GVP Deployment Guide

Configuring GVP Components 2

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Perform these advanced configuration procedures after installation and basic configuration.

Configuring the CTI Connector for Cisco ICM

When you install the CTI Connector, you can select the CTI Framework that is appropriate for your environment Genesys CTI or Cisco ICM. Use the procedure in this section to configure the CTI Connector if you selected Cisco ICM.

- Procedure: Configuring the CTI Connector for Cisco ICM Integration

Procedure: Configuring the CTI Connector for Cisco ICM Integration

1. Log in to Genesys Administrator.
2. On the Provisioning tab, select `Environment > Applications`.
3. Double-click the CTI Connector Application that you want to configure.
   The Configuration tab appears.
4. Click the Options tab.
5. If you want to use the Call Routing Interface (CRI), in the icmc section:
   - Configure the ICMInterface option with the CRI value.

> **Tip**
> During installation, when you select Cisco ICM, the Service Control Interface is initialize by default.

- Configure the TrunkGroupID option with an applicable value.

For Single Tenant Environments:

6. In the `Tenant1` section:
    - Enter the tenant name in the `TenantName` configuration option value field.
    - Change the value of the Ports configuration option, as required. For example, `8000` (or retain the default, `9000`).

For Multi-Tenant Environments:

7. Copy the `Tenant1` section and rename it for each additional tenant. For example: `Tenant2`, `Tenant3`, `Tenant4`.
8. For each newly created tenant:
    - Enter the tenant name in the TenantName configuration option value field.
    - Change the value of the Ports configuration option, as required. For example, `8000` (or retain the default, `9000`). Ensure that there are no duplicate ports configured across all tenants.

> Tip
>
> You can specify a comma-separated list of listener ports for a single tenant, one for each VRU-PG. For example, 8000, 9000, 10000.

9. Save the changes.
10. Configure an IVR Profile to support ICM call flows, see *Chapter 6 in the Genesys Voice Platform 8.5 User s Guide*.

Provisioning the PSTN Connector

The procedures in this section describe how to configure the mandatory parameters for the Public Switched Telephone Network (PSTN) Connector Application object and the how to integrate the PSTN Connector with SIP Server. There are many more configurable parameters for the PSTN Connector, all of which are optional. For a complete list and description of configuration options, see the *Genesys Voice Platform 8.5 User's Guide*.

The PSTN Connector component is required if you are planning to migrate from GVP 7.x Voice Communication Server (VCS), or a VoiceGenie (VG) TDM interface to GVP 8.1.2 or later.

- Procedure: Configuring the PSTN Connector
- Procedure: Configuring a Trunk DN for the PSTN Connector

Procedure: Configuring the PSTN Connector

Prepare the PSTN Connector to manage inbound and outbound calls for GVP.

1. Verify that:
    - All of the GVP components are installed. See Procedure: Using the Deployment Wizard to Install GVP, on page 221.
    - The connections to Message Server, SIP Server, and the SNMP Master Agent are configured in the PSTN Connector Application object. See *Procedure: Creating a Connection to a Server, on page 243*.
    - SIP Server is installed. See the Voice Platform Solution 8.1 Integration Guide.
2. Log in to Genesys Administrator.
3. On the Provisioning tab, select Environment > Applications.
4. Select the PSTN Connector Application object that you want to configure.
   The Configuration tab appears.
5. Click the Options tab, and from the View drop-down list, select the Mandatory Options view.
6. In the DialogicManager_Route1 and GatewayManager sections, enter the values for the mandatory options as shown in the table below.

Table: PSTN Connector Mandatory Parameters

Section	Option	Value
DialogicManager_Route1	RouteType	Specify one of three route types or call directions for this route, enter: • 0 for Inbound • 1 for outbound • 2 for In/Out (See Note below, in this table.)
	Signaling Type	Specify one of five signaling types, enter: • 0 for T1-ISDN (PRI) • 1 for Analog • 2 for E1-ISDN (PRI) • 3 for T1-RobbedBit • 4 for E1-CAS
	Channels	Specify the ports for this route by using the format, [<Card>:<PortRange>,<Card>:<PortRange>] You can provision more than one board in a route and a partial range of ports in a board for example: • 1:1-23 • 1:1-23,2:1-23 • 1:1-30,2:1-30

Section	Option	Value
		• 1:1-12,2:1-15
GatewayManager	SIP Destination IP Address	Enter the SIP endpoint IP address that will receive SIP calls from the PSTN Connector. (This is the IP address for SIP Server or the Resource Manager, depending on your configuration.)
	SIP Destination Port Number	Enter the SIP endpoint port number of the server that is configured in SIP Destination IP Address.
MediaManager	Supported Local Codec Type	Enter the audio format that is in use on the TDM trunk: • 0 - Mulaw • 8 - ALaw

