Ozeki VoIP SDK Data Sheet

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Ozeki VoIP SIP SDK	Product information
Product name	Ozeki VoIP SIP SDK
Category	Software Development Kit
Product website	http://www.voip-sip-sdk.com
Latest version	OZEKI VoIP SIP SDK v11.1.5
Release date	2014.12.01.
Package size	56.7 MB
Download url	http://www.voip-sip-sdk.com/p_21-download.html
Package contents	• Redistributable .DLL • Documentation • Example applications • Exe demo • Full source code (optional)
Main task	Makes it possible to build a VoIP client application or a PBX, based on the SIP protocol.
Connectivity	It connects to a supported VoIP PBX or to a VoIP service provider over the Internet. Supports firewall passthrough (STUN/TURN).
Supported client OS	• Windows server 2003 • Windows server 2008 • Windows server 2012 • Windows XP • Windows Vista • Windows 7 • Windows 8
Required .NET framework	At least .NET Framework 3.5 or any newer version
Supported programming languages and	Microsoft Visual Studio 2008, 2010, 2012, 2013 (C#, VB.NET, ASP.NET,...)

environments	
Source code	Full source code can be purchased. The source code of this VoIP SDK is in C#.Net.
Developer features	• Easy to use • Very easy to incorporate • Makes quick development possible • Supports all development environments with .NET support • Supports the development of WPF, Windows Form, Windows Service application, etc. • Free product version updates: one year free updates
Call features	• **Multiple simultaneous call support** • Make and receive Audio/Video calls • Make and receive peer-to-peer calls without a SIP server • Call reject • Hold/Unhold • Call forward • Call transfer (Attended and Blind transfer) • DTMF support: send/receive DTMF signals via RFC 2833, SIP INFO or inband • Do Not Disturb(DND) • Auto Answer • Redial • **Secure calls** (TLS, SRTP) • Codec priority change • Caller ID modification for outbound calls • Call history • **Multi-party voice conference support** (Conference split/join, locally mixed conferences)
SIP features	• Multiple SIP account support • Secure SIP connection via TLS • Message Waiting Indicator (for checking voicemail) • Send and receive instant messages • Digest authentication • Supported SIP methods: REGISTER, INVITE, ACK, CANCEL, OPTIONS, BYE, SUBSCRIBE, NOTIFY, REFER, INFO, MESSAGE • Direct access to incoming and outgoing SIP messages (add/modify SIP headers for inspect or repair) • Multipart SIP body handling • Outbound proxy server support • Subscription to SIP event packages

PBX features	• **Fully customized dial plan creation** (call and message routing) • Custom extension or custom connection creation (eg. voicemail, SIP trunk, echo test) • Rich call information and call event notification • Third-party call control (forward, hold, transfer, hangup) • **Direct access to caller and callee RTP stream outside of a call** • Authentication control • Music on hold
Network features	• Multiple network interface support • Supported protocols: UDP, TCP, TLS, SIP, SDP, RTP, SRTP, STUN, TURN, ICE • Configurable port range • Firewall/NAT passthrough (auto discovery, STUN, static IP setting)
Audio features	• **Microphone & Speaker device selection** (**on-the-fly as well during a conversation/conference**) • Device calibration (volume, level, mute, device change, format change) • **Play wav or mp3 files to remote party** • **Record audio** in wav or mp3 format • Text-To-Speech support (changing voice, setting speech rate, multiple TTS engines) • Speech-To-Text/Speech recognizer support (changing voice, multiple STT engines) • Play DTMF tones • **Play audio from multiple audio sources to remote party** • HD audio support (HD audio calls) • **Supports most audio formats** (8000-48000 Hz, 16bit, mono/stereo) • Automatic audio format conversion • Support access incoming and outgoing audio stream directly • Adaptive jitter buffer • Packet loss concealment
Video features	• **Camera device selection** (**on-the-fly as well during a conversation/conference**) • Device calibration (device change, resolution/frame rate change) • Play video files to remote party (mp4) • Record video in mp4 format • 3D video support • **Real-time video quality change** • Picture manipulation (rotate, flip) • 720p, SVGA, XVGA, VGA, CIF, QCIF video resolutions • Support access incoming and outgoing video stream directly
Advanced Digital Signal	• Auto Gain Control (AGC)

Processor features	• Noise Reduction • Voice Activity Detection (VAD) • Acoustic Echo Cancellation (AEC) • Answer Machine Detection (predictive dialer)
Supported audio codecs	• PCMA (G.711 aLaw) • PCMU (G.711 uLaw) • GSM • iLBC (20 and 30 ms) • Speex Narrowband (8kHz) • Speex Wideband (16kHz) • Speex Ultrawideband (32kHz) • G.722 • G.723 • G.726-16 • G.726-24 • G.726-32 • G.726-40 • G.728 • G.729 • L16
Supported video codecs	• H.264 • H.263-1998 • H.263
Fields of application	• Softphones • Webphones • Online chat communities (e.g.: dating, business meetings) • PBX servers • Automatic call distributors • Telephone and call centers • VoIP providers • Conferencing applications
Supported PBX systems	• Ozeki Phone System XE • Ozeki Phone System XE PBX • Cisco Unified CM PBX • Cisco Call Manager Express PBX • Asterisk PBX • 3CX PBX • AsteriskNow PBX • Kamailio PBX • FreeSwitch PBX • OpenSIPS PBX

	• Trixbox PBX • OpenSER PBX • PBXnSIP PBX • PBXpress PBX • SipX ECS PBX • Elastix PBX • FreePBX PBX • SwyxWare PBX • Aastra MX-One PBX
Example applications	• Silverlight Video Chat • SIP MediaGateway • Flash SIP client example • Flash chat example • WinForms DTMF IVR • VB.NET DTMF IVR • C# DTMF IVR • C# Voice recognition IVR
Standards	• RFC 2833:RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals • RFC 3261:Session Initiation Protocol • RFC 3263:SIP: Locating SIP Servers • RFC 3264:An Offer/Answer Model with the (SDP) • RFC 3265:SIP Event Notification • RFC 3420:Internet Media Type message/sipfrag • RFC 3428:SIP Instant Messaging • RFC 3489:STUN - Simple Traversal of UDP Through NATs • RFC 3515:SIP Refer Method • RFC 3550:Real-time Transport Protocol • RFC 3551:RTP Audio/Video Conference • RFC 3587:IPv6 Global Unicast • RFC 3666:SIP, PSTN, Call Flows • RFC 3725:Best Practices for Call Control • RFC 3842:Message Waiting Indication • RFC 3856:Presence Events in SIP • RFC 3891:The SIP Replaces Header • RFC 3892:SIP Referred-By Mechanism • RFC 3920:Extensible Messaging and Presence Protocol (XMPP): Core • RFC 4566:Session Description Protocol • RFC 5411:A Hitchhiker's Guide to the SIP

Try it now

Visit our webpage and download the demo version of Ozeki VoIP SIP SDK to test its functions:

Download URL:
http://www.voip-sip-sdk.com/p_21-download-ozeki-voip-sip-sdkvoip.html

Limitations in the trial version

The demo version has no limitations in the number of simultaneous calls. After the 4th call, one of every 4 calls plays a demo notification message. Another limitation is the time limit of 20 days for evaluation.

Cost structure

One time investment for the software license, optional technical support and version update service available.

The license cost depends on the number of simultaneous calls you wish to make, request a quotation at info@voip-sip-sdk.com!

For more information please visit

www.voip-sip-sdk.com

or contact us at

info@voip-sip-sdk.com!