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# Genesys Softphone Administrator's Guide

Configuration options reference

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Configuration settings by container and domain found in the `Softphone.config` file in the Genesys Softphone Installation Directory.

## Related documentation:

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For an example of the configuration file, see [Configuring Genesys Softphone](#).

The **Softphone.config** file is installed, along with **genesys\_softphone.exe**, by either the **Genesys Installation Wizard** or silently by command line. The contents of the **Softphone.config** file is generated by the choices specified in the wizard or by modifications made to the **genesys\_silent.ini** file.

In the **Softphone.config** file, the following attributes of the **Connector** section are set by **setup.exe**: `protocol`, `port`, and `certificate_search_value`, while `enable_sessionid` and `auto_restart` are not. The default value of these attributes are designed to address most business deployments. However, if you want to adjust their values, follow these steps to make a custom deployment:

1. Install Genesys Softphone on an administrator's machine.
2. Edit the **Softphone.config** file to change the values of the attributes in the **Connector** section.
3. Repackage Genesys Softphone with the custom **Softphone.config** file through an IT-controlled installation.
4. Push the custom package to the agent workstations.

## Basic Container

The first container ("Basic") holds the basic connectivity details that are required to connect to your SIP Server, optionally through a Session Border Controller (SBC). This container has at least one connection (Connectivity) element with the following attributes:

If you are using a configuration that supports Disaster Recovery and Geo-Redundancy, there may be multiple connection elements present with each specifying a separate possible connection.

You must make the following changes and save the updated configuration file before using Genesys Softphone:

- `user="DN"`: Supply a valid DN for the user attribute.
- `server="SERVER:PORT"`: Replace `SERVER` with the host name where your SIP Server or SBC is deployed, and `PORT` with the SIP port of the SIP Server or SBC host. The default SIP port value is 5060. For SRV

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resolution, specify the SRV record without including the port number in the server's URI. Also see SRV Resolution below.

- `protocol="TRANSPORT"`: Set the protocol attribute to reflect the protocol being used to communicate with SIP Server or SBC. Possible values are UDP, TCP, or TLS.

## SRV resolution

When using an SRV record for the **server** parameter, note the following:

- Do not specify the port in the server URI.
- Genesys Softphone does not take into account the **weight** field of an SRV record.
- You cannot combine IPv4 and IPv6 for a single FQDN.
- The maximum number of targets (SRV records) per service is 20.
- You can only specify SRV records in the **server** parameter of the **Connectivity** element. You cannot use SRV records for the mailbox section or the **vq\_report\_collector** setting.

## WebRTC

You will have to make the following changes and save the updated configuration file before using the Genesys Softphone:

- `user="DN"`—Supply a valid DN for the user attribute.
- `server="WEBRTC_GATEWAY_SERVER:WEBRTC_GATEWAY_PORT?sip-proxy-address="SIPPROXY_SERVER:SIPPROXY_PORT"`—Replace WEBRTC\_GATEWAY\_SERVER with the host name where the WebRTC Gateway is deployed, and PORT with the HTTPS port of the WebRTC Gateway. Also, replace SIPPROXY\_SERVER and SIPPROXY\_PORT (optional) with the connectivity parameters of the SIP Proxy that need to be contacted by the WebRTC Gateway to register this DN.
- `protocol="TRANSPORT"`—Set the protocol attribute to reflect the protocol being used to communicate with the WebRTC Gateway: HTTPS.

### 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, DN0@serverHostName0:port0>
		server		The SIP Server or

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Domain	Section	Setting	Default Value	Description
				Proxy location for the first user. Included in the SIP URI—for example, serverHostName0:port0>
		protocol		The transport protocol for the first user. For example, UDP, TCP, or TLS.

## Genesys Container

The second Container ("Genesys") holds a number of configurable settings that are organized into domains and sections. You don't have to change these settings but you can customize them.