Notes

- The Inbound & Outbound route type is supported only on ISDN (PRI) lines. If you select one of these route types, ensure that you use a compatible signaling type.
- If you are using a T1-Robbed Bit or E1-CAS interface, options in the T1rb options group must be configured, specifically the T1rbProtocolFile option, see the component metadata.
- For JCT boards only, a separate span is required to support ASR or recorded VoiceXML applications in T1-ISDN, E1-ISDN, or E1-CAS environments. The MediaVoxResourceBoard option must be configured with route number that is used for CSP.
- When JCT boards are used with the PSTN Connector, and the VoiceXML application uses ASR or recording media, not all the spans can be used for call handling. For each span that is configured to take calls, there must be another dedicated span for streaming echo cancelled audio to the Media Control Platform. Therefore, if Route1 is configured to handle calls on span1 (for example, ports=1:1-23), the `MediaVoxResourceBoard` option under the `DialogicManagerRoute1` section should be set to 2. Repeat the same steps if you have configured a DialogicManager_Route2 section.

This restriction does not apply in the following scenarios:

- The VoiceXML application is a pure DTMF application (does not use ASR or recording media).
- The VoiceXML application uses ASR but the JCT board is configured with the T1-Robbed Bit protocol.

7. Create additional `DialogicManager_Route`<N> sections as required (for example, you may want to create a section for inbound ports and one for outbound ports):
   a. From the View drop-down list, select either Advanced View (Options) or Advanced View (Annex).
   b. Right-click on the Section column heading and select New.
   c. Enter `DialogicManager_Route2` for the Section name, Description for the Option name, and Route2

Information for the Value.

d. Click **OK**.

e. Copy and paste all of the options from the `DialogicManager_Route1` and `DialogicManager_Route2` sections, modifying the mandatory values as required.

f. Repeat these steps as required.

> Tip
>
> **Dialogic Circular Buffer Size**
> When you configure the PSTN Connector application, set the value for [DialogicManager] DialogicTransferBufferSize to 2048. This specifies the size of the Dialogic Circular buffer which is used for transferring data to the Dialogic firmware. The default value of this parameter on Windows is different, but must be set to 2048 for Linux. If not, the Dialogic may return LOW_WATER_MARK warnings during initial play of the media, which interferes with normal audio play and may result in the prompt being cut off.

8. Save the configuration.
9. Configure the Trunk and Trunk Group DNs for the PSTN Connector. See Procedure: Configuring a Trunk DN for the PSTN Connector.

Procedure: Configuring a Trunk DN for the PSTN Connector

Configure SIP Server with a Trunk DN that points to the PSTN Connector Application object, to ensure that outbound calls can be routed to a specific PSTN Connector instance.

You can deploy multiple PSTN Connectors, however, you must ensure that SIP Server routes the outbound call to the same PSTN Connector instance as the inbound call. This procedure includes configuration options to enable this functionality.

1. Verify that the PSTN Connector is installed and configured. See Procedure: Configuring the PSTN Connector.
2. Log in to Genesys Administrator.
3. On the Provisioning tab, select `Environment > Switching > Switches`.
4. Double-click the switch that you want to configure.
   The Configuration tab appears.
5. On the DNs tab, select **New**.
6. In the General section, enter values for the mandatory fields, selecting **Trunk** from the Type drop-down list.
7. In the Switch pane, double-click the Trunk DN you created in Step 4.
8. On the Options tab, select Advanced View (Annex) from the View drop-down list.
9. Right-click on the Section column, and select **New**.
10. In the New Option dialog:
    - In the Section field, enter `TServer`.
    - In the Name field, enter `contact`.

- In the Value field, enter the IP address and port number of the PSTN Connector separated by a colon for example, `10.10.10.101:5060`

11. In the DNs pane, click **New**.
12. In the TServer section, add the following options and values:
    - `prefix = <xyzz>` where `<xyzz>` represents a number which, if present in the dial string of the outbound call, enables the SIP Server to route the call to the PSTN Connector instance that is configured with this prefix.
    - `replace-prefix = <empty String)` where `<empty String)` represents an empty string to ensure that the prefix added by Resource Manager to the destination number string is removed by the SIP Server before the call is forwarded to the same PSTN Connector instance.
13. Save the configuration. For information about how the PSTN Connector fits into a common VPS deployment architecture, see the Supported Architecture chapter in the Voice Platform Solution 8.1 Integration Guide.
14. If required, complete the post-installation activities for the Supplementary Services Gateway. See Provisioning the Supplementary Services Gateway.

