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SIP Voicemail Deployment Guide

SIP Voicemail 8.1.1

1/10/2022

Table of Contents

Welcome

SIP Voicemail Deployment Guide

Choose the right installation method for your environment - Complete Install or Manual Deployment - or access directly the configuration options for SIP Voicemail Server.

Choose Your Deployment

Complete Install

Genesys SIP Voicemail can be deployed using the SIP Voicemail Complete Install installation package on a single server.

Deployment Prerequisites

SIP Voicemail Complete Install

Manual Deployment

Genesys SIP Voicemail can also be deployed manually in a stand-alone mode using multiple servers. You can also deploy GSVM into an existing Genesys deployment consisting of Genesys Media Server and SIP Server.

Deployment Prerequisites

Manual Deployment

High-Availability Deployment

Available starting in release 8.1.1

Genesys SIP Voicemail can be deployed as part of a Solution HA scenario.

Configuration Options

Find detailed descriptions for all the configuration options available in the SIP Voicemail Server Application object.

SIP Voicemail Server Configuration Options

Migration Procedure

When updating from one version of SIP Voicemail to another, including hot fix updates, use the following procedure.

SIP Voicemail Migration Procedure

Choose Your Deployment
SIP Voicemail HA Deployment Guide

SIP Voicemail Deployment Prerequisites

This list applies to release 8.1.1.

Prerequisites

This section describes the prerequisites for the deployment of Genesys SIP Voicemail:

Prerequisite	Details
Operating System	Red Hat Enterprise Linux (RHEL), 64-bit Windows 2008 (64-bit)
C++	RELH5 Host must have C++ compatibility library (compat-libstdc++-33-3.2.3-61.i386.rpm) installed
RAM	Recommended 4GB of RAM or above available to the Java process.
Runtime Environment	Java Runtime Environment (JRE) v1.6 or above, 64-bit
Genesys Requirements	• Genesys Management Framework must be installed and configured. See the *Framework 8.1 Deployment Guide* for details. • A SIP Server instance for managing agents must be installed and configured. See the *Framework 8.1 SIP Server Deployment Guide* for details. **Note**: If you want to use an existing premise SIP Server to also process voicemail, you must use SIP Server version 8.1 or higher. • (Manual Deployment only) All application templates must be installed. Use the supplied templates for the SIP Voicemail Server. **Note**: The Complete Install automatically imports the required templates.

Next:	Go to the Manual Deployment or Complete Install task summary.

Maximize Performance

You can maximize performance from Genesys SIP Voicemail. The following hardware will handle 15,000 mailboxes, 200 concurrent calls, with a message duration of 30 seconds and an average of 15 messages per mailbox:

- 2 quad core CPUs, 2 GHz
- 16 GB RAM
- 500 GB HDD

All SIP Voicemail components need to be installed and running on all hardware.

New In This Release

Check out the new features that were added in the latest releases of SIP Voicemail.

New in Release 8.1.1

- High Availability. SIP Voicemail supports Solution HA, where all components are co-located on two servers.
- Pre-recorded prompts in many language options. English (US), English (UK), French, German, Spanish (Spain), Spanish (Latin America), Russian, Italian, Japanese, Portuguese (Brazil), Chinese (Mandarin). (Note that Nuance TTS is now removed from the product and no longer required).
- Windows 2008 (64-bit) Support.
- Message Waiting Indicator status synchronization. Synchronization occurs after specific scenarios.
- The MLCMD utility is now included in the SCS component as a replacement for the reboot option in HA Failure Scenarios.

SIP Voicemail Complete Install

This table lists the tasks that are required to deploy Genesys SIP Voicemail using the SIP Voicemail Complete Install.

Objective	Related procedures and actions
1. Ensure that your system meets the deployment prerequisites.	• See Deployment Prerequisites. • For Complete Install on Windows, the Wow 64 component must be installed. This enables 32-bit applications (other Genesys components) to be run on the 64-bit platform.
2. Copy CD to host.	Copy or mount the entire CD to the hard drive on the host computer where you want to run the install.

