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GVP Documentation Supplement

Genesys Voice Platform 85

7/10/2024

Table of Contents

Supplement to Documentation for: GVP 8.5 and Media Server 8.5

This document lists additions and changes to the docs for Genesys Voice Platform 8.5.x and Media Server 8.5.x, as part of Continuous Delivery.

Latest Available IPs

The following tables document the latest IPs available for the Genesys Media Server and Genesys Voice Platform products.

Genesys Media Server

Component	8.5.1 available	Latest IP Version	Windows	Linux
Resource Manager (RM)	Y	8.5.185.37	Y	Y
Media Control Platform (MCP)	Y	8.5.185.34	Y	Y
T-Server-CUCM to Media Server Connector	Y	8.5.184.06	Y	Y
Reporting Server (RS)	Y	8.5.181.77	Y	Y
GVP Reporting Plugin for GAX	Y	8.5.151.29	Y	Y
Management Information Base (MIB)	Y	8.5.130.40	Y	Y

See the Supported Operating Environment: Genesys Voice Platform page for more detailed information and a list of all supported operating systems.

Genesys Voice Platform

Component	8.5.1 available	Latest IP Version	Windows	Linux
MRCP Proxy	Y	8.5.185.06	Y	Y
Supplementary Services Gateway (SSG)	Y	8.5.160.97	Y	Y
CTI Connector (CTIC)	Y	8.5.160.80	N	Y
		8.5.160.73	Y	Y
Squid Caching Proxy	Y	8.5.100.06	Y	N/A
Call Control Platform (CCP)	Y	8.5.010.82	Y	N
		8.5.010.81	Y	Y
Policy Server (PS)	N	8.5.010.10	Y	Y

Note: See the *Genesys Supported Operating Environment Reference Guide* page for more detailed information and a list of all supported operating systems.

New in 8.5.1

Date	Feature
05 February 2016	• Users can now use the GVP Reporting plugin for GAX to provision IVR profiles and manage DID groups for IVR profiles. See the GVP Reporting plugin for GAX Release Note. **Note:** In large-scale deployments of GVP (with 10,000+ DIDs), users may experience up to 15-20 seconds response times when editing or saving DIDs in a DID group, through the GVP Reporting plug-in for GAX. • Tenant Administrators can now provision DIDs to DID Group mapping. **Note:** DID group provisioning succeeds when you remove the conflicting entries in overlapping ranges (example: 100-200 and 150-250). • Backend filtering no longer removes the call record from a report when you use an IVR profile or Tenant or component in the filter. Instead, certain filtering rules apply. See the GVP Reporting plug-in for GAX Release Note. • Resource Manager (RM) has three new configuration options in the CTI Connector (CTIC) Logical Resource Group (LRG) for handling CTIC failover. See the Resource Manager Release Note. • The TCP setup timer is now configurable. See the Resource Manager Release Note. • MRCP Proxy supports provisioning MRCP resources with the same URI, if the resource names or types are different.
January 2016	• MCP uses the SpiderMonkey 1.7 engine to perform ECMAScript (JavaScript) processing. SpiderMonkey 1.7 supports ECMA-262 3rd edition.
15 December 2015	• The `Sip.Body` value in the `[vxmli] session_vars` configuration option enables access to the body of SIP INVITE messages. See the MCP Release Note. • Use the configuration parameter `[callmgr] enable_sip_response_in_transfer_metric` to configure Media Control Platform to append the SIP response code (when it is available) to `transfer_result` metrics. See the MCP Release Note. • **TCP Timer Setup**—You can configure the wait time to keep a needed resource available, when waiting to establish a TCP or TLS connection. See the MCP Release Note.
16 November 2015	• You can use the GVP Reporting plugin for GAX to "self-service" provision IVR

Date	Feature
	profiles, Map individual DIDs to DID groups, and Map DID Groups to IVR profiles. Read complete details here.

28 August 2015	• The Supplementary Services Gateway now supports including custom HTTP response headers in the HTTP responses to HTTP requests. Enable this behavior with the `[http]ResponseHeaders configuration parameter`. Enter the value in this format: `Header1:Value1|Header2:Value2`
	Notes:
	  ◦ If your value includes the characters | or :, precede them with a backslash (\\\| or \\:).
	  ◦ The default value for this parameter will be set to `X-Frame-Options:DENY`—part of Genesys software defense against “clickjacking”.
	  ◦ This behavior also applies to accessing the SSG root page.
	• The Media Control Platform supports DTMF Masking while agent and caller are in conference with IVR. See Managing DTMF Clamping in MSML Recordings.
	• Now you can configure the duration of the MP3 recording buffer. Use the option `[mpc] mediamgr.recordmp3audiobuffer` (in the MCP application) to specify the duration of the audio buffer for MP3 recording, in milliseconds. `mediamgr.recordmp3audiobuffer` must be an integer `2000` or greater; the default is `4000`.
	Any change takes effect at start/restart.
	• The Media Control Platform's NGI VXML interpreter supports caching of JavaScript (JS) content, which improves performance.
	• The Media Control Platform supports mp3 compression at 8 kbps for mono recording. To enable 8 kbps mono recording, make these settings:
	  ◦ `[mpc] mp3.bitrate=8`
	  ◦ msml.record.channels=1 OR `msml.record.channels2=1`
	The default compression remains 16 kbps.
	• The Media Control Platform supports mono-channel recording during MSML GIR recording for all file formats.
	  ◦ Use the parameters (both in the `gvp.service-paramaters` section of the IVR profile `record`) to specify the channel information:
	Both options `recordingclient.channels=fixed` and `recordingclient.channels2=fixed` can have the possible value 1(mono) or 2(stereo, the default).
	If a parameter is missing from the IVR profile configuration, then MCP uses the configuration parameters `msml.record.channels` and `msml.record.channels2` (which specify the number of channels that MCP must use for MSML recording to dest2). These two have the same possible values (1=mono and 2=stereo, the default).
	Any change takes effect immediately.
	  ◦ Use the configuration option `[msml]record.amazonallowpublicaccess` to enable public download access to an MSML recording file that was uploaded to Amazon s3. Set to `true` to enable access to the uploaded recording file,

<table>
<tr><th>Date</th><th>Feature</th></tr>
<tr><td></td><td>and to <code>false</code> (the default) to disable access.
This option will grant access to both the primary recording destination (<code>recdest</code>) and the secondary recording destination (<code>recdest2</code>), if you configure both destinations to use the s3 URI format (<code>s3:bucketname</code>).
Any change takes effect at start/restart.</td></tr>
<tr><td>17 April 2015</td><td>Voiceprint Carrier Message Detection Setup</td></tr>
<tr><td>23 March 2015</td><td>The Media Server function Call Progress Detection (CPD) now performs voice print analysis and beep analysis to identify the specific preconnect carrier messages that occur in different countries.
Media Server's configurable database of preconnect tones is initiated during installation and loaded when Media Server starts. You can update the database with different carrier messages at any time, without stopping Media Control Platform.
Other features include the ability to leave postconnection messages such as voicemail.
You can read about additional CPD functionality in "Appendix C: Tuning Call Progress Detection" of the GVP 8.5 User's Guide.
Some new VoicePrint Configuration Options support this functionality.
The Reporting Server (RS) adds support for Windows 2012 64-bit and MS SQL Server 2012, in native 64-bit mode using the 64-bit version of the Java Virtual Machine. RS also adds support for Standard and Enterprise editions of the SQL Server 2012 database.
Resource Manager can now be configured to reject a call request if the geo-location of the targeted Logical Resource Group (LRG) does not match the geo-location attribute of the call request. This extends to all calls, a behavior that previously applied only to recording solution calls. Read about this behavior and the option that controls it, <code>reject-on-geo-location-nomatch</code> (new in release 8.5.1), in Locating Resources Using Geo-Location.</td></tr>
<tr><td>18 December 2014</td><td>GVP added support for Genesys Interactive Recording (GIR):
Media Control Platform added two new mp3 encoding compressions—16kbps (new default) and 24kbps—to the existing configuration option <code>[mpc]mp3.bitrate</code>.
VP Reporting Plugin for GAX added new support and compatibility requirements:
Added support for Management Framework 8.5 and Genesys Administrator Extension 8.5.
Added support for Windows Server 2012 R2 64-bit.
The plugin now requires GAX 8.5.0; support for GAX 8.1.4 is discontinued.</td></tr>
</table>

Date	Feature
	• VP Reporting Plugin for GAX Help is now online. • Management Information Base (MIB) added support for Windows Server 2012 R2 64-bit.
19 September 2014	• Added support for Windows Server 2012 64-bit for the Resource Manager (RM), Media Control Platform (MCP), and CTI Connector (CTIC) components.
15 July 2014	• Added support for Nuance Vocalizer 6.0.2 and Nuance Speech Server 6.2.5. • Route Unavailable Wakeup • Reliable Connection Retry • Third Party Recording with MCP and the Resource Manager • Configuration Options Changed

Documentation Corrections

Document	Corrections
Genesys Media Server 8.5 Deployment Guide	• Using the parallel-forking load-balance-scheme • Only one SNMP Master Agent per GVP Component • Remove Note about legacy GVPi functionality support • Horizontal version of table 1 on page 20 • File-based Call Recording • Remove references to Remdial
GVP 8.5 User's Guide	• Correct Name for transaction object ID • Multiple Parameter Support in Operational Parameter Management • Audio MP3 Support • Missing Audio File Solution • Correction to the `SSL Version` value in the GVP User's Guide • CCP Metric (1013) not logged or sent to RS • Limits to SRTP implementation • Recommendation about Database Retention Periods

<table>
<tr><th>Document</th><th>Corrections</th></tr>
<tr><td></td><td>Remove references to Remdial</td></tr>
<tr><td>GVP 8.1 Troubleshooting Guide</td><td>Documentation equalization about Reporting Server Options</td></tr>
<tr><td>GVP 8.1 Application Migration Guide</td><td>GVP 8.1+ Handles Play Treatment & Application Differently</td></tr>
<tr><td>GVP 8.1 Voice XML Help</td><td>How to access a nomatch DTMF result
New keyword Record for gvp:dest Attribute of VXML <log> tag
Correction to ECMAScript variable in <submit> tag
Corrections to the VXML tags <send>, <receive>, and <block>
Correction: The SSML <lexicon> tag does not have Audio as a Parent
Correction to <code>bargein</code> behavior with GVP interpreters
Enable MCP to accept the audio skip duration value specified in a VCR Control application
Update attribute to gvp:recordutterance
Remove "gvp:expr" <grammar> attribute</td></tr>
<tr><td>VP Solution 8.1 Integration Guide</td><td>Corrections to a TDM PBX Method Explanation
Resource Manager IP Address and Lost Host Name are no longer mandatory parameters</td></tr>
<tr><td>Multiple Documents</td><td>Warning: The Verbose level of debug tracing affects GVP production environment performance.
This warning applies to all mentions of the debug tracing setting <code>verbose</code>:

Warning: Debug tracing has a significant impact on GVP performance under load, especially the <code>verbose</code> setting. Genesys recommends that you enable debug tracing in GVP production systems only when recommended by Customer Care or Engineering.
Trace or tracing is mentioned in the context of debugging or a logging in several places, in the documentation listed below. In each instance, readers should be aware of its performance impact:
Genesys Media Server 8.5 Deployment Guide
[+] Procedure: Integrating the Connector with Resource Manager and Cisco T-Server

Appendix A: Deploying the T-Server-CUCM to Media Server Connector
Section: Deploying the Connector</td></tr>
</table>

Document	Corrections
	Procedure: Integrating the Connector with Resource Manager and Cisco T-Server, page 141 • **[+] Table 23: Selected T-Server-CUCM to Media Server Connector Options** Appendix A: Deploying the T-Server-CUCM to Media Server Connector Section: Customizing the Configuration Table 23: Selected T-Server-CUCM to Media Server Connector Options, page 144 • **[+] Task Summary: Configuring the Connector Common Features** Appendix A: Deploying the T-Server-CUCM to Media Server Connector Section: Customizing the Configuration Task Summary: Configuring the Connector Common Features, page 154 **GVP 8.5 User's Guide** • **[+] Table 8: Selected Configuration Options—log Section** Chapter 3: Configuring Common Features Section: Configuring Logging Table 8: Selected Configuration Options—log Section, page 63-67 • **[+] Table 9: Default Values for Selected log Options** Chapter 3: Configuring Common Features Section: Configuring Logging Table 9: Default Values for Selected log Options, page 68 • **[+] Table 15: Selected Policy Server Configuration Options** Chapter 5: Configuring Policy Server Important Policy Server Section: Configuration Options Table 15: Selected Policy Server Configuration Options, pages 100-101 • **[+] Table 17: IVR Profile Configuration Options for GVPi** Chapter 6: Provisioning IVR Profiles IVR Profile Section: Configuration for GVPi Table 17: IVR Profile Configuration Options for GVPi, page 125 • **[+] Table 25: Selected MRCP Proxy Configuration Options** Chapter 8: Configuring the MRCP Proxy Important MRCP Proxy Section: Configuration Options Table 25: Selected MRCP Proxy Configuration Options, pages 200-205 **Trace** also appears as a log setting in all sections of the Genesys Voice Platform Configuration Options online documentation, and in many sections of the Genesys Voice Platform Media Server Configuration Options online documentation.

Deployment Notes

Date	Features
19 December 2014	Follow the instructions in:

Date	Features
19 September 2014 15 July 2014	• GVP 8.5 Deployment Guide • Genesys Media Server 8.5 Deployment Guide • Migrating to GVP 8.5
15 July 2014	Two new features require some configuration, following deployment: • Route Unavailable Wakeup • Reliable Connection Retry

Tip

GVP 8.5.1 components are compatible with GVP 8.5 components not yet on version 8.5.1.

GVP Options Support IVR Recording

Features

This release of GVP introduces a new IVR Profile service-type: **record**, which applies to recording-enabled IVR Profiles.

New Options

Resource Manager Support for Recording IVR Profiles

Resource Manager (RM) can locate and select a recording-enabled IVR Profile. This functionality mimics GVP's existing usage of an MSML VoIP Service DN when the RURI parameter **media-service** is set to `record`.

When this functionality enabled, RM finds all IVR Profiles for the selected tenant whose variable **gvp.general.service-type** is set to `record`. Some details:

- If a call contains a value for **X-Genesys-geo-location**, RM selects the IVR Profile with a matching value in **gvp.general.geo-location**. If multiple profiles match, RM may choose any one of them.
- If a call does *not* contain a value for **X-Genesys-geo-location** and there is only one record-enabled IVR Profile, RM uses that one.
- If neither of the above situations listed above applies, RM selects the IVR Profile named **record**.
- Finally, if none of the above situations apply, RM selects the default IVR Profile as the recording IVR Profile.

gvp.general.geo-location
Configured where: IVR Profile-level
Valid values: Any valid string, such as "abcd" or "Kathmandu"
Default value: No default value (option not defined)

When **X-Genesys-geo-location** is provided in a call's initial INVITE, RM picks up a recording profile with the same value in `gvp.general.geo-location`.

gvp.policy.voicexml-recording-allowed
Configured where: IVR Profile-level (or at tenant-level). This is a TYPE-III policy parameter.
Valid values: `true`, `false`
Default value: `true`

Enables recording for the selected IVR Profile.

gvp.general.service-type
Configured where: IVR Profile-level
New additional valid value: `record`

This option reports the service type of an IVR Profile.

Existing Options, Extended

Use these existing options to configure IVR Recording.

callrec_default_type
Application: Media Control Platform
Section: conference
Valid values: any valid string
Default value: Empty (no value)
Takes effect: Immediately

Specifies the default recording type for MSML conference recording. Example formats: `audio/wav`, `audio/wav;codec=ulaw`, `audio/mp3`. If empty, the MCP uses the wave file from the default platform codec.

record.amazonpostmode
Application: Media Control Platform
Section: msml
Valid values: `http` or `https`
Default values: `http`
Takes effect: Immediately

Specifies the mode to use for uploading recording files to Amazon s3, during MSML call recording.

- When set to `https`, MCP uses the HTTPS protocol.
- When set to `http`, MCP uses the HTTP protocol (the default).
- If the primary and secondary recording destinations are both configured to use the s3 URI format, then MCP uses the value of this option to specify whether to use HTTP or HTTPS.

record.amazonallowpublicaccess
Application: Media Control Platform
Section: msml
Valid values: `true/false`
Default values: `false`
Takes effect: Immediately

Specifies the access permissions for the recording file that is uploaded to Amazon s3 during MSML call recording.

