Genesys 8.1

Genesys Voicemail Solution Configuration Note

Version 1.1

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Glossary & Acronyms

Term	Definition
FS	Feature Server
VM	Voicemail
VMS	Voicemail Server
MS	Media Server
MCP	Media Control Platform
RM	Resource Manager
GVP	Genesys Voice Platform
VXML	Voice XML
IWS	Interaction work space
XS	Extended Services
VMB	Voice Mail Box
MWI	Message Waiting Indicator
TUI	Telephony User Interface (aka Voice User Interface VUI)
DN	Directory Number

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1 Introduction

This whitepaper is intended to capture the end-to-end configuration and deployment of the Genesys Voicemail solution.

Genesys Voicemail solution provides three basic capabilities for a user: Deposit, Retrieval, and MWI.
Deposit allows a caller to leave messages for any user on the Genesys Voicemail System. Retrieval encompasses all functionality related to accessing a caller's voicemail messages. MWI functionality provides for the notification of voicemail messages. The Genesys solution eases the administration and maintenance of these voicemail features through a powerful web-based user interface, in addition to the normal voice user interface.

1.1 *Architecture*

The Genesys Voicemail solution is based on three key components: SIP Server, Feature Server, and Media Server. Figure 1 captures a high-level view of the Genesys Voicemail solution with these key components.

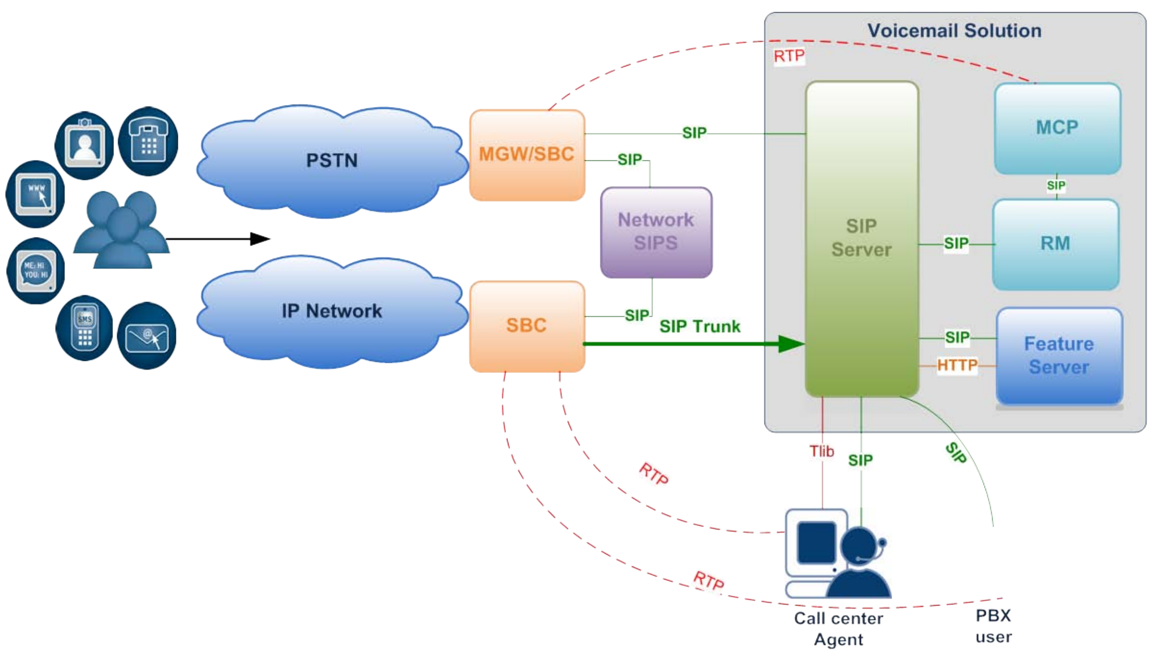


Figure 1 -Genesys Voicemail Solution - overview

1.1.1 SIP Server

SIP Server is a combined T-Server and call-switching component, in which the call-switching element functions as a SIP (Session Initiation Protocol) Back-to-Back User Agent (B2BUA). In concrete terms, this means that call switching and control is performed by Genesys—no third-party PBX or ACD system is required. A call's audio signal and its associated data travel on a single network, which eliminates the problems associated with synchronizing separate voice and data networks. Because SIP Server supports the Internet Engineering Task Force (IETF) SIP RFC 3261 suite, it is compatible with the most popular SIP-compatible, off-the-shelf hardware or software. SIP Server can operate with or without a third-party soft-switch. Genesys SIP Server gives the entire Genesys line of products access to SIP networks, offering a standards-based, platform-independent way to take advantage of the benefits of voice/data convergence

1.1.2 Feature Server

Genesys SIP Feature Server unifies SIP, Voice, and web-based technologies to provide key functionality such as configuration/provisioning of DNs, users and mailboxes, message-waiting indication (MWI) for endpoints, definition of dial-plans, deposit/retrieval of voicemail, and reporting/monitoring of Genesys components. It is an extension of the Genesys SIP Voicemail Server solution and employs the Cassandra DB to provide high availability and scalability of voicemail data across multiple data centers.

Feature Server also provides a web-based user interface for administrators and users. Administrators can perform such functions as creation of partitions/calling profiles, DN/Mailboxes, association of DN/mailboxes to users and maintenance of mailboxes. Users can access voicemail messages, manage forwarding settings and maintain their mailboxes through the user interface. The voice or telephony UI provides a seamless voice experience for access and deposit of voicemail messages, for both internal and external callers.

1.1.3 Media Server

When integrated with SIP Server, the Genesys Media Server provides Real-Time Protocol (RTP) streaming for a variety of media services like treatments, IVR application (VXML), conferences, call parking, call recording, call monitoring and Call Progress Detection (CPD), and it also provides support for voicemail when it is integrated with Feature Server. It provides support for both NETANN- and MSML-based media services.

2 Deployment of Genesys Voicemail solution

SIP Server uses the dial plan to identify calls that are to be routed to Voicemail. All Voicemail requests are forwarded via Genesys Media Server to Genesys Feature Server. Genesys Media Server uses HTTP to fetch and execute VXML documents to provide voicemail functionality.

Genesys Voicemail Solution offers flexibility and scalability through a choice of three different deployment modes:

1. Standalone with internal dial plan (SIP Server uses dial plan provided by SIP Server)
2. Standalone with Feature Server dial plan (SIP Server uses Feature Server dial plan)
3. SIP Cluster (Cluster SIP Server with Feature Server dial plan)

2.1 *Standalone mode (SIP Server uses its internal dial plan) – Deployment Architecture*

Voicemail deployment in standalone architecture is shown below. A SIP Server and a Feature Server instance will be deployed in a one-to-one relationship. In this case, SIP Server uses its internal dial-plan for configuring the voicemail access/deposit criteria (like forward on busy). All SIP Servers and Feature Servers can be deployed to use a single Resource Manager. Indeed the RM which is used for Media Service (like treatments, conference, etc) can also be used for voicemail service. Each Feature Server registers with its peer SIP Server to monitor the DNs and the Agents status, this is required to provide MWI notifications to appropriate SIP/Tlib clients.

In this architecture, Feature Server sends MWI to SIP Server through the Tlib interface; SIP Server forwards the MWI to its associated Tlib and SIP Clients. Voicemail boxes can be configured for the following objects in CME:

- DNs (Extensions or ACD Positions)
- Agent Logins
- Agent Groups

In this deployment mode, the following Feature Server functionalities are available:

- Voice Mail
- Feature Server Admin and User UI
- Environment monitoring

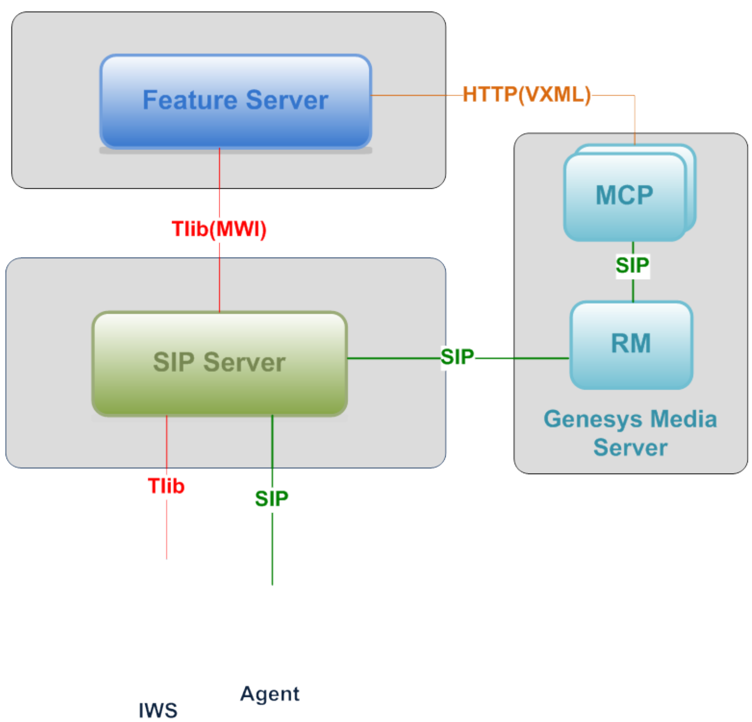


Figure 2 –Standalone mode Deployment (SIP Server dial plan)

2.2 *Standalone mode using Feature Server dial-plan – Deployment Architecture*

Voicemail deployment in standalone mode that uses the Feature Server dial-plan architecture is shown below. This deployment is very similar to the above mentioned standalone deployment [section 2.1] except that SIP Server uses Feature Server's dial-plan for configuring the voicemail access/deposit criteria (like forward on busy). Each Feature Server registers with its peer SIP Server to monitor the DNs and the Agents status, this is required to provide MWI notifications to appropriate SIP/Tlib clients.

In this architecture, Feature Server sends MWI to SIP Server through the Tlib interface; SIP Server forwards the MWI to its associated Tlib and SIP Clients. Voicemail boxes can be configured for the following objects in CME:

- DNs (Extensions or ACD Positions)
- Agent Logins
- Persons
- Agent Groups

The following functionality available on Feature Server can be used in this deployment:

- Dial Plan
- Forwarding
- Voice Mail
- Feature Server Admin and User UI
- Environment monitoring

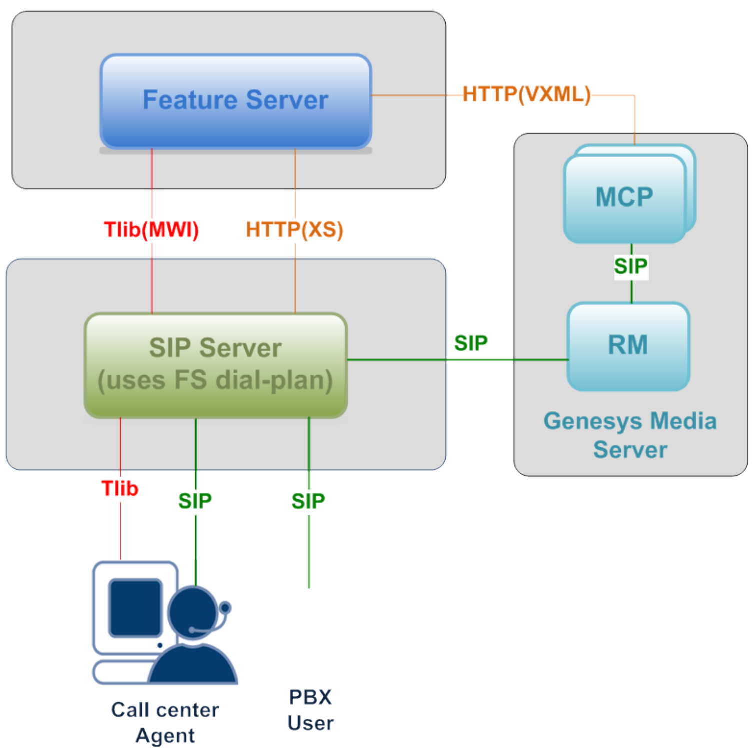


Figure 3 – Standalone mode using Feature Server dial-plan – Deployment

2.3 *SIP Cluster Mode – Deployment Architecture*

Voicemail solution for SIP cluster is captured in Figure 4. In this highly-scalable clustering architecture, a SIP Server node and a Feature Server will be deployed in a one-to-one relationship. All SIP Servers and Feature Servers can be deployed to use a single RM. Indeed the RM which is used for Media Service (like treatment, conference, etc) can also be used for voicemail service. Each Feature Server registers with its peer SIP Server to monitor the DNs and the Agents status, this is required to provide MWI notifications.

In this architecture, Feature Server sends SIP MWI to SIP Clients directly through SIP Proxy, while it sends MWI to T-Lib Clients through SIP Server.

Voicemail boxes can be configured for the following objects in SIP cluster:

- DNs (Extensions or ACD Positions)
- Persons
- Agent Groups

The voicemail boxes for DNs (Extensions or ACD Positions) and Persons can be configured in the Feature Server Administrator UI, whereas voicemail boxes for Agent Groups can be configured in CME. DNs are configured through the Feature Server Administrator UI while Persons/Agent Groups objects are configured in CME.

In this deployment mode, the following Feature Server functionalities are available:

- Dial Plan
- Forwarding
- Voice Mail
- Monitoring
- Feature Server Admin and User UI
- Environment monitoring

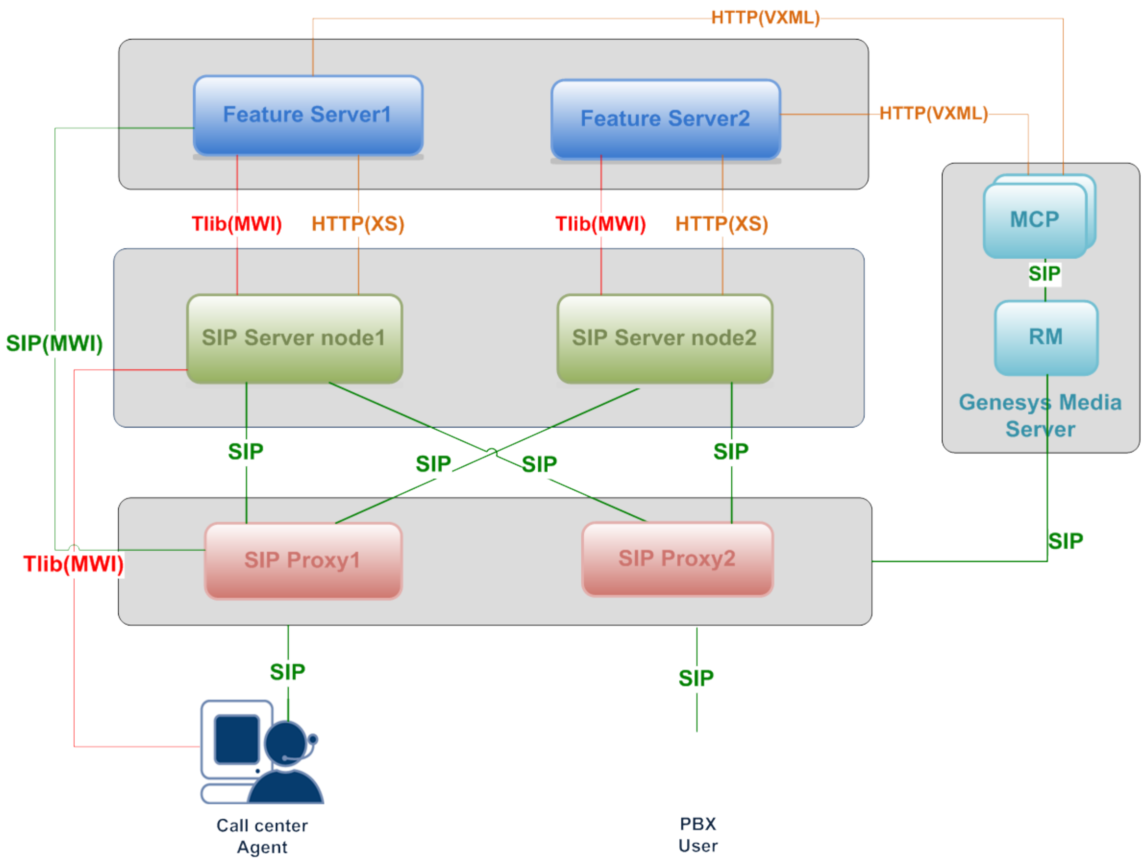


Figure 4 – SIP Cluster mode Deployment

2.4 *Standalone mode –Multi-site – Deployment Architecture*

A multi-site configuration involves deployment of two or more standalone SIP Servers sharing the same Genesys Framework (configuration database). This deployment is applicable for both the standalone deployments mentioned above [section 2.1 and section 2.2] and it is not applicable for SIP Cluster Deployment mentioned above [section 2.3]. A simple architectural view with single SIP Server and Feature Server on each site is shown below.

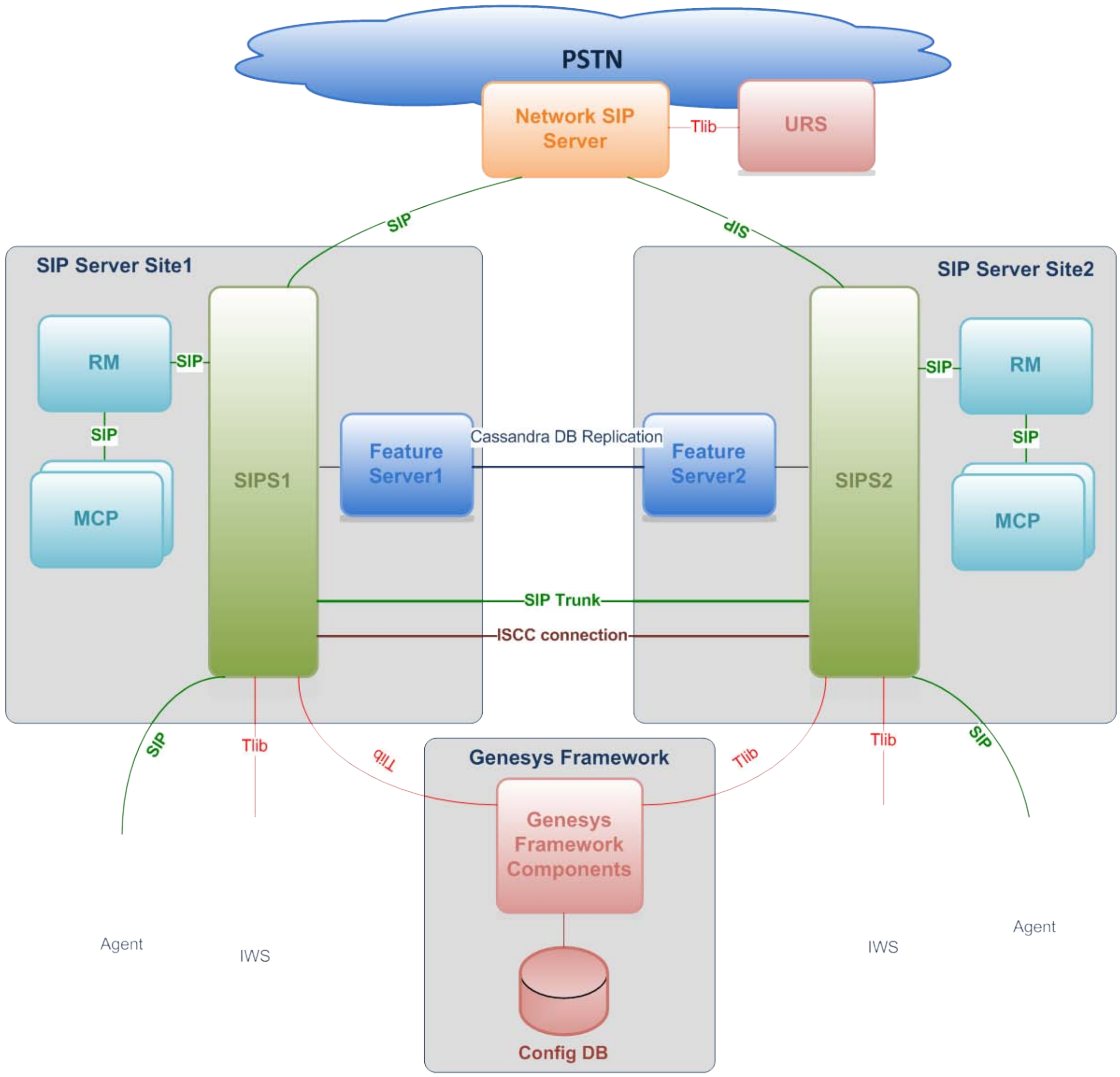


Figure 5 – Standalone mode multi-site – Deployment

Each Feature Server is connected to only one SIP Server. But all Feature Servers can be deployed to use a common Cassandra cluster which can be shared across the sites. In this case, Genesys voicemail solution serves as a global solution across sites, any user can access voicemail from any sites.
Feature Servers can also be deployed independently not to share Cassandra cluster. In this case, voicemail solution cannot serve as a global solution across sites.

Note: When TDM T-Servers and SIP Servers are deployed together in a Hybrid multi-site architecture, Genesys Voicemail cannot be used in T-Server site (because it is a pure IP-based solution).

3 Genesys Voicemail - High Availability and Business Continuity Deployments

3.1 *Genesys Voicemail High Availability – Deployment Architecture*

In the Genesys voicemail solution, the following components can be deployed in High-Availability mode:

1. SIP Server
2. Resource Manager
3. Feature Server

This is applicable for only standalone deployments [section 2.1 and section 2.2]
The following architecture describes the High Availability for Genesys voicemail solution.

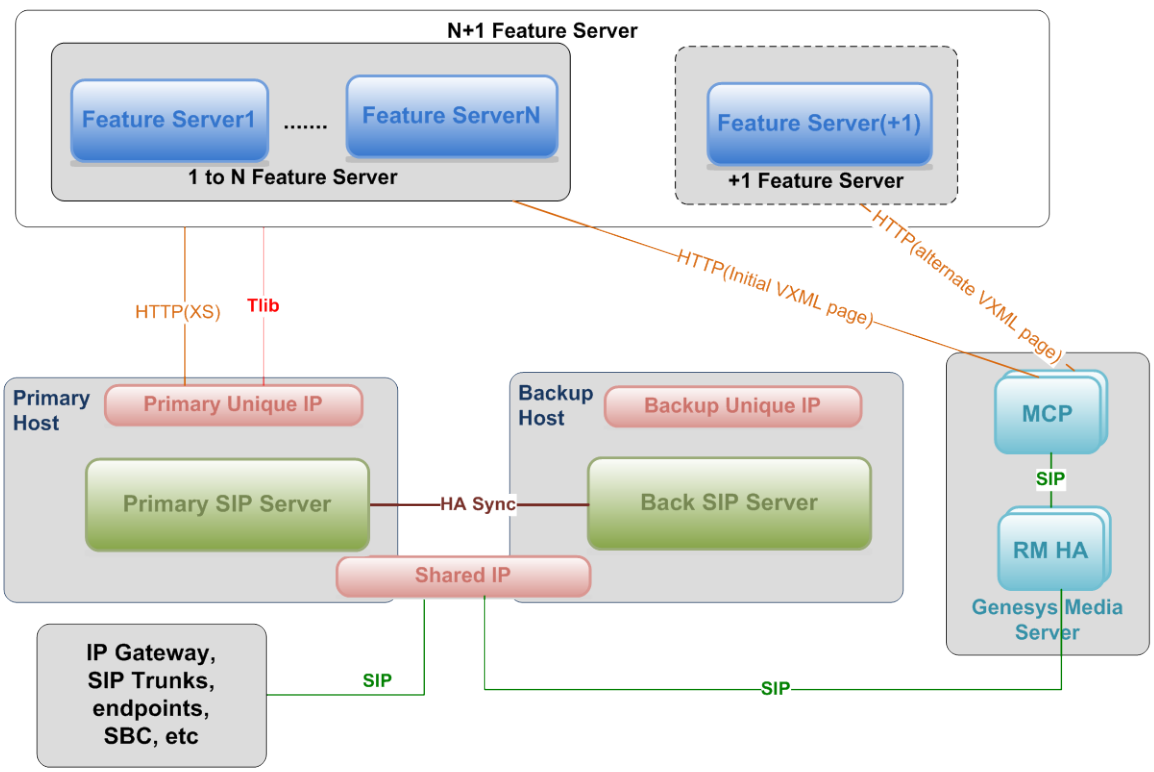


Figure 6 – High-Availability Deployment model for standalone Genesys Voicemail

3.1.1 SIP Server

SIP Servers can operate in a high-availability (HA) environment, providing redundant systems. High-availability (HA) architecture implies the existence of redundant applications: a primary and a backup. These applications are configured in such a way that if one fails; the other can take over its operations without significant loss of data or impact to business operations. The Framework Management Layer currently supports two types of redundant configurations: warm standby and hot standby.

