# Genesys Softphone Deployment Guide

Genesys Softphone 8.5.2

2/11/2022

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# Genesys Softphone Deployment Guide

Welcome to the Genesys Softphone Deployment Guide. This document describes how to deploy and use the Genesys Softphone in your environment.

## Contacting Genesys Customer Care

If you have purchased support directly from Genesys, please contact Genesys Customer Care.

Before contacting Customer Care, please refer to the Genesys Care Program Guide for complete contact information and procedures.

## About This Document

The following list explains different features of the Genesys Softphone:

### Overview

This section introduces you to the features of the Genesys Softphone.

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This section explains how to deploy the Genesys Softphone.

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### How to Use

This section explains how to use the Genesys Softphone.

Using the Genesys Softphone

# Overview

## Architecture

The Genesys Softphone sits on top of the SIP Endpoint SDK for .NET to enable it to take advantage of the SIP-based third-party call control functionality.

The following diagram illustrates the Genesys Softphone architecture:

file:Genesys_Softphone_Architecture.png

## Features and Functionality

### DTMF

The Genesys Softphone supports Dual-Tone Multi-Frequency (DTMF) signalling according to the RFC 2833 standard for third-party call control.

After receiving a NOTIFY with DTMF event, the Softphone Endpoint generates DTMF signals.

DTMF can be sent by using one of the three possible methods:

- InbandRTP
- RFC 2833
- SIP INFO message

### Third-party Call Control

When the Genesys Softphone Endpoint has registered on the Genesys SIP Server, it will support the following third-party call control scenarios:

- Make a call
- Answer a call
- Hold and retrieve a call
- Single step and two step transfers
- Participate in a conference that is provided by the GVP
- Play DTMF signals.

## SIP Voice

The Genesys Softphone supports the following codecs for SIP signaling:

- PCMU/8000 (G.711/mu-law)
- PCMA/8000 (G.711/A-law)
- G722/16000
- iLBC/8000 (iLBC — internet Low Bitrate Codec)
- iSAC/32000 ((iSAC/32kHz) — internet Speech Audio Codec)
- iSAC/16000
- G729/8000
- OPUS/48000/2

# Deploying the Genesys Softphone

> **Important**
> The Genesys Softphone 8.5.2 release is built based on SIP Endpoint SDK 8.5.2 version.

This section describes how to install and configure the Genesys Softphone in your environment.

## Prerequisites

## Environment Prerequisites

### Supported Operating Systems

- Windows 8 32-bit and 64-bit
- Windows 7 32-bit and 64-bit
- Windows 10 32-bit and 64-bit

### Other Prerequisites

To work with the Genesys Softphone, you must ensure that your system meets the software requirements established in the Genesys Supported Operating Environment Reference Manual, as well as meeting the following minimum software requirements:

- Microsoft.VC80.CRT:4053 "Microsoft Visual C++ 2005 Service Pack 1 Redistributable Package ATL Security Update"
- Microsoft.VC90.CRT:4148 "Microsoft Visual C++ 2008 Service Pack 1 Redistributable Package ATL Security Update"
- Microsoft Visual C++ 2005 Service Pack 1 Redistributable Package MFC Security Update
- Visual C++ Redistributable for Visual Studio 2012 Update
- Windows Media Player for ringtone playback.

> **Important**

You must install the Windows Media Player on the desktop with the Genesys Softphone to play ringtones.

# Installation

## Installing the Genesys Softphone

To install the Genesys Softphone:

1. Double-click the `setup.exe` file that is located in the `<Genesys Softphone Install Directory>\windows\` directory. The **Genesys Installation Wizard** displays the **Welcome to the Installation** window.
2. Click **Next**. The **Choose Destination Location** window appears.
3. Click **Next** to accept the default destination folder, or click **Browse** to select another destination location. The **Ready to Install** window appears.
4. Select **Install**. The wizard installs the Genesys Softphone and all associated files in the selected directory and displays the **Installation Status** window. The installation might take several minutes.
5. At the **Installation Complete** window, select **Finish**.

# Silent Installation

## Installing the Genesys Softphone in Silent Mode

To install the Genesys Softphone in silent mode, use the Installation Wizard silent arguments as follows:

1. Update the `genesys_silent.ini` file, and add the path to the Genesys Softphone installation directory—for example, `InstallPath=<Genesys Softphone Installation Directory>`.
2. Execute the following command:
   `setup.exe /s /z"-s 'FullPathToGenesysSilentConfigurationFile' -sl 'FullPathToGenesysSilentResultFile'"` where:
   - /s—Specifies that the installation is running in InstallShield Silent Mode.
   - /z—Passes the Genesys Silent Mode silent parameters to the installation.
   - -s—Specifies the full path to the silent configuration file. The `<Full path to Genesys Silent Configuration file>` is optional. If the `<Full path to Genesys Silent Configuration file>` parameter is not specified, the installation uses the `genesys_silent.ini` file in the same directory

where the `setup.exe` is located.

> **Important**
> Enclose the value of the <`Full path to Genesys Silent Configuration file`> parameter by apostrophes (') if the paramater contains white symbols.

