Callback User's Guide

Enable Outbound Calls

5/8/2025

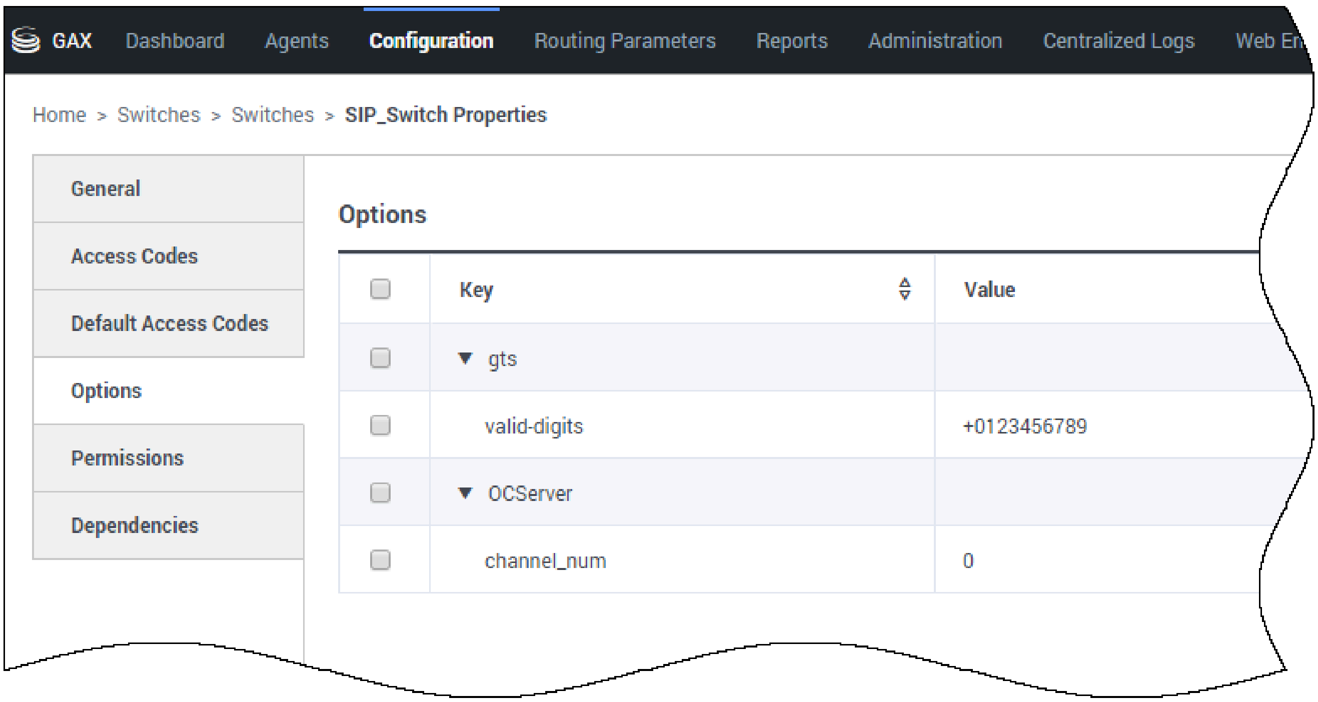
Enable Outbound Calls

Modified in 8.5.2

This configuration is required for voice scenarios. See the Genesys Voice Platform Deployment Guide for additional details.

Callback uses Media Server via SIP Server to make outbound calls. SIP Server communicates with Media Server using MSML and requires the following configuration to enable outbound calls.

Set Valid Digits (Optional)



Valid customer numbers should include a + sign if needed. If true, edit the `valid-digits` option in the **gts** section of your SIP Switch object:

```
[gts] valid-digits = +0123456789
```

Refer to ORS documentation for further details.

