)

Task Summary: IVR Behind, TDM-Connected Integration

Task Summary: TDM-Connected, IVR Behind, on page 152 provides an overview of the main steps that you must complete in order to integrate the VPS with IVR Server in the Behind mode, in a TDM deployment.

Task Summary: TDM-Connected, IVR Behind

Objective	Related Procedures and Actions
1. Log in to Genesys Administrator.	In your web browser, enter the application URL in the following format: http://<genesys_administrator_host>/wcm
2. Verify prerequisites.	Before starting the integration, verify the following prerequisites have been met: • The basic Voice Platform Solution is deployed. • The IVR Server and related objects have been created, and the IVR Server is installed. • The CTI Connector application has been created, and the application is installed. For a more detailed description of this task, see Procedure: Task Summary: Verify Prerequisites, on page 145.
3. Configure the SIP Server for Behind mode integration.	Go to: `Provisioning > Environment > Applications` Configure the SIP Server options as follows: • Set `override-to-on-divert` to `false`. • Set `event-ringing-on-100trying` to `true`. • Set `handle-vsp` to `all`. • Add connections to the URS, Stat Server, and Message Server. For the detailed procedure, see Procedure: Configuring the SIP Server for Behind mode, on page 169.

4. Configure `Trunk` DN.	Go to `Provisioning` > `Switching` > `Switches` > and select your SIP Server `Switch` object.
	1. On the `DNs` tab, click `New` and create a DN of type `Trunk`.
	2. On the `Options` tab, create a `TServer` section with the options:
	  • `contact`—Set this to the IP address and port number for Resource Manager.
	  • `prefix`—Set this to the initial digits that will match the range of port numbers coming in from the gateway. Prefix matching lets you create a single DN for a range of incoming port numbers.
	Key Actions
	• Before creating the `Trunk` DN, check whether the port numbers are coming in from the media gateway or the PBX.

Task Summary: TDM-Connected, IVR Behind (Continued)

Objective	Related Procedures and Actions
5. Configure the IVR Server.	For step-by-step procedures, see the following: 1. Procedure: Creating the dummy switching office, on page 171 2. Procedure: Creating the dummy switch for TServer_IVR, on page 171 3. Procedure: Configuring the TServer_IVR object, on page 172 4. Procedure: Adding required connections to the I-Server, on page 172 **Key Actions** • Create the DNs on the premise T-Server switch. • Associate `TServer_IVR` with a dummy switch. • In TServer_IVR, configure the `Router Timeout` option to a value of `7 s`. • In TServer_IVR, configure `active-release` to `false`. • In the `I-Server` object, add a connection to the premise T-Server.
6. Add connections to the premise T-Server.	In the premise T-Server `Application` object, add connections to the following: • Universal Routing Server • Stat Server • Message Server

7. Create the `IVR` object to represent GVP.	Go to: `Provisioning > Switching > IVRs`
	1. Create an `IVR` object to represent GVP:
	  • Select `Genesys Voice Platform` as the type.
	  • Select your `I-Server` application as the IVR Server.
	2. Click `Save` to register with the Configuration Layer.
	For a detailed procedure, see Procedure: Creating an IVR object to represent GVP in the solution, on page 173.

Task Summary: TDM-Connected, IVR Behind (Continued)

Objective	Related Procedures and Actions
8. Configure the CTI Connector.	Go to: Provisioning > Environment > Applications 1. Open your CTI Connector Application object. 2. On the Options tab, configure the following mandatory parameters: • Resource Manager IP Address • IVR Client Name • IVR Server Host IP Address • IVR Server Communication Port • Local Host Name 3. On the Options tab, in the CTIC section, set Fetch DNIS from IServer to true. For a detailed procedure, see Procedure: Configuring the CTI Connector for IVR Behind, on page 174.
9. Configure IVR Ports.	To configure IVR channels as Voice Treatment Port DNs, complete the following procedures: 1. Procedure: Creating Voice Treatment Port DNs, on page 175 2. Procedure: Mapping IVR Ports to their matching DNs, on page 177 3. Procedure: Configuring a Place object for each IVR Port, on page 178 4. Procedure: Configuring a Place Group as the target for IVR Ports, on page 178 **Key Rules** • Create the DNs in premise T-Server switch (no configuration is required). • Use identical numbering for the DNs, ports, and places. • Map each IVR port to the appropriate Voice Treatment Port DN on the premise switch. • Use continuous numbering (in increments of one) when naming the DNs.

Task Summary: TDM-Connected, IVR Behind (Continued)

Objective	Related Procedures and Actions
10. Configure Resource Manager for integration with CTI Connector.	Go to: `Provisioning > Voice Platform > Resource Groups` For step-by-step procedures, see the following: 1. Procedure: Creating a resource group for the CTI Connector, on page 179 2. Procedure: Configuring the gateway resource for CTI through IVR Server, on page 188 3. Procedure: Flagging the IVR Profile for CTI, on page 190 **Key Actions** • Create a resource group of type `CTI` for the CTI Connector. • Set `CTI Usage` to `Always On` (`use-cti=1`).
Additional Special Configuration	
Integrating SSG into CTI-C deployments.	Outbound calls initiated by SSG cannot go through CTI Connector. For deployments that include CTI-Connector, configure as follows: 1. Set `CTI Usage` on the SIP Server gateway resource group to `Based on DN Lookup` (`use-cti=2`). Resource Manager checks the IVR Profile to decide about CTI-C usage. 2. In the IVR Profile, configure `CTI Allowed` as follows: • For outbound calls—Set `CTI Allowed` to `false`. Resource Manager will bypass CTI-C. • For inbound calls—Set `CTI Allowed` to `true`. Resource Manager will process the call through CTI-C. For more information, see “Configuring CTI Flagging for IVR Profiles” on page 188.

Task Summary: TDM-Connected, IVR Behind (Continued)

Objective	Related Procedures and Actions
To fetch the script ID from URS...	If your switch cannot supply the DNIS for the incoming call, you must configure CTI Connector to fetch the script ID from URS. CTI Connector uses the value of this key as the DNIS in the `INVITE` to Resource Manager for IVR Profile selection: 1. In URS, define the key-value pair for the script ID: • Key—Define the key name that you will match in the CTI Connector application. The default is `scriptidkeyname`. • Value—Define the value that matches the DNIS used to select the desired IVR Profile in Resource Manager. 2. In the CTI Connector `Application` object, set the following options in the `IVRSC` section: • `Fetch Script ID from URS`—Enter `1` to enable CTI Connector to fetch the defined key from URS. • `Script Id Key Name`—Enter the key name as defined in URS. CTI Connector sends this key to retrieve the value for the script ID.
To configure the socket layer...	1. In the `TServer_IVR` application, configure the listening port for the socket layer in the GLI sections: • `gli_server > gli-server-mode`—Enter `circuit` to enable the server to open multiple circuits. • `gli_server > gli-n-servers`—Enter the number of circuit groups the server can open. 2. Create a `gli-server-group_n` section for each circuit group: • `gli-server-address`—Enter the IP address and listening port, in the format `<ip_address>:<port>`. 3. In the CTI Connector `Application` object, set the `iserversocket` option to the GLI port as configured in Step 2. (For configuration details, see Procedure: Configuring the CTI Connector for IVR Behind, on page 174.)

Task Summary: Integrating with IVR Server—In-Front Mode

With IVR Server in an In-Front configuration, the call arrives at the VPS directly from the PSTN, with the DNIS available for immediate IVR Profile mapping. No DN lookup through an IVR Profile is required for call setup. However, the call is still forwarded to CTI Connector and IVR Server for call registration—typically for reporting purposes.

For a sample configuration and a description of a typical call flow, see "Integration with IVR Server—In-Front Mode" on page 94.

Task Summary: Integrating IVR Server—In-Front Mode, on page 157 provides an overview of the main steps that you must complete in order to integrate the VPS with an IVR Server configured for In-Front mode.

Task Summary: Integrating IVR Server—In-Front Mode

Objective	Related Procedures and Actions
1. Log in to Genesys Administrator.	In your web browser, enter the application URL in the following format: `http://<genesys_administrator_host>/wcm`
2. Verify prerequisites.	Before starting the integration, verify the following prerequisites have been met: • The basic Voice Platform Solution is deployed. • The IVR Server and related objects have been created, and the IVR Server is installed. • The CTI Connector application has been created, and the application is installed. For a more detailed description of this task, see Procedure: Task Summary: Verify Prerequisites, on page 145.
3. Configure the IVR Server.	For step-by-step procedures, see the following: 1. Procedure: Creating the virtual switching office, on page 180 2. Procedure: Creating the virtual switch for TServer_IVR_InFront, on page 181 3. Procedure: Adding required connections to the TServer_IVR_InFront, on page 181 4. Procedure: Adding required connections to the I-Server_InFront, on page 182 **Key Actions** • Associate `TServer_IVR_InFront` with the virtual switch. Add a connection to the SIP Server. • In the `I-Server` object, add connections to SIP Server and `TServer_IVR_InFront`.
4. Create the `IVR` object to represent GVP.	Go to: `Provisioning > Switching > IVRs` 1. Create an `IVR` object to represent GVP: • Select `Genesys Voice Platform` as the type. • Select your `I-Server` application as the IVR Server. 2. Click `Save` to register with the Configuration Layer. For a detailed procedure, see Procedure: Creating an IVR object to represent GVP in the solution, on page 182

Task Summary: Integrating IVR Server—In-Front Mode (Continued)

Objective	Related Procedures and Actions
5. Configure the CTI Connector.	Go to: `Provisioning > Environment > Applications` 1. Open your CTI Connector `Application` object. 2. On the `Options` tab, configure the following mandatory parameters: • `Resource Manager IP Address` • `IVR Client Name` • `IVR Server Host IP Address` • `IVR Server Communication Port` • `Local Host Name` 3. On the `Options` tab, configure the following IVR port options: • `IVRPort Base Index` • `Max IVRPorts` **Note:** These IVR port options are optional for In Front configurations. If the gateway provides the IVRPort in the incoming `INVITE`, these options are not required. For a detailed procedure, see Procedure: Configuring the CTI Connector for IVR In-Front, on page 183.
6. Configure IVR Ports.	To configure mapping for IVR ports generated by the CTI Connector, complete the following procedures: 1. Procedure: Creating Voice Treatment Port DNs, on page 184 2. Procedure: Mapping IVR Ports to their matching DNs, on page 185 **Key Rules** • For In-Front mode, create `Voice Treatment Port` DNS on the virtual switch. • Use identical numbering for the DNs and ports. • Use continuous numbering (in increments of one) when naming the DNs.

Task Summary: Integrating IVR Server—In-Front Mode (Continued)

Objective	Related Procedures and Actions
7. Configure Resource Manager for integration with CTI Connector.	Go to: `Provisioning > Voice Platform > Resource Groups` For step-by-step procedures, see the following: 1. Procedure: Creating a resource group for the CTI Connector, on page 185 2. Procedure: Configuring the gateway resource for CTI through IVR Server, on page 188 3. Procedure: Flagging the IVR Profile for CTI, on page 190 **Key Actions** • Create a resource group of type `CTI` for the CTI Connector. • For TDM deployments, set `CTI Usage` to `Always On` (`use-cti=1`). • For carrier-connected deployments: 1. Set `CTI Usage` to `Based On DN Lookup (use-cti=2)`. 2. For non-CTI applications, set `cti-allowed` in the IVR Profile to `false`.
8. Enable midcall CTI routing for IVR-centric applications.	For midcall CTI transfers, you must configure Access Codes for interswitch communication between the SIP Server switch and the virtual switch for GVP (In-Front mode only): 1. Procedure: Configuring access from the SIP Server to GVP, on page 191 2. Procedure: Configuring access from GVP to the SIP Server, on page 192 3. Verify that the SIP Server switch includes a connection to the `TServer_IVR_InFront` object. 4. When configuring the `AccessNumGet` block in Composer, provision the required parameters as follows: • `Destination DN`—Enter the number for the `External Routing Point` DN you created on the SIP Server switch. • `Location`—Select the SIP Server switch from the drop-down list. 5. Configure the `<transfer>` tag in the VoiceXML application to use the number received in the `AccessNumGet` block. This can be either a Blind or Bridge transfer (applies to both NGI and GVPi applications).

Task Summary: Integrating IVR Server—In-Front Mode (Continued)

Objective	Related Procedures and Actions
Additional Special Configuration	
Integrating SSG into CTI-C deployments.	Outbound calls initiated by SSG cannot go through CTI Connector. For deployments that include CTI-Connector, configure as follows: 1. In this in front-mode configuration, `CTI Usage` is already set to `Based On DN Lookup` (`use-cti=2`). No further configuration is required. 2. In the IVR Profile, configure CTI Allowed as follows: • For outbound calls—Set `CTI Allowed` to `false`. Resource Manager will bypass CTI-C. • For inbound calls, your IVR Profiles should already be configured to allow or disallow CTI. No further configuration is required. For more information, see “Configuring CTI Flagging for IVR Profiles” on page 188.

Task Summary: Integrating with Network IVR Server

With IVR Server in a Network configuration, the IVR Server does not map incoming calls to IVR Port numbers. Instead of forwarding the port number to IVR Server as in other modes, the CTI Connector sends the number used to call the solution—either the DNIS or a toll free Number—as the `Called Number` in the `NewCall` request to the IVR Server.