3. Run the SIP Voicemail Complete Install tool.	Follow the Install instructions. When prompted, be prepared to provide the following information:
	• Configuration Server Hostname
	• Configuration Server Network port
	• Configuration Server User name
	• Configuration Server Password
	• License information for SIP Server:
	  ◦ Full path to the license file, or
	  ◦ License Manager port number and host name
	• Amount of RAM available for this process. If unspecified, the default of 4 GB is used.
	• Premise SIP Server Application Name
	• Type of used Audio Format:
	  ◦ `Mulaw` (North America), or
	  ◦ `Alaw` (Europe)
	• LCA port (default is 4999)
	• Genesys configuration environment Applications folder for creating applications (default is `Voicemail`)
	• Specify the HA role for this server: Primary or Backup. If this is a non-HA deployment, select Primary.

Objective	Related procedures and actions
	• Enter the Virtual IP Address for the GSVM HA Server pair. **Note**: For non-HA deployments, leave this field empty. • Full path of the destination directory for installation (the temporary directory used by the installation utility. Genesys IPs will be installed into predefined `/opt/genesys` directory.)
4. Configure DNs for the Agent SIP Server Application.	Complete the following procedure: Configuring DNs for the Agent SIP Server
5. Configure the Agent SIP Server Application.	On the `Options` tab, in the `TServer` section, configure the following: • `dial-plan=<dial-plan DN with voicemail rules>` • To use the `No-Answer-Supervision` feature, configure related configuration options, which could be set at an Application- or at DN-level. See the *Framework 8.1 SIP Server Deployment Guide* for details.
6. Configure DNs for the SIP Server Application dedicated to GSVM.	Complete the following procedure: Configuring DNs for the GSVM SIP Server Application

Configuring DNs for the Agent SIP Server Application

Prerequisites

- The Agent SIP Server `Application` object is created and associated with a `Switch` object of type `SIP Switch`.

Start

1. Configure a voicemail DN:
    a. Under the SIP Server `Switch` object, create a `Voice over IP Service` DN.
    b. On the `Annex` tab of the voicemail DN, in the `TServer` section, set the following configuration options:

- `contact` — Specify the contact URI of the SIP Server for GSVM, in the form of: `<ipaddress>:<SIP port>`
- `service-type` — Set this option to `voicemail`.

Create one voicemail DN for each Genesys SIP Voicemail.

Note: The Complete Install tool creates a default `Voice over IP Servce` DN 9999. You can use this DN as an example.

- Configure a dial-plan DNs:

If Genesys SIP Voicemail is integrated in your existing Genesys environment where you have dialing plans set up, go to **Step c**. For new users, create a new `DN` object to be associated with the voicemail dial plan:

a. Under the SIP Server `Switch` object, create a `Voice over IP Service` DN.
b. On the `Annex` tab of the dial-plan DN object, in the `TServer` section, set the configuration option `service-type` to `dial-plan`.
c. Specify the dial-plan rule to be used for consultation voicemail. This rule allows users to dial the access code to access the voicemail system. Use the following format:

    `dial-plan-rule-<n>=<access_code>=>gcti::voicemail`

    You can define several dial-plan rules to access the voicemail as necessary.

d. (Optional) For network forwarding using a dial plan, configure a dial-plan rule that is similar to this example:

```
dial-plan-rule-<n> = <dialing pattern>=>${DIGITS};timeout=5;ontimeout=
gcti::voicemail;ondnd= gcti::voicemail;onnoresp=
```

`gcti::voicemail;onunreg=gcti::voicemail`

Note: You may need to use a different rule, depending on how you want forwarding to voicemail to work. The variable `<dialing pattern>` is used in the example.

e. If not already done, associate the dial-plan to the caller by adding the option:

`dial-plan = <name of dial-plan-DN>` to either:

- The Agent/DN that is calling the digits (a trunk for inbound calls), in the TServer section of the Annex tab.
- The Agent SIPServer application (will apply to all DNs that have no DN or Agent level option set).

Refer to the SIP Server Deployment Guide for more information about dial-plans and dialing patterns in dial-plans.