Set the Prefix Dial Out Option

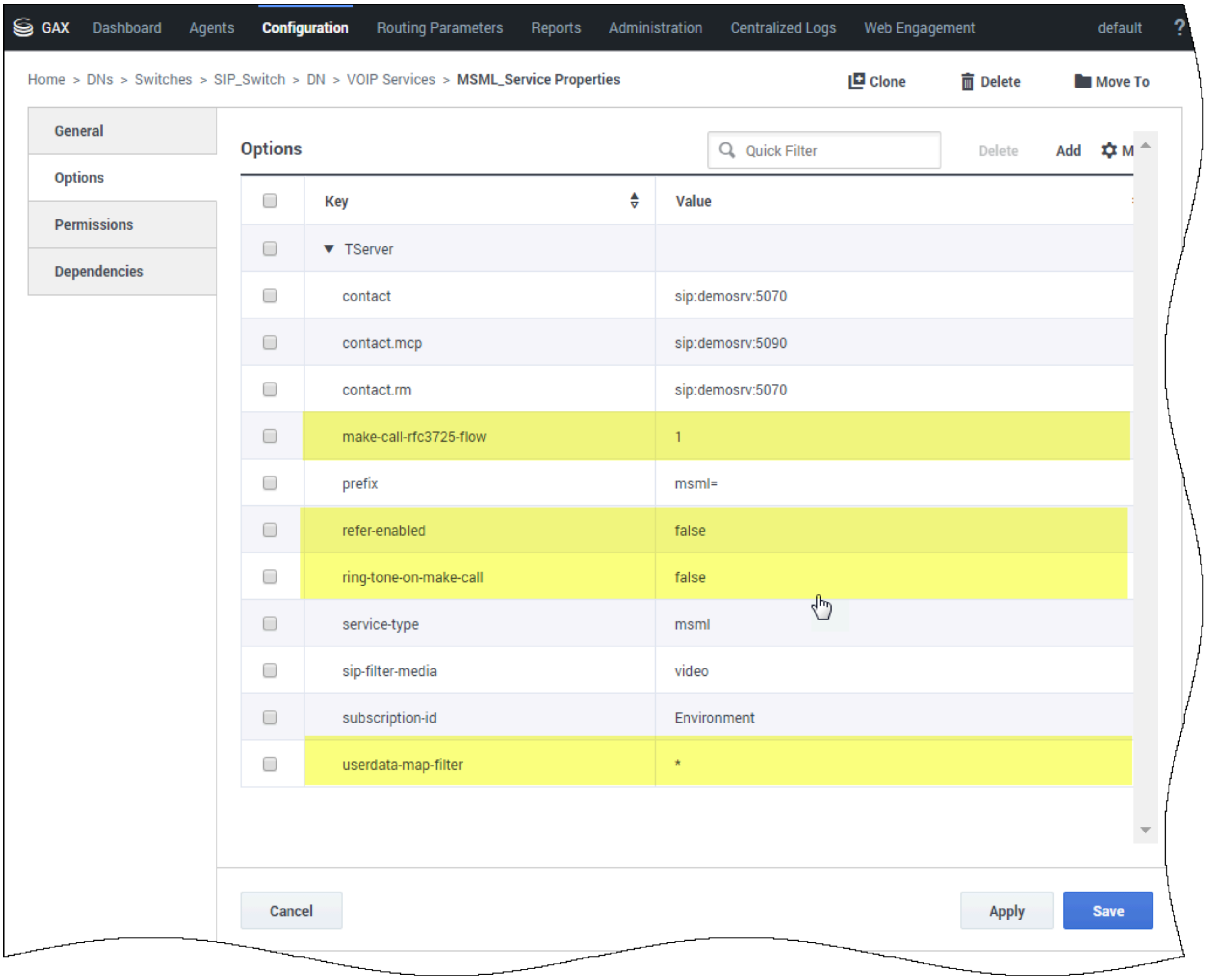
To make sure that the system will be able to call, configure the _prefix_dial_out option in your callback service with the Service Management UI.

Set the Default Country Option

By default starting in 8.5.108, callbacks for unreachable phone numbers and premium numbers are disabled (see _disallow_impossible_phone_numbers and _disallow_premium_phone_numbers options). Therefore, you must configure the _default_country option in your Callback service.

- Phone numbers are tested against Google's library for parsing, formatting, and validating international phone numbers.
- GMS 8.5.108.02 integrates version 7.2.8 and uses the Apache License Version 2.0.
- The list of premium numbers is available in Wikipedia.