- When set to `false`, MCP restricts access to the uploaded file to the s3 bucket owner only.
- When set to `true`, MCP allows public download access to the uploaded recording file.
- If both primary recording and secondary recording destinations are configured to use s3 URI format, then MCP grants the access permissions specified by this option to the two recording files uploaded to Amazon s3.

record.basepath
Application: Media Control Platform
Section: msml
Valid values: any valid string
Default value: `file://$installationRoot`
Takes effect: Immediately

Specifes the root directory path for recording media.

record.irrecoverablerecordpostdir
Application: Media Control Platform
Section: msml
Valid values: any valid string
Default value: `$installationRoot$/cache/record/failed`
Takes effect: Immediately

MSML call recordings are added to a list if they need to be posted to Amazon S3, Call Recording API, HTTP or HTTPS, or SpeechMiner. A separate posting thread specifies the list of recordings to be posted periodically.
This option specifies the directory for storing any recording files with irrecoverable errors during post attempts.

record.posttimeout
Application: Media Control Platform
Section: msml
Valid values: Integer > 0 and <= maximum integer (as defined by Genesys Administrator Help)
Default value: `120000` miliseconds (2 minutes)
Takes effect: Immediately

Specifies the post timeout for recordings that must be posted to Amazon S3, Call Recording API, HTTP or HTTPS, or SpeechMiner. When this timeout expires, the attempt to post is considered a recoverable error, and is retried.

record.updateheader
Application: Media Control Platform
Section: msml
Valid values: `true/false`
Default values: `false`
Takes effect: Immediately

- Set to `true` to update the recording file header on disk during MSML call recording.
- Set to `false` to *not* update the recording file header. The header update is performed (if needed) while saving the recording file to its final destination.

record.userecordcachedir
Application: Media Control Platform
Section: msml
Valid values: `true/false`
Default values: `false`
Takes effect: Immediately

- Set to `true` to use the record cache dir that is specified in `mpc.recordcachedir` for file-based MSML call recording. When recording completes, the recording file moves from the record cache directory to the final recording destination.
- Set to `false`, to *not* use the record cache dir for FILE based MSML call recording. The recording file is created directly at the final recording destination.

Note: For an HTCC post, MCP uses the `true` behavior of this option, regardless of the setting.

record.filenametemplate
Application: Media Control Platform
Section: msml
Valid values: string
Default values: `"$id$"`
Takes effect: Immediately

Specifies the default template for generating an MSML recording file name. Details:

- Any `gvp:param` present in the template is replaced with its value if specified using MSML.
- The parameters **AWSAccessKeyId(2**), **callrec_authorization**, **httpauthorization(2)** and **AWSSecretAccessKey(2)** are not replaced by their value, even if specified using MSML, due to security concerns.

> Example: The template `$id$$record$$MCPDateTime` produces the file name
> `basirecid12345_source_2013-09-13_08-10-15_.`
>
> ...where the ID is specified as `basicrecid12345` and record is specified as `source` using MSML gvp:param.

- Use of the $MCPDateTime parameter enables insertion of MCP local time in the generated file name.

Windows has a 260-character limit (including directories and extension) for the recording filename.

record.filenamepostfix
Application: Media Control Platform
Section: msml
Valid values: Any valid string
Default value: Empty
Takes effect: Immediately

Specifies the string to be appended to the file name to make it static. If this string is *not* empty, then it is appended to the file name. Otherwise, the logID is appended.

record.channels
Application: Media Control Platform
Section: msml
Valid values: 1 (mono) or 2 (stereo)
Default values: 2 (stereo)
Takes effect: Immediately

Specifies the number of channels for MSML recording to `dest` (the recording destination).

callrecording.dtmfhandling
Application: Media Control Platform
Section: msml
Valid values:

- `as-is`—Record everything as-is from the RTP stream. Inband DTMFs will be recorded, but RFC2833 digits will not.
- `no-digits`—Strip out all DTMF digits. This includes inband or RFC2833. NOTE: When telephone-event is negotiated on the call, if inband audio DTMFs are received, they will not be removed from the recording.
- `all-digits`—Record all DTMF digits, including inband, and generate audio for RFC2833 digits.

Default values: `as-is`
Takes effect: Immediately

Specifies the recording behavior for DTMFs in MSML Call Recording.

recordcachedir
Application: Media Control Platform
Section: mpc
Valid value: A valid directory path
Default values: `$installationRoot$/cache/record`
Takes effect: At start or restart

Sets the temporary recording cache directory for MSML call recording. When the recording completes, MCP places the recording files at the final recording destination and removes them from the cache directory.

cpd Service Parameters for IVR Profile Provisioning

Use these parameters to configure Call Progress Detection (cpd):

Description	Valid values	Default value	Takes effect
cpd-allowed (CPD Allowed)			
Controls whether CPD service is allowed	`true`: CPD service is allowed `false`: CPD service is not allowed	`true`	immediately
cpd-capability-requirement (CPD Capability Requirement)			
This parameter specifies the required capability for when the cpd service is invoked in the context of an application.	Use this format: `[cap_NameA]=[cap_ValueA1],...,[cap_ValueAm]; [cap_NameB]=[cap_ValueB1],...,[cap_ValueBn]; ; [cap_NameM]=[cap_ValueM1],...,[cap_ValueMl]` The set of `[cap_NameX]` must be unique.	(blank)	immediately
cpd-forbidden-respcode (CPD Forbidden Response Code)			
Expects parameter value of the form `sipcode;description` or just `sipcode`.		403	immediately
cpd-forbidden-set-alarm (Raise Alarm when CPD is Not Allowed)			
If set to `true`, an alarm will be raised for the corresponding policy violation	`true` `false`	`false`	immediately
cpd-level2-burst-limit (CPD level2 Usage Limit)			
CPD application level2 usage limit	The value must be an integer	(blank)	immediately
cpd-level3-burst-limit (CPD level3 Usage Limit)			
CPD application level3 usage limit	The value must be an integer	(blank)	immediately
cpd-usage-limit (CPD Usage Limit)			
CPD application usage limit	The value must be an integer	(blank)	immediately
cpd-usage-limit-exceeded-respcode (CPD Usage Limit Exceeded Response Code)			
Expects parameter value of the form sipcode;desc or just sipcode.		480	immediately
cpd-usage-limit-exceeded-set-alarm (Raise Alarm for CPD Usage Limit Exceeded)			

If set to `true`, an alarm will be raised for the corresponding policy violation	`true` `false`	`false`	immediately
cpd-usage-limit-per-session (CPD Usage Limit per Session)			
CPD application usage limit per call	The value must be an integer	(blank)	immediately

Provision DIDs to DID Group mapping

Now Tenant Administrators can provision DIDs to DID Group mapping. The implementation provides a GUI interface so a GAX user can create, edit, or delete DIDs and their assignments in DID Group mapping. This functionality uses overlap checking across multiple tenants to ensure that there are no duplicate DIDs.

To use the DID Groups feature, Tenant Administrators need the privilege PRIVILEGE_DID_GROUP_SERVICE. They can perform the following functions if they have also have these privileges:

- To access Configuration Manager: ACCESS_CONFIGMANAGER
- Accessing Voice Platform Profiles...
    To read Voice Platform Profiles: READ_VOICEPLATFORMPROFILES

    To modify General options and the state of Voice Platform profiles: MODIFY_VOICEPLATFORMPROFILES

    To modify items in the Options and Annex sections of Voice Platform Profiles: MODIFY_OPTIONS_VOICEPLATFORMPROFILES

    To create or hold Full Control of Voice Platform Profiles: CREATE_VOICEPLATFORMPROFILES

    To delete Voice Platform profiles: DELETE_VOICEPLATFORMPROFILES

Configure this feature with the <GAX application> options below.

Notes:

- All of these options require Read/Create/Update permission to the tenant object.
- Restart required to enact your changes.

```
[gvp-rpt]
did-source-type = "switch"
did-source = [
{name: <name of switch object>}
,
{name: <name of switch object>}
]
```

Notes:

- Declare `did-source-type = "switch"` in the **Application Options** tab of the GAX application object, *not* in the Annex.
- If `did-source-type` is set to `"switch"` and the `did-source` option is not set, then the plug-in will retrieve the DNs of type TrunkGroup from all the switches available to the user in the current tenant.
    - Setting `did-source` optional if you are using TrunkGroups.
- If customers use DNs of type `Trunk`, they should also use `did-source-type="tenant"` and manually provision the DIDs accordingly.

OR

```
[gvp-rpt]
did-source-type = "tenant"
```

Provision DIDs with the <tenant object> options below:

```
[gvp-dids]
111
112
113
... ''(and so on)''
```

Note: If `did-source-type` is not set, it will default to `tenant`.

Resource Manager Logical Resource Group Configuration options

These options (in CTI_LRG CTI and in GW LRGs) configure fallback mechanisms that handle scenarios where CTIC/ICM is unavailable. Logical Resources are configured one of two ways:

- As sections within the Resource Manager (RM). In this case, configuration options are available within these sections.
- Or as a named applications folder under Configuration Unit in a Tenant. In this case, configuration options are available within the `gvp.lrg` section of the named applications folder.

Gateway Resource Group Option

remove-ruri-capability-on-fallback

Section: Gateway Resource group section
Valid Values: `true` or `false` (default)

This parameter is used only for the Gateway resource group. It disables or enables the capability for selecting a VXML resource when falling back to VXML after CTIC returns a 404 error. This capability is specified in an INVITE Request URI (`gvp.rm.resource-req`).

- Set to `true` to disable Resource Manager's use of the `gvp.rm.resource-req` option for VXML fallback after a CTIC 404 error.
- Set to `false` to enable Resource Manager's use of `gvp.rm.resource-req`.

Note: This option is NOT available during configuration of a Gateway Resource Group via Genesys Administrator. Specify this parameter manually in the Gateway Logical Resource Group (GW LRG) section, using the format `remove-ruri-capability-on-fallback = true` (or `= false`).

CTI Connector Failover Options

Resource Manager (RM) has three new options in the CTI Connector (CTIC) Logical Resource Group (LRG) for handling CTIC failover.

Important: These options are *not* available during configuration of a CTIC Resource Group via Genesys Administrator. You must specify them manually in the CTIC LRG.

fail-over-cti-handling

Valid Values: `reject` (default), `answer`, `script;<service-type>;<URL>`

Takes Effect: After restart

This option specifies RM behavior when all attempts to use CTIC fail. For example: all CTICs are down, or port capacity of the CTIC LRG is exceeded, or all CTICs in the LRG were tried but failed.

- Set to `reject` to reject the call.
- Set to `answer` to answer the call.
- Set to `script;<service-type>;<URL>` to specify that RM redirects the call to the service `<service-type>` and informs that service to run the page at the URL (same behavior as `rm.cti-unavailable-action`).

cti-unavailable-respcode

Valid Values: No value specified (default), `none`, SIP response codes for which next CTI resource should *not* be retried.
Takes Effect: After restart

- Specifies a list of response codes to be intercepted and given special treatment. Separate each code in the list with a semicolon (;).
  If CTIC returns a response code matching a code provided in the list, RM does not retry any other CTIC; instead RM takes action based on the group-level option `cti-unavailable-action`, or based on the server-level option `rm.cti-unavailable-action`.
- When set to `empty` or `none`, RM retries the next CTIC available in the CTI LRG in response to any error from CTIC.
- When no value is specified, RM checks the server-level parameter `rm.cti-unavailable-respcode` and takes the action specified there.

Note: `cti-unavailable-respcode` overrides the server parameter `rm.cti-unavailable-respcode`.

cti-unavailable-action

Valid Values: `reject` (default), `answer`, `script;<service-type>;<URL>`
Takes Effect: After restart

Specifies the behavior expected when the SIP response code received from CTIC matches a response code that is configured in `rm.cti-unavailable-respcode`.

- Set to `reject` to reject the call.
- Set to `answer` to answer the call.
- Set to `script;<service-type>;<URL>` to specify that RM redirects the call to the service `<service-type>` and informs that service to run the page at the URL (same behavior as `rm.cti-unavailble-action`).

When no value is specified, RM uses the server-level parameter `rm.cti-unavailable-action`.

Note: This option overrides `rm.cti-unavailable-action`.

Managing DID Groups with GAX

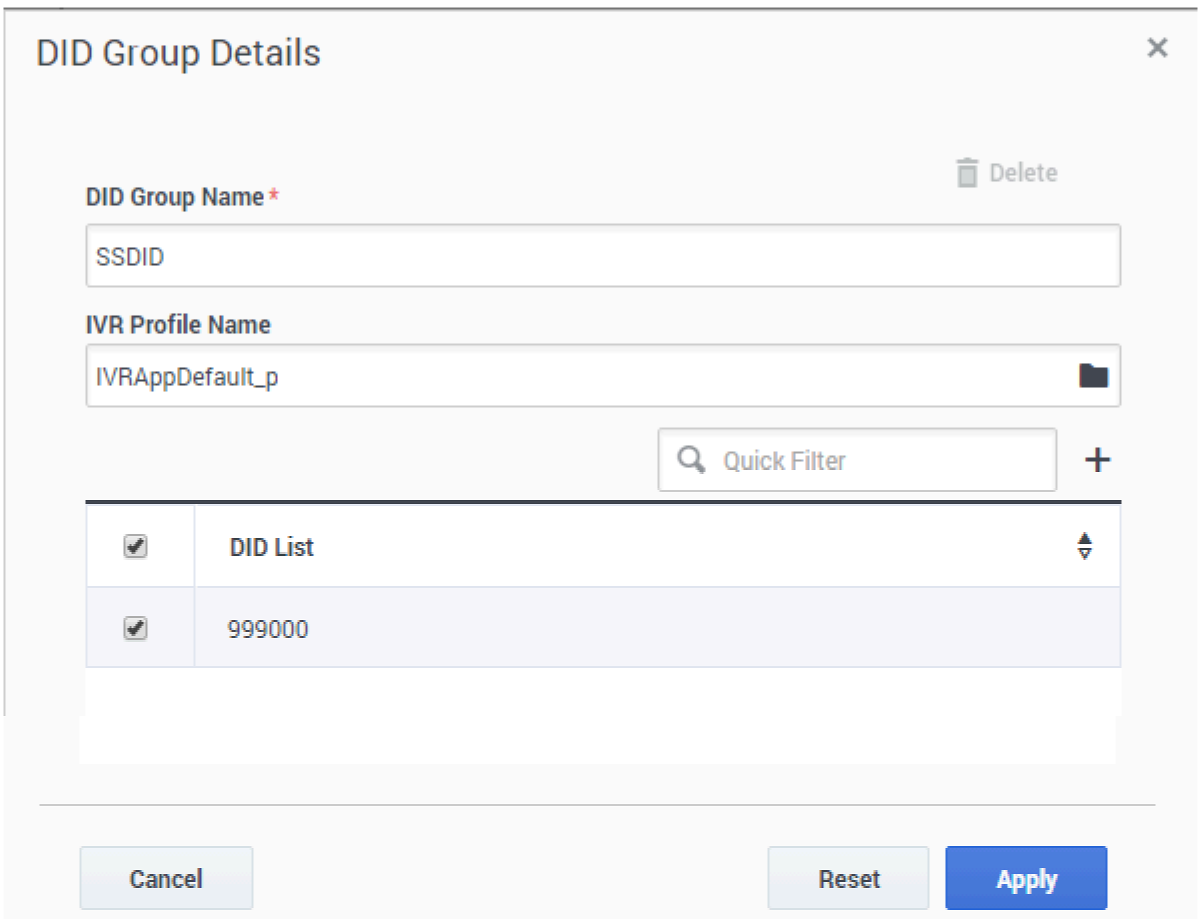
All procedures begin in the **DID Groups** main page. To get there:

1. Log into **Genesys Administrator Extension** (GAX) as a tenant user.
2. Select **DID Groups** from **Configuration > Voice Platform Tile** menu.

At the end of each procedure, click **Apply** to save changes.

You can sort the list view of DID Groups by using the arrowhead symbol in the DID Groups list view header in the top right corner.

