SIP Endpoint SDK Deployment Guide

SIP Endpoint SDK 8.1.2

2/11/2022

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Welcome

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 **This documentation is outdated and is included for historical purposes only.**

Installation

This deployment guide can be used to install SIP Endpoint SDK on your system and verify the installation. It contains chapters that describe how to do this on .NET and Apple's OS X. These chapters each include the following information:

- Deployment Information for .NET and OS X — Details related to the SIP Endpoint SDK installation, including prerequisites and links to related information. Genesys recommends reading this page before beginning your installation, to ensure that your system meets the minimum requirements for the SIP Endpoint SDK.
- Installation procedures for .NET and OS X
- Verification procedures for .NET and OS X — This includes tasks for walking through the installation process and verifying that components were installed correctly.

Next Steps

After you have successfully installed the SIP Endpoint SDK, you might want to do the following:

- Download the latest version of the Release Note (using links on the SIP Endpoint SDK Product Page) to see the most recent news and updates about this product.
- Review the pages on using the included .NET QuickStart Application and the OS X sample application.
- Find out more about how to configure SIP Endpoint SDK for .NET and OS X.
- Read the SIP Endpoint SDK API Reference for detailed information about the SIP Endpoint SDK.

New In This Release

Warning

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Check out the new features that have been added in the latest releases of SIP Endpoint SDK.

.NET

New in Release 8.1.2 for .NET

Release 8.1.200.25

- SIP Endpoint SDK 8.1.2 for .NET Release 8.1.200.25 allows you to select an audio layer for WebRTC using the Windows environment variable `GCTI_AUDIO_LAYER`.

Release 8.1.200.23

- SIP Endpoint SDK 8.1.2 for .NET Release 8.1.200.23 supports Genesys SIP Proxy in standalone mode.

Release 8.1.200.14

Note: SIP Endpoint SDK 8.1.2 for .NET Release 8.1.200.14 supports the following features. Additionally, support includes all new features supported by SIP Endpoint SDK 8.1.2 for Apple OS, with the exception of OS X 10.8 Mountain Lion.

- IPv6 support
- Forced DNS lookup for operating with SIP Cluster
- Support for long device names for Windows Vista SP1 or later
- Genesys SIP libraries and Google's WebRTC media stack are now used instead of the CounterPath SDK
- Support for basic first-party call control features:
    - Dial
    - Answer
    - Hold
    - Retrieve
    - Release

- VoIP monitoring of packet jitter and latency using RTP statistics
- Support for In-Dialog `SIP:INFO` message exchange using customer-provided information for the `Content-Type` and `Content headers`
- Access to raw video frames
- Support for the following Plantronics headset models in the standard headset mode:
    - Savi W740 and W745
    - Savi W440
    - Blackwire C310/C320
- Windows 8 compatibility
- Support for Picture-in-Picture (PiP) display
- Support for Microsoft's .NET WPF (Windows Presentation Foundation)
- Support for the creation of RTCP Extended Reports (RFC 3611) and the ability to publish them according to RFC 6035 at the end of each call. **Note:** The 8.1.2 implementation of this feature includes only MOS values (along with PacketLoss and Delay statistics, which are used for MOS estimation).

New in Release 8.1.1 for .NET

Release 8.1.100.04

- SIP Endpoint SDK for .NET now supports the collection of RTP audio and video statistics, giving users real-time access to these statistics during a call.

Release 8.1.100.03

- SIP Endpoint SDK for .NET now supports the G.729b voice codec with the `annexb=no` extension.

Release 8.1.100.02

- SIP Endpoint SDK for .NET now supports video using a new video control interface that alllows for the control of video source, frame events, window handles, and connectivity mode.
- When using the SIP Endpoint SDK for .NET, you can now encrypt SIP messaging and the media channel using TLS and SRTP, respectively.
- SIP Endpoint SDK for .NET now supports the RFC2617-style digest authentication that is currently used by SIP Server.

