SIP Endpoint SDK Developer's Guide

Default SipEndpoint.config Settings

5/4/2025

Contents

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Warning
This documentation is outdated and is included for historical purposes only.

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 inital 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:

Section	Setting	Values	Description
codecs — See Working with Codec Priorities			
device			
	audio_in_device	String	Microphone device name
	audio_out_device	String	Speaker device name
	headset_name	String	The name of the headset model
	manual_audio_devices_configure	Number	Valid values: 0 or 1. The setting is active if `use_headset=0`. Enables configuration of the preferred input and output devices that are set in `audio_in_device` and `audio_out_device`.
	use_headset	Number	Valid values: 0 or 1. If set to 1, the SDK uses a headset as the preferred audio input and output device.
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		Valid values: 0 – 4. Log levels: 0 = "Fatal"; 1 = "Error"; 2 = "Warning"; 3 = "Info"; 4 = "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.

Section	Setting	Values	Description
endpoint			
	audio_qos	Number	The 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	Number	If set to 1, the user agent field includes the OS version the client is currently running on. Default: 0.
	ip_versions	IPv4 IPv6 IPv4,IPv6 IPv6,IPv4 empty	A value of `IPv4` means that the application selects an available local IPv4 address; IPv6 addresses are ignored. A value of `IPv6` means that the application selects an available local IPv6 address; IPv4 addresses are ignored. 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. A value of `IPv6,IPv4` means that the application selects an IPv6 address if one exists. If not, an available IPv4 address is selected. Default: `IPv4,IPv6`. 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.
	rtp_inactivity_timeout	Number	Timeout interval for RTP inactivity. Valid values are integers from `0` to `150`. A value of `0` or values greater than `150` mean that this feature is not activated. A value in the range of `1` to `150` indicates the inactivity timeout interval in seconds. Default: `0`.
	rtp_port_min	Number	The integer value representing the minimum value for an

Section	Setting	Values	Description
			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 to initialize the endpoint.
	signaling_qos	Number	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	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

Section	Setting	Values	Description
			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.
	video_qos	Number	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
mailbox			
	password	String	Mailbox password
	server	String	Proxy server address and port for this mailbox
	timeout	Number	Registration timeout interval
	transport	udp tcp tls	Transport protocol to use when communicating with server

Section	Setting	Values	Description
	user	String	User ID for this mailbox
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
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 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`.
	reg_timeout	Number	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

Section	Setting	Values	Description
			is disabled, putting the endpoint in standalone mode.
security			
	cert_file	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	Number	If set to 1, connection with TLS transport will be registered. Default: `0`.
	use_srtp	String disabled optional mandatory	Indicates whether to use SRTP
session			
	agc_mode	0 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. NOTE: It is not possible to apply different AGC settings for different channels in multi-channel scenarios.
	auto_accept_video	Number	If set to 1, video calling is enabled and if set to 0, video calling is disabled
	auto_answer	Number	If set to 1, all incoming calls should be answered automatically.

Section	Setting	Values	Description
	dtmf_method	Rfc2833 Info InbandRtp	Method to send DTMF
	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 SDK should reject the incoming session if a USB headset is not available.
	sip_code_when_headset_na	Number	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	Number	Sets the degree of bandwidth reduction. Valid values: 0 – 3 — from 0 (conventional VAD) to 3 (aggressive high).

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.