SIP Cluster Solution Guide

Disabling recording and monitoring of outbound calls

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Disabling recording and monitoring of outbound calls

Starting with release 8.1.103.28, per GDPR compliance, SIP Server can disable recording and monitoring of outbound calls in SIP Cluster deployments. This feature is only supported for TMakeCall requests made through a Routing Point and then routed to an external number.

Sample Call Flow

1. An agent dials to an external number from a Workspace Web Edition (WWE) desktop.
2. WWE translates the destination and submits a TMakeCall request.
3. SIP Server receives the TMakeCall request and queues the call on the Routing Point.
4. The strategy on the Routing Point issues a TRouteCall request containing **make-call-cpd-required** = `true` and **UseDialPlan** = `agentid` in AttributeExtensions.
5. SIP Server processes the call destination and routes the call to the Trunk Group DN. The agent is connected with the Trunk Group DN. This is similar to the engaging call in Active Switching Matrix (ASM) mode.
6. SIP Server issues an internal TApplyTreatment request for playing standard music to the agent.
7. SIP Server issues an internal TMakePredictiveCall request with the Call Progress Detection (CPD) extensions and the IVR profile to be selected.
8. Once the agent is connected, SIP Server instructs a media server to perform the CPD. This is similar to the outbound call in ASM mode.
9. After the CPD is completed, SIP Server issues TApplyTreatment to play the prepared IVR profile. This profile contains the VXML script that plays the recording or monitoring opt-out prompt and collects the user input on it.
10. SIP Server merges both call parties by issuing TMergeCall.
11. On the completion of the merge call process, the agent and called party are connected.

Feature Configuration

1. Configure a Trunk Group DN for CPD functionality in the SIP Cluster switch. In this Trunk Group DN, add the following configuration options:
    - **ivr-profile-name**: the IVR profile that points to the VXML page responsible for a recording and monitoring opt out prompt.
    - **beep**: whether to apply a beep tone before connecting the agent from the engaging call with the called party from the outbound call.
    - **predictive-call-timeout**: the value to be included as an AttributeTimeout value in the TMakePredictiveCall request.

- **cpd-extensions**: a list of CPD-related AttributeExtensions that are included in TMakePredictiveCall.

2. Set the **make-call-cpd-dn** option to the Trunk Group DN (created in Step 1) on the Routing Point DN, where TMakeCall requests submitted by agents are queued.
3. Set the **make-call-cpd-required** key extension to `true` in TRouteCall.
4. If required, on the Trunk Group DN, set **AnsMachine**, **FaxDest**, and/or **SilenceDest** keys in the Extensions attribute of TMakePredictiveCall requests. For this feature, a value of `connect` (in addition to `drop`) is supported. These keys override respective Application-level configuration options: **am-detected, fax-detected, silence-detected**.

AttributeExtensions

Key: **make-call-cpd-required**
Type: String
Valid Values: `true, false`
Request: TRouteCall

Specifies whether SIP Server applies CPD functionality to the specified TRouteCall destination.

- If set to `true`, SIP Server applies CPD functionality to the specified TRouteCall destination by converting this request to the Active Switching Matrix (ASM) mode call flow.
  Note: Genesys recommends setting the **UseDialPlan** extension to `agentid` when **make-call-cpd-required** is set to `true`. SIP Server connects an agent with the configured Trunk Group DN and for the specified destination CPD is done through TMakePredictiveCall. Thus, the dial plan is not required and only an agent ID provided by SIP Feature Server is added to the response.
- If set to `false`, SIP Server performs the routing request and no CPD functionality is applied.

If this extension is not specified, then CPD functionality is not applied.

Configuration Options

make-call-cpd-dn

Setting: Routing Point DN > [TServer] section
Default Value: An empty string
Valid Values: Any Trunk Group DN
Changes Take Effect: For the next call

When this option is set to a valid Trunk Group DN, SIP Server invokes CPD functionality for the call routed to an external number. The call is queued on a Routing Point when a TMakeCall request is made by an agent.