- -sl—Specifies the full path to the installation results file. If the <`Full path to Genesys Installation Result file`> parameter is not specified, the installation creates the `genesys_install_result.log` file in the <`System TEMP folder`> directory.

> **Important**
> Enclose the value of the <`Full path to Genesys Installation Result file`> parameter by apostrophes (') if the paramater contains white symbols.

The `InstallShield setup.exe` installation starter requires that:

- there is *no* space between the `/z` argument and quotation mark. For example, `/z"-s"` is valid, while `/z "-s"` is not valid.
- there *is* a space between the -s,-sl parameters and quotation mark. For example, `/z"-s c:\temp\genesys_silent.ini"` is valid, while `/z "-sc:\temp\genesys_silent.ini"` is not valid.

For example,
`setup.exe /s /z"-s 'C:\8.5.000.05\windows\b1\ip\genesys_silent.ini' -sl 'C:\GSP\silent_setup.log'"`.

- After executing this command, verify that the Genesys Softphone is installed in the `C:\`<`Genesys Softphone Installation Directory`>, and that the `silent_setup.log` file created in the `C:\GSP\` directory.

# Configuration

## Configuring the Genesys Softphone

The Genesys Softphone installation includes an example configuration file (<`Genesys Softphone Installation Directory`>`/Genesys Softphone/GenesysSoftphone/Softphone.config`) with configuration settings that are applied to the Softphone when it starts.

> **Important**
>
> You can make changes to the configuration file, but you must restart the Softphone before any of the changes take effect.

The configuration file is broken into containers. Each container is split into domains that are, in turn, split into sections that hold the settings for a group of parameters. The following configuration file examples illustrate these settings:

For the description and valid values of each parameter, see Configuration Options Reference.

## Basic Container

The Basic container sets the Genesys Softphone user's DNs and the protocol used.

```
<Container name ="Basic">
    <Connectivity user ="DN0" server="Server0:Port0" protocol="Protocol"/>
    <Connectivity user ="DN1" server="Server1:Port1" protocol=" Protocol"/>
  </Container>
```

## Genesys Container

The Genesys container sets the policy, endpoint, session, device, codecs, proxy, mailbox, system and security parameters.

```
<Container name ="Genesys">
    <settings version="1.0" xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
                            xmlns="http://www.genesyslab.com/sip">
      <domain name="policy">
        <section name="endpoint">
          <setting name="public_address" value=""/>
          <setting name="ip_versions" value="ipv4"/>
          <setting name="include_os_version_in_user_agent_header" value="1"/>
          <setting name="include_sdk_version_in_user_agent_header" value="1"/>
          <setting name="sip_port_min" value="5060"/>
          <setting name="sip_port_max" value="5080"/>
          <setting name="rtp_port_min" value="8000"/>
          <setting name="rtp_port_max" value="9000"/>
          <setting name="rtp_inactivity_timeout" value="30"/>      <!-- seconds -->
          <setting name="sip_transaction_timeout" value="4000"/> <!--  msecs  -->
          <setting name="gui_tabs" value="status,calls,devices"/>
          <setting name="gui_call_lines" value="3"/>
          <setting name="vq_report_publish" value="0"/>
          <setting name="vq_report_collector"
                   value="collector@SipServer.domain.invalid:5060;transport=udp"/>
          <setting name="webrtc_audio_layer" value="0"/>
        </section>
        <section name="session">
          <setting name="auto_answer" value="0"/>
          <setting name="dtmf_method" value="rfc2833"/>
          <setting name="agc_mode" value="1"/>
          <setting name="dtx_mode" value="0"/>
          <setting name="vad_level" value="1"/>
          <setting name="echo_control" value="0"/>
```

```
    <setting name="noise_suppression" value="0"/>
    <setting name="reject_session_when_headset_na" value="0"/>
    <setting name="sip_code_when_headset_na" value="480"/>
    <setting name="ringing_enabled" value="1"/>
    <setting name="ringing_timeout" value="0"/>
    <setting name="ringing_file" value="ringing.wav"/>
  </section>
  <section name="device">
    <!-- The device priority depends on the element order
         in this section (highest priority listed first) -->

    <!-- Headset -->
    <setting name="use_headset" value="0"/>
    <setting name="headset_name" value="HeadsetName0"/>
    <setting name="headset_name" value="HeadsetName1"/>
    <!-- Mic -->
    <setting name="audio_in_device" value="InDeviceName0"/>
    <setting name="audio_in_device" value="InDeviceName1"/>
    <!-- Speaker -->
    <setting name="audio_out_device" value="OutDeviceName0"/>
    <setting name="audio_out_device" value="OutDeviceName1"/>
  </section>
</domain>
<domain name="codecs">
  <!-- The codec priority depends on the element order
       in this section (highest priority listed first) -->
  <section name="PCMU/8000"/>
  <section name="PCMA/8000"/>
  <section name="G722/16000"/>
  <section name="iLBC/8000">
    <setting name="payload_type" value="102"/>
  </section>
  <section name="iSAC/16000">
    <setting name="payload_type" value="103"/>
  </section>
  <section name="iSAC/32000">
    <setting name="payload_type" value="104"/>
  </section>
  <section name="g729/8000">
    <setting name="fmtp" value="annexb=yes"/>
  </section>
  <section name="opus/48000/2">
    <setting name="payload_type" value="120"/>
  </section>
</domain>
<domain name="proxies">
  <section name="proxy0">
    <setting name="reg_timeout" value="1800"/>
    <setting name="reg_interval" value="10"/>
    <setting name="password" value="<password>"/>
    <setting name="display_name" value="Genesys Softphone"/>
    <section name="nat">
      <setting name="ice_enabled" value="0"/>
      <setting name="stun_server" value="stun.example.com"/>
      <setting name="stun_server_port" value="3478"/>
      <setting name="turn_server" value="turn.example.com"/>
      <setting name="turn_server_port" value="3478"/>
      <setting name="turn_user_name" value="user"/>
      <setting name="turn_password" value="password"/>
      <setting name="turn_relay_type" value="1"/>
    </section>
  </section>
  <section name="proxy1">
```

```
          <setting name="reg_timeout" value="1800"/>
          <setting name="reg_interval" value="10"/>
          <setting name="password" value="<password>"/>
          <setting name="display_name" value="Genesys Softphone"/>
          <section name="nat">
            <setting name="ice_enabled" value="0"/>
            <setting name="stun_server" value="stun.example.com"/>
            <setting name="stun_server_port" value="3478"/>
            <setting name="turn_server" value="turn.example.com"/>
            <setting name="turn_server_port" value="3478"/>
            <setting name="turn_user_name" value="user"/>
            <setting name="turn_password" value="password"/>
            <setting name="turn_relay_type" value="1"/>
          </section>
        </section>
      </domain>
      <domain name="system">
        <section name="diagnostics">
          <setting name="logger_type" value="file"/>
          <setting name="log_file" value="logs/Softphone.log"/>
          <setting name="enable_logging" value="1"/>

          <!-- The levels: 0=Fatal 1=Error 2=Warning 3=Info(default) 4=Debug -->
          <setting name="log_level" value="3"/>
          <setting name="log_options_provider" value="gsip=2, webrtc=(error,critical)"/>
          <setting name="log_segment" value="10 MB"/>
          <setting name="log_expire" value="10"/>
          <setting name="log_time_convert" value="local"/>
          <setting name="log_time_format" value="time"/>
        </section>
        <section name="security">
          <setting name="cert_file" value="<valueOfCertificateThumbprint>"/>
          <setting name="use_srtp" value="allowed"/>
        </section>
        <section name="media">
          <setting name="ringing_file" value="ringing.wav"/>
        </section>
      </domain>
    </settings>
  </Container>
```

