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# SIP Feature Server Deployment Guide

SIP Feature Server 8.1.2Legacy

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# Legacy SIP Feature Server Deployment Guide

## Warning

This Deployment Guide applies ONLY to the Legacy SIP Feature Server 8.1.200.xx release. Unless you have been told to use this documentation, use the current [8.1.2 Deployment Guide](#)

SIP Feature Server works in conjunction with SIP Server and other Genesys components to provide voicemail and other services.

You can set up multiple Feature Servers to handle load balancing. You can also achieve high availability through an N+1 configuration.

### Planning and pre-installation

[Plan and prepare](#) your environment.

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[Known issues and recommendations](#)

[Architecture](#)

[Hardware and software prerequisites](#)

### Deploying

[Deploy](#) SIP Feature Server.

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[Review the configuration options](#)

# Known issues and recommendations

## Warning

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## SIP Feature Server 8.1.2 known issues and recommendations

This document includes SIP Feature Server issues and recommendations applicable to the legacy 8.1.2 release.

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### Important upgrade step

If you are upgrading from a restricted release of SIP Feature Server 8.1.2 (any version prior to 8.1.200.83), you must manually restore the `vms_host` parameter, as follows:

1. Open `launcher.xml`.
2. In the `vms_host` section, in the line `<format type="string" default="localhost" />`, replace "localhost" with "0.0.0.0" or a specific IP address to restrict Feature Server web application access to that address.

If you are doing a fresh installation or upgrading from 8.1.200.83 or later, you can omit this step.

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### Feature Server upgrade requires manual Cassandra file backup and restoration

Before you upgrade Feature Server, you must first back up your Cassandra configuration and database files, then restore them after the upgrade. Complete these steps:

1. On the Feature Server master node (which is also the Cassandra seeds node), back up all files in the **etc** folder, which includes the `cassandra.yaml` file.
2. Back up the database files for all nodes in the Cassandra cluster (because the upgrade propagates changes to all nodes in the cluster). Note the file locations.
3. After completing the Feature Server upgrade, restore the backed up configuration and database files to their original locations.
4. Restart Feature Server and use the Feature Server administrative web application to verify that it is running.
5. Repeat these steps on each node.

(SIPVM-2006)

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### Use IP addresses

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Use IP Address or local host name, not FQDN, to access the Web application. FQDNs can cause unexpected logouts.

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### **Group Mailbox Administrator privileges**

Only users with the Group Mailbox Administrator role can use the web application or TUI to upload or change greetings and passwords.

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### **RequestPrivateService MWI notification generated with incorrect values**

In a multi-site environment, RequestPrivateService MWI notification for deposit and retrieval calls is intermittently generated with incorrect values. (SIPVM-1757)

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### **When the network is disabled on the host on which Feature Server is running, Windows may terminate unexpectedly**

Windows may terminate unexpectedly if the host on which Feature Server is running is removed from the network by clicking Disable from the Windows Local Area Connection dialog box. (SIPVM-1333)

#### *Workaround*

Stop Feature Server before disconnecting the network.

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### **Message priority is not being considered during retrieval**

The message priority selected during a call deposit is not taken into consideration during voicemail retrieval. (SIPVM-1248)

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In the Telephone User Interface (TUI), User and Group mailboxes can be accessed only with mailbox credentials (mailbox number). DN, agent, and user credentials to access mailboxes is not supported. (SIPVM-859)

# Planning and pre-installation

## Warning

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Before installing and configuring SIP Feature Server, you must plan your environment and install required hardware and software.

In your planning, you must:

- Meet [hardware and software prerequisites](#) such as operating system, hardware, and optional Genesys components.
- Review the current [known issues and recommendations](#) for this release.

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# Architecture

## Warning

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SIP Feature Server provides a scalable, load-balanced architecture that enables you to add Feature Server instances as your needs expand. A Cassandra-based data cluster maintains data consistency and performance across all instances.

## Provisioning and dial plan setup

Initial program configuration occurs primarily in Genesys Administrator (GA). You can create users, devices, and mailboxes only in GA. To set up your dial plan, a highly configurable set of call disposition patterns, you can select either of two methods:

- the Feature Server administrative web application
- the existing SIP Server methodology, using GA

Administrators can also use the administrative web application to provision users with devices, voicemail, and call settings.

