



This PDF is generated from authoritative online content, and is provided for convenience only. This PDF cannot be used for legal purposes. For authoritative understanding of what is and is not supported, always use the online content. To copy code samples, always use the online content.

Outbound Contact Deployment Guide

Overview of the GVP VoIP/SIP Server Deployment

4/16/2025

Overview of the GVP VoIP/SIP Server Deployment

Outbound Contact supports a VoIP deployment that enables automated outbound dialing when using SIP Server 8.0 and GVP 8.1 (Media Server).

In this environment, Outbound Contact and Universal Routing components function on top of SIP Server, Media Server, and a SIP Media Gateway.

The **VoIP Environment/Outbound Contact Architecture** figure illustrates the architecture. Instead of requiring new dialing modes for a VoIP deployment, Outbound Contact uses the existing dialing modes and runs these as VoIP modes when Outbound Contact is configured as VoIP-ready (see the procedure [Configuring Outbound Contact and SIP Server for a VoIP Deployment](#)).

GVP 8.0 uses SIP Server and MCP/Media Server(s) call processing in a VoIP environment. It can perform call-progress detection on outbound calls and can process those calls that are connected to customers, based on the logic that is built into VoiceXML scripts. These scripts allow GVP to manipulate user data and perform call control; for example, the call can be completed or transferred to the live agent for further processing. GVP 8.0 can also communicate with OCS by using HTTP or HTTPS (see [Outbound Contact and HTTP Server](#)), which enables automated outbound record processing.

Dialing Algorithm

This deployment supports two dialing modes: Power GVP and Progressive GVP. In this environment, calls originate on the Trunk Group DN and are communicated to the GVP platform after the established CPD is completed using the ApplyTreatment T-Library API.

Calculating the Number of Records

- For the Power GVP (VoIP) dialing mode, the number of records is calculated the same way they are calculated when using the Power GVP dialing mode with GVP 7.6.
- For the Progressive GVP dialing mode: When a dialing session is loaded for a Campaign Group object configured to use this dialing mode, OCS calculates the number of records for retrieval from the database as a percentage (specified in the Optimal Record Buffer Size dialog box) of the total number of channels that are available for this Campaign Group. If more than one Calling List is used in the Campaign, the number of records retrieved from each Calling List is determined by the specified list weight.

OCS then submits outbound call requests to SIP Server based on the total number of channels that are available, while trying to keep as many calls in progress as there are available channels.

OCS retrieves records from the Calling List table in the database to replenish its buffer when the number of records in the OCS buffer is less than the number calculated as a percentage (specified in Minimum Record Buffer Size dialog box) of the total number of channels that are available for this Campaign Group.

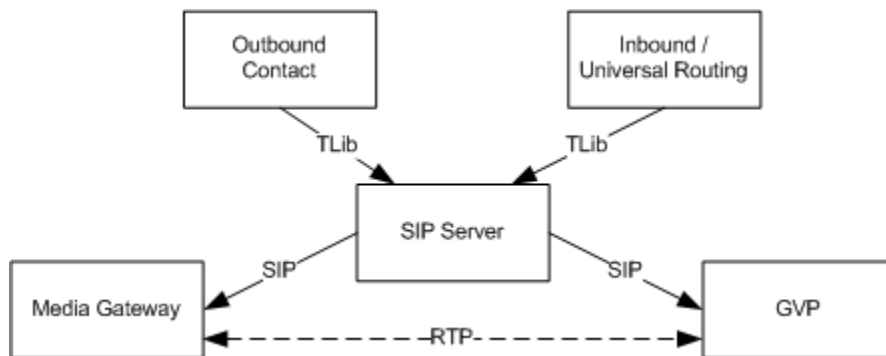
In a VoIP-ready deployment, OCS is responsible for initiation of outbound and engage calls dialing and partial control of call distribution.

Note:

For more information on dialing modes functioning in a VoIP environment, see [VoIP Dialing Modes](#).

If the Campaign Group object is not configured with a Trunk Group DN, the dialing modes function as they would in a non-VoIP environment.

The Genesys components/applications function as shown in the following figure.



VoIP Environment/Outbound Contact Architecture

- Outbound Contact components handle the following:
 - Outbound call pacing
 - Outbound call initiation toward the customer (Transfer, ASM, and Proactive Contact modes)
 - Engaging call initiation toward agents (ASM mode)
 - Outbound call distribution toward agents (Transfer and ASM modes) and toward the voice platform (Power GVP and Progressive GVP modes)
- The Universal Routing components handle routing outbound and engaging calls to the agent.
- SIP Server controls all SIP signaling for initiated calls and also acts as an interface between solution components.
- GVP Media Server handles the following:
 - Playing the basic announcement (ASM Mode)
 - Call recording
 - Call-progress detection
 - Media bridging (ASM mode)
 - Playing voice applications (Power GVP and Progressive GVP dialing modes)

In addition to Media Server, the GVP components also include Resource Manager, Fetching Module, and Squid and Reporting Server. For more information on the GVP components, see the Genesys Voice Platform 8.1 Deployment Guide.

Note:

GVP Resource Manager manages Media Server

	resources. If two or more Media Servers are configured to serve a Trunk Group, Resource Manager distributes the engaging calls and customer calls among them.
--	---------------------------------------------------------------------------------------------------------------------------------------------------------------

Supported Deployments

Outbound supports several types of VoIP deployments for agent connectivity including:

- IP agents located on the same SIP Server.
- IP agents who are on multiple SIP Servers.
- Both IP agents on SIP Server and TDM agents who are located on T-Server.

Resources

The resources available for each dialing session/outbound campaign are determined by the Trunk Group DN that is specified in the Advanced tab of the Campaign Group object (see [Campaign Group Object](#)).

Note:	For GVP 8.1.1, the Trunk Group DN must be named Environment working in any tenant; in GVP 8.1.2 there is no restriction for Trunk Group DN name.
-------	--------------------------------------------------------------------------------------------------------------------------------------------------

On behalf of the Trunk Group DN, SIP Server provides OCS with media-resources usage (total ports and available ports). OCS initially receives this information in EventRegistered when it registers this DN with SIP Server. Once OCS registers Trunk Group DNs, SIP Server automatically provides updated resource/port information for Trunk Group DNs, which OCS uses to run dialing sessions/campaigns. In addition to this information, a dialing session limits the number of ports it can use to the number specified in the Number of Channels parameter in the Campaign Group object (Configuration tab/Advanced section).