Provisioning the Supplementary Services Gateway

The following procedures describe how to configure the mandatory parameters for the Supplementary Services Gateway Application object and the how to integrate the Supplementary Services Gateway with SIP Server. There are many more configurable parameters for the Supplementary Services Gateway, all of which are optional. For a complete list and description of configuration options, see the GVP 8.5 User's Guide.

The Supplementary Services Gateway is an optional component and is required only if you intend to support outbound call campaigns in your deployment.

> **Tip**
> Multiple instances of the Supplementary Services Gateway can be installed on the same host; however, the HTTPPort and HTTPSPort parameters must have unique values and cannot be the same for more than one instance.

This section contains the following procedures:

- Configuring the Supplementary Services Gateway
- Configuring DNs for the Supplementary Services Gateway
- Configuring a Routing Point DN
- Configuring a Voice Over IP Service DN
- Configuring a Voice Treatment Ports DN

Procedure: Configuring the Supplementary Services Gateway

Prepare the Supplementary Services Gateway to receive and respond to outbound call initiation requests from TAs.

1. Verify that:
   - All other GVP components are installed. See Procedure: Using the Deployment Wizard to Install GVP, on page 221.
   - The connections to Message Server, SIP Server, and the SNMP Master Agent are configured in the Supplementary Services Gateway Application object. See Procedure: Creating a Connection to a Server.
   - SIP Server is installed. See the Voice Platform Solution 8.1 Integration Guide.
2. Log in to Genesys Administrator.
3. On the Provisioning tab, select `Environment > Applications`.
4. Select the Supplementary Services Gateway Application object that you want to configure. The Configuration tab appears.
5. Click the Options tab, and select Advanced View (Options) from the View drop-down list.
6. In the Tenant1 section, enter Environment in the Value field of the TGDN option. (The value that is configured in the TGDN parameter is used as the tenant name.)
7. Create additional `Tenant<`*n*`>` sections, as required—for example: Tenant2, Tenant3, and so on. To create new sections by copying and pasting options, see Step 6 in Procedure: Configuring the PSTN Connector.
8. Save the configuration.
9. Configure the Trunk and Trunk Group DNs to route for Resource Manager and external numbers. See Procedure: Configuring DNs for the Supplementary Services Gateway.

Procedure: Configuring DNs for the Supplementary Services Gateway

Configure SIP Server with a Trunk Group DN that points to the Resource Manager Application object, and a Trunk DN that is used to route calls to external numbers.

The Trunk Group DN must have the same name as the tenant for which the Supplementary Services Gateway will make calls. This configuration enables the SUBSCRIBE and NOTIFY messages between SIP Server and the Resource Manager.

1. Verify that the Supplementary Services Gateway is installed and configured. See Procedure: Provisioning the Supplementary Services Gateway.
2. Log in to Genesys Administrator.
3. On the Provisioning tab, select Environment > Applications.
4. Select the SIP Server Application object you want to configure. The Configuration tab appears.
5. On the Options tab, in the View drop-down list, select Advanced View (Options).
6. In the TServer section, change the value of the following Options:
   - `am-detected = connect`

- `fax-detected = connect`
- `cpd-info-timeout = 7`
- `sip-invite-treatment-timeout = 30`

7. Configure a DN on the SIP switch of type Trunk Group, and, for the DN name, enter the name of the tenant for example, `Tenant1`.
8. Configure a DN on the SIP switch of type Trunk as the endpoint. The endpoint is the destination of the outbound call for example, a softphone.
   Configure Trunk Group DN
9. After the <TenantName> Trunk Group DN is configured, enter the values for the following parameters in the TServer section of the Annex, as shown in the figure below.
   - `subscription-id = <TenantName>` Must be the name of the tenant that is associated with the Resource Manager. (The DN, tenant, and subscription-id must all have the same name: <TenantName>.

   > **Tip**
   > Tenant names are case sensitive and must be used consistently, especially in outbound scenarios. For example, if you create the tenant names, tenant1 and Tenant1, they are treated as two separate tenants. The value that is configured in the TGDN parameter is used as the tenant name.