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

Domain	Section	Setting
policy	endpoint	audio_qos
		include_os_version_in_user_agent_header
		include_sdk_version_in_user_agent_header
		ip_versions
		public_address
		include_mac_address
		refer_to_proxy
		rtp_inactivity_timeout
		rtp_port_min
		rtp_port_max
		tcp_port_min
		tcp_port_max
		signaling_qos
		sip_port_min
		sip_port_max
		sip_transaction_timeout
		video_max_bitrate
		video_qos
		vq_report_collector
		vq_report_publish
		vq_alarm_threshold
		webrtc_audio_layer
		answer_sdp_priority

Domain	Section	Setting
		sip_port_binding
		defer_device_release
	<b>session</b>	
		agc_mode
		auto_accept_video
		auto_answer
		auto_answer_delay
		callwait_tone_enabled
		callwait_tone_file
		dtmf_feedback
		dtmf_method
		echo_control
		noise_suppression
		dtx_mode
		reject_session_when_headset_na
		sip_code_when_headset_na
		vad_level
		ringback_enabled
		ringback_file
		ringing_enabled
		ringing_timeout
		ringing_file
		ringing_while_call_held
		restart_audio_if_stuck
		reject_session_when_busy

Domain	Section	Setting
		number_sessions_for_busy
		sip_code_when_busy
		rx_agc_mode
	device	
		audio_in_device
		audio_out_device
		capture_device
		headset_name
		include_headset
		use_headset
codecs		
— See SIP Endpoint SDK for .NET 9.0.0NET—Working with Codec Priorities.		
proxies		
	proxy"	
		display_name
		domain
		password
		reg_interval
		reg_match_received_rport
		reg_timeout
		mailbox (sub-section of proxy)
		server
		timeout
		transport



Domain	Section	Setting
		user
		<b>nat (sub-section of proxy)</b>
		ice_enabled
		stun_server
		stun_server_port
		turn_password
		turn_relay_type
		turn_server
		turn_server_port
		turn_user_name
		system
	<b>diagnostics</b>	
		enable_logging
		log_file
		log_filter
		log_level
		log_options_provider
		log_options_endpoint
		logger_type
		log_segment
		log_expire
		log_time_convert
		log_time_format
	<b>security</b>	
		certificate

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Domain	Section	Setting
		tls_enabled
		tls-target-name-check
		use_srtp
	media	
		ringing file

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## policy Domain

### endpoint Section

audio\_qos

Valid Values: Integer

Integer value representing the DSCP bits to set for RTP audio packets. **Note:** QoS is not supported for Windows Vista, Windows 7, or higher.

include\_os\_version\_in\_user\_agent\_header

Valid Values: 0, 1

Default Value: 1

If set to 1, the user agent field includes the OS version the client is currently running on.

include\_sdk\_version\_in\_user\_agent\_header

Valid Values: 0, 1

Default Value: 1

If set to 1, the user agent field includes the SDK version the client is currently running on.

ip\_versions

Valid Values: IPv4, IPv6, IPv4, IPv6, IPv6, IPv4, or empty

Default Value: IPv4, IPv6

- IPv4—the application selects an available local IPv4 address; IPv6 addresses are ignored.
- IPv6—the application selects an available local IPv6 address; IPv4 addresses are ignored.
- IPv4, IPv6 or an empty—the application selects an IPv4 address if one exists. If not, an available IPv6 address is selected.
- IPv6, IPv4—the application selects an IPv6 address if one exists. If not, an available IPv4 address is selected.

**Note:** This parameter has no effect if the **public\_address** option specifies an explicit IP address.

public\_address

Valid Values: See description below

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Default Value: Empty string which is fully equivalent to the `$auto` value

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 SDK uses to automatically obtain the public address.

*Valid Values:*

This setting may have one of the following explicit values:

- An IP address. For example, 192.168.16.123 for IPv4 or FE80::0202:B3FF:FE1E:8329 for IPv6.
- A bare host name or fully qualified domain name (FQDN). For example, epsipwin2 or epsipwin2.us.example.com.