## Configuring the Agent's DN

Set the following `TServer` section option for the DNs of the Place to which the agent is logging in:

- `sip-cti-control` = `talk,hold,dtmf`

> **Important**
> This option is mandatory to use third-party call control on the SIP device.

For information about configuring DN objects, see the Genesys Administrator Extension Help.

# Configuring SIP Server

Genesys recommends setting the following SIP Server options:

- `dual-dialog-enabled=true` (default value)
- `make-call-rfc3725-flow=1` (allows for better and/or simpler codec negotiation)
- `ring-tone-on-make-call=true` (default value)
- `use-register-for-service-state=true`

For more information about these options, see the SIP Server Deployment Guide.

## Suppressing the Ringtone

The ringtone is generated for all incoming call to the Genesys Softphone. To suppress the ringtone for third-party call control for the originating DN, configure the following SIP Server option:

- `make-call-alert-info=<urn:alert:service:3pcc@genesys>`

or

- `make-call-alert-info=<file://null>;service=3pcc`

> **Important**
>
> If at least one Genesys Softphone in the contact center is configured with the `ringing_enable` option set to 1, the SIP Server `make-call-alert-info` option should be set to one of the specified values.

# Configuration Options Reference

This section lists and describes, by container and then by domain, the configuration settings found in the `<Genesys Softphone Installation Directory>/Genesys Softphone/GenesysSoftphone/Softphone.config` file. For an example of the configuration file, see Configuring Genesys Softphone.

## Basic Container

> **Important**
> Your environment can have up to six SIP URIs (Connectivity sections) that represent six endpoint connections with SIP Server.

| Domain | Section | Setting | Default Value | Description |
|---|---|---|---|---|
| | Connectivity | user | | The first user's DN extension as configured in the configuration database. Included in the SIP URI—for example, <sip:**DN0**@serverHostName0:por |
| | | server | | The SIP Server or Proxy location for the first user. Included in the SIP URI—for example, <sip:DN0@**serverHostName0:p** |
| | | protocol | | The transport procotcol for the first user. For example, UDP, TCP, or TLS. |
| | | For more information, see the Basic Container description in the SIP Endpoint SDK for .NET Developer's Guide. | | |

## Genesys Container

The second Container ("Genesys") holds a number of configurable settings that are organized into domains and sections. These settings do not have to be changed, but can be customized.

