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SIP Cluster Solution Guide

Configuring SIP Feature Server

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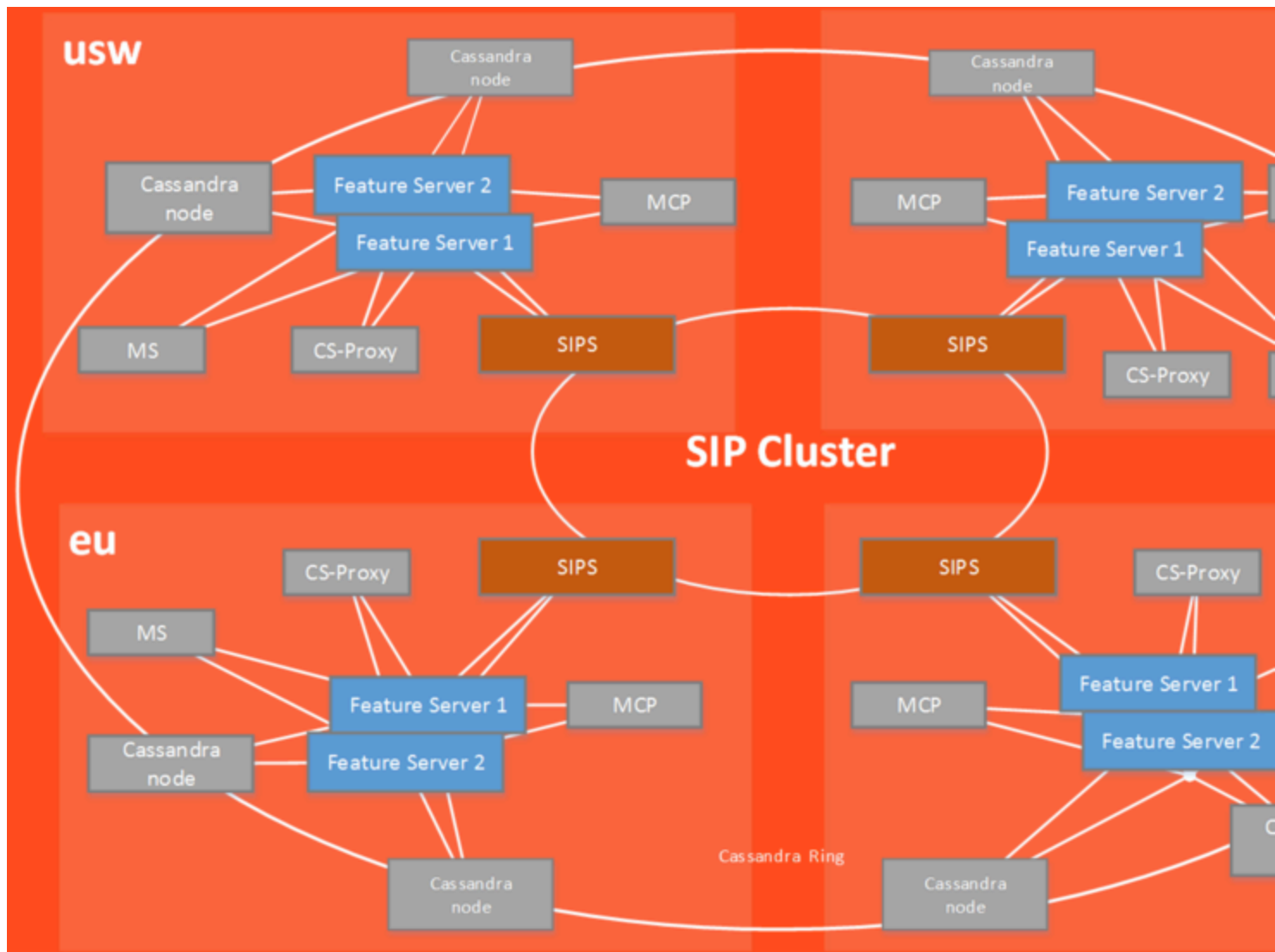
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Configuring SIP Feature Server

Genesys SIP Feature Server integrates with Genesys SIP Cluster to provide a SIP-based voicemail and SIP feature manager for Genesys contact centers and enterprise environments. Callers leave voicemail, and users retrieve and manage that voicemail. Administrators manage users, devices, voicemail, and call disposition (the dial plan). A distributed architecture enables resiliency and enhances performance.