This setting may have one of the following special values:

- `$auto`—The SDK selects the first valid IP address on the first network adapter that is active (status=up) and has the default gateway configured. IP family preference is specified by the `policy.endpoint.ip_versions` setting.
- `$ipv4` or `$ipv6`—Same behavior as the `$auto` setting but the SDK restricts the address to a particular IP family.
- `$host`—The SDK retrieves the standard host name for the local computer using the `gethostname` system function.
- `$fqdn`—The SDK retrieves the fully qualified DNS name of the local computer. The SDK uses the `GetComputerNameEx` function with parameter `ComputerNameDnsFullyQualified`.
- 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 `$fqdn`.  
If the value is an explicit host name, FQDN, or `$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 `$host`, the resolved IP address is used for Contact. The value in SDP is always the IP address.
- `$net:subnet` - The SDK will select the IP address matching the given network ( from any local interface). The *subnet* is the full CIDR name as per RFC 4632. For example, `$net:192.168.0.0/16`.

## include\_mac\_address

Valid Values: 0, 1

Default Value: 0

If set to 1, the MAC address is included in the Contact header of the REGISTER message of the host's network interface in a format compatible with RFC 5626.

## refer\_to\_proxy

Valid Values: 0, 1

Default Value: 0

Specifies the destination of a referred INVITE.

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- 0—Send the INVITE to the URL specified in the Refer-To header of the REFER message.
  - 1—Send the INVITE to your configured SIP Proxy.

### rtp\_inactivity\_timeout

Valid Values: 5-150

Default Value: 150

Suggested Value: 30

Timeout interval in seconds for RTP inactivity.

### rtp\_port\_min

Valid Values: 9000-65535

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

Valid Values: 9000-65535

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 to initialize the endpoint.

### tcp\_port\_min

Valid Values: 0-65535

The integer value representing the minimum value for a TCP client-side port range. Must be within the valid port range of 1 to 65535. If set to 0 (default) or if the configured range is not valid, SIP connections over TCP and TLS use ephemeral ports, assigned by the operating system.

### tcp\_port\_max

Valid Values: 0-65535

The integer value representing the maximum value for a TCP client-side port range. Must be within the valid port range of 1 to 65535. If set to 0 (default) or if the configured range is not valid, SIP connections over TCP and TLS use ephemeral ports, assigned by the operating system.

If the value is non-zero and greater than the *tcp\_port\_min* value, this value specifies the maximum value for a TCP client-side SIP port range that will be used for all outgoing SIP connections over TCP and TLS transport.

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## signaling\_qos

Valid Values: Integer

The integer value representing the DSCP bits to set for SIP packets. **Note:** QoS is not supported for Windows Vista, Windows 7, or higher.

## sip\_port\_min

Valid Values: 1-65535

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

Valid Values: 1-65535

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.

## sip\_transaction\_timeout

Valid Values: 1-32000

Default Value: 4000

SIP transaction timeout value in milliseconds. Valid values are 1 through 32000, with a default value of 4000. The recommended value is 4000.

## video\_max\_bitrate

Valid Values: Integer

Integer value representing the maximum video bitrate.

## video\_qos

Valid Values: Integer

The integer value representing the DSCP bits to set for RTP Video packets. **Note:** QoS is not supported for Windows Vista, Windows 7, or higher.

## vq\_report\_collector

See SIP Endpoint SDK for .NET—Producing RTCP Extended Reports.

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## vq\_report\_publish

See SIP Endpoint SDK for .NET—Producing RTCP Extended Reports.

## vq\_alarm\_threshold

Valid Values: 0 or a number from 1.0 to 5.0

Default Value: 0

Specifies the MOS threshold for generating Voice Quality Alarms. A 0 value disables the alarms. The recommended threshold value is 3.5. Genesys recommends that you avoid using values above 4.2 as an MOS that high might not be obtainable with some codecs, even in perfect network conditions.