For a sample architecture diagram and a description of a typical call flow, see “Integration with IVR Server—In-Front Mode” on page 94.

Task Summary: Integrating with Network IVR Server provides an overview of the main steps that you must complete in order to integrate the VPS with a Network IVR Server.

Task Summary: Integrating with Network IVR Server

Objective	Related Procedures and Actions
1. Log in to Genesys Administrator.	In your web browser, enter the application URL using the following format: `http://<genesys_administrator_host>/wcm`

Task Summary: Integrating with Network IVR Server (Continued)

Objective	Related Procedures and Actions
2. Create an IVR Network switch.	Go to: `Provisioning > Switching > Switching Offices` 1. Create a `Switching Office` object with the switch type `GenSpec XML`. 2. Create a `Switch` object and: • Associate it with the Network IVR switching office. • Select `TServer_IVR_810_Network` as the associated T-Server. For more information, see the "Sample Configurations" chapter in the IVR Interface Option 8.1 IVR Server System Administrator's Guide.
3. Configure inter-switch access.	Go to: `Provisioning > Switching > Switches` 1. In the Network T-Server switch, configure the Access Code used to reach the IVR Network switch, and then create the corresponding access resource DN in the Network T-Server switch. 2. In the IVR Network switch, configure the access code used to reach the Network T-Server switch, and then create the corresponding access resource DN in the Network T-Server switch. For more information, see the "Configuring Multi-site Support" chapter in the *IVR Interface Option 8.1 IVR Server System Administrator's Guide*.
4. Configure the CTI Connector.	Go to: `Provisioning > Environment > Applications` 1. Open your CTI Connector `Application` object. 2. On the `Options` tab, configure the following mandatory parameters: • `Resource Manager IP Address` • `IVR Client Name` • `IVR Server Host IP Address` • `IVR Server Communication Port` • `Local Host Name` 3. On the `Options` tab, configure the following network mode option: • `Use Called Number`—If customers use a toll-free number to contact the VPS, set this option to `TFN`. For a detailed procedure, see Procedure: Configuring the CTI Connector for Network IVR Server, on page 187.

Task Summary: Integrating with Network IVR Server (Continued)

Objective	Related Procedures and Actions
5. Configure the IVR Profile.	If customers use a toll-free number to contact the VPS, you must configure this number in the IVR Profile. • From the `Service Properties` page in the IVR Profile Wizard, enter the number in the `Toll Free Number` box.

Task Summary: Configuring Midcall CTI Functions

Task Summary: Configuring Midcall CTI Functions provides an overview of the different midcall CTI actions available for IVR-centric applications, and any additional configuration steps required to enable them.

Task Summary: Configuring Midcall CTI Functions

Objective	Related Procedures and Actions
Configuring Peek/Get stat results	1. Create a new section in the `I-Server` application for the statistic you want to gather—for example: `ExpectedWaitTime`. 2. For this sample section, configure the following options: • `obj_id`—Set this to `<dn>@<switch>`. • `obj_type`—Set this to `SObjectQueue`. • `server_name`—Set this to `<stat server name>`. • `stat_type`—Set this to `ExpectedWaitTime`. • `update_frequency`—Set this to `5`. 3. Create a matching section in the Stat Server application object—for example: `ExpectedWaitTime`, with the above options. 4. Verify that the required connections are in place: • TServer_IVR — Message Server • I-Server — TServer_IVR, Stat Server • URS — TServer_IVR, Stat_Server, Message Server • Stat Server — TServer_IVR, Message Server 5. Configure the `Peek/Get Stat` block in the voice application, using Composer.

Task Summary: Configuring Midcall CTI Functions (Continued)

Objective	Related Procedures and Actions
Enable midcall CTI routing for IVR-centric applications.	For midcall CTI transfers, you must configure Access Codes for inter-switch communication between the SIP Server switch and the virtual switch for GVP (IVR In-Front mode only): 1. Procedure: Configuring access from the SIP Server to GVP, on page 191 2. Procedure: Configuring access from GVP to the SIP Server, on page 192 3. Verify that the SIP Server switch includes a connection to the `TServer_IVR_InFront` object. 4. When configuring the `AccessNumGet` block in Composer, provision required parameters as follows: • `Destination DN` —Enter the number for the `External Routing Point` DN you created on the SIP Server switch. • `Location` —Select the SIP Server switch from the drop-down list.

Task Summary: Configuring Transfers

Task Summary: Configuring the IVR Profile for Transfer Type provides an overview of the service parameter configuration steps required to enable different transfer types in the IVR Profile for your voice application. 0

Note: The VPS uses the default agent number that you specify in the `Default Agent` option in the IVR Profile as the target for the transfer, in cases where the IVR Server returns the default agent number due to a timeout or other error.

Task Summary: Configuring the IVR Profile for Transfer Type

Transfer Type	Related Procedures and Actions
Blind Transfer (Using `OneStepXfer` through CTI)	On the `Options` tab of the IVR Profile: 1. In the `gvp.policy` section, configure the following options: • `Transfer Allowed`—Set the value for this option to `true`. 2. In the `gvp.service-parameters` section, configure the following options: • `cti.DefaultAgent`—Set the value for this option to `fixed,<Agent DN>` • `cti.TransferOnCTI`—Set the value for this option to **`fixed,yes`**. **For GVPi applications only:** • `voicexml.gvpi.$transfer-type$`—For **GVPi** applications only, set the value for this option to `fixed,1SignalChannel`. • `voicexml.gvpi.$transfer-option$`—For **GVPi** applications only, set the value for this option to `fixed,SipRefer`.
Blind Transfer (Using `REFER` on SIP Server	On the `Options` tab of the IVR Profile: 1. In the `gvp.policy` section, configure the following options: • `Transfer Allowed`—Set the value for this option to `true`. 2. In the `gvp.service-parameters` section, configure the following options: • `cti.DefaultAgent`—Set the value for this option to `fixed,<Agent DN>` • `cti.TransferOnCTI`—Set the value for this option to **`fixed,no`**. **For GVPi applications only:** • `voicexml.gvpi.$transfer-type$`—For **GVPi** applications only, set the value for this option to `fixed, 1SignalChannel`. • `voicexml.gvpi.$transfer-option$`—For **GVPi** applications only, set the value for this option to `fixed,SipRefer`.

Task Summary: Configuring the IVR Profile for Transfer Type (Continued)

Transfer Type	Related Procedures and Actions
Bridge Transfer (Using INVITE on SIP Server)	On the Options tab of the IVR Profile: 1. In the gvp.policy section, configure the following options: • Transfer Allowed—Set the value for this option to enabled. • outboundcallallowed—Set the value for this option to enabled 2. In the gvp.service-parameters section, configure the following options: • Default Agent—Set the value for this option to fixed, <Agent DN> • voicexml.gvpi.$transfer-type$—For **GVPi** applications only, set the value for this option to fixed, 2SignalChannel. • voicexml.gvpi.$transfer-option$—For **GVPi** applications only, set the value for this option to not set.
Media Redirect Transfer	**For GVPi applications:** 1. In the gvp.service-parameters section of the IVR Profile, set voicexml.gvpi.$transfer-type$ to 2 Signal Channel. 2. In the MCP application, set defaultbridgexfer to MEDIAREDIRECT. **Note**: If voicexml.gvpi.$cti_endcall_on_agentleg_hup$ is set to false in the IVR Profile, MCP will ignore the MEDIAREDIREC parameter, and force a BRIDGE transfer instead. **For NGI applications:** 1. Set the method attribute in the VoiceXML <transfer> tag to MEDIAREDIRECT. OR In the MCP application, set set defaultbridgexfer to MEDIAREDIRECT. 2. Set the connectwhen attribute in the VoiceXML <transfer> tag to answered. (If you set this to immediate, the transfer will fail.) 3. In the sessmgr section of the MCP application, set sendsdpininvite to false.

Task Summary: Configuring Voice Treatments

Task Summary: Additional Voice Treatment Configuration, on page 166 provides an overview of the solution-level configuration steps required in addition to the configuration of basic Interaction Routing Designer (IRD) Voice Treatment object parameters. For information about basic Voice Treatment block configuration, see the "Voice Treatment Options" section of the *Universal Routing 8.1 Reference Manual*.

Task Summary: Additional Voice Treatment Configuration

Treatment Type	Related Procedures and Actions
Play Application	**GVPi Applications** For GVPi applications, you must configure both the `APP_ID` parameter in the routing strategy and the `ScriptID` variable in the Studio application with the same value. For example: 1. In the `Play Application` block of the routing strategy, set `APP_ID` to `2`. 2. In the `Branching` block of the Studio application, set the `ScriptID` to `2`. **NGI Applications** For NGI applications, you can configure Play Application treatments in one of two ways: • In the routing strategy `Play Application` block, configure the `{s}APP_URL` parameter with a value that specifies the fully qualified URL of the VoiceXML application. OR • If you do not include the `{s}APP_URL` in the routing strategy, you must configure the IVR Profile to include the URL instead: 1. In the `gvp.service-prerequisite` section of the IVR Profile, set the `script-url` option to the URL of the VoiceXML application that will provide the treatment. 2. If the `script-url` option is not configured, Resource Manager instead uses the value of the `initial-page-url` option from the IVR Profile. **Sending User Data back to the Application** Using CTI Connector, `PlayApplication` treatments can pass all forms of interaction data (`CED`, `UData`, `ExtnsEx`) back to the strategy.

Task Summary: Additional Voice Treatment Configuration (Continued)

<table>
<tr><th>Treatment Type</th><th>Related Procedures and Actions</th></tr>
<tr><td>Play Announcement</td><td>GVPi Applications
1. In the routing strategy, add a <code>Play Announcement</code> block with empty parameter values.
2. In the Studio application, configure a property in the <code>Branching</code> block as follows:
• <code>Node</code>—Can be any identifier.
• <code>Variable</code>—<code>ScriptID</code>
• <code>Value</code>—<code>TRT:PlayAnnounce</code>, chosen from the drop-down list

NGI Applications
MCP provides a default VoiceXML application to handle <code>PlayAnnounce</code> treatments. If you want to use a different application, configure the following:
1. In the <code>IVRSC</code> section of the CTI Connector <code>Application</code> object, set the <code>PlayAnnounce Resource Path</code> parameter to the location of the VoiceXML application you want to use.
2. If this parameter is not set, Resource Manager instead uses the value of the <code>script-url</code> option from the IVR Profile.

Sending User Data back to the Application
No interaction data is passed back to the strategy at the end of this treatment.</td></tr>
</table>

Task Summary: Additional Voice Treatment Configuration (Continued)

Treatment Type	Related Procedures and Actions
Play Announcement and Collect Digits	**GVPi Applications** 1. In the routing strategy, add a `Play Announcement and Collect Digits` block with empty values. 2. In the Studio application, configure a property in the Branching block as follows: • `Node`—Can be any identifier. • `Variable`—`ScriptID` • `Value`—`TRT:PlayAnnounceAndDigits`, chosen from the drop-down list **NGI Applications** MCP provides a default VoiceXML application to handle `PlayAnnounceAndDigit` treatments. If you want to use a different application, configure the following: 1. In the IVRSC section of the CTI Connector Application object, set the PlayAnnounceAndDigit Resource Path parameter to the location of the VoiceXML application you want to use. 2. If this parameter is not set, Resource Manager instead uses the `script-url` option from the IVR Profile. **Sending User Data back to the Application** For NGI—`CED` interaction data is passed back using the `<exit>` tag with the `namelist` attribute, which is translated into the body of the `INFO` message at the end of the treatment. For GVPi—Use the `treatment result` block in Studio to specify what data should be sent.

Task Summary: Additional Voice Treatment Configuration (Continued)

Treatment Type	Related Procedures and Actions
Play Music	**GVPi Applications** 1. In the routing strategy: • Add a `Play Music` block with empty values. • Add a `Pause` block, with a duration a few seconds longer than the treatment itself. 2. In the Studio application: • Configure the `Branching` block with a new property, where the `ScriptID` equals `TRT:Music`. • Specify the same duration in this block as you did in the routing strategy. **NGI Applications** Configure the routing strategy as follows: 1. Add a `Music` block with the following parameters: • `MUSIC_DN`—Enter the http path of the music file. • `DURATION`—Enter the duration of the music file in ms. 2. Add a `Pause` block with the following parameters: • `DURATION`—Enter a duration a few seconds longer than the duration of the music file itself.

Integrating with IVR Server in Behind Mode—Procedures

Complete the procedures in this section to integrate the Voice Platform Solution with the IVR Server in a Behind mode configuration.

Procedure: Configuring the SIP Server for Behind mode

Purpose: To perform additional configuration of the SIP Server `Application` object for VPS integration with IVR Server in a Behind mode configuration.

Start of procedure

1. Go `Provisioning > Environment > Applications`, and click the SIP Server `Application` object.

2. On the `Options` tab, in the `TServer` section, configure the options as described in Table 7.

Table 7: SIP Server Options—TServer Section

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

3. In the `extrouter` section, configure the `handle-vsp` option as described in Table 8.

Table 8: SIP Server Options—Extrouter Section

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

4. Click `Save and close`.

End of procedure

Next Steps

- Procedure: Creating the dummy switching office, on page 171

Procedure: Creating the dummy switching office

Start of procedure

1. Go to `Provisioning > Switching > Switching Offices`.
2. Click `New`.
3. Enter a name, and select any type of switch. Genesys recommends `Virtual Switch for IVR In-Front`.
4. Click `Save`.