Apple OS

New in Release 8.1.2 for Apple OS

- SIP Endpoint SDK 8.1.2 for Apple OS supports OS X Mountain Lion (10.8)
- This release of SIP Endpoint SDK for Apple OS supports the following codecs:

    - G.711 (PCMA, PCMU)
    - G.722
    - iLBC (Internet Low Bitrate Codec)
    - iSAC (Internet Speech Audio Codec)
    - VP8 video

- SIP Endpoint SDK 8.1.2 for Apple OS supports the following Plantronics headset models in the standard headset mode:
    - Savi W440
    - Savi W7xx
    - Blackwire C320

SIP Endpoint SDK 8.1.2 for Apple OS also supports the following features:

- TLS 1.2 protocol (RFC 6176)
- AGC (Automatic Gain Control)
- MWI (Message Waiting Indicator)
- You can now specify these behaviors when a SIP Endpoint user does not have a working USB headset:
    - Whether SIP Endpoint should automatically reject an incoming call
    - The SIP error code to be sent to the inviting party
- `INVITE` messages now have an additional header that contains user data. This data can be obtained by using the following new method of `GSSessionControlService`:

```
- (GSResult) dialFrom:(id<GSConnection>)connection
to:(NSString*)destination withData:(NSString *)data;
```

- Hangup on RTP inactivity timeout
- Configuration of:
    - RTP port ranges
    - SIP port ranges
- Continuous Registration
- DTMF tones can now be sent using SIP INFO
- VoIP monitoring of packet jitter and latency using RTP statistics
- NAT (Network Address Translation) traversal methods:
    - ICE (Interactive Connectivity Establishment)
    - STUN (Session Traversal Utilities for NAT)
    - TURN (Traversal Using Relay NAT)
- Connection to Genesys SIP Cluster

In addition, this release of SIP Endpoint SDK for Apple OS uses a SIP stack that has been developed by Genesys.

New in Release 8.1.1 for Apple OS

- SIP Endpoint SDK 8.1.1 for Apple OS supports OS X Lion (10.7)
- This release of SIP Endpoint SDK for Apple OS supports the following voice codecs:
    - G.711 (PCMA, PCMU)
    - G.722
    - iLBC (Internet Low Bitrate Codec)
- When devices and services use multiple voice codecs, the SIP Endpoint SDK for Apple OS supports the negotiation of the voice codec that will be used between them.
- The SIP Endpoint SDK for Apple OS supports Quality of Service (QoS), which helps guarantee that packet traffic for a voice or other media connection will not be delayed or dropped due to interference.
- The SIP Endpoint SDK for Apple OS supports additional security signaling and media encryption via SRTP (Secure Real-Time Transport Protocol).
- The SIP Endpoint SDK for Apple OS supports both first party call control (1PCC) and third party call control (3PCC).

Deploying SIP Endpoint SDK for .NET

Warning

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

This chapter shows how you can deploy and verify the SIP Endpoint SDK for .NET.

SIP Endpoint for .NET Deployment Information

Warning

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Introduction

For the 8.x release, the SIP Endpoint SDK allows you to develop applications by using .NET technology. To assist you with development, the SIP Endpoint SDK is packaged with a SIP Endpoint SDK API Reference document (`SipEndpointNet.chm`) that allows you to find reference information, coding recommendations, and code snippets in a single location. For your convenience, the SIP Endpoint SDK also includes a Visual Studio Starter Kit that contains a project template and code snippets. This Starter Kit can help you get up and running during early application development. Finally, every Genesys product also includes a Release Note that provides any late-breaking product information that could not be included in the manual. This product information can often be important. To view it, open the `read_me.html` file in the application home directory, where you will find a link to the latest Release Note for this product. These development aids can be located on the Developer Documentation Library CD, as well as from the Genesys Developer website that is located at http://www.genesyslab.com/developer.

What You Should Know

This guide is written for software developers and application architects that have an understanding of the Genesys platform and the basics of SIP telephony before using this SDK. Before working with the SIP Endpoint SDK, you should know how to use the logging functionality of the Platform SDK. In addition, the following document can be useful in understanding the Genesys SIP server environment:

- Framework 8.0 SIP Server Deployment Guide

Environment Prerequisites

Supported Operating Systems

- Windows 8 32 bit
- Windows 2008 Server (32 and 64 bit)
- Windows 7 64 bit
- Windows XP 32 bit

Other Prerequisites

To work with Release 8.1.2 of the SIP Endpoint SDK, 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:

- Genesys SIP Server 7.6.x, 8.0.2, or higher
- Genesys Voice Platform (GVP) 8.0 or higher is required for video conference support, which makes use of the GVP Resource Manager and Media Control Platform
- Microsoft .NET Framework version 3.5 Service Pack 1 or higher
- Microsoft Visual Studio® .NET 2008 or higher
- Microsoft Win32 ® API
- Microsoft Windows SDK for Windows 7 and .NET Framework 4 (http://www.microsoft.com/en-us/download/details.aspx?id=8279)
- DirectX Software Development Kit (http://www.microsoft.com/en-us/download/details.aspx?id=6812)
- Microsoft.VC80.CRT:4053 "Microsoft Visual C++ 2005 Service Pack 1 Redistributable Package ATL Security Update" (http://www.microsoft.com/en-us/download/details.aspx?id=14431)
- Microsoft.VC90.CRT:4148 "Microsoft Visual C++ 2008 Service Pack 1 Redistributable Package ATL Security Update" (http://www.microsoft.com/en-us/download/details.aspx?id=11895)

Important Note: The computers that will run your application must have the Microsoft Visual C++ 2008 SP1 Redistributable Package (x86). To download this package, do one of the following things:

- Go to http://www.microsoft.com/downloads and in the `Search Download Center` field, enter `Visual C++`. The link for the redistributable package will be listed. Make sure you download the SP1 version that is dated 9/16/2008, and not the pre-SP1 version dated 11/29/2007.
- Go directly to the following link, which is valid as of September 1, 2010: http://www.microsoft.com/downloads/details.aspx?familyid=A5C84275-3B97-4AB7-A40D-3802B2AF5FC2&displaylang=en

Related Resources

- SIP Endpoint SDK Developer's Guide

Installing SIP Endpoint SDK for .NET

Warning
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Prerequisites

- Check the list of environment prerequisites, and confirm that your system meets these standards prior to installing SIP Endpoint SDK.

Procedure

Start of procedure

1. Run the installation file named `setup.exe` located in the `\SIPEndpointSDK\DotNet\windows\` directory on your product CD. The `Genesys Installation Wizard` is displayed to guide you through the installation and setup process.
2. Click `Next` at the `Welcome` dialog to display the `Genesys License Agreement` dialog.
3. Check the `I accept Genesys License Agreement` box to accept the conditions of the agreement.
4. Click `Next` at the `Genesys License Agreement` dialog. The `Choose Destination Location` dialog is displayed, showing the default destination, `C:\Program Files\GCTI\SIP Endpoint SDK`.
5. Click `Next` if you want to accept the default destination folder that is specified. If you prefer to install the SIP Endpoint SDK in a different location than the default directory, complete the following steps:
    1. Click `Browse` to open the `Choose Folder` dialog.
    2. Navigate to and select a directory path.
    3. Click `OK` to return to the `Choose Destination Location` dialog.
    4. Click `Next` to accept the destination folder that you have selected.
6. At the `Ready to Install` dialog, click `Install`. The Wizard installs the SIP Endpoint SDK, and all associated files, in the directory you selected. When the installation is finished, the `Installation Complete` dialog appears.
7. Click `Finish`.

End of procedure Next Steps

- To review the installation and confirm the location of your SIP Endpoint SDK files, continue by verifying the installed components.

Verifying Installed SIP Endpoint SDK for .NET Components

Warning

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

Prerequisites

- You must first complete the procedure that is found at Installing SIP Endpoint SDK for .NET.

Procedure

1. Expand the archive containing the SIP Endpoint SDK installation.
2. Examine each folder (including the root installation folder) to confirm its contents. The SIP Endpoint SDK Folder Contents table below gives a description of the expected result.

SIP Endpoint SDK Folder Contents

Folder	Contents
\Doc	This directory contains the SIP Endpoint SDK API Reference (`SipEndpointNet.chm`), which has detailed information about the structure and usage of the SIP Endpoint SDK.
\QuickStart	Visual Studio source files for the SIP Endpoint SDK QuickStart application.
\QuickStartExe	A compiled and ready-to-run version of the SIP Endpoint SDK QuickStart application.

Next Steps

- None

SIP Proxy Support

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

SIP Endpoint SDK can now be used with Genesys SIP Proxy, which provides high availability without requiring a virtual IP address.

Configuration

Domain Names

SRV records are not currently supported by SIP Endpoint SDK, which supports only A (or AAAA) records. Because of this, Genesys SIP Proxies should be configured with a single fully qualified domain name (FQDN) that resolves via DNS into multiple IP addresses via A (or AAAA) records in DNS server.