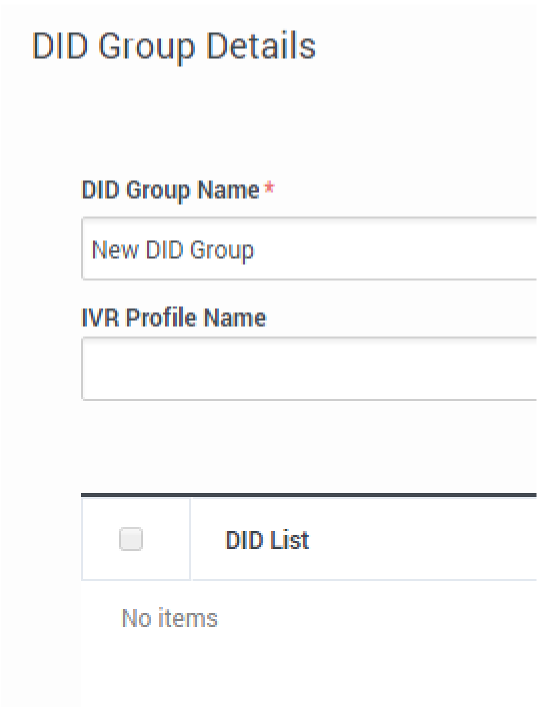
Creating a New DID Group



1. Click **New**.

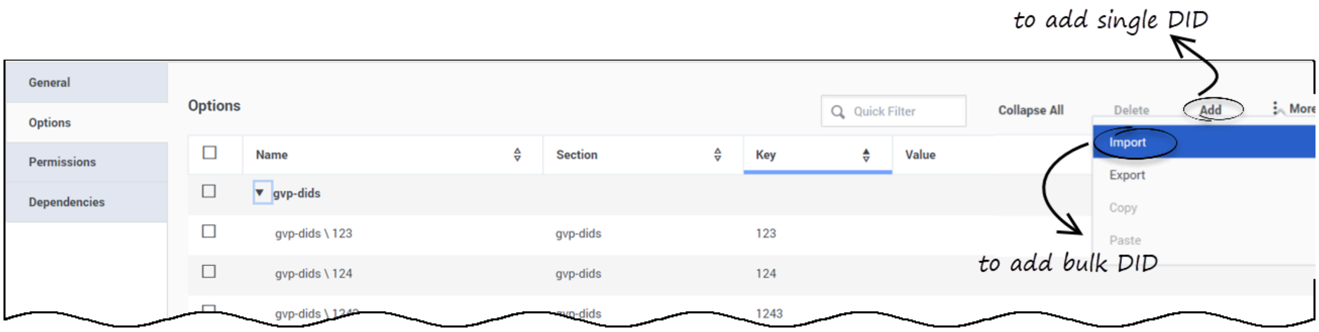
2. Open the **DID Group Details** dialog. Specify the following details to create a new DID Group:
   - **DID Group Name** - Mandatory to specify a unique name of the DID Group to be created.
   - **IVR Profile Name** - Optional to specify an existing IVR Profile Name for the DID Group.
3. Use the '+' icon to add a single DID from the available list. Optionally you can use the quick filter to filter the DIDs in the list to include or exclude by selecting and clearing the box to associate with the newly created DID Group.
   - You can also associate multiple or bulk DIDs to a DID Group. For more details on adding a single or multiple DIDs to a group, refer to Adding DIDs to a DID group. Refer to Add DIDs to add either a single or bulk DIDs in GAX.
4. Click **Apply** to save the DID Group.

Modifying an Existing DID Group



1. Double click the DID Group to edit.
2. Make the necessary changes to the DID Group.
3. Click **Apply** to save the changes.

Adding DIDs



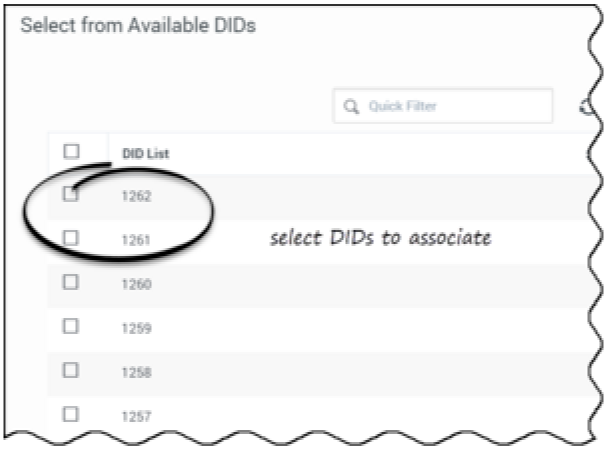
1. In GAX, navigate to **Environment > Tenants > Option**.
2. In the **gvp-dids** section, add the required DIDs.
    - You can add a single DID or a range (for example, 1000-1025), by clicking the **Add** button.
    - Or, add multiple or bulk DIDs by using the **More** > **Import** feature by importing the bulk DIDs from a **.CSV** file.

      Note: The CSV file is formatted to contain the columns: **Section, Key, and Value**. You can also define a DID range in the Key column in the CSV file.

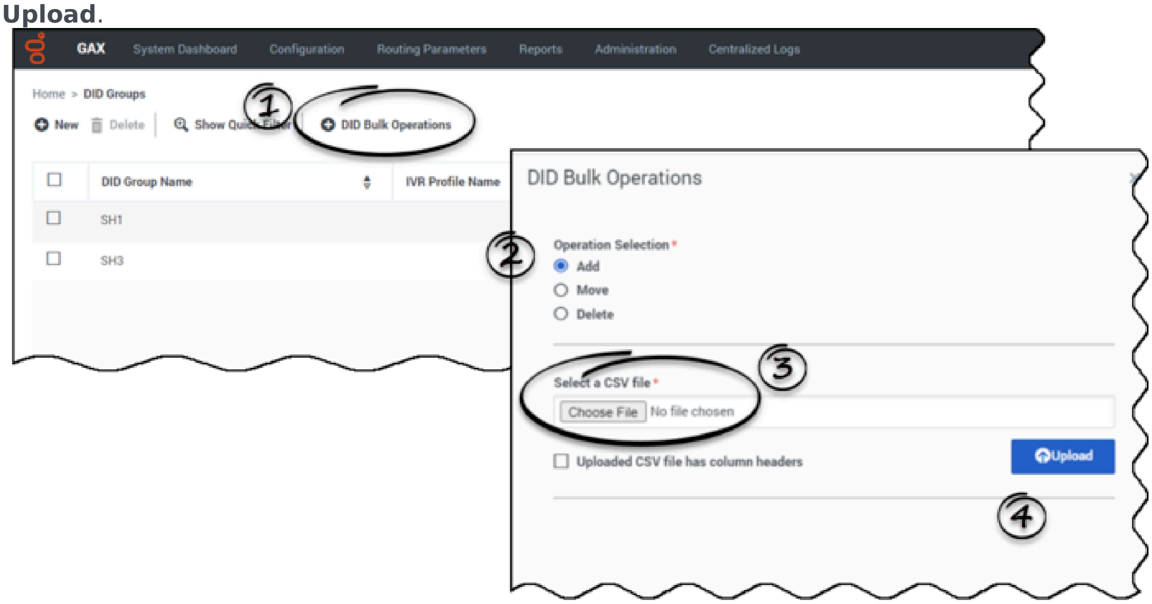
Adding DIDs to a DID group

After adding DIDs in GAX, you can associate them to DID Groups.

1. In GAX, navigate to **Configuration > Voice Platform Tile > DID Groups**.
2. In the **DID Groups** page, double click the **DID Group** record to display the **DID Group Details** page.
3. If you want to associate a single DID to the group,
    1. Use the '+' icon to display the **Select from Available DIDs** dialog.
    2. Select the required DIDs from the list and click **Apply**. Note that this list doesn't display the DID range, you must use the **DID Bulk Operations** feature to associate a DID range.



4. If you want to associate multiple or bulk DIDs to a DID Group,
    1. Click the **DID Bulk Operations** feature from the **DID Groups** page.
    2. In the **DID Bulk Operations** dialog, select the type of operation you want to perform on the DIDs. You can select one of the following:
        - **Add** - to associate the DIDs to the DID Group.
        - **Move** - to move the given DIDs from one DID Group to another.
        - **Delete** - to delete the DIDs association from the DID Group.
    3. **Select a CSV file** - select the **.CSV** file that contains the bulk DID details to be associated with the DID Group.
       Note: The .CSV file is formatted to contain the columns: DID, DID Group, Tenant, and IVR Profile Name.
    4. Click **Upload**.



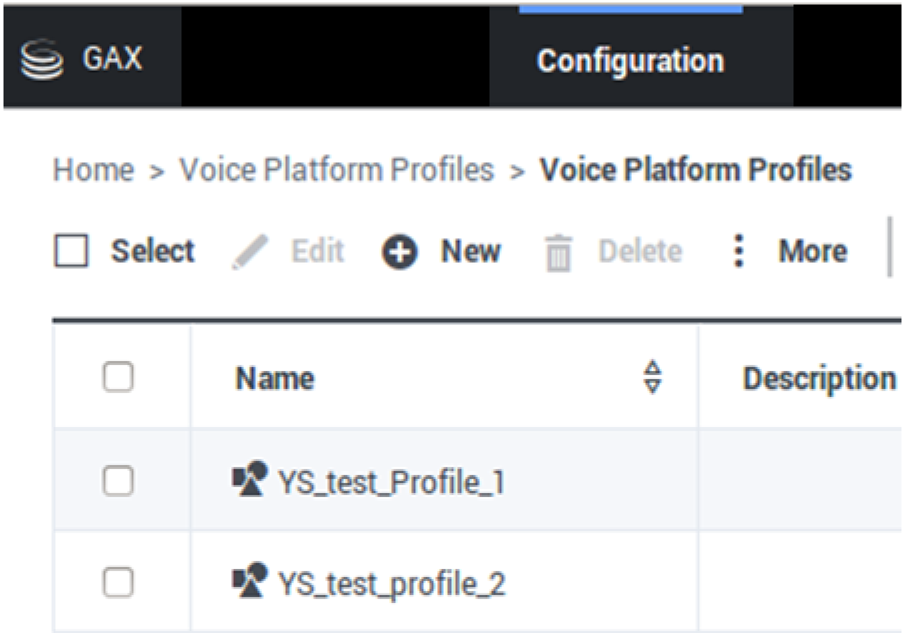
Important

You can associate a DID range (for example, 1000-1025) to a DID Group by defining the range value in the CSV file. However, the DID range is not displayed in the **Select from Available DIDs** dialog for direct selection.

Provision IVR Profiles Yourself

Now you can use GAX to provision IVR profiles and Map Direct Inward Dialing (DID) Groups to IVR profiles. You don't have to ask a manager or a system administrator.

How to Provision IVR Profiles



Before You Begin

> **Important**
> All “Voice Platform Profile Edit” permissions are provided in the XML file in the VP Reporting Plugin GAX installation package. You must import the package in order to see the permissions shown on this page.

- Acquire the appropriate permissions:
  - GVP_CFG_IVR_PROFILE_ALL (Voice Platform Profile Edit All Access)—If you need to see all tabs on the IVR Profile edit page, including Options and Permissions.
  - GVP_CFG_IVR_PROFILE_UPDATE (Voice Platform Profile Edit)—If you need to update initial and alternate page URLs on the Voice Platform tab in existing IVR profiles. You will be able to see DID Group assignments, but unable to modify them.

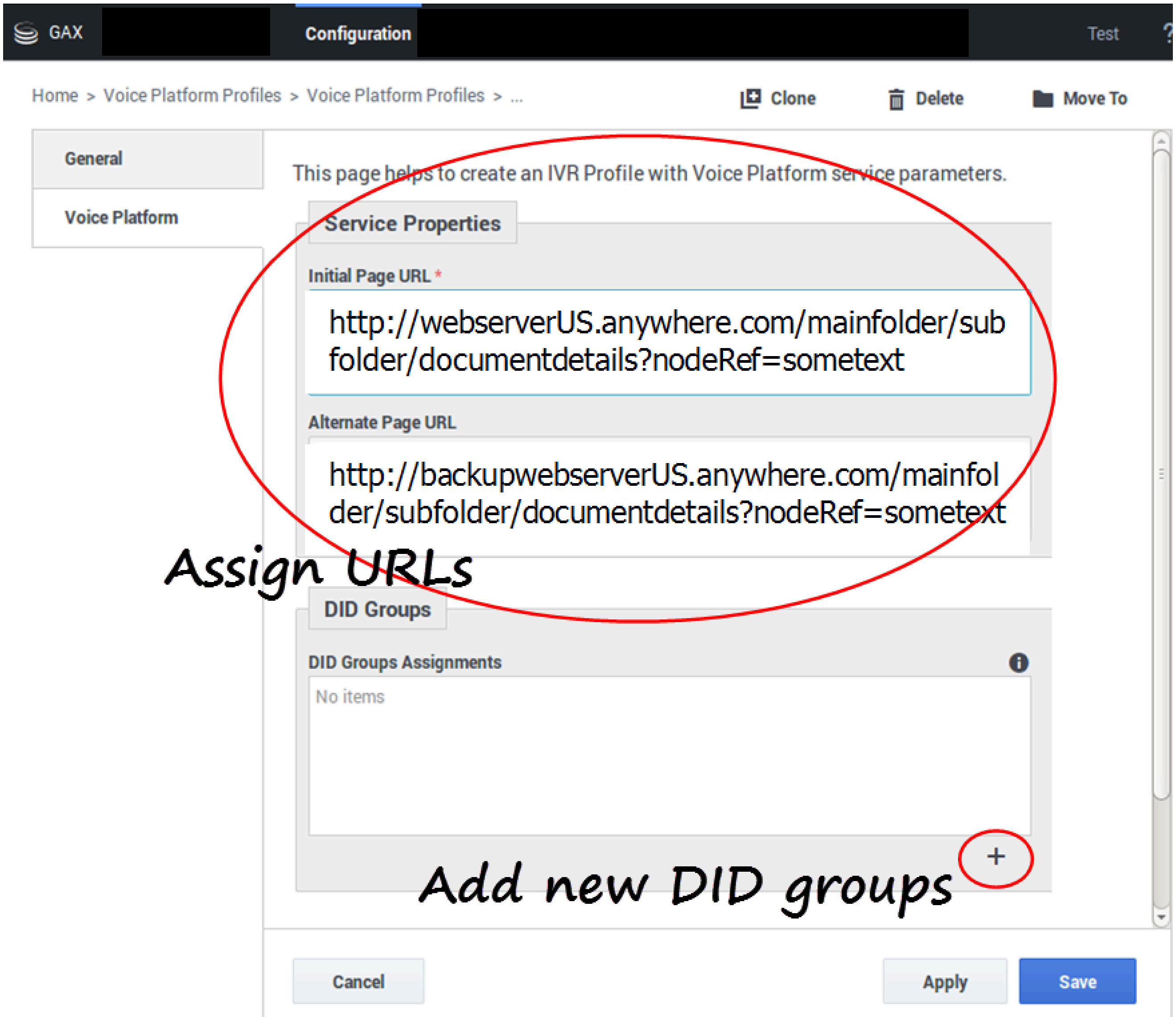
    - GVP_IVR_PROFILE_DID_GROUP_UPDATE (Voice Platform Profile Edit DID Group assignments)—If you need only to update DID Group assignments in existing IVR Profiles. You will be able to see initial and alternate page URL assignments but unable to modify them.

- Install GAX 8.5.220.20.

Log in as a tenant and select the Voice Platform Profile icon on the Main Menu page.

- The Voice Platform Profiles Main Page lists all existing IVR profiles that belong to you (the logged-in tenant).
- The Edit, New, Delete, and More commands (above the list) act on the IVR profile(s) that you select.

How to Create New IVR Profiles



Creating a new IVR profile requires completing two forms.
Click **New** and be ready to enter information that the forms need:

On the Service Properties Page

- Initial page URL (only http, https and file protocols supported)
- Alternate Page URL (only http, https and file protocols supported)
- DID Groups
  - Add to the list—click the **+** button (plus) near the bottom right. An Available DID Groups dialog will pop up.

    - Remove from the list—roll over the item with the cursor and click the **X**.
- **Save** also moves you to the IVR profile listing page.

On the Available DID Groups Page Popup Dialog

- Select the DID groups to assign and click **OK**. Your selections appear in the DID Groups Assignments list.
- **Save** also returns you to the VP Profiles Main Page, which now includes your newly created / newly assigned DID groups in the list.

Managing DTMF Clamping in MSML Recordings

DTMF Clamping (also known as Masking) guards a customer's sensitive credit card information from an agent's ears, and from call recording, by inserting silence into the agent and recording legs of the call while the customer is entering credit card numbers.