There are several options available for setting up a high-availability SIP Server deployment. Each approach has its own benefits and drawbacks. For more information about different HA models and configurations, refer to the Framework SIP Server High-Availability Deployment Guide.

3.1.2 Resource Manager

The RM will be deployed as a coordinating cluster of nodes. It will use cluster virtual IP technology so that it appears as a single entity to SIP Server making use of GVP services. A pair of RMs can be configured in either active/standby mode or active/active mode. In active/standby mode, one RM node will process the requests while the other node will act as backup. In active/active mode, both RM nodes will process the requests simultaneously.

3.1.3 Feature Server

High Availability is provided by installing multiple Feature Servers (usually N+1 Feature Servers accessible though a FQDN or multiple URLs configuration) per SIP Server HA pair. All the Feature Server nodes share the same Cassandra database; hence if one Feature Server is down, the other Feature Servers can operate without loss of functionality.

3.1.3.1 For HTTP (VXML) interface with Media Server:

A dedicated Feature Server (+1 FS) can be deployed to act as a backup for the voicemail services using Media Server though HTTP (VXML). This backup Feature Server will process voicemail requests (VXML) when all of the N Feature Servers becomes unavailable. The GVP IVR Profile allows two URL parameters to be configured for VXML service type: “initial-page-url” and “alternatevoicexml”. GVP will try the alternate URL provided if the primary URL fails. GVP will return to the primary node as soon as it starts responding again.

The “initial-page-url” will be configured as a FQDN of N Feature Servers pool. The “alternatevoicexml” will be configured to an individual FS (+1 FS) instance that is not part of the FQDN pool.

3.1.3.2 For HTTP (XS) interface with SIP Server:

N+1 Features Servers are deployed with SIP Server in a load-balancing mode for dial-plan interface (HTTP–XS). When any of the Feature Servers becomes unavailable, SIP Server simply sends a request to another active Feature Server in the N+1 pool.

3.1.4 Sample Failover scenarios

3.1.4.1 Primary SIP Server is down or switchover occurs

If the primary SIP Server fails or switchover occurs, then the backup SIP Server will take over the Primary Server's operations without significant loss of data. It will continue to process the Tlib, HTTP (XS), and SIP traffic.

3.1.4.2 One Feature Server is down

N+1 Feature Servers can be accessible through a FQDN/multiple URLs from a SIP Server HA pair. If one Feature Server is down, the other Feature Servers in the N+1 pool can process the HTTP (XS) and Tlib operations from SIP Server without loss of functionality.
A dedicated Feature Server (+1) can be deployed to act as a backup for voicemail services using Media Server through HTTP (VXML). When all of the N Feature Servers becomes unavailable then the backup FS (+1) will process the voicemail requests (HTTP) from Media Server.

3.1.4.3 RM is down

In the case of active-standby mode, if one RM fails in the RM HA pair, then another standby RM will take over to complete operations. In the case of active-active mode, if one RM fails, then another active RM will take over the operations

3.1.4.4 MCP is down

MCPs are deployed in a farm/pool; RM monitors the health of the pool. If one MCP fails in a pool, RM will make use of another available MCP from the pool.

3.2 *Genesys Voicemail Business Continuity – Deployment Architecture*

SIP Business Continuity provides the ability for a group of agents to continue offering critical business functions to customers in the event of a loss of all Genesys components running at a particular site. The SIP Business Continuity architecture uses a synchronized, two-site deployment, where Genesys switch and server components are mirrored at each site in an active-active configuration, so that any agent can log in to either switch, at any time.

Genesys Voicemail supports business continuity deployment and it will enable Voicemail to continue serving the users to perform their critical business functions even with a loss of all Genesys components including SIP Server and Feature Server at a particular site. Feature Server will provide redundancy for accessing voicemail messages across sites and within a site, through the deployment of a Cassandra cluster. Redundant sites will have the Cassandra database replicated, to provide a consistent view of the underlying data.

In the event of a failure at one site (all Genesys components go down), agents connected to the failed site are re-logged in automatically to the surviving site. Although any active calls on the failed site are terminated at the moment of failure (including calls on the surviving site that include the failed SIP Server in the signaling path), the surviving site is able to process all new calls, with minimal impact to queue wait times. Agents will receive MWI snapshots from the surviving site and they can access their voicemail from the surviving site.

The Business Continuity deployment for Genesys voicemail solution is captured below. A standalone SIP Server can be deployed with or without Feature Server dial plan.

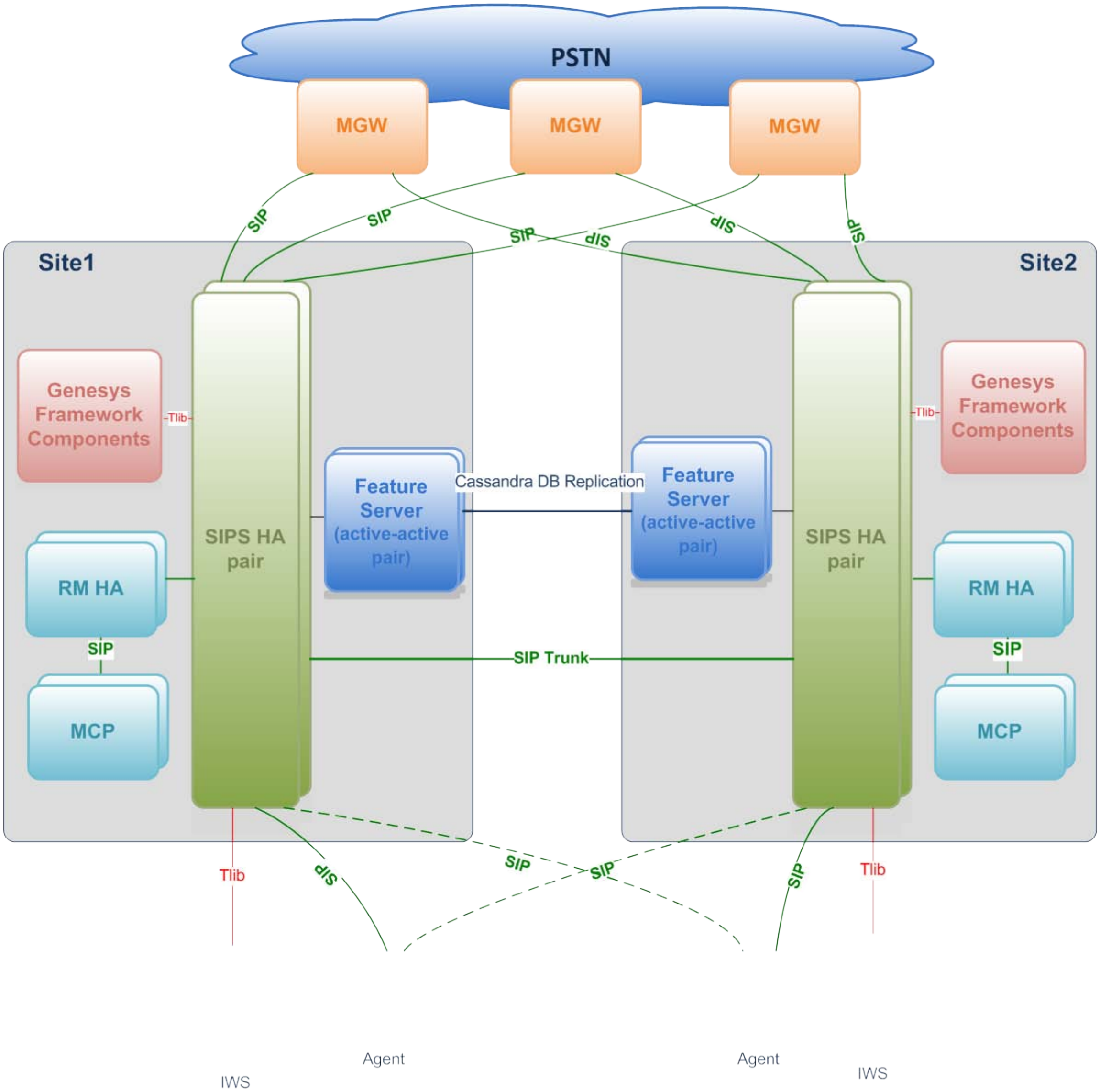


Figure 7 – Business Continuity Deployment model for standalone Genesys Voicemail

In this mode, Feature Server is deployed in an active-active configuration in each BC site. SIP Server and RM are deployed in HA configuration. Each Feature Server has an embedded Cassandra database which will be synced frequently between the sites and this ensures the real-time availability of all data in Site1 and Site2.

4 Supported Voicemail Features

Genesys Voicemail Solution provides the following basic voicemail functionalities:

- Voicemail Deposit
- Voicemail Retrieval
- Voicemail notifications (SIP and Tlib MWI notifications)
- Group Voicemail

4.1 *Voicemail Deposit*

Voicemail can be deposited for any mailboxes configured in the following objects.

- DNs (Extensions or ACD Positions)
- Agent Logins
- Persons
- Agent Groups

The following table shows the supported mailbox configurations in each deployment:

Mailbox Configuration	Standalone	Cluster	Standalone mode uses FS dial plan
Extension or ACD Position	✓	✓	✓
User(Person)	✗	✓	✓
Agent Login	✓	✗	✓
Agent Group	✓	✓	✓

When a voicemail deposit is triggered, the Feature Server provides a step-by-step Voice UI for leaving messages and reviewing them. The Voice UI call flow for deposit is shown in the section 13.1.

4.2 *Voicemail Retrieval*

Through voicemail retrieval functionality users can listen to and manage their messages. Voicemail messages deposited for mailboxes defined on User/Person, Agent Login, DN or Agent Group can be retrieved and listened and deleted or saved. The step-by-step Voice UI for retrieval is shown in section 13.2. Feature Server also provides robust web-based user interface, for a user to listen to messages. Any user defined with a Person object in CME, can log in to the Feature Server User UI to listen to messages and act on these messages. A sample web-based user interface is shown in section 7.6.3.

4.3 *Voicemail notifications*

Endpoints can subscribe for voicemail notifications (MWI Message Waiting Indication) that alert the user on the status of his mailbox. MWI notifications are triggered automatically on actions such as Agent Login/Logout and deposit/Retrieval of voice messages. Subscription for these notifications can be either from SIP endpoints or Tlib clients. Feature Server triggers MWI notifications for all the endpoints that have subscribed to receive notifications on the mailbox.

4.4 *Group Voicemail*

Group Voicemail allows for agents or a set of users to have access to a common mailbox called the group mailbox. A group of users is represented as an Agent Group object in Genesys CME. Any number of agents/users represented by Person objects can be added to an Agent Group and a mailbox can be defined for the Agent Group object. All users in the Agent Group have access to the messages in the group mailbox. If they are subscribed for notifications, they will also receive MWI notifications for messages deposited in the group mailbox.

As shown in section 13.2, group mailboxes are accessible in the Voice UI through the Group Mailbox Menu, when the user is associated with one or more agent groups. Depending on the user role assigned by the administrator of the Genesys Voicemail Solution, the users will have varying level of privileges on the group mailbox. For example, a User with administrator privileges is allowed to change password and greetings for the mailbox through the Voice UI, while normal users of the group mailbox cannot change password or greetings for the group mailbox.

Group mailboxes are also accessible through the Feature Server web interface. Through the Feature Server Admin interface, an administrator can perform functions such as deletion, changing password, or setting maximum message limits. Through the Feature Server user UI, a user (Person in CME) can access his group mailbox messages, in addition to his personal mailbox messages (explained in section 7.6.3).

5 Call Flows

The following section provides very basic step-by-step call flows involved in all the deployment modes. The call flow will be similar for both deposit and retrieval, so it will be combined in a single diagram. Even though call flow is similar, there are a few differences between deposit and retrieval scenarios.

- Retrieval INVITEs are dialed using Voicemail Access numbers, and the dial plan (internal or Feature Server dial plan) selects the voicemail service (RM) and forwards to RM; whereas deposit INVITEs are intended to internal DNs, if the call meets the deposit condition then the dial plan (internal or Feature Server dial plan) selects the voicemail service (RM) and forwards to RM.
- User experience (prompts) will be different for retrieval and deposit scenarios.
- The MWI notification call flow will be similar for both retrieval and deposit. As soon as a voicemail is deposited or retrieved, a MWI notification will be sent to the clients (Tlib and SIP Clients).

5.1 *Standalone mode (SIP Server uses its internal dial plan) Deployment – Call flow*

Figure 8 explains the step-by-step voicemail call flow involved in the Standalone mode (uses internal dial plan) deployment.

Feature Server registers (through Tlib) with SIP Server (all DNs) to monitor the DNs; this is required to provide MWI notifications to the appropriate Tlib clients. SIP Phone REGISTERs to SIP Server and SUBSCRIBEs to SIP Server for MWI notification. Tlib clients like Interaction Workspace (IWS) registers (Tlib) to SIP Server to receive Tlib MWI notifications.

When a voicemail deposit or retrieval INVITE is requested [1] from SIP Endpoints, SIP Server selects the appropriate voicemail service DN based on the internal dial plan condition and dials the voicemail INVITE to RM [2]. RM proxies the INVITE to MCP [3]. Upon receiving the INVITE, MCP parses and sends the DN number, mailbox number and call condition through an HTTP request to Feature Server. Based on the information, Feature Server serves VXML pages to MCP to provide appropriate media to the caller [4].
When the caller completes the deposit/ retrieval operation, MCP notifies the Feature Server of the completion through the HTTP interface. Feature Server consequently sends Tlib MWI notification to the SIP Server [5]. SIP Server forwards the MWI notification to the appropriate SIP and Tlib clients [6 & 7].

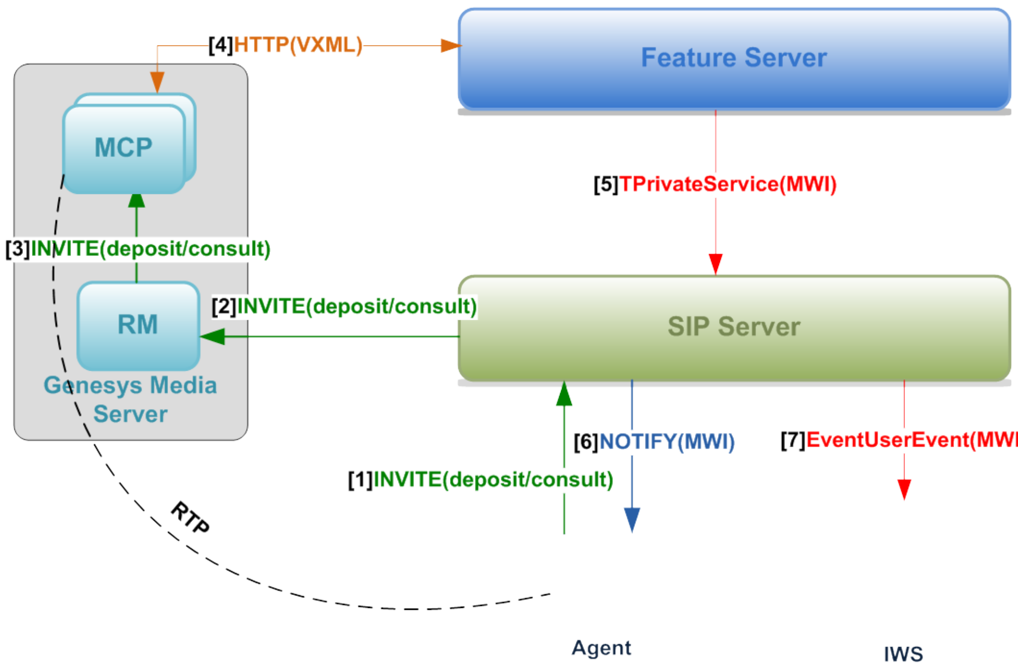


Figure 8 - Standalone mode (SIP Server uses its internal dial plan) Deployment – Call flow

5.2 *Standalone mode using Feature Server dial plan Deployment– Call flow*

Figure 9 explains the step-by-step voicemail call flow involved in the Standalone mode using Feature Server dial plan deployment.

Feature Server registers (through Tlib) with SIP Server to monitor the DNs and the Agents status; this is required to provide MWI notifications to the appropriate Tlib clients. SIP Phone REGISTERs to SIP Server and it SUBSCRIBEs to SIP Server for MWI notification. Tlib clients like IWS registers (Tlib) to SIP Server to receive Tlib MWI notifications.

When a voicemail deposit or retrieval INVITE is requested [1] from SIP Endpoints, SIP Server sends an XS dial-plan request to Feature Server to get the appropriate destination to dial [2]. Based on the call condition and configuration, Feature Server responds with the voicemail service DN in the XS response [3].

SIP Server then uses the voicemail service DN to send the voicemail INVITE to RM [4]. RM proxies the INVITE to MCP [5]. On receiving the INVITE, MCP parses and sends the DN number, mailbox number and call condition through HTTP request to Feature Server. Based on the information, Feature Server serves the appropriate VXML pages to MCP to provide the appropriate media to the caller [6].

When the caller completes the deposit/ retrieval operation, MCP notifies the Feature Server of the completion through the HTTP interface. Feature Server then sends Tlib MWI notification to SIP Server [7]. SIP Server forwards the MWI notification to the appropriate SIP and Tlib clients [8 & 9].

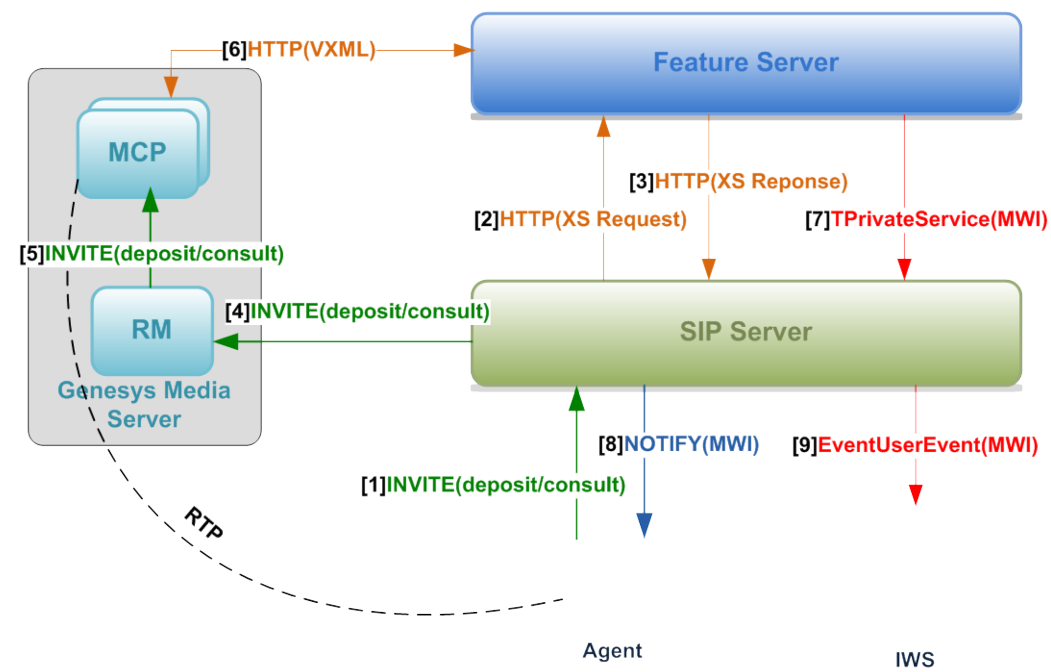


Figure 9 - Standalone mode using Feature Server dial plan Deployment – Call flow

5.3 *SIP Cluster Deployment – Call flow*

Figure 10 explains the step-by-step voicemail call flow involved in the SIP Cluster deployment. Feature Server registers (via Tlib) with SIP Server to monitor the DNs and the Agents status. This is required to provide MWI notifications to appropriate Tlib clients. SIP Phones REGISTER to SIP Proxy and it sends MWI SUBSCRIBE directly to Feature Server through SIP Proxy. Tlib clients like IWS register (Tlib) with SIP Server to receive Tlib MWI notifications.

When a voicemail deposit or retrieval INVITE is requested [1] from SIP Endpoints, SIP Proxy forwards the INVITE to SIP Server [2]. SIP Server sends an XS dial-plan request to Feature Server to get the appropriate destination to dial [3]. Based on the call condition and configuration, Feature Server responds with the voicemail service DN as XS response [4].

SIP Server then uses the voicemail service DN to send the voicemail INVITE to SIP Proxy [5]. SIP Proxy forwards the INVITE to RM [6].RM proxies the INVITE to MCP [7].

On receiving the INVITE, MCP parses and sends the DN number, mailbox number and call condition through HTTP request to Feature Server. Based on the information, Feature Server serves the appropriate VXML pages to MCP to provide appropriate media to the caller [8].

When a caller completes the deposit/ retrieval operation, MCP notifies the Feature Server of the completion through HTTP interface. Feature Server then sends the SIP MWI notification directly to SIP Endpoints through SIP Proxy [9 & 10] and Feature Server sends Tlib MWI notification to SIP Server [11]. SIP Server forwards the Tlib notification to the appropriate Tlib clients (IWS, etc).