An overview of the settings in this container and the valid values for these settings is provided here:

| Domain | Section | Setting | Values | Description |
|---|---|---|---|---|
| **policy** | | | | |
| | **endpoint** | | | |
| | | include_os_version_in_user_agent_header | Number | If set to 1, the user agent field includes the OS version the client is currently running on. Default: 1. |
| | | gui_call_lines | Number from 1 to 7 | This option controls the number of phone lines in the First Party Call Control tab.<br><br>**Valid values:** Integer between 1 and 7<br><br>**Default value:** 3 |
| | | gui_tabs | Comma-separated list of tab names | This option controls what tabs are shown in the GUI and their order.<br><br>**Valid values:** Comma-separated list of tab names in any order. The tab names are `status`, `calls`,and `devices`. Names may be shortened to `stat`, `call`, and dev. The value is case-sensitive. This option ignores unrecognizable and duplicate tab names. If the setting is present but has an incorrect value, the value will fall back to the single tab `status`.<br><br>**Default value:** `status,calls,devices` |
| | | include_sdk_version_in_user_agent_header | Number | If set to 1, the user agent field includes the SDK version the client is currently running on. Default: 1. |
| | | ip_versions | IPv4 | A value of IPv4 means that the |

| Domain | Section | Setting | Values | Description |
|---|---|---|---|---|
| | | | IPv6<br>IPv4,IPv6<br>IPv6,IPv4<br>empty | application selects an available local IPv4 address; IPv6 addresses are ignored.<br><br>A value of IPv6 means that the application selects an available local IPv6 address; IPv4 addresses are ignored.<br>A value of IPv4,IPv6 or an empty value means that the application selects an IPv4 address if one exists. If not, an available IPv6 address is selected.<br>A value of IPv6,IPv4 means that the application selects an IPv6 address if one exists. If not, an available IPv4 address is selected.<br>Default: IPv4.<br>NOTE: This parameter has no effect if the public_address option specifies an explicit IP address. |
| | | public_address | String | Local IP address or Fully Qualified Domain Name (FQDN) of the machine. This setting can be an explicit setting or a special value that the GSP uses to automatically obtain the public address.<br><br>**Valid Values:**<br>This setting may have one of the following explicit values:<br><br>• An IP address. For example, 192.168.16.123 for IPv4 or FE80::0202:B3FF:FE1E:8329 for IPv6.<br>• A bare host name or fully qualified |

| Domain | Section | Setting | Values | Description |
| --- | --- | --- | --- | --- |
| | | | | domain name (FQDN). For example, `epsipwin2` or `epsipwin2.us.example.com`.<br><br>This setting may have one of the following special values:<br><br>• `$auto`—The GSP selects the first valid IP address on the first network adapter that is active (status=up) and has the default gateway configured. IP family preference is specified by the **policy.endpoint.ip_versions** setting.<br><br>• `$ipv4` or `$ipv6`—Same behavior as the `$auto` setting but the GSP restricts the address to a particular IP family.<br><br>• `$host`—The GSP retrieves the standard host name for the local computer using the `gethostname` system function.<br><br>• `$fqdn`—The GSP retrieves the fully qualified DNS name of the local computer. |

| Domain | Section | Setting | Values | Description |
|---|---|---|---|---|
| | | | | The GSP uses the `GetComputerNameEx` function with parameter `ComputerNameDnsFullyQualified`.<br>• An adapter name or part of an adapter name prefixed with `$`. For example, `$Local Area Connection 2` or `$Local`. The specified name must be different from the special values `$auto`, `$ipv4`, `$host`, and `$fqdn`.<br>**Default Value:** Empty string which is fully equivalent to the `$auto` value.<br>If the value is specified as an explicit host name, FQDN, or `$fqdn`, the `Contact` header includes the host name or FQDN for the recipient of SIP messages (SIP Server or SIP proxy) to resolve on their own. For all other cases, including `$host`, the resolved IP address is used for `Contact`. The value in SDP is always the IP address. |
| | | rtp_inactivity_timeout | Number | Timeout interval for RTP inactivity. Valid values are positive integers. A value of `0` means that this feature is not activated. A value 1 or higher indicates the inactivity timeout interval in |

| Domain | Section | Setting | Values | Description |
|---|---|---|---|---|
| | | | | seconds. Default: 0. Suggested values: 1 through 150. |
| | | rtp_port_min | Number | The integer value representing the minimum value for an RTP port range. Must be within the valid port range of 9000 to 65535. If the minimum and maximum values are not specified or are set to an invalid value, the default minimum (9000) and maximum (minimum value + 999) are used. Setting the minimum to a value that is larger than the maximum is considered an error and will result in a failure to initialize the endpoint. |
| | | rtp_port_max | Number | The integer value representing the maximum value for an RTP port range. Must be within the valid port range of 9000 to 65535. If the minimum and maximum values are not specified or are set to an invalid value, the default minimum (9000) and maximum (minimum value + 999) are used. Setting the maximum to a value that is less than the minimum is considered an error and will result in a failure |