The credit card verifier hears the card numbers as the customer enters them, but the agent and the call recording hear silence.

An MSML conference request enables clamping, and three new configuration parameters support the new MSML element <clamp> under the previously supported <stream> element. You can read more here about configuring and managing DTMF clamping and its behavior in a conference call.

Define DTMF Handling of Call Recording

Use the option `[msml] callrecording.dtmfhandling` to specify the recording behavior for DTMF clamping during MSML Call Recording and GIR Recording.

Valid values:

-

- as-is (the default): Record everything as-is from the RTP stream.
  Inband DTMFs are recorded, but not RFC 2833 digits.
- no-digits: Strip out all DTMF digits, including inband and RFC 2833 digits.
  However, when a telephone-event is negotiated on the call, inband audio DTMFs are not removed from the recording.
- all-digits: Record all DTMF digits, including inband, and generate audio for RFC 2833 digits.

Any change takes effect immediately.

Set the Number of Silence Packets before Clamped DTMFs

Use the option `[msml] clampdtmf.presilencepackets` to specify the number of audio packets to be replaced with silence before clamped DTMFs.

For example, use when the DTMF tone appears before a DTMF RFC 2833 event, which may happen when a SIP gateway converts DTMF tones to a DTMF RFC 2833 event.

Valid values:

- An integer `1–50`, or `0` (the default).
  Be cautious: increasing the value introduces more audio delays into a conference.
  One packet usually = 20 milliseconds, but that can vary with traffic and other factors.

Any change takes effect immediately.

Set the Number of Silence Packets after Clamped DTMFs

Use the option `[msml] clampdtmf.postsilencepackets` to specify the number of audio packets to be replaced with silence after clamped DTMFs.

For example, use when a DTMF tone appears after a DTMF RFC2833 event.

Valid values:

- A positive integer, or `0` (the default).
  One packet usually = 20 milliseconds, but that can vary with traffic and other factors.

Any change takes effect immediately.

Only One SNMP Master Agent per GVP Component

SUMMARY: A note requiring one SNMP Master Agent per Media Server instance should extend to all GVP components.

The next publication of the Media Server Deployment Guide will include this revision:

CHAPTER: Chapter 4: Deploying Genesys Media Server
SECTION: Provisioning Media Server
REPLACE THE NOTE on page 114 with this note:

> **Note:** A single SNMP Master Agent can serve a single component only. Therefore, you must have an SNMP Master Agent installed and a connection configured for every instance of every component. Think of “One Media Control Platform (MCP) connected to one SNMP Master Agent”, “One Resource Manager connected to one SNMP Master Agent”, “One Reporting Server connected to one SNMP Master Agent”, and so on.

Table 1 in the Media Server Deployment Guide

Horizontal version

Table 1: DNs Supported by GVP and Media Server

DN Type	Solution applies to:	Protocol	Usage	Can be recorded by SIP Server?	Generates TEvents?	Allows ICON reporting
VoIP Service	SIP Server, GQM, HPE	SIP/MSML	Media services including music-on-hold, conferencing, call parking (treatments), call recording	No	No	No
Voice Treatment Port (VTP)	GVP	SIP	Legacy IVR ports for both inbound and outbound IVR calls	Yes	Yes	Yes
Trunk Group DN	GVP, OCS, HPE	SIP/MSML	CPD for outbound solution: ASM mode, transfer mode, and proactive notification Outbound GVP IVR calls (SSG) HPE also uses this DN type for inbound GVP IVR calls	Yes	Yes	Yes
Trunk DN	GVP	SIP	Inbound GVP IVR calls	No	No	No

File-based Call Recording

SUMMARY: Add missing word to text.
The next publication of the Media Server Deployment Guide will include this revision:
CHAPTER: Chapter 3: Media Server Functions
SECTION: File-based Call Recording
EDIT Add the word "local" before "storage location" so the following line reads:
When you specify the destination using file://, the recording is placed in a local storage location.

Remove references to Remdial

SUMMARY: Remdial is not supported. Remove all references to Remdial.
The next publication of the Media Server Deployment Guide, GVP 8.5 User's Guide, Genesys Voice Platform 8.1 User's Guide will include this revision:

Genesys Media Server Deployment Guide

CHAPTER: Chapter 4: Deploying Genesys Media Server
SECTION: Provisioning Media Server
EDIT In the procedure titled "Start of Procedure" under the sub section **Procedure: Integrating Media Server with the Resource Manager**, remove step 5 "If `Remdial` is used, select sip to find the `routeset` option". Renumber the steps accordingly.

Genesys 8.5 User's Guide

SUMMARY: Remdial is not supported. Remove all references to Remdial.
CHAPTER: Chapter 3: Configuring Common Features
SECTION: Configuring Session Timers and Timeouts
EDIT On page 79, in the sub section **Additional Timeouts**, delete reference to remdial.

Change:

This timer should be greater than the connect timeout of the outbound call (depending on how the outbound call is initiated, the connect timeout can be specified in the transfer tag, or in the remdial command).

To:

This timer should be greater than the connect timeout of the outbound call (depending on how the outbound call is initiated, the connect timeout can be specified in the transfer tag).

CHAPTER: Chapter 7: Configuring the Media Control Platform
SECTION: Important Media Control Platform Configuration Options
EDIT On page 154, remove the bullet following bullet:

• remdial—Parameters determine remote dialer behavior.

CHAPTER: Chapter 7: Configuring the Media Control Platform
SECTION: Important Media Control Platform Configuration Options
EDIT On page 171, remdial section from Table 23: Selected Media Control Platform Configuration Options. A half-page contains the remdial Section--four rows of Table 23: Selected Media Control Platform Configuration Options, describing Remdial Port, Remdial Max Calls, Remdial Max Client, and Remdial Telnet.

CHAPTER: Appendix A: Module and Specifier IDs

SECTION: Media Control Platform
EDIT

- On page 388, remove the REMDIAL row from Table 70: Media Control Platform Application Module Names and IDs.
- On page 394, remove the REMDIAL section from Table 71: Media Control Platform Specifier Names and IDs.

CHAPTER: Appendix K: SIP Customizable Headers and Parameters
SECTION: Session Variables for VXML
EDIT

- On page 562, remove the highlighted text:
  This feature is supported for transfers and calls initiated using RemDial, and can be enabled by configuring `sip.out.invite.headers`, `sip.out.invite.params`, `sip.out.refer.headers` and `sip.out.refer.params`.
- On page 563, remove the highlighted text:
  - Below is an example Media Control Platform configuration for customizing request URI's paramA, request URI's paramB and HeaderC in outgoing INVITE messages (for <transfer> involving two call legs and remdial calls):
  - If the following signaling variables are defined (or the equivalent name/value list is defined and appended to the remdial call request):

CHAPTER: Index
EDIT On pages 576, 578, and 584, remove the multiple index entries the refer to remdial.

Genesys 8.1 User's Guide

SUMMARY: Remdial is not supported. Remove all references to Remdial.
CHAPTER: Chapter 3: Configuring Common Features
SECTION: Configuring Session Timers and Timeouts
EDIT On page 83, in the sub section **Additional Timeouts**, delete reference to remdial.

Change:

This timer should be greater than the connect timeout of the outbound call (depending on how the outbound call is initiated, the connect timeout can be specified in the transfer tag, or in the remdial command).

To:

This timer should be greater than the connect timeout of the outbound call (depending on how the outbound call is initiated, the connect timeout can be specified in the transfer tag).

CHAPTER: Chapter 7: Configuring the Media Control Platform
SECTION: Important Media Control Platform Configuration Options
EDIT On page 158, remove the bullet following bullet:

• remdial—Parameters determine remote dialer behavior.

CHAPTER: Chapter 7: Configuring the Media Control Platform
SECTION: Important Media Control Platform Configuration Options
EDIT On page 175, remdial section from Table 23: Selected Media Control Platform Configuration Options. A half-page contains the remdial Section--four rows of Table 23: Selected Media Control Platform Configuration Options, describing Remdial Port, Remdial Max Calls, Remdial Max Client, and Remdial Telnet.

CHAPTER: Appendix A: Module and Specifier IDs
SECTION: Media Control Platform
EDIT

- On page 392, remove the REMDIAL row from Table 70: Media Control Platform Application Module Names and IDs.
- On page 398, remove the REMDIAL section from Table 71: Media Control Platform Specifier Names and IDs.

CHAPTER: Appendix K: SIP Customizable Headers and Parameters
SECTION: Session Variables for VXML
EDIT

- On page 566, remove the highlighted text:
  This feature is supported for transfers and calls initiated using RemDial, and can be enabled by configuring `sip.out.invite.headers`, `sip.out.invite.params`, `sip.out.refer.headers` and `sip.out.refer.params`.
- On page 567, remove the highlighted text:
  - Below is an example Media Control Platform configuration for customizing request URI’s paramA, request URI’s paramB and HeaderC in outgoing INVITE messages (for <transfer> involving two call legs and remdial calls):
  - If the following signaling variables are defined (or the equivalent name/value list is defined and appended to the remdial call request):

CHAPTER: Index
EDIT On pages 580, 582, and 588, remove the multiple index entries the refer to remdial.

Stereo .wav files are NOT supported by MCP

SUMMARY: Stereo .wav files are NOT supported by MCP. This must be documented in the GVP User Guide.
The next publication of the GVP 8.5 User's Guide will include this revision:
CHAPTER: Appendix B Media Control Platform Reference Information
SECTION: Audio and Video File Formats

Add the following Note after Table 89 "Supported Audio File Formats – Play":

Note: Only mono (single-channel) is supported for container/file types .au, .avi, .nist, and .wav – i.e., stereo is not supported.

MSML Dialog Base Package

SUMMARY: Update the MSML Dialog Base Package section.

The next publication of the Media Server Deployment Guide will include this revision:

APPENDIX: Appendix B MSML Specification
SECTION: MSML Dialog Base Package

Make the two following changes:

- One page 167, update "uri" with this information - "Valid schemes are file, http, rtsp and dtmf".
  Child Elements:
  `<audio>`
  Attributes:
  ```
  uri: Valid schemes are file, http, rtsp and dtmf.
      format
          iterate
  ```

- On page 168, update "uri" with this information - "Valid schemes are file, http and rtsp".
  `<video>`
  Attributes:
  ```
  uri: Valid schemes are file, http and rtsp.
      format

      iterate
  ```

Remove Note about GVPi Legacy Functionality

SUMMARY: Remove a note about Legacy GVPi functionality that is no longer supported.

The next publication of the Genesys Media Server 8.5 Deployment Guide will include this revision:
CHAPTER: Chapter 3: Media Server Functions
SECTION: Media Interfaces
REMOVE THIS NOTE on page 47:

~~**Note:** GVPi was not included in the GVP 8.1.5 installation package, but is still supported when deployed with Media Server 8.1.4.~~

Improved RURI Capability for VXML

SUMMARY: Add a note about a new parameter to the GVP User Guide.

DOCUMENT: The next publication of the GVP 8.5 User's Guide will include these revisions.

CHAPTER: Chapter 4, Chapter 4: Configuring the Resource Manager.

SECTION: Configuring Logical Resource Groups

Add this note to *Table 14: Logical Group Section Configuration Options*, on page 93:

You can now direct Resource Manager to *not* use the capability specified in the `gvp.rm.resource-req param` of an INVITE Request URI for selecting a VXML resource when falling back to VXML after CTIC returns a 404 error. Use a new configuration option in the Gateway LRG section in order to achieve this functionality. **remove-ruri-capability-on-fallback**
Section: Gateway Resource group section
Valid Values: `true` or `false` (default)
Set to `true` to disable Resource Manager's use of the `gvp.rm.resource-req` option for VXML fallback after a CTIC 404 error.
Set to `false`, to enable Resource Manager's use of `gvp.rm.resource-req`.
Note: This option is NOT available during configuration of a Gateway Resource Group via Genesys Administrator. Specify this parameter manually in the Gateway Logical Resource Group (GW LRG) section, using the format `remove-ruri-capability-on-fallback = true` (or `= false`).

Document CPA behavior

SUMMARY: Add CPA behavior information to the user guide.

DOCUMENT: The next publication of the GVP 8.5 User's Guide will include these revisions.

CHAPTER: Appendix C: Tuning Call Progress Detection

SECTION: *Detecting a Human, Not a Machine*

Add a new section title "Detecting a Human, Not a Machine", and add the following information to the section:

The Media Control Platform detects a human when all three of these conditions are true:

- There is at least one signal.
- One (1) second has passed since the start of the first signal (hardcoded and used in CPA when `postconnectpref="no_machine"`).
- The fax duration has passed since the start of the first signal.

Remove Note from page 42

SUMMARY: Remove a note from page 42 of the GVP User Guide.

DOCUMENT: The next publication of the GVP 8.5 User's Guide will include these revisions.

CHAPTER: Chapter 3: Configuring Common Features

SECTION: *Enabling Secure Communication*

Remove this note:

Note: Although, the GVP components support SIPS, the Genesys SIP Server does not. Before you enable SIPS in your GVP deployment, contact your Genesys Sales Representative for more information.

Correct Name for transaction object ID

SUMMARY: `Transaction_dbid` is improperly named in the GVP User Guide.
The next publication of the GVP 8.5 User's Guide will include this revision:
CHAPTER: Chapter 6: Provisioning IVR Profiles
SECTION: IVR Profile Configuration Options
EDIT the final row of **Table 16: IVR Profile Configuration Option** (pages 109-123) as follows:

Option Name	Description Valid	Values and Syntax
	OPM	
~~transaction object ID~~ transaction_dbid	Specifies the transaction or list object DBID that will be referenced during runtime of this profile.	Any string of integers (except 0 [zero]). Default value: Empty

Multiple Parameter Support in Operational Parameter Management

The next publication of the GVP 8.5 User's Guide will include these revisions:
CHAPTER: Chapter 6: Provisioning IVR Profiles
SECTION: Operational Parameter Management and Self-Service Applications, on page 123

EDIT the third paragraph in this section as follows:
Once configured in GAX, an OPM set can be assigned to a routing strategy or configured within a GVP IVR Profile, or even both. GVP supports ~~up to two parameter sets~~ multiple parameters when using OPM in GAX, but only a single parameter group per IVR Profile (application) and is configured as a setting in the IVR Profile. When the VoiceXML application executes, the GVP interpreter fetches the named parameter set for use as operational variables within the application.

ADD this note below the third paragraph:

> **Tip**
>
> Look for the `dbid` for the list object in GAX, not in GA. To make this value appear in GAX, modify your User Preferences, under the Configuration Manager category, to `Show DBID`.

Audio MP3 Support

SUMMARY: A table in the GVP 8.5 User's Guide does not include support for the MP3 format.

The next publication of the GVP 8.5 User's Guide will include this revision:

CHAPTER: Appendix B: Media Control Platform Reference Information
SECTION: Combined Audio and Video Formats—Play
ADD A ROW to *Table 92: Supported Audio File Formats—Record* on pages 439-441:

MIME type	Recorded file format	Sample size	Encoding	File extension
. . .				
`audio/mp3` `audio/mpeg`	Audio MP3	**sample rates**: 8, 11.025, 12, 16, 22.05, 24, 32, 44.1, and 48 KHz **bit rates**: 16, 24, 32, 48, 64, 96, 128, 160, 192, 256, and 320 kb/s For mono recording **8kb/s** bitrate is also supported. **Default** bit rate is 16kb/s and sample rate is 16KHz.	**MPEG 1 – Layer 3**: For sampling rates of 32 KHz or higher. **MPEG 2 – Layer 3**: For sampling rates of 24 KHz or lower.	.mp3

Missing Audio File Solution

Return to Corrections Table of Contents

SUMMARY: Now you can compensate for a missing default audio file for an announcement, to avoid a silent announcement. (GVP-21227)

The next publication of the Genesys Media Server 8.5 Deployment Guide will include this revision:

CHAPTER: Chapter 3, Media Server Functions
SECTION: Media File Archives
ADD THESE PARAGRAPHS:

If a codec is negotiated but the codec-specific audio file is missing, Media Server must make a substitution, and it looks for a default file in the location specified by the request URI.

