What is SIP Interconnect?

OpenTok SIP Interconnect enables interoperability between WebRTC endpoints and existing telephony systems so that users can make in-context SIP-based audio calls from wherever they are, while simultaneously browsing the website or mobile application.

Core Features

- **Interoperability**
  
  Interoperability between the OpenTok platform and other telephony systems including PSTN, IMS, and PBX.

- **Multi-Party Calls**
  
  Invite an expert, or host a videoconference with multiple users across WebRTC & SIP endpoints.

- **Archiving**
  
  Our industry-leading Archiving API enables secure call recording.

- **Video, Voice Calls, & Text Chat**
  
  Offer a variety of communications options with reliable video and audio calls or text chat.

- **Co Browsing & Content Sharing**
  
  Enable richer and more collaborative interactions with co-browsing and content sharing.

What is WebRTC?

WebRTC (Web Real-Time Communication) is an open standard for embedding real-time communications directly into web browser and mobile applications. WebRTC offers better video quality than predecessor technologies, up to 6x faster connection times, reduced audio/video latency and complete customizability.
Architecture - How Does It Work?

The standard way to connect to the OpenTok platform is with the OpenTok client SDKs that use the proprietary signaling interfaces provided by TokBox. With SIP Interconnect, the OpenTok platform exposes a SIP interface that enables access to existing 3rd party telephony systems. This interface enables users, who are connected via such 3rd party telephony platforms, to participate in the audio portion of an OpenTok session. For example, an application would make a SIP call to a customer contact center, which routes the call to an agent, using existing queuing logic. The agent then accepts the phone call and participates in the session by voice. The agent has the option to communicate via live video by joining through a regular OpenTok WebRTC session.

1. End users call through an OpenTok-powered “click-to-call” interface on Customer’s website or mobile app.
2. OpenTok initiates the request to Customer’s SIP network. OpenTok transcodes WebRTC media (audio) to SIP and vice versa.
3. Customer’s call center infrastructure bridges the call between the end user and agent through OpenTok SIP Interconnect.
4. Call center agent answers the incoming call. The agent can invite a third person to join the call for further assistance.

Benefits

Seamless Integration
SIP Interconnect embeds seamlessly into existing call center infrastructure so their existing systems can be reused efficiently.

Greater Access
Offer convenient access to contact center agents from websites & mobile apps.

Context
Collect contextual data from visitor’s web or mobile-application session so that the call can be routed to the best agent who knows where the customer is calling from.

Reduced time to resolution
Resolve customer issues quickly & efficiently, saving valuable time and money.

Increased customer satisfaction
With click-to-call capabilities and calls in context, customers no longer have to dial in, wait in a long call queue & explain their problem, or download any third party software.
## Technical Specifications

### Media
- RFC 3550 (RTP/RTCP) support
- Support for encrypted (SRTP) & unencrypted (plain RTP)

### Audio-Stream Encoding
- Opus, G.711 and G.722

### Signaling
- RFC 3561 (SIP) support including UDP, TCP & TLS transports
- SIP dialog initiated by the OpenTok SIP gateway supported
- SIP header manipulation

### Media Applications & Services
- Call recording
- Support for both one-to-one and multi-party

### Duration
- Calls > 6 hours will be closed

### Security
- Signaling encryption security using TLS
- Media encryption using SRTP
- Exchange of security keys using DTLS and SDES
- IP whitelisting

### Note:
- All of the existing functionality of the OpenTok Platform, including multiparty sessions and archiving, are compatible with SIP Interconnect.
- SIP Interconnect supports audio only through the SIP interface, and does not currently support video.
- SIP Interconnect does not include built-in PSTN functionality, such as phone numbers, SMS, etc.