Standard Feature Server deployment includes a High Availability (HA) installation with the SIP Server in cluster mode and at least one active-active pair of Feature Servers should be deployed to serve one SIP Cluster node. Refer to [Deployment options](#) for more details.

You can deploy SIP Feature Server across multiple data centers, in an active-active configuration. The following image shows Feature Server deployment in External Cassandra cluster mode.



Feature Server in SIP Cluster Environment

Configuration and provisioning

Initial program configuration occurs primarily in Genesys Administrator (GA). You can create users, DNSs, and mailboxes only in GA, but can use the Feature Server plug-in for Genesys Administrator Extension (GAX) to provision them with devices, voicemail, and call settings.

Dial plan

SIP Feature Server enables the configuration of the dial plan, a highly configurable set of call disposition patterns. You configure the dial plan using either of two methods:

- the Feature Server plug-in for Genesys Administrator Extension (GAX)
- the existing SIP Server methodology, using GA

Voicemail

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

Agents and other users can use the Feature Server GAX plug-in and the Telephone User Interface (TUI) to manage their personal and group voice mailboxes and call settings.

Data management

SIP Feature Server uses [Apache Cassandra](#) data clusters to replicate data across the environment, achieving resiliency and high availability. Refer to [Data management](#) for more details. It is recommended to use External Cassandra cluster in the SIP Cluster environment.

Feature Server configuration options

See [Configuration options](#) for information about the SIP Feature Server configuration options.

Deploying SIP Feature Server

This section explains how to install and configure the SIP Feature Server instances for SIP Cluster.

Planning and pre-installation

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 Genesys and third-party components.
- Review the current [known issues and recommendations](#).

Deployment steps

Complete these steps to install and configure the SIP Feature Server instances for SIP Cluster.

- [Configure SIP Feature Server applications](#)

Important

For deployments using employee ID as agent login code, set the use-employee-login option in the **[VoicemailServer]** section to **TController** on all Feature Server Application objects.

- [Deploy a co-located/external Cassandra cluster](#)

- [Configure SIP Feature Server for a co-located/external Cassandra cluster](#)

Important

It is recommended to use External Cassandra cluster.

- [Start and verify SIP Feature Server](#)
- [Configure SIP Server for Feature Server](#)
- [Configure voicemail](#)
- [Configure Message Waiting Indicator \(MWI\)](#)
- [Provision users](#)
- [Provision DNs](#)
- [Provision mailboxes](#)

Configure voicemail

To configure voicemail in Cluster mode:

1. In Genesys Administrator, navigate to the `rm` section of the Options tab of the Resource Manager (RM) application, and set the option value of the `sip-header-for-dnis` option to `request-uri`.
2. Create a resource group of type `Media control platform` to make the Media Control Platform (MCP) instances in the cluster available for RM instances.
3. Create a resource group of type `gateway` between SIP Server and each RM instance in the cluster.
4. Under the SIP Cluster Switch, create a unique DN of type **Voice over IP Service** with the same name as the configured Direct Inward Dialing (DID). Create two DNs (DID) to Feature Server HA pair. The voicemail DNs will be used for configuring voicemail.
 - In the Annex > **TServer** section, configure the following options:
 - **contact** = `::msml`
 - **geo-location** = <A string identifying the data center to which this DN belongs>
 - **service-type** = `voicemail`
5. Each SIP node will be connected to one Feature Server HA pair having two voicemail IVR profiles per Feature Server HA pair. Under the Voice Platform tab, create IVR profile for Feature Server instances in the cluster as shown below:
 - `service-type,value voicexml`
For Feature Server 1
 - `initial-page-url,value [http: https://FQDN1 or //FS1 IP address:port/fs`
FQDN1 is the FQDN you created while configuring Feature Server applications, if your environment includes more than two Feature Server instances per SIP switch.
 - `alternatevoicexml,value [http: https://FQDN2 or //FS2 IP address:port/fs`
FS2 IP address is the IP address of the "extra" Feature Server instance that is not included in *FQDN1*.

