



This PDF is generated from authoritative online content, and is provided for convenience only. This PDF cannot be used for legal purposes. For authoritative understanding of what is and is not supported, always use the online content. To copy code samples, always use the online content.

Genesys Softphone Deployment Guide

Configuring Workspace Desktop Edition to use Genesys Softphone

5/9/2025

Configuring Workspace Desktop Edition to use Genesys Softphone

This article describes how to set up **Workspace Desktop Edition** (WDE) to work with Genesys Softphone instead of the Workspace SIP Endpoint to handle SIP Voice-over-IP calls. Genesys Softphone provides agents with interface elements in the WDE Voice Interaction window, including muting and volume control for both the microphone channel and the speaker channel of the selected audio device(s) on the agent workstation.

Tip

Any USB headset that is supported by the Windows Operating System works normally with Genesys Softphone.

Other SIP VoIP features included with Genesys Softphone: automatic gain control, beep tone, auto-answer, unavailable headset detection, log-level support, Real-time Transport Protocol (RTP) support, and speaking detection.

Workspace Desktop Edition deployment template

Workspace Desktop Edition provides three templates from which you can choose when you deploy WDE, one for the application, and two optional ones for the Workspace SIP Endpoint. Use the **Workspace Desktop Edition_SEP85_851.apd** template when you deploy WDE to use Genesys Softphone. After deploying WDE with this template, install Genesys Softphone and configure WDE as described in this topic.

USB headset configuration

You can use the following options to configure Workspace Desktop Edition to use a headset:

- `sipendpoint.policy.device.use_headset`: Specifies whether a USB head set is used for voice calls.
- `sipendpoint.policy.device.headset_name`: Specifies what type of USB headsets that you support in your environment. Use the "|" character to separate the names of different headsets if more than one type is supported. For example: 'Plantel|Jabra'.

If these options are set, and the corresponding USB headset is connected to the agent workstation at start-up time, the headset is selected automatically.

If the configured USB headset is not connected to the agent workstation, then the behavior depends on the following configuration option in the **interaction-workspace** section of the Workspace

Application object:

- `sipendpoint.headset-enforce-configured-usage`: This option specifies whether the agent must plug in the specified USB headset to complete logging in. When it is set to **false**, and if the headset is not plugged in at start-up time, the default audio devices that are available on the workstation, if any, are selected. When the option is set to **true**, and if the headset is not plugged in when the agent logs in, Workspace Desktop Edition waits for the headset to be plugged in before finalizing the login of the voice channel.

Genesys Softphone enables agents to switch to a preconfigured Not Ready state if the USB headset becomes unplugged after the agent has logged in to the SIP Voice Media. The agent will remain logged in to other eServices media such as email and chat.

Use the following configuration options in the **interaction-workspace** section of the Workspace Application object to control the behavior of this feature:

- `sipendpoint.headset-unplugged.not-ready-reason`: Specifies the Not Ready reason to be set to the SIP DN if the USB headset that is used by the agent becomes unplugged.
- `sipendpoint.headset-unplugged-set-not-ready`: Specifies whether the SIP DN of the agent is set automatically to **Not Ready** if the USB Headset that is used by the agent becomes unplugged.
- `sipendpoint.headset-replugged-set-ready`: Specifies whether the SIP DN of the agent is set automatically to **Ready** if the USB Headset that is used by the agent is plugged back in.

Genesys Softphone can be configured to retain volume setting of the USB headset between agent sessions.

Use the following configuration options in the `interaction-workspace` section of the Workspace Application object to control the behavior of this feature:

- `sipendpoint.retain-volume-settings-between-sessions`: Specifies whether the volume settings are saved for both microphone and speaker, when the agent logs out.

Important

When an agent logs in to Workspace Desktop Edition, the application creates a list of headsets that are plugged into the workstation. If an agent wants to use a different headset, they must exit Workspace, plug in the new headset, and then relaunch Workspace.