- Define the mailbox for DNs and/or Agent Logins.

a. Under the SIP Server `Switch` object, select the existing or create an `Extension` DN or Agent Login.

b. On the `Annex` tab of the `Extension` DN or Agent Login, in the `TServer` section, set the configuration option to represent the callers voicemail boxes.

- `gvm_mailbox` Set this option to the mailbox ID. Note that only digits are supported. It must be unique, but it can be assigned to multiple DNs.
- (Optional) To use the No-Answer-Supervision feature, configure related configuration options, which could be set at an Application- or at DN-level. See the 'Framework 8.1 SIP Server Deployment Guide for details.

Note: The Complete Install tool creates a default `Extension` DN 8899. You can use this DN as an example.

- Repeat **Step e** until all necessary mailboxes for `Extension` DNs and/or Agent Logins are set up.

End

Back	Return to the Manual Deployment or Complete Install task summary.

Configuring DNs for the GSVM SIP Server Application

Prerequisites

- The SIP Server Application object dedicated to GSVM is created and associated with the same Switching Office object as the Agent SIP Server Application.

Start

1. Create a Trunk Group DN:
    a. Under the SIP Server Switch object that is associated with the GSVM SIP Server Application, create a Trunk Group DN. It will be used for the connection with the Resource Manager.
    b. On the Annex tab, in the TServer section, create the following configuration option:

- contact: Specify the contact URI of the Resource Manager, in the form of: <ipaddress>:<SIP port>

- sip-to-pass-through: Set this to true. SIP Server passes the To header (with the mailbox-id) through to Resource Manager without any modifications. GVP resolves the mailbox-id to the correct IVR Profile.

Note: The Complete Install tool creates a default Trunk Group DN 9999. You can use this DN as an example.

- Create a Trunk DN:

a. Under the same Switch object, create a Trunk DN object. It will be associated with the Agent SIP Server.
b. On the Annex tab, in the TServer section, create the following configuration options:

- contact: Specify the contact URI of the Agent SIP Server, in the form of: <ipaddress>:<SIP port>
- force-register: Set this option to the Agent SIP Server voicemail DN, in the form of:

sip:<voicemail_DN>@<AgentSIP_Server_IP:Port>;user=voicemail

Note: The Complete Install tool creates a sample Trunk DN Premise-9999. You can use this DN as an example.

End

Back	Return to the Manual Deployment or Complete Install task summary.

SIP Voicemail Manual Deployment

These steps apply to release 8.1.1.

Manual Deployment Task Summary Table

Complete the following steps to deploy Genesys SIP Voicemail manually.

Objective	Related procedures and actions
1. Ensure that your system meets the deployment prerequisites.	See Deployment Prerequisites.
2. Install SIP Voicemail Server.	Complete one of the following procedures, as per your operating system: Installing SIP Voicemail Server (on Linux) Installing SIP Voicemail Server (on Windows)
3. Configure the SIP Voicemail Server Application.	Complete the following procedure: Configuring the SIP Voicemail Server Application object
4. Configure the Agent SIP Server Application.	On the `Options` tab, in the `TServer` section, configure the following options to activate the Message Waiting Indication (MWI) functionality: • `mwi-implicit-notify=true` • `subscription-event-allowed=*` • `dial-plan=<voicemail DN>` To use the No-Answer-Supervision feature, configure related configuration options, which could be set at an Application- or at DN-level. See the *Framework 8.1 SIP Server Deployment Guide* for details
5. Configure DNs for the Agent SIP Server Application.	Complete the following procedure: Configuring DNs for the Agent SIP Server Application
6. Configure the GSVM SIP Server Application.	On the `Options` tab, in the `TServer` section, configure the following options, configure the following options to activate the MWI functionality:

Objective	Related procedures and actions
	• mwi-implicit-notify="" • subscription-event-allowed=*
7. Configure DNs for the SIP Server Application dedicated to GSVM.	Complete the following procedure: Configuring DNs for the GSVM SIP Server Application

