



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.

SIP Endpoint SDK Developer's Guide

Default `SipEndpoint.config` Settings

12/14/2025

Contents

- 1 Default SipEndpoint.config Settings
 - 1.1 Using the Default Configuration File
 - 1.2 **policy** Domain
 - 1.3 **codecs** Domain
 - 1.4 **proxies** Domain
 - 1.5 system Domain
 - 1.6 Additional Configuration Options

Default SipEndpoint.config Settings

Using the Default Configuration File

You can find the default configuration file in the following location:

```
<installation folder>/Configuration/SipEndpoint.config
```

This file contains XML configuration details that affect how your SIP Endpoint SDK application behaves. The initial settings are the same as those specified for use with the [QuickStart application](#) that is included with your SIP Endpoint SDK release.

Configuration settings are separated into two containers: the [Basic Container](#) holds the connectivity details that are required to connect to your SIP Server, while the [Genesys Container](#) holds a variety of configuration settings.

Basic Container

The first Container ("Basic") holds the basic connectivity details that are required to connect to your SIP Server. This container has at least one connection (Connectivity) element with the following attributes:

```
<Connectivity user="DN" server="SERVER:PORT" protocol="TRANSPORT"/>
```

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. Refer to the [configuration settings](#) of that feature for details. You will have to make the following changes and save the updated configuration file before using the SIP Endpoint SDK:

- `user="DN"` — Supply a valid DN for the user attribute.
- `server="SERVER:PORT"` — Replace SERVER with the host name where your SIP Server is deployed, and PORT with the SIP port of the SIP Server host. (The default SIP port value is 5060.)
- `protocol="TRANSPORT"` — Set the protocol attribute to reflect the protocol being used to communicate with SIP Server. Possible values are UDP, TCP, or TLS.

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 to take full control over your SIP Endpoint SDK applications.

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
		refer_to_proxy
		rtp_inactivity_timeout
		rtp_port_min
		rtp_port_max
		signaling_qos
		sip_port_min
		sip_port_max
		sip_transaction_timeout
		video_max_bitrate
		video_qos
		vq_report_collector
		vq_report_publish
		webrtc_audio_layer
		answer_sdp_priority
		sip_port_binding
	session	
		agc_mode
		auto_accept_video

Domain	Section	Setting
		auto_answer
		auto_answer_delay
		dtmf_method
		echo_control
		noise_suppression
		dtx_mode
		reject_session_when_headset_na
		sip_code_when_headset_na
		vad_level
		ringing_enabled
		ringing_timeout
		ringing_file
		restart_audio_if_stuck
		reject_session_when_busy
		number_sessions_for_busy
		sip_code_when_busy
	device	
		audio_in_device
		For more information, see Audio Device Settings
		audio_out_device
		capture_device
		headset_name
		use_headset
codecs		

Domain	Section	Setting
— See Working with Codec Priorities		
proxies		
	proxy<n>	display_name
		domain
		password
		reg_interval
		reg_match_received_rport
		reg_timeout
		mailbox (sub-section of proxy<n>)
		password
		server
		timeout
		transport
		user
		nat (sub-section of proxy<n>)
		ice_enabled
		stun_server
		stun_server_port
		turn_password
		turn_relay_type
		turn_server
		turn_server_port
		turn_user_name

Domain	Section	Setting
system	diagnostics	
		enable_logging
		log_file
		log_level
		log_options_provider
		log_options_endpoint
		logger_type
		log_segment
		log_expire
		log_time_convert
		log_time_format
	security	
		cert_file
		tls_enabled
		use_srtp
	media	
		ringing_file

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

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.

refer_to_proxy

Valid Values: 0, 1

Default Value: 0

Specifies the destination of a referred INVITE.

- 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.

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 [Producing RTCP Extended Reports](#).

vq_report_publish

See [Producing RTCP Extended Reports](#).

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_AUDIOIO_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.

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.

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

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.

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.

ringing_enabled

Valid Values: 0, 1, 2, 3

Default Value: 1

Specifies whether to enable the ringing tone.

- 0—event Ringing disabled
- 1—event Ringing enabled
- 2—play ringtone internally (event Ringing disabled)
- 3—play ringtone internally and enable event Ringing.

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.

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.

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

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.

device Section

audio_in_device

Valid Values: String

Microphone device name.

For more information, see [Audio Device Settings](#)

audio_out_device

Valid Values: String

Speaker device name.

capture_device

Valid Values: String

Capture device name.

headset_name

Valid Values: String

The name of the headset model.

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.

codecs Domain

See [Working with Codec Priorities](#)

proxies Domain

proxy<n> 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 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

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<n>** 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

Transport protocol to use when communicating with the server.

user

Valid Values: String

User ID for this mailbox.

nat Sub-section

Important

nat is a sub-section of the **proxy<n>** 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.

`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

system Domain

diagnostics Section

enable_logging

Valid Values: 0 or 1

Disable or enable logging.

log_file

Valid Values: String

Log file name, for example, SipEndpoint.log.

log_level

Valid Values: 0-4

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.

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
- <number> or <number> KB: Size in kilobytes
- <number> MB: Size in megabytes
- <number> 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.
- <number> or <number> file: Sets the maximum number of log files to store. Specify a number from 1–1000.
- <number> 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).

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

`cert_file`

Valid Values: String

Thumbprint value 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`

Valid Values: 0, 1

Default Value: 0

If set to 1, connection with TLS transport will be registered.

`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

media Section

ringing_file

Valid Values: Empty, String file name

Default Value: rining.mp3

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

Additional Configuration Options

The default configuration file may not contain all settings that may be used with the SIP Endpoint SDK; additional settings can be added to change certain behaviors. Check [Configuring SIP Endpoint SDK for .NET](#) for a discussion of these additional settings.