## webrtc\_audio\_layer

Valid Values: 0, 1, 2, 1000, 2000, 3000

Default Value: 0

Specifies which audio layer is used for WebRTC.

- 0 — The audio layer is defined by the GCTI\_AUDIO\_LAYER environment variable — Core audio is used if this environment variable is not specified.
- 1 — Wave audio layer is used.
- 2 — Core audio layer is used.
- 1000 — Instructs the audio layer to open the microphone channel when the endpoint starts up, using the audio layer type defined by option 0, and to keep it open until the endpoint is terminated.
- 2000 — Opens the speaker channel for the life of the endpoint, using the audio layer type defined by option 0. Eliminates any delay in opening the audio device when an incoming or outgoing call is connected, for example in environments where audio device startup is slow due to a required restart of the Windows MMCSS service.
- 3000 — Opens the microphone and speaker channels for the life of the endpoint, using the audio layer type defined by option 0.

### Important

Keeping the audio channels permanently open eliminates any delay in connecting audio device to the call works around any issues with device occasionally not starting (or stopping) properly, at the cost of very small performance penalty.

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## answer\_sdp\_priority

Valid Values: config, offer

Default Value: config

- config—the endpoint selects the first codec from the codec configuration listed in both the codec configuration and the SDP offer.
- offer—the endpoint selects the first codec in the SDP offer listed in both the codec configuration and the SDP offer.

## sip\_port\_binding

Valid Values: 0, 1

Default Value: 0

- 0—open the SIP port to listen on any interface.
- 1—the SIP port binds to the interface specified by the public\_address setting and listens only on this IP address.

## defer\_device\_release

Valid Values: Any integer

Default Value: 200

If set to a non-zero value, releasing of audio devices will be deferred for a given time (in milliseconds) after the audio stream has been stopped, to avoid any potential service interruptions when the audio is going to be quickly restarted, and if audio device operations are too slow on the user workstation or have other problems with restart. A zero value disables the deferred device release.

## session Section

### agc\_mode

Valid Values: 0, 1

Default Value: 1

If set to 0, AGC (Automatic Gain Control) is disabled; if set to 1, it is enabled. 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.

**Note:** It is not possible to apply different AGC settings for different channels in multi-channel scenarios.

### auto\_accept\_video

Valid Values: 0, 1



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This setting is only used in auto-answer scenarios when `auto_answer=1`.

If `auto_accept_video` is set to 1, both audio and video streams are accepted, otherwise incoming calls are answered as audio only, even if video is present in the offer.

`auto_accept_video` applies to a 3pcc answer when `make-callrfc3275` is configured to 1 on the originating DN and a video codec is configured in the endpoint. `auto_accept_video` is not applied to a 3pcc answer when `make-call-rfc3275` is configured to 2 on an originating DN, even if `auto_accept_video` is set to 1 and a video codec is configured in the endpoint.

`auto_answer`

Valid Values: 0, 1

If set to 1, all incoming calls should be answered automatically.

`auto_answer_delay`

Valid Values: Number in milliseconds

Default Value: 1500

Time in milliseconds to wait before auto-answering. The recommended and default value is 1500 milliseconds.

`callwait_tone_enabled`

Valid Values: 0, 1

Default Value: 1

Specifies whether the call waiting tone is enabled (1) or disabled (0). This configuration is applied at startup.

`callwait_tone_file`

Valid Values: Empty, or the path to the call waiting sound file. The path may be a file name in the current directory or the full path to the sound file.

Default Value: `callwait.wav`

Specifies the audio file that is played when the call waiting tone is enabled by the `callwait_tone_enabled` option.

Note: WebRTC does not support MP3 playback. The `callwait` file for built-in ringing should be a RIFF (little-endian) WAVE file using one of the following formats:

- `kWavFormatPcm` = 1, PCM, each sample of size `bytes_per_sample`
- `kWavFormatALaw` = 6, 8-bit ITU-T G.711 A-law
- `kWavFormatMuLaw` = 7, 8-bit ITU-T G.711 mu-law

Uncompressed PCM audio must 16-bit mono or stereo, and have a frequency of 8, 16, or 32 kHz.