8. Configure the GVP Resource Manager Application.	Reorganized this section to separate mandatory step from single-site-only steps.
	Configure the following (mandatory):
	• Indicate that the DNIS for the IVR Profile is fetched from the Request-URI message: [rm].sip-header-for-dnis = request-uri
	Configure the following only if SIP-Server and Resource Manager are deployed on the same host:
	• On the Options tab, in the proxy section, modify the following options by entering new port numbers. For example:
	  • sip.localport = 5260
	  • sip.localsecureport = 5261
	  • sip.transport.0 = transport0 udp:any:5260
	  • sip.transport.1 =transport1 tcp:any:5260
	  • sip.transport.2 = transport2 tls:any:5261 cert=$InstallationRoot$/config/x509_certificate.pem key=$InstallationRoot$/config/x509_private_key.pem

9. Configure GVP resources.	Complete the following procedure: Configuring GVP Resources
10. Configure A GVP IVR Profile to point to SIP Voicemail Server.	On the Options tab, specify these parameters in their respective sections: • [gvp.general]service-type = voicexml • [gvp.service-prerequisite]initial-page-url = http://<Voicemail Server IP>:<8080>/voicemail-web **Note**: For HA deployments, use the Virtual IP address for the GSVM Server HA pair.

Objective	Related procedures and actions
11. Configure a GVP DID Group.	• On the Options tab, specify this parameter in its respective section: [gvp.dn-groups]<name of the group> = <DNs in the group> where <DNs in the group> are configured DNs under a Switch with which the voicemail SIP Server Application is associated. • Associate the IVR Profile with the DID Group, by specifying the following parameter in its respective section: [gvp.dn-group-assignments]voicemail=<DBID of the Voicemail IVR Profile>
12. Configure a GVP MCP object.	• On the Options tab, in the gvp.rm section, configure the following parameter: aor=sip:<MCP_IP_Address:MCP_Port> • Make sure the Squid is disabled by specifying the following parameter: fm.http_proxy="" • On the Connections tab, add the TTS Application object • On the Advanced tab, in theApplication Parameters dialog box, specify the following parameter: provisiontype=primary

Installing SIP Voicemail Server (on Windows)

Start

1. Using Configuration Manager or Genesys Administrator, create an `Application` object of type `Genesys Generic Server` using the supplied templates for the SIP Voicemail Server.
2. Install the SIP Voicemail Server on the target machine by launching the `setup.exe` file available on the product CD.
3. When prompted, input the following information:
    a. Enter the connection parameters to the Configuration Server associated with this SIP Voicemail Server: Configuration Server name, port, username and password.
    b. Select which application to install.
    c. Select the amount of RAM available (or press Enter to accept the default of 4).
    d. Specify the HA role for this server: primary or backup. If this is a non-HA deployment, select primary.
    e. Enter the full path of the destination directory for the installation. For example, /opt/genesys/VM
    f. Select the product version: 32-bit or 64-bit.
4. Verify that `Cassandra.yaml` found in the `<voicemail_install_path>/etc` directory to ensure that the following directories are defined and point to the Genesys SIP Voicemail installation folder:

    - `data_file_directories` — `<voicemail_install_path>/storage/data`
    - `commitlog_directory` — `<voicemail_install_path>/storage/commitlog`
    - `saved_caches_directory` — `<voicemail_install_path>/storage/saved_caches`

- When the installation is complete, start SIP Voicemail Server (for example, as a Service).
- After SIP Voicemail Server starts successfully, in a web browser, type the URL to the SIP Voicemail Server default home page"or example:

```
http://<target machine IP>:8080/voicemail-web
```

End

Next Steps

Configuring the SIP Voicemail Server Application Object OR Back to Task Table

Installing SIP Voicemail Server (on Linux)

Start

1. Using Configuration Manager or Genesys Administrator, create an `Application` object of type `Genesys Generic Server` using the supplied templates for the SIP Voicemail Server.
2. Install the SIP Voicemail Server on the target machine by running the `install.sh` script (on Linux).
3. When prompted, input the following information:
    a. Enter the connection parameters to the Configuration Server associated with this SIP Voicemail Server: Configuration Server name, port, username and password.
    b. Select which application to install.
    c. Select the amount of RAM available (or press Enter to accept the default of 4).
    d. Specify the HA role for this server: primary or backup. If this is a non-HA deployment, select primary.
    e. Enter the full path of the destination directory for the installation. For example, /opt/genesys/VM
    f. Select the product version: 32-bit or 64-bit.

