SIP Endpoint SDK Overview

Overview

12/17/2025

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Overview

The SIP Endpoint SDK enables you to build a SIP endpoint that can seamlessly connect agent desktop applications with the Genesys SIP Server in order to handle audio and video calls. This overview will help you understand its design goals and its architecture.

Design Goals

Many SIP softphones that are currently available require an agent to interact with their own separate user interface in addition to that of the agent desktop application. For instance, the agent might have to use the SIP phone interface to answer a call, while other actions — such as holding or releasing the call — would have to be done via agent desktop interface.

In contrast to that, the SIP Endpoint SDK is designed to be integrated into an agent desktop so the agent can use a single user interface to control calls. Genesys recommends that this be done in a way that leaves actual control in the hands of a T-Lib–based agent desktop application, which has a fuller feature set and is also fully supported by Genesys.

The SIP Endpoint SDK is also designed to integrate with the Genesys SIP Server. It supports the SIP, SDP, and RTP/RTCP protocols.

In addition to these principal design goals, the SIP Endpoint SDK supports the following features:

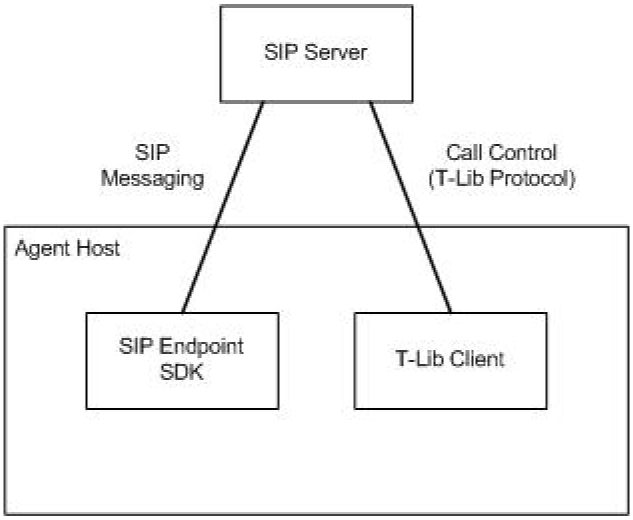
- Traditional call control functionality such as:
    - Establishing inbound and outbound calls
    - Hold and retrieve
    - Transfers and conference calls. SIP Endpoint SDK supports all 1pcc and 3pcc transfer scenarios.
- Multi-line call handling

Note: Multi-line call handling is available up to the limitations on the number of simultaneous connections supported by the licensing available to Genesys at this time.

Architecture

The SIP Endpoint SDK is a software component that resides on the agent's computer. It uses the SIP protocol to communicate with the Genesys SIP Server and supports both thin (or 3-tier) and thick (or rich) T-Lib clients.

In SIP messaging terms, the SIP Endpoint is a user agent. The contact center agent's computer should also host a Genesys T-Lib client that works with SIP Server and provides Genesys agent-related functionality, as shown below.



As you can see, the SIP Endpoint SDK handles the SIP messaging, and the T-Lib Client uses the T-Lib protocol to handle call control.

SDK Components

The components of the SIP Endpoint SDK are described on the following page:

- SIP Endpoint SDK for NET Components

Supported Codecs

SIP Endpoint SDK 9.0.0 supports the following codecs:

- G722/16000 (G.722)
- G.729 (including Annexes A and B)
- H.264 video (Baseline profile only. If any other profile is configured, the default Baseline profile will be used and SIP Endpoint SDK will print an error message in the log file.)
- ILBC/8000 (iLBC — internet Low Bitrate Codec)
- ISAC/16000 (iSAC/16kHz — internet Speech Audio Codec)
- ISAC/32000 (iSAC/32kHz)
- OPUS/48000/2
- PCMA/8000 (G.711/A-law)
- PCMU/8000 (G.711/mu-law)
- VP8 video
- VP9 video

> **Important**
>
> Forward error correction is supported using ULPFEC (RFC 5109). SIP Endpoint SDK automatically adds this support as long as `ulpfec` is included in the codec list in the configuration file.

Supported RFCs

SIP Endpoint SDK 9.0.0 partially or fully supports the following RFCs:

Section	Name	Description
Media		
	RFC 1889	RTP: A Transport Protocol for Real-Time Applications
	RFC 2327	SDP: Session Description Protocol
	RFC 2833	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
	RFC 3264	An Offer/Answer Model with the Session Description Protocol (SDP)
	RFC 3550	RTP: A Transport Protocol for Real-Time Applications [replaces RFC 1889]
	RFC 3951	Internet Low Bit Rate Codec (iLBC)
	RFC 3952	Real-time Transport Protocol (RTP) Payload Format for internet Low Bit Rate Codec (iLBC) Speech
	RFC 5109	RTP Payload Format for Generic Forward Error Correction
	RFC 5245	Interactive Connectivity Establishment (ICE)
Network		
	RFC 1035	Domain names—implementation and specification
	RFC 2327	SDP: Session Description Protocol
SIP		
	draft-ietf-sipping-cc-transfer	Session Initiation Protocol Call Control—Transfer draft-ietf-sipping-cc-transfer-12
	RFC 2617	HTTP Authentication: Basic and Digest Access Authentication (for SIP)

Section	Name	Description
	RFC 2976	The SIP INFO Method
	RFC 3261	SIP: Session Initiation Protocol
	RFC 3265	Session Initiation Protocol (SIP): Specific Event Notification
	RFC 3420	Internet Media Type message/ sipfrag
	RFC 3515	The Session Initiation Protocol (SIP) Refer Method
	RFC 3891	The Session Initiation Protocol (SIP) "Replaces" Header
	RFC 3892	The Session Initiation Protocol (SIP) Referred-By Mechanism
	RFC 5168	XML Schema for Media Control

Working with the SIP Endpoint SDK for .NET

The SIP Endpoint SDK for .NET distribution includes the following files, which you can use "as is" in your custom applications:

- Genesyslab.Sip.Endpoint.dll
- Genesyslab.Sip.Endpoint.Provider.Genesys.dll

These files are located in the `\Bin` directory at the root level of the SIP Endpoint SDK directory.

Learning More

To continue learning about the SIP Endpoint SDK, we recommend you read the pages describing SIP-Based Third-Party Call Control to understand messaging patterns. After that, you should be ready to use the Deployment Guide to install the SDK on your system.

Once you have installed the SDK, you may want to check the following pages from the Developer's Guide:

- SIP Endpoint SDK Configuration for .NET
- Disaster Recovery and Geo-Redundancy