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UC Connector Genesys Lync Integration Deployment Guide

SIP Server Configuration

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SIP Server Configuration

This section describes the list of **SIP Server** options that need to be configured for Lync / Skype for Business integration. No software changes are necessary in SIP Server in order to integrate with Lync / Skype for Business; however, a specific configuration is necessary as described below.

Set up MSML-Enabled Treatments

Set the following configuration option on the SIP Server Application level:
`TServer\msml-support = true`

Enable Music on Hold

To enable music to be played when the caller is on hold, set the following configuration option on the SIP Server Application level:
`TServer\sip-enable-moh = true`

Disable Early Media Support

In integration with Microsoft Lync / Skype for Business, early media support should be disabled in the SIP Server. For details, see **Current Limitations**.

On the SIP Server Application level, set the following configuration option:
`TServer\sip-enable-100rel = false`

Set Softswitch Properties

Purpose: To set up a softswitch DN for Lync or Skype for Business.

A softswitch configuration simplifies the common configuration required on Endpoint DNs. Use a DN of type **Voice over IP Service**, with `TServer\service-type = softswitch`. Create Extension DNs with the number corresponding to the Skype for Business Enterprise Agent's Phone number, with no sections added to them specifically.

These options are configured in **Genesys Administrator** under the **Switch > DNs** associated with a SIP Server. For detailed information on these options, consult the **SIP Server Deployment Guide**.

Under the **Options** tab, select **Advanced View (Annex)** and add the following options:

- `TServer/contact`—The contact should point to the Mediation Server host and port, which by default

uses port 5068 for TCP transport, and port 5067 for TLS transport.

- `TServer/dual-dialog-enabled`—The dual-dialog setting should be set to `false`. As with most PSTN devices, Mediation Server handles one call at a time. This makes SIP Server reuse the same dialog for the consultation call. This is also required to have a Media bypass applied to the consult call. Otherwise, by default, SIP Server sends a consultation call INVITE message without the SDP.
- `TServer/make-call-rfc3725-flow`—The call flow should be set to 1, to make third-party call control calls without sending an initial INVITE with the black hole SDP to the Mediation Server.
- `TServer/refer-enabled`—The REFER support is set to `false`, to make the RFC 3725 call flow effective.
- `TServer/reuse-sdp-on-reinvite`—The Mediation Server does not apply Media Bypass for calls, which go by Late Media. In order for a valid SDP from the caller to reach the Mediation Server, the SDP is reused. The value for this option should be set to `true`.
- `TServer/service-type`—The service type is set to `softswitch`.

Create a Trunk DN for Mediation Server

Purpose: To create a Trunk DN on SIP Server pointing to the IP address and port of the Mediation Server.

These options are configured in Genesys Administrator under the Switch > DNs associated with a SIP Server. For detailed information on these options, consult the [SIP Server Deployment Guide](#).

Under the Options tab, select `Advanced View (Annex)` and add the options as shown below:

- `TServer/contact`—The contact should point to the Mediation Server host and port, which by default uses port 5068 for TCP transport, and port 5067 for TLS transport.
- `TServer/dual-dialog-enabled`—The dual-dialog setting should be set to `false`.
- `TServer/make-call-rfc3725-flow`—The call flow should be set to 1.
- `TServer/prefix`—A string should contain any characters allowed in a user part of the SIP URI (according to RFC 3261). When configured on a Trunk DN, the value of this option is used by SIP Server to select the proper Trunk for an outgoing call. For each available Trunk, SIP Server compares the value of this option with the initial characters of the call's destination name; the Trunk with the longest possible match is selected.
- `TServer/refer-enabled`—The REFER support is set to `false`, to make the RFC 3725 call flow effective.
- `TServer/reuse-sdp-on-reinvite`—The Mediation Server does not apply Media Bypass for calls, which go by Late Media. In order for a valid SDP from the caller to reach the Mediation Server, the SDP is reused. The value for this option should be set to `true`.

Create a DN for MSML VoIP Service

Purpose: To provision GVP/Media Server for treatment of the Inbound Calls.

These options are configured in [Genesys Administrator](#) under the Switch > DNs associated with a SIP Server. For detailed information on these options, consult the [SIP Server Deployment Guide](#).

Under the Options tab, select Advanced View (Annex) and add the following options:

- TServer/contact—The contact should point to the Resource Manager IP address and port.
- TServer/cpd-capability—This should be set to mediaserver.
- TServer/make-call-rfc3725-flow—The call flow should be set to 1.
- TServer/prefix—This should be set to msml=.
- TServer/refer-enabled—The REFER support is set to false, in order to make the RFC 3725 call flow effective.
- TServer/ring-tone-on-make-call—This should be set to false.
- TServer/service-type—This should be set to msml.
- TServer/subscription-id—This should be set to the name of the tenant to which SIP Server switch belongs.

Create a DN for Recorder VoIP Service

Call recording can be configured using NETANN. This is how the recording test cases were tested during the SIP Server qualification tests with Skype for Business. Additional and more advanced recording capabilities can be configured, but were not tested officially during the qualification tests.

These options are configured in Genesys Administrator under the Switch > DNs associated with a SIP Server. For detailed information on these options, consult the [SIP Server Deployment Guide](#).

Under the Options tab, select Advanced View (Annex) and add the following options:

- TServer/contact—The contact (SIP URI) should point to the recorder server.
- TServer/request-uri—The value of the Request-URI address to be used in the INVITE message, if that address is different from the address where the message will be sent.
- TServer/service-type—The service type is set to recorder.

Create a Routing Point DN

When a call is made from a Skype for Business Client to a SIP Endpoint, the call progresses through the Skype for Business Server and lands on a SIP Server's Routing Point.

Create a DN of type Routing Point and set the Register property to true.