Session Border Controller configuration

Genesys Softphone supports connecting to SIP Server through a Session Border Controller (SBC) (refer to [Server 8.1 Deployment Guide](#)).

You must configure Workspace Desktop Edition to connect to SIP Server through an SBC instead of directly to SIP Server. If you do not configure Workspace Desktop Edition to connect to SIP Server by using an SBC, Genesys Softphone connects directly to SIP Server to register the agent SIP Endpoint by using the **TServer/sip-address** and **TServer/sip-port** options of the corresponding SIP Server application. However, when you configure Workspace Desktop Edition to connect by using an SBC

you decouple the address and port information that is sent to the SIP REGISTER from SIP Server and Workspace Desktop Edition obtains the host address and port from the configuration.

Configure the following two options in the **interaction-workspace** section of the Application, Tenant, Agent Group, or User object:

- `sipendpoint.sbc-register-address`: Specifies the address of your SBC to which Genesys Softphone connects.
- `sipendpoint.sbc-register-port`: Specifies the port on your SBC to which Genesys Softphone connects.

To set the Domain/Realm of your contact center instead of an IP when Genesys Softphone tries to register through a session border controller (SBC) device, set the value of the following two options to represent valid SIP domain names to specify a 'request-uri' in the SIP REGISTER request that is decoupled from the SIP Proxy address that is contacted:

- `sipendpoint.proxies.proxy0.domain`
- `sipendpoint.proxies.proxy1.domain`

Genesys SIP Proxy configuration

Genesys Softphone supports Genesys SIP Proxy. This feature enables SIP high availability (HA) without requiring a virtual IP address. Refer to the [SIP Proxy 8.1 Deployment Guide](#) for information about deploying and using SIP Proxy.

DNS SRV

You can configure the Genesys Softphone with one of the following:

- A standard DNS A-Record. The final URI form is: **sip:user@<host_fqdn>:<port>** where **<host_fqdn>** can be virtual and can represent multiple physical addresses behind the scenes, but the **:<port>** is mandatory, or
- A **DNS SRV** (Service record) as specified in the [Genesys SIP Proxy Architecture](#). The final URI form is: **sip:user@<host_fqdn>**

Limitations

- Genesys SIP Proxy currently does not support scenarios with switchover mid-transaction; therefore, call ANSWER and CANCEL probably will not work; however, BYE is fully supported.

Provisioning

Configure the connection to the SIP Proxy by using the following Workspace Desktop Edition configuration options:

- `sipendpoint.sbc-register-address`: Specifies the IP Address, Host Name of the SIP Proxy or the FQDN of the SIP Proxy farm.

- `sipendpoint.sbc-register-port`: Specifies the port of the SIP Proxy. For a SIP Proxy farm, all SIP Proxy instances must have the same SIP Port. For a DNS SRV, set this option to **0**.
- `sipendpoint.sbc-register-address.peer`: Specifies the IP Address, Host Name of the DR peer SIP Proxy or the FQDN of the DR peer SIP Proxy farm.
- `sipendpoint.sbc-register-port.peer`: Specifies the port of the DR peer SIP Proxy. In case of DNS SRV, set this option to **0**.

These options were introduced in Workspace Desktop Edition to support Session Border Controller; therefore, they are not specific to SIP Proxy.

Genesys recommends that you set the value of the `sipendpoint.policy.endpoint.rtp_inactivity_timeout` option to the default value of **30**.

Enabling an agent to use Genesys Softphone

Prerequisites

- A working knowledge of Genesys Administrator Extension.
- A Workspace **Application** object exists in the Configuration Database.