| Domain | Section | Setting | Values | Description |
|---|---|---|---|---|
| | | | | to initialize the endpoint. |
| | | sip_port_min | Number | The integer value representing the minimum value for a SIP port range. Must be within the valid port range of 1 to 65535. If the minimum and maximum values are not specified or are set to an invalid value, the default minimum (5060) and maximum (minimum value + 6) are used. Setting the minimum to a value that is larger than the maximum is considered an error and will result in a failure to initialize the endpoint. |
| | | sip_port_max | Number | The integer value representing the maximum value for a SIP port range. Must be within the valid port range of 1 to 65535. If the minimum and maximum values are not specified or are set to an invalid value, the default minimum (5060) and maximum (minimum value + 6) are used. Setting the maximum to a value that is less than the minimum is considered an error and will result in a failure to initialize the endpoint. |

| Domain | Section | Setting | Values | Description |
|---|---|---|---|---|
| | | sip_transaction_timeout | Number | SIP transaction timeout value in milliseconds. Valid values are 1 through 32000, with a default value of 4000. The recommended value is 4000. |
| | | vq_report_collector | | See SIP Endpoint SDK for .NET—Producing RTCP Extended Reports |
| | | vq_report_publish | | See SIP Endpoint SDK for .NET—Producing RTCP Extended Reports |
| | | webrtc_audio_layer | 0<br>1<br>2 | Valid values:<br>0—the audio layer is defined by environment variable "GCTI_AUDIO_LAYER"<br>1—Wave audio layer is used<br>2—Core audio layer is used |
| | **session** | | | |
| | | agc_mode | 0<br>1 | If set to 0, AGC (Automatic Gain Control) is disabled; if set to 1, it is enabled. Default: 1. Other values are reserved for future extensions. This configuration is applied at startup, after which time the agc_mode setting can be changed to 1 or 0 from the main sample application.<br>NOTE: It is not possible to apply different AGC settings for different channels in multi-channel scenarios. |

| Domain | Section | Setting | Values | Description |
|---|---|---|---|---|
| | | auto_answer | Number | If set to 1, all incoming calls should be answered automatically. |
| | | dtmf_method | Rfc2833<br><br>Info<br>InbandRtp | Method to send DTMF |
| | | echo_control | 0<br>1 | Valid values: 0 or 1. If set to 1, echo control is enabled. |
| | | noise_suppression | 0<br>1 | Valid values: 0 or 1. If set to 1, noise suppresion is enabled. |
| | | dtx_mode | Number | Valid values: 0 or 1. If set to 1, DTX is activated. |
| | | reject_session_when_headset_na | Number | Valid values: 0 or 1. If set to 1, the GSP should reject the incoming session if a USB headset is not available. |
| | | sip_code_when_headset_na | Number | Defaul Value: `480`<br><br>If a valid SIP error code is supplied, the GSP rejects the incoming session with the specified SIP error code if a USB headset is not available. |
| | | vad_level | Number | Sets the degree of bandwidth reduction. Valid values: 0 – 3 — from 0 (conventional VAD) to 3 (aggressive high). |
| | | ringing_enabled | Number | Valid values: 0, 1, 2, or 3.<br><br>0 = None, disable ringtone<br>1 = Play ringtone through system default device only. Configure media in `system.media.ringing_file`.<br>2 = Play ringtone |

| Domain | Section | Setting | Values | Description |
|---|---|---|---|---|
| | | | | through communication device (headset) only. Configure media in `policy.session.ringing_file`.<br>3 = Play ringtone through both devices at the same time.<br>Default Value: 1<br>Specifies whether to enable the ringing tone and on which device to play the media file. |
| | | ringing_timeout | Number | Valid Values: Empty, 0, or a positive number<br><br>Default Value: 0<br>Specifies the duration, in seconds, of the ringing tone. If set to `0` or if the value is empty, the ringing time is unlimited. |
| | | ringing_file | String | Valid values: Empty or the path to the ringing sound file for the audio out device (headset). The path may be a file name in the current directory or the full path to the sound file.<br><br>Default Value: `ringing.wav`<br>Specifies the audio file that is played in the audio out device (headset) when the ringing tone is enabled with the `ringing_enabled` option.<br>Note that WebRTC does not support MP3 playback. The ringtone file for built-in ringing should be a RIFF (little-endian) WAVE file using one of the following formats:<br><br>kWavFormatPcm = 1, PCM, each sample of size bytes_per_sample<br>kWavFormatALaw = 6, 8-bit ITU-T G.711 A-law<br>kWavFormatMuLaw = 7, 8-bit ITU-T G.711 |