End of procedure

Next Steps

- Procedure: Creating the dummy switch for TServer_IVR

Procedure: Creating the dummy switch for TServer_IVR

Start of procedure

1. Go to `Provisioning > Switching > Switches`.
2. Click `New` and provision the switch as follows:
   - `Name`—Enter a name for the dummy switch.
   - `Switching Office`—Click the browse icon and select the dummy switching office that you created in "Creating the dummy switching office".
   - `TServer`—Click the browse icon and select the `Tserver_IVR`.
3. Click `Save`.

   Tip: This dummy switch performs a logical function for IVR Server. No DNs or Agent Logins are required.

End of procedure

Next Steps

- Procedure: Configuring the TServer_IVR object, on page 172

Procedure: Configuring the TServer_IVR object

Purpose: To add the required connections and configure mandatory options required to integrate the TServer_IVR into the VPS (with IVR Server in Behind mode).

Suggested Configuration Option Settings for Behind Mode

For integration with IVR Server in Behind mode, Genesys recommends setting the following options:

- `Router Timeout`—Genesys recommends increasing the length of time that elapses before URS will route to a default DN, in cases where there is a delay in providing the next treatment request to IVR Server.
- `active-release`—Set this option to `false` to prevent the IVR Server from forwarding an end call request received from GVP, since GVP should receive these messages from SIP Server only. This avoids the race condition where the end call message is sent/received by IVR Server.

Start of procedure

1. Go to `Provisioning > Environment > Applications` and select `TServer_IVR`.
2. On the `Options` tab, in the `Timers` section, create the following new option:
   - `Router Timeout`—Change the default value for this option from `4 s` to `7` s. IVR Server waits this length of time before it performs default routing for the call.
3. On the `Options` tab, in the `IServer` section, create the following new option:
   - `active-release`—Set this to `false`.
4. Click `Save and close`.

End of procedure

Next Steps

- Procedure: Adding required connections to the I-Server

Procedure: Adding required connections to the I-Server

Start of procedure

1. Go to `Provisioning > Environment > Applications and select I-Server`.

2. On the `Configuration` tab, under `Connections`, add connections as follows:
   - CTI flagging enabled (carrier-connected)—Add a connection to the `TServer_IVR`.
   - CTI flagging disabled (carrier-connected)—Add a connection to the SIP Server `Application` object.
   - TDM-connected—Add a connection to the premise T-Server `Application` object.
3. Click `Save`.

End of procedure

Next Steps

- Procedure: Creating an IVR object to represent GVP in the solution

Procedure: Creating an IVR object to represent GVP in the solution

Start of procedure

1. Go to `Provisioning > Switching > IVRs`.
2. Click `New`, and configure the parameters `Configuration` tab as follows:
   - `Name`—Enter a name for this IVR.
   - `Description`—Enter a useful description of this IVR. For example, the switch configuration, or the IVR mode.
   - `Type`—Select `Genesys Voice Platform` as the type.
   - `Version`—Enter the software release number for the GVP that this IVR will represent. For example, `8.1`.
3. Click `Save`.

End of procedure

Next Steps

- After you save the `IVR` object, the `IVR Port` tab appears. Use this tab to create the IVR Ports that you map to the corresponding DNs on the SIP Server switch.

  Before creating the IVR Ports, configure CTI Connector for integration with the IVR Server in Behind mode. See Procedure: Configuring the CTI Connector for IVR Behind, on page 174.

Procedure:
Configuring the CTI Connector for IVR Behind

Purpose: To configure the CTI Connector `Application` object for integration with the IVR Server in Behind mode.

Summary

In an IVR behind mode integration, the key configuration option is `Fetch DNIS from IServer`. When set to `true`, the VPS sends a request to the IVR Server in order to retrieve the DNIS required to map the IVR Profile. In a TDM deployment, where the DNIS is not available at the outset of the call, this setting is mandatory. In carrier-connected deployments, however, where the DNIS is available with the incoming call, you can configure the gateway resource to perform CTI flagging in order to skip CTI Connector (and IVR Server) for Standard VoiceXML applications that do not require any CTI functionality. In this case, you can set `Fetch DNIS from IServer` to `false`, and the VPS will retrieve the DNIS from the SIP header.

Prerequisites

- The CTI Connector `Application` object is created and the component is installed. For an overview of the deployment process, see Task Summary: Verify Prerequisites, on page 146.

Start of procedure

1. Go to `Provisioning > Environment > Applications > <CTI_Connector>`.
2. On the `Options` tab, select `Mandatory Options` from the `View` drop-down list and configure the following mandatory parameters:
   - `Resource Manager IP Address`—Enter the IP address and SIP port for the Resource Manager in the following format:
     `<RM_ip_address>:<RM_sip_port>`
     For example:
     `10.10.10.10:5060`
   - `IVR Client Name`—Enter the login name of the IVR Server `Application` object, as it appears in Management Framework.
   - `IVR Server Host IP Address`—Enter the IP address of the IVR Server host.
   - `IVR Server Communication Port`—Enter the GLI port number as configured in the `gli-server-address` option of the `TServer_IVR` object.
   - `Local Host Name`—Enter the IP address of the CTI Connector host.

Tip: The application template provides an ISserver_Sample section that you can configure as-is, rename, or use as a model. Add a new section for every additional IVR Server in your configuration. If you rename this section, or add any new sections, make sure you add the section name to the Customer IServers List parameter in the IVRSC section.

3. For TDM-connected deployments, select Advanced View (Options) from the View drop-down list and configure the following parameters:
   - Fetch DNIS From IServer—Set this option to true.
   - Default DNIS—If Fetch DNIS from IServer is enabled, enter a default DNIS to be used in case the IVR Server fails to return the DNIS during the fetch operation.

   Tip: For carrier-connected deployments (CTI flagging enabled as recommended), skip this step. The default settings are acceptable.

4. For carrier-connected deployments, configure the following parameters:

Carrier-Connected with CTI Flagging

- IVRPort Base Index—Set the starting number for the range of IVR Ports to match the Voice Treatment Port DN range that you will later create in the SIP Switch and in the dummy switch for TServer_IVR.
- Max IVRPorts—Set the maximum number of IVR Ports to the total number of Voice Treatment Port DNs that GVP allows (you will create this number of DNs in the next procedure).

Tip: For carrier-connected deployments, in which CTI flagging is not required, you can skip this step and instead set Fetch DNIS from IServer to true and add a value for Default DNIS (see Step 3).

5. Click Save.

End of procedure

Next Steps

- Procedure: Creating Voice Treatment Port DNs

Procedure: Creating Voice Treatment Port DNs

Purpose: To create and configure Voice Treatment Port DNs on the appropriate switch. These DNs will be mapped to IVR Ports configured in the IVR object.

Summary

Carrier-Connected For carrier-connected deployments with CTI flagging enabled, create one set Voice Treatment Port DNs in the SIP Server switch.

TDM-Connected For TDM-connected deployments, create one set of `Voice Treatment Port` DNs in the premise T-Server switch (no option configuration required). The numbers for these DNs must match the port numbers coming in from the media gateway or PBX.

Start of procedure

1. Go to `Provisioning > Switching > Switches > <your_switch>`.
2. On the `DNs` tab, click `Add`.
3. On the `Configuration` tab, configure the following mandatory parameters:
   - `Number`—Assign the new DN a number.
   - `Type`—For a regular DN that is always ready to receive calls (no agent login required), select `Voice Treatment Port` from the drop-down list.
4. If you created the DN in the SIP Server switch, on the `Options` tab, click `New` and do the following:
   a. Enter `TServer` as the section name.
   b. Enter `contact` as the option name.
   c. For the value, enter the IP address and port of the Resource Manager:
      `sip:<RM_ip_address>:<RM_sip_port>`

   Tip: Skip this step if creating DNs in either the premise T-Server switch or in the dummy switch for `TServer_IVR`.

5. On the `Options` tab, in the `TServer` section, click `New` and add the following option:
   - `event-ringing-on-100trying`—Set this option to `true`.
6. To create another DN, click `Save and New`. Create as many new `Voice Treatment Port` DNs as GVP allows, provisioning each as you did in the preceding steps.

   Tip: Use continuous numbering (in increments of one) when naming all subsequent DNs.

7. Click `Save and Close`.

End of procedure

Next Steps

- For carrier-connected deployments with CTI flagging enabled, you must create another set of matching DNs in the dummy switch for `TServer_IVR`. Repeat the above configuration steps, but in the dummy switch instead.

- If no CTI flagging is required in your deployment, continue at Procedure: Mapping IVR Ports to their matching DNs.

Procedure: Mapping IVR Ports to their matching DNs

Purpose: To create IVR Ports in the IVR object and link them to the Voice Treatment Port DNs configured on the switch. SIP Server manages these ports for incoming calls, forwarding the port number to the Resource Manager in the INVITE.

Start of procedure

1. Go to Provisioning > Switching > IVRs.
2. Double-click the IVR object you created to represent GVP in the solution.
3. On the IVR Ports tab, click Add.
4. Enter the Port Number as follows:
   - For carrier-connected deployments—enter the same number as you did for the Voice Treatment Port DN that you want to map this port to.
   - For TDM-connected deployments—Enter the port number of the channel on the switch (or media gateway).
5. To enter the Associated DN do the following:
   a. Click the browse icon, then select one of the following switches:
      - CTI flagging enabled (carrier-connected)—Select the dummy switch for TServer_IVR.
      - CTI flagging disabled (carrier-connected)—Select the SIP Server switch.
      - TDM-connected—Select the premise T-Server switch.

   b. In the DNs folder, select the matching Voice Treatment Port DN.
6. Click Save & New. Create additional IVR Ports until all matching Voice Treatment Port DNs are mapped.
7. Click Save and Close.

End of procedure

Next Steps

- Procedure: Configuring a Place object for each IVR Port, on page 178

Procedure: Configuring a Place object for each IVR Port

Purpose: To create a Place object for every corresponding Voice Treatment Port DN mapped to an IVR Port. Incoming calls arriving on a Routing Point DN will be targeted to these places, allowing SIP Server to pass the port number to the Resource Manager.

Note: For TDM-connected deployments, the call arrives at a Trunk DN. The routing strategy to select a Voice Treatment Port DN is not required. You can skip this procedure.

Start of procedure

1. Go to Provisioning > Switching > Places.
2. Click New.
3. Provision the new Place as follows:
   - Name—Enter a number that matches the Voice Treatment Port DN that you will map this Place to.
   - DNs—Click Add, select the SIP Server switch, and then select the corresponding Voice Treatment Port DN.
4. Click Save & New. Create a new Place to match every Voice Treatment Port DN mapped to an IVR Port.
5. Click Save and Close.

End of procedure

Next Steps

- Procedure: Configuring a Place Group as the target for IVR Ports

Procedure: Configuring a Place Group as the target for IVR Ports

Purpose: To create the Place Group object that the routing strategy can use to distribute calls to IVR ports for treatment. Stat Server determines the availability of the individual Places.

Start of procedure

1. Go to Provisioning > Switching > Place Groups.
2. Click New.

3. Enter a name for this Place Group.
4. Click `Save` to register the Place Group with the Management Layer.
   Once registered, the `Places` tab appears in the `Place Group` object.
5. On the `Place` tab, click `Add`.
6. Hold down the `Shift` or `Ctrl` key while selecting the Places that you created in Procedure: Configuring a Place object for each IVR Port, on page 178.
7. Click `Save & Close`.

End of procedure

Next Steps

- Procedure: Creating a resource group for the CTI Connector

Procedure: Creating a resource group for the CTI Connector

Purpose: To create the resource group on Resource Manager that is used to define the connection to CTI Connector.

Start of procedure

1. Go to `Provisioning > Voice Platform > Resource Groups`.
2. Click `New` to open the `Resource Group Wizard`.
3. Follow the wizard instructions to create a new `Resource Group` for the CTI Connector:
   a. Click `New`, then `Next`, and select the Resource Manager to which this group will belong.
   b. Enter a name for the group and select `CTI Connector` as the `Group Type`, and click `Next`.
   c. Keep the defaults for `monitoring method` and `load balancing scheme`, and click `Next`.
   d. Select the CTI Connector that you want to add, configuring as follows:
      i. Select `sip` as the `scheme`.
      ii. Select the SIP `Port` used to communicate with CTI Connector (typically `5080`)
      iii. Click `Next`.

Tip: For more information about using this wizard, see the “Creating Resource Groups” procedure in the *Genesys Voice Platform 8.1 Deployment Guide*.

4. At the end of the wizard, click `Finish`.

End of procedure

Next Steps

- For TDM deployments, all calls must go through CTI Connector. Make sure that `CTI Usage` in the gateway resource group for the SIP Server is set to `Always on` (`use-cti=1`).
- For carrier-connected deployments, you can configure the gateway resource group to send applications either to SIP Server or to CTI Connector, depending on the application. To configure this feature, see Procedure: Configuring CTI Flagging for IVR Profiles.

Integrating with IVR Server in In-Front Mode—Procedures

Complete the procedures in this section to integrate the Voice Platform Solution with the IVR Server in an In-Front mode configuration.

Procedure: Creating the virtual switching office

Start of procedure

1. Go to `Provisioning > Switching > Switching Office`.
2. Click `New`.
3. Enter a `Name`.
4. For the `type`, select `Virtual Switch for IVR In-Front`.
5. Click `Save`.