How to Configure the MSML Service

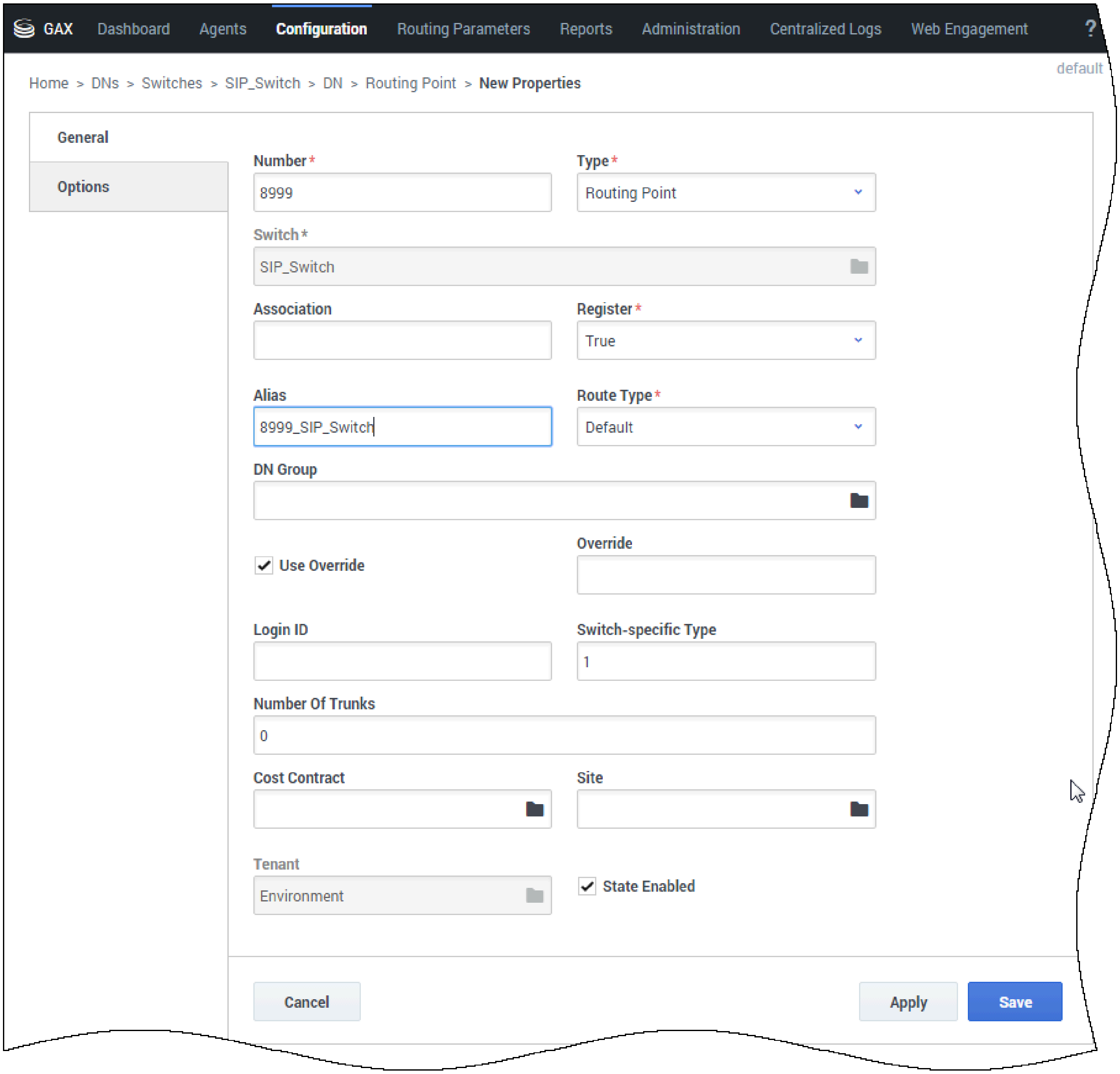


Open Genesys Administrator:

- Navigate to **Switching > Switches > SIP_Switch > DN > VOIP Service** and edit the `MSML_Service` object.
- Make sure that the following options are configured for **MSML_Service** to enable outbound:

```
make-call-rfc3725-flow=1
refer-enabled=false
ring-tone-on-make-call=false
userdata-map-filter=*
```

Create a Routing Point DN Dedicated to Outbound Calls



Navigate to **Switching > Switches > SIP_Switch > DN > Routing Point** and create a **Routing Point** object with, for instance, **name** set to 8999 and **alias** set to `8999_SIP_Switch`.