| Domain | Section | Setting | Values | Description |
|---|---|---|---|---|
| | | | | mu-law<br><br>Uncompressed PCM audio must 16 bit mono or stereo and have a frequency of 8, 16, or 32 KHZ. |
| | **device** | | | |
| | | audio_in_device<br><br>For more information, see SIP Endpoint SDK for .NET—Audio Device Settings | String | Microphone device name |
| | | audio_out_device | String | Speaker device name |
| | | headset_name | String | The name of the headset model |
| | | use_headset | Number | Valid values: 0 or 1. If set to 0, the audio devices specified in `audio_in_device` and `audio_out_device` are used by the SDK. If set to 1, the SDK uses a headset as the preferred audio input and output device and the audio devices specified in `audio_in_device` and `audio_out_device` are ignored. |
| **codecs**<br>— See SIP Endpoint SDK for .NET—Working with Codec Priorities | | | | |
| **proxies** | | | | |
| | **proxy<*n*>** | | | |
| | | display_name | String | Proxy display name |
| | | password | String | Proxy password |
| | | reg_interval | Number | The period, in seconds, after which the endpoint starts a new registration cycle |

| Domain | Section | Setting | Values | Description |
| --- | --- | --- | --- | --- |
| | | | | when a SIP proxy is down. Valid values are integers greater than or equal to `0`. If the setting is empty or negative, the default value is `0`, which means no new registration cycle is allowed. If the setting is greater than `0`, a new registration cycle is allowed and will start after the period specified by `regInterval`.<br>**Important**<br>The re-registration procedure uses a smaller timeout (half a second) for the first re-try only, ignoring the configured `reg_interval` setting; the `reg_interval` setting is applied to all further retries. |
| | | reg_match_received_rport | Number | Valid Values: `0` or `1`<br>Default Value: `0`<br>This setting controls whether or not SIP Endpoint SDK should re-register itself when receiving a mismatched IP address in the `received` parameter of a REGISTER response. This helps resolve the case where SIP Endpoint SDK for .NET has multiple network interfaces and obtains the wrong local IP address. A value of 0 (default) disables this feature and a value of 1 enables re-registration. |
| | | reg_timeout | Number | The period, in seconds, after which registration should expire. A new REGISTER request will be |

| Domain | Section | Setting | Values | Description |
| --- | --- | --- | --- | --- |
| | | | | sent before expiration. Valid values are integers greater than or equal to `0`. If the setting is `0` or `empty/null`, then registration is disabled, putting the endpoint in standalone mode. |
| | **nat** | | | |
| | | ice_enabled | Boolean | Enable or disable ICE |
| | | stun_server | String | STUN server address. An empty or null value indicates this feature is not being used. |
| | | stun_server_port | String | STUN server port value |
| | | turn_password | Number | Password for TURN authentication |
| | | turn_relay_type | Number | Type of TURN relay |
| | | turn_server | String | TURN server address. An empty or null value indicates this feature is not being used. |
| | | turn_server_port | String | TURN server port value |
| | | turn_user_name | String | User ID for TURN authorization |
| **system** | | | | |
| | **diagnostics** | | | |
| | | enable_logging | Number | Valid values: 0 or 1. Disable or enable logging. |
| | | log_file | String | Log file name, for example, `SipEndpoint.log` |
| | | log_level | Number | Valid values: 0 – 4. Log levels: 0 = "Fatal"; 1 = "Error"; 2 = "Warning"; 3 = "Info"; 4 = |

| Domain | Section | Setting | Values | Description |
|---|---|---|---|---|
| | | | | "Debug". |
| | | log_options_provider | String | Valid values for webrtc = (warning, state, api, debug, info, error, critical). For example: `gsip=2, webrtc=(error,critical)` |
| | | logger_type | `file` | If set to `file`, the log data will be printed to the file specified by the `log_file` parameter. |
| | | log_segment | `false`<br>Number<br>Number in KB,MB, or `hr` | Valid Values:<br>`false`: No segmentation is allowed<br><`number`> or <`number`> KB: Size in kilobytes<br><`number`> MB: Size in megabytes<br><`number`> hr: Number of hours for segment to stay open<br>Deafult Value: 10 MB<br>Specifies the segmentation limit for a log file. If the current log segment exceeds the size set by this option, the file is closed and a new one is created. This option is ignored if log output is not configured to be sent to a logfile. |
| | | log_expire | `false`<br>Number<br>Number `file`<br>Number day | Valid Values:<br>`false`: No expiration; all generated segments are stored.<br><`number`> or <`number`> file: Sets the maximum number of log files to store. Specify a number from 1–`1000`.<br><`number`> day: Sets the maximum number of days before log files are deleted. Specify a number from 1—`100`<br>Deafult Value: 10 (store 10 log fragments and purge the rest)<br>Determines whether log files expire. If they do, sets the measurement for determining when they expire, along with the |