4. Verify that `Cassandra.yaml` found in the `<voicemail_install_path>/etc` directory to ensure that the following directories are defined and point to the Genesys SIP Voicemail installation folder:

- `data_file_directories` — `<voicemail_install_path>/storage/data`
- `commitlog_directory` — `<voicemail_install_path>/storage/commitlog`
- `saved_caches_directory` — `<voicemail_install_path>/storage/saved_caches`

- When the installation is complete, start SIP Voicemail Server, by executing the `run.sh` script from the installation folder.
- After SIP Voicemail Server starts successfully, in a web browser, type the URL to the SIP Voicemail Server default home page"or example:

```
http://<target machine IP>:8080/voicemail-web
```

End

Next Steps

Configuring the SIP Voicemail Server Application Object

OR

Back to Task Table

Configuring the SIP Voicemail Server Application Object

Prerequisites

- Both the Agent SIP Server and the GSVM SIP Server `Application` objects have been created in the configuration environment.

Start

1. In the SIP Voicemail Server `Application` object, on the `Connections` tab, add both the SIP Server Applications dedicated to managing agents and dedicated to GSVM.
2. On the `Advanced` tab, in the `Application Parameters` dialog box, configure the following parameter only to the SIP Server dedicated to GSVM:

   - `voicemail=true`

- On the `Options` tab, in the `VoicemailServer` section, configure the following options:

   - locale
   - security-account-lockout-duration
   - security-admin-access-group
   - security-password-length-min
   - voice-can-deposit-during-extended-absence
   - voice-enrollment-enabled
   - voice-message-max-duration
   - voice-mailbox-message-count
   - voice-record-path

End

Back	Return to the Manual Deployment task summary.

Configuring DNs for the Agent SIP Server Application

Prerequisites

- The Agent SIP Server `Application` object is created and associated with a `Switch` object of type `SIP Switch`.

Start

1. Configure a voicemail DN:
   a. Under the SIP Server `Switch` object, create a `Voice over IP Service` DN.
   b. On the `Annex` tab of the voicemail DN, in the `TServer` section, set the following configuration options:

- `contact` — Specify the contact URI of the SIP Server for GSVM, in the form of: `<ipaddress>:<SIP port>`
- `service-type` — Set this option to `voicemail`.

Create one voicemail DN for each Genesys SIP Voicemail.

Note: The Complete Install tool creates a default `Voice over IP Servce` DN 9999. You can use this DN as an example.

- Configure a dial-plan DNs:

If Genesys SIP Voicemail is integrated in your existing Genesys environment where you have dialing plans set up, go to **Step c**. For new users, create a new `DN` object to be associated with the voicemail dial plan:

a. Under the SIP Server `Switch` object, create a `Voice over IP Service` DN.
b. On the `Annex` tab of the dial-plan DN object, in the `TServer` section, set the configuration option `service-type` to `dial-plan`.
c. Specify the dial-plan rule to be used for consultation voicemail. This rule allows users to dial the access code to access the voicemail system. Use the following format:

   `dial-plan-rule-<n>=<access_code>=>`gcti::voicemail

   You can define several dial-plan rules to access the voicemail as necessary.
d. (Optional) For network forwarding using a dial plan, configure a dial-plan rule that is similar to this example:

```
dial-plan-rule-<n> = <dialing pattern>=>${DIGITS};timeout=5;ontimeout=
gcti::voicemail;ondnd= gcti::voicemail;onnoresp=
```

```
gcti::voicemail;onunreg=gcti::voicemail
```

Note: You may need to use a different rule, depending on how you want forwarding to voicemail to work. The variable <dialing pattern> is used in the example.

e. If not already done, associate the dial-plan to the caller by adding the option:

dial-plan = <name of dial-plan-DN> to either:

- The Agent/DN that is calling the digits (a trunk for inbound calls), in the TServer section of the Annex tab.
- The Agent SIPServer application (will apply to all DNs that have no DN or Agent level option set).