You must specify the filename extension (and thus, its media type). Thus, you have two tasks:

1. Place a default audio file in the location specified by the request URI, with the appropriate extension for the media type (.au or .wav).
2. Specify the extension by using this option:
   Option: `annc.defaultaudioext`
   Application: Media Control Platform
   Section: `netann`
   Valid Values: `.wav` (default) or `.au`

Correction to SSL Version Values

Return to Table of Contents

SUMMARY: Corrections to the valid values listed for the option `SSL Version`.

The next publication of the GVP 8.5 User's Guide will include this revision:

CHAPTERS: 7 and 9
SECTION: Two tables in the GVP 8.5 User's Guide require the same corrections: "Table: Selected Media Control Platform Configuration Options" on page 160 in chapter 7, and "Table: Call Control Platform Configuration Options" on page 217 in chapter 9.

Option Name	Description	Valid Values and Syntax
. . .		
SSL Version	Specifies the Secure Socket Layer version to use.	• 0—Automatically detect version • 1—Force TLSv1 • 2—Force ~~TLSv2~~ SSLv2 • 3—Force ~~TLSv3~~ SSLv3 Default value: 0

CCP Metric 1013 not logged or sent to Reporting Server

SUMMARY: Explain why the CCP Metric (event 1013) is not logged nor sent to Reporting Server.

The next publication of the GVP 8.5 User's Guide will contain this correction:
CHAPTER: Chapter 3: Configuring Common Features
SECTION: Configuring Reporting
ADD THIS NOTE to the end of Table 6: Default Log and Metrics Filters on page 60-61:

Note concerning the CCP metric 1013: In general, all events must associate with a session ID. But this CCP metric 1013 occurs before a session ID is created, and at that moment the session ID is NULL. Generating an event report fails when any session ID value is NULL; for that reason, CCP metric 1013 is not logged and is not sent to Reporting Server.

Limits to SRTP implementation

Return to Corrections Table of Contents

SUMMARY: GVP implementation of SRTP has some limitations not fully documented. (GVP-21356)

The next publication of the GVP 8.5 User's Guide will include this revision:

CHAPTER: Appendix I: Specifications and Standards
SECTION: Related Standards
ADD THIS NOTE below the line "RFC 3711 The Secure Real-Time Protocol (SRTP)":
Notes:

- The Media Server (MS) does not support some options in the SRTP specification, including: the Master Key Index (MKI), key derivation rates other than zero, the cipher F8, anti-replay lists with sizes other than 128, the use of the packet index to select between master keys.
- MS does not support the FEC_ORDER and FEC_KEY session parameters.
  ...where FEC stands for Forward Error Correction. Refer to Section 6 of RFC 4568 for details.
- MS ignores the WSH (Window Size Hint) parameter and assumes a value of 128.
- MS supports only the default master key lifetime.
- MS rejects the media line containing the `crypto:` attribute if the RTP profile that was signaled is not RTP/SAVP.
- MS rejects the media line if the keying method is not `inline:` (the only keying method that is specified in RFC 4568).

Documentation equalization about Reporting Server Options

SUMMARY: Documenting a set of RS options (hidden parameters) that were used to fix an issue in customer's environment.

The next publication of the GVP 8.1 Troubleshooting Guide will include this revision.

CHAPTER: Appendix B: Frequently Asked Questions
SECTION: Troubleshooting options
CHANGE TO MAKE on page 51:

Add a new section "Troubleshooting options" after the section "GVP Dashboard", and include the following information on RS options:

Sometimes in some specific cases, engineering might request some "hidden" options adjustments to troubleshoot and/or fine tune a customer's environment. Those options are not described in the options reference guide due to a safeguard measure. A few of them are described here, for an informational purpose.

> **Warning**
> Never change these options unless specifically instructed by Genesys Customer Care or Genesys Engineering.

Version: 8.5.130.61
Section: persistence

hibernate.connection.isolation
Default Value: 2
Valid Values: 1, 2, 4, 8
Changes Take Effect: at start/restart

This parameter can be used to configure the JDBC transaction isolation level for MS SQL Server. Valid integer values are 1 (READ UNCOMMITTED), 2 (READ COMMITTED), 4 (REPEATABLE READ), and 8 (SERIALIZABLE).

hibernate.remote.database
Default Value:
Valid Values: Database Name
Changes Take Effect: at start/restart

The name of remote database that will be used for RS. Used to help construct the JDBC connection URL.

A minor change is required when Reporting Server is deployed with an Oracle RAC database. When *the hibernate.remote.database configuration option is used, the Reporting Server internally appends some parameters to the value of the hibernate.remote.url option, including the value of the* hibernate.remote.database option. Therefore, ensure the hibernate.remote.url option is properly configured for use with Oracle RAC by configuring the hibernate.remote.database option with the value blank.

To support SCAN addresses used in Oracle, set these Reporting Server options:

[persistence] hibernate.remote.database <make it empty>

[persistence] hibernate.remote.url = jdbc:oracle:thin:@<SCAN address>:<port>/<service name>

The first of these means to set the hibernate.remote.data option to blank, [mentioned already in the config file] where as for the second option you should replace the <SCAN address> with the FQDN of the scan address you've set up for Oracle RAC, <port> is the port number to access Oracle, and <service name> is the name of the Oracle service that has been set up to be used by the Reporting Server.

For example: jdbc:oracle:thin:@dbgenesys-scan.pdc.hcnet.vn:1521/gvreport.pdc.hcnet.vn [Not related to your configuration]

Then try to start Reporting Server.

hibernate.remote.dialect
Default Value:
Valid Values: org.hibernate.dialect.SQLServerDialect, org.hibernate.dialect.Oracle10gDialect
Changes Take Effect: at start/restart

The dialect Hibernate should use when interacting with the database.

hibernate.remote.driver
Default Value:
Valid Values: com.microsoft.sqlserver.jdbc.SQLServerDriver, oracle.jdbc.driver.OracleDriver
Changes Take Effect: at start/restart

SQL Driver Hibernate should use when interacting with the database.

hibernate.remote.url
Default Value:
Valid Values: JDBC URL
Changes Take Effect: at start/restart

The JDBC URL that RS should use to connect with the database. For Oracle, the final URL is constructed by appending a colon plus the hibernate.remote.database string to this option's value.

For SQL Server, the final URL is constructed by appending ';databasename=' plus the hibernate.remote.database string to this option's value. The JDBC connection URL will equal the hibernate.remote.url string if the hibernate.remote.database parameter is set to empty.

See also hibernate.remote.database.

hibernate.remote.user
Default Value:
Valid Values: User Name
Changes Take Effect: at start/restart

The user name that RS should use to connect the the remote database.

hibernate.show_sql
Default Value: false
Valid Values: false, true
Changes Take Effect: at start/restart

Enables output of the SQL statement generated by Hibernate to the console.

password
Default Value:
Valid Values: User Password
Changes Take Effect: at start/restart

The password that RS should use to connect the the remote database.

rs.histonly.enabled
Default Value: false
Valid Values: false, true
Changes Take Effect: at start/restart

Configures the RS to run in HIST-Only Mode. The RS will never write to the remote database, but will continue to support historical report queries. The HIST-Only RS does not support writing CDR, OR, SQA, or log data. It does not support data summarization or data purging. It does not support realtime (RT) call reports.

rs.nodb.enabled
Default Value: false
Valid Values: false, true
Changes Take Effect: at start/restart

Enables the 'no DB' feature for running the RS without a remote DB. This option controls 'no DB' mode. In 'no DB' mode, the RS functions without a remote DB, however, only a limited number of reporting services are available.

rs.partitioning.enabled
Default Value: false
Valid Values: false, true
Changes Take Effect: at setup

Enables CDR/VAR/Upstream partitioning feature. This option must not be changed after initial RS installation (unless the database is reconfigured).

rs.partitioning.partitions-per-day
Default Value: 8
Valid Values: 1, 2, 3, 4, 6, 8, 12, 24
Changes Take Effect: at start/restart

Number of partitions a day for CDR/VAR/CustomVAR/EVENT storage. It must be a divider of 24. The higher values are suitable for smaller partitions and high capacity storage. Smaller values correspond to larger partitions and lower storage capacity.

rs.partitioning.upstream-partition-number
Default Value: 1
Valid Values: 1, 5, 10, 50 and 100

Changes Take Effect: at start/restart

Number of Event Log partitions per a single CDR partition. Higher number intended for larger number of event metrics per call. At the same time higher number would result in heavier DB activity associated with storing and querying metric results. The number can be 1 (default), 5, 10, 50 and 100.

rs.partitioning.upstream-start-time.enabled
Default Value: true
Valid Values: false, true
Changes Take Effect: at setup

Enables faster VAR CDR/CDR EVENT queries by using START_TIME query hint.

rs.storage.metricsfilter
Default Value: *
Valid Values: ...
Changes Take Effect: at start/restart

This option controls the metrics that should be persisted to the database backend.

RS uses the string provided to filter metrics before saving to the database. The string uses the same format as in Reporting Client, such as 0-16,18,25,35,36,41,52-55,74,128,136-141. The default value is "*". When the parameter is missing, the default value will be assumed and all metrics received will be saved to the database.

rs.storage.upstream-serializer.watermark
Default Value: 60000
Valid Values: ...
Changes Take Effect: at start/restart

Defines the maximal number of calls cached in the serializer component before new calls are no longer processed.

GVP 8.1+ Handles Play Treatment and Play Application Differently

Return to Corrections Table of Contents

SUMMARY: The GVP 8.x Next Generation Interpreter (NGi) handles treatments differently. (GVP-15176)

The next publication of the GVP 8.1 Application Migration Guide will include this revision:
CHAPTER: Chapter 3: Migrating to GVP 8.1 NGi
SECTION: VoiceXML $-Variables
ADD THIS NOTE to *Table 11: Mapping $-Variable Functionality*, below `$sid$` and above `$scriptdata$`:

<table>
<tr><th>Variable</th><th>Description</th><th>NGI Equivalent</th></tr>
<tr><td><code>sid</code></td><td>The value of the Script ID as generated by the Queue Adapter or the IVR Server Client.</td><td>Not supported</td></tr>
<tr><td colspan="3">Note: Script ID is a parameter supplied by the routing strategy during play treatment and play application.

• GVP 7.x uses the GVP Interpreter (GVPi), which executes treatments in the 7.x studio application branches, based upon <code>script ID</code>.

• GVP 8.1 and later use the Next Generation Interpreter (NGi), and the routing strategy supplies the treatment/application URL directly. The URL is sent by IVR Server to CTIC and then to MCP, where it is executed. (MCP still receives the Script ID, but does not use it.)</td></tr>
<tr><td><code>$scriptdata$</code></td><td>The URL link to the page that the CFA generated in response to run script requests from the IVR Server Client.</td><td>Not supported.</td></tr>
</table>

New keyword Record for gvp:dest Attribute of VXML <log> tag

SUMMARY: To support IVR Recording the <log> tag attribute gvp:dest has a new value: **record**.

The next publication of the GVP 8.1 Voice XML Help will include these revisions:

HELP TOPIC: Standard VoiceXML > VoiceXML Tags
<log>
SECTION: Attributes
This table gets a new item in an existing row...

Attribute	Description
gvp:dest	This attribute specifies a space-separated list of files (<dest1> ..<destN>) to which the message generated by this log should be written. The valid values are: . . . • **Record**—To support IVR recording, the **record** destination specifies these recording commands, which do what their names suggest: Start, Pause, Resume, and Stop.

Correction to ECMAScript variable in the <submit> tag

SUMMARY:' The <submit> tag description contains inaccurate information:
Submitted ECMAScript objects in the namelist are *not* flattened.

The next publication of the GVP 8.1 Voice XML Help will include these revisions:

REMOVE this text:

~~When submitting ECMAScript objects in the namelist, the object will be flattened (all the object's properties will be submitted separately). For example, if object o has two properties, p1 and p2, such that o.p1="value1" and o.p2="value2", and the following transition is made:~~

```
<submit next="test.jsp" namelist="o" ...>
```

~~The HTTP request will contain o.p1=value1&o.p2=value2. Remember to retrieve the properties separately (as o.p1 and o.p2) in the server-side code used to generate the next page.~~

REPLACE with this text:

When an ECMAScript variable is submitted to the server its value is first converted into a string before being submitted. If the variable is an ECMAScript Object, its value would take on the ECMAScript string “[object]”. For example, if object o has two properties, p1 and p2, such that o.p1="value1" and o.p2="value2", and it should be submitted separately as o.p1 & o.p2

```
<submit next="test.jsp" namelist="o.p1 o.p2" >
```

Corrections to the VXML tags send, receive, and block

SUMMARY: "Form" is *not* a Parent of the <send> and <receive> tags; "Send" and "Receive" *are* Children of the <block> tag.

- REMOVE "form" from the Parent list for the <send> and "receive" tags.
- ADD "send" and "receive" to the Child list for the <block> tags.

The next publication of the GVP 8.1 Voice XML Help will include these revisions:

HELP TOPIC: Standard VoiceXML > VoiceXML Tags
<send>
SECTION: Parent

Parent

Block, Catch, Error, Filled, Foreach, ~~Form,~~ Help, If, Noinput, and Nomatch

HELP TOPIC: Standard VoiceXML > VoiceXML Tags
<receive>
SECTION: Parent

Parent

Block, Catch, Error, Filled, Foreach, ~~Form,~~ Help, If, Noinput, and Nomatch

HELP TOPIC: Standard VoiceXML > VoiceXML Tags
<block>
SECTION: Child

Child

Assign, Clear, Data, Disconnect, Exit, Foreach, Goto, If, Log, Prompt, Receive, Reprompt, Return, Script, Send, Submit, Text (#PCDATA), Throw, Value, and Var

Correction: The SSML lexicon tag does not have Audio as a Parent

SUMMARY: The <lexicon> tag does not have Audio as a Parent.

The next publication of the GVP 8.1 Voice XML Help will include this revision:

HELP TOPIC: Standard VoiceXML > Properties <lexicon>
SECTION: Parent
REMOVE Audio from the list of Parents, leaving just four:

Parent

~~Audio,~~ Enumerate, Foreach, Prompt, and Speak

Correction: `bargein` property's behavior with interpreters

SUMMARY: Behavior of the `bargein` property with GVP interpreters is missing from the VXML Help. (GVP-7433)

The next publication of the GVP 8.1 Voice XML Help will include this revision:

HELP TOPIC: Standard VoiceXML > Properties
SECTION: Prompt and Collect Properties Barge-in
ADD THE RED TEXT to the existing table row:

Property	Description	Default Value
`bargein`	Controls whether user input can be collected before prompts have finished playing: • `true`—Any user input can barge in during prompts. • `false`—No user input can barge in during prompts. **How `bargein` interacts with interpreters** In the legacy VoiceGenie interpreter, the application can set the value of the `bargein` property to `dtmf`. When this value is set, only DTMF input stops the prompt, whereas a start-of-speech that is detected by the speech recognizer does not. The Next Generation Interpreter (NGI) does not support this feature. In NGI, VCR controls are enabled even though `bargein` is set to `true`. For example, pressing `5` will pause audio that is playing.	`true`

Enable MCP to accept the audio skip duration value specified in a VCR Control application

SUMMARY: How to correctly specify the audio skip duration value.
The next publication of the GVP 8.1 Voice XML Help will include this revision:
TOPIC: Properties
SECTION: Audio Control Properties
ADD THIS TEXT:
When MCP does not accept the audio skip duration value specified in a VCR Control application, some possible reasons are:

- For the Next Generation Interpreter (NGI), you must specify property names in lowercase.
- You must use `com.genesyslab`.
- When you specify a time, you must specify a unit of measurement (no unit is assumed by default). Specify `s` (seconds) or `ms` (milliseconds) at the end of the value.

For example, this incorrect specification:
`<property name= "COM.VOICEGENIE.AUDIO.SKIPDURATION" value = "9000">`
should be corrected to read:
`\<property name= "com.genesyslab.audio.skipduration" value = "9000ms" />`

Update attribute to gvp:recordutterance

SUMMARY: Update unknown attribute saveutterance to recordutterance.