This option can be configured on the Routing Point DN in the SIP Cluster switch. Or, this option can be specified in the SIP Cluster DN (the DN with **service-type** set to `sip-cluster-nodes`) to apply to all Routing Point DNs under a SIP Cluster switch. The Routing Point DN setting takes precedence over the SIP Cluster DN setting.

ivr-profile-name

Setting: Trunk Group DN > [TServer] section
Default Value: An empty string
Valid Values: Any valid IVR profile
Changes Take Effect: For the next call

Specifies the name of the IVR profile that is added in the Request-URI sent to the Media Server.

Sample URI format: sip:msml@<RM:FQDN>;media-service=cpd;gvp-tenant-id=<ivr-profile-name>

beep

Setting: Trunk Group DN > [TServer] section
Default Value: `on`
Valid Values: `on, off`
Changes Take Effect: For the next call

When set to `on`, SIP Server applies a beep tone for the specified duration to the agent before processing the TMergeCall request. When set to `off`, SIP Server merges both the engaging and outbound call without playing a beep tone.

predictive-call-timeout

Setting: Trunk Group DN > [TServer] section
Default Value: `20`
Valid Values: `0-1800`
Changes Take Effect: For the next call

Specifies, in seconds, the value to be included as an AttributeTimeout value in the TMakePredictiveCall request. If this timeout expires before the call is answered, or if SIP Server receives a BYE message from the Media Server, SIP Server terminates the call.

This AttributeTimeout value is applied only when the **predictive-timerb-enabled** option is set to `false` in the Trunk Group DN. When **predictive-timerb-enabled** is set to `true`, SIP Server uses the 32-second timer and ignores the timeout specified in this option.

cpd-extensions

Setting: Trunk Group DN > [TServer] section
Default Value: An empty string
Valid Values: A comma-separated key-value pairs of TMakePredictiveCall without spaces
Changes Take Effect: For the next call

Specifies CPD-related AttributeExtensions to be included in TMakePredictiveCall. SIP Server applies default values to the non-configured extensions. For example:
`cpd-record=on,call_answer_type_recognition=positive_am_detection,cpd-on-connect=off,call_timeguard_timeout=2000,AnsMachine=connect,FaxDest=drop,SilenceDest=drop`

CPD AttributeExtensions

Key: **cpd-record**
Default Value: `off`
Valid Values: `on, off`
Request: TMakePredictiveCall
Enables or disables the recording of the call progress detection phase of the call.

Key: **call_answer_type_recognition**
Type: String
Default Value: `positive_am_detection`
Valid Values: `no_progress_detection, no_am_detection, positive_am_detection, full_positive_am_detection, accurate_am_detection, telephony_preset`
Request: TMakePredictiveCall
Specifies answer, answering machine, and fax detection settings when dialing using SIP Server.

Key: **cpd-on-connect**
Type: String
Default Value: `off`
Valid Values: `on, off`
Request: TMakePredictiveCall
Specifies when call progress analysis is started.

Key: **call_timeguard_timeout**
Type: String
Default Value: `3000`
Valid Values: Time interval in msec
Request: TMakePredictiveCall
Enables setting a timeout for post-connect call progress detection. The call is transferred to a queue when the timeout expires, regardless of the call result or the completion of call progress detection.

Key: **AnsMachine**
Type: String
Valid Values: `connect, drop`
Request: TMakePredictiveCall
Specifies whether SIP Server connects or drops a call if the CPD result shows that the predictive call reached an answering machine.

Key: **FaxDest**
Type: String
Valid Values: `connect, drop`
Request: TMakePredictiveCall
Specifies whether SIP Server connects or drops a call if the CPD result shows that the predictive call reached a fax machine.

Key: **SilenceDest**
Type: String
Valid Values: `connect, drop`
Request: TMakePredictiveCall
Specifies whether SIP Server connects or drops a call if the CPD result shows that silence is detected.

Feature Limitations

- This feature supports only direct outbound calls made by an agent to an external destination through TMakeCall requests.
- Outbound calls to an external destination through T-Library requests—TSingleStepConference, TInitiateConference, TSingleStepTransfer, TInitiateTransfer—are not supported.
- If a Trunk Group DN is configured with **call_answer_type_recognition**=`no_progress_detection`, CPD analysis is not done, but SIP Server still generates TApplyTreatment to play the prepared IVR profile. This is considered as misconfiguration.