Procedure

To enable an agent to use the Genesys Softphone to send and receive SIP-based interactions, perform the following steps:

1. Install Genesys Softphone on the agent workstation in **connector mode**.
2. During installation, specify the **Connector port** and configure the port for either **http** or **https**.
3. In the **GenesysSoftphone.config** file, in the **connector** section, set the value of the **enable_sessionid** option to **0**.
4. Configure the options `sipendpoint.standalone.port` and `sipendpoint.standalone.protocol` according to the values specified for the Connector at Genesys Softphone installation time.
5. If required, configure the other SIP Endpoint options in the **interaction-workspace** section of the Workspace **Application** object (refer to the Genesys Softphone **configuration option reference** for a list of SIP Endpoint options and a description of how to configure them).
6. If required, configure SIP Endpoint for **SIP Proxy** support.
7. Set the following **TServer** section options for the DN of the Place to which the agent is logging in:
 - **sip-cti-control = talk,hold**
 - **voice = true**

Running Workspace and Genesys Softphone in a VDI

Environment

If the goal is to run Workspace and Genesys Softphone in a VDI environment, the `sipendpoint.standalone.vdi-detection-model` option must be set to **localhost** and Genesys Softphone must be installed using the appropriate VDI type.

In a Citrix environment where agents are using Windows sessions on the same Windows Server, the Virtual IP Loopback feature must be activated to allow successful communication between WDE and Genesys Softphone when multiple users are assigned to the same Windows Server. For more information, see the [Virtual IP and virtual loopback](#) page in the Citrix documentation.

In the IP Loopback configuration, register the following executables:

- **interactionworkspace.exe**
- **genesyssoftphone.exe**

Overriding Genesys Softphone option values

You can override the following Genesys Softphone options when you [provision Workspace Desktop Edition options](#):

- In the **proxies** and **system** domains, you can override all options.
- In the **policy** domain, you can override **endpoint**, **session**, and **device** sections.

Important

Options in the **Connector** section of the **policy** domain must be specified in the configuration file; these cannot be overridden. WDE implicitly controls configuration for options in the **Basic container** to enable single sign-on with WDE.

Overriding an option

To override a Genesys Softphone option when provisioning WDE, convert the option to the following format:

```
sipendpoint.<domain>.<section>.<setting>
```

For example, to override the **ringing_file** setting in the **session** section, configure **sipendpoint.policy.session.ringing_file** in your WDE provisioning. See the [options reference](#) for a list of Genesys Softphone settings.

Codec priority

To specify the priority of enabled codecs, use the **sipendpoint.codecs.enabled.audio** option in the Configuration Layer.

For example:

```
sipendpoint.codecs.enabled.audio = "iLBC,G722"
```

Or use the **enabled** section of the **codecs** domain in the **Softphone.config** configuration file to specify the order in which audio codecs are given priority.

For example:

```
<domain name="codecs">
  <section name="enabled">
    <setting name="audio" value="opus,pcmu,pcma,G722,iSAC/16000,G729"/>
  </section>
  <section name="PCMU/8000"/>
  <section name="PCMA/8000"/>
  <section name="G722/16000"/>
```

Warning

Any codec that is not explicitly included in the **enabled** section will not be used, even if the section for that codec is present in the configuration file or the Genesys Configuration Layer.

To use the **enabled** section of the "codecs" domain, follow these guidelines:

- Codec names are *case-insensitive*. You can omit the clock rate portion of the section name unless needed to discriminate between two sections with the same name. The clock rate portion must be provided for **iSAC**.
- Specify codec parameters as a comma-separated list in parenthesis after an equals sign. You can use abbreviations such as "pt" for "payload_type".
- If there are codec conflicts, the value in the **enabled** section takes precedence over value in corresponding codec section, regardless of whether those values come from the configuration file or the Genesys Configuration Layer. For example:

```
<setting name="audio" value="g729=(fmt='annexb=no'),opus=(pt=125),pcmu,pcma"/>
<setting name="video" value="h264=(pt=120,fmt='profile-level-id=420028')"/>
```

- If codec parameters are specified in-line (or a particular codec does not require any parameters, such as the PCMU and PCMA codecs), then a separate codec section is not necessary. In any case, codecs specified in the "enabled" section do not require presence of corresponding section to take effect.