Then, use this DN to set the option _route_point in your Callback service. For example, `_route_point = 8999_SIP_Switch`.

> **Important**
>
> This routing point is dedicated to callback outbound calls and you must not configure any other strategies in its **Annex** tab.

How to Configure Calls Placed from Agent DNs

Added in 8.5.108.02

Outbound calls will be placed from agent DNs if you configure the following options in your callback service:

```
_userterminated_first_connect_party=AGENT
_agent_preview_via_rp=false
_agent_first_via_rp=false
```

Additionally, for agents involved in this callback scenario, set the following configuration in each agent DN Annex:

```
section TServer
refer-enabled=false
make-call-rfc3725-flow=1
```

Advanced Settings for Agents on External Switch

For a user-terminated callback with option `_userterminated_first_connect_party` set to `CUSTOMER`, the outbound call will be placed from the route point specified by option `_route_point`. This route point must be on a **SIP Server** type switch.

If the agents are on a different switch, you must set the callback service option `_ixn_redirect_confirm` to `false`. This is due to a limitation in how the routing to agent operates. In this scenario, a new call is created, the callback SCXML will not receive the events for this call and will not be able to confirm that the agent answered.

To handle the case where the agent does not answer, you can set the option `_ixn_redirect_hints` option to enable particular handling by the other switch. For example, you can set the following value for a Cisco switch.

```
_ixn_redirect_hints = {"extensions" : {"NO_ANSWER_TIMEOUT" : "5","NO_ANSWER_ACTION" :
"notready","NO_ANSWER_OVERFLOW" : "some DN"}}
```

This configuration enables a **no answer** timeout of 5 seconds, sets the agent to `Not Ready Upon No Answer`, and, upon no answer, routes the call to the DN specified.

Outbound Call Ringing Period Timeout (No Answer)

You can modify the ringing period timeout for the outbound call by changing the value of the _ixn_createcall_timeout option (in Advanced Parameters). If the call is not answered within this period, the outbound call attempt will end in error and will be retried later, according to the values set for _max_dial_attempts and _dial_retry_timeout. The default and maximum value of `_ixn_createcall_timeout` is 32 seconds.

The _call_timeguard_timeout option sets the timeout (msec) for the Call Progress Detection (CPD) result to be determined after the call is answered. If this timeout occurs, the result will be set to `human answer`. The recommended value for callback CPD is 5000 or greater.

The Orchestration application option `cti-transaction-timeout` sets the maximum time for the outbound call request to completed. Set this to a value greater than `_ixn_createcall_timeout`.

Increasing the Ringing Period Timeout

For some environments, a ringing period timeout of 32 seconds is not long enough for the call to reach voice mail. This is particularly true when calling a mobile phone, for example. A ringing period timeout greater than 32 seconds can be achieved by the following configuration.

- Set _ixn_createcall_timeout to the desired value (and the Orchestration application option cti-transaction-timeout accordingly, see above).
- For the following refer to "Increasing Ringing Period for Predictive Calls" in the SIP Server Deployment Guide.

Configure Voice over IP Service DN

- This is the DN used to issue the outbound call request to to Resource Manager / Media Server.
- In the Annex T-Server section, configure `predictive-timerb-enabled=false`.

Configure MCP Application

- Configure the `sip.timer_si` option to a value greater than the _ixn_createcall_timeout value.
- Configure the `sessmgr.acceptcalltimeout` option to a value greater than the `sip.timer_si` value. This prevents the MCP application from interfering with the SIP level timers.