SIP Feature Server Deployment Guide

Configure voicemail

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To configure Feature Server for voicemail and group voicemail:

Voicemail

1. In Genesys Administrator, in the `rm` section of the `Options` tab of the Resource Manager (RM) application, set `sip-header-for-dnis` to `request-uri`.
2. Create a resource group of type `gateway` between SIP Server and RM.
3. Create a resource group of type `Media control platform` between MCP and RM.
4. Create an IVR profile using `Define New IVR Profile` and create the following options:
   - `service-type`,value `voicexml`
   - `initial-page-url`, value `[http: https:]//`*`FQDN1 or FS1 IP address`*`:`*`port`*`/fs`
     FQDN1 is the FQDN you created while configuring Feature Server applications, if your environment includes more than two Feature Server instances per SIP switch.
   - `alternatevoicexml`,value `[http: https:]//`*`FS(n+1) IP address`*`:`*`port`*`/fs`
     FS(n+1) IP address is the IP address of the "extra" Feature Server instance that is not included in *FQDN1*.
5. Create a DID group and add the IVR profile created in the step above:
   - Add a DID with the same value as the one set for the VoIP DN option in the previous step. The recommended option and value are `service-type` and `voicexml`.
6. To configure external voicemail deposit, configure a URS strategy that considers the DN/Agent dial-plan settings for business calls that land at a route point. In a TRouteCall to SIP Server, the strategy must respond to the value `full` with `UseDialPlan`.
7. Configure a URS strategy to retrieve an individual or group voicemail. This strategy must respond with a TRouteCall to the `P-Alcatel-CSBU` header.
   For example, on this routing point the strategy should generate the following TRouteCall:

```
message RequestRouteCall

AttributeThisDN '1519'

AttributeConnID 010a02020db9f03c

AttributeOtherDN 'gcti::voicemail'

AttributeLocation ''

AttributeExtensions [152] 00 04 00 00..

'SIP_HEADERS' 'P-Alcatel-CSBU'

'P-Alcatel-CSBU' 'call_condition=localdirect;categparty=internal;rd=unconditional'

AttributeDNIS '1519'

AttributeRouteType 1 (RouteTypeDefault)
```

Group voicemail

1. To configure group voicemail deposit and retrieval, log into GAX as an administrator (`*GAX IP address*:*port*/gax`).
2. Under **Configuration > Configuration Manager**, create a new agent group named `Agent_Group_1`. You can also use a virtual agent group, but **only** a virtual agent group based on skill expressions.
3. Create an option:
   - Section: `T-Server`
   - Name: `gvm_mailbox`
   - Value: `2200` (the mailbox number)
4. Add agents to the `Agent_Group_1` group.
5. Search for any user added to the `Agent_Group_1` group. Verify that the group mailbox is associated with that user.
6. Create a routing point (for example, 5000).
7. Configure a URS strategy to deposit a group voicemail when none of the agent group members answer the call. This strategy must respond with `TRouteCall` to `gcti::voicemail`.
   - Configure the `UseDialPlan Extensions` attribute with its value set to `false`.
   - Configure the `gvm_mailbox` with its value set to the group mailbox number.

     These configurations ensure that SIP Server does not use the SIP Feature Server dial plan and builds an `INVITE` to the mailbox number provided by the strategy.
8. Load the routing point with the above strategy.
9. Make a call to the route point loaded with this strategy. If no agent in the group answers, after the timeout the call reaches the group mailbox (2200).

Dial plan processing

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

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

- If `false`, the call is routed directly to the target mentioned in the TRouteCall.
- If `partial`, and the SIP Server dial plan is in effect, SIP Server performs number translation and authorization check on the target processes; if the SIP Feature Server dial plan is in effect, Feature Server performs the number translation and authorization check on the target processes.

- If `full`, and the SIP Server dial plan is in effect, SIP Server performs number translation, authorization, and call forwarding; if the SIP Feature Server dial plan is in effect, Feature Server performs number translation, authorization, and call forwarding.
- If `agentid`, Feature Server returns the agent ID of the agent and does not apply digit translation and destination rules. If the agent ID is not available, then Feature Server does not include the **agent-id** field in the XS response, and digits translation and destination rules are not applied.