PortSIP VoIP/IMS SDK

For developing SIP/3GP/IMS/VoLTE applications

The award-winning PortSIP VoIPis a powerful and highly versatile set of tools to dramatically accelerate SIP application development on ***Windows, iOS*** and ***Android*** platforms. It includes a suite of stacks, SDKs, add-ons and sample projects. Each one enables developers to combine all the necessary components to create an ideal development environment for every application's specific needs. The SDK complies with IETF and 3GPP standards, and is IMS-compliant (3GPP/3GPP2). These high performance tools provide multiple API layers for full user control and flexibility.

The PortSIP VoIP SDK is comprised of building blocks that simplify and dramatically reduce development time of applications, enabling engineers to develop complex voice, video, IM and Presence applications. High Level APIs hide SIP and Audio, Video complexity to accelerate development time. The SDK allows the developer to build his own graphical user interface and applications, and implement different calling behaviors for features such as call answer, call transfer, conferencing and more.

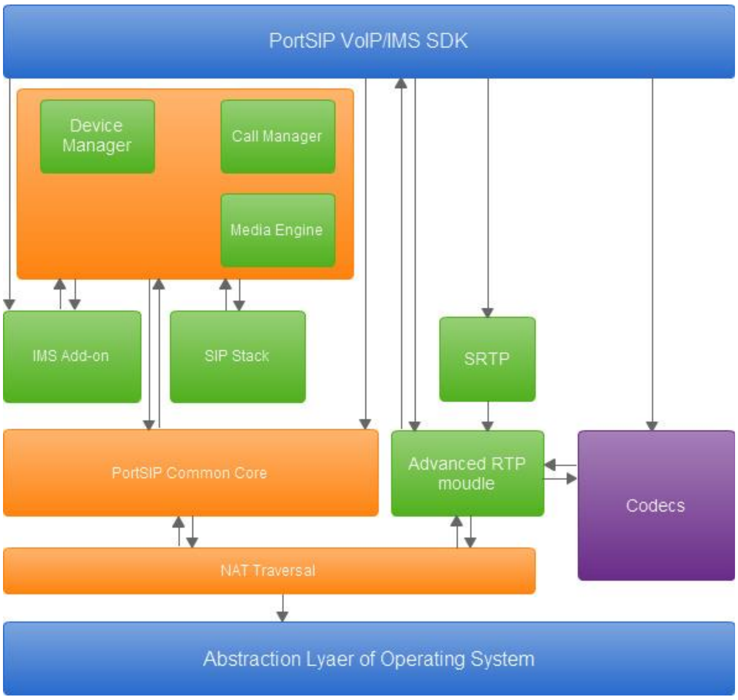
Architectural Overview

PortSIP VoIP SDK is built base on open standards, leveraging on SIP (Session Initiation Protocol) to create, run and terminate multi-media sessions. They are built on ITU codec standards for voice and video (G.711 aLaw/uLaw, G.729(a/b), iLBC, GSM, G.722.1, G.722, ISAC, ISAC-WB, AMR， AMR-WB, SPEEX, SPEEX-WB,OPUS, H.263, H.263-1998, VP8, H.264), and the SIP SIMPLE standards for Instant Messaging and Presence.

- **Support Windows, iOS and Android platforms**
- **Crystal clear HD audio, video, no any echo**
- **Carrier-grade audio and video codecs**
- **Security and encryption features (TLS, SRTP, usually use to avoid the SIP blocking)**
- **Enhanced Quality of service (QoS)**
- **Conferencing - audio and video conferencing**
- **Bandwidth scaling**
- **Contact Management**
- **IM and Presence (SIMPLE and XMPP)**
- **STUN NAT Traversal**
- **Open standards-based**
- **Interoperable with all of the major equipment vendors**
- **Rapid market deployment**
- **Simplified integration with Microsoft applications**
- **Simplified integration with SIP and other**
- **VoIP applications**
- **Easy integration with third party web applications**
- **Detailed Developer Guide**
- **Sample code**
- **Developer support options**

STUN Nat Traversal

PortSIP VoIP SDK support STUN (Simple Traversal Underneath NAT) NAT Traversal, it's a complete NAT traversal solution for developers, comprising the latest NAT-related solutions available for SIP, RTP and others. The PortSIP SDK NAT Traversal feature



complies with IETF Standard RFC 3489 and the latest draft-ietf-behave-rfc3489bis-05. This enables support for both existing STUN servers as well as those being developed, and serves as a foundation for ICE (Interactive Connectivity Establishment), which is based on the updated STUN standard.

Presence & Instant Messaging

PortSIP VoIP SDK supports SIP for Instant Messaging and Presence Leveraging Extensions(SIMPLE), it focuses on applying the Session Initiation Protocol
(SIP, RFC 3261) to services collectively known as Instant Messaging and Presence (IMP). Since the services SIP is used for share much in common with IMP, and it is such a mature and widely deployed protocol, the adaptation of SIP for IMP is a natural choice for IMP. PortSIP VoIP SDK implements logic/information layer for PUA (Presence User Agent).

SIP Stack

An internally multi-threaded (configurable) library containing all SIP-specific functionality, including message encoding and decoding, transaction and call management and SIP extensions.
PortSIP VoIP SDK using a very stable SIP stack, it’s full implementation of the RFC3261, and support more IETF RFCs, this SIP implementation supports UDP, TCP and TLS transport protocols, as well as IPV4 and IPV6.

High-definition Audio

PortSIP VoIP SDK support G.722.1, AMR-WB and SPEEX-WB, ISAC-WB audio codecs for the HD audio in 16kHZ. The codecs offer high-quality and low-bandwidth performance that works for the call.