   - `contact = sip:172.24.133.50:5060` The IP address and port number of the Resource Manager. (The value shown here for contact is only an example.)
   - `cpd-capability = mediaserver` This configuration designates the Media Server module of the Media Control Platform as the CPD provider.
   - `request-uri = sip:msml@172.24.133.50:5060;gvp-tenant-id=TenantName` The value of this parameter should point to the Resource Manager, and the user part must contain msml.
   - `userdata-map-format = sip-headers-encoded`
   - `make-call-rfc3725-flow = 1`
   - `refer-enabled = false`
   - `ring-tone-on-makecall = false`

   Configure Trunk DN
10. After the Trunk DN is configured, enter the value for the following parameter in the TServer section of the of the Advanced View (Annex):
   - `contact = sip:172.21.193.61:10000` The IP address and port number of the external party. (The value shown here for contact is only an example).

> **Tip**
> If CPD is enabled at Media Gateway, configure CPD at the Trunk. If CPD is enabled at both Trunk and Trunk Group, CPD enabled on the media gateway takes precedence by default.

For information about how to configure the Supplementary Service Gateway within the VPS, see the VP Solution 8.1 Integration

Guide.

11. (Optional) Configure a Routing Point DN. See Configuring a Routing Point DN.
12. Complete the post-installation activities for the Call Control Platform if you intend to use it for outbound calling. See Procedure: Configuring the Call Control Platform.

Procedure: Configuring a Routing Point DN

Create a Routing Point DN to use for outbound calls with the legacy GVPi.

Use this procedure if you want to support your outbound solution by making outbound calls through a Routing Point DN (instead of a Trunk Group DN) by using the legacy GVPi and CTI functionality. Create a Routing Point DN for each tenant in your environment.

1. Verifythat a Trunk DN exists on the SIP switch for outbound calls. See the VP Solution 8.1 Integration Guide.

   Configure the Routing Point DN

2. Log in to Genesys Administrator.
3. On the Provisioning tab, select `Switching > Switches`.
4. Double-click the SIP Server Switch object you want to configure.
   The Configuration tab appears.
5. Click the DNs tab and select **New**.
6. On the DN Configuration tab:
   a. In the Number field, enter valid Route Point DN number. (Ensure there is no conflict with the Trunk Group DN that was created for receiving port details.)
   b. In the Type field, select **Routing Point** from the drop-down list.
7. On the Options tab, click **New**.
8. In the New Option dialog box:
   a. In the Section field, enter `TServer`.
   b. In the Name field, enter `partition-id`.
   c. In the Value field, enter the name of the tenant (for which this Route Point DN is configured). This configuration enables SIP Server to select an MSML service with the same partition-id.

   Configure the SSG Application

9. On the Provisioning tab, select `Environment > Applications`.
10. Select the Supplementary Services Gateway Application object that you want to configure.
    The Configuration tab appears.
11. Click the Options tab, and select **Advanced View (Options)** from the View drop-down list.
12. In the Tenant1 section:
    a. For the RPDN option, enter a value that matches the RPDN. For example, `RP 2000`.
    b. For the TGDN option in the Value field, enter the tenant name. The value that is configured in the TGDN parameter is used as the tenant name.

13. If you want to send outbound calls for multiple tenants, create additional `Tenant <N>` sections configured with the RPDN option, as required for example, Tenant2, Tenant3, and so on. To create new sections by copying and pasting options, see Step 6 in Procedure: Configuring the PSTN Connector.
14. Save the configuration.
15. Configure a VoIP Service DN. See Procedure: Configuring a Voice Over IP Service DN, below:

Procedure: Configuring a Voice Over IP Service DN

Create and configure a VoIP Service DN that is used by SIP Server to initiate MSML dialogs (to obtain CPD) when a request is received on the Routing Point DN.

The VoIP Service DN works in conjunction with the Routing Point DN. For outbound calls, SIP Server selects the MSML service type with the value that is configured in the partition-id option (either no partition-id or the partition-id that is configured in the SIP Server application).

1. Verify that a Routing Point DN has been configured. See Configuring a Routing Point DN.

   Configure the Voice Over IP Service DN
2. Log in to Genesys Administrator.
3. On the Provisioning tab, select `Switching > Switches`.
4. Double-click the SIP Server Switch object you want to configure.
   The Configuration tab appears.
5. Click the DNs tab and select **New**.
6. On the DN Configuration tab:
   a. In the Number field, provide a unique, valid number for the DN.
   b. In the Type field, select Voice Over IP Service from the drop-down list.
7. On the Options tab, click **New**.
8. In the New Option dialog box:
   a. In the Section field, enter `TServer`.
   b. In the Name field, enter `contact`.
   c. In the Value field, enter the IP address and port number of the Resource Manager, for example, `sip:172.24.133.50:5060`.
9. Repeat the two previous steps to add the following options and values:
   - `cpd-capability = mediaserver`
   - `partition-id = <tenant_name>`
   - `service-type = msml`
   - `subscription-id = <tenant_name>`
10. Click **Save** and **Close**.
11. Repeat Steps 4 to 9 to create a second VoIP Service DN to play Treatments (`service-type = treatment`).