For Feature Server 2

- `initial-page-url`, value [`http: https://FQDN2 or //FS2 IP address:port/fs`]
- `alternatevoicexml`, value [`http: https://FQDN1 or //FS1 IP address:port/fs`]

6. For each IVR profile created above, configure a unique DID.

7. Configure a GVP DID Group by specifying the following parameter in the Annex tab of the tenant:

- Under the **[gvp.dn-groups]** section configure the <DNs in the group> value for each FS application in the Feature Server HA pair, where <DNs in the group> are configured DNSs under a Switch with which the voicemail SIP Server Application is associated.

Option	Value
FS1 application name	DID 1
FS2 application name	DID 2

- Under the **[gvp.dn-group-assignments]** section, configure the <DBID of the Voicemail IVR Profile> value for each FS application in the Feature Server HA pair.

Option	Value
FS1 application name	DBID of IVR profile created for Feature Server 1
FS2 application name	DBID of IVR profile created for Feature Server 2

Configure Message Waiting Indicator (MWI)

To configure Feature Server to issue a Message Waiting Indicator (MWI):

1. In your **SIP Proxy** application, select `Options > sipproxy > feature-server-address`. Configure the Feature Server IP address with port 5160.
2. To use a different SIP port from the default SIP port on the SIP Feature Server, create a `[sip]` section on the SIP Feature Server and add a **localport** option to assign the SIP port.

Notes:

- The SIP MWI is supported only for individual mailboxes.
- For group mailboxes, only the T-Library MWI is supported.
- Feature Server does not accept subscriptions for device numbers (except where the mailbox number matches the device number).
- To support SIP MWI notification, the SIP endpoints must be configured to subscribe to the voice mailbox number directly.
- The subscription should be sent to SIP Proxy—for example, `mailbox number@SIP Proxy IP`.

Configuring dial-plan in Feature Server

- See **Dial plan** for information about creating partitions and calling profiles, and editing dial plan settings.

- [Configuring dial-plan DNs](#)
- [Calling profiles and its associations](#)
- [Sample basic dial-plan configuration](#)

Configuring dial-plan DNs

The SIP Cluster Switch might contain one or more dial-plan DNs. A typical SIP Cluster deployment will have one dial-plan DN configured per geo-location.