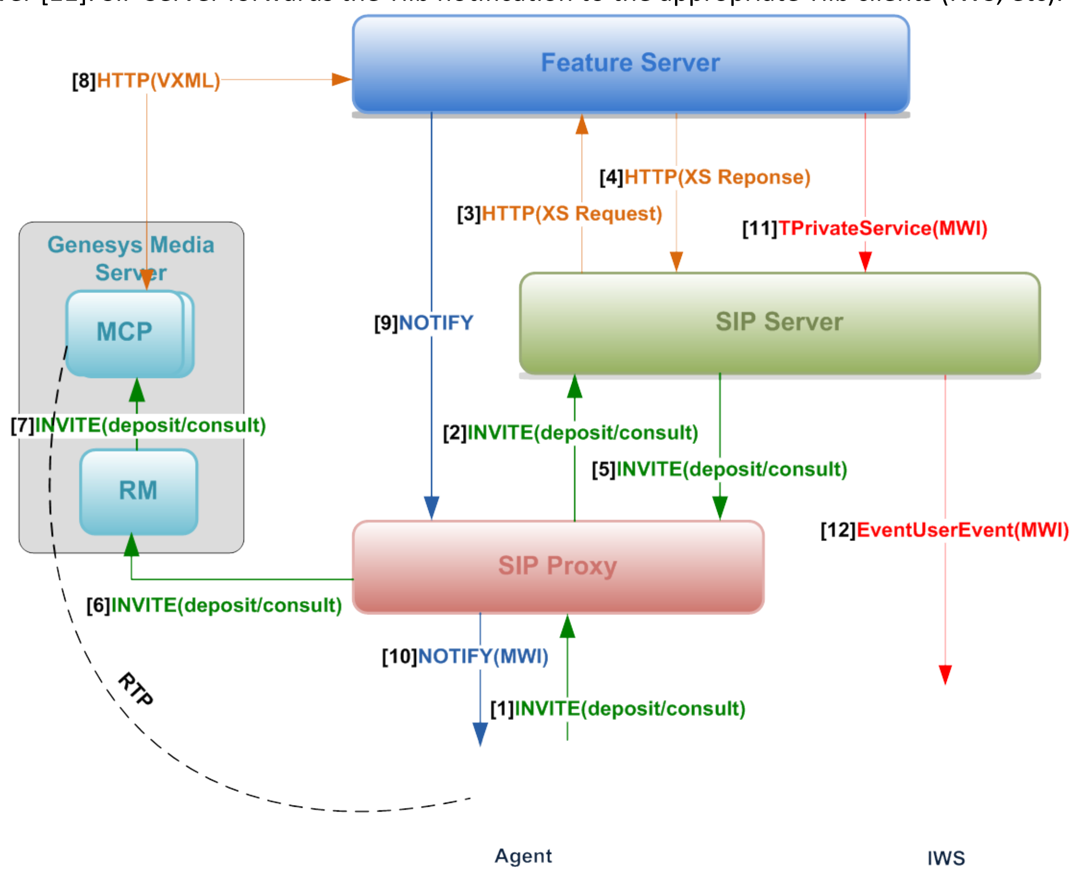


Figure 10 - SIP Cluster Deployment Call Flow

6 Prerequisites for Genesys Voicemail

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

- Genesys Management Framework must be installed and configured. See the Framework Deployment Guide for details.
- If you integrate Genesys Voice mail in your existing environment, make sure Genesys Media Server is installed and configured. See the Genesys Voice Platform Deployment Guide for details.
- A SIP Server instance for managing agents must be installed and configured with one of the above mentioned deployments [section 2]. See the Framework SIP Server Deployment Guide for details.
- Feature Server instances must be installed and configured with one of the above mentioned deployments [section 2]. See the Feature Server Deployment Guide for details.
- All application templates must be installed. Use the supplied templates for the Feature Server.

7 Configuration – Single switch deployment (Standalone) with internal dial plan

The configuration steps provided below are expected to be used along with the SIP Server Deployment Guide. Please refer to the SIP Server Deployment Guide to create a SIP Server application and a switch that is associated with SIP Server.

7.1 *Sample deployment*

Let us assume the following sample deployment:

A call center agent 2000 is configured with:

- SIP IP Phone with Extension number (DN): 1000
- Agent Login id: 2000
- Personal VMB number: 11000 (configured under “Extension” object)
- Agent VMB number: 22000 (configured under “Agent Login” object)
- Group VMB: 72000 (the agent belongs to “CreditCard_Dept” Agent Group where group voicemail is configured)
- Voicemail Access number : 9999 (configured in the dial-plan)
- Other Genesys objects, person: 2000 (Employee Id : 1814, name : John Doe), place: 1000
- Distribution number: RP – 5000

A call center agent 4000 is configured with:

- An Agent (4000) with phone extension: 3000 and his personal VMB: 31000, he also belongs to the agent group “CreditCard_Dept” where group VMB is configured as 72000

Servers:

- SIP Server is configured to use its internal dial plan
- Feature Server is configured in a standalone mode
- RM and MCPs are configured

7.1.1 Sample Personal Voicemail Deposit scenario

1. Consider only personal VMB 11000 is configured and agent VMB 22000 is not configured.
2. Agent's (2000) phone (DN 1000) REGISTERs to SIP Server.
3. Agent's (2000) phone (DN 1000) activates MWI Subscription for personal VMB 11000 to SIP Server.
4. Agent's (2000) phone (DN 1000) receives snapshot of MWI for the personal Mailbox 11000.
5. Agent (2000) logs in to DN 1000 through IWS.
6. Agent's (2000) IWS receives snapshot of MWI for the personal Mailbox 11000.
7. Agent 2000 or phone (DN1000) is configured with one of the VM forwarding conditions described in section 7.5 (forwarding to personal VMB).
8. Inbound call comes to the queue and gets distributed to the agent's phone (DN 1000).
9. Call reaches the forwarding condition and it gets forwarded to the personal VMB 11000.
10. Caller hears agent's personal VM prompt and leaves a voicemail.
11. Caller disconnects.
12. Feature Server sends MWI notification to SIP Server.
13. SIP Server distributes MWI notification for the personal VMB (11000) to both the Agent's phone (DN 1000 - Phone's LED blinks) and Agent's IWS (2000).

7.1.2 Sample Personal Voicemail Deposit scenario when Agent-login mailbox used

1. Both Extension VMB -11000 and agent-login VMB -22000 is configured.
2. Agent's (2000) phone (DN 1000) REGISTERs to SIP Server.
3. Agent's (2000) phone (DN 1000) activates MWI Subscription for personal VMB 11000 to SIP Server.
4. Agent's (2000) phone (DN 1000) receives snapshot of MWI for the personal Mailbox 11000.
5. Agent (2000) logs in to DN 1000 through IWS.
6. Agent's (2000) IWS receives snapshot of MWI for the agent Mailbox 22000.
7. Agent's (2000) phone (DN 1000) receives snapshot of MWI for the agent Mailbox 22000 (it overrides the personal VMB 11000 status which is already received).
8. Agent 2000 or phone (DN1000) is configured with one of the VM forwarding conditions described in section 7.5 (forwarding to VMB).
9. Inbound call comes to the queue and gets distributed to the agent's phone (DN 1000).
10. Call reaches the forwarding condition and it gets forwarded to the agent VMB 22000.
11. Caller hears agent's VM prompt and leaves a voicemail.
12. Caller disconnects.
13. Feature Server sends MWI notification to SIP Server.

14. SIP Server distributes MWI notification for the agent VMB (22000) to both the Agent's phone (DN 1000 - Phone's LED blinks) and Agent's IWS (2000).

7.1.3 MWI notifications - Order of Precedence when device level mailbox and agent level mailbox are configured

If an Agent Login has its own gvm_mailbox and logs in on a device which has its own mailbox, as long as the agent is logged in, the state of the agent's extension reflects the state of the agent login mailbox. When the agent logs out, the agent's extension reflects the state of the device's mailbox.

If an Agent Login has no mailbox and logs in on a device having a mailbox, as long as the agent is logged in, the state of the device (Extension) reflects the state of the device's mailbox.

The following table shows which mailbox is used for MWI notification when VMB is configured at an Agent-Login level and at Extension level.

Agent –Login 2000 gvm_mailbox:	**Extension 1000 gvm_mailbox:**	**When agent 2000 logs in on Extension 1000**
22000	11000	SIP MWI(NOTIFY) for VMB-22000 sent to SIP Phone (DN 1000) Tlib MWI (EventUserEvent) for VMB-22000 sent to IWS (DN 1000 and Agent 2000)
22000	Not defined or ""	SIP MWI for VMB-22000 sent to SIP Phone (DN 1000) Tlib MWI for VMB-22000 sent to IWS (DN 1000 and Agent 2000)
Not defined or ""	11000	SIP MWI for VMB-11000 sent to SIP Phone (DN 1000) Tlib MWI for VMB-11000 sent to IWS (DN 1000 and Agent 2000)
Not defined or ""	Not defined or ""	Nothing is sent

Note: SIP Server selects mailbox during deposit based on the above order of precedence.

7.1.4 Sample Group Voicemail Deposit scenario

1. Agent's (2000) is linked to the AgentGroup "CreditCard_Dept" where group mailbox is configured.
2. Agent's (4000) is linked to the AgentGroup "CreditCard_Dept" where group mailbox is configured.
3. Consider only personal VMB 11000 is configured and agent VMB 22000 is not configured.
4. Agent's (2000) phone (DN 1000) REGISTERs to SIP Server.

5. Agent’s (2000) phone (DN 1000) activates MWI Subscription for personal VMB 11000 to SIP Server.
6. Agent’s (2000) phone (DN 1000) receives snapshot of MWI for the personal Mailbox 11000.
7. Agent’s (2000) phone (DN 1000) activates MWI Subscription for group VMB 72000 to SIP Server.
8. Agent’s (2000) phone (DN 1000) receives snapshot of MWI for both personal Mailbox 11000 and the group Mailbox 72000.
9. Agent (2000) loges in to SIP Server on DN 1000 using IWS.
10. Agent’s (2000) IWS receives snapshot of MWI for both personal Mailbox 11000 and the group Mailbox 72000.
11. Inbound call comes to the queue, no agents are available in the Agent group “CreditCard_Dept”.
12. Caller selects a VM option and URS selects the voicemail as destination with group VMB (72000) configured for the agent group “CreditCard_Dept”.
13. Call gets forwarded to group voicemail box 72000.
14. Caller hears agent's group VM prompt and leaves a voicemail in group VMB 72000.
15. Caller disconnects.
16. Feature Server sends two separate MWI notifications to agent 2000 and agent 4000 who belong to AgentGroup “CreditCard_Dept”.
17. SIP Server distributes MWI notification to the Agent 2000’s phone (DN 1000 - Phone’s LED blinks) and Agent’s IWS (2000) for the group VMB (72000).
18. SIP Server distributes MWI notification to the Agent 4000’s phone (DN 3000 - Phone’s LED blinks) and Agent’s IWS (4000) for the group VMB (72000).

7.1.5 Sample Retrieval Scenario for both personal and group VM

1. A caller leaves a message in VMBs 11000 (personal), 22000 (agent), and 72000 (group).
2. Agent 2000 dials 9999 from his extension to retrieve personal voicemail message (VMB 11000):
    a. Agent 2000 enters the personal mailbox password (by default the password equal to mailbox number 11000).
    b. Agent 2000 hears the voicemail retrieval menu for VMB 11000.
3. Agent 2000 dials 9999 from his extension to retrieve an agent VM message (VMB 22000):
    a. Agent 2000 enters the agent mailbox password (by default the password equal to mailbox number 22000)
    b. Agent 2000 hears the voicemail retrieval menu for VMB 22000.
4. Agent 2000 dials 9999 from his extension to retrieve a group VM message (VMB 72000):
    a. Agent 2000 enters the personal mailbox password (by default, the password is equal to the mailbox number 11000)
    b. Agent follows the prompts. Option 6 allows listening to group VMB 72000 without entering password of group VMB.
    c. Agent 2000 hears the voicemail retrieval menu for VMB 72000.

The examples are given in the following configuration sections are based on the above sample deployment.

7.2 *Switch configuration*

7.2.1 Voicemail Service DN

Create a DN of type VOIP Service to represent RM. At least, one DN of this type is required. The name of the DN can be any string (alphabetic or alphanumeric). The following parameters need to be configured in this DN:

- contact =<RM contact> - Contact must be contact information of Resource Manager (RM), Contact can be IP address and port, or it can be a FQDN
- service-type=<voicemail> – the value should be “voicemail”
- geo-location=<location> – geo-location of the RM. This is an optional parameter and this will be used when more than one Voicemail Service DN is configured. There could be a matching geo-location configured in the inbound Trunk. In this case, SIP Server selects the Resource Manager based on the geo-location of the incoming call. This functionality is used to select the co-located media servers, so that we can optimize the RTP path. Example of Voicemail Service DN configuration is shown below.

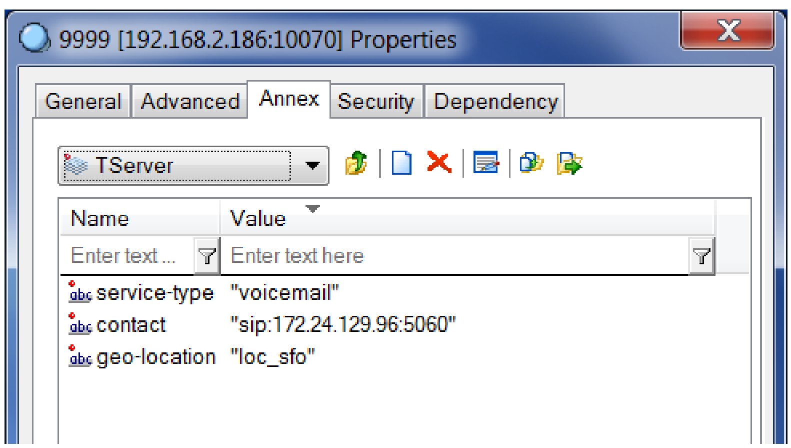


Figure 11 - Sample Voicemail Service DN configuration

Note: If more than one voicemail service DN is configured, IVR profile needs to be configured for each voicemail DN (DNIS) in GVP. The same IVR profile can also be used by configuring a unique number in the user part of the “request-uri” parameter in all of the voicemail DNs.

7.2.2 Voicemail boxes configuration

7.2.2.1 Personal voicemail box

Personal voicemail boxes are configured as DN objects of type “Extension” or “ACD Position” for device-specific voicemail boxes. It can also be configured as an “Agent Login” object for agent-

specific voicemail boxes. The following option needs to be configured under “Extension” type: DN or “Agent Login” object to represent voicemail box for a device/agent
gvm_mailbox – number of the voicemail box of an agent or a device. Note that only digits are supported. It must be unique.

Example of the personal voicemail box configuration when “Extension/ACD Position” DN object is used:

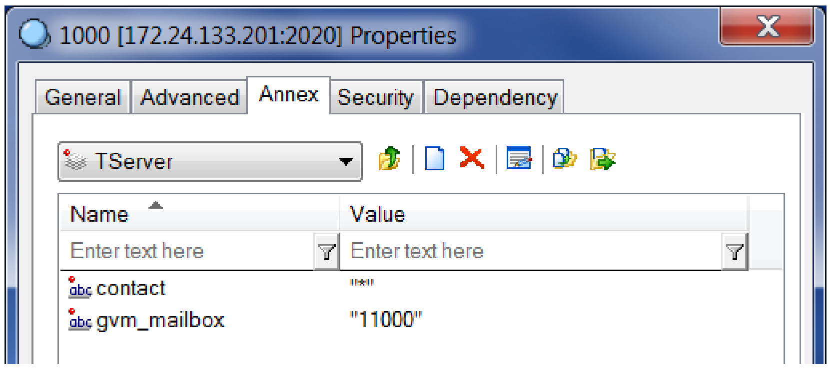


Figure 12 - Sample personal VMB configuration on Extension

Example of the personal mail box configuration when “Agent Login” object is used:

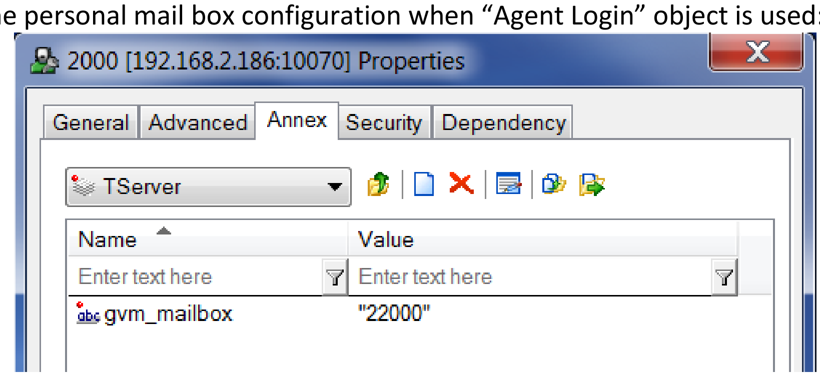


Figure 13 - Sample personal VMB configuration on Agent Login

7.2.2.2 Group voicemail box

An “Agent Group” object will be configured to represent a group of agents who belongs to the same working group. The group voicemail is configured under the “Agent Group” object. The following option needs to be configured in the “Agent Group” object to represent the group voicemail for a particular working group:
gvm_mailbox – number of the voice mail box of the working group.
Note: This option name is same for both personal and group voicemail box configurations, but the configuring DN object differentiates whether it is for personal or group voicemail box.

Example of the group voicemail configuration for an Agent Group is shown below.

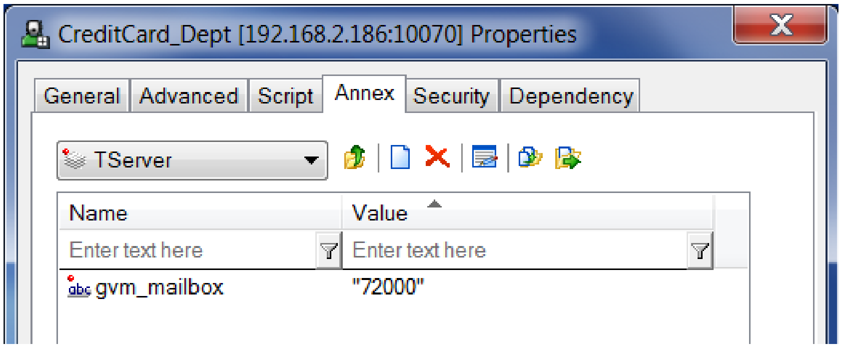


Figure 14 - Sample Group VMB configuration on Agent Group

Note: Group voicemail functionality is not applicable for virtual agent groups which are configured using an Agent Group object. Please refer to the Stat Server user manual for the explanation of virtual agent groups.

7.3 *SIP Server application level configurations*

The following parameters needs to be configured in the SIP Server application:

- subscription-event-allowed=<*> - This is an existing SIP Server application-level option which specifies what are the subscription packages allowed by SIP Server. Genesys recommends configuring *(asterisk) for voicemail integration. This means to allow all type of subscriptions. The default value of this option is empty.
- mwi-implicit-notify=<>(empty(default value)). This is an existing SIP Server application level option which specifies whether implicit MWI notifications are required. Genesys recommends configuring this as an empty value (i.e. to leave the default value) for voicemail integration. This needs to be set to true only if you require MWI but your phone doesn't send SUBSCRIBE.
- mwi-subscribe-vmb=<true>. This option specifies whether SIP Server needs to accept the MWI subscription submitted for voicemail box rather than DN number. This option is required for the phone to subscribe to multiple VMBs like one personal and several group VMBs. This option needs to be set to true if the group voicemail functionality is used. This option works only when the above option "mwi-implict-noify" is empty. The default value of this option is false.

Example of the SIP Server application configuration is shown below.

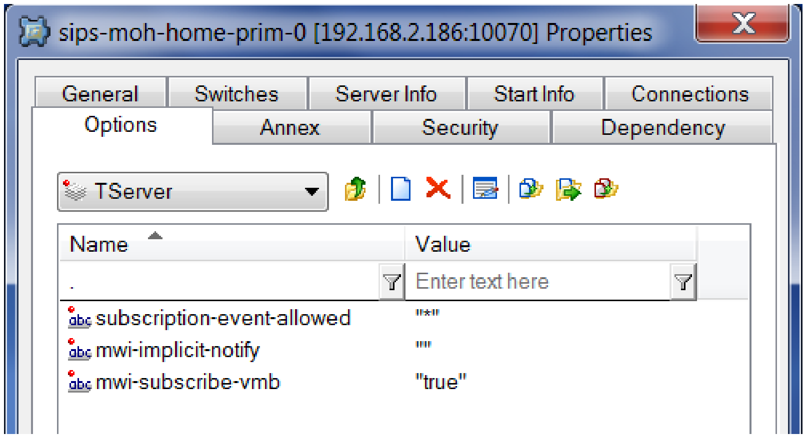


Figure 15 - Sample SIP Server application-level configuration

7.4 *Voicemail access and forwarding numbers*

A unique number (like 9999) is used for voicemail access and voicemail forwarding. This number is used to access both personal and group voicemail boxes. A number translation dial-plan needs to be configured to convert the voicemail forwarding/access number to the special voicemail service DN (gcti::voicemail), like below:

Voicemail forwarding/access number => voicemail service DN

This dial-plan is configured in SIP Server. Create a Voice over IP Service DN for a dial-plan DN, in the TServer section.
1. Set the configuration option service-type to dialplan.
2. Specify the dial-plan rule to be used for access/forward voicemail. This rule allows users to dial the access code to access the voicemail system. Use the following format:
dial-plan-rule-<n>=<acces_code>=>gcti::voicemail

Example of the dial plan:
dial-plan-rule-1=9999=>gcti::voicemail

Example of the dial-plan DN configuration is shown below.

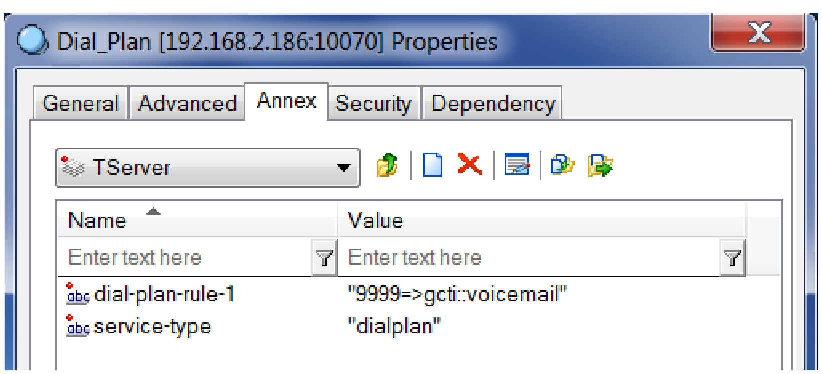


Figure 16 - Sample dial-plan DN configuration

7.5 *Voicemail deposit (forwarding) condition - configurations*

There are two types of VM forwarding can be used widely.
1. Endpoint (Agent) initiated forwarding
2. Network (SIP Server) initiated forwarding

7.5.1 Agent initiated forwarding (Configured from Telephone user interface (TUI))

7.5.1.1 SIP Endpoint initiated forwarding

SIP phones users can set an immediate forward or a conditional forward (on no answer, on busy) to their voice mailbox. In this case, SIP endpoint answers with the redirection response 302. These configurations are specific to the SIP Phones used in the environment, Please refer to the respective SIP Phones user guide for this configuration.

7.5.1.2 IWS initiated forwarding

Call forwarding to voicemail can be configured under IWS using TCallSetForward Tlib request. Call forwarding can be set based on the forwarding modes like unconditional, onbusy, noanswer, etc.
Example request set on IWS (DN 1000):
CallSetForward("1000","9999",OnBusy)
Where 1000 –extension number, 9999- voicemail forwarding number.

