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Callback User's Guide

User Terminated Agent First with Implicit Reservation

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User Terminated Agent First with Implicit Reservation

The Callback service first calls an agent with Implicit Reservation Information (ISCC) and starts an outbound consultation call with the customer. Then, Callback merges the two calls.

Call flow

This Callback scenario is an outbound service that goes through the following stages:

Start Callback

- Callback Service: Creates a call from trunk to the agent.
- Callback Service: Holds the call (implicit reservation).
- Callback Service: Starts a consultation call from trunk to the customer.
- Callback Service: Waits for the customer to accept the call.
- Callback service: Transfers and merges the held interaction with the consultation call to connect the agent with the customer.

Create your Scenario

In the **Admin UI > Services > Configured Services** tab, add a Callback service with User-Terminated-Immediate as the **Common Default Configuration** (see [Adding a Service](#) for details).

Enter a service name. This name is the callback execution name of your service and will be used in URLs to access this service. For example, if you set this name to agent-first, your service URL will be:

`http://host:port/{base-web-application}/service/callback/agent-first`

When you add this service and default configuration, many options are automatically populated with the appropriate default values.

Configuration Options

Predefined Values

These are the default values, which are automatically populated when using the pre-defined User-Terminated-Immediate service. You do not need to change these values.

| Option | Description |
|---|---|
| <code>_media_type=voice</code> | <p>Media type of the interaction that the service is expected to handle. This option enables URS to select an agent who has the appropriate media capabilities. This is a default value, automatically populated when using the predefined User-Terminated scenario. You do not need to change this value.</p> <p>This option is mandatory.</p> |
| <code>_wait_for_agent = true</code> | <p>True to wait for an agent to connect. If this option is set to true, the service will wait for the agent to initiate the interaction and to send the notification to the customer. If the option is set to false, the interaction can start right after the creation of the service instance. In voice scenarios, the access information will be returned immediately with the service ID.</p> <p>This option is mandatory.</p> |
| <code>_wait_for_user_confirm = false</code> | <p>True to wait for confirmation of the customer's availability. If this option is set to true, the service sends a push notification to the customer's device to get confirmation that the customer is ready to have a conversation with the agent. This scenario is possible only if the <code>_wait_for_agent</code> option is set to true.</p> |
| <code>_agent_preview = false</code> | <p>Enables Agent Preview. If set to true, the Preview Dialog with caller information is displayed to the agent.</p> |
| <code>_call_direction = USERTERMINATED</code> | <p>This is a default value, automatically populated when using the predefined User-Terminated scenario. You do not need to change this value.</p> <ul style="list-style-type: none">• If this option is set to <code>USERORIGINATED</code>, the customer's device will initiate the call to get connected to the agent.• If this option is set to <code>USERTERMINATED</code>, the agent or the system will initiate the call to contact the customer. |
| <code>_ttl = 86400</code> | <p>Duration (in seconds) for which the service will be kept in storage after the Desired Time is passed (Time To Live).</p> |

| Option | Description |
|--|---|
| | <p>Once expired, the service is removed from the system. For example, if you want the callbacks to be visible in the Service Management UI for one week past the execution time, then you should set 7 days of Time To Live, which means <code>_ttl=604800</code>.</p> <p>This option is mandatory.</p> |
| <code>_type = ors</code> | <ul style="list-style-type: none"> For Genesys Mobile Services-based services: builtin For Orchestration Server-based services: ors |
| <code>_provide_code= false</code> | <p>If true, returns a randomly generated code to be used for the authentication of the user originated (inbound) call.</p> <p>This option is mandatory.</p> |
| <code>_cpd_enable = false</code> | <p>Enables CPD. If this option is set to true, CPD will be performed on a callback made to the customer.</p> <ul style="list-style-type: none"> If CPD results in a human or silence detection, the call will be routed to the agent. If a fax is detected, the call will be disconnected and marked complete. If an answering machine is detected, the answering machine treatment is played. <p>This option is mandatory.</p> |
| <code>_use_debug_push_certificate = false</code> | Use debug certificates for the push notification provider |

Additional Required Options

You must enter a string value for the following options:

| Option | Description |
|---------------------------------------|--|
| <code>_agent_first_via_tg=true</code> | <p>If true, enables the call dialing from the trunk group (configured in the <code>_trunk_group</code> option) in the following user-terminated scenario. When the trunk group dials the call to the customer, it makes a call to the agent first where the agent preview mode is disabled, and the agent can consult the call to the customer. Finally, the agent can merge the two</p> |

| Option | Description |
|---|--|
| | calls. If the option is false, the call is dialed from the agent's DN. |
| <code>_trunk_group= "{TRUNK Route Point}@{Telephony Switch}"</code> | Trunk Group from which the system can create a user-terminated (outbound/inbound) call. If you configured <code>_agent_first_via_tg = true</code> , this option is mandatory. |
| <code>_route_point= "{Route Point}@{Telephony Switch}"</code> | Optional - if <code>_agent_first_via_tg=false</code> Route point from which the system can create a user-terminated (outbound) call. This option is mandatory. |
| <code>_agent_first_via_rp=true</code> | Enables dialing of the call from the route point (set in the <code>_route_point</code> option) in a user-terminated scenario <i>connect to agent first</i> where the agent preview mode is disabled. Otherwise, the call will be dialed directly from the agent's DN. This option is mandatory. |
| <code>_ixn_redirect_hints</code> | The extensions parameters of the JSON object must include values for implicit reservation information (ISCC). For example: <pre>{ "extensions": { "iscc-ar-duration": 15000, "iscc-ar-agent-dn": "", "iscc-ar-agent-id": "", "iscc-ar-place": "", "iscc-ar-priority": 1000, "iscc-ar-priority-1": 0, "iscc-ar-priority-2": 0 } }</pre> Note: <code>iscc-ar-agent-dn</code> , <code>iscc-ar-agent-id</code> , and <code>iscc-ar-place</code> options are set in the SCXML strategy. |

Additionally, edit your SIP server configuration and set `sip-enable-moh=false` in the T-Server section.

Troubleshooting

How to display the correct ANI on the agent and customer's end in a GMS Agent-First Scenario?

If you have a pool of external numbers and one of them is used for an outbound customer call, if you want the customer to get the correct ANI displayed, follow the instructions below.

For each external number of the pool, create a **DN** of type trunk in your configuration.

- Each trunk DN must have a unique prefix option, an empty replace-prefix option, and a unique cpn option.
- Other options can be identical for all of the trunks.

In this scenario, the strategy adds the matching trunk to the dialing number prefix with the proper cpn and prefix options. Then, after finding the matching trunk, the SIP Server removes the prefix option by applying the replace-prefix empty option. As a result, the SIP Server uses the cpn value as an invite for the username.