The next publication of the GVP 8.1 Voice XML Help will include this revision:

TOPIC: Recognizing Recorded Utterance
SECTION: Usage
Description:
The "GVP 8.1 Voice XML Help" document contains a "Recognizing Recorded Utterance" topic. In the 3rd example (Speech input from the recording of the caller's earlier input), it suggests using "saveutterance" as an attribute of <field>:

```
<?xml version="1.0"?>
<vxml version="2.1" xmlns="http://www.w3.org/2001/vxml">
  <meta name="maintainer" content="<yourname>@<yourserver.com>"/>
  <meta name="application" content="Recorded Audio Input 3"/>
  <property name="ASRENGINE" value="SPEECHWORKS"/>
  <property name="bargein" value="false"/>
  <form>
     <grammar xml:lang="en-US" version="1.0" root="ROOT"
              xmlns="http://www.w3.org/2001/06/grammar" type="application/srgs+xml">
        <rule id="ROOT" scope="public">
           <item> goodbye <tag>command='goodbye';</tag> </item>
        </rule>
     </grammar>
     <field name="field1" saveutterance="true" slot="command">
        <prompt>
           Please say goodbye.
        </prompt>
        <catch event="nomatch noinput">
           Try again.
           <reprompt/>
        </catch>
        <filled>
           I recognized goodbye with a confidence of
           <value expr="field1$.confidence"/>.
           Let's make sure I'm a consistent recognizer.
        </filled>
     </field>
     <field name="field2" audioinexpr="field1$.utteranceaudio" slot="command">
        <prompt>
           Please say goodbye.
        </prompt>
        <filled>
           I recognized goodbye with a confidence of
           <value expr="field2$.confidence"/>.
           <if cond="field1$.confidence == field2$.confidence">
              See, I am a consistent recognizer!
           <else/>
              I guess I'm not a consistent recognizer.
           </if>
        </filled>
     </field>
  </form>
</vxml>
```

This should be updated to "gvp:recordutterance".

The namespace extension should be added to the <vxml> tag. That is, change:

```
<vxml version="2.1" xmlns="http://www.w3.org/2001/vxml">
```

To:

```
<vxml version="2.1" xmlns="http://www.w3.org/2001/vxml" xmlns:gvp="http://www.genesyslab.com/
2006/vxml21-extension">
```

Also, change saveutterance to recordutterance in the following line in the Getting Started section, on the same page:

The shadow variable `$.utteranceaudio` from an earlier `<field>`, if `saveutterance` is enabled.

Remove "gvp:expr" <grammar> attribute

SUMMARY: Remove the "gvp:expr" <grammar> attribute from the help document

The next publication of the GVP 8.1 Voice XML Help will include this revision:

TOPIC: VoiceXML Tags
SECTION: <Grammar>
Description:
The "GVP 8.1 Voice XML Help" document contains the "gvp:expr" <grammar> attribute. This should be removed as it only worked with VGi and non-vxml 2.1. Delete the following information from the document:

(Optional) This attribute is an ECMAScript expression to be evaluated and used as the initial value of this grammar. This record will be visited only if the expression evaluates to undefined.

The default value is undefined.

This attribute is a GVP extension and must be qualified by the GVP-namespace when used in the tag.

Audioinexpr attribute is not valid

SUMMARY: The tag "audioinexpr" should be updated to reflect the proper tag of "gvp:audioinexpr".

The next publication of the GVP 8.1 Voice XML Help will include this revision:

TOPIC: VoiceXML Tags

SECTION: The "Field" and "Recognizing Recorded Utterance" sections

Description: In the "GVP 8.1 Voice XML Help" document, there are references of the attribute `audioinexpr` in the "Field" and the "Recognizing Recorded Utterance" sections.

While almost all references in the "Field" section properly label the attribute as `gvp:audioinexpr`, the references in the "Recognizing Recorded Utterance" section label it incorrectly as `audioinexpr`.

If you try to build an application with the attribute `audioinexpr` the validation will fail in both Composer and GVP. So, all references to `audioinexpr` must be updated to reflect `gvp:audioinexpr` in both these sections, including the use cases where sample vxml is provided for testing.

Update content in gvp:dest attribute description in <LOG> tag

SUMMARY: Include a cross-reference to the "record" value in the gvp:dest attribute description in <LOG> tag

The next publication of the GVP 8.1 Voice XML Help will include this revision:

TOPIC: VoiceXML Tags
SECTION: <LOG>
Description:

The gvp:dest attribute description in <LOG> tag in the "GVP 8.1 Voice XML Help" document currently does not contain "record" as one of the valid values.

Add the following cross-reference to the "record" value in the gvp:dest attribute description in <LOG> tag.

To know about the "record" value, refer to the "VoiceXML Syntax" section in IVR Recording of the Genesys Interaction Recording Solution guide.

Corrections to a TDM PBX Method Explanation

SUMMARY: Corrections to a TDM PBX method explanation.
The next publication of the VP Solution 8.1 Integration Guide will include these revisions.
CHAPTER: Chapter 4: Supported Call Flow Scenarios
SECTION: Time-Division Multiplexing PBX
MODIFY two paragraphs on pages 53-54:

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~~ routing point at SIPServer 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 or agent 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~~.

Resource Manager IP Address and Local Host Name no longer mandatory parameters

SUMMARY: These parameters should be removed from a procedure because they are no longer mandatory.

The next publication of the VP Solution 8.1 Integration Guide will contain this correction:
CHAPTER: Chapter 12: Configuration Tasks for CTI Through IVR Server
SECTION: Integrating with IVR Server in In-Front Mode
REMOVE THESE BULLET POINTS from *Procedure: Configuring the CTI Connector for IVR In-Front* on pages 183-184:

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.

~~*__Local Host Name__—Enter the IP address of the CTI Connector host.~~

Setting Up Voiceprint Carrier Message Detection

With Voiceprint Carrier Message Detection, you can configure Genesys Media Server to recognize prerecorded messages that carriers in a region may provide, and tailor your outbound response accordingly.

How It Works

During the pre-connect period of a call, the Genesys Media Server could receive one of many highly specific carrier messages. These messages vary with each situation and country and region, but they are definable and you can build a database of them. Using that database, your system can match the message that it hears during preconnect with a known recording—thus identifying a known situation—and then use that information to take the appropriate action.

If the remote party is an answering machine (AM), Media Server will know that and wait for the appropriate AM signal before delivering a message. If the remote party is an out-of-service number and Media Server finds a matching recording, you can configure it to take a different action.

Steps to Configure

You can use Voiceprint configuration options to define the database of sounds that MCP matches against incoming pre-connect messages, in real time.

1. Acquire or develop a list of pre-connect carrier messages that your system can hear.

   > **Tip**
   > To compile your list of carrier messages, Genesys suggests seeking help from your Partner or Genesys Professional Services, or by recording the various carrier messages in that country or region.

2. Open Genesys Administrator (GA) or Genesys Administrator Extension (GAX) and navigate to your Media Control Platform (MCP) application, and within it, the **mpc** section.
3. Enable Voiceprint with this exact configuration:
   - `cpa.enable_carrier_messages = true`
4. Assign a name to each carrier message with configurations like these examples:
   - `cpa.carriermsg.0 = file://c:\media\carriermsgs\IsBusy.wav`
   - `cpa.carriermsg.1 = http://someserver/media/carriermsgs/IsBusy2.mp3`

- `cpa.carriermsg.2 = http://someserver/media/carriermsgs/IsBusy3.wav`
- `cpa.carriermsg.3 = http://someserver/media/carriermsgs/NotInUse.wav`
- `cpa.carriermsg.4 = http://someserver/media/carriermsgs/CannotbeReached.wav`
- `cpa.carriermsg.5 = http://someserver/media/carriermsgs/Operator_NotInUse.wav`

  You can define up to 100 carrier messages (from `carriermsg.0` to `carriermsg.99`). We recommend defining more than 10 messages for best results. Gaps or empty locations in your configuration are legal, as in this example with nothing to the right of the equals sign:
  `cpa.carriermsg.0 =`

> **Tip**
>
> If your configuration includes multiple MCPs, consider the advantages of standardization for filenames and directory paths. You can achieve this by defining the audio file database and setting the preconnect lists on a single MCP, testing to be sure that's what you want, and finally exporting the related options from that MCP object. From there, you simply import your designed and tested configuration into the other Media Servers and MCPs that perform call progress detection.

5. Add your result lists, using configurations that match specific results with specific carrier messages. Consider these examples:
   - `[mpc]cpa.preconnectresult.busy.list = 0 1 2`
     BUSY is the response to `cpa.carriermsg.0` through `cpa.carriermsg.2`
   - `[mpc]cpa.preconnectresult.custom1.list = 3 4 5`
     CUSTOM1 is the response to `cpa.carriermsg.3` through `cpa.carriermsg.5`
   - `[mpc]cpa.postconnectresult.machine.list` **`= 7`**
     MACHINE is the response to `cpa.carriermsg.7` (See Configuring for Answering Machine Detection below.)

   Success! You have instructed Media Server how to identify up to 100 carrier messages and what action to take in each case.

> **Important**
>
> **One Number, One List:** Every value in every list must be unique. For example, you can place the number 1 (meaning `cpa.carriermsg.1`) in only one list. Duplicates corrupt VoicePrint detection results.

Configuring for Answering Machine Detection

If an answering machine takes the call, Media Server waits until the "leave-a-message" audio plays, then delivers the appropriate audio. Consider the following details:

- You must configure a postconnect carrier message result list, which could look like this:
  `cpa.postconnectresult.machine.list` `= 0 1 2 4`
  In this configuration, detecting **`carriermsg.0`**, **`1`**, **`2`**, or **`4`** triggers the result **`Machine`**.
  The One Number, One List rule applies here, too.
- Carrier message detection still occurs in preconnect mode—it *always* does. But as stated above, the

results are delayed.

- MCP must be configured to start in preconnect mode, and after AM detection, move to postconnect mode while running, allowing the carrier message detection results to be returned after the call is connected.

Optional Parameters

You can customize your setup to meet specific needs in your organization or to address troubleshooting issues. See Voiceprint Analysis Configuration Options and the Media Control Platform Configuration Options documentation for all details on these parameters.

`[mpc]`**`cpa.cm_enable_initial_tone_filter`** `= [true | false] (Default = false)`
This parameter enables initial tone filtering, which suppresses the generic single tone at the beginning of the audio frames for carrier message detection. (To recognize the multiple ring tones in some countries, MCP must remove the generic single tone that precedes all of them.)

`[mpc]`**`cpa.cm_initial_silence_suppression_level`** `= [integer value] (Default = 64)`
This parameter specifies the initial level needed for silence suppression in carrier message detection. Increasing the value will result in higher energy noise being suppressed.

`[mpc]`**`cpa.cm_match_percent`** `= [integer value] (Default = 80) <percent>`
This parameter specifies the how strict we are in carrier message matching. A higher value means stricter matching, which may restrict the results. A lower value means more lenient matching, which yields more results, but may reduce accuracy.

`[mpc]`**`cpa.readduration`** `= [integer value] (Default = 60)`
This parameter specifies the maximum number of seconds that MCP reads a carrier message before stopping. This prevents delays when MCP reads carrier messages that are too long.

`[mpc]`**`cpa.nframes_cm_detection`** `= [integer value] (Default = 50)`
This parameter specifies how many frames from the audio files and real-time incoming audio that MCP to uses for detection. A higher number means better accuracy but more time consumed in detection. A lower number means less time consumed but also less accuracy.

Troubleshooting Tips

- **Bad carrier message files:** Ensure that the carrier message audio files are in the correct format for MCP to use them.
- **Duplicate indices:** Duplicate indices will cause matching to not work properly. Ensure that there are no duplicates in result lists. Follow the One Number, One List rule.
- **Accurate Indices:** Ensure that the indices that you are trying to match are listed in the correct result lists.
- **File location:** Ensure that MCP can access the files (particularly HTTP files) and that the paths are correct. ERRORs in the logs will tell you which files have issues.
- **MCP in the preconnect state:** MCP must be in the preconnect state for preconnect carrier message

detection to work. An MCP configured in the wrong state may send the incorrect input to the call progress analysis, and cause detection failures.

Voiceprint Analysis Configuration Options

Note: Find all of these options in the section `mpc`.

Option: `cpa.nframes_cm_detection`
Valid Value: An integer 25-100
Default Value: 50
Takes effect: At start/restart

Specifies the number of frames needed for carrier messages detection.

- Increasing the number of frames improves precision, but delays the result.

Option: `cpa.cm_initial_silence_suppression_level`
Valid Value: An integer 0-100, common range = 40-100.
Default Value: 64
Takes effect: At start/restart

Specifies the initial level for silence suppression in carrier messages detection.

- Increasing the value of this parameter results in high energy noise being suppressed.

Option: `cpa.cm_enable_initial_tone_filter`
Valid Values: `true` (default), `false`
Takes effect: At start/restart

This ringtone filter suppresses the generic single tone at the beginning of incoming frames, for carrier message detection.
`True` enables the initial ringtone filter during carrier messages detection.
`False` disables the initial ringtone filter during carrier messages detection.

- The suppression starts when 10 consecutive frames (400msec) of a given frequency is detected.
- There is no limit for tone duration, so both continuous and intermittent tone (like ring tones) are suppressed.

Option: `cpa.cm_match_percent`
Valid Value: An integer 40-80, common range = 70-80
Default Value: 80
Takes effect: Immediately

Specifies matching percent for carrier messages detection. Specifies the matching criteria as a parameter for tuning matching conditions.

- Increasing the number improves precision, but delays the result.

Option: `cpa.enable_carrier_messages`
Valid Values: `true`, `false` (default)
Takes effect: Immediately

Enables/disables carrier messages detection.
`True` enables carrier messages detection.
`False` disables carrier messages detection.

Option: `cpa.carriermsg.0`
Option: `cpa.carriermsg.1`
Option: `cpa.carriermsg.2`
Option: `cpa.carriermsg.3`
Option: `cpa.carriermsg.4`
Option: `cpa.carriermsg.5`
Option: `cpa.carriermsg.6`
Option: `cpa.carriermsg.7`
Option: `cpa.carriermsg.8`
Option: `cpa.carriermsg.9`
Valid Value: A valid pathname and filename
Default Value: None
Takes effect: Immediately

Specifies the full path to the file containing carrier message #0. Path should include prefix `file://` `http://`, or `https://`.
Examples:

```
cpa.carriermsg.0 = file://E:\cpatest\carrierMsg\cm_NotInUse_Operator1.pcm16
cpa.carriermsg.0 = file://opt/msg/cm_NotInUse_Operator1.pcm16
cpa.carriermsg.0 = http://localhost/cm_NotInUse_Operator1.wav
```

- You can configure as many as 100 carrier messages. The limitation of 100 files is due to performance purpose.
- Carrier messages should be defined sequentially (`cpa.carriermsg.0` followed by `cpa.carriermsg.1`, and so on).

Option: `cpa.carriermsg.readduration`
Default Value: 60
Valid Value: a non-negative integer, 60-300 inclusive.
Takes effect: Immediately, but wait at least a few minutes while the coefficients for all carrier messages are recalculated.

Specifies how much time MCP will spend reading a carrier message file, to prevent MCP from reading carrier message files that are too long. Very long carrier message files can impact matching accuracy.

- This parameter does not govern HTTP request timeouts.
- Do not rely on this parameter to precisely measure how long to read each file.
- Make sure that the read duration is longer than your longest carrier message file.
- Coefficients for all the carrier message are recalculated following a change to this parameter.

Option: `cpa.preconnectresult.custom1.list`
Option: `cpa.preconnectresult.custom2.list`
Option: `cpa.preconnectresult.custom3.list`
Option: `cpa.preconnectresult.custom4.list`
Valid Value: A string of single integers, separated by spaces.
Default Value: None
Takes effect: Immediately

Specifies a list of carrier messages that will return `result = sit.custom1` in preconnect mode.
Example: `cpa.preconnectresult.custom_1.list = 0 2 4` ... is a sequence of integers equal to the index of carrier messages previously specified. It specifies that if an incoming message matches with `carriermsg.0, carriermsg.2` or `carriermsg.4` then CPA outputs `result = sit.custom1`.

Option: `cpa.preconnectresult.busy.list`
Valid Value: A string of single integers, separated by spaces.
Default Value: None
Takes effect: Immediately

Specifies a list of carrier message files associated with `result = busy` in preconnect mode.
Example: cpa.preconnectresult.busy.list = 0 2 4 ...is a sequence of integer number equal to the index of carrier message previously specified. It means that if incoming message matches with `carriermsg.0, carriermsg.2` or `carriermsg.4` then CPA outputs `result = busy`.