7.5.1.3 On demand forwarding from IWS (using TRedirectcall Tlib request)

Call can be forwarded to voicemail on the fly by the agent using the T-Lib request: TRedirectCall(dn, VoiceMail DN). This request will redirect the calls to voicemail without an answer. The agent can also dynamically specify the voicemail box number in the request by adding the voicemail box number in the AttributeExtensions. The key is gvm_mailbox; the value is voicemail box number.This key-value pair should be added in the ArributeExtensions of the above Tlib request. The request looks like:

```
message RequestRedirectCall
AttributeThisDN    '1000'
AttributeConnID    006a01e20b468001
AttributeOtherDN    '9999'
AttributeReferenceID    13
AttributeExtensions  [16]      00 03 00 00..
        'gvm_mailbox'   '11000'
```

Where 1000 – agent extension number, 9999- voicemail forwarding number.

If none of the voicemail numbers are specified then the voicemail box number configured under the agent's device/agent login will be used to forward to voicemail.

7.5.1.4 Call forwarding from call center routing logic (URS strategy)

Calls can be forwarded to voicemail by configuring call center routing logic. A URS strategy can be configured for this purpose. Please refer to section [7.7] to create such a URS strategy.

7.5.2 SIP Server initiated forwarding

SIP Server offers a few call forwarding options.

7.5.2.1 Dial plan redirection to voice mailbox

SIP Server allows configuring a dial-plan for VM forwarding conditions like on no answer/on timeout, on busy, and on DND. To create a Voice over IP Service DN for dial-plan DN, in the TServer section:

1. Set the configuration option service-type to dial-plan.
2. Configure the following dial-plan rule (this rule can be customized based on their needs): dial-plan-rule-<n>=XXXX=>${DIGITS};timeout=5;ontimeout=9999;ondnd=9999;onnoresp=9999;onunreg=9999

Where 9999 is a voicemail forwarding number.

Note: When a call is routed to any destination using TRouteCall (business calls) then the above Dial-Plan functionality is not activated at all. This Dial-plan restriction is configurable through an AttrubuteExtension of TRouteCall “UseDialPlan = partial”. The possible values of UseDialPlan extension are:
UseDialPlan = partial/full/false (default=false for internal Dial-plan functionality)
When the value “partial” is configured, SIP Server will process the above dial-plan.

Example of dial-plan DN configuration is shown below.

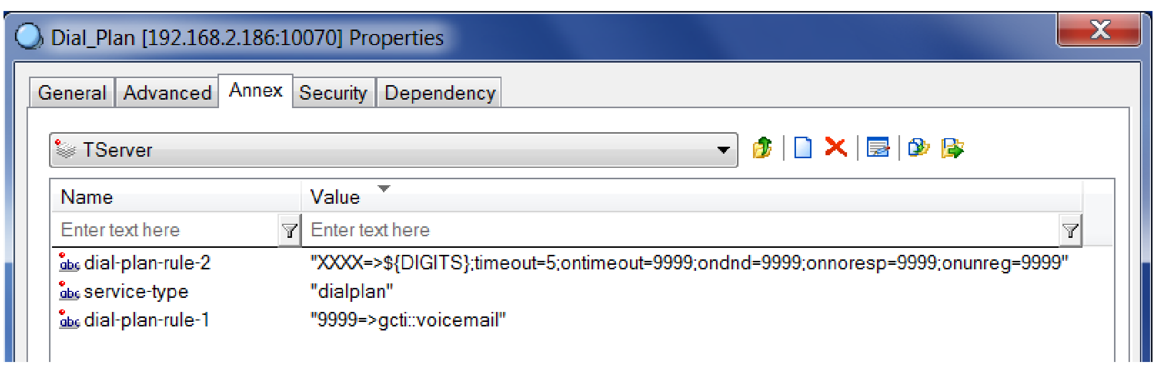


Figure 17 - Sample dial-plan DN configuration

Please refer to the SIP Server deployment guide for dial-plan functionality related information.

7.5.2.2 SIP Server global call redirection to voice mailbox, using TServer options

A global redirection strategy can be applied in the case of no answer to all calls or to some specific type of calls (internal, inbound, outbound) using TServer options.

In the SIP Server configuration objects, two options need to be configured:

- TServer/extn-no-answer-timeout = <timeout>

The value of this option is the delay in seconds SIP Server is waiting before redirecting the call to voicemail.

- TServer/extn-no-answer-overflow = <Voice Mail forwarding number>
  The value of this option is the destination where SIP Server redirects the call when it is not answered after timeout

These options are only effective for business calls (i.e. calls received from a queue or a routing point). When a call is routed to agent DNs using TRouteCall (business calls) then the voicemail forwarding is restricted using the above options. This voicemail forwarding restriction is configurable through an AttrubuteExtension of TRouteCall: “gvm_forwarding = true”. The possible values of gvm_forwarding extension are:
gvm_forwarding = true/false (default=false), When “gvm_forwarding = true” is configured, SIP Server activates the voicemail forwarding using above options.

To allow voicemail forwarding for other calls, the following options must be set, according to the kind of call supposed to be redirected:

- TServer/internal-bsn-calls = true
  Enable voicemail forwarding for internal calls
- TServer/inbound-bsn-calls = true
  Enable voicemail forwarding for inbound calls
- TServer/outbound-bsn-calls = true
  Enable voicemail forwarding for outbound calls

Please refer to No-Answer Supervision functionality in the SIP Server Deployment Guide for more information related to the above feature.

7.5.3 Voicemail forwarding configurations (voicemail deposit) for Group mail box

Group voicemail forwarding can be configured using call center routing logic. For example, if a group of agents configured under an AgentGroup (CreditCard_Dept) doesn’t answer a call then the routing logic will forward the call to voicemail. A URS strategy can be configured to forward the calls to group voicemail. Please refer to section [7.7] to configure this URS strategy.

7.6 *Voicemail retrieval (access) for both personal and group VMs*

A common voicemail access number (9999) [mentioned in section 7.1.1] can be used to access both personal and group voicemail.

7.6.1 Through the phone (TUI)

Access to the mailbox is available from any phone connected to the site or it can be done from external number.
For example, an Agent can listen to personal VMs or group VMs from his assigned phone, all other phones in the call center, or from a home phone.

A mailbox is accessed from a phone by dialing the unique access code (9999).

- If an Agent is calling from a phone assigned to him:
  a. The Agent enters the personal mailbox password (by default, the password is equal to the mailbox number).

b. Next, the Agent follows the prompts . Option number 6 allows listening to Group VMB without entering password of Group VMB.

- If an Agent uses a phone not assigned to him or is calling from an external number:
  a. Agent presses '*'. Next, the Agent follows the prompts (enter desired MB#, which may be the personal VMB as well, and the password of personal VMB)
  b. Agent presses '*'. Next, the Agent follows the prompts (enters desired MB#, which may be Group VMB number as well and the password of group VMB)

7.6.2 Through IWS

Voicemail also can be accessed from IWS using the TMakeCall(Extension, voicemail access number) Tlib request. A gvm_mailbox AttributeExtensions must be associated to this request; this is required to identify which mailbox is accessed. This mechanism is used to access both personal and group mailboxes. The Request looks like below.

```
message RequestMakeCall
AttributeThisDN    '1000'
AttributeOtherDN    '9999'
AttributeMakeCallType    0                (MakeCallRegular)
AttributeReferenceID    9
AttributeExtensions[371] 00 0B 00 00..
        'gvm_mailbox' '72000'
```

Where 1000- agent extension, 9999- voicemail access number, 72000 – group VMB.

7.6.3 Through FS UI

Voicemail can be accessed from the FS UI. The "user login" of FS UI is used for accessing a mail box. The following URL is used to access user interface

User interface: http://<FSIP:8080>/fs/user

In the case of a multisite deployment or with an N+1 Feature Server architecture, the UI can be accessed by any node at any time.

The user needs to enter his user name and password to retrieve voicemail. The user name and password are configured under the person object in CME. The following figure shows the example UI after user logged in. The user has to select the highlighted mailbox 11000 (personal) or 22000 (agent login) or 72000 (group VM) to access the voicemail.

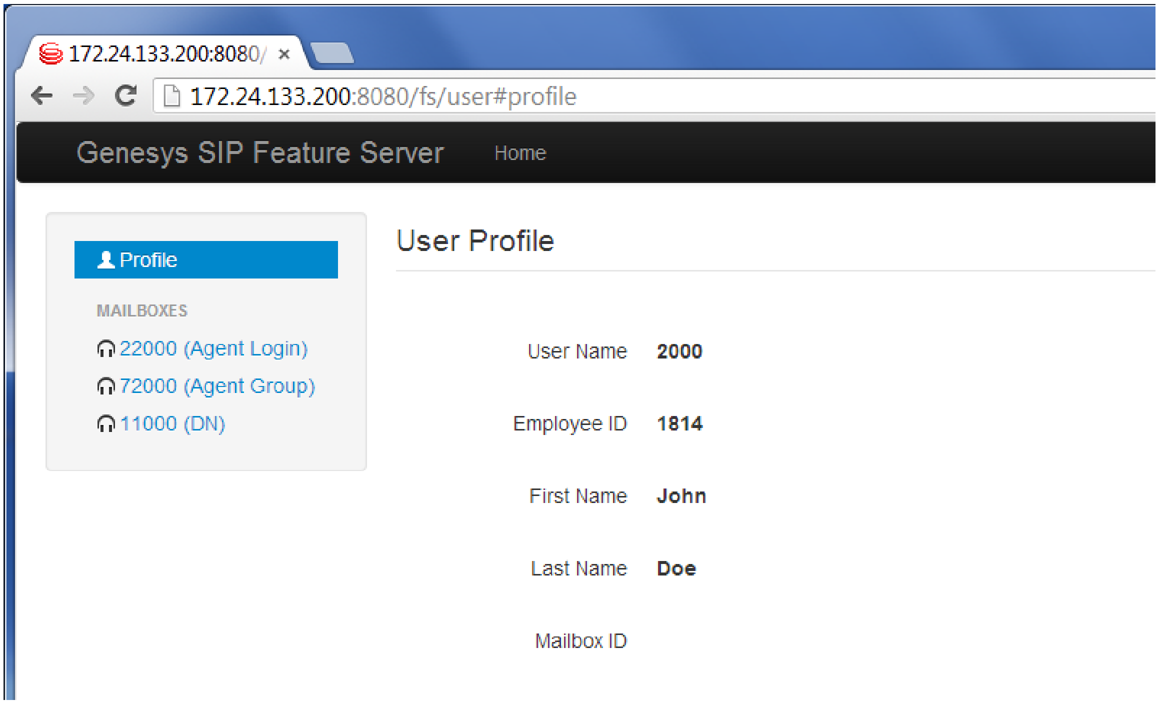


Figure 18 - Sample FS User login Interface to access VMB

7.7 *Using URS to deposit Voicemails*

The following URS strategy needs to be configured for voicemail deposit. This URS strategy can be used to deposit a personal voicemail box or a group voicemail box. The mail box which is going to be deposited should be mentioned as gvm_mailbox in the AttributeExtensions of TRouteCall. The gvm_mailbox can be a personal voicemail box or group voicemail box.

The URS strategy needs to generate TRouteCall like below.

```
message RequestRouteCall
AttributeThisDN        '5000'
AttributeConnID        010a02020db9f03a
AttributeOtherDN       ‘gcti::voicemail’
AttributeExtensions  [136] 00 03 00 00..
        'SIP_HEADERS'  'P-Alcatel-CSBU'
        'P-Alcatel-CSBU'    'call_condition=forwardeduser;categparty=internal;rd=unconditional'
        'gvm_mailbox'  '72000'
AttributeDNIS          '5000'
AttributeRouteType  1 (RouteTypeDefault)
```

ThisDN – Route Point number
OtherDN – Voicemail forwarding DN, “gcti::voicemail” is a special forwarding number used instead of regular forwarding number mentioned in section 7.4
AttributeExtensions->gvm_mailbox – group voicemail box number

By default, SIP Server generates the P-Alcatel-CSBU header with call_conidtion of “forwardeduser” and redirection type (rd) of “unconditional”. If you want customize the P-Alcatel-CSBU header then the following AttributeExtensions will be used. Otherwise the below AttributeExtensions are not required.
AttributeExtensions->SIP_HEADERS – Specifies it is a SIP header
AttributeExtensions->P-Alcatel-CSBU - The special header required for deposit with forwarding condition

The URS deposit strategy to create TRouteCall looks like below

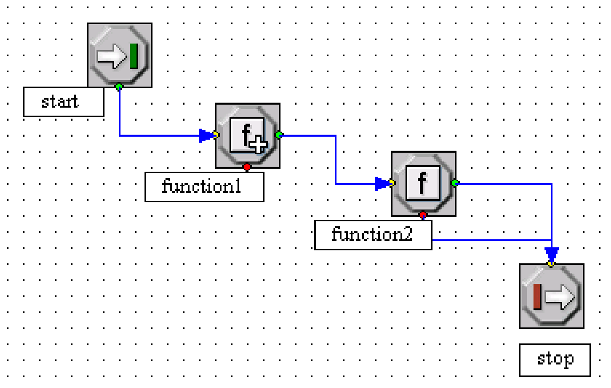


Figure 19 - Sample URS strategy to deposit VM

7.7.1 Function1

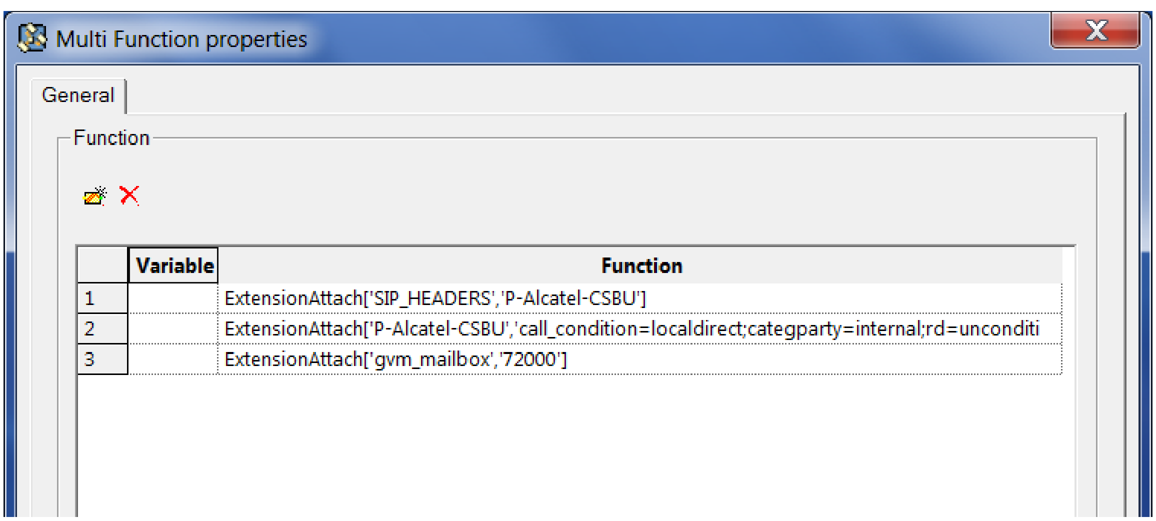


Figure 20 - Function1 of the strategy (Multi Function)

7.7.2 Function2

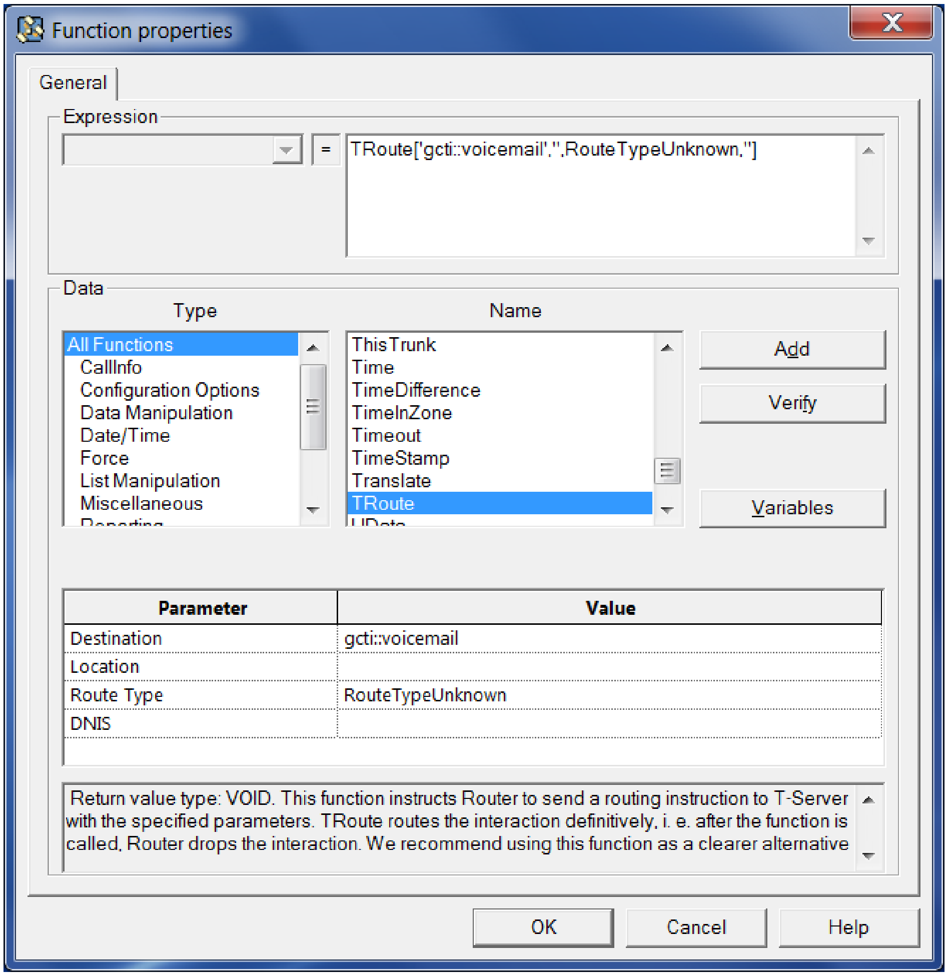


Figure 21 - Function2 of the strategy

7.7.3 Configuring group voicemail deposit

Basically the group mailbox is configured under an AgentGroup object where SIP Server doesn't have control; URS is the one which maintains the AgentGroup information, so it is the URS responsibility to notify SIP Server if a group of agents doesn't pick the call. It needs to provide a voicemail forwarding number ("gcti::voicemail") as a destination to route to when the group of agents do not pick the call. It also needs to provide the group VMB number (gvm_mailbox) in the AttributeExtensions of TRouteCall.

Normally for Group Voicemail deposit, a URS strategy will be created with the target Agent Group. In this strategy, on no agents available during designated timeout event, TRoute function is performed. The routing destination is a VM forwarding number with group VMB attached in the request.

7.7.4 When a regular call forwarding number(9999) is used instead of gcti::voicemail

For voicemail deposit using 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 – mentioned in section 7.4) instead of gcti::voicemail then the number conversion dial-plan mentioned in section 7.4 (dial-plan-rule-1=9999=>gcti::voicemail) doesn't work by default. The following needs to be configured to support regular forwarding number.

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

The URS strategy needs to generate TRouteCall like below.

```
message RequestRouteCall
AttributeThisDN       '5000'
AttributeConnID       010a02020db9f03a
AttributeOtherDN      '9999'
AttributeExtensions   [136] 00 03 00 00..
        'SIP_HEADERS'   'P-Alcatel-CSBU'
        'P-Alcatel-CSBU'    'call_condition=forwardeduser;categparty=internal;rd=unconditional'
        'gvm_mailbox'   '72000'
        'UseDialPlan '  'partial'
AttributeDNIS         '5000'
AttributeRouteType    1 (RouteTypeDefault)
```

7.8 *User experience during deposit and retrieval of VMs (personal and group VMs)*

The user experience will differ in case of voicemail access and deposit. It will also differ in case of personal VMB and group VMB. Section 13 explains the user interface in a detailed manner.

7.9 *Modify mailbox settings using FS Web UI*

Several mailbox settings like max message, optout phone, etc can be edited through the FS Web UI.
The following URL is used to access web admin UI
Administrator interface: **http://<FSIP:8080>/fs/admin**
Both personal and group mailboxes can be edited as shown below.

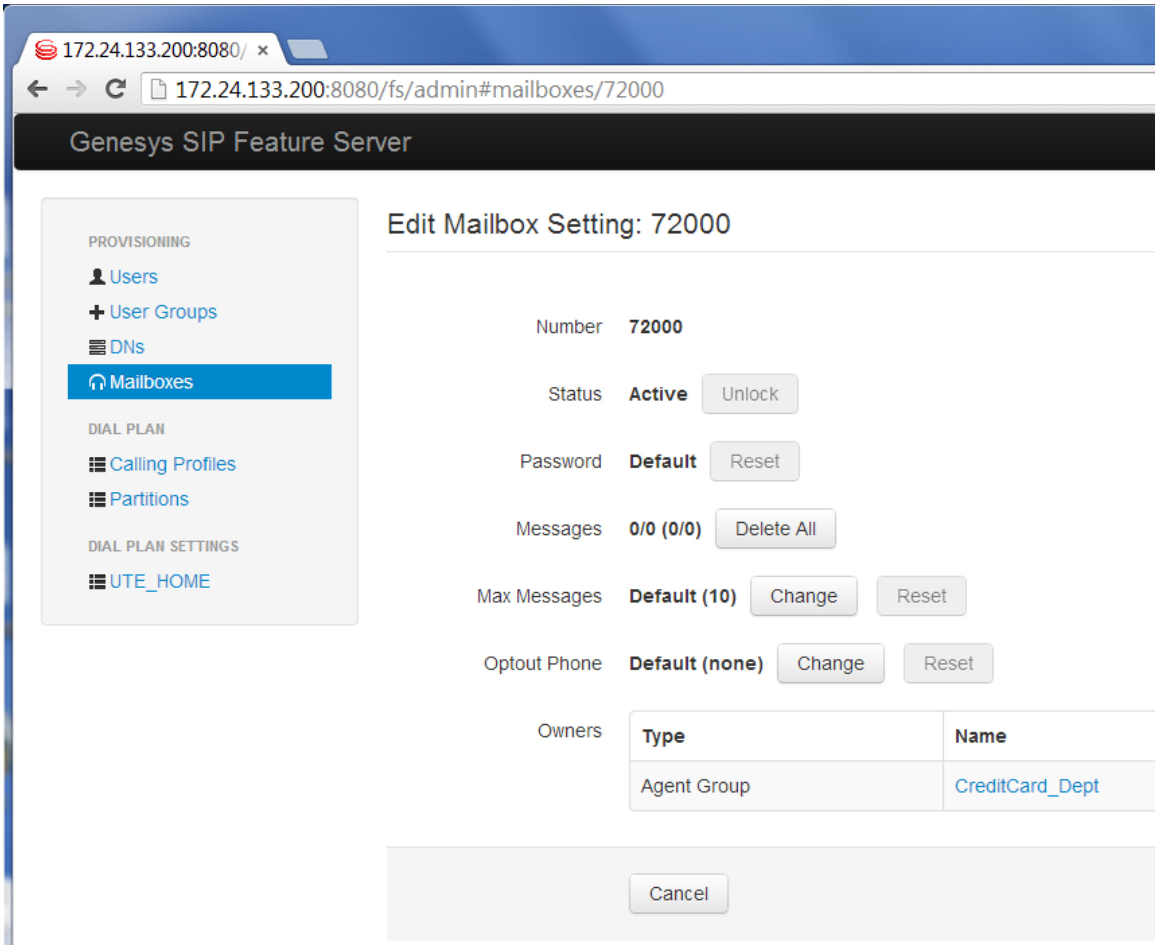


Figure 22 - FS Web Admin UI - Edit Mailbox Settings

7.9.1 Modify Mailbox password using FS user UI

Both personal and group mailboxes can be modified using FS web user login UI. The following figure shows the sample mailbox password change screen.