For information on how to configure SIP Proxy, consult the SIP Proxy Deployment Guide.

SIP Transaction Timeout

Name	Description	Valid Values	Default	Recommended
sip_transaction_timeout	SIP transaction timeout value in milliseconds.	1 through 32000	32000 (32 seconds)	**4000 (4 seconds)**

Because the default SIP transaction timeout value is too long for use by customer-facing applications, **Genesys strongly recommends that you set the `sip_transaction_timeout` value to `4000`, as shown here**:

```
<domain name="policy">
  <section name="endpoint">
      ...
    <setting name="sip_transaction_timeout" value="4000"/>
      ...
  </section>
```

Limitations

- Genesys SIP Proxy does not support scenarios involving switchovers in mid-transaction. Because of this, call answer and `CANCEL` may not work properly in such situations. In particular, the incoming call cannot be answered under these circumstances and must be released. Also, the outgoing call may be stuck on SIP Server for an unpredictable length of time. (Please note that this is a limitation of Genesys SIP Proxy and not of SIP Endpoint SDK.)
- You must configure the `reg_interval` parameter (in the SIP Endpoint configuration file) to a positive value (for example, `30`) if you want your SIP Endpoint to resend REGISTER and SUBSCRIBE messages to a new SIP Proxy when the current SIP Proxy is down. SIP Endpoint does not resend the REGISTER or SUBSCRIBE if `reg_interval` is configured to `0`.
- The SIP Endpoint SDK always retries INVITE once, regardless of the number of proxies configured.

Technical Background

The following background information describes certain features of the SIP Endpoint SDK internals that might be helpful in planning your application.

> **Warning**
> This information is subject to change without notice and is not supported by Genesys.

High Availability

Because the SIP Endpoint SDK already supports DNS queries for each transaction, the load balancing aspect of High Availability is already taken care of, although it may require configuring an IP address rotation in DNS server. The missing parts for High Availability support are:

- **Temporarily blacklisting the IP address on failure (no response), so the next SIP message is sent to a different proxy:** The proposed method for this assigns a specific penalty to each IP address in the list from the DNS, based on how recently that IP address has failed. When a selection must be made, the one with the lowest penalty value is chosen. If there are no penalty-free IP addresses, the algorithm chooses the least-recently blacklisted address, which may now be available. Because of this algorithm, there is not much sense in making the blacklist interval configurable, and it is currently hard-coded to 10 minutes. (After that, the address is fully cleared and can be tried again).
- **You must set the `sip_transaction_timeout` parameter to a value less than 32000 milliseconds**, as described in the Configuration section above, with the recommended setting being `4000`.
- **Automatic re-transmission of failed (timed out) SIP messages to a new proxy:** Given the current GSIPLIB architecture, this re-transmission must be done separately for each message type. Thus, it must be tested for all possible use cases, notably:
  - REGISTER and SUBSCRIBE renewal—by design, switching to a new SIP proxy obeys the configured `reg_interval` parameter, so if re-registration is disabled by a value of `0`, the endpoint does not resend the REGISTER or SUBSCRIBE message.

- Initial INVITE to be retried once to a different SIP proxy (and reported as failed in case of a double failure).
  Note: To give application code full visibility to the SIP call ID , an `in-progress` state is reported twice for the same session ID (with the state reported as `disconnected` in between them).
- Mid-call INVITE for Hold and Retrieve operations to be retried once, transparent to application code.
- Retrying a 200 OK response to an initial INVITE (answering the call) and call-terminating BYE and CANCEL requests work differently from other requests. These retry operations work only when the `sip_transaction_timeout` parameter is set to a value lower than 32000 milliseconds (as described in the Configuration section above), with the recommended setting being `4000`. These requests are retried continuously for 32 seconds total (cycling throught the list of configured proxies), after which the call is abandoned.

ICMP Messages

It is not currently possible to intercept ICMP messages using GSIPLIB, because exceptions are processed on the transport level, but the reaction must be implemented on the transaction level and there is no easy way to pass control between those two levels. Because of this, these failures may be detected in the current release by timeout only. This is not much of a limitation, however, as ICMP messages are generated only when:

- The server is on Windows
- No firewall is blocking them
- The host is alive

Therefore, these messages are unreliable and their use would add very little to the timeout method.

Deploying SIP Endpoint SDK for Apple OS

Warning

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This chapter shows how you can deploy and verify the SIP Endpoint SDK for Apple OS.