Option: `cpa.preconnectresult.fast_busy.list`
Valid Value: A string of single integers, separated by spaces.
Default Value: None
Takes effect: Immediately

Specifies a list of carrier messages that will return `result = busy` in preconnect mode.
Example: `cpa.preconnectresult.fast_busy.list = 0 2 4` is a sequence of integers equal to the index of carrier messages previously specified. If the incoming message matches with `carriermsg.0, carriermsg.2` or `carriermsg.4` then CPA outputs `result = busy`.

Option: `cpa.preconnectresult.sit_nocircuit.list`
Valid Value: string of single integers, separated by spaces.
Default Value: None
Takes effect: Immediately

Specifies a list of carrier messages that will return the `result = sit.nocircuit` in preconnect mode.
Example: cpa.preconnectresult.sit_nocircuit.list = 0 2 4 is a sequence of integers equal to the index of carrier messages previously specified. It means that if an incoming message matches with `carriermsg.0, carriermsg.2` or `carriermsg.4` then CPA outputs result = sit.nocircuit.

Option: `cpa.preconnectresult.sit_vacantcircuit.list`
Valid Value: A string of single integers, separated by spaces.
Default Value: None
Takes effect: Immediately

Specifies a list of carrier messages that will return `result = sit.vacantcircuit` in preconnect

mode.
Example: cpa.preconnectresult.sit_vacantcircuit.list = 0 2 4 is a sequence of integers equal to the index of carrier message previously specified.
If an incoming message matches with `carriermsg.0, carriermsg.2` or `carriermsg.4` then CPA outputs result = sit.vacantcircuit.

Option: `cpa.preconnectresult.sit_operatorintercept.list`
Valid Value: A string of single integers, separated by spaces.
Default Value: None
Takes effect: Immediately

Specifies a list of carrier messages that will return `result = sit.operatorintercept` in preconnect mode.
Example: `cpa.preconnectresult.sit_operatorintercept.list = 0 2 4` is a sequence of integers equal to the index of carrier message previously specified. If an incoming message matches with `carriermsg.0, carriermsg.2` or `carriermsg.4` then CPA outputs result = sit.operatorintercept.

Option: `cpa.preconnectresult.sit_reorder.list`
Valid Value: A string of single integers, separated by spaces.
Default Value: None
Takes effect: Immediately

Specifies a list of carrier messages that will return `result = sit.reorder` in preconnect mode.
Example: cpa.preconnectresult.sit_reorder.list = 0 2 4 is a sequence of integers equal to the index of carrier message previously specified. If the incoming message matches with `carriermsg.0`, `carriermsg.2` or `carriermsg.4` then CPA outputs `result defined = sit.reorder`.

Option: `cpa.postconnectresult.machine.list`
Valid Value: A string of single integers, separated by spaces.
Default Value: None
Takes effect: Immediately

Specifies a list of carrier messages that will return `result = machine` in postconnect mode.
Example: `cpa.postconnectresult.machine.list = 0 10` is a sequence of integers equal to the index of carrier message previously specified. If the incoming message matches with `carriermsg.0 or carriermsg.10`, then CPA outputs `result = machine`.

Revise the table Task Summary: Specifying Windows Services and Settings

DONE HERE

SUMMARY: The GVP Deployment Guide description of Windows Services and Settings that can be Specified needs several cautionary notes.

The next publication of the GVP 8.5 Deployment Guide will contain this correction:
CHAPTER: Chapter 5: Preparing the Operating System for GVP
SECTION: Windows Services and Settings
MAKE THESE ADDITIONS to the table *Task Summary: Specifying Windows Services and Settings* on pages 204-206:

- Add a note to the top of the table (page 204):

  Note: Some of the services shown here may no longer be available on Windows 2008/2012.

- Add the footnote mark **"A"** to the row containing `IIS Admin Service` (page 205), and add the corresponding footnote following the table (page 206):

  A: Not required for the default setup of MCP component. If you require inline grammars on the MCP, then consult Microsoft documentation on how to add the appropriate roles and features in Windows 2008/2012 to enable the Internet Information Services (IIS). Also, refer to the GVP 8.5 User's Guide or the Voice Platform Media Control Platform Configuration Options for how to configure inline grammars.

- Add the footnote marked **"B"** to the row containing `World Wide Web Publishing Service` (page 205), and add the corresponding footnote following the table (page 206):

  B: Inline grammars, when enabled, must be hosted on the local web server (MCP only). Please refer to the Microsoft documentation on how to add the appropriate roles and features in Windows 2008/2012 to enable the World Wide Web Publishing Service. Also, refer to the GVP 8.5 User's Guide or the Voice Platform Media Control Platform Configuration Options for how to configure inline grammars.

Corrections to `request-uri` Value

DONE HERE under GVP as a Self-service IVR

SUMMARY: Corrections to the valid values listed for the option `request-uri`.

DOCUMENT: The next publication of the GVP 8.5 Deployment Guide will include these revisions.

CHAPTER: Appendix E: Resource Manager High Availability

SECTION: *GVP as a Self-service IVR*

The table **Trunk Group DN Configuration (Outbound)** on page 405 of the GVP 8.5 Deployment Guide requires this correction:

For HPE, configure GVP as a Trunk Group DN in SIP Server:
Table: Trunk Group DN Configuration (Outbound)

Parameter Name	Parameter Value
...	
request-uri	~~`sip:msml@<RMHost>:<RMport>;gvp-tenantid=<Tenant-Name>`~~ `sip:msml@<RMHost>:<RMport>;gvp-tenant-id=[<Tenant-Name>]`

SECTION: *GVP as an Outbound Media Server*

The unnamed table beneath "Configure GVP as a Trunk Group with these parameters:" on page 407 requires this correction:

Parameter Name	Parameter Value
...	
request-uri	~~`sip:msml@<RMHost>:<RMport>;gvp-tenantid=<Tenant-Name>`~~ `sip:msml@<RMHost>:<RMport>;media-service=cpd;gvp-tenant-id=[<Tenant-Name>]` **Note:** If Call Progress Analysis will not occur, then remove the item `media-service=cpd` from the `request-uri` value, to avoid false reporting in Resource Manager.

Correction: SSL Use No Longer Affects Performance

DONE HERE under Secure Communications

SUMMARY: SIP Server now supports secure SIP capabilities.

The next publication of the GVP 8.5 Deployment Guide will include this revision:

CHAPTER: Chapter 2: GVP Architecture
SECTION: Secure Communications
On page 63, remove the first line of text from the first bullet point in this section:

Considerations and Usage Notes

Before you implement widespread use of HTTPS in your GVP deployment, consider the following:

- ~~Complete use of SSL will affect platform performance and capacity.~~ Lags in fetch times and high CPU usage are normal when SSL is used, because the web server must encrypt every byte of data, and the platform must then decrypt the received data. In addition, an SSL handshake takes place between the web server and the platform before data transmission starts.

Enable Inline Grammar Access by URL

Follow these steps to enable inline grammar access by URL:

Windows IIs Environment

1. Verify that IIS is installed and started, and that MCP is installed.

2. Create and add the directory `inlinetmp` under `C:\Program Files\Common Files\GCTI\www\gvp\mcp\<MCP_application>\grammar\`.

Steps 3 and 4 are automatically performed by the MCP installation in a Windows 32-bit environment, but must be performed manually in a 64-bit environment:

3. Add the application `mcp` in IIS, which points to `C:\Program Files\Common Files\GCTI\www\gvp\mcp`.

4. Enable **Directory Browsing** and add the mime type `application/octet-stream` for the extension * to the application `mcp` in IIS.

Linux Environment

1. Verify that MCP is installed.

2. Create the virtual directory `/var/www/gvp/mcp/` by adding these lines of code to the file `/etc/http/conf/httpd.conf`:

```
Alias /mcp/ "/var/www/gvp/mcp/"
<Directory "/var/www/gvp/mcp/">
   Options Indexes MultiViews
   AllowOverride None
   Order allow,deny
   Allow from all
   ExpiresActive On
   ExpiresDefault "now plus 5 minutes"
</Directory>
```

3. Enable the `httpd` service
(run `chkconfig` and specify `level 345 httpd on`) ????????

4. Start the `httpd` service (run `start` in the directory `/etc/init.d/httpd`).

5. Add the directory `inlinetmp`, with grant read and access permission granted to all files and directories beneath `/var/www/gvp/mcp/<MCP_application>/grammar/`.

Debug Tracing Affects Performance

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CAUTION: Enabling debug logging may impact GVP performance, especially the `Verbose` setting. Therefore enable debug logging in a production environment **only** when advised to do so by an Engineering or Technical Support team member, for troubleshooting a specific issue. For best performance, use the application object's default logging level for GVP modules: `interaction` for MCP and `standard` for other components.

GVP Backward Compatibility

DONE HERE under Voice Platform Solution Components

SUMMARY: The current GVP version may be compatible with previous versions of SIP Server, Genesys Administrator and Configuration Server. (GVP-21392)

The next publication of the GVP 8.5 Deployment Guide will include this revision:

CHAPTER: Chapter 4: Prerequisites and Planning
SECTION: Voice Platform Solution
ADD THIS NOTE above Table 12—Versions Compatible With GVP:

> **Important**
>
> The table below lists the versions of Management Framework components and SIP Server that are recommended for each GVP release. However, the newest GVP version may still be compatible with a previous version of SIP Server, Genesys Administrator or Configuration Server. Please verify with Genesys Customer Care if you wish to keep a previous version of any of these components.

Secure SIP Capabilities Support

DONE HERE

SUMMARY: SIP Server now supports secure SIP capabilities.

The next publication of the GVP 8.5 Deployment Guide will include this revision:

CHAPTER: Chapter 7: Post-Installation Configuration of GVP
SECTION: Configuring the GVP Components
REMOVE THIS NOTE:

Note: Although the GVP components support secure SIP capabilities, the external SIP Server does not. Before you enable Secure SIP (SIPS) in your deployment, contact your Genesys sales representative for more information.

Contents of GVP and MS Release DVDs

DONE HERE

SUMMARY: The contents of the release DVDs for GVP and Media Server have changed over time.

The next publication of the Genesys Voice Platform 8.5.0 Deployment Guide will include this revision:

CHAPTER: Chapter 4, Prerequisites and Planning
SECTION: GVP Installation DVDs
CHANGE THE TABLE "CD Contents" on page 185 of the PDF:

Table 9: Contents of Release DVDs

Component	8.5.0	8.1.7	8.1.6	8.1.5	8.1.4	8.1.3
Genesys Voice Platform (DVD #1)						
Call Control Platform (CCP)	✔	✔	✔	✔	✔	✔
Reporting Server (RS)				✔	✔	✔
Squid Caching Proxy	✔	✔	✔	✔	✔	✔
Supplementary Services Gateway (SSG)	✔	✔	✔	✔	✔	✔
Computer Telephony Integration (CTI) Connector	✔	✔	✔	✔	✔	✔
Public Switched Telephone Network (PSTN) Connector					✔	✔
Policy Server (PS)	✔	✔	✔	✔	✔	
Media Resource Control Protocol (MRCP) Proxy	✔	✔	✔	✔	✔	
Genesys Media Server (DVD #2)						

Component	8.5.0	8.1.7	8.1.6	8.1.5	8.1.4	8.1.3
Resource Manager (RM)	✔	✔	✔	✔	✔	✔
Reporting Server (RS)	✔	✔	✔			
Media Control Platform (MCP)	✔	✔	✔	✔	✔	✔
GVP Reporting Plugin for GAX	✔	✔	✔			
Management Information Bases (MIB)	✔	✔	✔	✔	✔	✔

ASR and TTS Support

DONE HERE

SUMMARY: Genesys support of specific Nuance ASR/TTS software versions has changed.

The next publication of the GVP 8.5 Deployment Guide will include this revision:

BOOK: GVP 8.5 Deployment Guide
CHAPTER: Chapter 4: Prerequisites and Planning
SECTION: Prerequisites
CHANGES: The ASR and TTS sections of *Table 10: Software Requirements—Windows* and *Table 11: Software Requirements—Linux* are changed as follows:

Table 10: Software Requirements for Windows

Category	Requirements and comments
The upper portions of this table are not changed.	
Automatic speech recognition (ASR) (Optional)	Genesys recommends that the ASR servers are installed and operational before you install the Genesys Voice Platform. Genesys has validated the following third-party ASR software: • Nuance Recognizer 10.2.4 with Nuance Speech Server (NSS) 6.2.5 • Nuance Recognizer 9.0.18 with Nuance Speech Server (NSS) 5.1.7 • Telisma Telispeech ASR 2.0 SP1. • IBM WebSphere Voice Server (WVS) 6.1.1 ASR or higher. **Notes:** • It is your responsibility to obtain the software and the appropriate licenses. • Media Resource Control Protocol version 1 (MRCPv1) supported. • MRCPv2 supported only with Nuance NSS. • For more speech information, see the Genesys Supported Media Interfaces Reference Manual.
Text-to-speech (TTS) (Optional)	Genesys recommends that the TTS servers are installed and operational before you install the Genesys Voice Platform. Genesys has validated the following third-party TTS software: • Nuance Vocalizer 6.0.2 with Nuance Speech Server (NSS) 6.2.5

Category	Requirements and comments
	• Nuance Vocalizer 5.7.3 with Nuance Speech Server (NSS) 6.2.5 • Nuance Vocalizer 5.0.5 with Nuance Speech Server (NSS) 5.1.7 • IBM WebSphere Voice Server 6.1.1 TTS or later, with IBM TTS connector **Notes:** • It is your responsibility to obtain the software and the appropriate licenses. • MRCPv1 supported. • MRCPv2 supported only with Nuance NSS. • For more speech information, see the Genesys Supported Media Interfaces Reference Manual. • RealSpeak is now at End of Life for both Nuance and Genesys.

Software Requirements for Linux

The table below summarizes the software requirements for GVP 8.5 deployments on Linux:

Table 11: Software Requirements Linux

Category	Requirements and comments
The upper portions of this table are not changed.	
Third-party Supporting Components	
Automatic speech recognition (Optional)	Genesys recommends that the ASR servers are installed and operational before you install the Genesys Voice Platform. Genesys has validated the following third-party ASR software: • Nuance Recognizer 10.2.4 with Nuance Speech Server (NSS) 6.2.5 • Nuance Recognizer 9.0.18 with Nuance Speech Server (NSS) 5.1.7 • Telisma Telispeech ASR 2.0 SP1. • IBM WebSphere Voice Server (WVS) 6.1.1 ASR or higher. **Notes:** • It is your responsibility to obtain the software and the appropriate licenses. • MRCPv1 supported.

Category	Requirements and comments
	• MRCPv2 supported only with Nuance NSS. • For more speech information, see the Genesys Supported Media Interfaces Reference Manual.
Text-to-speech (Optional)	Genesys recommends that the TTS servers are installed and operational before you install the Genesys Voice Platform. Genesys has validated the following third-party TTS software: • Nuance Vocalizer 6.0.2 with Nuance Speech Server (NSS) 6.2.5 • Nuance Vocalizer 5.7.3 with Nuance Speech Server (NSS) 6.2.5 • Nuance Vocalizer 5.0.5 with Nuance Speech Server (NSS) 5.1.7 • IBM WebSphere Voice Server 6.1.1 TTS or later, with IBM TTS connector **Notes:** • It is your responsibility to obtain the software and the appropriate licenses. • MRCPv1 supported. • MRCPv2 supported only with Nuance NSS. • For more speech information, see the Genesys Supported Media Interfaces Reference Manual.

Route Unavailable Wakeup

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The Problem: Previously, when both Resource Managers (RMs) were marked offline, the MCP waited until the value of `transport.routerecoverytime` had expired, then tried the RMs again. The result was an unnecessary delay.

The Solution: Now, if both RMs are marked offline (i.e., "this route is unavailable"), MCP ignores the route `transport.routerecoverytime` setting (default=30 seconds) and immediately tries again, one RM at a time.

The Technical Description: The Route Unavailable Wakeup feature controls the availability of static route groups and DNS SRV domains. "Unavailable" is a self-explanatory state, except that it may not always be `true`, or not `true` *yet*. When enabled (the recommended—and default—setting) this feature ignores both the RM's state of "unavailability," and the pre-set route recovery wait time, no matter what value either holds.

Enable and disable this feature by setting the configuration options `sip.transport.unavailablewakeup` and `transport.unavailablewakeup`:

Option: `sip.transport.unavailablewakeup`
Application: `Resource Manager`
Section: `proxy`
Valid values: `true` (default and recommended setting) or `false`
Takes effect: After restart.
This option affects Resource Manager for all SIP requests, and mostly commonly applies to requests of a SIP Server that was configured using the DNS SRV domain.