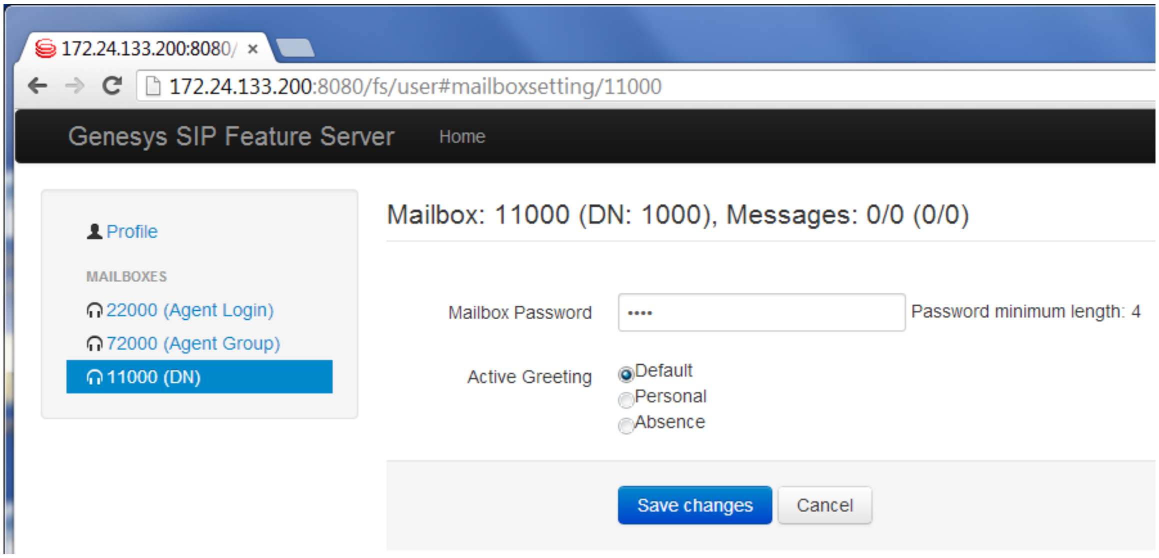


Figure 23 - FS User interface - Modify mailbox password

8 Configuration - Single switch deployment (standalone) with FS dial plan

8.1 *Sample deployment*

This deployment is exactly the same as the sample deployment mentioned in section 7.1 except that SIP Server uses Feature Server's dial plan. All the sample scenarios/configurations mentioned in section 7 are the same except the following:

- Additional SIP Server configurations are required to use FS dial-plan.
- The dial-plan conditions are configured under FS GUI.
- Voicemail boxes can be configured under person objects as well.

Let us assume the sample deployment mentioned in the standalone section 7.1 along with following

- Person object 2000 is configured with VMB : 33000
- SIP Server is configured to use FS dial-plan

Sample deployment when an IPPBX user is connected to SIP Server:
An IPPBX user is configured with:

- SIP IP Phone is configured with Extension number(DN): 6000
- Device VMB number: 11000 (configured under "Extension" object)
- User VMB number: 33000 (configured under "Person" object)
- Voicemail Access number : 9999 (configured in the dial-plan)
- Person : 6677 (Employee Id: 1814, name: John Doe)
- Place : 6000, Person (6677) is configured with default place as 6000
- Distribution number: RP – 5000

8.1.1 Sample personal voicemail deposit/retrieval scenario when person level VMB is configured

The sample scenarios mentioned in the standalone section 7.1 are applicable for this deployment. Apart from that the following sample scenario also applicable for this deployment.

1. Extension VMB -11000, agent-login VMB -22000 and person VMB -33000 are configured
2. Agent's (2000) phone (DN 1000) REGISTERs to SIP Server.
3. Agent's (2000) phone (DN 1000) activates MWI Subscription for personal VMB 11000 to SIP Server.
4. Agent's (2000) phone (DN 1000) receives snapshot of MWI for the personal Mailbox 11000.
5. Agent logs in to the DN 1000 through a Tlib client using person object 2000.
6. Agent's Tlib client receives snapshot of MWI for the person Mailbox 33000.
7. Agent's (2000) phone (DN 1000) receives snapshot of MWI for the person Mailbox 33000 (Overrides the personal VMB status using person VMB status).
8. Agent 2000 or phone (DN1000) is configured with one of the VM forwarding conditions.
9. Inbound call comes to the queue and gets distributed to the agent's phone (DN 1000).
10. Call reaches the forwarding condition and it gets forwarded to the person's VMB 33000.
11. Caller hears person's VM prompt and leaves a voicemail.
12. Caller disconnects.
13. FS sends MWI notification to SIP Server.
14. SIP Server distributes MWI notification for the person VMB (33000) to the Agent's phone (DN 1000 - Phone's LED blinks) and Agent's Tlib client.

8.1.2 Order of Precedence when device level mailbox, agent level mailbox and person level mailbox are configured

When Extension, agent-login, and person level mailboxes are configured, FS selects the appropriate VMB to deposit, based on the following order of precedence:

1. FS will use User's (person) mailbox if agent is logged-in using person object.
2. Fs will use Agent-login mailbox if agent is logged-in using agent-login id.
3. FS will use Device (Extension) mailbox if the above are not configured or the agent is not logged-in.

FS selects the mailbox for MWI notifications based on the above order.

8.1.3 Sample voicemail deposit/retrieval scenario when an IPPBX user is connected and Extension level mailbox is configured

1. Extension VMB -11000 is configured.
2. User's phone (DN 6000) REGISTERs to SIP Server.
3. User's phone (DN 6000) activates MWI Subscription for device VMB 11000 to SIP Server.
4. User's phone (DN 6000) receives snapshot of MWI for the device Mailbox 11000.
5. User's phone (DN 6000) is configured with one of the VM forwarding conditions.
6. Inbound call comes to the queue and gets distributed to the user's phone (DN 6000).
7. Call reaches the forwarding condition and it gets forwarded to the device VMB 11000.

8. Caller hears device's VM prompt and leaves a voicemail.
9. Caller disconnects.
10. FS sends MWI notification to SIP Server.
11. SIP Server distributes MWI notification for the device VMB (11000) to the user's phone (DN 6000 - Phone's LED blinks).

8.1.4 Sample voicemail deposit/retrieval scenario when an IPPBX user is connected and both Extension level mailbox and Person level mailbox are configured

1. Extension VMB -11000 and person VMB -33000 are configured.
2. User's phone (DN 6000) REGISTERs to SIP Server.
3. User's phone (DN 6000) activates MWI Subscription for device VMB 11000 to SIP Server.
4. User's phone (DN 6000) receives snapshot of MWI for the device Mailbox 11000.
5. User's phone (DN 6000) receives snapshot of MWI for the person Mailbox 33000.
6. User's phone (DN 6000) is configured with one of the VM forwarding conditions.
7. Inbound call comes to the queue and gets distributed to the user's phone (DN 6000).
8. Call reaches the forwarding condition and it gets forwarded to the person VMB 33000.
9. Caller hears person's VM prompt and leaves a voicemail.
10. Caller disconnects.
11. FS sends MWI notification to SIP Server.
12. SIP Server distributes MWI notification for the person VMB (33000) to the user's phone (DN 6000 - Phone's LED blinks).

8.1 *Switch configuration*

The entire configuration explained in section 7.2 is applicable for this deployment. The following configurations mentioned in section 7.2 vary in this deployment:

1. Dial-plan DN mentioned in the section 7.5.2.1 is not applicable for this deployment; instead, the Feature Server DN mentioned in section 8.1.1 is used.
2. The voicemail forwarding conditions using SIPS dial-plan configuration mentioned in section 7.5.2.1 is not applicable for this deployment; instead, the VM forwarding condition configuration under Feature Sever GUI is used here [mentioned in section 8.3].
3. SIP Server application level configurations mentioned in section 7.3 vary for this deployment [mentioned in section 8.2].

8.1.1 Feature Server DN

Create a DN of type VOIP Service to represent Feature Server. The name of the DN can be anything. This DN name needs to be configured as the value of the SIP Server application level option "dial-plan"; this is explained in section 8.2.

Following parameters need to be configured in this DN:

- url=<FS url> – url must be the url of Feature Server(FS). Url is like http://<FS Node>:8800, where FS node – IP address/FQDN of FS, 8800 – default XS dial-plan port which can be modified in FS application.

For n+1 High Availability (HA), FS can be configured as url, url-1, url-2, … url-N, where url, url-1, url-2, … url-N must be urls of the FS1, FS2, … FSN respectively. If more than one url is configured, SIP Server will load-balance XS dial-plan requests across those lists.

- service-type=<feature-server> – The value should be “feature-server”.

Example of Feature Server DN configuration is shown below.

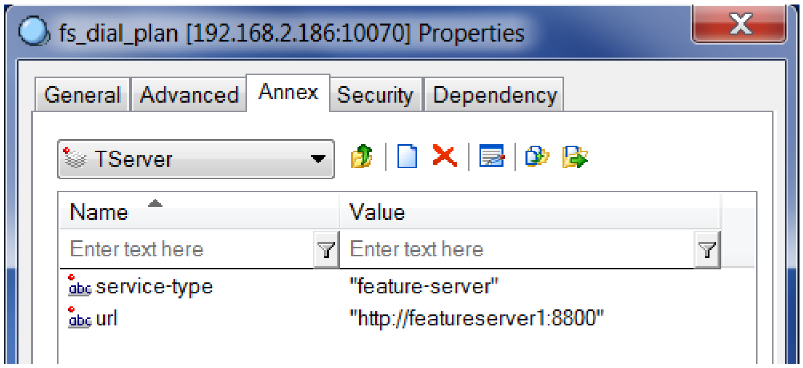


Figure 24 - Sample Feature Server DN configuration

If multiple FS URLs are configured, the configuration looks like below.

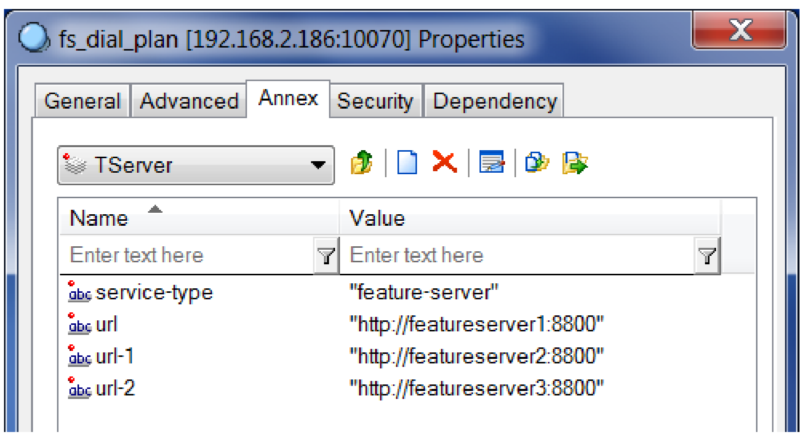


Figure 25 - Sample Feature Server DN when multiple FS URLs configured

8.1.2 Voicemail box configuration under person object

Personal voicemail boxes are configured under “Person” objects. Personal VMBs can also be configured under “Extension” type of DNs and “Agent Login” objects as mentioned in section 7.2.2.1 . FS selects the appropriate VMB to deposit based on the order of precedence mentioned in section 8.1.2
gvm_mailbox – number of the voice mail box. Note that only digits are supported. It must be unique.

Example personal voicemail box configuration under “Person” Object:

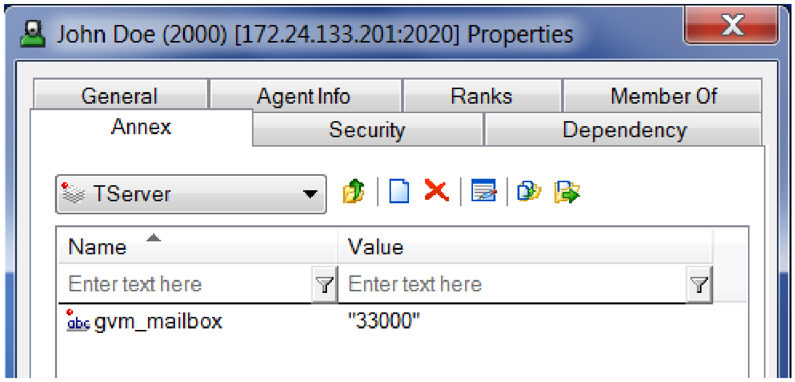


Figure 26 - Sample VMB configuration under person object

8.2 *SIP Server application level configurations*

The configurations explained in the standalone section 7.3 are applicable for this deployment excepts the “dial-plan” option. The dial-plan option needs to point to the FS dial-plan DN in this deployment.

- dial-plan = <Feature Server DN> - Specifies which dial-plan DN will be used for calls. This option decides which Feature Server can be used for dial-plan functionality. It must point to Feature Server DN[8.1.1], ie the name of the Feature server DN must be configured as a value.

Example of SIP Server application configuration is shown below.

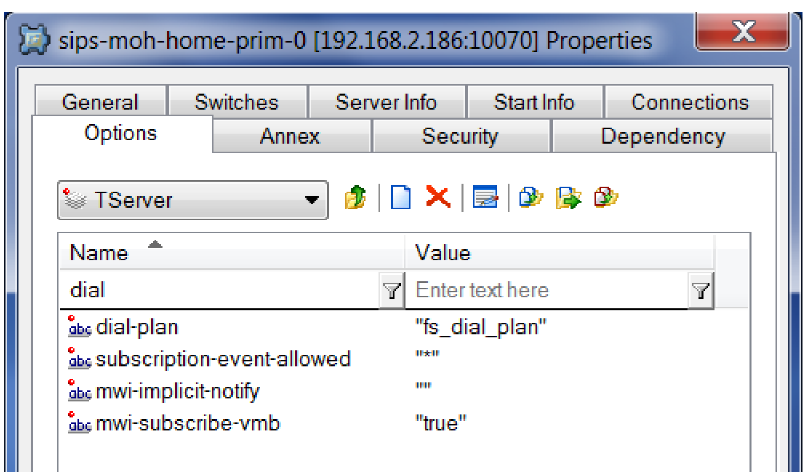


Figure 27 - Sample SIP Server application configuration

8.3 *Dial-Plan configuration for VM in the FS*

8.3.1 Configuring VM Access and forwarding numbers using Profiles and Partitions in FS Web UI

Detailed configuration for creation and administration of calling Profiles and partitions is captured in the feature server deployment guide [References in section 14].

The mechanism used for creating and assigning the dial-plan rule are similar to the mechanism which is used in the standalone mode where SIPS uses its internal dial plan. The partitions are similar to the dial-plan rule in the SIPS internal dial-plan model [mentioned in section 7.4], and calling profiles are similar to the VOIP Service dial-plan DN which can be configured globally to all calls or assigned to a specific device.

For voicemail functionality at least two basic partitions are necessary.

1. Voicemail partition for VM forwarding and VM access
2. Default partition to allow calls

The following operations are done using FS admin UI. The URL is used to access web admin UI.
Administrator interface: http://<FSIP:8080>/fs/admin

8.3.1.1 Voicemail partition for VM forwarding and VM access

An example of a voicemail partition that maps a generic access and forwarding number to voicemail service is shown below. This partition indicates that dialing 9999 from an internal endpoint/external phone would allow the user access/retrieve to his personal and group mailboxes.

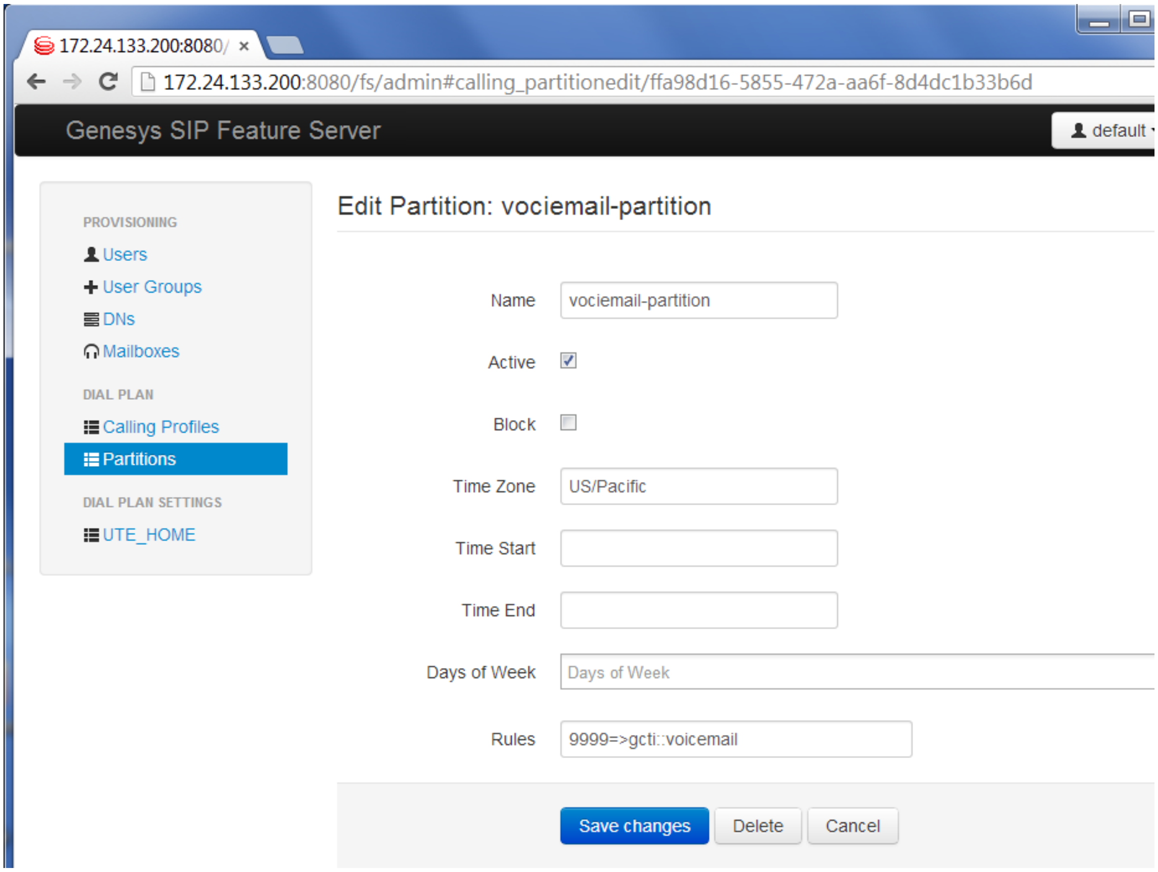


Figure 28 - Sample Voicemail access/forwarding partition

8.3.1.2 Default partition to allow calls

Create a partition with the pattern (. =>$digits) to allow calls to any phone like that shown below. It can be used as default partition.

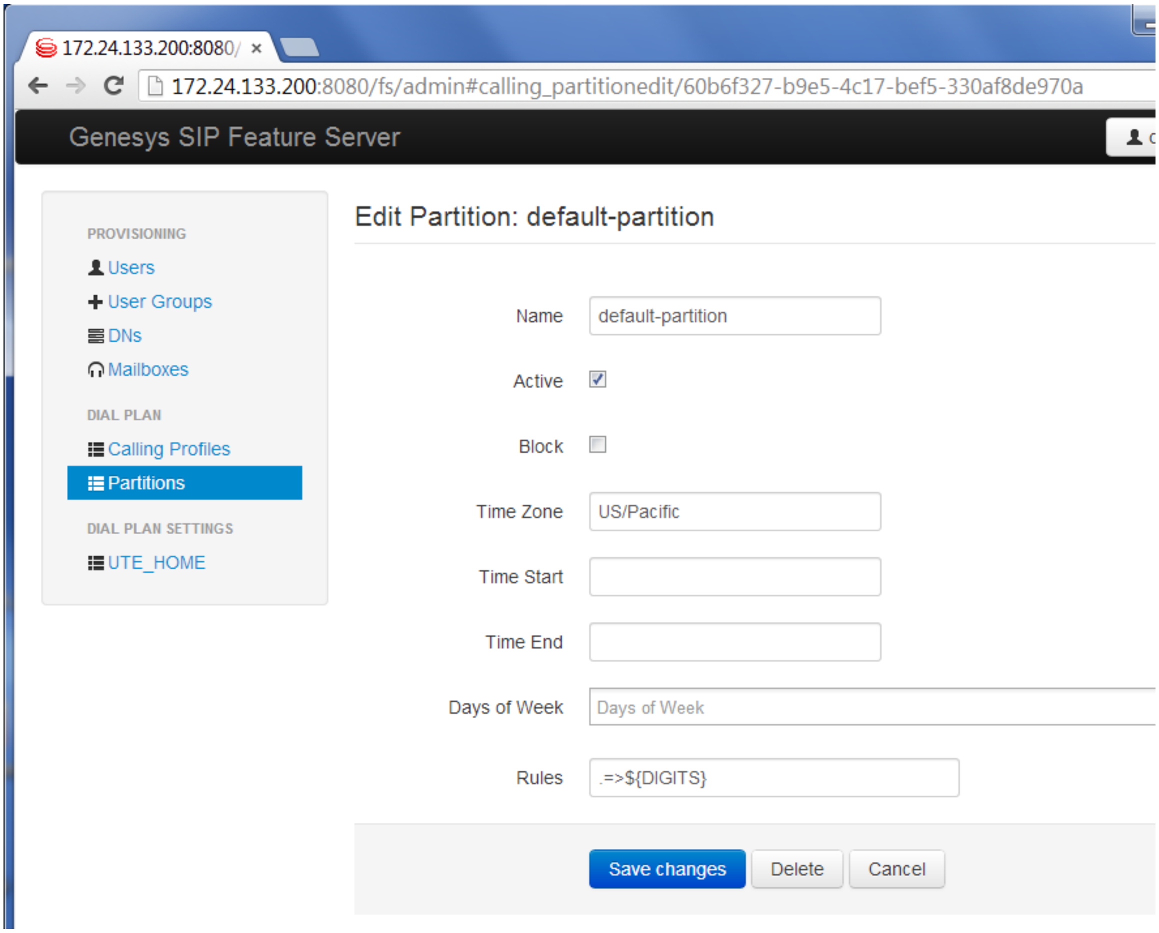


Figure 29 - Default Partition creation

8.3.1.1 Calling profile creation

The calling profiles are created based on the partitions. The below example shows the creation of a calling profile using the partitions already created in the above two steps [sections 8.3.1.1 and 8.3.1.2]. This calling profile can be used as a default calling profile which can be assigned to all users or an individual user.