End of procedure

Next Steps

- Procedure: Creating the virtual switch for TServer_IVR_InFront, on page 181

Procedure:
Creating the virtual switch for TServer_IVR_InFront

Purpose: To create the virtual switch that represents GVP when the VPS is integrated with IVR Server in an In-Front mode configuration. The CTI Connector uses this virtual switch to manage the `Voice Treatment Port` DNs used to map an IVR port for each incoming call.

Start of procedure

1. Go to `Provisioning > Switching > Switches`.
2. Click `New`, and provision the switch as follows:
   - `Name`—Enter a name for the virtual switch.
   - `Switching Office`—Click the browse icon, and select the virtual switch that you created in Procedure: Creating the virtual switching office, on page 180.
   - `TServer`—Click the browse icon, and select the `Tserver_IVR`.
3. Click `Save`.

End of procedure

Next Steps

- Procedure: Adding required connections to the TServer_IVR_InFront

Procedure:
Adding required connections to the TServer_IVR_InFront

Start of procedure

1. Go to `Provisioning > Environment > Applications`, and select `TServer_IVR`.
2. On the `Configuration` tab, under `Connections`, add connections to:
   - TServer `Application` object for the premise switch
   - Message Server.
3. Click `Save and Close`.

End of procedure

Next Steps

- Procedure: Adding required connections to the I-Server_InFront, on page 182

Procedure: Adding required connections to the I-Server_InFront

Start of procedure

1. Go to Provisioning > Environment > Applications, and select I-Server.
2. On the Configuration tab, under Connections, add a connection to:
   - TServer_IVR_InFront
3. Click Save.

End of procedure

Next Steps

- Procedure: Creating an IVR object to represent GVP in the solution

Procedure: Creating an IVR object to represent GVP in the solution

Start of procedure

1. Go to Provisioning > Switching > IVRs.
2. Click New, and configure the parameters on the Configuration tab as follows:
   - Name—Enter a name for this IVR.
   - Description—Enter a useful description of this IVR. For example, the switch configuration, or the IVR mode.
   - Type—Select Genesys Voice Platform.
   - Version—Enter the software release number for the GVP that this IVR will represent. For example, 8.1.
3. Click Save.

End of procedure

Next Steps

- After you save the IVR object, the IVR Port tab appears. Use this tab to create the IVR Ports that you map to the corresponding DNs on the virtual switch for GVP.

  Before creating the IVR Ports, configure the CTI Connector for integration with the IVR Server in the In-Front mode. See Procedure: Configuring the CTI Connector for IVR In-Front, on page 183.

Procedure:
Configuring the CTI Connector for IVR In-Front

Purpose: To configure the CTI Connector `Application` object for integration with the IVR Server in the In-Front mode. You will specify the range of IVR Ports that CTI Connector will generate in order to map each call to a specific port and attach the port number in the `NewCall` message to IVR Server.

Prerequisites

- The CTI Connector `Application` object is installed and basic configuration is completed. For an overview of the deployment process, see Table Task Summary: on page 146.

Start of procedure

1. Go to `Provisioning > Environment > Applications > <CTI_Connector>`.
2. On the `Options` tab, select `Mandatory Options` from the `View` drop-down list and configure the following mandatory parameters:
    - `Resource Manager IP Address`—Enter the IP address and SIP port for the Resource Manager in the following format:
      `<RM_ip_address>:<RM_sip_port>`
      For example:
      `10.10.10.10:5060`
    - `IVR Client Name`—Enter the login name of the IVR Server `Application` object, as it appears in Management Framework.
    - `IVR Server Host IP Address`—Enter the IP address of the IVR Server host.
    - `IVR Server Communication Port`—Enter the GLI port number as configured in the `gli-server-address` option of the `TServer_IVR` object.
    - `Local Host Name`—Enter the IP address of the CTI Connector host.

    > **Tip:** The application template provides an `IServer_Sample` section that you can configure as is, rename, or use as a model. Add a new section for every additional IVR Server in your configuration. If you rename this section, or add any new sections, make sure you add the section name to the `Customer IServers List` parameter in the `IVRSC` section.

3. Select `Advanced View (Options)` from the `View` drop-down list and provision the number of IVR ports as follows:
    - `IVRPort Base Index`—Set the starting number for the range of IVR Ports that CTI Connector will generate. Later, you will create matching ports/DNs in the Configuration Layer for IVR port mapping.

- Max IVRPorts—Set the maximum number of IVR Ports that the VPS can allow into the system. Later, you will create this number of matching ports/DNs in the Configuration Layer.

Note: These IVR port options are optional for In Front configurations. If the gateway provides the IVRPort in the incoming INVITE, these options are not required.

4. Click Save.

End of procedure

Next Steps

- Procedure: Creating Voice Treatment Port DNs

Procedure: Creating Voice Treatment Port DNs

Purpose: To create and configure Voice Treatment Port DNs on the virtual switch for GVP. These DNs will be used to map the IVR ports generated by CTI Connector to the ports configured in the IVR object.

Start of procedure

1. Go to Provisioning > Switching > Switches > <virtual_switch_for_GVP>.
2. On the DN tab, click New.
3. On the Configuration tab, configure the following mandatory parameters:
   - Number—Assign the new DN a number, starting with the first number in the range you defined in the IVRPortBaseIndex option in CTI Connector (see Step 3 on page 183).
   - Type—For a regular DN that is always ready to receive calls (no agent login required), select Voice Treatment Port from the drop-down list.
4. To create another DN, click Save & New. Create as many new Voice Treatment Port DNs as GVP allows, provisioning each as you did in Step 3.

   Tip: Use continuous numbering (in increments of one) when naming all subsequent DNs. Create as many DNs as you defined in the MaxIVRPorts option in CTI Connector (Step 3 on page 183).
5. Click Save.

End of procedure

Next Steps

- Procedure: Mapping IVR Ports to their matching DNs

Procedure: Mapping IVR Ports to their matching DNs

Purpose: To create IVR Ports in the `IVR` object and link them to the `Voice Treatment Port` DNs configured on the virtual switch. CTI Connector manages these ports for incoming calls, forwarding the port number to the IVR Server in the `NewCall` request.

Start of procedure

1. Go to `Provisioning > Switching > IVRs`.
2. Click the `IVR` object you created to represent GVP in the solution.
3. On the `IVR Port` tab, click `New`.
4. Provision the new IVR port as follows:
   - `Port Number`—Enter the same number here that you entered for the `Voice Treatment Port` DN that you want to map this port to.
   - `Associated DN`—Click browse, and select the virtual switch for GVP; then, in the `DNs` folder, select the matching `Voice Treatment Port` DN.
5. Click `Save & New`. Create additional IVR Ports until all matching `Voice Treatment Port` DNs are mapped.
6. Click `Save and Close`.

End of procedure

Next Steps

- Procedure: Creating a resource group for the CTI Connector

Procedure: Creating a resource group for the CTI Connector

Purpose: To create the resource group on Resource Manager that is used to define the connection to the CTI Connector.

Start of procedure

1. Go to `Provisioning > Voice Platform > Resource Groups`.
2. Click `New` to open the `Resource Group Wizard`.

3. Follow the wizard instructions to create a new `Resource Group` for the CTI Connector as follows:
   a. Click `New`, then `Next`, and select the Resource Manager to which this group will belong.
   b. Enter a name for the group and select `CTI Connector` as the `Group Type`, and click `Next`.
   c. Keep the defaults for `monitoring method` and `load balancing scheme`, and click `Next`.
   d. Select the CTI Connector that you want to add, configuring as follows:
      i. Select `sip` as the `scheme`.
      ii. Select the SIP `Port` used to communicate with CTI Connector (typically `5080`)
      iii. Click `Next`.

   Tip: For more information about using this wizard, see the “Creating Resource Groups” procedure in the *Genesys Voice Platform 8.1 Deployment Guide*.

4. At the end of the wizard, click `Finish`.

End of procedure

Next Steps

- Check that a `Trunk` DN to contact Resource Manager has been created in the SIP Server switch. If it has not, go to Procedure: Configuring a GVP DN for Standard VoiceXML applications, on page 116.
- If the `Trunk` DN exists in the SIP Server switch, continue at Procedure: Configuring CTI Flagging for IVR Profiles, on page 188.

Integrating with Network IVR Server—Procedures

Complete the procedures in this section to integrate the Voice Platform Solution with the Network IVR Server.

Procedure: Configuring the CTI Connector for Network IVR Server

Prerequisites

- The CTI Connector `Application` object is installed, and basic configuration is completed. For an overview of the deployment process, see Table Task Summary:, "Verify Prerequisites," on page 146.

Start of procedure

1. Go to `Provisioning > Environment > Applications > <CTI_Connector>`.
2. On the `Options` tab, select `Mandatory Options` from the `View` drop-down list and configure the following mandatory parameters:
   - `Resource Manager IP Address`—Enter the IP address and SIP port for the Resource Manager in the following format:
     `<RM_ip_address>:<RM_sip_port>`
     For example:
     `10.10.10.10:5060`
   - `IVR Client Name`—Enter the login name of the IVR Server `Application` object, as it appears in Management Framework.
   - `IVR Server Host IP Address`—Enter the IP address of the IVR Server host.
   - `IVR Server Communication Port`—Enter the GLI port number as configured in the `gli-server-address` option of the `TServer_IVR` object.
   - `Local Host Name`—Enter the IP address of the CTI Connector host.

   Tip: The application template provides an `IServer_Sample` section that you can configure as-is, rename, or use as a model. Add a new section for every additional IVR Server in your configuration. If you rename this section, or add any new sections, make sure you add the section name to the `Customer IServers List` parameter in the `IVRSC` section, separating each section name by a semicolon (;).

3. Select `Advanced View (Options)` from the `View` drop-down list and provision the number that the VPS sends to IVR Server as follows:
   - `Use Called Number`—If customers use a toll-free number to contact the VPS, set this parameter to `TFN`. Otherwise, leave the default value of `DN`. Depending on this value, either the toll-free number or the DNIS used to contact the solution is forwarded to the IVR Server, instead of the port number, which is forwarded for other modes.
4. Click `Save`.

End of procedure

Next Steps

- If you are configuring CTI Connector for use with a toll-free number, verify that the `tollfreenum` parameter is also set in the corresponding IVR Profile.

Configuring CTI Flagging for IVR Profiles

Complete the procedures in this section to configure the rules that Resource Manager uses to determine whether it should forward calls to the CTI Connector.

Depending on your deployment architecture and the applications that it services, you can flag the gateway resource as follows:

- Always Off—For CTI through SIP Server only deployments, where CTI Connector and IVR Server are not included in the architecture. Resource Manager performs IVR Profile mapping based on the DNIS provided in the incoming call.
- Always On—For deployments where CTI through IVR Server is always required. For example, if your deployment only services IVR-centric applications, where CTI through IVR Server is required for midcall CTI functionality. Or in TDM-connected configurations, where the DNIS must be retrieved from the IVR Server in all cases.
- Based on DN Lookup—For deployments with architecture configurations that support both CTI through IVR Server and CTI through SIP Server call flows, and that include both Standard VoiceXML and IVR-centric voice applications. For this option, you must set the IVR Profile for a specific application as either CTI or non-CTI. For non-CTI applications (self-service only), Resource Manager bypasses CTI Connector and instead maps the IVR Profile directly.

Outbound Applications

The `Based on DN Lookup` setting is also required for deployments that include SSG for outbound calls. When SSG initiates an outbound call, it cannot go through CTI Connector. In this case, the IVR Profile for an outbound VoiceXML application must be set as non-CTI. CTI-enabled applications are still available for inbound calls, and must be configured accordingly.

Procedure:
Configuring the gateway resource for CTI through IVR Server

Purpose: To create a gateway resource that Resource Manager uses to communicate with SIP Server, in CTI though IVR Server configurations. This includes flagging the resource group so that Resource Manager can determine whether it should send the call to CTI Connector or, for voice applications that do not require CTI through IVR Server, bypass CTI Connector altogether.

Start of procedure

1. Go to `Provisioning > Voice Platform > Resource Groups` panel.
2. Click `New` to start the `Resource Group` Wizard.
3. When the wizard opens, click `Next`.
4. On the `Resource Manager Selection` page, select the Resource Manager for which you want to configure the resource group, then click `Next`.
5. On the `Group Name and Type` page, give your resource group a name, and select `Gateway` as the type. Then click `next`.
6. On the `Group Properties` page, configure the parameters as follows:
   - `Monitoring Method`—Accept the default of `None`.
   - `Load Balancing Scheme`—Accept the default of `Round Robin`.
   - `CTI Usage`—Configure this option as follows:
     - `Always On`—For CTI through IVR Server-only deployments, set `use-cti` to `1`.

     Note: Do not set `CTI Usage` to `Always On` (`use-cti=1`) for deployments that include SSG for outbound calls. Calls initiated by SSG cannot go through CTI Connector.

     - `Based on DN Lookup`—For CTI through IVR Server deployments that can also include CTI through SIP Server call flows, set `use-cti` to `2`.

       For outbound calls using SSG, set `use-cti` to `2`.
7. On the Resource Assignment page, do the following:
   a. Select your SIP Server from the list.
   b. Select the `Scheme`—either SIP or SIPS (secure SIP).
   c. Select the SIP `Port` number—typically 5060.
   d. Select the `Max Ports` for this SIP Server gateway group.
   e. Click `Next`.
8. On the `Confirmation` page, click `Finish`.