SIP Endpoint for Apple OS Deployment Information

Warning

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Introduction

With Release 8.1.1, the SIP Endpoint SDK allows you to develop applications for OS X Lion (10.7). To assist you with development, the SIP Endpoint SDK is packaged with an API Reference, which is located in the `Doc/html` folder. For your convenience, the SIP Endpoint SDK also includes a Sample application that can help you get up and running during early application development. Finally, every Genesys product also includes a Release Note that provides any late-breaking product information that could not be included in the manual. This product information can often be important. To view it, open the `read_me.html` file in the application home directory, which contains a link to the latest Release Note for this product.

What You Should Know

This guide is written for software developers and application architects who have already developed an understanding of the Genesys platform and the basics of SIP telephony. Before working with the SIP Endpoint SDK, you should also know how to use the logging functionality of the Platform SDK. In addition, the following document can be useful in understanding the Genesys SIP server environment:

- Framework 8.0 SIP Server Deployment Guide

Environment Prerequisites

To work with the SIP Endpoint SDK, 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:

- Genesys SIP Server 7.6.x, 8.0.2, or higher

- OS X Lion (10.7) or higher
- Xcode version 4.2.3 or higher

Related Resources

- SIP Endpoint SDK Developer's Guide

Installing SIP Endpoint SDK for Apple OS

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Prerequisites

- Check the list of environment prerequisites, and confirm that your system meets these standards prior to installing SIP Endpoint SDK.

Procedure

Start of procedure

1. Copy the zip archive containing the SIP Endpoint SDK for Apple OS from your product CD to your desired installation location.
2. Unzip the archive.
3. Copy the `ip_description.xml` and `read_me.html` files from your product CD into the root level of the unzipped archive.
4. Copy the `Log4Cocoa.framework` and `SipEndpoint.framework` folders located in the `Bin` folder of your installation to the `/Library/Frameworks` folder on your computer.
5. Copy the `libresample.dylib.1` file located in the `ThirdParty` folder of your installation to the `/usr/lib` folder on your computer.

End of procedure Next Steps

- To review the installation of your SIP Endpoint SDK files, continue by verifying the installed components.

Verifying Installed SIP Endpoint SDK for Apple OS Components

Warning

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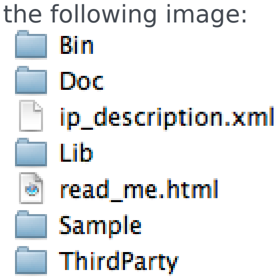
Prerequisites

- You must first complete the procedure that is found at Installing for Apple OS.

Procedure

Start of procedure

1. Use the Finder to locate the folder that you unzipped during the SIP Endpoint SDK installation.
2. Confirm that the SIP Endpoint SDK components contained in this folder look similar to what is shown in the following image:


3. Examine each folder (including the root installation folder) to confirm their contents. The SIP Endpoint SDK Folder Contents table below gives a description of the expected result.
4. Verify that the `/Library/Frameworks` folder on your computer contains the `Log4Cocoa.framework` and `SipEndpoint.framework` folders.
5. Verify that the `/usr/lib` folder on your computer contains the `libresample.dylib.1` file.

End of procedure Next Steps

- None

SIP Endpoint SDK Folder Contents

Folder	Contents
/	The root directory contains the following two files: • `ip_description.xml`—This file contains data for the `read_me.html` file. • `read_me.html`—This Read Me file identifies the build number, platform compatibility, and a link to the latest Release Note.
/Bin	This directory contains the `Log4Cocoa.framework` and `SipEndpoint.framework`, as well as the `SipEndpoint Sample.app`
/Doc	The `html` folder inside this directory contains the SIP Endpoint SDK API Reference, which has detailed information about the structure and usage of the SIP Endpoint SDK for Apple OS. This directory also contains the Third Party Software Notices.
/Lib	Contains a copy of `SipEndpoint.framework`.
/Sample	Contains the source code for the sample application. <ref name="sample_app">For more information about the sample application included with this release, see SIP Endpoint SDK OS X Sample Application.</ref>
/ThirdParty	This directory contains the following two files: • `libresample.dylib.1`, which should be copied into `/usr/lib` • `ReadMeFirst.rtf`, which gives the path where `libresample.dylib.1` has to be copied

<references />