## Voicemail

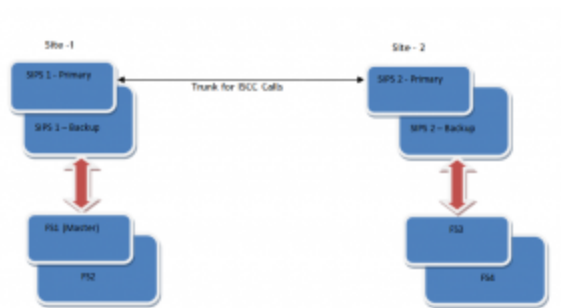
SIP Feature Server combines with Genesys Voice Platform (GVP) and SIP Server to handle voicemail tasks.

The User administrative web application and the Telephone User Interface (TUI) enable users to specify mailbox settings and manage their voicemail.

## Data management

SIP Feature Server uses [Apache Cassandra](#) data clusters to replicate data across the environment, achieving scalability and high availability. The SIP Feature Server installer deploys these clusters.

## Multi-site deployment



### Multi-site architecture

You can deploy SIP Feature Server across multiple data centers, in an active-active configuration.

In multi-site environments, Feature Server supports voicemail functionality, including deposit, retrieval and deletion of voice messages, and MWI, for ISCC calls. Voicemail configuration is the same for both multi-site and single-site deployments.



# Hardware and software prerequisites

## Warning

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Genesys SIP Feature Server requires the following hardware and software. Other system components, such as Genesys Management Framework, have their own hardware and software requirements.

## Sizing

To calculate the database disk space requirements for voicemail and monitoring in your environment, enter your information in column B of the [SIP Feature Server 8.1.2 Sizing Guide](#).

Category	Prerequisite
Host	Install Feature Server on its own dedicated host, unless you are deploying to a small or lab environment. Do not co-locate any other applications on this host.
Operating System	<ul style="list-style-type: none"> <li>Red Hat Enterprise Linux (RHEL) 5 or 6, 64-bit, or</li> <li>Windows 2008 (64-bit)</li> </ul> <p>All Feature Servers in your environment must run on either Linux or Windows. You cannot mix Linux and Windows machines.</p> <p>If you are running RHEL 6, you must:</p> <ul style="list-style-type: none"> <li>Install all RHEL 6 compatibility packages</li> <li>Create a symbolic link for <code>libcurl.so.3</code> (64-bit only): <pre>cd /usr/lib64  ln -s /usr/lib64/libcurl.so.4.1.1 ./libcurl.so.3</pre> </li> </ul> <p>If you are running RHEL 5, you must also install the C++ compatibility library (<code>compat-libstdc++-33-3.2.3-61.i386.rpm</code>)</p>
RAM	4GB of RAM or above available to the Java process.
Runtime Environment	Java Runtime Environment (JRE) v1.7 or above, 64-bit
Cassandra cluster	The Cassandra cluster that SIP Feature Server installs requires two disks, one for the commit log

Category	Prerequisite
	and the other for the data directories. At a minimum, place the commit log in its own partition.
Genesys Requirements	<p>Install and configure:</p> <ul style="list-style-type: none"> <li>• Genesys Management Framework. See the <i>Framework 8.1 Deployment Guide</i> for details.</li> <li>• Genesys Voice Platform components: Resource Manager and Media Control Platform. See the Genesys Voice Platform Deployment Guide.</li> <li>• A SIP Server instance for managing agents. See the <i>Framework 8.1 SIP Server Deployment Guide</i> 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.</li> <li>• All application templates. Use the supplied templates for the SIP Feature Server.</li> </ul>

Feature Server uses the following default ports. Do not use these ports for any other applications that share the host on which you install Feature Server.

Port	Used for
7000	Cassandra Storage_port
7001	Cassandra SSL_Storage_port
8080	http
8443	https
8800	Dial plan
9160	Cassandra rpc_port
9192	JMX remote port

# Deploying SIP Feature Server

## Warning

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Complete these steps to configure the SIP Feature Server instances for standalone operation.