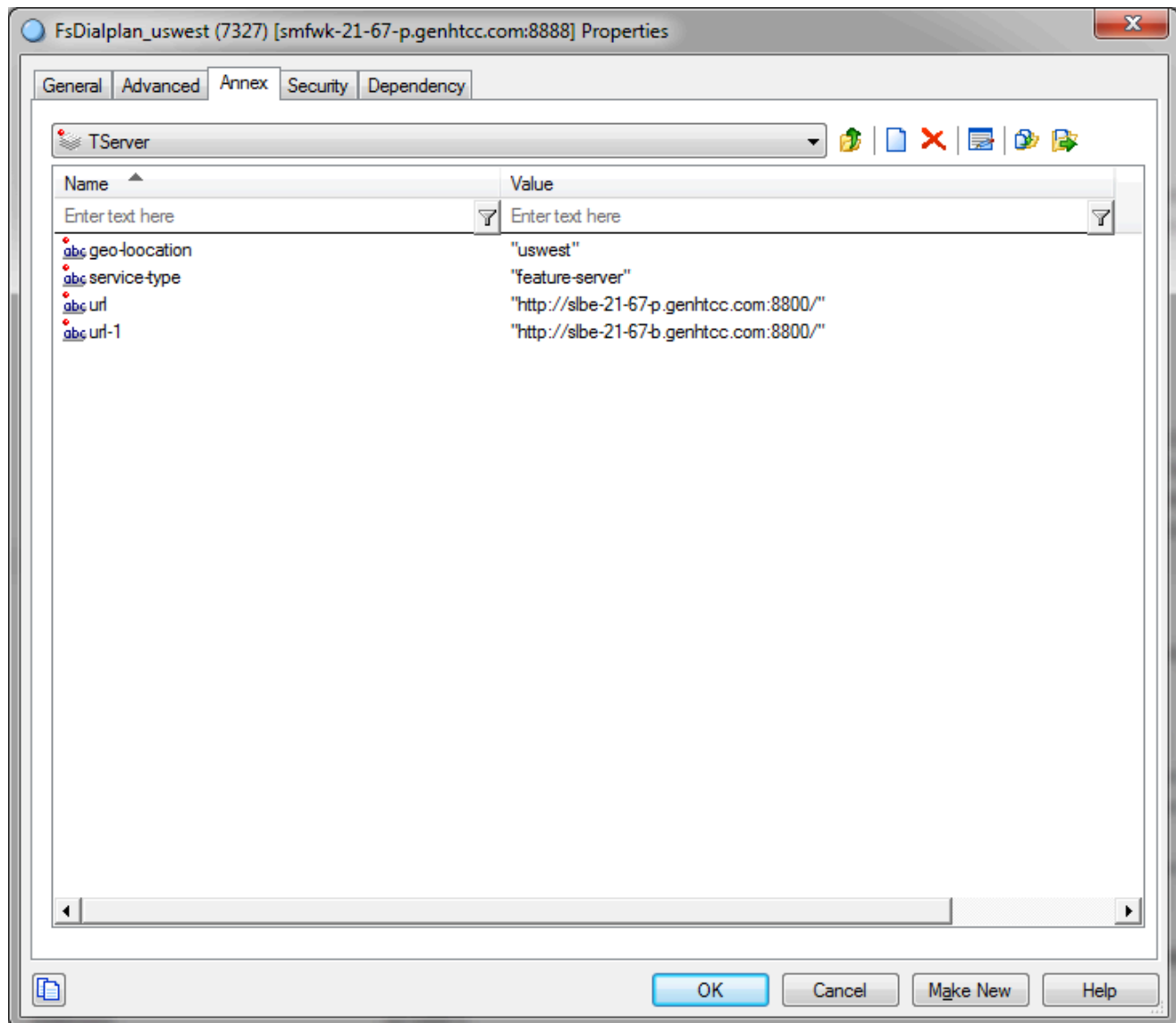
1. Under the SIP Cluster Switch, create a DN of type **Voice over IP Service** named `dial-plan-<datacenter>`, for each region. For example, **dial-plan-sfo**.
2. In the **Annex > TServer** section, configure the following mandatory options:

Name	Value	Notes
geo-location		A string identifying the data center to which this DN belongs.
service-type	feature-server	
url	A URL of a local SIP Feature Server	Set this option to the URL for the local SIP Feature Server in this data center. If there is more than one SIP Feature Server in the data center, create as many url options as needed to accommodate all their addresses in one DN. All consecutive options must be named url-<n> , depending on the number of deployed Feature Server instances (see url-1 below).
url-1	A URL of the second SIP Feature Server	Set this option to the URL for the second SIP Feature Server in this data center. All consecutive options must be named url-<n> , depending on the number of deployed Feature Server instances.
url-N	A URL of the Nth SIP Feature Server	Set this option to the URL for the N+1th SIP Feature Server in this data center. All consecutive options must be named url-<n> , depending on the number of deployed Feature Server instances.

Important

For each SIP Feature Server, the pool of connection of URLs depends on the number of SIP Feature Servers available.