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## dtmf\_feedback

Valid Values: 0, 1 (default). If set to 1, DTMF feedback (audio tones played locally when sending DTMF signals to the remote side of the conversation) is enabled.

## dtmf\_method

Valid Values: Rfc2833, Info, InbandRtp

Method to send DTMF.

## echo\_control

Valid Values: 0, 1

If set to 1, echo control is enabled.

## noise\_suppression

Valid Values: 0, 1

If set to 1, noise suppression is enabled.

## dtx\_mode

Valid Values: 0, 1

If set to 1, DTX is activated.

## reject\_session\_when\_headset\_na

Valid Values: 0, 1

If set to 1, the SDK should reject the incoming session if a USB headset is not available.

## sip\_code\_when\_headset\_na

Valid Values: SIP Error Code

Default Value: 480

If a valid SIP error code is supplied, the SDK rejects the incoming session with the specified SIP error code if a USB headset is not available.

## vad\_level

Valid Values: 0-3

Sets the degree of bandwidth reduction, from 0 for conventional VAD to 3 for aggressive high.

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## ringback\_enabled

Valid Values: 0, 1, 2, 3, 4, 6

Default Value: 2

Specifies whether the ringback tone is enabled for outgoing calls.

- 0 — The ringback is not played when the INVITE dialog is not yet established. In scenarios where ringback is provided by Media Server, the ringback tone would be still present.
- 1 — The incoming media stream is played if provided by the Media gateway in a reliable provisional response with SDP.
- 2 — A local file is used for the ringback.
- 3 — The ringback is always played using either a local file or media provided by the gateway, if the provisional response is reliable.
- 4 — Same as 1, but the incoming media stream is played even if the provisional response from Media gateway is not reliable.
- 6 — The ringback is always played using either a local file or media provided by the gateway (regardless of whether the provisional response is reliable or not).

## ringback\_file

Valid Values: Empty or the path to the ringback sound file. The path can be a file in the current directory or the full path to the sound file.

Default Value: Empty

Specifies the audio file that is played when the ringing tone is enabled with the `ringing_enabled` option.

WebRTC does not support MP3 playback. The ringtone file for built-in ringback must be a RIFF (little-endian) WAVE file using one of the following formats:

- `kWavFormatPcm` = 1, PCM, each sample of size `bytes_per_sample`
- `kWavFormatALaw` = 6, 8-bit ITU-T G.711 A-law (8 KHz sampling rate)
- `kWavFormatMuLaw` = 7, 8-bit ITU-T G.711 mu-law (8 KHz sampling rate)

Uncompressed PCM audio must be 16-bit mono or stereo with a sampling rate of 8, 16, or 32 KHz.

## ringing\_enabled

Valid Values:

- 0: None, disable ringtone.
- 1: (default) Play ringtone through system default device only. Configure media in `system.media.ringing_file`.
- 2: Play ringtone through communication device (headset) only. Configure media in `policy.session.ringing_file`.

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- 3: Play ringtone through both devices at the same time (the combination of values 1 and 2).
  - 4: Play ringtone through a separate ringer device, specified by `policy.device.ringer_device`.
  - 5: Play ringtone through system default device and lay ringtone through a separate ringer device (the combination of values 1 and 4).
  - 6: Play ringtone through the communication device (headset) once only for the full duration (`policy.session.ringing_timeout` is ignored, and ringing does not stop when the call is answered). Configure media in `policy.session.ringing_file`.
  - 7: Play ringtone once for the full duration through both system default device and communication device (headset) (`policy.session.ringing_timeout` is ignored, and ringing does not stop when call is answered). Configure media in `system.media.ringing_file` and `policy.session.ringing_file`.