High-definition Video

PortSIP VoIP SDK support HD Video(720P) on standard PCs. Natural, full-motion video up to 2Mbps @ Maximum 30fps

Telephony Features

- Multiple SIP lines
- Hold
- Do not disturb
- Call Ignore
- Auto Answer
- Call history
- Call transfer (Blind transfer and Attended transfer)
- Voice/video call recording
- Voice/video conferencing
- 3PCC

Extensions

- Send/Receive Out-Of-Dialog REFERS (with/without modification) & other SIP messages
- Subscription to SIP event packages during a specific call
- SIP preprocessor functionality-inspection/repair of SIP messages
- Access the incoming audio stream directly
- Access the incoming video stream directly
- Send the PCM stream directly to instead of microphone input
- Access the incoming SIP message and SIP message header directly
- Add/modify the SIP message headers.
- Support access incoming aduio stream directly.
- Support access incoming SIP message directly.
- Support access incoming video stream directly.
- Support play AVI file to remote side.
- Support play wave file to remote side.
- Support adding custom SIP header.
- Support modify SIP header.
- Support QoS.
- Audio conferencing, support maximum 100 participants audio conference. (recommended 8 parties)
- Video conferencing, support maximum 100 participants video conference. (recommended 3rd)
- Support send INFO and OPTIONS message.
- Support P2P call without SIP proxy server
- IM Support: SIMPLE(Presence, Subscribe, Pager message) and XMPP.
- Message waiting Indicator(MWI)
- Authentication: HTTP Basic, Digest Authentication.
- DTMF support: Send DTMF tone (RFC2833 and SIP INFO method), detect DTMF tone(RFC2833 and SIP INFO method).
- Multiple Call
- P2P call without SIP server
- Audio Tuning Wizard
- Video Tuning Wizard
- Microphone & Speaker Device Selector
- Microphone & Speaker Volume control
- Acoustic Echo Cancellation
- Automatic gain control
- Comfort Noise Generation
- Voice Activity Detector
- STUN support
- Outbound proxy server support
- Jitter buffer
- Free product version upgrades: one year free upgrades.
- Support develop WPF, Windows Form, Windows Service application, Android and iOS application.

Features and Specifications

- Support iOS, Android and Windows XP/Vista/7/8/10e.
- Support servers: Cisco CallManager, Kamailio, OpenSIPS, OpenSER, Asterisk, SIPX, PortaOne, Radvision, Nortel, Mitel, FreeSwitch, Avaya and other SIP Platforms.
- Support development tools:
  MS Visual Basic 2005/2008/2010/2013/2015
  MS C#
  MS Visual C++(8.0/9.0、10)
  Delphi XE
  Apple XCode
  Eclipse
  Android Studio
  Xamarin
- Audio call: G.711 aLaw/uLaw, G.729(a/b), iLBC, GSM, G.722.1, G.722, AMR，AMR-WB, SPEEX, SPEEX-WB, ISAC, ISAC-WB, Opus
- Video Call: H.263, H.263-1998, VP8, H.264
- Call transfer: Attended transfer, Blind transfer.
- PRACK support
- Call forwarding.
- Call hold, mute speaker, mute microphone.
- Do not disturb(DND), Auto answer(AA).
- Audio record: record audio as wave file.
- Video record: record video as AVI file.
- Support 720P, SVGA, XVGA, VGA, QVGA, CIF, QCIF video resolution.
- Support TLS/SRTP(usually use to avoid SIP blocking)

A part of customers

HP: HP purchased PortSIP VoIP SDK to integrate VoIP feature with Telecom Audio to HPVR.

Philips: Philips purchased PortSIP VoIP SDK for Philips CareServant in 2007.

Cisco: Cisco(Brazil) purchased PortSIP VoIP SDK for their products in 2010.

Siemens: Siemens(India) purchased PortSIP VoIP SDK for their products in 2011.

Agilent: Agilent purchased PortSIP VoIP SDK in 2011.

Qualcomm: Qualcomm Incorporated. purchased PortSIP VoIP SDK in 2012 for VoLTE project.

Fujitsu: Fujitsu purchased our "Customization Softphone" in October 2012.

Alibba: Alibba Group purchased PortSIP VoIP SDK in 2013.

NEC: NEC purchased PortSIP VoIP SDK in 2013.

CITI: CITI bank purchased PortSIP VoIP SDK in 2014.

Unify: Unify Inc. (Siemens Enterprise Communications GmbH & Co. KG) purchased PortSIP VoIP SDK in 2015.

KPMG: KPMG (Taiwan) purchased PortSIP VoIP SDK in Aug, 2015.

Company

PortSIP is a leading provider of VoIP, Unified communication, video conferencing and telepresence technologies over IP and wireless networks, offering end-to-end visual communications that help businesses collaborate more efficiently.

PortSIP is constantly improving its products, quickly responding to the needs and suggestions of established customer base that spans 50 countries on 6 continents. The quality of our products and technical support is greatly appreciated by our customers.

PortSIP's customers include some of the world's largest telecommunications service providers and IT companies, including HP, Philips, CallCentric, Qualcomm, NEC, Cisco(Brazil), Siemens(India), Unify Inc., KMPG(Taiwan), Agilent, Keysight, Fujitsu, Alibaba Group.

We serve 2,800+ companies customers around the world in past 6 years. Our diverse product line will meet your need regardless if you are a small, local new regional ITSP looking for a cost effective competitively featured client solution or a traditional large telecommunication company looking to enhance your branded triple threat options with customized state-of-the-art voice and video solutions integrated into your existing "look and feel" voice, video services platform.

Our potent combination of technology and services is supported by strategic alliances with other innovative companies. Together with our partners, PortSIP is opening the door to the next generation of global communication.

For more information, please visit:

PortSIP Solutions, Inc.

http://www.portsip.com

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