Here is a sample screenshot:



Calling profiles and its associations

Calling profile can be assigned to the following contact center objects:

- **Switch**—Generic rules that are applicable for all calls in the cluster deployment.
- **User**—Specific set of rules that need to be configured for a set of users.
- **Extension**—Specific set of rules that need to be configured for a set of extensions.
- **Trunk**—Customized dial-plan for a particular trunk because external calling profile is used for all inbound calls.
- **Softswitch**—Customized dial-plan for a particular softswitch. For example, 10-digit numbers must be converted into E.164 to match PSTN carrier requirement.

- **Trunk Group**—Calling profile used for predictive calls.
- **Route Point**—Calling profile used only for predictive calls. This calling profile does not affect call routing.

Example of minimum required basic dial-plan configuration

The following table explains the mandatory configuration to use dial-plan.

Name	Description	Rule
Basic	Used for digit translation.	.=>\${DIGITS}
Voicemail	Used for voicemail access number to gcti::voicemail translation.	5555=>gcti::voicemail

Partitions

Basic Partitions

"Basic" Partition makes no number translation (Rules: **.=>\${DIGITS}**). It is active 24x7.

SIP Voicemail & Call Settings

Home / Partitions / Partition Properties: basic

General

Name *

Basic

☒ Active

☐ Block

Time Zone

Not Set

Time Start

00:00

Time End

End of the day

Days of Week

Select Some Options

Rules *

.*=>\${DIGITS}

Save changes Delete Cancel

Voicemail Partition

"Voicemail" Partition makes 5555 Voicemail access number (Rules: **5555=>gcti::voicemail**). It is active 24x7.

The screenshot shows the 'SIP Voicemail & Call Settings' page in the GAX System Dashboard. The navigation bar at the top includes 'GAX', 'System Dashboard', 'Agents', 'Configuration', and 'Administration'. The breadcrumb trail is 'Home / Partitions / Partition Properties: Voicemail'. The 'General' tab is selected, showing the following fields:

- Name ***: Text input containing 'Voicemail'.
- Active**: Checked checkbox.
- Block**: Unchecked checkbox.
- Time Zone**: Dropdown menu showing 'Not Set'.
- Time Start**: Text input containing '00:00'.
- Time End**: Text input containing 'End of the day'.
- Days of Week**: Text input containing 'Select Some Options'.
- Rules ***: Text input containing '5555=>gcti::voicemail'.

At the bottom of the form are three buttons: 'Save changes' (light blue), 'Delete' (dark blue), and 'Cancel' (light grey).

Calling Profile

Create Calling Profile

Create Calling Profile System adding Basic and Voicemail Partitions.

The screenshot shows the 'SIP Voicemail & Call Settings' page in the GAX system dashboard. The page has a dark header with navigation links: 'GAX', 'System Dashboard', 'Agents', 'Configuration', and 'Administration'. The user is logged in as 'default'. The breadcrumb trail is 'Home / Calling Profiles / Calling Profile Properties: System'. The main content area is titled 'SIP Voicemail & Call Settings' and has a 'General' tab selected. A text field labeled 'Name' contains the value 'System'. Below this is a 'Partitions' table with columns for selection, index, movement, name, active status, block status, and time zone. Two partitions are listed: 'Basic' and 'Voicemail', both of which are active. At the bottom right of the table is an 'Add' button. At the bottom of the form are 'Save changes' and 'Cancel' buttons.

				Name	Active	Block	Time Z
<input type="checkbox"/>	1	↑	↓	Basic	✓		
<input type="checkbox"/>	2	↑	↓	Voicemail	✓		

Assign Calling Profile

Assign "System" Calling Profile to SIP Cluster Switch as both **System Internal Calling Profile** and **External Caller Calling Profile**.

GAXSystem DashboardAgentsConfigurationAdministrationdefault?

SIP Voicemail & Call Settings

Home / Dial Plan Properties: sa_demo_02_switch

GeneralCall Settings

System Internal Calling Profile

System

External Caller Calling Profile

System

Not Set

System

dummy

+

Save changesCancel