Default Value: 1

Specifies whether to enable the ringtone and on which device to play the media file. This option applies to both calls that are auto-answered by Softphone or by other applications such as Workspace Web Edition and by calls that are manually answered by an agent.

## Suppressing the Ringtone

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

- `make-call-alert-info=@genesys>`

or

- `make-call-alert-info=;service=3pcc`

### Important

If at least one application based on SIP Endpoint in the contact center is configured with the `ringing_enabled` option set to a non-zero value, the SIP Server `make-call-alert-info` option should be set to one of the specified values.

## ringing\_timeout

Valid Values: Empty, 0, or a positive number

Default Value: 0

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

Valid Values: Empty or the path to the ringing sound file. The path may be a file name in the current directory or the full path to the sound file.

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Default Value: ringing.wav

Specifies the audio file that is played when the ringing tone is enabled with the `ringing_enabled` option.

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:

- `kWavFormatPcm` = 1, PCM, each sample of size `bytes_per_sample`
- `kWavFormatALaw` = 6, 8-bit ITU-T G.711 A-law
- `kWavFormatMuLaw` = 7, 8-bit ITU-T G.711 mu-law

Uncompressed PCM audio must 16 bit mono or stereo and have a frequency of 8, 16, or 32 KHZ.

`ringing_while_call_held`

Valid Values: 0, 1

Default Value: 1

Specifies whether to play ringtone or not when a call is held and a new call arrives.

- 0 - call wait tone (if configured) is played instead of ringtone.
- 1 - ringtone is played whenever a new call arrives and there are no other active calls; held calls are not considered active in this case.

`restart_audio_if_stuck`

Valid Values: Empty, 0, 1

Default Value: 0

- 0 or Empty—disable auto restart for stuck audio
- 1—enable auto restart for stuck audio

`reject_session_when_busy`

Valid Values: Empty, 0, 1

Default Value: 0

- 0 or Empty—disable rejection of a session when busy
- 1—enable rejection of a session when busy

`number_sessions_for_busy`

Valid Values: Positive integer

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Default Value: 1

Sets the number of sessions before busy. Must be a positive integer.

sip\_code\_when\_busy

Valid Values: Empty, 4xx, 5xx, 6xx

Default value: Empty

SIP error response code to use when busy. Can be set to any valid SIP error response code in the 4xx, 5xx, or 6xx range, for example, 486.

rx\_agc\_mode

Valid Values: 0, 1

Default value: 0

When set to 1, the SDK enables the receiving-side AGC allowing the volume of the received RTP stream to be adjusted automatically. When set to 0 (default), the feature is disabled.

## **device** Section

audio\_in\_device

Valid Values: A regex that matches the ECMAScript standard.

Microphone device name.

audio\_out\_device

Valid Values: A regex that matches the ECMAScript standard.

Speaker device name.

capture\_device

Valid Values: A regex that matches the ECMAScript standard.

Capture device name.

headset\_name

Valid Values: A regex that matches the ECMAScript standard.

The name of the headset model.

include\_headset

Valid Values: A pair of device names or name parts, with microphone and speaker names separated

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by a colon, or a comma-separated list of name pairs. For example: External Mic:Headphones

If the names include delimiter characters such as quotes, colons, or comma, they must be enclosed in single or double quotes.

Specifies the list of audio in / out devices to be considered as a headset for automatic device selection. This option is applicable to the case when **use\_headset** = "1"

ringer\_device

Valid Values: A valid ringer device name: can be either the device proper name or a regular expression. This option is applicable when ringing\_enabled = 4

use\_headset

Valid Values: 0, 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.

exclude\_headset

Valid Values: Any valid regular expression.

Default Value: Empty

The name of a headset model or built-in audio device (example: Realtek). The specified device is excluded from being recognized or automatically selected as a valid headset. Note that components of an excluded device, such as a microphone or speaker, can still be selected manually (or automatically, if Softphone does not find a better device). However, even if a component of an excluded device is selected, Softphone does not recognize the excluded device as an available headset.