End of procedure

Next Steps

- If you configured the gateway resource to send the call to CTI Connector based on DN lookup (set `CTI Usage` to `Based On DN Lookup` (`use-cti=2`)), you must also flag the IVR Profile so that the lookup knows what to do with the application. See Procedure: Flagging the IVR Profile for CTI.

Procedure: Flagging the IVR Profile for CTI

Purpose: To set the IVR Profile for a specific application as either CTI or non-CTI. In flexible architectures where the gateway resource (SIP Server) can handle both CTI through SIP Server and CTI through IVR Server call flows, this setting enables the Resource Manager to determine where to send the call.

Inbound Calls Enabling this setting is only available in VPS deployments where the SIP Server is able to get the DNIS from the incoming call. In deployments where this is not possible—for example, a TDM-connected configuration—all calls must go through CTI Connector, including calls with no midcall CTI requirement.

Outbound Calls For outbound calls initiated by SSG, you must configure the voice application that connects to the called number so that it does not allow CTI. Calls initiated by SSG cannot pass through CTI Connector.

Prerequisites

- In Resource Manager, the gateway resource group for SIP Server must be configured to allow both CTI and non-CTI calls. Set the `CTI Usage` option in the resource group to `Based on DN Lookup` (`use-cti=2`).

Start of procedure

1. Go to `Provisioning > Voice Platform > IVR Profile`.
2. On the `Options` tab, in the `gvp.policy` section, configure the `CTI Allowed` option as follows:
   - `true`—For IVR-centric applications requiring mid-call CTI functionality, set `CTI Allowed` to `true`.
   - `false`—For voice applications that do not require any mid-call CTI functionality (for example, a Standard VoiceXML application with no CTI extensions), set `CTI Allowed` to `false`.

     For outbound voice applications, set `CTI Allowed` to `false`.
3. Click `Save`.

End of procedure

Next Steps

- This completes the basic procedures for a CTI through IVR Server in Behind mode integration.
- For CTI through IVR Server in the In-Front mode, additional configuration is required to enable CTI transfers from an IVR-centric application to the

SIP Server. See Procedure: Enabling Midcall CTI Routing (IVR-centric Applications).

Enabling Midcall CTI Routing (IVR-centric Applications)

When the VPS is integrated with IVR Server in the In-Front mode, GVP is configured as a virtual switch. In the voice application, transfers between this virtual switch for GVP and the SIP Server switch are treated as inter-switch transfers. Inter-switch transfers require an additional request—AccessNumGet—in order to prepare the call before the routerequest starts the actual transfer.

Procedure: Configuring access from the SIP Server to GVP

Purpose: To configure SIP Server access to the virtual switch for GVP, and to create the access resource that the virtual switch uses to reach the SIP Server switch.

Start of procedure

1. Go to Provisioning > Switching > Switches > <SIP_Server_switch>.
2. On the Configuration tab, under Switch Access Code, click the ADD icon.
3. In the Switch Access Code dialog box, configure the required parameters as follows:
   - Switch—Click the browse icon and select the virtual switch for GVP.
   - Target Type—Select Target ISCC from the drop-down list.
   - Route Type—Select DNIS Pooling from the drop-down list.
   - ISCC Protocol Parameters—Enter dnis-tail=4.

   Tip: In this case, no prefix is required to reach the DN on the virtual switch, so you can leave the Access Code box empty.
4. Click Ok.
5. On the DN tab, click New, and create a a DN of type External Routing Point.

   The virtual switch for GVP will use this DN to reach the SIP Server switch.
6. Click Save and Close.
7. Click Save and Close.

End of procedure

Next Steps

- Procedure: Configuring access from GVP to the SIP Server

Procedure: Configuring access from GVP to the SIP Server

Purpose: To configure the virtual switch for GVP access to the SIP Server switch, and to create the access resource that the SIP Server switch uses to reach the virtual switch.

Start of procedure

1. Go to `Provisioning > Switching > Switches > <virtual_switch_GVP>`.
2. On the `Configuration` tab, under `Switch Access Code`, click the `ADD` icon.
3. In the `Switch Access Code` dialog box, configure the required parameters as follows:
   - `Switch`—Click the browse icon and select your SIP Server `Switch` object.
   - `Access Code`—Enter the prefix required to reach the DN on the SIP Server switch.
   - `Target Type`—Select `Target ISCC` from the drop-down list.
   - `Route Type`—Select `Route` from the drop-down list.
4. Click `Ok`.
5. Create the `Access Resource` DN that the SIP Server switch uses to reach the virtual switch:
   a. Click `New`.
   b. On the `Configuration` tab, configure the required parameters as follows:
      - `Number`—Enter a number for the DN.
      - `Type`—Select `Access Resource` from the drop-down list.
      - `Resource Type`— Enter `dnis`.
   c. Enter a number for the DN.
   d. Click `Save and Close`.
6. Click `Save and Close`.

End of procedure

Next Steps

1. Verify that the SIP Server includes a connection to the `TServer_IVR_InFront` application.

2. When configuring the AccessNumGet block in Composer, configure the required parameters as follows:
   - Destination DN—Enter the number for the External Routing Point DN you created on the SIP Server switch.
   - Location—Select the SIP Server switch from the drop-down list.
3. Configure the <transfer> tag in the VoiceXML application to use the number received in the AccessNumGet block. You can configure this as either a Blind or Bridge transfer (applies to both NGI and GVPi applications).

Chapter

13 Integrating with SSG

This chapter describes the configuration steps required to support outbound call functionality from the Voice Platform Solution 8.1, using the Supplementary Services Gateway (SSG).

This chapter includes the following sections:

Task Summary: SSG Integration

The following table provides an overview of the main steps that you must complete in order to integrate the SSG into the Voice Platform Solution, for outbound call functionality.

Task Summary: Integrating the SSG for Outbound Calls

Objective	Related Procedures and Actions
1. Log in to Genesys Administrator	In your web browser, enter the application URL in the following format: `http://<genesys_administrator_host>/wcm`

Task Summary: Integrating the SSG for Outbound Calls (Continued)

Objective	Related Procedures and Actions
2. Verify prerequisites.	To verify that the baseline solution is working: 1. Go to `Provisioning > Environment > Applications`. 2. Check that the green `Started` bar appears under the `Status` column for the following applications: • SIP Server • Resource Manager • Media Control Platform/Call Control Platform • Supplementary Services Gateway 3. If any of these applications are not shown as started, click the application, and then click the green `Start` arrow. If any application does not start, recheck the configuration.
3. Configure the SIP Server Application.	Go to: `Provisioning > Environment > Applications > <your_SIP_Server_application> > TServer` section 1. Configure the following mandatory options: • `sip-invite-treatment-timeout`—Set this option to `60`. This sets the timeout for the treatment that SSG connects the customer to. • `msml-support`—Set to true. • `divert-on-ringing`—Set to false. • `userdata-map-trans-prefix`—To enable the passing of user data back to the application, set to `X-Genesys-`. 2. Configure the default behavior for how SIP Server responds on receiving CPD result from the gateway or MCP: • `am-detected`—For answering machine detection, set this option to `connect`. • `fax-detected`—For fax machine detection, set this option to `connect`. • `cpd-info-timeout`—Set this option to the length of time, in seconds, that SIP Server will wait for the CPD results. Set this to `7` or higher for CPD on MCP, leave the default of `3` for CPD on media gateway. For a detailed procedure, see Procedure: Configuring SIP Server for outbound calls through SSG, on page 205.

Task Summary: Integrating the SSG for Outbound Calls (Continued)

Objective	Related Procedures and Actions
4. Create a Trunk Group DN to represent GVP.	Go to: Provisioning > Switching > Switches To configure GVP as a DN in the SIP Server switch for making outbound calls: 1. Create a Trunk Group DN with the name Environment. 2. On the Options tab, in the TServer section, configure the following mandatory parameters: • contact • request-uri • subscription-id • make-call-rfc3725-flow • refer-enabled • ring-tone-on-make-call • userdata-map-filter 3. To enable CPD on the MCP, on the Options tab, configure the following optional parameter: • cpd-capability—Set this to mediaserver. For a detailed procedure, see Procedure: Creating a Trunk Group DN for outbound calls, on page 207.
5. Configure the SSG Application object.	Point the SSG to the Trunk Group DN on the SIP Server instance from which outbound calls will be placed: 1. Add a connection to SIP Server. 2. Configure the following mandatory options: • TenantName • TGDN **Key Actions and Rules** • Create a Tenant<n> section for every tenant that this SSG will service. • The variable <n> ranges from 1 to 200. For a detailed procedure, see Procedure: Integrating SSG with SIP Server, on page 208.

Task Summary: Integrating the SSG for Outbound Calls (Continued)

Objective	Related Procedures and Actions
6. Configure the gateway `Trunk` DN.	Go to: `Provisioning > Switching > Switches` **Basic Trunk Configuration** 1. `Number`—Enter a text-based name (word or letter) for this trunk. 2. `Type`—Select `Trunk` from the drop-down menu. 3. `Contact`—Enter the IP address and port of the media gateway. 4. `event-ringing-on-100trying`—Set to true. 5. `prefix`—(for multiple Trunks) Set this to the initial 3 or 4 digits of the TelNum provided in the HTTP request. **Note:** Prefix is just one parameter used by SIP Server to select among available trunks. For more information about working with multiple devices, see the *Framework 8.1 SIP Server Deployment Guide*. **Enabling CPD** To make predictive calls using a supported gateway, additional configuration includes: • `cpd-capability`—Set this option to one of the following supported media gateway types: `audiocodes` or `paraxip`. • `cpd-info-timeout`—Set this option to the length of time, in seconds, that SIP Server will wait for the `INFO` with CPD results, after the `200 OK` from the media gateway. **Note**: VPS supports CPD on Paraxip and Audiocodes media gateways only. Paraxip has been tested with CCXML applications only. For a detailed procedure, see Procedure: Configuring a Media Gateway Trunk for CPD, on page 209

Task Summary: Integrating the SSG for Outbound Calls (Continued)

Objective	Related Procedures and Actions
7. Configure trigger application	**Mandatory Parameters** The `HTTP POST` sent by the trigger application must be designed to include the following mandatory parameters: • Request URI must contain the `TenantName` as a query string parameter. In a multi-tenant setup, use `Environment` or an equivalent identifier. In a single-tenant setup, substitute this value with `Resources` or an equivalent name. • `IVRProfileName`—Specifies the name of the IVR Profile of the voice application to be used for the outbound call. • `Telnum`—Specifies the telephone number used to make the outbound call. • `NotificationURL`—An encoded URL that the SSG will use to send asynchronous notifications to the TA to indicate the success or failure of the outbound call. For a description of all available `CreateRequest` attributes that can be included in the `HTTP POST` request, see the HTTP Interface section of the *Genesys Voice Platform 8.1 User's Guide*. **CPD Control Parameters** For Call Progress Detection, the trigger application must include the following parameters in the `CreateRequest` that it sends to SSG: • `preconnect`—Specifies when to start CPD. If set to true, SSG includes the `cpd-on-connect` Extension in the `TMakePredictiveCall` request. • `detect`—Specifies the action that SSG will take on the outbound call when CPD is detected. SSG translates these parameters into `Extensions` attributes in the T-Library MakeCall request that it sends to SIP Server For a description of all available custom parameters supported by the VPS, see the Call Progress Detection section of the *Genesys Voice Platform 8.1 User's Guide*.

Task Summary: Integrating the SSG for Outbound Calls (Continued)

Objective	Related Procedures and Actions
Additional Special Configuration	
Configure the voice application.	**NGI Applications** For NGI applications, the application can transfer the call to an agent using the `<transfer>` tag only. You can create the VoiceXML application using Composer. **Legacy GVPi Applications** For GVPi applications, you must use the Routing Point DN call flow. Additional configuration is required. See the following procedures: • Task Summary: Additional Configuration for Routing Point Call Flow, on page 201
Integrating SSG into CTI-C deployments.	For CTI through IVR Server deployments, you must use the `Routing Point` DN call flow. Additional configuration is required. See the following procedures: • Task Summary: Additional Configuration for Routing Point Call Flow, on page 201.
Configure SSG for SNMP monitoring.	*Optional: This procedure is only required if you are capturing alarm and trap information for SSG.* 1. Go to the `Configuration` tab of the SSG Application object. 2. Under `Connections`, click `Add` and fill the parameters that point to the SNMP Master Agent installed on this host. **Note:** If no SNMP Master Agent is available, you might need to configure SNMP monitoring for the baseline solution. See “Configuring SNMP Monitoring” on page 122.

Task Summary: SSG Integration, Routing Point Call Flow

For CTI through IVR Server deployments, or when using legacy GVPi applications, you must configure VPS to use the `Routing Point` DN call flow for SSG initiated outbound calls.

Warning! Legacy GVPi applications that use attached data must be configured as "IVR-centric," i.e. they apps require CTIC to exchange attached data with Framework.

Task Summary: Additional Configuration for Routing Point Call Flow, on page 201 provides an overview of the additional steps you must complete to use this Routing Point call.

Task Summary: Additional Configuration for Routing Point Call Flow

Objective	Related Procedures and Actions
1. Verify baseline configuration.	Complete the basic configuration steps for SSG integration. • Verify that Steps 1 to Step 7 are completed in the following task table: ◦ Task Summary: Integrating the SSG for Outbound Calls, on page 195 **Note**: The `Trunk Group` DN will not be used to place the call, but is still required for getting available/total port details for the tenant.