## Configuring SIP Feature Server in standalone mode

<multistep> Configure SIP Feature Server applications.=

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To deploy one standalone SIP Feature Server:

1. Ensure that you have met all the [installation prerequisites](#) for SIP Feature Server. If you assign port numbers, ensure that you do not use any of the [reserved ports](#).
2. Configure the SIP Feature applications: In Genesys Administrator, use the supplied template (**Templates > GenesysSIPFeatureServer\_812.apd**) to create an application. Note the Application object name, which the application installer requires.
3. Create a Host for the target machine.
4. If this is the first Feature Server instance you are deploying, designate it to handle the initial synchronization of data from Configuration Server to all Feature Servers on a given switch.
  - In the Cluster > Options tab of the Feature Server Application, set master to true. See [Configuration options](#).
  - For other Feature Servers in the cluster, in the Cluster > Options tab of the Feature Server Application, set master to false and confsync to true.  
**Important:** Ensure that you designate only one Feature Server instance as the master.
5. In the Feature Server Connections tab, add the SIP Server to the Feature Server. Ensure that the PortID is the default SIP Server port.
6. Review the current [known issues and recommendations](#) for this release.
7. To install Feature Server, run `install.sh` (Linux) or `setup.exe` (Windows) from the product DVD. Follow the installer prompts, using the default values except for:

- Set the deployment mode to Standalone.
- Supply the Cassandra cluster name (use the same name for all Cassandra instances in the Cassandra cluster).
- Supply the IP address of the master Cassandra server.
- Supply a Cassandra storage location other than the installation directory.

8. In the file **<Cassandra storage location>/etc/Cassandra.yaml**:

- Verify that the file contains these properties, and update the path set at installation as needed:
  - `data_file_directories`: - <Cassandra storage location>/storage/data
  - `commitlog_directory`: - <Cassandra storage location>/storage/commitlog
  - `saved_caches_directory`: - <Cassandra storage location>/storage/saved\_caches
  - `storage_port`: - 7000 (set each Cassandra node to 7000)
- Enable the Cassandra replication of data between SIP Feature Server instances by supplying all the IP addresses of all feature servers after the instance is started:  
 <SIPFS1 IP Address>,<SIPFS2 IP Address>,<SIPFS\_N IP Address>

For example:

135.39.18.1,135.39.18.2,135.39.18.3

9. If this is the master Feature Server, disable the server firewall.
10. To enable a Transport Layer Security autodetect (upgradable) connection with a Configuration Server, configure the Configuration Server autodetect port. See Introduction to Genesys Transport Layer Security in [Genesys 8.1 Security Deployment Guide](#).
11. Set the remaining [Feature Server configuration options](#).

|–| Configure environment monitoring.=

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To configure monitoring for the components that you want to monitor from within the SIP Feature Server administration web application:

1. In Genesys Administrator, add the components to be monitored in the Connections tab of Feature Server applications.
2. Using the following table, create an http-port ID in the applications of the components that are to be monitored, where <port> is any free port in the host machine of a specific component:

Application	Navigate to	Value
SIP Server	Options > TServer	<port>
Stat Server	Server Info	<port>
ICON	Server Info	<port>
URS	Server Info	<port>
SIP Proxy	Server Info	<port>

|–| Start and verify SIP Feature Server.=

### Warning

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To start and verify SIP Feature Server:

### Warning

Do not start Feature Server until you have set the configuration options *replicationStrategyClassName* and *replicationOptions*. See [Cassandra options](#).

