OZEKI Systems Ltd.

Ozeki VoIP SDK Data Sheet

Document version	v.3.0.0.

Ozeki VoIP SIP SDK	Product information
Product name	Ozeki VoIP SPI SDK
Category	Software Development Kit
Product website	https://www.voip-sip-sdk.com
Download url	https://www.voip-sip-sdk.com/p_7021-download-ozeki-voip-sip-sdk.html
Package contents	Redistributable .DLL Documentation Example applications Exe demo Full source code (optional)
Main task	Makes it possible to build VoIP client software 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 11, Windows 10, Windows 2008, Windows Server 2022, Windows Server 2019, Windows Server 2016
Required .NET framework	.NET Framework 3.5 or .NET Framework 4.0
Supported programming languages and environments	Microsoft Visual Studio 2022, 2019, 2017, 2015, (C#, VB.NET, J#, ASP.NET,...) MS Visual Studio 6(VC6, VB6, ...) Borland C++ 5/6/7 Borland Delphi 6/7 CodeGear Delphi 2007 CodeGear C++ Builder 2007 Web technologies (ActiveX)
Source code	Full source code can be purchased. The source code of this VoIP SDK is in C#.Net.
Basic Telephony and telephone functions	Hold, Transfer, Do Not Disturb(DND), Auto answer, Redial, Multiple SIP lines, Call Ignore, Call history, Voice call recording, Voice conferencing. PC-PC or PC-phone calls, Caller ID with name, Quick calling and

	re-dialing, Call history, Connect to PSTN and mobile phones, Video calls
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 ❑ Voice Activation Detection
Fields of application	❑ Softphones ❑ Webphones ❑ Online Chat Communities (e.g.: dating, business meetings) ❑ VoIP Providers ❑ VoIP Devices ❑ Embedded Phones ❑ Conferencing Applications
Audio file play and record	YES (Supports .wav and .mp3 files) Audio format can be: ❑ 8K 16bit mono PCM ❑ 8k 8bit mono mulaw/alaw
Supported SIP Methods	REGISTER, INVITE, CANCEL, INFO, BYE, ACK, SUBSCRIBE, OPTIONS.
Authentication	HTTP authentication (BASIC).
RTP Package Access	Support access incoming and outgoing RTP audio stream directly. And support change RTP audio stream to integrate TTS and ASR engine.
Extensions	❑ Send/Receive Out-Of-Dialog REFERS (with/without modification) & other SIP messages ❑ Subscription to SIP event packages during a specific all SIP preprocessor functionality-inspection ❑ 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
Easy, familiar, event-driven call control	❑ Easy to use; quick development ❑ Support for all development environments with .Net support ❑ Very easy to incorporate
Rich call control feature set	❑ Multi-party voice conference support (Conference split/ join, locally mixed conferences) ❑ Multi-line support (multiple simultaneous calls) ❑ Multiple lines for multiple concurrent calls ❑ SIP Instant messaging ❑ Locally mixed conferences ❑ Hold/Mute ❑ Call transfer ❑ Call forwarding and rejection
Audio features	❑ Adaptive jitter buffer ❑ Packet loss concealment for voice and video ❑ Automatic Gain Control (AGC) for voice ❑ Voice Activity Detection (VAD) ❑ Predictive dialing (answering machine detection) ❑ Acoustic Echo Cancellation (AEC) ❑ Narrow band and wide-band voice codec choice: G711A, G711U, iLBC, Speex, Speex-wb, GSM, G.729a, L16, G.723, G.726-16, G.726-24, G.726-32, G.726-40, G.728
Video features	❑ HD video phoning ❑ Jitter buffer ❑ Supported video codecs: H.263, H.264 ❑ Picture rotate, flip ❑ 720p, svga, xvga, vag, cif, QCIF video resolution
Supported PBX systems	❑ Ozeki Phone System ❑ Cisco Unified CM PBX ❑ Asterisk PBX ❑ 3CX PBX ❑ AsteriskNow PBX ❑ Kamailio PBX ❑ FreeSwitch PBX ❑ OpenSIPS PBX ❑ SipX ECS PBX ❑ Trixbox PBX ❑ OpenSER PBX ❑ PBXnSIP PBX

	❑ PBXpress PBX ❑ Elastix PBX ❑ FreePBX PBX
Example applications	❑ C# WPF softphone ❑ Windows Forms Softphone Sample ❑ Windows Forms Softphone VB NET ❑ Visual C++.Net Softphone ❑ Silverlight Video Chat Example ❑ SL Mediagatewayeasyconnection example ❑ MediaGateway SIP Example ❑ Flash SIP Client Example ❑ MediaGateway SDK Flash chat example ❑ WinForms DTMF IVR ❑ C# DTMF IVR ❑ C# Voice recoginition IVR ❑ C# Autodialer example ❑ C# Speech to text ❑ C# Voice conference room ❑ C# Command line caller ❑ SIP SMS Example ❑ C# Callback Form ❑ ASP VoIP example
Standards	❑ RFC 2833:RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals ❑ RFC 3261:Session Initation 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
Features and Specifications	❑ Audio call: G.711 aLaw/uLaw, G.729(b), iLBC, GSM, G.722, SPEEX, SPEEX-WB. ❑ Call hold, mute speaker, mute microphone ❑ Do not disturb(DND), Auto answer(AA) ❑ Audio record: record audio as wave file ❑ Support access incoming audio stream directly ❑ Support access incoming SIP message directly ❑ Support play wave file to remote side ❑ Support adding custom SIP header ❑ Support modify SIP header ❑ Audio conferencing ❑ Message waiting Indicator(MWI) ❑ Authentication: HTTP Basic, Digest Authentication ❑ DTMF support: Send DTMF tone(RFC2833), detect DTMF tone(RFC2833) ❑ Multiple Call ❑ Microphone & Speaker Device Selector ❑ Microphone & Speaker Volume control ❑ Acoustic Echo Cancellation ❑ Automatic gain control ❑ Comfort Noise Generation ❑ Voice Activity Detector ❑ STUN/TURN support ❑ Outbound proxy server support ❑ Jitter buffer ❑ Free product version updates: one year free updates. ❑ Support develop WPF, Windows Form, Windows Service. application, etc. ❑ Call forward ❑ Full HD

Try it now

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

On-line demo URL:
https://www.voip-sip-sdk.com/p_7021-download-ozeki-voip-sip-sdk.html

Limitations in the trial version

The demo version allows to make 4 simultaneous calls. More simultaneous calls are available in the licensed versions.

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!