Refer to the SIP Server Deployment Guide for more information about dial-plans and dialing patterns in dial-plans.

- Define the mailbox for DNs and/or Agent Logins.

a. Under the SIP Server Switch object, select the existing or create an Extension DN or Agent Login.

b. On the Annex tab of the Extension DN or Agent Login, in the TServer section, set the configuration option to represent the callers voicemail boxes.

- gvm_mailbox Set this option to the mailbox ID. Note that only digits are supported. It must be unique, but it can be assigned to multiple DNs.
- (Optional) To use the No-Answer-Supervision feature, configure related configuration options, which could be set at an Application- or at DN-level. See the 'Framework 8.1 SIP Server Deployment Guide for details.

Note: The Complete Install tool creates a default Extension DN 8899. You can use this DN as an example.

- Repeat **Step e** until all necessary mailboxes for Extension DNs and/or Agent Logins are set up.

End

Back	Return to the Manual Deployment or Complete Install task summary.

Configuring DNs for the GSVM SIP Server Application

Prerequisites

- The SIP Server Application object dedicated to GSVM is created and associated with the same Switching Office object as the Agent SIP Server Application.

Start

1. Create a Trunk Group DN:
    a. Under the SIP Server Switch object that is associated with the GSVM SIP Server Application, create a Trunk Group DN. It will be used for the connection with the Resource Manager.
    b. On the Annex tab, in the TServer section, create the following configuration option:

- contact: Specify the contact URI of the Resource Manager, in the form of: <ipaddress>:<SIP port>
- sip-to-pass-through: Set this to true. SIP Server passes the To header (with the mailbox-id) through to Resource Manager without any modifications. GVP resolves the mailbox-id to the correct IVR Profile.

Note: The Complete Install tool creates a default Trunk Group DN 9999. You can use this DN as an example.

- Create a Trunk DN:

a. Under the same Switch object, create a Trunk DN object. It will be associated with the Agent SIP Server.
b. On the Annex tab, in the TServer section, create the following configuration options:

- contact: Specify the contact URI of the Agent SIP Server, in the form of: <ipaddress>:<SIP port>
- force-register: Set this option to the Agent SIP Server voicemail DN, in the form of:

sip:<voicemail_DN>@<AgentSIP_Server_IP:Port>;user=voicemail

Note: The Complete Install tool creates a sample Trunk DN Premise-9999. You can use this DN as an example.

End

Back	Return to the Manual Deployment or Complete Install task summary.

Configuring GVP Resources

Start

1. In the GVP Resource Manager Application object, add the SIP Server for GSVM Application to its connections.
2. On the Advanced tab, in the Application Parameters dialog box, specify the following parameters: logical-resource-section=SIPServer;aor=sip:<Voicemail SIP Server IP:Port;redundancy-type=active;port-capacity=<port capacity>.
   The port capacity specifies the maximum number of simultaneous active calls allowed on the Voicemail server. If set to 100, then the maximum number of simultaneous calls will be 100.
3. Create a new section with the value specified in the logical-resource-section (Step 1) to represent the gateway resources (such as a SIP Server connection to the Resource Manager). Specify the following options in that section:
   - load-balance-scheme=round-robin
   - monitor-method=none
   - port-usage-type=in-and-out
   - service-types=gateway
   - use-cti=0
4. Under the Environment folder, create a new configuration object named GVP. On the Annex tab, in the gvp.resources section, specify the parameter rm_dbid=<DBID of RM>.
   Use Genesys Administrator to perform the above step. See the GVP method of creating GVP configuration unit and resources, which is described in the GVP Deployment Guide.
5. Within the GVP configuration object, create a folder"or example, VMGroup, to represent the resources managed by Resource Manager, such as the Media Server. On the Annex tab, in the gvp.lrg section, specify the following parameters:
   - geo-location=""
   - load-balance-scheme=round-robin
   - monitor-method=option
   - port-usage-type=outbound
   - service-types=voicexml;ccxml;announcement;conference
6. Create the MCP Application object within this logical resource group folder under the GVP configuration unit.