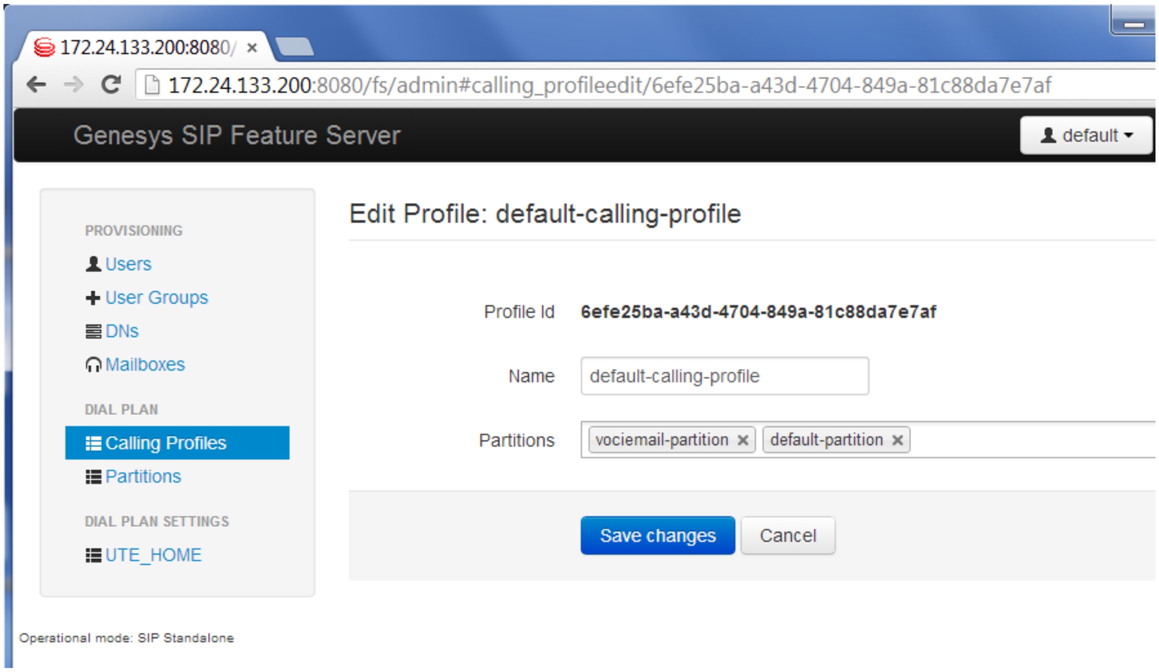


Figure 30 – Sample Default calling profile

8.3.1.2 Assigning the calling profile globally (for all users in a switch)

The above [8.3.1.1] created calling profile (default-calling-profile) can be assigned as the default profile for all users in a switch [as shown below].

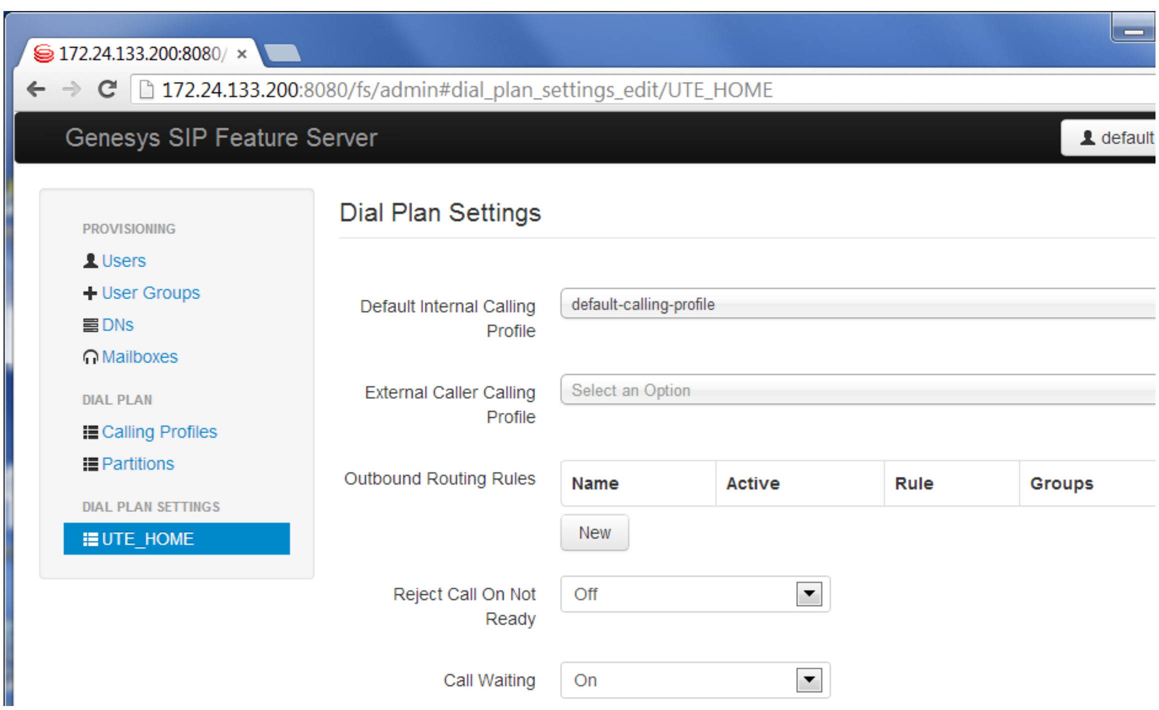


Figure 31 – calling profile association for all users

8.3.1.3 Assigning the calling profile for individual users

The above [8.3.1.1] created calling profile (default-calling-profile) can be assigned to individual users as shown below.

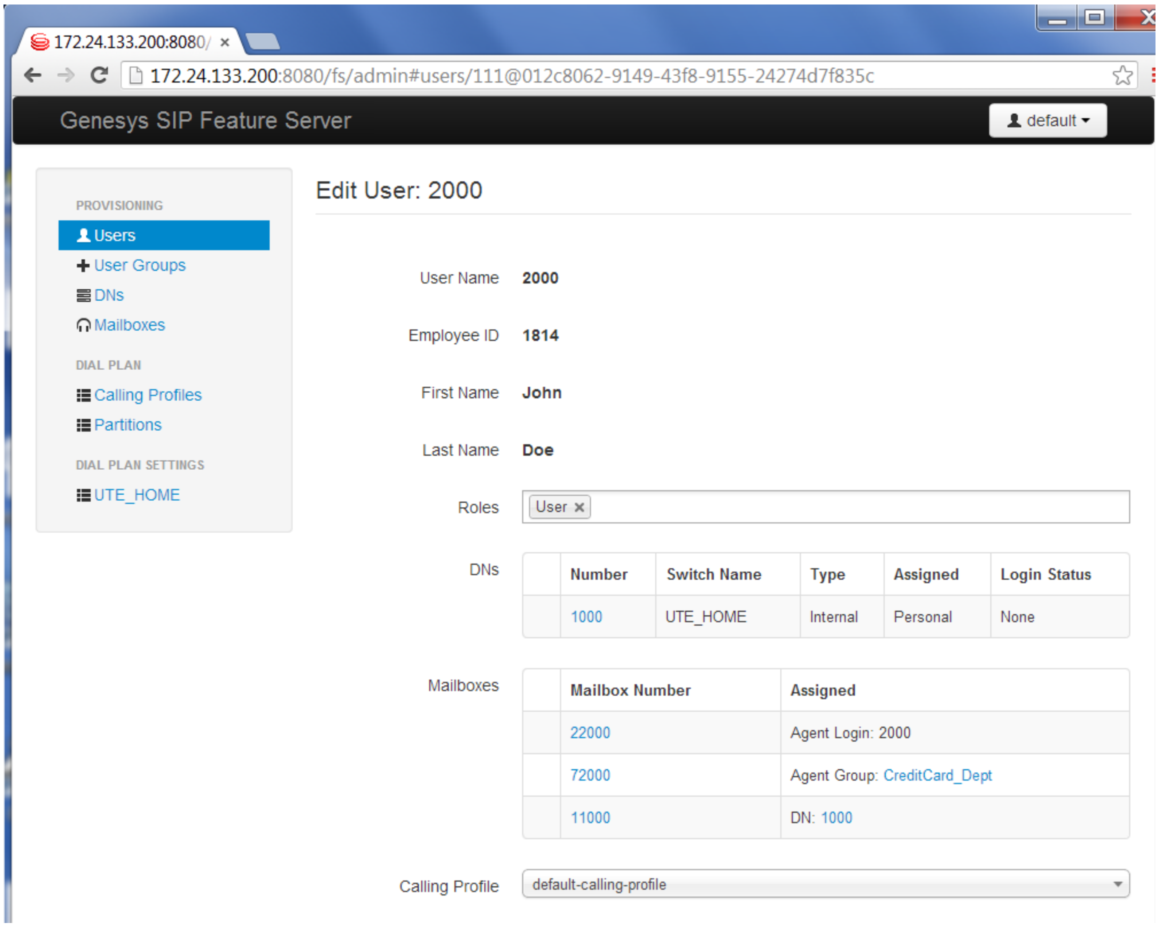


Figure 32 – Calling profile association for an individual User

8.3.2 Configuration for forwarding to voicemail using FS Dial-plan

Forwarding to voicemail can be done in any of the following scenarios using Feature Server dial-plan:

1) Forwarding on Busy
2) Forwarding on no Answer
3) Unconditional forwarding [Forward All calls]

Forwarding settings can be edited /configured both by the administrator or user in the FS Web UI. In the below example, forwarding on no answer with a time-out of 15 seconds is set on the user.

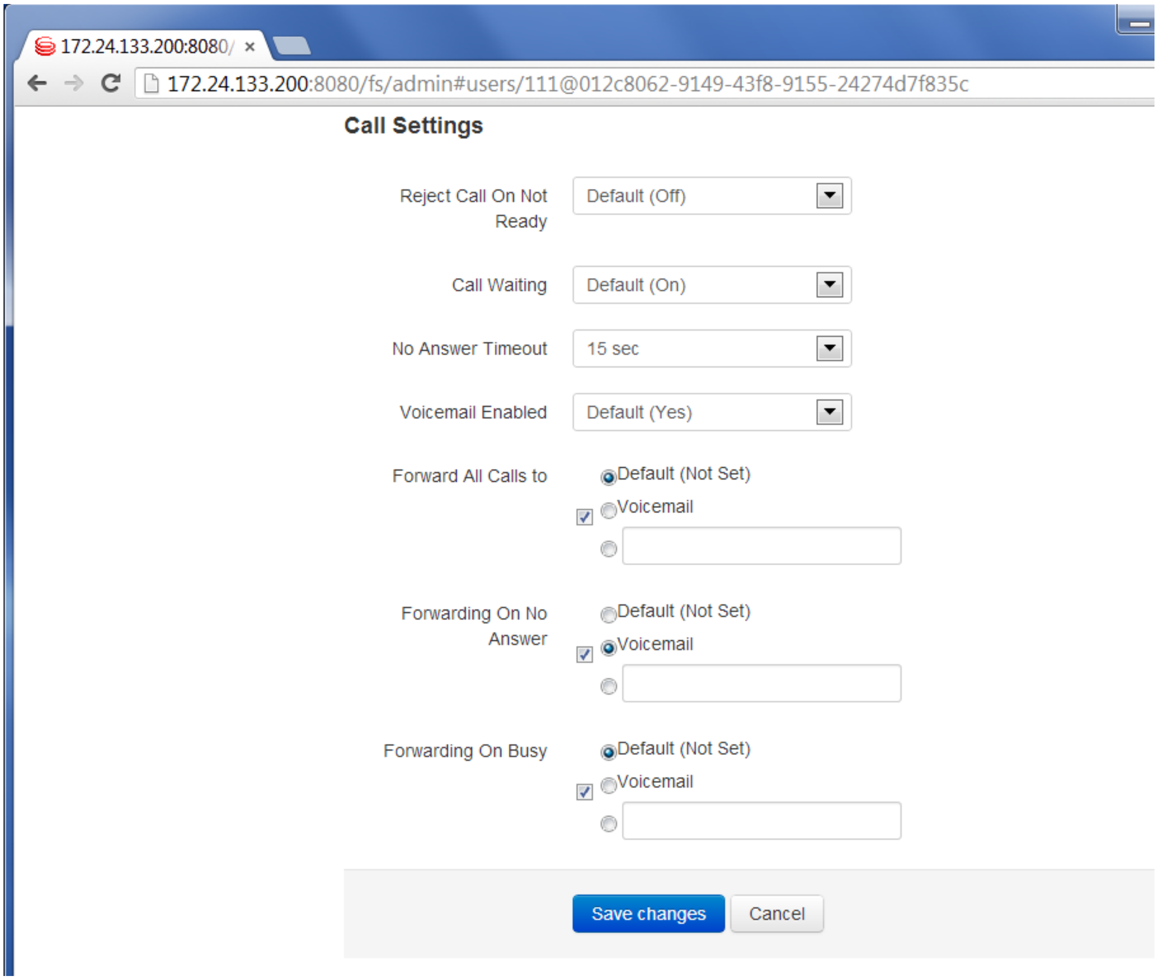


Figure 33 - Forwarding to Voicemail on no answer

Note: When a call is routed to any destination using TRouteCall (business calls) then the above call forwarding to voicemail dial-plan is not activated by default. This Dial-plan restriction is configurable through an AttributeExtension of TRouteCall “UseDialPlan = full”. The possible values of UseDialPlan extension are:
UseDialPlan = partial/full/false (default=partial when FS Dial-plan is used)

If false, the call will be routed directly to the target mentioned in the TRouteCall.
If ‘partial’, Feature Server will be requested to apply number translation and authorization check on target.
If ‘full’, Feature Server will have full authority to perform number translation, authorization, and call forwarding.

The URS strategy needs to generate TRouteCall like below.

```
message RequestRouteCall
AttributeThisDN       '5000'
AttributeConnID       010a02020db9f03a
AttributeOtherDN      '1000'
```

```
AttributeExtensions  [136] 00 03 00 00..
        'UseDialPlan '   'full'
AttributeDNIS        '5000'
AttributeRouteType   1 (RouteTypeDefault)
```

8.3.3 Voicemail forwarding configurations (voicemail deposit) for Group mail box

Group voicemail forwarding can be configured using call center routing logic. For example, if a group of agents configured under an AgentGroup (CreditCard_Dept) does not answer a call then the routing logic will forward the call to voicemail. A URS strategy can be configured to forward the calls to group voicemail. Please refer to section 7.7 to configure this URS strategy.

Note: For this deployment and SIP cluster deployment, the value of TRouteCall AttributeExtensions “UseDialPlan” should be configured as false, so that SIP Server skips the FS dial plan and builds the appropriate INVITE with the URS provided voicemail box number and custom header. In this case, the TRouteCall request looks like below.

```
message RequestRouteCall
AttributeThisDN      '5000'
AttributeConnID      010a02020db9f03a
AttributeOtherDN     'gcti::voicemail'
AttributeExtensions  [136] 00 03 00 00..
        'SIP_HEADERS'   'P-Alcatel-CSBU'
        'P-Alcatel-CSBU'   'call_condition=forwardeduser;categparty=internal;rd=unconditional'
        ‘UseDialPlan’ ‘false’
        'gvm_mailbox'   '72000'
AttributeDNIS        '5000'
AttributeRouteType   1 (RouteTypeDefault)
```

8.4 *Functional comparison of internal and FS DPs*

The following table shows the VM forwarding options available in both SIPS internal dial-plan and FS dial-plan deployments. This also provides differences between the two deployments.

No	Standalone SIPS uses internal dial-plan	Standalone SIPS uses FS dial-plan
	VM forwarding options	
1	VM forwarding number is configured using SIPS internal dial-plan	VM forwarding number is configured using SIPS FS dial-plan
2	SIP endpoint initiated forwarding is supported (302 from Endpoint to VM)	SIP endpoint initiated forwarding is supported
3	IWS initiated forwarding is supported (using TCallSetForward request)	IWS initiated forwarding is supported (using TCallSetForward request)
4	On demand forwarding from IWS (using TRedirectcall Tlib request) is supported	On demand forwarding from IWS (using TRedirectcall Tlib request) is supported

5	Voicemail forwarding using internal dial-plan can be configured for scenarios like on timeout, on busy, on DND, on un-reachable, and on no SIP Registration. For business calls (through TRouteCall) to make use the above dial-plan "UseDialPlan = partial" needs to be specified in AttributeExtensions of TRouteCall	Voicemail forwarding using FS dial-plan can be configured for scenarios like on timeout, on busy, and unconditional. For business calls (through TRouteCall) to make use the above dial-plan "UseDialPlan = full" needs to be specified in AttributeExtensions of TRouteCall
6	SIP Server global call redirection to voice mailbox using application level option no-answer-timeout supported.	SIP Server global call redirection to voice mailbox using application level option no-answer-timeout is supported.
7	Group VM forwarding using URS strategy is supported	Group VM forwarding using URS strategy is supported
	Differences between the deployments	
1	VMB can be configured under Extensions, Agent Logins and Agent Group	VMB can be configured under Extensions, Agent Logins, Persons and Agent Group
2	Mailbox number is determined by SIPS [using SIPS dial plan] and forwarded to FS for deposit or retrieval	Feature Server Dial Plan provides mailbox number to SIPS which is again forwarded back to FS for deposit or retrieval

9 Multi-site deployments

Multi-site deployment is applicable for both the standalone mode deployments explained above in sections 7 and 8.

9.1 *Sample deployment*

Let us assume the following sample deployment.

9.1.1 Site 1 configuration

- SIPServer_Site1 is configured
- Featureserver_Site1 is configured with SIPServer_Site1 in standalone mode
- RM_Site1 and MCPs for Site1 are configured

A call center agent is configured with:

- SIP IP Phone is configured with Extension number(DN): 1000
- Agent Login id: 2000
- Personal VMB number: 11000 (configured under "Extension" object, it can be the same number of Extension or some other unique number)
- Agent VMB number: 22000 (configured under "Agent Login" object, it can be the same number of Agent Login or some other unique number)

- Group VMB: 72000 (he belongs to “CreditCard_Dept” Agent Group where group voicemail is configured)
- Voicemail Access number : 9999 (configured in the dial-plan)
- Other Genesys objects, person: 2000 (Employee Id : 1814, name : John Doe), place: 1000
- Distribution number: RP – 5000
- Voicemail service DN(VOIP DN) - 4444

9.1.2 Site 2 configuration

- SIPServer_Site2 is configured
- Featureserver_Site2 is configured with SIPServer_Site2 in standalone mode
- RM_Site2 and MCPs for Site2 are configured

A call center agent is configured with:

- SIP IP Phone is configured with Extension number(DN): 3000
- Agent Login id: 4000
- Personal VMB number: 31000 (configured under “Extension” object, it can be the same number of Extension or some other unique number)
- Agent VMB number: 42000 (configured under “Agent Login” object, it can be the same number of Agent Login or some other unique number)
- Group VMB: 72000(he belongs to “CreditCard_Dept” AgentGroup where group voicemail is configured)
- Voicemail Access number : 9999 (configured in the dial-plan)
- Other Genesys objects, person:6000 (Employee Id :2824, name : Tony Clifford), place:3000
- Distribution number: RP – 5000
- Voicemail service DN(VOIP DN) - 5555

Note: The site1 agent-2000 and site2 agent-4000 belongs to the same Agent Group “CreditCard_Dept”. Distinct voicemail DN needs to be configured in each site to select proper IVR profile in GVP.

9.1.3 Group VM deposit and MWI notifications for group VM when the agents distributed across the sites belong to same Agent Group

The sample scenarios mentioned in the standalone section 7.1 and 8.1 are applicable for this deployment. Apart from that the following sample scenario is the special for this deployment.

1. Agent’s (2000) belongs to site1 and is linked to Agent Group “CreditCard_Dept” where group mailbox 72000 is configured.
2. Agent’s (4000) belongs to site2 and is linked to Agent Group “CreditCard_Dept” where group mailbox 72000 is configured.
3. Inbound call comes to the queue on Site1, no agents are available in the Agent Group “CreditCard_Dept”.
4. Caller selects VM option and URS selects the voicemail as destination with group VMB (72000) configured for the Agent Group “CreditCard_Dept”.
5. Call gets forwarded to group voicemail box on Featureserver_Site1.

6. Caller hears agent's group VM prompt and leaves a voicemail in group VMB 72000.
7. Caller disconnects.
8. Featureserver_Site1 sends MWI notification for Agent (2000) to SIPServer_Site1 for VMB 72000 status.
9. SIPServer_Site1 forwards the MWI to agent's (2000) IWS as well as agent's SIP Phone (DN 1000).
10. Featureserver_Site1 sends MWI notification for Agent (4000) to SIPServer_Site1 through a special Tlib request TNetworkPrivateService for VMB 72000 status.
11. SIPServer_Site1 forwards the special request to SIPServer_Site2.
12. SIPServer_Site2 generates MWI to agent's (4000) IWS as well as agent's SIP Phone (DN 3000).

Sample TNetworkPrivateService mentioned in the above step-10 looks like below.

```
message RequestNetworkPrivateService
        AttributePrivateMsgID   6002
        AttributeThisDN ' 4444'
        AttributeHomeLocation  'Site2'
        AttributeExtensions     [157] 00 06 00 00..
                'Mailbox'                   '72000'
                'Messages-Waiting'          'true'
                'Voice-Message'             '1/0 (0/0)'
                'Message-Account'           'sip:72000@127.0.0.1:5060'
                'NewMessages'               1
                'OldMessages'               0
                'DestinationDN'             '3000'
        AttributeReferenceID    7
```

9.2 *Configurations*

No special configurations are required for voicemail functionality. Please refer to SIP Server Deployment Guide for multi-site configuration. There is a configuration required in FS to support multi-site deployment which is described in section 10.3.

10 HA and BC deployments

HA and BC deployment are applicable for both the standalone mode deployments explained in sections 7 and 8.

10.1 *Sample BC deployment*

Let us assume the following simple BC deployment:
Site 1:

- SIPServer_Site1 is configured.
- FeatureServer_Site1 is configured with SIPServer_Site1 in a standalone mode.
- RM_Site1 and MCPs for Site1 are configured.

- An Agent (2000) with phone extension 1000 is associated with Site1 and his personal VMB – 11000, he belongs to Agent Group “CreditCard_Dept” where group VMB is configured as 72000.

Site 2:

- SIPServer_Site2 is configured.
- FeatureServer_Site2 is configured with SIPServer_Site2 in a standalone mode.
- RM_Site2 and MCPs for Site2 are configured.
- An Agent (4000) with phone extension 3000 is associated with Site2 and his personal VMB – 31000, he belongs to Agent Group “CreditCard_Dept” where group VMB is configured as 72000.

10.1.1 Sample group VM deposit /MWI Notification scenario when the Agent Group agents distributed across the sites

The following scenario is special when considering VM solution in BC deployment:

1. Agent’s (2000) logs in to site1, he is linked to AgentGroup “CreditCard_Dept” where group mailbox 72000 is configured.
2. Agent’s (4000) logs in to site2, he is linked to AgentGroup “CreditCard_Dept” where group mailbox 72000 is configured.
3. Inbound call comes to the queue on Site1 , no agents are available in the Agent Group “CreditCard_Dept”.
4. Caller selects VM option and URS selects the voicemail as destination with group VMB (72000) configured for the agent group “CreditCard_Dept”.
5. Call gets forwarded to group voicemail box on FeatureServer_Site1.
6. Caller hears agent's group VM prompt and leaves a voicemail in group VMB 72000.
7. Caller disconnects.
8. FeatureServer_Site1 sends MWI notification for Agent (2000) to SIPServer_Site1 for VMB 72000 status.
9. SIPServer_Site1 forwards the MWI to Agent’s (2000) IWS as well as Agent’s SIP Phone(DN 1000).
10. Featureserver_Site1 sends MWI notification for Agent (4000) to SIPServer_Site1 for VMB 72000 status.
11. SIPServer_Site1 requests to its DR peer SIPServer_Site2 using TNetworkPrivateService to send MWI notification to the Agent (4000).
12. SIPServer_Site2 generates MWI to Agent’s (4000) IWS as well as Agent’s SIP Phone(DN 3000).

