

Ozeki VoIP SDK Data Sheet

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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 (IPv4 and IPv6 supported). 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 4.8 or .NET 8
Supported programming languages and environments	Microsoft Visual Studio 2026, 2022 (C#, VB.NET,...) JetBrains Rider
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	<input type="checkbox"/> Select media input/output devices (on-the-fly as well during

configuration support	<p>a conversation/conference)</p> <ul style="list-style-type: none"> <input type="checkbox"/> Configurable ports (RTP, SIP UDP, SIP TCP, SIP TLS, STUN, TURN, ICE) <input type="checkbox"/> SIP proxy
Advanced digital voice processing features	<ul style="list-style-type: none"> <input type="checkbox"/> AGC (auto gain controller) <input type="checkbox"/> AES (Acoustic echo cancellation or suppression) <input type="checkbox"/> Noise cancellation or suppression <input type="checkbox"/> Voice Activation Detection
Fields of application	<ul style="list-style-type: none"> <input type="checkbox"/> Softphones <input type="checkbox"/> Webphones <input type="checkbox"/> Online Chat Communities (e.g.: dating, business meetings) <input type="checkbox"/> VoIP Providers <input type="checkbox"/> VoIP Devices <input type="checkbox"/> Embedded Phones <input type="checkbox"/> Conferencing Applications
Audio file play and record	<p>YES (Supports .wav and .mp3 files) Audio format can be:</p> <ul style="list-style-type: none"> <input type="checkbox"/> 8K 16bit mono PCM <input type="checkbox"/> 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	<ul style="list-style-type: none"> <input type="checkbox"/> Send/Receive Out-Of-Dialog REFERS (with/without modification) & other SIP messages <input type="checkbox"/> Subscription to SIP event packages during a specific all SIP preprocessor functionality-inspection <input type="checkbox"/> Access the incoming audio stream directly <input type="checkbox"/> Access the incoming video stream directly <input type="checkbox"/> Send the PCM stream directly to instead of microphone input <input type="checkbox"/> Access the incoming SIP message and SIP message header directly <input type="checkbox"/> Add/modify the SIP message headers
Easy, familiar, event-	<ul style="list-style-type: none"> <input type="checkbox"/> Easy to use; quick development

<p>driven call control</p>	<ul style="list-style-type: none"> <input type="checkbox"/> Support for all development environments with .Net support <input type="checkbox"/> Very easy to incorporate
<p>Rich call control feature set</p>	<ul style="list-style-type: none"> <input type="checkbox"/> Multi-party voice conference support (Conference split/join, locally mixed conferences) <input type="checkbox"/> Multi-line support (multiple simultaneous calls) <input type="checkbox"/> Multiple lines for multiple concurrent calls <input type="checkbox"/> SIP Instant messaging <input type="checkbox"/> Locally mixed conferences <input type="checkbox"/> Hold/Mute <input type="checkbox"/> Call transfer <input type="checkbox"/> Call forwarding and rejection
<p>Audio features</p>	<ul style="list-style-type: none"> <input type="checkbox"/> Adaptive jitter buffer <input type="checkbox"/> Packet loss concealment for voice and video <input type="checkbox"/> Automatic Gain Control (AGC) for voice <input type="checkbox"/> Voice Activity Detection (VAD) <input type="checkbox"/> Predictive dialing (answering machine detection) <input type="checkbox"/> Acoustic Echo Cancellation (AEC) <input type="checkbox"/> Narrow band and wide-band voice codec choice: G711A, G711U, OPUS, iLBC, Speex, Speex-wb, GSM, G.729a, L16, G.723, G.726-16, G.726-24, G.726-32, G.726-40, G.728
<p>Video features</p>	<ul style="list-style-type: none"> <input type="checkbox"/> HD video phoning <input type="checkbox"/> Jitter buffer <input type="checkbox"/> Supported video codecs: H.263, H.264 <input type="checkbox"/> Picture rotate, flip <input type="checkbox"/> 720p, svga, xvga, vag, cif, QCIF video resolution
<p>Supported PBX systems</p>	<ul style="list-style-type: none"> <input type="checkbox"/> <u>Ozeki Phone System</u> <input type="checkbox"/> <u>Cisco Unified CM PBX</u> <input type="checkbox"/> <u>Asterisk PBX</u> <input type="checkbox"/> <u>3CX PBX</u> <input type="checkbox"/> <u>AsteriskNow PBX</u> <input type="checkbox"/> <u>Kamailio PBX</u> <input type="checkbox"/> <u>FreeSwitch PBX</u> <input type="checkbox"/> <u>OpenSIPS PBX</u> <input type="checkbox"/> <u>SipX ECS PBX</u> <input type="checkbox"/> <u>Trixbox PBX</u> <input type="checkbox"/> <u>OpenSER PBX</u> <input type="checkbox"/> <u>PBXnSIP PBX</u> <input type="checkbox"/> <u>PBXpress PBX</u> <input type="checkbox"/> <u>Elastix PBX</u> <input type="checkbox"/> <u>FreePBX PBX</u>

Example applications

- [C# WPF softphone](#)
- [Windows Forms Softphone Sample](#)
- [Windows Forms Softphone VB NET](#)
- [Visual C++.Net Softphone](#)
- [Silverlight Video Chat Example](#)
- [SL Mediatgatewayeasyconnection example](#)
- [MediaGateway SIP Example](#)
- [Flash SIP Client Example](#)
- [MediaGateway SDK Flash chat example](#)
- [WinForms DTMF IVR](#)
- [C# DTMF IVR](#)
- [C# Voice recognition 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 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](#)

Features and Specifications

- ❑ Audio call: G.711 aLaw/uLaw, G.729(b), iLBC, GSM, G.722, SPEEX, SPEEX-WB, OPUS.
- ❑ 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!