Task Summary: Additional Configuration for Routing Point Call Flow (Continued)

<table>
<tr><th>Objective</th><th>Related Procedures and Actions</th></tr>
<tr><td>2. Configure the <code>Voice over IP Service</code> DN for CPD.</td><td>Go to: <code>Provisioning</code> > <code>Switching</code> > <code>Switches</code> > and double-click your SIP Server <code>Switch</code> object

SIP Server uses this DN to contact RM to initiate MSML dialog for CPD on the Genesys Media Server:

1. On the <code>DNs</code> tab, click <code>Add</code> and create a new DN of the type <code>Voice over IP Service</code>.
2. On the <code>Options</code> tab, in the <code>TServer</code> section, configure the following mandatory parameters:
• <code>contact</code>—Set this to the IP address and SIP port for Resource Manager:
<code>sip.<RM_IP>:<RM_port></code>
• <code>make-call-rfc3725-flow</code>—Set to <code>1</code>.
• <code>refer-enabled</code>—Set to <code>false</code>.
• <code>request-uri</code>—Point this to RM, with <code>msml</code> in the user part, and identify the tenant-id as the name of the tenant:
<code>sip:msml@<RMHost>:<RMport>;gvp-tenant-id=<Tenant_Name></code>
• <code>ring-tone-on-make-call</code>—Set to <code>false</code>.
• <code>service-type</code>—Set to <code>msml</code>.
• <code>subscription-id</code>—Set this to the name of the tenant.
• <code>partition-id</code>—Set this to the name of the tenant.
3. To enable CPD on the MCP, on the <code>Options</code> tab, configure the following optional parameter:
• <code>cpd-capability</code>—Set this to <code>mediaserver</code>.</td></tr>
<tr><td>3. Configure the <code>Voice over IP Service</code> DN for playing treatments.</td><td>Go to: <code>Provisioning</code> > <code>Switching</code> > <code>Switches</code>

SIP Server uses this DN to contact RM to initiate MSML dialog for playing treatments on the Genesys Media Server:

1. On the <code>DNs</code> tab of the <code>Switch</code> object, click <code>Add</code> and create a new DN of the type <code>Voice over IP Service</code>.
2. On the <code>Options</code> tab, in the <code>TServer</code> section, configure the following mandatory parameters:
• <code>contact</code>—Set this to the IP address and SIP port for Resource Manager:
<code>sip.<RM_IP>:<RM_port></code>
• <code>service-type</code>—Set to <code>treatment</code>.</td></tr>
</table>

Task Summary: Additional Configuration for Routing Point Call Flow (Continued)

Objective	Related Procedures and Actions

4. Configure GVP as a series of `Voice Treatment Port` DNs.	Go to: `Provisioning > Switching > Switches`
	VPS uses this set of DNs to select the IVR Profile in order to play the VoiceXML application for the connected customer.
	1. On the `DNs` tab of the `Switch` object, click `Add` and create a new DN of the type `Voice Treatment Port`.
	2. On the `Options` tab, in the `TServer` section, configure the following mandatory parameters:
	• `contact`—Set this to the IP address and SIP port for Resource Manager:
	`sip.<RM_IP>:<RM_port>`
	• `prefix`—Set to the name of the DN.
	• `request-uri`—Use the following format:
	`sip:<vtp_DN_name>@<RM_contact>;gvp-tenant-id=<tenant_name>`
	• `event-ringing-on-100trying`—Set to `true` when CTI Connector is used for the call.
	• `cpd-capability`—Set to mediaserver.
	• `userdata-map-filter`—Set to the required GVP headers, using a constant string. Mandatory values are as follows: `gsw-ivr-profile-name, gsw-session-dbid`
	You can also include the following additional parameters to this option:
	`OutboundData`—Include to pass user data from SSG to IVR Application.
	`AnswerClass`—Include to pass CPD result back to VoiceXML application.
	`GVP-IVRPort`—**Must** include if SSG is integrated with PSTN Connector and CTI Connector.
	`GVP-PSTNC-DBID`—**Must** include if SSG is integrated with PSTN Connector and CTI Connector.

Task Summary: Additional Configuration for Routing Point Call Flow (Continued)

Objective	Related Procedures and Actions
5. Configure the Place Group used by the routing strategy.	1. Configure a Place object for each Voice Treatment Port DN. a. Go to Provisioning > Switching > Places. b. Click New and create a new Place with the same name as the corresponding Voice Treatment Port DN. c. Under DNs, click Add and browse to select the corresponding Voice Treatment Port DN. d. Repeat for as many Voice Treatment Ports 2. Configure a Place Group as the target for the Voice Treatment Port DNs. a. Go to Provisioning > Switching > Place Groups. b. Click New, and give the Place Group a name. c. On the Places tab, click Add and browse to select all the Places you created in Step 1.
6. Configure the Routing Point DN.	Go to: Provisioning > Switching > Switches The solution uses this DN to place the outbound call request. 1. On the DNs tab of the switch object, click Add. 2. Create a new DN of the type Routing Point. **Key Rules** • You will use the name of this DN RPDN parameter in the SSG application. • If serving multiple tenants, create a separate Routing Point DN for each tenant. In the TServer section, set the partition-id to the name of the tenant.
7. Configure the SSG Application object.	Go to: Provisioning > Environment > Applications > your SSG application Point the SSG to the Routing Point DN through which the outbound call will be placed. • On the Options tab, in the Tenant<n> section, configure the following: • RPDN—Set to the Routing Point DN you configured in Step 6. **Note**: The other mandatory parameters should already be configured as described in Step 5 of the baseline integration.

Task Summary: Additional Configuration for Routing Point Call Flow (Continued)

Objective	Related Procedures and Actions
8. Create a routing strategy	Use Interaction Routing Designer to: 1. Create the routing strategy that URS uses to select an available `Voice Treatment Port` DN. 2. Load the strategy on the `Routing Point` DN that you created in Step 6. For an example, see "Sample Routing Strategy" on page 210.

Integrating SIP Server with SSG

To integrate SSG with SIP Server, complete the following procedures.

1. Procedure: Configuring SIP Server for outbound calls through SSG
2. Procedure: Integrating SSG with SIP Server
3. Procedure: Creating a Trunk Group DN for outbound calls
4. **Additional configuration for Routing Point call flow**: Task Summary: SSG Integration, Routing Point Call Flow, on page 201

Procedure:
Configuring SIP Server for outbound calls through SSG

Purpose: To configure the SIP Server `Application` object to support outbound calls initiated through the Supplementary Services Gateway.

Start of procedure

1. Go to `Provisioning > Environment > Applications`, and double-click on your SIP Server `Application` object.
2. On the `Options` tab, in the `TServer` section, configure the options as described in Table 9 on page 206.

Table 9: SIP Server Options—TServer Section

Option	Value	Description
sip-invite-treatment-timeout	30	Set this option to `30` and SIP Server will wait this length of time for a response to the `INVITE` it sends to start a treatment, before the call times out. If the call times out, SIP Server disconnects the call leg, sends an error response back to SSG, and retries the call.
am-detected	`connect`	Set this option to `connect`. This sets the default to be used by SIP Server, which gives the trigger application control over whether SSG will drop or connect the call, depending on the CPD result.
fax-detected	`connect`	Set this option to `connect`. This sets the default to be used by SIP Server, which gives the trigger application control over whether SSG will drop or connect the call, depending on the CPD result.
cpd-info-timeout	3, 7 or higher	**CPD on MCP** Set this option to `7` (seconds) or higher if CPD is to be performed on the MCP. The MCP requires up to 6 seconds to detect CPD. **CPD on Media Gateway** Set this option to the default of `3` (seconds) if CPD is to be performed on a supported media gateway (Paraxip or Audiocodes). **Note:** Paraxip has been tested with CCXML applications only.
userdata-map-trans-prefix	`X-Genesys-`	SIP Server maps any header with this `X-Genesys-` prefix to corresponding T-Library messages.

3. Click `Save and close`.

End of procedure

Next Steps

- Procedure: Creating a Trunk Group DN for outbound calls

Procedure:
Creating a Trunk Group DN for outbound calls

Purpose: To create the Trunk Group DN from which SIP Server sends the outbound INVITE to the customer endpoint.

Start of procedure

1. Go to Provisioning > Switching > Switches, and double-click your SIP Server Switch object.
2. Create a Trunk Group DN:
   a. On the DNs tab, click Add.
      A New DN window opens.
   b. On the Configuration tab, enter the following information:
      - Number—Name the DN according to the tenant from which the outbound call will be made.
      - Type—Select Trunk Group from the drop-down list.

Mandatory Options

3. On the Options tab, click New and create a new section called TServer, then add new options as follows:
   - Contact—Set this option to the Resource Manager IP address and sip port (typically 5060), using the following format:
     sip:<RM_ip_address>:<RM_sip_port>
   - make-call-rfc3725-flow—Set this option to 1. This instructs SIP Server to use the 3pcc call flow as defined in the RFC 3725
   - refer-enabled—Set this option to false. It forces SIP Server to use the re-INVITE method instead of REFER, as required for 3pcc calls.
   - ring-tone-on-make-call—Set this option to false (no ring tone is required for scenarios that may include CPD)
   - request-uri—Set the user part of the URI to msml, and identify the tenant-id as the name of the tenant. Format the value of this option as follows: sip:msml@<RMHost>:<RMport>;gvp-tenant-id=<Tenant_Name>

     SIP Server sends an INVITE to Resource Manager with the Request-uri modified to tell GVP to act as media server for the call.
   - subscription-id—Set this to the name of this Trunk Group DN. In this case, set the value for this option to Environment.
   - userdata-map-filter—Set this to the following string:
     gsw-ivr-profile-name,gsw-session-dbid,CustomerParam

     This ensures that these required parameters are provided as UserData parameters to Resource Manager. It also allows any custom data provided by the Trigger Application to be passed through the CustomParam attribute in the TMakePredictiveCall.

Enable CPD

4. On the Options tab, in the TServer section, add the following option:
   - cpd-capability—Set this option to mediaserver. This enables CPD analysis to be performed by the MCP.

     Tip: If CPD is also configured for the media gateway, then SIP Server selects the gateway for CPD, over MCP.

5. Click Save and Close to save all changes.

End of procedure

Next Steps

- Procedure: Integrating SSG with SIP Server

Procedure: Integrating SSG with SIP Server

Purpose: To point the SSG Application object to the SIP Server instance and related Trunk Group DN from which outbound calls will be made.

Prerequisites

- A Trunk Group DN for outbound call flow. See Procedure: Creating a Trunk Group DN for outbound calls, on page 207.

Start of procedure

1. Go to Provisioning > Environment > Applications, and double-click your SSG Application object.
2. On the Configuration tab, under Connections, add a connection to the SIP Server Application object.
3. On the Options tab, in the Tenant<n> section, configure the following mandatory options:
   - TenantName—Set to the name of the tenant this section belongs to.
   - TGDN—Set this to the name of the prerequisite Trunk Group DN.

   Note: Create a Tenant<n> section for every tenant that this SSG will service. The variable <n> can range from 1 to the maximum number of tenants, 200.

4. Click Save and close.

End of procedure

Next Steps

- Procedure: Configuring a Media Gateway Trunk for CPD

Enabling CPD on the Media Gateway

CPD can be performed on either the MCP or using one of two supported media gateways: Paraxip or Audiocodes. If CPD is enabled for both MCP and the media gateway, SIP Server chooses the gateway to perform the analysis.

Note: Paraxip has been tested with CCXML applications only.

Procedure: Configuring a Media Gateway Trunk for CPD

Purpose: To configure a Trunk DN that the VPS will use to perform call progress analysis for outbound calls using SSG.

Note: This procedure describes the minimum configuration required for a Trunk DN in order to support CPD on a media gateway. For a more complete description of the options available when configuring a gateway, see the procedure "Configuring a gateway" in the *SIP Server 8.1 Deployment Guide*.

Start of procedure

1. Go to Provisioning > Switching > Switches, and double-click the SIP Server Switch.
2. On the DNs tab, click Add.
3. On the Configuration tab, enter parameters as follows:
   - Number—Enter a name for this DN.

     Tip: Genesys recommends entering a text-based name for Trunk DNs (letters or words: for example, My_Trunk), to differentiate from regular DNs.

   - Type—Select Trunk from the drop-down list.
4. On the Options tab, create a TServer section, and add new options as follows:
   - contact—Set this to the IP address and port of the media gateway.
   - cpd-capability—Set this to the exact string that identifies the type of supported gateway you are using: audiocodes or paraxip.

- `cpd-info-timeout`—Set this to the length of time, in seconds, that SIP Server will wait for the `INFO` with CPD results, after the `200 OK` from the media gateway.

  Tip: This DN-level option takes precedence over the application-level configuration. If `call_timeguard_timeout` is included in the `TMakePredictiveCall` request, then it takes precedence over either option setting.

5. Click `Save and close`.

End of procedure

Sample Routing Strategy

For deployments that use the `Routing Point` DN call flow (see "Outbound Calls Using the Supplementary Services Gateway" on page 58), the following sample routing strategy provides some basic details about the required routing blocks.

