WebRTC SDKs

Improve team collaboration through live interactions by adding real-time video and audio to your conferencing options. Increasing personal experience and engagement has been shown to make teams significantly more effective and productive.

See how we can help you succeed

Telestax offers web, iOS and Android SDKs and templates that allow you to leverage the features and advantages of WebRTC. We take care of the underlying complexities for you so you can concentrate on building real-time voice and video apps.

Secure audio and video provides peace of mind

Provides continuous voice and video encryption. The Secure RTP protocol (SRTP) is used for encryption and authentication, which is especially beneficial over WiFi networks because it prevents eavesdropping.

Advanced voice and video quality for a better experience

Utilizes the computer's adjustable built-in microphone settings to allow for much better sound quality than Flash. It also takes advantage of advanced codecs to produce high fidelity voice quality.

Interoperability with UC solutions enhances existing apps

Interoperates with Unified Communications solutions allowing it to be seamless integrated with a company's existing technology infrastructure. With RestcommONE SDKs you can be in the WebRTC business in mere minutes.

Interoperability with VoIP and video for easy integration

A huge value is its promise of interoperability with existing voice and video systems. This includes devices using SIP, Jingle, XMPP, and the PSTN. If the existing voice and video devices are utilizing standard protocols, they will most likely work with WebRTC-based devices.

Reliable session establishment increases productivity

Support for reliable session establishment. This is true for Network Address Translators (NAT), something that hinders and may block other communications and collaboration protocols. The reliable operation reduces latency and increases quality. It also reduces the server load.

Platform and device independence offer choice

Web services applications utilizing WebRTC-enabled browsers can direct the browser to create a real-time audio or video connection to another WebRTC device or to a WebRTC media server – no matter the operating system. The browser operating system is not relevant. Developers write HTML5 code that executes on desktops and mobile devices.

Multiple media streams ensure quality

Compensates and adjusts to changing network conditions. It adjusts the communications quality, responds to bandwidth availability, detecting and avoiding congestion. The sending and receiving browsers communicate conditions that are then analyzed and responded to changes in network conditions.