1. To run Feature Server in secure (https) mode:
  - Open the `start.ini` file and uncomment `etc/jetty-ssl.xml`
  - In the IVR Profile, set `initial-page-url = https://Feature Server IP address or host name:8443/fs`
2. Use Genesys Administrator, not the command line, to start SIP Feature Server. If you are running more than one Feature Server, start the Master first.
3. In Genesys Administrator, verify that the Feature Server is running.
4. Verify that the administration Web interface is running by logging in as the Default administrator (in other words, the Default user in Configuration Server):
 

```
<Feature Server IP address>:<port>/fs/admin
```

To enable other users to log in as administrators, [assign the Administrator role](#) to them.

|–| Configure SIP Server for Feature Server.=

### Warning

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release. Unless you have been told to use this documentation, use the current [8.1.2 Deployment Guide](#)

To configure the SIP Server application and SIP switch DNs:

1. On the SIP switch that is associated with the SIP Server, in the Options > TServer section, create a DN of type VoIP Service (VOIPDN = 9999, for example) to point to the Resource Manager IP address and port, and configure these options:
  - contact = <Resource Manager IP:Port>
  - service-type = voicemail
2. To use the Feature Server administrative web application to configure and administer your dial plan, on the SIP switch that is associated with the SIP Server, create a VoIP Service DN named mixed-dialplan and configure these options:
  - service-type = extended
  - url = http://<FS Node>:8800/

For n+1 High Availability (HA), add the following parameters:

  - url-1 = http://<FS Node2>:8800/
  - url-2 = http://<FS Node3>:8800/
  - url-<n> = http://<FS Node\_N>:8800/
3. To use your existing SIP Server dial plan to administer your dial plan, create a DN of type VoIP Service named standalone-dialplan. Specify one or more dial plan rules. See the Dial-Plan Rule section in the [Framework 8.1 SIP Server Deployment Guide](#). The following example directs all calls to 2001 to be forwarded to voicemail; 9999 refers to the voicemail VoIP DN configured on the switch. If you use a different number, change the dial plan accordingly.
  - service-type = dial-plan
  - rulename = "2001=>2001;onbusy=9999;ondnd=9999;ontimeout=9999;timeout=1"
4. In the SIP Server Application object, on the Options > TServer tab, configure these options:
  - dial-plan = mixed-dialplan or standalone-dialplan (the name of the dial-plan created above)
  - mwi-implicit-notify = true
  - subscription-event-allowed = "\*"
5. On the SIP switch that is associated with the SIP Server, in the Options > TServer section, create a DN of the type Extension and configure these options:
  - contact = "\*"
    - gvm\_mailbox = <mailbox ID>

| - | Configure for voicemail.=

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To configure Feature Server for voicemail:

1. In Genesys Administrator, in the `rm` section of the Options tab of the Resource Manager (RM) application, set `sip-header-for-dnis` to `request-uri`.
2. Create a resource group of type `gateway` between SIP Server and RM.
3. Create a resource group of type `Media control platform` between MCP and RM.
4. Create an IVR profile using `Define New IVR Profile` and create the following options:
  - `service-type`, value `voicexml`
  - `initial-page-url`, value `[http: https:]/<Feature Server1 IP address>:8080/fs/`
  - (N+1 HA only) `alternatevoicexml`, value `[http: https:]/<Feature Server2 IP address>:8080/fs/`
5. Create a DID group and add the IVR profile created in the step above:
  - Add a DID with the same value as the one set for the VoIP DN option in the previous step. The recommended option and value are `service-type` and `voicexml`.
6. To configure external voicemail deposit, configure a URS strategy that considers the DN/Agent dial-plan settings for business calls that land at a route point. In TRouteCall to SIP Server, the strategy must respond to the value `full` with `UseDialPlan`.
7. Configure a URS strategy to retrieve an individual or group voicemail. This strategy must respond with TRouteCall to the P-Alcatel-CSBU header.

For example, on this routing point the strategy should generate the following TRouteCall:

```
message RequestRouteCall
AttributeThisDN '1519'
AttributeConnID 010a02020db9f03c
AttributeOtherDN 'gcti::voicemail'
AttributeLocation
AttributeExtensions [152] 00 04 00 00..
'SIP_HEADERS' 'P-Alcatel-CSBU'
'P-Alcatel-CSBU' 'call_condition=localdirect;categparty=internal;rd=unconditional'
AttributeDNIS '1519'
AttributeRouteType 1 (RouteTypeDefault)
```

## Partial dial plan processing

For voicemail deposit using the URS strategy, the destination of TRouteCall needs to be provided as a special voicemail forwarding number "gcti::voicemail". If the customer wants to use the regular forwarding number (9999, for example) instead of gcti::voicemail then the number conversion dial plan (dial-plan-rule-1=9999=>gcti::voicemail) does not work by default. The following needs to be configured to support regular forwarding number.

