Conaito Technologies

Introduces powerful and highly versatile VoIP SIP Client SDK

VoIP SIP Client SDK

A powerful and highly versatile VoIP SDK to accelerate development of SIP applications and websites - Release v1.6

Software Product: Conaito VoIP SIP Client SDK - Version: 1.6
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VoIP SIP Client SDK

A powerful and highly versatile VoIP SDK to accelerate development of SIP applications and websites

The VoIP SIP Client SDK provides a powerful and highly versatile solution to add quickly SIP (Session Initiation Protocol) based dial and receive phone calls features in your software applications and websites. It accelerates the development of SIP/RTP compliant soft phone with a fully-customizable user interface and brand name.

The SDK contains a high performance VoIP conferencing client capable of delivering crystal clear sound even for both low and high-bandwidth users and SIP compatible devices (hardware and software). It enables a worldwide communication over the internet or intern networks either and delivers superior voice quality by integrating digital voice processing features including auto gain controller (AGC), acoustic echo suppression (AES) and noise suppression. It supports multiple lines, multi-party voice conference, call hold, call forwarding and transfer, DTMF, Packet Loss Concealment (PLC), adaptive jitter buffer, record and playing WAV and much more!

Conaito VoIP SIP Client SDK is based on IETF standards (SIP, RTP, STUN, TURN, ICE etc.), so it should be compatible with other standard based products such as: SER, Sip EXpress, OpenSER and Asterisk.

New features of the version 1.6:

- Multiple sip accounts registration support (unlimited)
- Recording voice conversation into mp3 (.mp3) file (without or ID3v2.1 tag supported)
- Playing mp3 (.mp3) files to the remote end (without or ID3v2.1 tag supported)
- Codec selection and sorting
- Fixed VB6 example bug
- Fixed STUN event bug
- Additional minor improvements as well file size optimization and performance of the SDK

Key features of the VoIP SIP Client SDK:

- Easily make and receive SIP (Session Initiation Protocol) based phone calls through any SIP gateway or SIP compliant IP-Telephony service provider
- VoIP conferencing with crystal clear sound even for both low and high-bandwidth users
- Supported audio codec´s: G729, G723, G711 A-Law, G711 U-Law, Speex, Speex-wb, GSM6.10 and iLBC
- Open standards-based and interoperable with all of the major equipment vendors
- UDP and TCP support
- Multi-party voice conference support (Conference split and join, locally mixed conferences)
- Multi-line support (multiple simultaneous calls)
- SIP Instant/Chat Messaging with send/receive controlling
- Integrated STUN, TURN and ICE NAT Traversal

- Comes with SIP Server demo to provide in bundle with the Conaito SIP Client a ready up own SIP VoIP and Instant Messaging network solution.
- P2P support for directly connections between 2 SIP clients without SIP Server (no SIP Server necessary)
- Outbound proxy server support
- Encrypted SIP account settings (encrypted SIP account settings in your webpage)
- Line Hold/Un-hold support
- Call forwarding and rejection
- Call transfer support
- Select media input/output devices - on-the-fly (as well during a conversation/conference)
- Microphone and Speaker volume with Mute (including level indicator)
- Auto-answer
- DND (Do Not Disturb)
- Adaptive Jitter buffer
- PLC (Packet Lost Concealment)
- AGC (auto gain controller)
- AES (Acoustic echo cancellation or suppression)
- Noise cancellation or suppression
- DTMF tones support (generation/detection)
- Recording voice conversation into PCM WAVE and mp3 (.wav and .mp3) file
- Playing PCM WAVE and mp3 (.wav and .mp3) files to the remote end
- Audio file memory cache
- Extended SIP URL functions
- Registration on SIP Server (SIP Registrar)
- Log file on/off setting
- Keep-alive packets to NAT/firewall
- Fully-customizable user interface
- Production-ready Microsoft Authenticode Certificate certified
- Works with all kind of Internet connections
- Friendly to NAT and other firewalls
- Royalty free licensing
- No Yearly/Monthly fee
- Very easy to incorporate
- SDK comes in bundle with ActiveX control (Webdemo with ready-up signed CAB included), native DLL with .NET Interface for easy usage in .NET development (no ActiveX registration necessary)
- Fully sample applications for various programming languages such as sample source code for C#, VB.NET, C++, VB 6.0, Delphi and HTML/JavaScript (Webdemo) for ActiveX, DLL and .NET version
- For .NET framework as well and all development environments with native DLL, .NET or ActiveX support

Easy, familiar, event-driven call control ActiveX, DLL and .NET

- Easy to use; quick development
- Powerful .NET Interface for easy .NET development (no ActiveX registration necessary)
- Support for .NET framework as well and all development environments with native DLL, .NET or ActiveX support
- Very easy to incorporate

Rich call control feature set

- Multi-party voice conference support (Conference split and join, locally mixed conferences)
- Multi-line support (multiple simultaneous calls)
- SIP Instant messaging
- Locally mixed conferences
- Hold/Mute
- Call transfer and redirect
- Call forwarding and rejection

Industry leading SIP support

- RFC3261 compliant SIP stack
- RFC2833 out-of-band DTMF signaling
- Integrated STUN, TURN and ICE support

Comprehensive configuration support

- Select media input/output devices - on-the-fly (as well during a conversation/conference)
- Configurable ports (RTP, SIP UDP, SIP TCP, STUN, TURN, ICE)
- SIP proxy

Advanced digital voice processing features

- AGC (auto gain controller)
- AES (Acoustic echo cancellation or suppression)
- Noise cancellation or suppression

… and much more!

Having the above features available makes it simple to develop any type of VoIP-enabled application, like e.g. a SIP softphone, IVR solution, teaching tool, live support, voip chat, meeting tool or any other type of application which requires users being able to talk and type messages to each other.

For Conaito VoIP SIP clients to be able to interact with each other they must connect to a SIP gateway or SIP based IP-Telephony service provider.

Just relax!
Please, don't hesitate trying our VoIP SIP Client SDK at once and get yourself, as well as your customers, the exciting experience of easy, fast and high quality standard applications which VoIP-enable your application and website.

We hope you enjoy the new Conaito VoIP SIP Client SDK – A powerful and highly versatile VoIP SDK to accelerate development of SIP applications and websites.

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