10.2 *HA and BC deployment configurations in SIP Server*

No special configurations are required to set up voicemail solution in BC and HA deployment. Please refer to the SIP Server HA deployment guide for the configurations of HA and BC deployments.
Multi-site deployment can also be deployed using Business Continuity model.

10.3 *Multi-site, High Availability and Business Continuity Configurations in FS*

There are two configuration parameters on Feature Server that control the import of CME data on start-up and synchronization with CME. Please follow the recommendations below for the use of these parameters.

- Generally all instances of FS in a deployment share the same Cassandra database.
- There has to be only one feature server that is a master node [master=true] for the entire deployment [Cassandra cluster] even across switches.
- Generally master and confSync parameters both will not be true for a Feature Server instance.
- For every Switch, there should be at least one FS instance where either master or confSync is true.

Some example configurations are captured below:

Deployment	**Master/confSync recommendation**
Simple SIP Cluster: 1 FS per switch and only 1 switch SwitchX, FSx	**FSx** Set [cluster]master = true Set [cluster]confSync = false
SIP Cluster: 2 or more FS per switch SwitchX, FSx1, FSx2	**FSx1** Set [cluster]master = true Set [cluster]confSync = false **FSx2** Set [cluster]master = false Set [cluster]confSync = false
Standalone BC : 1 FS per switch, 2 switches SwitchX, FSx SwitchY, FSy	**SwitchX.FSx** Set [cluster]master = true Set [cluster]confsync=false **SwitchY.FSy** Set [cluster]master = false Set [cluster]confSync = true

Standalone Multi-site: 2 or more switches per site [all FS in same Cassandra cluster] SwitchX, FSx1, FSx2 SwitchY, FSy1, FSy2	**SwitchX.FSx1** Set [cluster]master = true Set [cluster]confSync=false **SwitchX.FSx2** Set [cluster]master = false Set [cluster]confSync=false **SwitchY.FSy1** Set [cluster]master = false Set [cluster]confSync=true **SwitchY.FSy2** Set [cluster]master = false Set [cluster]confSync=false

Note: SIP Cluster configuration is explained in section 11.

11 Cluster

11.1 *Sample deployment*

Let us assume the following simple cluster deployment

Node1:

- SIPServer_Node1 is configured
- FeatureServer_Node1 is configured with SIPServer_Node1 in cluster mode

Node2:

- SIPServer_Node2 is configured
- FeatureServer_Node2 is configured with SIPServer_Node2 in cluster mode

Other Servers:

- SIPProxy1 and SIP Proxy2 are configured for Node1 and Node2
- RM and MCPs are configured for SIPServer_Node1 and SIPServer_Node1 are configured

Switch configuration:

- An Agent (2000) with phone extension 1000 and his personal VMB – 11000 (configured in FS GUI), he belongs to Agent Group "CreditCard_Dept" where group VMB is configured as 72000.
- An Agent (4000) with phone extension 3000 and his personal VMB – 31000 (configured in FS GUI), he belongs to Agent Group "CreditCard_Dept" where group VMB is configured as 72000.

11.1.1 Sample Personal Voicemail Deposit scenario

1. Consider personal VMB 11000 is configured for DN 1000 in FS GUI.
2. Agent's (2000) phone (DN 1000) REGISTERs to SIPProxy1.

3. Agent’s (2000) phone (DN 1000) activates MWI Subscription for personal VMB 11000 to FeatureServer_Node1.
4. Agent’s (2000) phone (DN 1000) receives snapshot of MWI for the personal Mailbox 11000 from FeatureServer_Node1.
5. Agent (2000) logs in to the DN 1000 through IWS.
6. Agent’s (2000) IWS receives snapshot of MWI for the personal Mailbox 11000.
7. Agent 2000 or phone (DN1000) is configured with one of the VM forwarding conditions.
8. Inbound call comes to the queue and gets distributed to the Agent's phone (DN 1000) through SIPServer_Node1.
9. Call reaches forwarding condition and it gets forwarded to the personal VMB -11000 using FeatureServer_Node1.
10. Caller hears agent's personal VM prompt and leaves a voicemail.
11. Caller disconnects.
12. FeatureServer_Node1 sends SIP MWI notification for the personal VMB (11000 to the Agent’s phone (DN 1000 - Phone’s LED blinks).
13. FeatureServer_Node1 sends MWI notification to SIPServer_Node1.
14. SIP Server distributes MWI notification for the personal VMB (11000) to Agent’s IWS (2000).

11.1.2 Sample Group Voicemail Deposit scenario

1. Agent (2000) is linked to the Agent Group “CreditCard_Dept” where group mailbox 72000 is configured.
2. Agent (4000) is linked to the Agent Group “CreditCard_Dept” where group mailbox 72000 is configured.
3. Consider personal VMB 11000 is configured for the DN 1000.
4. Agent’s (2000) phone (DN 1000) REGISTERs to SIPProxy1.
5. Agent’s (2000) phone (DN 1000) activates MWI Subscription for personal VMB 11000 to FeatureServer_Node1.
6. Agent’s (2000) phone (DN 1000) receives snapshot of MWI for the personal Mailbox 11000 from FeatureServer_Node1.
7. Agent’s (2000) phone (DN 1000) activates MWI Subscription for group VMB 72000 to FeatureServer_Node1.
8. Agent’s (2000) phone (DN 1000) receives snapshot of MWI for both personal Mailbox 11000 and the group Mailbox 72000 from FeatureServer_Node1.
9. Agent (2000) logs in to SIPServer_Node1on the DN 1000 using IWS.
10. Agent’s (2000) IWS receives snapshot of MWI for both personal Mailbox 11000 and the group Mailbox 72000.
11. Inbound call comes to the queue on SIPServer_Node1 ,no agents are available in the Agent group “CreditCard_Dept”.
12. Caller selects VM option and URS selects the voicemail as destination with group VMB (72000) configured for the agent group “CreditCard_Dept”.
13. Call gets forwarded to group voicemail box.

14. Caller hears agent's group VM prompt and leaves a voicemail in group VMB 72000.
15. Caller disconnects.
16. FeatureServer_Node1 sends SIP MWI notifications to the phone 1000 and the phone 3000 for group VMB 72000.
17. FeatureServer_Node1 sends two separate MWI notifications to SIP Server (one for 1000 another for 3000).
18. SIPServer_Node1 forwards the MWI notification to the agent 2000's IWS and agent 4000's IWS.

11.2 *Switch configuration*

The entire configuration explained standalone uses FS dial-plan in section 8.1 is applicable for SIP Cluster deployment. The differences from the FS dial-plan deployment are:

- Optional geo-location can be configured in the Feature Server DN in the SIP cluster Deployment.
- Extensions(DNs) are created under FS web admin GUI and mailboxes for the Extensions(device level mailboxes) are configured under FS web admin GUI
- Persons/Agent Groups are configured in CME; voicemail boxes for persons can be configured under CME or FS web admin UI. Voicemail boxes for Agent Groups are configured under CME only.

11.2.1 Feature Server DN with geo-location

Create a DN of type VOIP Service to represent Feature Server. Name of the DN can be anything. This DN name needs to be configured as the value of the SIP Server application level option "dial-plan", this is explained in the section 11.3.

Following parameters need to be configured in this DN:

- geo-location=<locations> –This parameter is used to identify a data center in SIP Cluster deployment. This is an optional parameter and this is used only in SIP cluster deployment. There must be a matching geo-location configured at the SIP Server application level to represent the data center. In this case, SIP Server selects the local Feature Server which belongs to the particular data center. This option is not required for a single data-center.
- The parameters "url" and "service-type" are explained in section 8.1.1.

  Example Feature Server DN configuration is shown below.

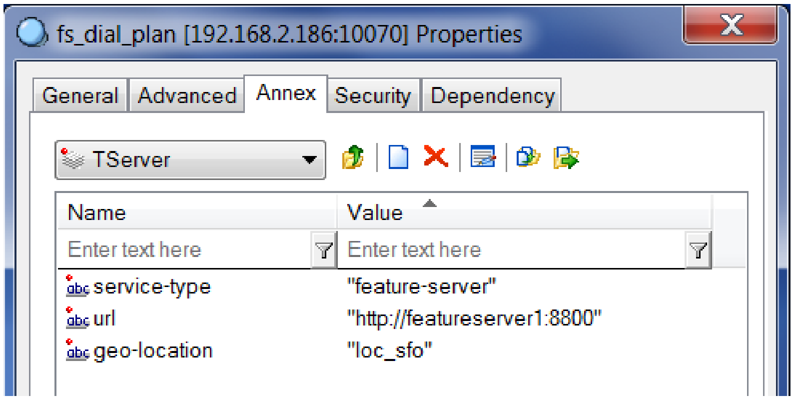


Figure 34 - Sample Feature Server DN configuration in cluster

11.2.2 Extension (DN) creation and mailbox association

In cluster deployment, Extensions (DN) are created through the FS web admin UI. Mailboxes for the Extensions can be assigned at the time of DN creation. The following screenshots shows the create DN page.

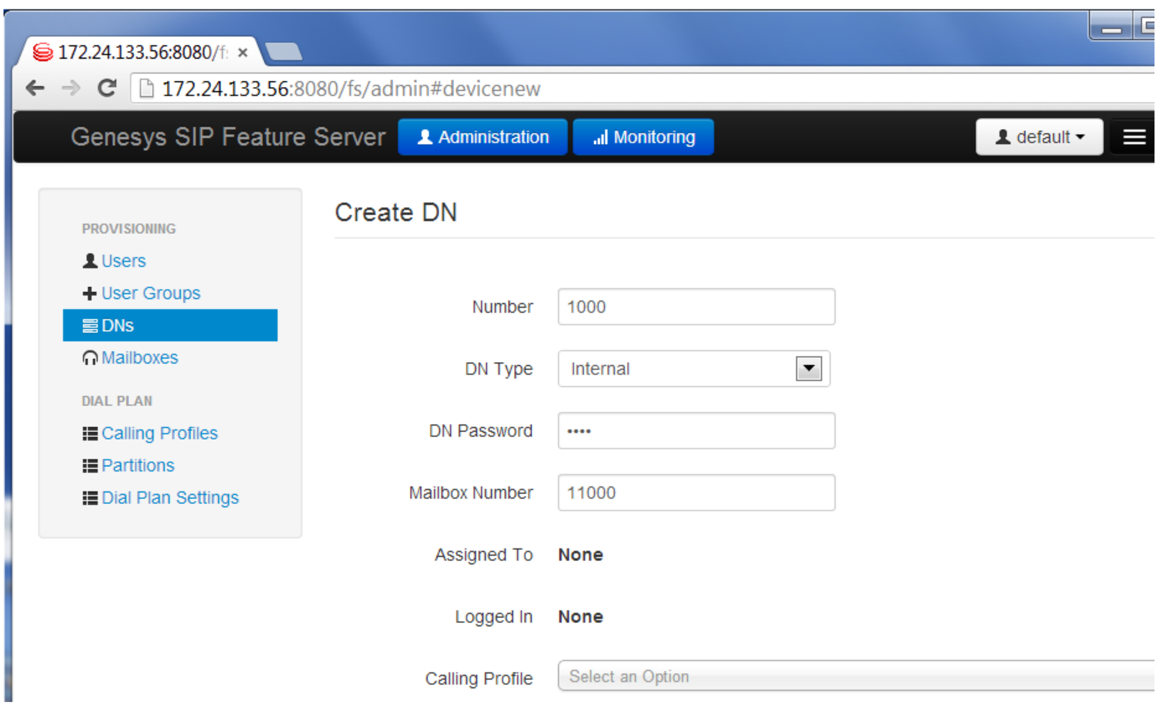


Figure 35 - Create DN in FS web admin UI

Also a bulk upload option is provided to create multiple Extensions (DNs) along with associated mailboxes using a csv file. The format for the definition of the csv file is captured in below screenshot.

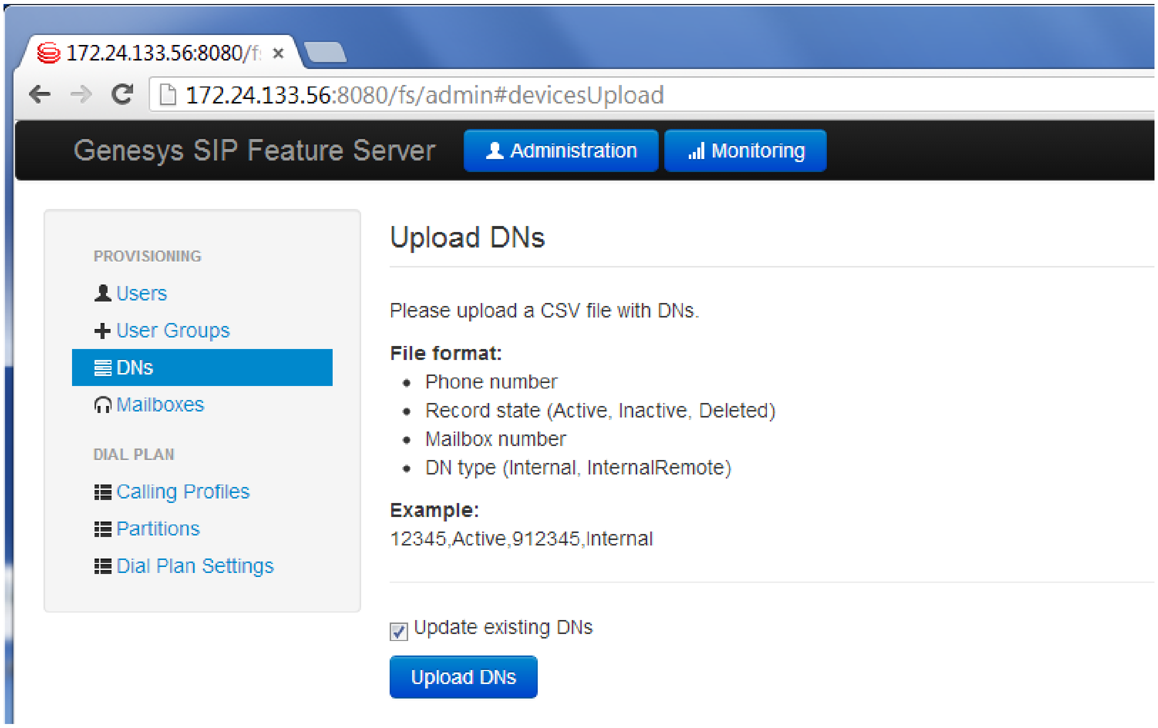


Figure 36 - Bulk DN creation using CSV upload

11.2.3 Mailboxes for Users

At the user level, mailboxes can be configured through CME as shown in section 8.1.2 or through the FS Web admin UI as shown below. Mailboxes can be assigned to users using Edit User->Mailboxes->Assign option as shown below.

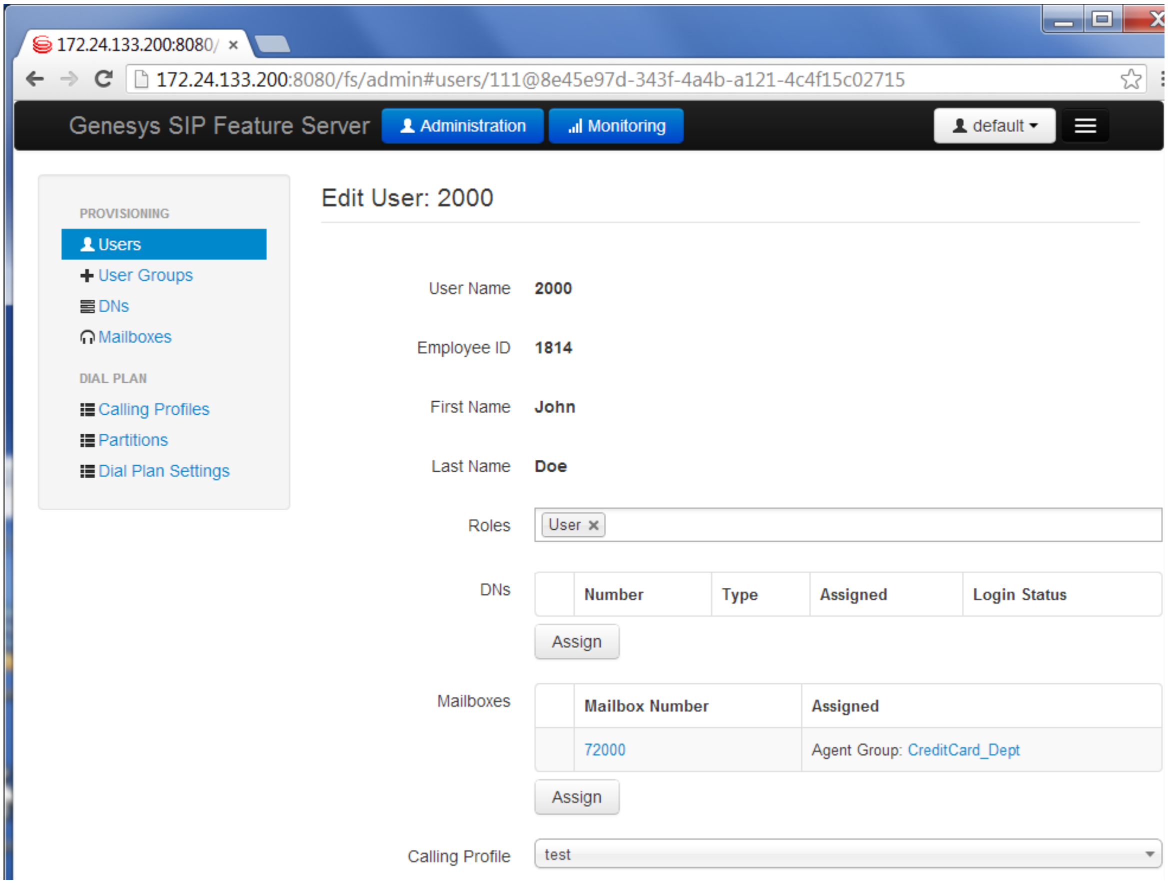


Figure 37 - Assigning user level mailboxes

11.2.4 Group Mailboxes configuration

Group mailboxes can be configured only through CME. Please refer to section 7.2.2.2 for mailbox configuration for Agent Groups in CME.

11.3 *SIP Server application level configurations*

Following parameters need to be configured in the SIP Server application for cluster:

- dial-plan = <Feature Server DN> – Specifies which dial-plan DN will be used for incoming calls. This option decides which Feature Server can be used for dial-plan functionality. It must point to Feature Server DN[7.1.2], ie the name of the Feature Server DN must be configured as a value. If this option is not configured, SIP Server selects Feature Server DN based upon matching geo-location.

Example SIP Server application level configuration is shown below.

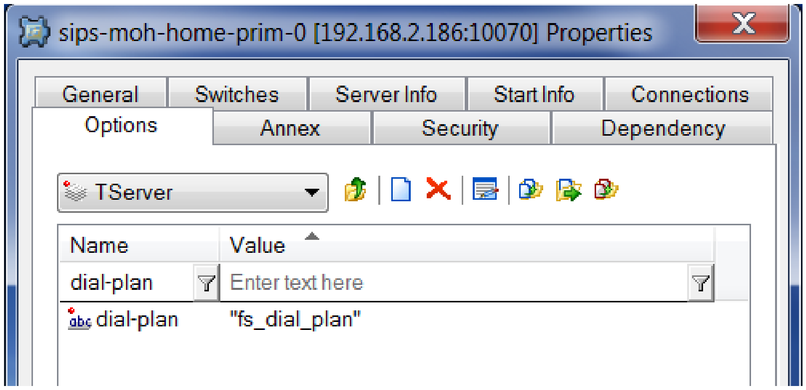


Figure 38 - Sample SIP Server application level configuration in cluster

12 Configuring Polycom Phone

12.1 *Polycom phone Configuration – Standalone single site deployment*

This section explains how the Polycom phones can be configure to support both personal and group voicemail boxes. The below configurations are suggested only when Polycom Sound Point IP phones 450/550/650 are used and UCS (firmware) version 4.0.0 and above is used.

Let us assume the following deployment:

- A Polycom phone agent ”John Doe” assigned to an Extension number 1000 and his personal voicemail box number is 11000 and he belongs to an Agent Group where the group voice mailbox number is configured as 72000
- SIP Server IP: sipserver.com
- SIP Server port: 5060
- Voicemail access number: 9999

The following section explains how the personal VMB and group VMB are configured to receive MWI notifications.

12.1.1 Personal mail box configuration (Line1)

Configure SIP Server address, ports and Expires as mentioned below in the Server1 section where Expires defines the expire interval for REGISTER and SUBSCRIBE dialogs. Configure the personal mail box number in the section “Message center->Subscription Address” and configure the voicemail access number in the section “Message center->Callback Contact”. The configuration looks like below.

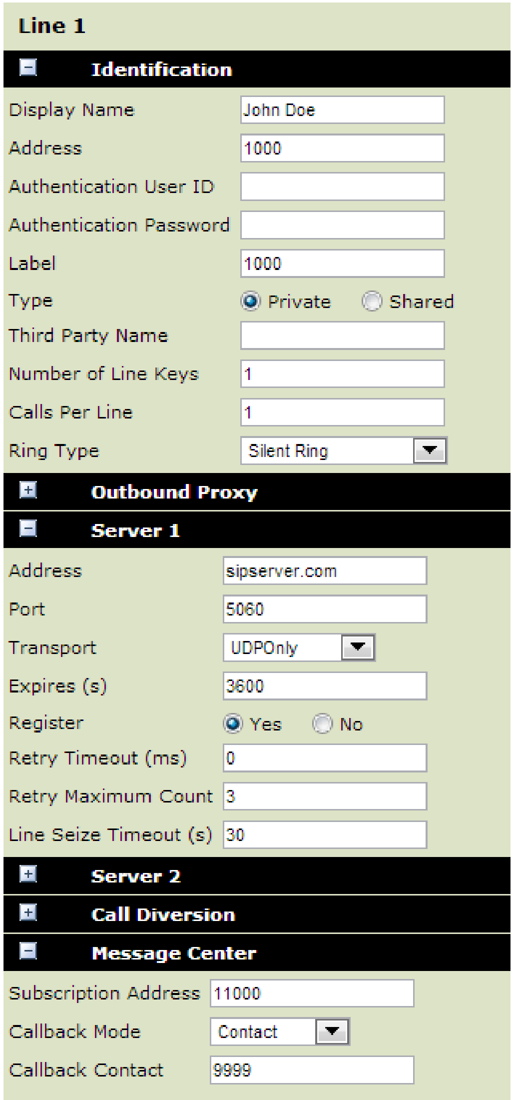


Figure 39 - Polycom Line1 configuration for user

12.1.2 Group mail box configuration

One of the Polycom's phone Lines can be configured to display the group voicemail notifications. No voice can be configured for this Line and this Line will be only used to display group voicemail notifications. Configure "Identification->Address" as the user's Extension number and "Identification->Label" as some name which is different from the extension number (Like GroupVM).
Configure SIP Server address, ports and Expires as mentioned below in the Server1 section where Expires defines the expire interval of the SUBSCRIBE dialog. Register should be set as "no", since no registration is required for this Line.
Configure the group mail box number in the section "Message center->Subscription Address" and configure the voicemail access number in the section "Message center->Callback Contact". The configuration looks like below.