Option: `sip.transport.unavailablewakeup`
Application: `Media Control Platform`
Section: `vrmrecorder`
Valid values: `true` (default and recommended setting) or `false`
Takes effect: After restart.
This option affects Media Control Platform for SIP requests for third-party recording *only*.

Option: `transport.unavailablewakeup` *(Note: No "sip" in this option name)*
Application: `Media Control Platform`
Section: `sip`
Valid values: `true` (default and recommended setting) or `false`
Takes effect: After restart.
This option affects Media Control Platform for all SIP requests *except* third-party recording.

For All Three Options:

- Set to `true` to enable (when one or more static route groups, or one or more DNS SRV domains, are also configured; and only if all the resources are unavailable when a new request is received). All destinations corresponding to the static route group(s) or DNS SRV domain(s) are marked available for new requests, *before* `sip.transport.routerecoverytime` expires.

(Previously, they were unavailable until *after* `sip.transport.routerecoverytime` expired.)

- Set to `false` to disable this feature. All destinations corresponding the static route group(s) or DNS SRV domain(s) are marked unavailable, and they remain unavailable until *after* `sip.transport.routerecoverytime` expires.

How are these options different from each other? They specify where the Unavailable Wakeup functionality is enabled—and that is determined by which of the parameters is set.

- If the parameter is set in the `vrmrecorder` section, then it applies to third party recording requests.
- If the parameter is set in the `sip` section, then it applies to other SIP requests.

Reliable Connection Retry

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SUMMARY: Resource Manager (RM) can now resend a failed SIP request message—specifically, a TCP message for non-INVITE client transactions—using a reliable channel, instead of immediately sending an error to the originator of the request. (GVP-21311)
Three new gateway parameters enable this behavior.

- Configure Reliable Connection Retry
- Parameters Used in Reliable Connection Retry

Configure Reliable Connection Retry

Follow these steps to specify the parameters correctly:

1. Open Genesys Administrator.
2. Select the **Resource Manager** Application.
3. Open the section **Connections** and select the Gateway where the parameters will be added.
4. Click **Edit** and select the **Advanced** tab.
5. Enter all three parameters listed below into the **Application Parameters** text box, separated by semicolons:
    - `retrydest-on-conerr=<`*`ipaddress:port`*`>`
    - `connerr-retry-timeout=<`*`timeout`*`>` (in milliseconds)
    - `connerr-max-retries=<`*`attempts`*`>`

    For example:
    `retrydest-on-conerr=192.000.0.000:8008; connerr-retry-timeout=3000; connerr-max-retries=10`

    See Parameters Used in Reliable Connection Retry for details about these parameters.

With this feature enabled, the message is re-sent until...

- **...the connection succeeds** — RM forwards the received successful response to the originator of the request.
  OR
- **...the retries number** (`connerr-max-retries`) **is reached** — RM tries to connect again, every time that `connerr-retry-timeout`is reached.
  OR
- **...the timeout value** (`Timer_F`) **is reached** — RM returns a 408 Request Timeout response code to the originator of the request.

> **Tip**
>
> RM adds these parameters to the X-Genesys-Session-ID header while forwarding the INVITE request to MCP from a specific SIP-S gateway resource. Thus, any SIP request that is made from MCP to SIP-S via RM for the call would contain these configured parameters inside the message.

Parameters Used in Reliable Connection Retry

Parameter: `retrydest-on-conerr`
Valid values: <*`IPAddress:Port`*>)
...where

- *`IPAddress`* is the same IP Address where the message was previously sent. If not specified or incorrect, then Resource Manager does not retry.
- *`Port`* is a valid port. If not specified or invalid, Resource Manager uses the default `5060`.

Specifies the destination that Resource Manager uses to retry the connection for sending a message, following a connection error.
Required: `connerr-max-retries > 1`.

Parameter: `connerr-retry-timeout`
Valid values: A positive integer that specifies milliseconds. The default is `3000` (3 seconds).
Specifies how much time Resource Manager waits, after a connection error, for the before it retries the connection.

Parameter: `connerr-max-retries`
Valid values: A non-negative integer.
Specifies the number of retries. Any positive number >1 enables this function. Any other value, including omission or no value (the default), disables it.

Third-Party Recording with MCP and RM

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SUMMARY: A new configuration option enables MCP to use the same Resource Manager for third-party recording that was used for incoming requests. (GVP-21310)

Option: `userouteonrecording`
Section: `sip`
Valid Values: `true` or `false`
Takes effect: immediately, on a per-session basis.

- Set to `true` to specify that for third-party recording, MCP will use the same Resource Manager that was used for incoming requests, which increases the likelihood of reaching an active Resource Manager. **Note:** This setting overrides `vrmrecorder.sip.routeset`.
- Set to `false` to specify that MCP use `vrmrecorder.sip.routeset` if present. Otherwise, MCP will not set the Route header.

Changed GVP Configuration Options

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The configuration options `routefailovertime` and `routerecoverytime` are now unhidden in the MCP configuration template.

Option: `transport.routefailovertime`
Application: Media Control Platform
Section: `sip`
Valid values: 1-32
Takes effect: At restart
Specifies the failover time, in seconds, for SIP static routing and DNS HA routing. If a SIP request has not received a response within the failover time, and SIP static routing or DNS HA routing is enabled, then the SIP request will be retransmitted to an alternate route.

Option: `transport.routerecoverytime`
Application: Media Control Platform
Section: `sip`
Valid Values: 1-600
Takes effect: At restart.
Specifies the recovery time, in seconds, for SIP static routing and DNS HA routing. When SIP static routing or DNS HA routing is enabled and the route is marked as unavailable due to an error or a SIP response timeout, then the route will be marked as available again after the recovery time.

These two options were added to Resource Manager for release 8.5.0:

Option: `pass-tenantid-parameter-to-gateway`
Application: Resource Manager
Section: `rm`
Valid Values: `true` (default) or `false`

- Set to `false` to specify that RM does not pass the tenant-id RURI parameter (specified by `gvp-tenant-id`) in the request to the Gateway or an external SIP service.
- Set to `true` to specify that RM passes the tenant-id RURI parameter in the request to the Gateway or an external SIP service.

Option: `ignore-ruri-tenant-dbid`
Application: Resource Manager
Section: `rm`
Valid Values: `true` (default) or `false`

- Set to `true` to specify that RM does not use the `tenant-dbid` parameter received through the incoming INVITE RURI.
- Set to `false` to specify that RM uses the `tenant-dbid` parameter.

Mid-call Change to MSML Call Not Reflected

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SUMMARY: The word "not" was omitted from the description of this behavior, reversing the meaning.

The next publication of the GVP 8.5 Deployment Guide will include this revision:

CHAPTER: Chapter 2: GVP Architecture
SECTION: Reporting Server Functions
ADD "NOT" TO THIS NOTE:

> **Note:** GVP reporting is unable to track Media Server services use at the tenant level (by tenant or by application). Applications that use URS centric routing have the following reporting issue:
>
> During an MSML call into GVP, if SIP Server changes the `X-Genesys-gsw-ivr-profile-name` or the `X-Genesys-gvp-tenant-id` parameters in the middle of the call (e.g. applying different treatments that use different IVR profiles), the change is **not** reflected by Resource Manager, Media Control Platform or Reporting Server. All reporting for the call will be against the original IVR profile.

Limits to UTF-8 Support

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SUMMARY: Media Control Platform requires all characters in filenames to be encoded using UTF-8. (UTF-8 is a variable-width encoding for the Unicode character set, and the default-by-concensus for the world wide web.)

Also note that:

- Windows does not support UTF-8 local file names explicitly, but UTF-8 characters can be used (even if they do not display well on Windows).
- This does not impact UTF-8 coming from non-local filenames.

Latest IPs: Genesys Voice Platform

Genesys Voice Platform

Component	8.5.1 available	Latest IP Version	Windows	Linux
MRCP Proxy	Y	8.5.185.06	Y	Y
Supplementary Services Gateway (SSG)	Y	8.5.160.97	Y	Y
CTI Connector (CTIC)	Y	8.5.160.80	N	Y
		8.5.160.73	Y	Y
Squid Caching Proxy	Y	8.5.100.06	Y	N/A
Call Control Platform (CCP)	Y	8.5.010.82	Y	N
		8.5.010.81	Y	Y
Policy Server (PS)	N	8.5.010.10	Y	Y

Note: See the *Genesys Supported Operating Environment Reference Guide* page for more detailed information and a list of all supported operating systems.

Latest IPs: Genesys Media Server

The following table serves as a virtual media image documenting the latest Installation Packages (IPs) available for the Genesys Media Server CD, where available.

Genesys Media Server

Component	8.5.1 available	Latest IP Version	Windows	Linux
Resource Manager (RM)	Y	8.5.185.37	Y	Y
Media Control Platform (MCP)	Y	8.5.185.34	Y	Y
T-Server-CUCM to Media Server Connector	Y	8.5.184.06	Y	Y
Reporting Server (RS)	Y	8.5.181.77	Y	Y
GVP Reporting Plugin for GAX	Y	8.5.151.29	Y	Y
Management Information Base (MIB)	Y	8.5.130.40	Y	Y

See the Supported Operating Environment: Genesys Voice Platform page for more detailed information and a list of all supported operating systems.

GVP Compatibility with Management Framework

Return to Corrections Table of Contents

If Migrating to this GVP version:	Upgrade to Management Framework version of:	
Genesys Administrator	Configuration Server	
8.5.1	8.1.3	8.1.3
8.5.0	8.1.3	8.1.3
8.1.7	8.1.3	8.1.3

Recommendation about Database Retention Periods

SUMMARY: Reporting Server DB settings can affect performance.

The Genesys Voice Platform 8.1 User's Guide and the Genesys Voice Platform 8.5 User's Guide both require this revision:

CHAPTER: Chapter 14: Configuring the Reporting Server
SECTION: Configuring Database Retention Policies
ADD THIS NOTE below *Table 34: Default Maximum Units, by Granularity Level* on page 274:

> ### Important
> Genesys recommends using default values for Reporting Server database retention parameters. Using higher retention periods may have impact on RS performance and therefore not supported.

Call Progress Detection section

SUMMARY: Two changes to text The next publication of the GVP 8.5 User's Guide will include this revision:
CHAPTER: Chapter 11: Configuring the Supplementary Services Gateway
SECTION: Call Progress Detection

The following two changes are required:

- Call Progress Detection (Page 251)
  SIP Server selects a CPD provider (Media Gateway or Media Server), and sets the appropriate CPD mode. If it is configured for both providers, the Media Gateway takes precedence as the CPD provider. SIP Server is also responsible for receiving the CPD result from the CPD provide, and pass it to the Supplementary Services Gateway through the corresponding T-Event.
  In the above paragraph, provide should be replaced with provider.

- Table 31: CPD Attributes (page 253)
  Attribute - postconnecttimeout
  Description - Specifies the timeout interval (in seconds or milliseconds) for the post connect scenario. This value is passed to SIP Server in the TMakePredictiveCall request. SIP Server starts the timer starts when the outbound call is connected. If the timer expires without a call result detected, SIP Server sends the EventEstablsihed TEvent with the CallState attribute set to Unknown to the Supplementary Services Gateway
  The word starts after timer is not required.

CPA Configuration Options That Can be Overwritten

SUMMARY: Two changes to text The next publication of the The Genesys Voice Platform 8.1 User's Guide will include this revision:
CHAPTER: Appendix B: Media Control Platform Reference Information
SECTION: CPA Configuration Options That Can be Overwritten

The following change is required on page 447:

Call Progress Analysis (CPA) configuration options can be overwritten by the gvp.service parameters in the IVR Profile. The IVR Profile service parameter must be prefixed by voicexml.gvp.config for VoiceXML services, and by msml.gvp.config for MSML services.

In the above text, change "and by msml.gvp.config for MSML services" to "and by cpd.gvp.config for MSML services".

Update list of valid services for a Capability Requirement

SUMMARY: Update list of valid services for a Capability Requirement in the GVP User Guide.
The next publication of the GVP 8.5 User's Guide will include this revision:
CHAPTER: Chapter 6: Provisioning IVR Profiles
SECTION: IVR Profile Configuration Options
EDIT: Table 16 IVR Profile Configuration Options on page 117 of GVP 8.5 User's Guide does not include a full list of valid values for <service> for option <service> Capability Requirement.

Following is the current list of valid values for <service>:

- ccxml
- conference
- voicexml
- msml
- announcement

Update the list of valid values for <service> as follows:

- ccxml
- conference
- voicexml
- msml
- announcement
- CPD
- Recording Client
- Recording Server
- Treatment

Move MCP configurations options to the respective table

SUMMARY: A few MCP configuration options are listed in the MRCP Server Configuration Options table. Move them to the MCP Configuration Options table.

The next publication of the GVP 8.5 User's Guide will include this revision:

CHAPTER: Chapter 7: Configuring the Media Control Platform

SECTION: Important MRCP Server Configuration Options

On page 191, **Table 24: Selected MRCP Server Configuration Options** includes the following two MCP configuration options:

- ASR Resource Reservation (page 191)
- TTS Resource Reservation (page 192)

Move these two options to **Table 23: Selected Media Control Platform Configuration Options** on page 156:

- Add ASR Resource Reservation after ASR Engine Default on page 156
- Add TTS Resource Reservation after TTS Engine Default on page 187

GENERIC CORRECTION

SUMMARY: Write a one-sentence summary of the correction, here.

DOCUMENT: The next publication of the DOCUMENT LINK|DOCUMENT NAME will include these revisions.

CHAPTER: Chapter name goes here

SECTION: *Section name goes here*

Put the correction here.

How to Access a `nomatch` DTMF Result

SUMMARY: The GVP 8.1 Genesys VoiceXML Help states a variable incorrectly.

The next publication of the **GVP 8.1 Voice XML Help** will include this revision:
HELP TOPIC: Standard VoiceXML > VoiceXML Tags > <field>
scroll down 3/4 of the way to the bottom of this topic

CORRECTION: The text in red will be added to the next published version of GVP Genesys VoiceXML Help.
. . .
"The field shadow variable is only set if the user's input is recognized and the field name variable is set, according to the Field Variable Assignment rules. The `styleapplication.lastresult$.nomatchdtmf` object has corresponding properties which are set after every utterance, even if a nomatch occurs or the recognition result is not used to set the field variable. Also, when any shadow variable is not set, its value will be ECMAScript undefined."

What is this variable used for?

When someone dials a wrong number or makes a similar mistake, GVP generates the wrong DTMF result, `nomatch`. Some customers require that information to resolve that problem, and the variable containing that information is misnamed in Genesys VXML documentation. The Genesys variable that contains this information is `styleapplication.lastresult$.nomatchdtmf`.

Remove support for SSLv2

SUMMARY: Remove support for SSL v2.

The next publication of the Genesys Media Server 8.5 Deployment Guide will include this revision:

CHAPTER: Appendix A: Deploying the T-Server-CUCM to Media Server Connector
SECTION: Secure Communications
CHANGE TO MAKE
Two changes:
1). On page 21, remove SSL v2.
2). On page 133, add the following section "Secure Communications" before the section "Supported Media Operations".

Secure Communications
UCMC supports the following protocols for secure SIP communications:

- Secure SIP (SIPS) — SIP over the Transport Layer Security (TLS) protocol for media service requisition between UCMC and Resource Manager.
- Non-secure SIP — SIP over User Datagram Protocol (UDP) and Transport Control Protocol (TCP) for media service request messaging between UCMC and Resource Manager.
- Secure Socket Layer (SSL) — SSL version 3 (SSL v3), SSL version 23 (SSL v23), TLS v1, TLS v1.1, and TLS v1.2.

Note: SSL version 2 (SSL v2) is no longer supported.

Key and Certificate Authentication (this is sub section of **Secure Communications**)
UCMC ships with a generic private key and SSL certificate. Default SIP transports for TLS are configured in the UCMC Application object. Therefore, basic security is implemented without having to configure it.

For more stringent security, UCMC 8.1.5 and above supports using the attributes of the **sip.transport.configuration** option to configure a password for key and certificate authority to perform server authentication.

Note: password=[password] Applicable to SIPS only and is optional. The password is associated with the certificate and key pair, and is required only if the key file is password protected.

For more information about obtaining SSL keys and certificates, and configuring UCMC to use SIPS in your deployment, see the section on enabling secure communications in the GVP 8.5 User's Guide.