End

Back	Return to the Manual Deployment task summary.

SIP Voicemail Server Configuration Options

Configuration options specific to the SIP Voicemail Server functionality are set in Configuration Manager or Genesys Administrator, in the `VoicemailServer` section on the `Options` tab of the SIP Voicemail Server `Application` object.

VoicemailServer section

This section must be called `VoicemailServer`.

locale

- Default Value: `en-US`
- Valid Values: A string
- Changes Take Effect: Immediately
- Specifies the default locale for the Telephone User Interface (TUI).

security-account-lockout-duration

- Default Value: `600000`
- Valid Values: Any positive integer
- Changes Take Effect: Immediately
- Specifies the time interval, in milliseconds, that SIP Voicemail Server waits before unlocking the account, which has been locked because of the incorrect login entries. The user is locked out after three failed login attempts.

security-admin-access-group

- Default Value: An empty string
- Valid Values: <Tenant>\<Access Group Name>
- Changes Take Effect: Immediately
- Specifies the members of the Access Group that will have permission to access the Admin interface of the voicemail.

security-password-check-internal-call

- Default Value: `true`
- Valid Values: `true, false`

- Changes Take Effect: Immediately
- Specifies whether the password check request is played when a user dials the voicemail system from an internal phone.

security-password-length-min

- Default Value: 4
- Valid Values: Any positive integer
- Changes Take Effect: Immediately
- Specifies the minimum length of the password digits.

voice-can-deposit-during-extended-absence

- Default Value: `true`
- Valid Values: `true, false`
- Changes Take Effect: Immediately
- When set to `true`, enables the voicemail deposit after the absence greeting is played.

voice-enrollment-enabled

- Default Value: `true`
- Valid Values: `true, false`
- Changes Take Effect: Immediately
- When set to `true`, enables the voicemail enrollment.

voice-greeting-extended-max-duration

- Default Value: `20`
- Valid Values: Any valid integer
- Changes Take Effect: Immediately
- Specifies the length of time, in seconds, allowed for the user to record their extended greeting. After the timeout ends, a prompt is played asking the user to confirm their recorded message.

voice-greeting-personal-max-duration

- Default Value: 12
- Valid Values: Any valid integer
- Changes Take Effect: Immediately

- Specifies the length of time, in seconds, allowed for the user to record their peronal greeting. After the timeout ends, a prompt is played asking the user to confirm their recorded message.

voice-message-max-duration

- Default Value: `10`
- Valid Values: Any positive integer
- Changes Take Effect: Immediately
- Specifies, in seconds, the message maximum duration.

voice-mailbox-message-count

- Default Value: `10`
- Valid Values: Any positive integer
- Changes Take Effect: Immediately
- Specifies the maximum number of voicemail messages per mailbox.

voice-record-path

- Default Value: An empty string
- Valid Values: A string
- Changes Take Effect: Immediately

Specifies the GVP directory where recorded files will be stored.

The GVP directory above refers to the MCP configuration parameters `[vxmli] recording.basepath` and defaults to `<MCPInstallationPath\record>`.

SIP Voicemail Migration Procedure

Start

1. Stop the GSVM Server. If installed in HA mode, stop both primary and backup GSVM Server instances.
2. Install the latest version of GSVM Server in a different directory than prior versions (in other words, do not install in the same directory as any old versions).
3. All previously deposited voicemails are kept in the `Storage` folder in the installation path for that install version. Copy the Storage folder from the previous GSVM version to the installation folder for the newly installed GSVM Server.
4. Start the primary voicemail server.
5. All data copied from the previous version of GSVM Server should now be available from the new, running version.
6. For HA deployments, complete these steps for the primary server instance only. The backup server instance will synchronize data with the primary automatically.

End

Back	Return to the Deployment Guide overview.