### Important

This option is only available on Windows installations.

## codecs Domain

See SIP Endpoint SDK for .NET 9.0.0.NET—Working with Codec Priorities.

## proxies Domain

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Configure a proxy section for each connectivity line. For example, for three connectivity lines, configure sections for proxy0, proxy1, and proxy2.

When the proxy section does not exist in the configuration file for a particular connectivity line, the framework takes the configurations settings from the proxy0 section. You can use this feature in use cases where the proxy sections are the same for all connectivity lines.

## proxy Section

### display\_name

Valid Values: String

Proxy display name.

### domain

Valid Values: Any valid SIP domain

Default Value: Empty

A SIP domain is an application layer configuration defining the management domain of a SIP proxy. The configured value should include hostport and may include uri-parameters as defined by RFC 3261. The scheme, userinfo, and transport URI parameters are included automatically.

If set to an empty string, SIP Endpoint SDK for .NET uses the parameters from the Connectivity section to construct the SIP domain value as it did in previous versions.

### password

Valid Values: String

Proxy password.

### reg\_interval

Valid Values: Integer

Default Value: 0

The period, in seconds, after which the endpoint starts a new registration cycle 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.

### reg\_match\_received\_rport

Valid Values: 0 or 1

Default Value: 0

This setting controls whether or not SIP Endpoint SDK should re-register itself when receiving an IP



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address (in the received parameter of a REGISTER response) that is different from the address supplied in the Contact header and does not match any local network interfaces. A value of 0 (default) disables this feature and a value of 1 enables re-registration.

Starting from 9.0.003, this setting is deprecated and is not recommended for use, unless suggested by Genesys Technical Support to fix specific problems. When the received parameter of a REGISTER response matches a local IP address, changing the IP address and re-registering is now done automatically.

reg\_timeout

Valid Values: Number in seconds

The period, in seconds, after which registration should expire. A new REGISTER request will be 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.

## mailbox Sub-section

### Important

**mailbox** is a sub-section of the **proxy** section.

password

Valid Values: String

Mailbox password.

server

Valid Values: String

Proxy server address and port for this mailbox.

timeout

Valid Values: Number in seconds

Default Value: 1800

Subscription expiration timeout in seconds. If the setting is missing or set to 0, the SDK uses a default timeout of 1800 seconds (30 minutes).

transport

Valid Values: udp, tcp, tls

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Transport protocol to use when communicating with the server.

user

Valid Values: String

Mailbox ID for this mailbox.

## **nat** Sub-section

### Important

**nat** is a sub-section of the **proxy** section.

ice\_enabled

Valid Values: Boolean

Enable or disable ICE.

stun\_server

Valid Values: String

STUN server address. An empty or null value indicates this feature is not used.

stun\_server\_port

Valid Values: Valid port number

Default Value: 3478

STUN server port value.

turn\_password

Valid Values: String

Password for TURN authentication.

### Warning

Starting from 9.0.012.02, this setting is deprecated and is not recommended for use, unless suggested by Genesys Technical Support to fix specific problems. Use the GCTI\_TURN\_PASSWORD environment variable to set the password for TURN authentication.

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## turn\_relay\_type

Valid Values: 0, udp, 1, or tcp

Type of TURN relay.

- 0 or udp for TURN over UDP.
- 1 or tcp for TURN over TCP.

## turn\_server

Valid Values: String

TURN server address. An empty or null value indicates this feature is not used.

## turn\_server\_port

Valid Values: Valid port number

Default Value: 3478

TURN server port value.

## turn\_user\_name

Valid Values: String

User ID for TURN authorization

### Warning

Starting from 9.0.012.02, this setting is deprecated and is not recommended for use, unless suggested by Genesys Technical Support to fix specific problems. Use the GCTI\_TURN\_USERNAME environment variable to set the username for TURN authentication.