| Domain | Section | Setting | Values | Description |
|---|---|---|---|---|
| | | | | maximum number of files (segments) or days before the files are removed. This option is ignored if log output is not configured to be sent to a log file. |
| | | log_time_convert | `local`<br>`utc` | Valid Values:<br><br>`local`: The time of log record generation is expressed as a local time, based on the time zone and any seasonal adjustments. Time zone information of the application's host computer is used.<br>`utc`: The time of log record generation is expressed as Coordinated Universal Time (UTC).<br>Default Value: `local`<br>Specifies the system in which an application calculates the log record time when generating a log file. The time is converted from the time in seconds since the Epoch (00:00:00 UTC, January 1, 1970). |
| | | log_time_format | `time`<br>`locale`<br>`ISO8601` | Valid Values:<br><br>`time`: The time string is formatted according to the HH:MM:SS.sss (hours, minutes, seconds, and milliseconds) format<br>`locale`: The time string is formatted according to the system's locale.<br>`ISO8601`: The date in the time string is formatted according to the ISO 8601 format. Fractional seconds are given in milliseconds.<br>Default Value: `time`<br>Specifies how to represent, in a log file, the time when an application generates log records. A log record's time field in the ISO 8601 format looks like this: `2001-07-24T04:58:10.123`. |
| | **security** | | | |
| | | cert_file | String | Thumbprint value |

| Domain | Section | Setting | Values | Description |
|---|---|---|---|---|
| | | | | of the Public endpoint certificate file, which is used as a client-side certificate for outgoing TLS connection and server-side certificate for incoming TLS connections. For example: `78 44 34 36 7a c2 22 48 bd 5c 76 6b 00 84 5d 66 83 f5 85 d5` |
| | | tls_enabled | Number | If set to 1, connection with TLS transport will be registered. Default: `0`. |
| | | use_srtp | String<br>disabled optional mandatory | Indicates whether to use SRTP |
| | **media** | | | |
| | | ringing_file | String | Valid Values: Empty or String file name<br>Defaul Value: `ringing.mp3`<br>The Ringing sound file name in the current directory or the full local path to the ringing sound file. Specifies the audio file that is played in the defualt audio device (speakers) when the default device ringing tone is enabled with the `ringing_enabled` option. |

For more information about these options, see SIP Endpoint SDK for .NET Developer's Guide.

# Using the Genesys Softphone

This section describes how to use the Genesys Softphone.

## Starting the Genesys Softphone

You can start the Genesys Softphone in one of two ways:

- Double-click the GenesysSoftphone.exe file found in the `<Genesys Softphone Installation Directory>/Genesys Softphone/GenesysSoftphone/` directory
- Execute the following command:

```
C:<Genesys Softphone Installation Directory>Genesys Softphone\GenesysSoftphone\
GenesysSoftphone.exe   C:<Genesys Softphone Installation Directory>Genesys Softphone\
GenesysSoftphone\\ConfigFileName.config
```

To open the Genesys Softphone UI, right-click the Genesys Softphone (file:Spicon.png) icon from the Icon Tray:

file: Softphone_icon.png

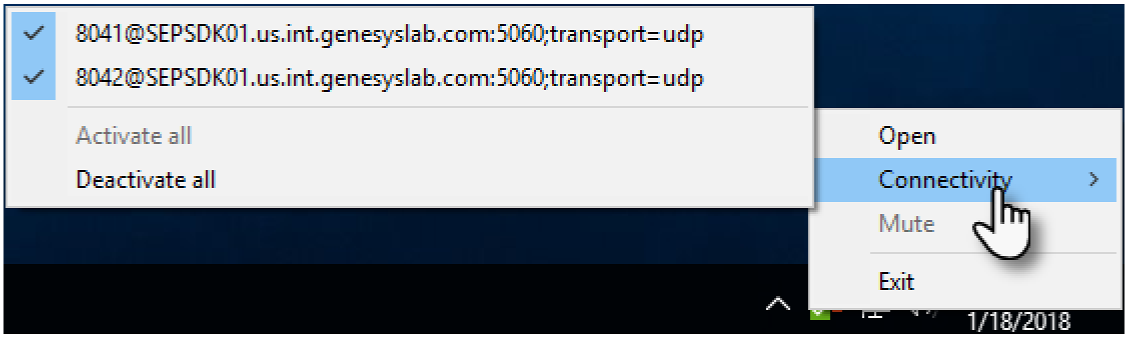
and select **Open**.

## Activating and Registering the User

When the Genesys Softphone first starts, it reads the user's information from the `Softphone.cfg` file, and automatically registers the user.

To verify that the user is registered:

- After starting the Genesys Softphone, right-click on the softphone icon from the Icon Tray and hover over the **Connectivity** menu. You can register or un-register a connection by clicking and toggling the check marks. The notification area shows that the Softphone is active and ready to take calls.




## Selecting the Input and Output Devices

The Genesys Softphone configures the input and output devices during start-up when it reads the list of devices from the Softphone.config file. However, if required, the softphone user can change the brand of device used while the Genesys Softphone is running.

To select an input or output device:

1. In the application, click on the **devices** tab.

   file:Softphone_GUI_devices_tab.png
2. Select the appropriate microphone from the **Input Device** drop-down list.
3. Select the appropriate speaker from the **Output Device** drop-down list.

## Viewing the Softphone Users and Status

Each Genesys Softphone instance can have up to six SIP user accounts configured.

