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SIP Server Deployment Guide

Dial Plan enhancements including support for SIP Feature Server Dial Plan

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Contents

SIP Server now offers the option to use SIP Feature Server as an "external dial plan" as an alternative to the internal SIP Server dial plan. Each choice offers distinct advantages to consider when choosing which dial plan to use. (Note that dial plans may not be combined.)

Feature Server Dial Plan Highlights:

- User-based calling preferences for Call Waiting and Call Forwarding (including Find-Me-Follow-Me)
- Flexible rules with pattern matching logic for choosing a trunk for outgoing calls
- Enhanced support for deployments where voicemail mailboxes are assigned to users (but not to DNs)

SIP Server Dial Plan Highlights:

- Many supported parameters for advanced dial-plan rules, such as "onbusy", "type", "calltype", "clir", and more
- Native support by SIP Server (smaller footprint, less complexity if Feature Server is not required for the deployment)

SIP Server offers additional control over how a dial plan is applied to the destination of TRouteCall and/or to multi-site (ISCC) calls that are routed through an External Routing Point with two configuration options:

- The **rp-use-dial-plan** configuration option changes the default behavior of the dial plan to any one of the following:
  - SIP Server does not apply any dial plan.
  - SIP Server applies only the digit translation to a dial plan target.
  - SIP Server applies the digit translation and forwarding rules to a dial plan target.

    The **rp-use-dial-plan** option applies to both SIP Server and SIP Feature Server dial plans. If the **UseDialPlan** key-value pair is present in AttributeExtensions of TRouteCall, then it takes priority over the **rp-use-dial-plan** option.
- The **enable-iscc-dial-plan** option enables SIP Server to apply the dial plan to the target destination when a call is routed from an External Routing Point to a DN at the destination site.

Feature Configuration

Using Feature Server Dial Plan

1. Administer the SIP Feature Server dial plan as described in the SIP Feature Server Administration Guide.
2. Configure the SIP Server that is associated with the Feature Server by setting the following option in the **[TServer]** section of the SIP Server Application:
   - **dial-plan**—Set this option to `fs-dialplan`, as described in the SIP Feature Server Deployment Guide.

3. Under a SIP Server Switch object that is associated with the SIP Server, create a VOIP Service DN named **fs-dialplan** and configure these options:
   - **service-type**—Set this option to `extended`.
     Important: Ensure that you add the final slash character (/) to the end of each of the following URLs.
   - **url**—Set this option to `http://`*`FS Node:port/`*
     For n+1 High Availability (HA), add the following parameters:
     - **url-1** = `http://`*`FS Node2:port/`*
     - **url-2** = `http://`*`FS Node3:port/`*
     - **url-n** = `http://`*`FS Node_N:port/`*

     Important: A Feature Server's dial plan URL must be configured only on a VOIP Service DN that was created on the Switch controlled by the SIP Server that is connected to that particular Feature Server.
4. If required, configure the following options in the SIP Server Application object, the **[TServer]** section:
   - **rp-use-dial-plan**—Set this option to a value suitable for your environment.
   - **enable-iscc-dial-plan**—Set this option to `true` to enable SIP Server to apply the dial plan to multi-site (ISCC) calls that are routed through an External Routing Point.
5. (Optional) In a routing strategy, set the **UseDialPlan** key extension in TRouteCall. The key extension setting takes priority over configuration options.

Using SIP Server Dial Plan

1. Configure the Dial Plan feature as described in the Framework 8.1 SIP Server Deployment Guide.
2. If required, configure the following options in the SIP Server Application, the **[TServer]** section:
   - **rp-use-dial-plan**—Set this option to a value suitable for your environment.
   - **enable-iscc-dial-plan**—Set this option to `true` to enable SIP Server to apply the dial plan to multi-site (ISCC) calls that are routed through an External Routing Point.
3. (Optional) In a routing strategy, set the **UseDialPlan** key extension in TRouteCall. The key extension setting takes priority over configuration options.

Configuration Options

rp-use-dial-plan

Setting: **TServer** section, Application level
Default Value: `default`
Valid Values: `default, full, partial, false`
Changes Take Effect: Immediately

Specifies how SIP Server applies the dial plan:

- `default`—For a SIP Server dial plan, the same as the `false` value. For a Feature Server dial plan, the same as the `partial` value.

- `full`—The dial plan is applied to the destination of TRouteCall, including the digit translation and forwarding rules.
- `partial`—Only the digit translation is applied to a dial-plan target. Forwarding rules, such as forwarding on no answer (`ontimeout`), forwarding on busy (`onbusy`), forwarding on DND (`ondnd`), forwarding on no response (`onunreach`), and forwarding on not SIP registered (`onnotreg`) are not applied. Valid for both SIP Server and SIP Feature Server dial plans.
- `false`—No dial plan is applied to the destination of TRouteCall.

> **Important**
> If the SIP Server dial plan is used, SIP Server selects the dial plan assigned to the caller. This is the dial plan configured for the DN/Agent Login of the DN for internal calls, or the Trunk DN for inbound calls, or the Application-level option if no DN/Agent-Login-level dial plan is configured.

enable-iscc-dial-plan

Setting: **TServer** section, Application level
Default Value: `true`
Valid Values: `true, false`
Changes Take Effect: At the next call

Specifies whether SIP Server applies the dial plan to the agent destination of multi-site (ISCC) calls that are routed through an External Routing Point (**cast-type**=`route-notoken`), as follows:

- If set to `true`, the dial plan (full, including the digit translation and forwarding rules) is applied.
- If set to `false`, the dial plan is not applied.

This option must be configured on the remote (destination) site. SIP Server applies the dial plan when a call is routed from an External Routing Point to a DN at the destination site.

> **Important**
> SIP Server will still apply the dial plan to the External Routing Point destination of multi-site (ISCC) calls, and this will take priority over the agent DN destination dial-plan rule regardless of the setting of **enable-iscc-dial-plan**.

AttributeExtensions

Key: **UseDialPlan**
Type: String
Values: `full, partial, false`
Request: TRouteCall

Specifies how SIP Server applies the dial plan:

- `full`—The dial plan is applied to the destination of TRouteCall, including the digit translation and forwarding rules.
- `partial`—Valid for both SIP Server and SIP Feature Server dial plans. Only the digit translation is applied to a dial-plan target. Forwarding rules, such as forwarding on no answer (`ontimeout`), forwarding on busy (`onbusy`), forwarding on DND (`ondnd`), forwarding on no response (`onunreach`), and forwarding on not SIP registered (`onnotreg`) are not applied.
- `false`—No dial plan is applied to the destination of TRouteCall.

Important

- For ISCC calls, this extension is applied only to calls routed through an External Routing Point (**cast-type**=`route-notoken`).
- This extension is not supported in Business Continuity deployments.