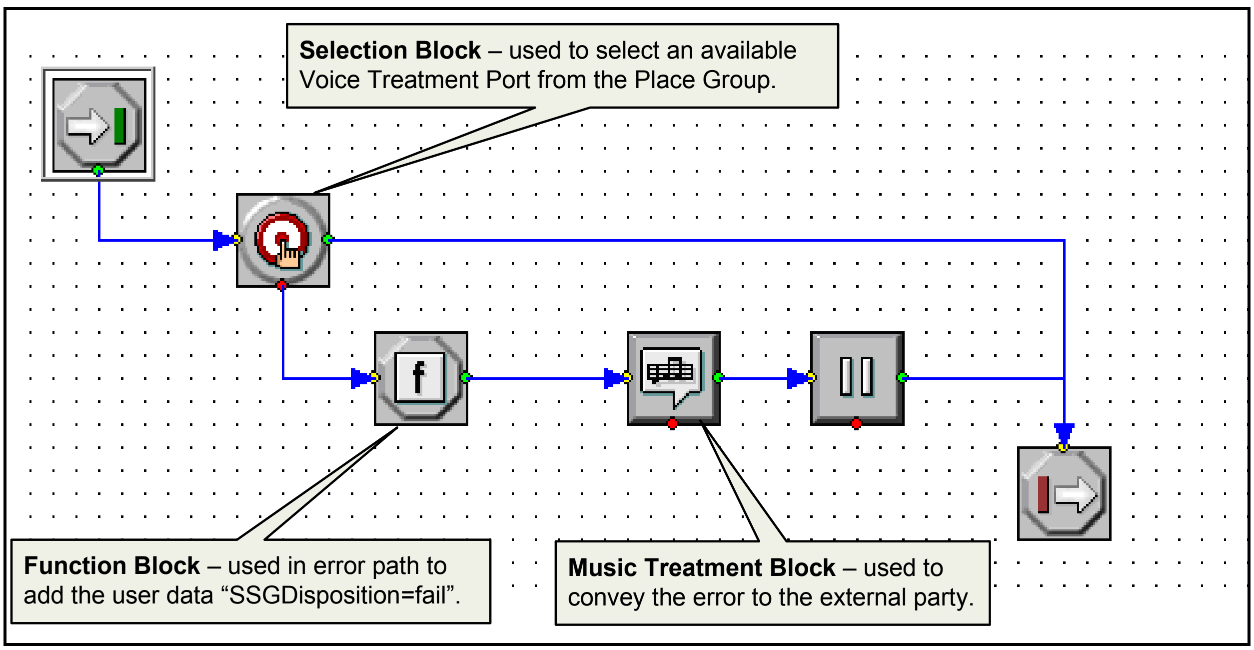


Figure 50: Routing Point Call Flow—Sample Routing Strategy

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Chapter

14 Integrating with PSTN Connector

This chapter describes the configuration steps required to integrate the PSTN Connector into the solution.

Note: It is beyond the scope of this document to provide detailed configuration steps for all scenarios that the PSTN Connector supports. The procedures in this chapter can be used to integrate the PSTN Connector into a station-side connected architecture, as an example.

For general deployment procedures, see the *Genesys Voice Platform 8.1 Deployment Guide*.

This chapter includes the following sections:

Task Summary: Integrating With PSTN Connector

The Task Summary: Integrating With PSTN Connector, on page 212 provides an overview of the main steps required to integrate the PSTN Connector into the solution.

Note: For an overview of how PSTN Connector fits into a TDM-connected, IVR Behind solution, see Chapter 8, “PSTN Connector,” on page 101.

Task Summary: Integrating With PSTN Connector

<table>
<tr><th>Objective</th><th>Related Procedures and Actions</th></tr>
<tr><td>1. Verify baseline configuration.</td><td>Before integrating the PSTN Connector, the rest of the solution must be properly integrated.

In keeping with the “Sample Deployment” on page 101 deployment described in this guide, verify that the following procedures have been completed:

• Task Summary: IVR Behind, TDM-Connected Integration, on page 151

Key Actions

Some small changes to the baseline configuration are required for integration with PSTN Connector:

• GVP <code>Trunk</code> DN—set the <code>prefix</code> option to match the first few digits of the Default DNIS you will configure later in Step 4.
• Gateway Resource Group—Set <code>CTI Usage</code> to <code>Always On</code> (<code>use-cti</code> set to 1).</td></tr>
<tr><td>2. Install Dialogic card.</td><td>For information about installing and configuring Dialogic hardware and software, visit the vendor’s website.

Required Software Version: Dialogic® SR 6.0 PCI for Windows SU 241 (2003/2008)

Special Configuration for Windows Server 2008

After installation on Windows Server 2008, you must disable the Physical Address Extension (PAE) on the server.

1. From the command line interface (CLI), enter:
• <code>C:\bcdedit /set nx OptOut</code>
• <code>C:\bcdedit /set pae ForceDisable</code>
2. Restart the server.

For information about supported Dialogic cards, see the Genesys Voice Platform 8.1 Deployment Guide.</td></tr>
</table>

Task Summary: Integrating With PSTN Connector (Continued)

<table>
<tr><th>Objective</th><th>Related Procedures and Actions</th></tr>
<tr><td>3. Deploy PSTN Connector.</td><td>
1. Install and configure the PSTN Connector application and object, with the following connections:
• Message Server (optional)
• SIP Server
• SNMP Master Agent
2. Configure DialogicManager_Route1 section:
• <code>Channels</code>—Set this to the port numbers for this route, using the format <code>Card:PortRange</code>. For example: <code>1:1-23</code>
• <code>Routetype</code>—Set this the call direction for this route: <code>Inbound</code>, <code>Outbound</code>, or <code>In/Out</code>
• <code>Signaling type</code>—Set this to the signaling type for this route:
  • <code>T1-ISDN (PRI),</code>
  • <code>0;Analog,</code>
  • <code>1;E1-ISDN (PRI),</code>
  • <code>2;T1-RobbedBit,</code>
  • <code>3;E1-CAS,4</code>
3. Configure GatewayManager section:
• <code>SIP Destination IP Address</code>—Set this to the SIP Server IP address.
• <code>SIP Destination Port Number</code>—Set this to the SIP Server port number.
4. Configure MediaManager section:
• <code>Supported Local Codec Type</code>—Set to <code>Alaw</code> or <code>MuLaw</code>

Sample screenshot: See Figure 51 for an example of a configured application.

For more information: For detailed procedures, go to theGenesys Voice Platform 8.1 Deployment Guide for information about the following:
• Using the deployment wizard to install GVP
• Creating a connection to a server
• Configuring the PSTN Connector
</td></tr>
</table>

Task Summary: Integrating With PSTN Connector (Continued)

<table>
<tr><th>Objective</th><th>Related Procedures and Actions</th></tr>
<tr><td>4. Configure default DNIS.</td><td>Go to <code>Provisioning</code> > <code>Environment</code> > <code>Applications</code> > and select your PSTN Connector application.
• On the <code>Options</code> tab, in the <code>DialogicManager</code> section, configure the following option:
◆ <code>DefaultDNIS</code>—Set this to any number to be used as the default DNIS (for example, 74388485). You must set the prefix in the GVP Trunk to the first few digits of this number (see Step 1).
Note: This is only required for behind the switch deployments, where the DNIS is not immediately available from the switch.</td></tr>
<tr><td>5. Configure the PSTN Connector Trunk.</td><td>Go to <code>Provisioning</code> > <code>Switching</code> > <code>Switches</code> > and select your SIP Server switch object.
1. On the <code>DNs</code> tab, click <code>New</code> and create a DN of the type <code>Trunk</code> .
2. On the <code>Options</code> tab of the DN, create a <code>TServer</code> section with the following options:
◆ <code>contact</code>—Enter the IP address and port for the PSTN Connector.
◆ <code>sip-replaces-mode</code>—Set this to <code>1</code>. For consultation transfers, SIP Server forwards the <code>REFER</code> with <code>Replaces</code> to the external destination.
◆ <code>oosp-transfer-enabled</code>—Set this to <code>true</code>.
For deployments with multiple PSTN Connectors
◆ <code>prefix</code>—Used internally by GVP/SIP Server in bridge transfer scenarios. This setting ensures that the outbound leg of the transfer is routed to the same PSTN Connector where the inbound call came from.
◆ <code>replace-prefix</code>—Create this option, but leave the value as an empty string. SIP Server removes the prefix added by Resource manager before forwarding the call to the PSTN Connector instance.</td></tr>
</table>

Sample PSTN Connector Configuration

Figure 51 on page 215 shows an example of how the mandatory options for PSTN Connector might be configured.

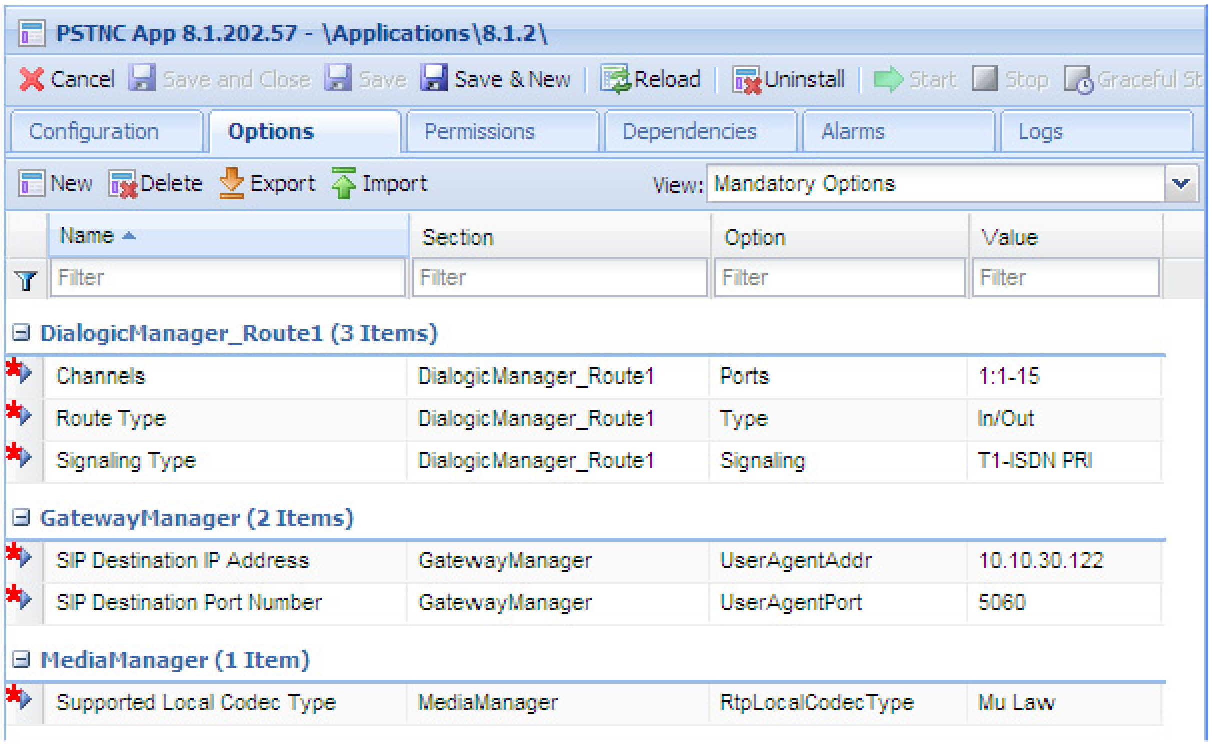

Figure 51: Sample PSTN Connector Configuration

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Part

3 Appendixes

Use the following appendixes for additional information related to the solutions:

- Appendix A, “Sample User Data Mapping,” on page 219
- Appendix B, “Configuration Options,” on page 225

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-CustomerPassword: 1234
X-Genesys-CustomerZipcode: 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'
         'CustomerPassword'  '1234'
         'CustomerZipCode'   '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 = new Object();
     userdata.CustomerName = "John Doe";
     userdata.CustomerPassword = "1234";
     userdata.CustomerZipCode = "90210";
     </script>
     <transfer bridge="true"
dest="sip:5555@" gvp: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'
      'CustomerPassword'     '1234'
      'CustomerZipCode'      '90210'
      'CallUUID'     'DHU3UHQ5TT5DFBQ70P3BP9P7NC00000B'
      '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'DHU3UHQ5TT5DFBQ70P3BP9P7NC00000G'
   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-99E50A51A3
23-10
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-CustomerPassword: 1234
X-Genesys-CustomerZipcode: 90210
X-Genesys-CallUUID: DHU3UHQ5TT5DFBQ7OP3BP9P7NC00000G
Session-Expires: 1800;refresher=uac
Min-SE: 90
```

Sample VoiceXML Session Variables

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

```
session.com.genesyslab.userdata.customername = 'John Doe';
session.com.genesyslab.userdata.customepassword ='1234'
session.com.genesyslab.userdata.customerzipcode ='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&CustomerPassword=1234&__reason=disconnect
```

Sample T-Library Event

