GENESYS™

Microsoft Skype for Business Deployment Guide

Using Telephony Objects

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Using Telephony Objects

Contents

Deploying T-Server manually requires that you configure a number of different objects in the Configuration Layer prior to setting up your T-Server objects and then installing T-Server. Use Configuration Manager or Genesys Administrator for creating telephony objects as described below.

1. Create a Switching Office object.
2. Create two Switch objects: one for T-Server and one for SIP Server.
3. Create devices for T-Server.
4. Create devices for SIP Server.
5. Create Agent Logins.

Switching Office

Configure a Switching Office object of type **SIP Switch** that accommodates your Switch object under Environment. Until you have done this, you cannot register a Switch object under Resources (single-tenant environment) or a Tenant (multi-tenant environment).

Switches

1. Configure two Switch objects for this deployment. Assign one **Switch** object to T-Server for Skype for Business, and assign another **Switch** object to SIP Server.
2. On the Annex tab of the Switch for T-Server, create the following sections:
    - **[conference-services]**
    - **[connector]**
    - **[log]**

   You will add configuration options as required for your deployment or functionality.
3. In the **[conference-services]** section, add two configuration options:
    - count
    - uri-pattern

> ### Important
>
> - Trusted Application Endpoints to be used for conference services must be created using the **count** and **uri-pattern** options.
> - Genesys recommends that any changes to options related to conference services are made during a scheduled maintenance window. After any change to these options, restart all Connectors to ensure that they are all operating with the same

> configuration.

4. If implementing the multi-site configuration, specify access codes for all switches on the network so that the call-processing applications can route and transfer calls between switches.
   Two types of access codes exist in a Genesys configuration:
   - Default access codes that specify how to reach this switch from any other switch in the Genesys environment.
   - Switch-to-switch access codes that specify how to reach a particular switch from any other switch. Use this type when either a nondefault dial number or routing type is required between any two locations. When a switch-to-switch access code is configured, its value has a higher priority than that of a default access code. See Multi-Site Support for step-by-step instructions.

Devices for T-Server

> **Important**
>
> 1. T-Server internally converts all DN names to lower case because only lower case URIs are supported by Skype for Business. If two or more DNs are configured with names that only differ in the case of some characters (like `sip:dn1` and `sip:DN1`), T-Server creates the LMS message MSG_TS_COMMON_DN_MISCONFIGURED and only one of these DNs will be in service. If such DNs are created, in order to avoid unpredictable behavior, you must delete all of them and then create a single DN with the correct name.
> 2. T-Server does not support On Demand registration. The **register** flag must be set to `true` for devices to be in service.

T-Server supports the following DN types that you configure in the Genesys configuration environment under the **Switch** object assigned to T-Server for Skype for Business:

- **Extension**—Create a DN of type Extension for each Skype for Business user with the number corresponding to the SIP URI of the Skype for Business user that was created and enabled for Skype for Business.
  For example:
  `sip:alice@skype.lab`
  `sip:andrew.smith@skype.lab`
- **Routing Point**—Create a DN of type Routing Point with the number corresponding to the SIP URI of the Trusted Application Endpoint.
  For example, `sip:RP1@skype.lab`

- **External Routing Point**—Create a DN of type External Routing Point with the number corresponding to the SIP URI of the Trusted Application Endpoint.
  For example, `sip:ERP1@skype.lab`
    - Configure the **Association** field with the dialable phone number assigned to a referenced line. This value will be used by SIP Server as the user part of the URI in SIP INVITE to reach the Skype for Business External Routing Point. This number corresponds to the `LineURI` parameter value in the corresponding Trusted Application Endpoint.
- **Virtual Queue**—Used for routing and reporting. Support of this DN has no specifics related to Skype for Business.
- **Voice Over IP Service**—Used for different services and configuration tasks. For example, for presence mapping configuration. Every DN of this type must have option **service-type** set to support a particular service, such as Music on Hold, Media Server, and so on.
- **Trunk Group**—Must be named **gcti::park**. It is used for parking of chat during media escalation.

You must configure all DNs that agents and their supervisors use in day-to-day contact center operation—so-called seat-related DNs—such as Extensions. All Extensions, Routing Points, External Routing Points, and Trunk Groups must be configured under the Microsoft Skype for Business object with a name corresponding to its SIP URI.

Devices for SIP Server

1. Under the **Switch** object assigned to SIP Server, create a DN of type **Trunk** for Microsoft Skype for Business.
2. In the **[TServer]** section of the Trunk DN, configure the following options:
    - **contact**—For correct communication with a Mediation Server or Mediation Server Pool, you must configure a SIP SRV record to be used in the contact properties. The SRV record protocol should be configured as `tcp` (typically port 5068), or in case of secure connections use `tls` (typically port 5067) to match the Mediation Server configuration.
      For example:

      contact=`genesys.com;transport=tcp`

      contact=`genesys.com;transport=tls`

      Refer to the “DNS Name Resolution” and “Transport Layer Security for SIP Traffic” chapters in the Framework 8.1 SIP Server Deployment Guide for more information.
    - **prefix**—Set this option to the initial digits of the number matching this SIP Server outbound trunk. For example, `001`.
    - **sip-proxy-headers-enabled**—Set this option to `false`.
3. Under the Switch object assigned to SIP Server, create a **Switch Access Code** to be used by SIP Server to reach Skype for Business. The Access Code must match the appropriate SIP Server outbound trunk. For example, `001`.

Agent Logins

It is recommended, but not mandatory, to create an Agent Login object for each agent. An administrator can, however, enforce strict policy for Agent Login and forbid using non-configured agent names. The Application option agent-strict-id in the **[TServer]** section controls this policy.

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