12. (Optional) Configure VTP Ports. See Procedure: Configuring a Voice Treatment Port's DN, below:

Procedure: Configuring a Voice Treatment Ports DN

Create and configure a Voice Treatment Ports (VTP) DN to play IVR Profile VoiceXML dialogs. You can create any number of VTP DNs to control the number of simultaneous outbound requests that can be placed on the route point.

1. Log in to Genesys Administrator.
2. On the Provisioning tab, select `Switching > Switches`.
3. Select the SIP Server Switch object you want to configure.
   The Configuration tab appears.
4. Click the DNs tab and select **New**.
5. On the DN Configuration tab:
   a. In the Number field, enter a valid Voice Treatment Port as the number.
   b. In the Type field, select **Voice Treatment Port** from the drop-down list.
6. On the Options tab, click **New**.
7. In the New Option dialog box:
   a. In the Section field, enter `TServer`.
   b. In the Name field, enter `contact`.
   c. In the Value field, enter the IP address and port number of the Resource Manager, for example, `sip:172.24.133.50:5060`.
8. Repeat Steps 6 and 7 to add the following options and values:
   - `prefix = mediaserver`
   - `request-uri = sip:<vtp DN name>@<RM contact>gvp-tenant-id=<tenantName>`
   - `event-ringing-on-100trying = true` (when CTI Connector is used on the call, otherwise, false)
   - `cpd-capability = mediaserver` (the Media Control Platform performs CPD)
   - `userdata-map-filter = constant string "gsw-ivr-profile-name,gsw-session-dbid, OutboundData,AnswerClass,outbound-ivr-call"`

     Where:
   - `OutboundData` is the filter that is provided to pass user data from the Supplementary Services Gateway to the IVR Application. For information about attaching user data to outbound calls, see the Voice Platform Solution 8.1 Integration Guide.
   - `outbound-ivr-call` must be passed to the Resource Manager to ensure the media service call is not dropped until the route is used when a temporary double-counting of IVR Profile usage can happen when the IVR VoiceXML call leg is placed.
   - `AnswerClass` is required if the CPD result must be passed to VoiceXML application. Add the `GVP-IVRPort` parameter if the Supplementary Services Gateway is integrated with the PSTN Connector and the `GVP-PSTNC-DBID` parameter if the Supplementary Services Gateway is integrated with the CTI Connector, for example: `userdata-map-filter = constant string "gsw-ivr-profile-`

```
name,GSW_SESSION_DBID,gsw-session-dbid,OutboundData,AnswerClass,GVP-IVRPort,GVP-
PSTNC-DBID"
```

> **Tip**
>
> Genesys recommends that you create a strategy to select a available VTP DN (from the set of VTP DNs that you created) to load on the Routing Point DN. For more information about routing strategies, see Chapter 11 of the Voice Platform Solution 8.1 Integration Guide.

Preparing the Call Control Platform for Outbound Calling

This section describes how to prepare the Call Control Platform to make outbound calls. The Call Control Platform is an optional component.

Procedure: Configuring the Call Control Platform

Configure the Call Control Platform Application object to make outbound calls.

1. Verify that:
    - All of the GVP components are installed. See Procedure: Using the Deployment Wizard to Install GVP.
    - The Call Control Platform is integrated with the Resource Manager. See Procedure: Integrating Application Objects with Resource Manager.
    - A Call Control Platform connection to the Message Server is created. See Procedure: Creating a Connection to a Server.
2. Log in to Genesys Administrator.
3. On the Provisioning tab, select `Environment > Applications`.
4. Select the Call Control Platform Application.
5. Click the Options tab, and scroll to the `mediacontroller` section.
6. Click the Value field of the `sipproxy` option, and then enter <IP_RM>:<SIPPort_RM>
   Where `IP_RM` is the IP address of the Resource Manager, and `SIPPort_RM` is the SIP port of the Resource Manager.
7. Click the Value field of the `bridge_server`, and then enter <IP_RM>:<SIPPort_RM>
   Where `IP_RM` is the IP address of the Resource Manager, and `SIPPort_RM` is the SIP port of the Resource Manager (`5060` is the default).
8. Click **Apply**.
9. Save the configuration.
10. Complete the post-installation activities for the Resource Manager. See the next section, *Using Resource Groups*.