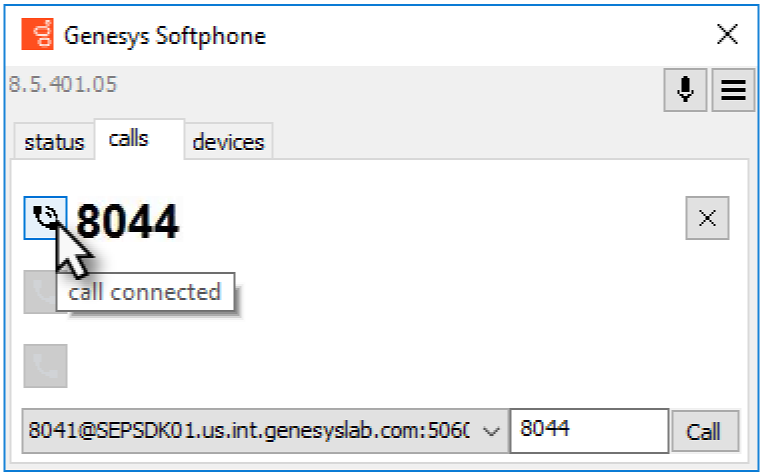
To view the number of users configured and their statuses:

1. Right-click the softphone icon, and click **Open**. The **Genesys Softphone** window displays. Click on the **status** tab.

   file:Softphone_GUI_status_tab.png

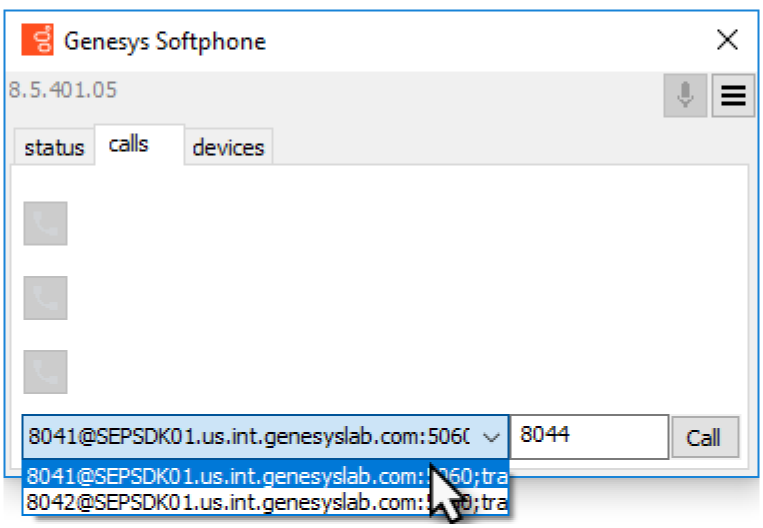
## Making and Receiving Calls

You can make and receive calls from the **calls** tab.



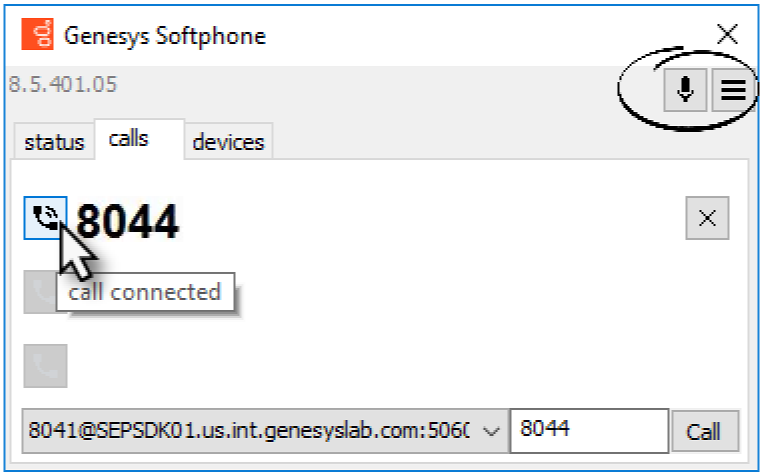

From this tab, you can perform the following operations:

- Answer an incoming call—click on the button of an *alerting* call to answer. If you were on another call, that call will be placed on hold.
- Hold a call—when you switch to another call, the currently active call is placed on hold.
- Retrieve a call—click on the the line button of a call on hold to retrieve that call.
- Hangup a call—click on the hangup button to terminate a call. You can terminate calls that are on hold.
- Dial and make a call—you can make a call by selecting an originating account (connection) from the connections combo box, entering a destination number, and clicking **Call**. Making a new call while another call is active places the existing call on hold.




## Muting the Microphone

The microphone button shows the current mute status, either muted or un-muted. Clicking the microphone button changes the status.




Mute/un-mute functionality works on the application level and not the system level:

- The mute button is only available when there is an active call.
- Muting the microphone in the Softphone is done on the session level. The mute status does not depend on the selected devices nor on device presence and status. A session may be muted even if a microphone is not plugged in.

You may also mute/un-mute the microphone from the tray icon menu. To mute/un-mute the input device:

1. Right-click on the Softphone icon, and click **Mute**.
2. From the same menu, click **Un-mute** un-mute the input device.

> ### Important
> The mute menu item is clickable only when the Genesys Softphone is in an active session.