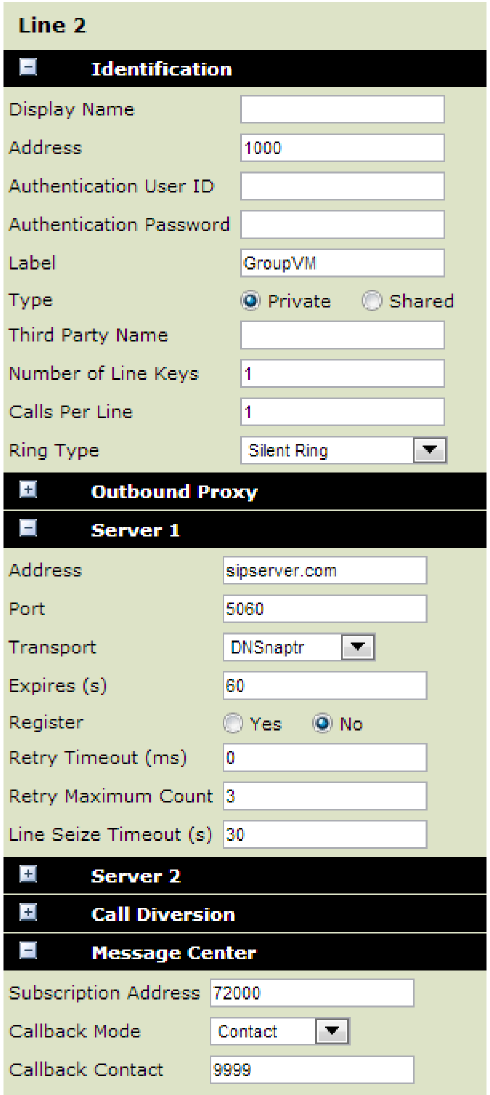


Figure 40 - Polycom Line2 configuration for group VM

This configuration generates the following SUBSCRIBE which is addressed to voicemail rather than DN number. From contains the extension number of the user.

SUBSCRIBE sip:72000@192.168.2.186:5060 SIP/2.0
Via: SIP/2.0/UDP 172.24.128.21:5060;branch=z9hG4bK52a4a499830A1FAC
From: "**1000**" <sip: 1000@172.24.128.21>;tag=E9DE28E8-19D89333
To: <sip:**72000**@192.168.2.186:5060>
CSeq: 1 SUBSCRIBE
Call-ID: 937b0d17-436e7d9a-836177f5@172.24.128.21
Contact: <sip:1000@172.24.128.21:5060>
Event: message-summary
User-Agent: PolycomSoundPointIP-SPIP_450-UA/3.1.2.0392
Accept-Language: en
Accept: application/simple-message-summary

```
Max-Forwards: 70
Expires: 3600
Content-Length: 0
```

12.1.3 Sample phone’s display after the above configuration

The screen of the Polycom phone after the above configuration would look like below.

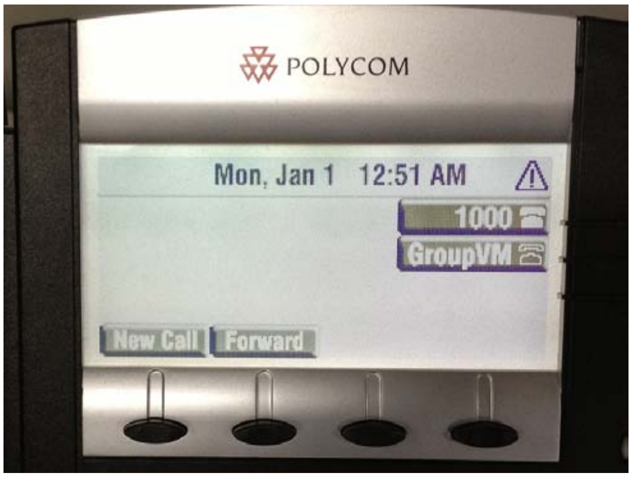


Figure 41 - Sample Polycom screen after the above configuration

12.2 *Polycom phone Configuration – Business Continuity mode*

All Polycom SoundPoint IP phones must be configured to work in fail-over mode. This is configured under <server/> section of the Polycom configuration file (sip.cfg). It looks like below.

<server> voIpProt.server.1.address="sipserver.site1.com" voIpProt.server.1.port="5060" voIpProt.server.1.transport="DNSnaptr"
….
…
voIpProt.server.x.failOver.failBack.mode=”registration”
...
<server/>

Where failback mode = registration means the phone tries the primary server again when the registration renewal signaling begins, ie when the primary is up again.
In this mode phones register and subscribe on one Site only. In case of SIP Server failure or network disconnect, the phones reconnect (register and subscribe) to DR-peer Site.
As soon as the problem is resolved, phones reconnect to “preferred” site back.
Selection of this “preferred” Site depends on the IP address configured in section Server1 of the Polycom phone. A sample scenario is mentioned below. In a SIP BC environment FQDNs are used.

DNS Server should support the following:

- enable-fixed-rrset
- with-fixed-rrset

Let us assume:

- SIP Server IP address on Site1 - 172.24.128.75
- SIP Server IP address on Site2 - 172.24.128.96
- A Polycom user with Extension 1000 is associated with Site1 preferred and his personal VMB – 11000, group VMB- 72000
- Another Polycom user with Extension 3000 is associated with Site2 preferred and his personal VMB – 31000, group VMB- 72000

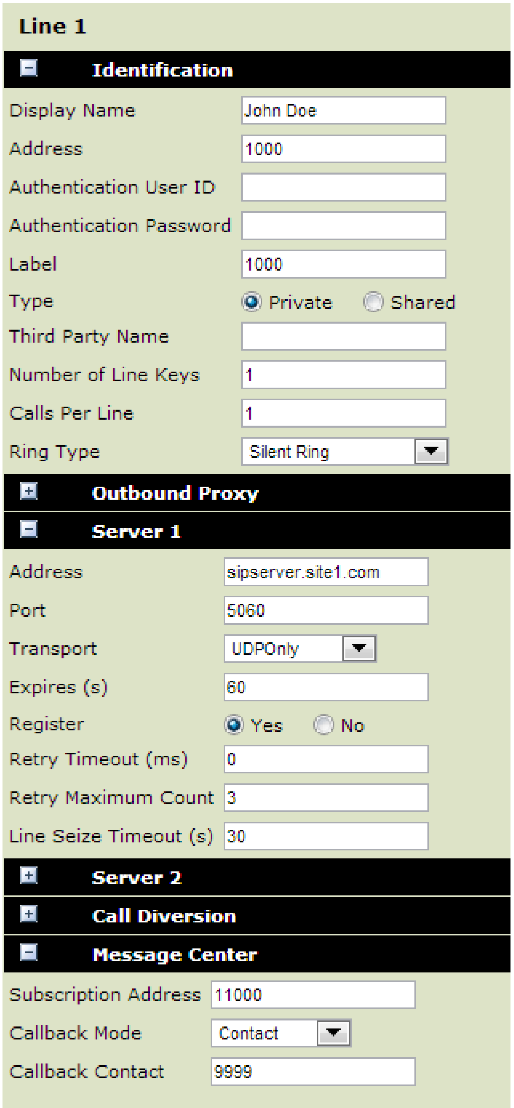
Two FQDNs are added to the DNS table with “A” records of two IP addresses. This will resolve the same IP addresses in different order:
sipserver.site1.com
172.24.128.75
172.24.128.96
sipserver.site2.com
172.24.128.96
172.24.128.75

Sample Site failure scenario:

1. Phone 1000 Registers to 172.24.128.75-SIP Server (Site1 preferred) and subscribes for personal VMB -11000 and group VMB – 72000 on Site1.
2. Site1 is down.
3. Phone 1000 Re-Registers to 172.24.128.96-SIP Server and subscribes for personal VMB -11000 and group VMB – 72000 on Site2.
4. Site1 is up again.
5. Phone 1000 Re-Registers to 172.24.128.75-SIP Server (Site1 preferred) and subscribes for both personal VMB -11000 and group VMB – 72000 on Site1.

12.2.1 Phones configured to work with the preferred Site1– Personal mail box

Phone 1000’s configuration for Line1 with personal mailbox is shown below. The Expires interval configured below should be less like 60 seconds, so that Polycom detects the site failure within 60 seconds (if site failure happens). The Expires defines the expire interval for REGISTER and SUBSCRIBE dialogs.



Line 1

Identification

Display Name	John Doe
Address	1000
Authentication User ID	
Authentication Password	
Label	1000
Type	(•) Private ( ) Shared
Third Party Name	
Number of Line Keys	1
Calls Per Line	1
Ring Type	Silent Ring

Outbound Proxy

Server 1

Address	sipserver.site1.com
Port	5060
Transport	UDPOnly
Expires (s)	60
Register	(•) Yes ( ) No
Retry Timeout (ms)	0
Retry Maximum Count	3
Line Seize Timeout (s)	30

Server 2

Call Diversion

Message Center

Subscription Address	11000
Callback Mode	Contact
Callback Contact	9999

Figure 42 - Polycom Line1 configuration for site1 preferred

12.2.2 Phones configured to work with the preferred Site1– Group mail box

Phone 1000’s configuration for Line2 with group mailbox is shown below. This line is used only for viewing the MWI status and not used for voice.

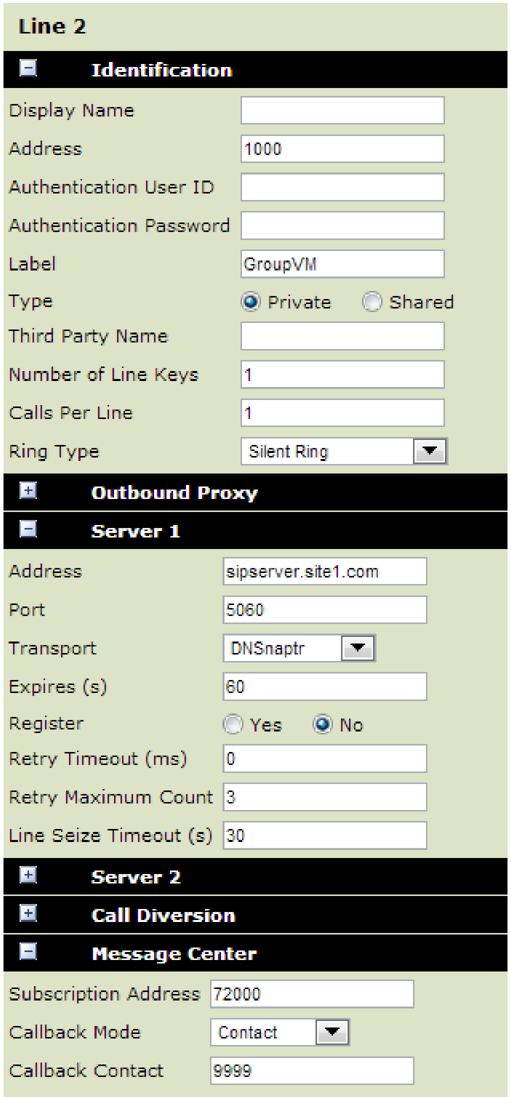


Figure 43 - Polycom Line2 configuration for site1 preferred (group VM)

12.2.3 Phones configured to work with the preferred Site2 – Personal mail box:

Phone 3000’s configuration for Line1 with personal mailbox is shown below.

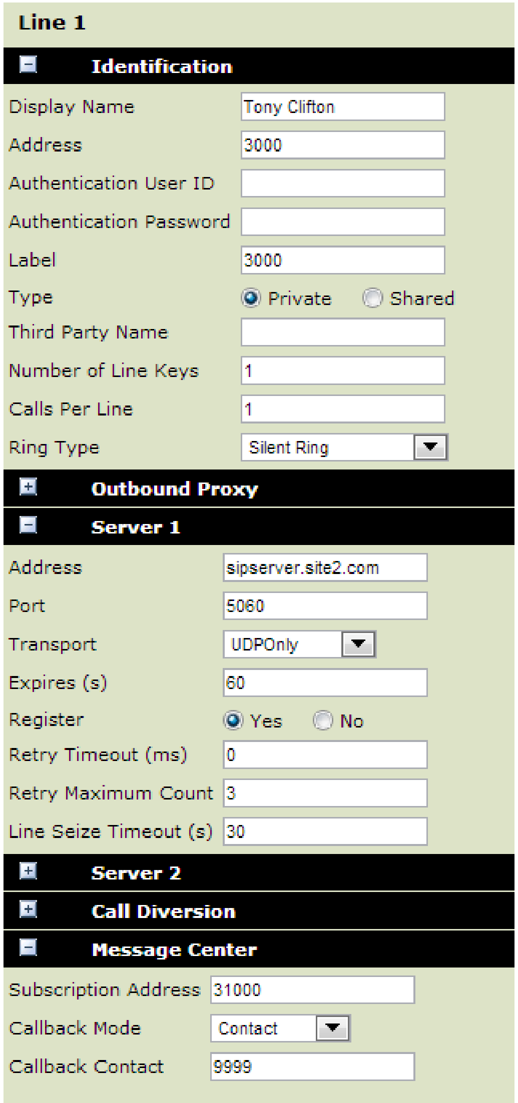


Figure 44 - Polycom Line1 configuration for site2 preferred

12.2.4 Phones configured to work with the preferred Site2 - Group mail box

Phone 3000's configuration for Line2 with group mailbox is shown below. This line is used only for viewing the MWI status and not used for voice.

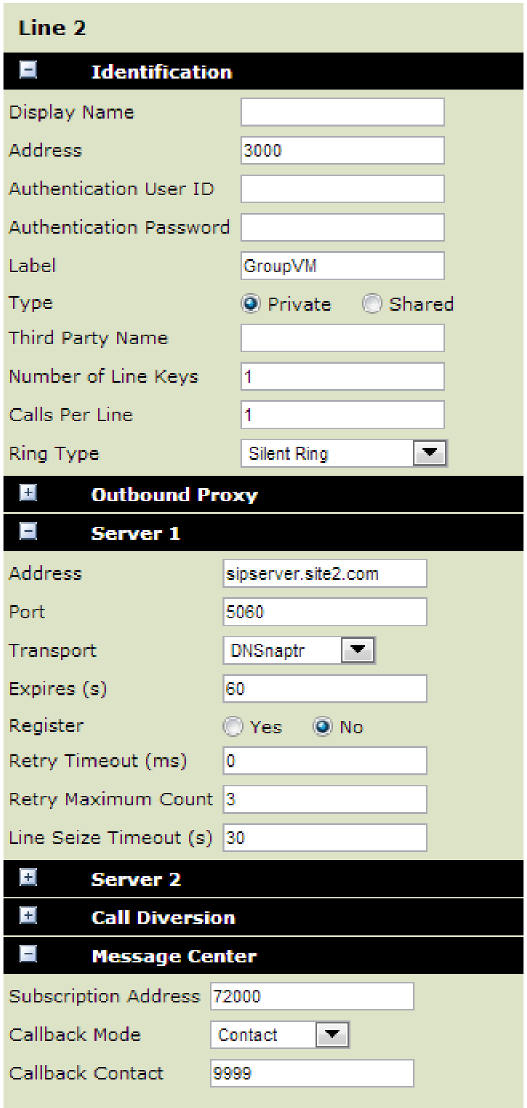


Figure 45 - Polycom Line2 configuration for site2 preferred

13 User Experience

13.1 User experience – Voice UI during voicemail deposit

A sample Voice UI call flow for VM deposit is shown below. The deposit voice menu allows the user to record voicemail, delete and rerecord voicemail and set priority to the voicemail.

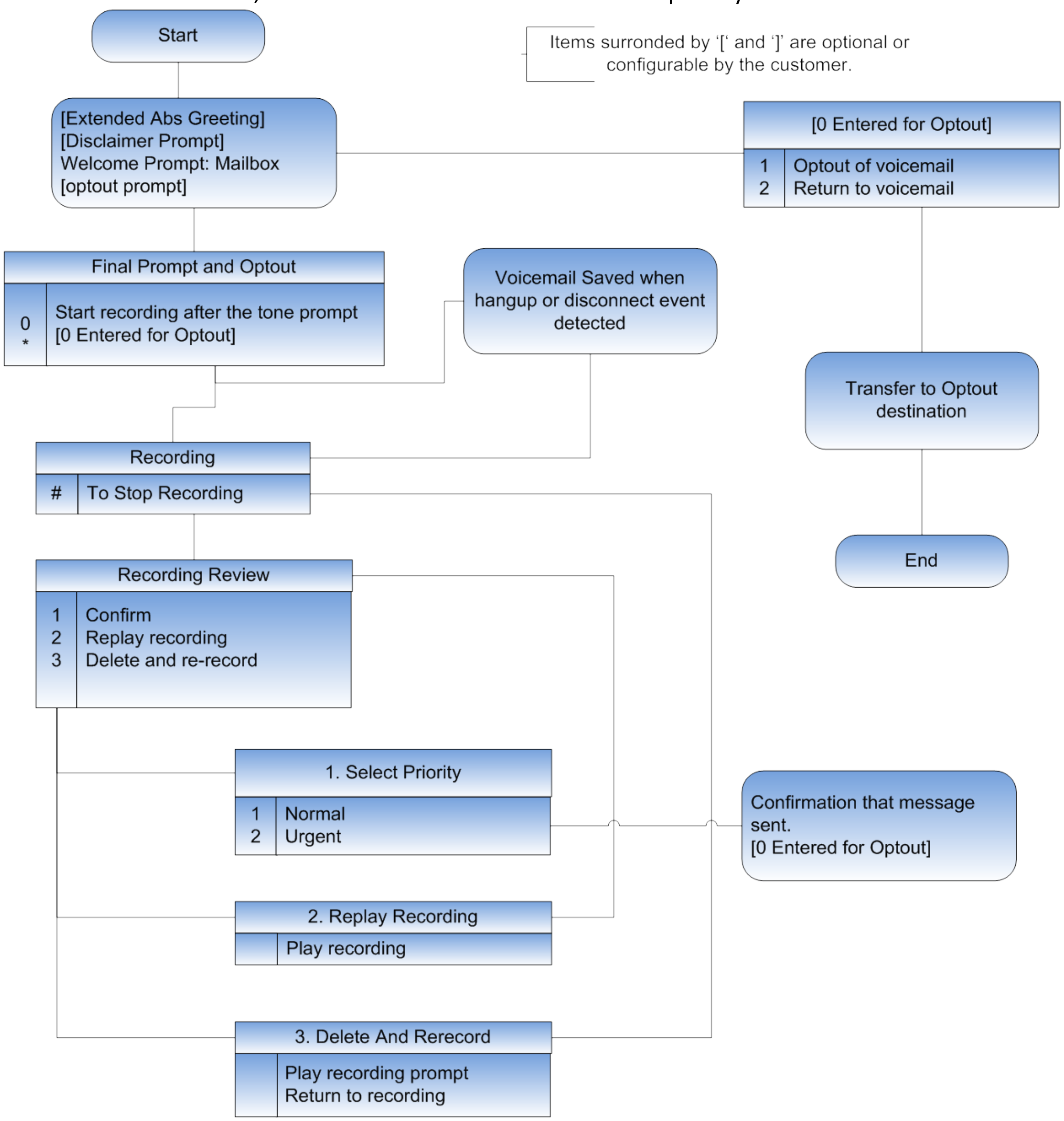


Figure 46 –Deposit Voice UI

13.2 *User experience – Voice UI during voicemail retrieval*

A sample Voice UI call flow for VM retrieval is shown below. In the main retrieval menu, the user will be greeted with the number of new and old messages on a personal mailbox configured on agent/DN.

In the main menu, the user could press an option to access group mailboxes. A further menu will be provided where the user can enter a group mailbox number or choose to list all the available mailboxes. This way the user could browse through both personal mailbox and group mailboxes in the same session.

If the user is dialing from his default Extension, only password validation is required and mailbox number is not requested. If the user is dialing from external device then the mail box number should be entered followed by the mail box password.

When a user does not have any group mailboxes then the group mail voice menu [6] will not be played.

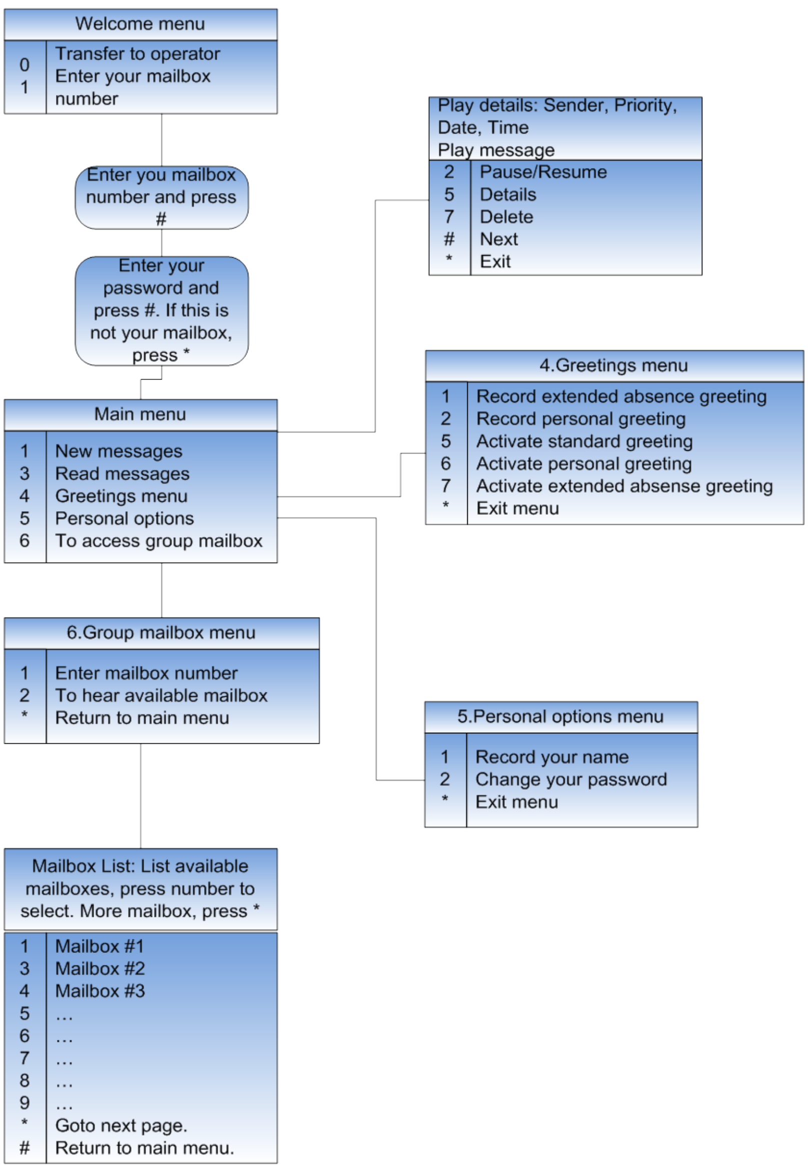


Figure 47 –Retrieval Voice UI

14 References

1. Feature Server configuration for SIP Cluster deployment:

http://docs.genesys.com/Documentation/SIPC/8.1.1/Solution/cfgFS

Online access is restricted; only logged-in users can access the online documentation. Please check with Genesys Support to obtain a PDF copy.

2. Feature Server Deployment Guide for standalone deployments:

http://docs.genesys.com/Documentation/FS/8.1.2/Deploy/fsdep

Online access is restricted, only logged-in users can access the online documentation. Please check with Genesys Support to obtain a PDF copy.