When a call is routed to any destination using TRouteCall (business calls), then the SIP Server dial plan functionality is not activated. This dial plan restriction is configurable through an AttributeExtension of TRouteCall "UseDialPlan = partial". The possible values of the UseDialPlan extension are: UseDialPlan = partial/full/false (default=false for internal dial plan functionality)

- If false, the call is routed directly to the target mentioned in the TRouteCall.
- If partial, SIP Server processes the above dial plan, and requests Feature Server to apply number translation and authorization check on target.
- If full, Feature Server has full authority to perform number translation, authorization, and call forwarding.

-| Configure for group voicemail.=

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To configure group voicemail:

1. Create a new agent group named Agent\_Group\_1 under the Agent Groups folder.
2. Add a TServer section on the Annex tab of the new Agent\_Group\_1 group.
3. Create a gvm\_mailbox option with the value set to 2200 (the mailbox number) in the T-Server section.
4. Add agents to the Agent\_Group\_1 group.
5. Log on to the SIP Feature Server administration page at `http://<Feature Server IP address>:8080/fs/admin`.
6. Search for any user added to the Agent\_Group\_1 group. You can see that the group mailbox is associated with that user.
7. Create a routing point (for example, 5000).
8. Configure a URS strategy to deposit a group voicemail when none of the Agent group members answer the call. This strategy must respond with TRouteCall to gcti::voicemail.
  - a. Configure the UseDialPlan Extensions attribute with its value set to false.
  - b. Configure the gvm\_mailbox with its value set to the group mailbox number.



These configurations ensure that SIP Server does not use the SIP Feature Server dial plan and builds an INVITE to the mailbox number provided by the strategy.

9. Load the routing point with the above strategy.
10. Make a call to the route point loaded with this strategy. If no agent in the group answers, the call reaches the group mailbox (2200) after timeout.

|</multistep>

# Configuration options

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Set these configuration options in Genesys Administrator, in the Options tab of the SIP Feature Server Application object.

### Cassandra section

Option	Values (default value in bold)	Description
logFile	<b>cassandra.log</b>	The log file name; applies only to Cassandra logs.
logLevel	<b>error</b> , none, debug, info, warning, critical	The log level; applies only to Cassandra logs.
maxFileSize	<b>20000000</b>	The threshold, in bytes, after which a new log file is created; applies only to dial plan logs.
maxFileCount	<b>2</b>	The number of log files kept at one time; applies only to Cassandra logs.
replicationStrategyClassName	<b>SimpleStrategy</b> , NetworkTopologyStrategy	<p>Specifies the strategy used in distributing data to Cassandra nodes:</p> <ul style="list-style-type: none"><li>In single-data center installations, SimpleStrategy uses the value of <i>replicationOptions</i> to determine the number of Cassandra nodes to which the application distributes data.</li><li>In multi-data center installations, NetworkTopologyStrategy uses the value of <i>replicationOptions</i> and the values entered in the file <b>resources/cassandra-topology.properties</b> to determine the identity and number of Cassandra nodes to which the application distributes</li></ul>

Option	Values (default value in bold)	Description
		data.
replicationOptions	<b>replication_factor=2</b> , replication_factor=3, replication_factor=4, ... <b>or</b> <i>data center 1 name=number of replication nodes, data center 2 name=number of replication nodes, ...</i>	Specifies the number of Cassandra nodes used in the replication scheme, depending on the value of <i>replicationStrategyClassName</i> : <ul style="list-style-type: none"> <li>When <i>replicationStrategyClassName</i> is set to <i>SimpleStrategy</i>, set any of the first group of values to specify the number of Cassandra nodes used in the replication scheme.</li> <li>When <i>replicationStrategyClassName</i> is set to <i>NetworkTopologyStrategy</i>, set any one of the second group of values to specify the data center name and the number of Cassandra nodes used in the replication scheme for each data center.</li> </ul>

### Cluster section

Option	Values (default value in bold)	Description
master	<b>false</b> , true	When set to <i>true</i> , this option designates this Feature Server as the server that coordinates the synchronization of real-time non-switch data (such as people, agents, and agent groups) from Configuration Server. <b>Important:</b> Designate only one server as master.
confsync	<b>false</b> , true	When set to <i>true</i> in a non-master Feature Server, this option enables the server to handle the synchronization of real-time non-switch data with Configuration Server whenever the master server is not operating.