## system Domain

### diagnostics Section

## enable\_logging

Valid Values: 0 or 1

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Default Value: 1

Disable or enable logging.

log\_file

Valid Values: String

Log file name, for example, `SipEndpoint.log`.

log\_filter

Valid Values: *Empty*, dtmf

Default Value: *Empty*

Specifies the list of log filters to be applied to hide sensitive data from the endpoint log. Currently the only supported filter is dtmf, which hides all occurrences of DTMF data from the log (by replacing entered digits with 'x').

log\_level

Valid Values: 0-4

Default Value: 3

Log levels: 0 = "Fatal"; 1 = "Error"; 2 = "Warning"; 3 = "Info"; 4 = "Debug"

log\_options\_provider

Valid Values: Valid values for webrtc, warning, state, api, debug, info, error, critical

Example value: `gsip=2, webrtc=(error,critical)`

log\_options\_endpoint

Valid Values: 0-4, same as **log\_level**

Default Value: 2

Log levels: 0 = "Fatal"; 1 = "Error"; 2 = "Warning"; 3 = "Info"; 4 = "Debug"

5 = Logging disabled.

This setting should not be set higher than log\_level setting.

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## logger\_type

Valid Values: file

If set to file the log data will be printed to the file specified by the **log\_file** value.

## log\_segment

Valid Values: false, number, or number in KB,MB, or hr

Default Value: 10 MB

- false: No segmentation is allowed
- or KB: Size in kilobytes
- MB: Size in megabytes
- hr: Number of hours for segment to stay open

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

Valid Values: false, number, number file, number day

Default Value: 10 (store 10 log fragments and purge the rest)

- false: No expiration; all generated segments are stored.
- or file: Sets the maximum number of log files to store. Specify a number from 1—1000.
- day: Sets the maximum number of days before log files are deleted. Specify a number from 1—100.

Determines whether log files expire. If they do, sets the measurement for determining when they expire, along with the 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

Valid Values: local, utc

Default Value: local

- 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.
- utc: The time of log record generation is expressed as Coordinated Universal Time (UTC).

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).

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## log\_time\_format

Valid Values: time, locale, ISO8601

Default Value: time

- **time**: The time string is formatted according to the HH:MM:SS.sss (hours, minutes, seconds, and milliseconds) format.
- **locale**: The time string is formatted according to the system's locale.
- **ISO8601**: The date in the time string is formatted according to the ISO 8601 format. Fractional seconds are given in milliseconds.

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 Section

### Important

SIP Endpoint SDK no longer uses the **tls\_enabled** setting.

## certificate

Valid Values: String

Thumbprint value of the Public endpoint certificate file which is used as a client-side certificate for outgoing TLS connections 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

This option replaces the **cert\_file** option from previous versions. For backwards compatibility, the SDK accepts **certificate** or **cert\_file**.

## tls-target-name-check

Valid Values: no, host

Default Value: host

Specifies if the Common Name in the subject field and/or the Subject Alternate Names of the server's certificate will be compared to the target host name (option value host). If they are not identical, the connection fails. If the option is set to no, a comparison is not made, and the connection is allowed.

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## use\_srtp

Valid Values: optional, allowed, disabled, off, elective, both, enabled, force, mandatory

Indicates whether to use SRTP:

- optional or allowed—do not send secure offers, but accept them
- disabled or off—do not send secure offers and reject incoming secure offers
- elective or both—send both secure and non-secure offers and accept either
- enabled—send secure offers, accept both secure and non-secure offers
- force or mandatory—send secure offers, reject incoming non-secure offers

Adding either `',UNENCRYPTED_SRTCP'` (long form) or `',UEC'` (short form) to any value (for example, `"enabled,UEC"`), would result in the **UNENCRYPTED\_SRTCP** parameter being added to that offer. When this parameter is negotiated, RTCP packets are not encrypted, but are still authenticated.

## media Section

### ringing\_file

Valid Values: Empty, String file name

Default Value: ringing.mp3

The Ringing sound file name in the current directory or the full local path to the ringing sound file.