```
AttributeANI       '9059683348'
   AttributeDNIS   '8000'
   AttributeUserData[65] 00 03 00 00..
      'CustomerName'     'Jane Doe'
      'CustomerPassword'    '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="CustomerPassword" expr="1234"/>
   <exit namelist="CustomerName CustomerPassword"/>
</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="CustomerPassword" expr="1234"/>
   <vg:send namelist="CustomerName CustomerPassword"/>
</form>
...
```

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

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Appendix

B Configuration Options

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

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 8.1 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 “at” symbol (`@`) is required to delimit the user part from the host part of the address.

referxferhold

Default Value: `0`
Valid Values: `0, 1`
Display Name in Genesys Administrator: `Refer Transfer Hold`
Changes Take Effect: At start/restart.

Specifies whether to place the original caller on hold (INVITE hold) before sending the `REFER` message to transfer the call.

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:5070`
- `transport.1=transport1 tcp:any:5070`

- `transport.2=transport2 tls:any:5071 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.

Logical Resource Group Options

You can modify these resource group options using the Genesys Administrator Resource Group Wizard: `Provisioning > Voice Platform > Resource Groups`.

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, msml`
Changes Take Effect: After restart

Specifies the types of service provided by resources in this 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.
- `msml`—Media streaming services using the Media Server Markup Language (MSML).

CTI Connector Options

ClientName

Default Value: None
Valid Values: Any valid string
Display Name in Genesys Administrator: IVR ClientName
Changes Take Effect: After restart

Specifies the IVR Group Name that is configured in Genesys Administrator.

DefaultDNIS

Default Value: None
Valid Values: Any DNIS
Display Name in Genesys Administrator: Default DNIS
Changes Take Effect: After restart

Specifies the default DNIS that CTI-Connector will send to Resource Manager, in case the IVR Server in Behind mode fails to return the DNIS during the fetch DNIS operation. This mandatory option must be set for all IVR Server modes, though it is only effective for Behind mode.

fetchscriptidfromurs

Default Value: `0`
Valid Values: `0, 1`
Display Name in Genesys Administrator: `Fetch Script ID from URS`
Changes Take Effect: After restart

Specifies whether CTI Connector needs to fetch the user-defined key value to represent the DNIS from the Management Layer. This option is used in situations where the switch cannot provide the DNIS through the CTI link. CTI Connector sends the key to URS, where the predefined value is returned—typically the DNIS that is required for IVR Profile mapping.

To set this option, you must perform the following additional configuration:

- Define a key-value pair in URS.
- Define the script ID key name (`scriptidkeyname`) to be sent to URS.

GetDNISFromIServer

Default Value: `False`
Valid Values: `True, False`
Display Name in Genesys Administrator: `Fetch DNIS from IServer`
Changes Take Effect: After restart

Specifies whether CTI Connector needs to get the DNIS value that is required to map an IVR Profile from the IVR Server. Set this option to `true` only for integration with IVR Server in Behind mode—required for TDM integrations, optional for carrier-connected (if CTI flagging is disabled). For other IVR Server mode integrations, leave this option set to the default `false`, so that CTI Connector obtains the DNIS from the `history-info` or `to` header of the `INVITE` message (the default behavior).

iserveraddr

Default Value: None
Valid Values: <IServer_ip_address>
Display Name in Genesys Administrator: `IServer Host IP Address`
Changes Take Effect: After restart

Specifies the IP address of the host machine where the IVR Server is running.

iserversocket

Default Value: No default (9090 is recommended)
Valid Values: <IServer_port>
Display Name in Genesys Administrator: IServer Communication Port
Changes Take Effect: After restart

Specifies the gli_server_address port number as configured in the IVR Server Application object. Typically the port used is 9090.

IVRPortBaseIndex

Default Value: -1
Valid Values: Any integer
Display Name in Genesys Administrator: IVRPort Base Index
Changes Take Effect: After restart

Specifies the starting IVR port number. Each port number increments by one after this is set. If this parameter is set to -1, CTI Connector will not generate an IVR Port, instead it will take the port base on DNIndicator.

Note: Use this option when the port information is unavailable. IVR Server uses this port number to pass the DNIS information. For deployments using multiple CTI Connectors, the port range must be distinct.

This parameter is applicable when IVR Server is deployed in-front-of-switch mode.

localhostname

Default Value: None
Valid Values: <CTIC_host_ip_address>
Display Name in Genesys Administrator: Local Hostname
Changes Take Effect: After restart

Specifies the IP address of the CTI Connector host machine.

MaxIVRPorts

Default Value: 2000
Valid Values: Any integer
Display Name in Genesys Administrator: Max IVR Ports
Changes Take Effect: After restart

Specifies the upper limit to the number of IVR ports that CTI Connector will generate for IVR In-Front mode integrations.

playannouncepath

Default Value: file://../treatments/PlayAnn.vxml
Valid Values: <path_to_playannaounce_vxml_file>
Display Name in Genesys Administrator: PlayAnnounce Resource Path
Changes Take Effect: After restart

Specifies the path to the default resource that will play the announcement treatments triggered from the IVR Server application.

playannounceanddigitpath

Default Value: file://../treatments/PlayAnnDigits.vxml
Valid Values: <path_to_playannaounceanddigit_vxml_file>
Display Name in Genesys Administrator: PlayAnnounceAndDigit Resource Path
Changes Take Effect: After restart

Specifies the path to the default resource that will play the announcement and collect digit treatments triggered from the IVR Server application.

RMIPAddr

Default Value: None
Valid Values: <RM_ip_address>:<RM_sip_port>
Display Name in Genesys Administrator: Resource Manager IP Address
Changes Take Effect: After restart

Specifies the IP address and port information of the Resource Manager. The value should take the following form: <RM_ip_address>:<RM_sip_port>

Example:

```
10.10.10.10:5060
```

scriptidkeyname

Default Value: None
Valid Values: <script_id_key_name>
Display Name in Genesys Administrator: Script Id Key Name
Changes Take Effect: After restart

Specifies the key name, configured in Framework side that will be used by the IVR Server Client in UdataGet messages. This option is applicable only when the IVR Server is running in Behind mode.

UseCalledNumAs

Default Value: DN
Valid Values: DN, TFN
Display Name in Genesys Administrator: Use Called Number
Changes Take Effect: After restart

Specifies whether CTI Connector will send the toll free number (TFN) or the DNIS. To send the TFN, set the value to TFN. To send the DNIS, set the value to the default DN.

Supplementary Services Gateway Options

These options should be created in a section called `Tenant<n>`, where each section corresponds to a particular tenant services by this instance of SSG. For example, the Environment tenant would appear in SSG as section `Tenant1`, with `TenantName` set to `Environment`.

TenantName

Default Value: No default value
Valid Values: Name of tenant
Changes Take Effect: On restart

Specifies the tenant to which the options in this section apply. For example, if this section is to control the Environment tenant, set this to the name of the tenant: `Environment`.

RPDN

Default Value: No default value
Valid Values: Name of valid Routing Point DN
Changes Take Effect: On restart

Specifies the `Routing Point` DN that the solution uses when placing the outbound call through SSG, in cases where the Routing Point call flow is required. For example, use this option for CTI through IVR server deployments, or when connecting the customer to GVPi applications.

For outbound calls through Trunk Group call flow, do not configure this option.

Note: CPD results are not processed by SSG when TMPC request is placed on Routing Point.

TGDN

Default Value: No default value
Valid Values: Name of valid `Trunk Group` DN
Changes Take Effect: On restart

Specifies the `Trunk Group` DN that the solution uses when placing outbound calls through SSG, in cases were the Trunk Group call flow is required. For example, use this option for CTI through SIP Server deployments, where the called customer is connected to an NGI application.

Supplements

Related Documentation Resources

The following resources provide additional information that is relevant to this software. Consult these additional resources as necessary.

SIP Server

- *Framework 8.1 SIP Server Deployment Guide*, which provides information to configure and install SIP Server.

Genesys Voice Platform

- *Genesys Voice Platform 8.1 Deployment Guide,* which provides information about installing and configuring Genesys Voice Platform (GVP).
- *Genesys Voice Platform 8.1 User's Guide,* which provides information about configuring, provisioning, and monitoring GVP and its components.
- *Genesys Voice Platform 8.1 Genesys VoiceXML 2.1 Reference Help,* which provides information about developing Voice Extensible Markup Language (VoiceXML) applications. It presents VoiceXML concepts, and provides examples that focus on the GVP Next Generation Interpreter (NGI) implementation of VoiceXML.
- *Genesys Voice Platform 8.1 Legacy Genesys VoiceXML 2.1 Reference Manual*, which describes the VoiceXML 2.1 language as implemented by the Legacy GVP Interpreter (GVPi) in GVP 7.6 and earlier, and which is now supported in the GVP 8.1 release.
- *Genesys Voice Platform 8.1 CCXML Reference Manual,* which provides information about developing Call Control Extensible Markup Language (CCXML) applications for GVP.
- *Genesys Voice Platform 8.1 Troubleshooting Guide,* which provides information about Simple Network Management Protocol (SNMP) Management Information Bases (MIBs) and traps for GVP, as well as troubleshooting methodology.

- *Genesys Voice Platform 8.1 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 8.1 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.

Composer

- *Composer 8.1 Deployment Guide,* which provides information about installing and configuring Composer.
- *Composer 8.1 Help,* which provides information about using Composer, a GUI for developing applications based on VoiceXML and CCXML.

Genesys

- *Genesys Technical Publications Glossary,* which ships on the Genesys Documentation Library DVD, provides a comprehensive list of the Genesys and computer-telephony integration (CTI) terminology and acronyms used in this document.
- *Genesys Migration Guide*, which ships on the Genesys Documentation Library DVD, provides documented migration strategies for Genesys product releases. Contact Genesys Technical Support for more information.
- Release Notes and Product Advisories for this product, which are available on the Genesys Technical Support website at `http://genesyslab.com/support`.

Genesys Voice Platform 8.1 Web Services API wiki, which describes the Web Services API that the Reporting Server supports.

Composer

- *Composer 8.1 Deployment Guide,* which provides installation and configuration instructions for Composer.
- *Composer 8.1 Help,* which provides online information about using Composer, an Integrated Development Environment used to develop applications for GVP and Universal Routing.

Open Standards

- *W3C Voice Extensible Markup Language (VoiceXML) 2.1, W3C Recommendation, 19 June 2007,* which is the World Wide Web Consortium (W3C) VoiceXML specification that GVP NGI supports.
- *W3C Voice Extensible Markup Language (VoiceXML) 2.0, W3C Recommendation, 16 March 2004,* 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 2005,* which is the W3C CCXML specification that GVP supports.
- *W3C Semantic Interpretation for Speech Recognition (SISR) Version 1.0, W3C Recommendation, 5 April 2007,* which is the W3C SISR specification that GVP supports.
- *W3C Speech Recognition Grammar Specification (SRGS) Version 1.0, W3C Recommendation, 16 March 2004,* which is the W3C SRGS specification that GVP supports.

Genesys

- *Genesys Technical Publications Glossary,* which ships on the Genesys Documentation Library DVD and provides a comprehensive list of the Genesys and computer-telephony integration (CTI) terminology and acronyms that are used in this document.
- *Genesys Migration Guide*, which ships on the Genesys Documentation Library DVD, and which provides documented migration strategies for Genesys product releases. Contact Genesys Technical Support for more information.
- Release Notes and Product Advisories for this product, which are available on the Genesys Customer Care website at `http://genesyslab.com/support`.

Information about supported operating systems and third-party software is available on the Genesys Customer Care website in the following documents:

Genesys Supported Operating Environment Reference Guide

Genesys Supported Media Interfaces Reference Manual

For additional system-wide planning tools and information, see the release-specific listings of System Level Documents on the Genesys Technical Support website, accessible from the system level documents by release tab in the Knowledge Base Browse Documents Section.

Genesys product documentation is available on the:

- Genesys Customer Care 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.
- Genesys Online Documentation at docs.genesyslab.com.

Document Conventions

This document uses certain stylistic and typographical conventions—introduced here—that serve as shorthands for particular kinds of information.

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:

80fr_ref_06-2008_v8.0.001.00

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

Screen Captures Used in This Document

Screen captures from the product graphical user interface (GUI), as used in this document, may sometimes contain minor spelling, capitalization, or grammatical errors. 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.

Type Styles

Table 10 describes and illustrates the type conventions that are used in this document.

Table 10: Type Styles

Type Style	Used For	Examples
Italic	• Document titles • Emphasis • Definitions of (or first references to) unfamiliar terms • Mathematical variables Also used to indicate placeholder text within code samples or commands, in the special case where angle brackets are a required part of the syntax (see the note about angle brackets on page 238).	Please consult the *Genesys 8 Migration Guide* for more information. Do *not* use this value for this option. A *customary and usual* practice is one that is widely accepted and used within a particular industry or profession. The formula, $x + 1 = 7$ where x stands for . . .
Monospace font (Looks like `teletype` or `typewriter text`)	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. • Code samples. Also used for any text that users must manually enter during a configuration or installation procedure, or on a command line.	Select the `Show variables on screen` check box. In the `Operand` text box, enter your formula. Click `OK` to exit the `Properties` dialog box. T-Server distributes the error messages 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. Enter `exit` on the command line.
Square brackets ([ ])	A particular parameter or value that is optional within a logical argument, a command, or some programming syntax. That is, the presence of the parameter or value is not required to resolve the argument, command, or block of code. The user decides whether to include this optional information.	`smcp_server -host [/flags]`

Table 10: Type Styles (Continued)

Type Style	Used For	Examples
Angle brackets (< >)	A placeholder for a value that the user must specify. This might be a DN or a port number specific to your enterprise. **Note:** In some cases, angle brackets are required characters in code syntax (for example, in XML schemas). In these cases, italic text is used for placeholder values.	`smcp_server -host <confighost>`

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