### dialplan section

Except for *active*, which indicates whether you are using the Feature Server dial plan or the SIP Server dial plan, these options apply only if you are using the Feature Server dial plan.

Option	Values (default value in bold)	Description
active	<b>true</b> , false	Determines whether Feature Server starts and displays the Feature Server dial plan. Setting the option to false disables the Feature Server dial plan and hides dial plan-related settings in the Feature Server Administration and User web applications. Value change requires restart.
heartbeatFailInterval	<b>60</b>	The timeout, in seconds, between heartbeat queries from Feature Server to the dial plan when the previous request failed.
heartbeatOkInterval	<b>300</b>	The timeout, in seconds, between heartbeat queries from Feature Server to the dial plan when the previous request succeeded.
logFile	<b>none</b> , <dial plan script working directory>, <another directory>	The log file location on the Feature Server host; applies only to dial plan logs.
logLevel	<b>none</b> , debug, info, warning, error, critical	The log level; applies only to dial plan logs.
maxFileSize	<b>1000000</b>	The threshold, in bytes, after which a new log file is created; applies only to dial plan logs.
maxFileCount	<b>2</b>	The number of log files kept at one time; applies only to dial plan logs.
port	<b>8800</b> , <other valid http port>	The port specific to the dial plan; must differ from the Feature Server http port.

## Log section

Feature Server uses standard Genesys logging, with the following exceptions to the default values. For details of all logging options, see [Common Configuration Options log section](#).

Option	Values (default value in bold)	Description
internal	<b>Error</b> , Info, Debug	The PSDK logging level; events appear in the same log file.
verbose	<b>trace</b> , all, debug, interaction, standard, none	The verbose logging level.

## Monitoring section

Option	Values (default value in bold)	Description
active	<b>true</b> , false	Determines whether Feature Server collects environment statistics for monitoring. When set to true, the option enables Feature Server

Option	Values (default value in bold)	Description
		environment monitoring to occur even when the monitoring interface is unavailable. To conserve disk space, set the option to false if you are not using environment monitoring.
logFile	<b>monitoring.log</b> , <string>.log	The log file name in the Feature Server application directory; applies only to monitoring logs.
logLevel	<b>debug</b> , info, warning, error, critical	The log level; applies only to monitoring logs.
maxFileSize	<b>1000000</b>	The threshold, in bytes, after which a new log file is created; applies only to monitoring logs.
maxFileCount	<b>2</b>	The number of log files kept at one time; applies only to monitoring logs.

### VoicemailServer section (Application object)

Set these options in the VoicemailServer section on the Options tab of the SIP Feature Server Application object.

Option	Values (default value in bold)	Description
advance-audio-control	<b>true</b> , false	Displays advanced audio controls (forward, rewind, pause, and volume) to the web application user. Overrides the value set at the switch level.
language	< <b>server locale</b> >, de, en-GB, en-US, es, es-LA, fr, it, ja, pt, ru, zh-CN	The language used for voicemail prompts; overrides the language set at the switch level, and overridden by the language set for a mailbox.
locale	<b>en-US</b> , other locale strings	Specifies the default locale for the Telephone User Interface (TUI).
security-account-lockout-duration	<b>10</b> , zero or any positive integer	Specifies the time interval, in minutes, that Feature Server waits before unlocking an account that has been locked because of incorrect login entries; the user is locked out after four failed login attempts. A value of 0 (zero) means that only an administrator can unlock the mailbox.
security-password-check-internal-call	<b>true</b> , false	Specifies whether the password check request is played when a user dials the voicemail system from an internal phone.
security-password-length-min	<b>4</b> , any positive integer	Specifies the minimum length of

Option	Values (default value in bold)	Description
		the password digits.
time-zone	<b>&lt;server time zone&gt;</b> , any valid time zone (see <a href="http://joda-time.sourceforge.net/timezones.html">http://joda-time.sourceforge.net/timezones.html</a> )	Specifies the default time zone for voicemail messages (applies to the entire application, and overrides any value set at the switch); overridden when the time zone of the user or mailbox is set to a value other than the default.
voice-can-deposit-during-extended-absence	<b>true</b> , false	When set to <b>true</b> , enables the voicemail deposit after the absence greeting is played.
voice-enrollment-enabled	<b>true</b> , false	When set to <b>true</b> , enables the voicemail enrollment.
voice-greeting-extended-max-duration	<b>24</b> , any positive integer	Specifies the length of time, in seconds, allowed for the user to record their extended greeting; after the period ends, a prompt asks the user to confirm their recorded message.
voice-greeting-personal-max-duration	<b>12</b> , any positive integer	Specifies the length of time, in seconds, allowed for the user to record their personal greeting; after the period ends, a prompt asks the user to confirm their recorded message.
voice-message-max-duration	<b>10</b> , any positive integer	Specifies, in seconds, the maximum message length.
voice-message-priority-enabled	<b>false</b> , true	Enables callers to specify whether the voicemail they are leaving is urgent or normal priority. When set to <b>true</b> it plays the message priority selection menu to choose between normal or urgent delivery. When <b>false</b> , messages are all sent with normal priority.
voice-mailbox-message-count	<b>10</b> , any positive integer	Specifies the maximum number of voicemail messages per mailbox.
voicemail-optout-destination	<b>empty (means that the feature is disabled)</b> , any valid destination URI in one of these formats: <ul style="list-style-type: none"> <li>• sip:NNNN@&lt;IP address&gt;:&lt;port&gt;</li> <li>• sips:NNNN@&lt;IP address&gt;:&lt;port&gt;</li> <li>• sip:&lt;user name&gt;@&lt;host&gt;:&lt;port&gt;</li> <li>• sips:&lt;user</li> </ul>	When set, enables a caller to transfer out of voicemail to the specified destination at any moment during a call; overridden when the optout destination of the switch has a value or when the Optout Phone field of the Mailbox settings page has a value. See <a href="#">Provisioning mailboxes</a> .

Option	Values (default value in bold)	Description
	name>@<host>:<port>	

### VoicemailServer section (Switch)

Set these options in the VoicemailServer section on the Options tab of the SIP Switch object, not in the Application object.

Option	Values (default value in bold)	Description
advance-audio-control	<b>true</b> , false	Displays advanced audio controls (forward, rewind, pause, and volume) to the web application user. Overridden by the value set at the application level.
language	< <b>server locale</b> >, de, en-GB, en-US, es, es-LA, fr, it, ja, pt, ru, zh-CN	The language used for voicemail prompts; overridden by the language set at the application or mailbox level.
play-disclaimer	<b>true</b> , false	When true, enables the system to play a recorded disclaimer; see <a href="#">Provisioning mailboxes</a> .
time-zone	< <b>server time zone</b> >, any valid time zone (see <a href="http://joda-time.sourceforge.net/timezones.html">http://joda-time.sourceforge.net/timezones.html</a> )	Specifies the default time zone for voicemail messages (applies to the switch); overridden when the time zone of the user, mailbox, or application is set to a value other than the default.
voicemail-optout-destination	<b>empty (means that the feature is disabled)</b> , any valid destination URI in one of these formats: <ul style="list-style-type: none"> <li>sip:NNNN@&lt;IP address&gt;:&lt;port&gt;</li> <li>sips:NNNN@&lt;IP address&gt;:&lt;port&gt;</li> <li>sip:&lt;user name&gt;@&lt;host&gt;:&lt;port&gt;</li> <li>sips:&lt;user name&gt;@&lt;host&gt;:&lt;port&gt;</li> </ul>	When set, enables a caller to transfer out of voicemail to the specified destination at any moment during a call; overrides the optout destination value set at the application level, but overridden when the Optout Phone field of the Mailbox settings page has a value. See <a href="#">Provisioning mailboxes</a> .