# WebRTC Interconnect Framework

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WebRTC Interconnect Framework powered by Softil's award-winning toolkits allows developers of Unified Communications (UC) solitons to quickly and effectively enable seamless connectivity between popular WebRTC endpoints and enterprise UC systems. Such integration of the WebRTC communication capabilities into the workflow of enterprise communications allows modern enterprise service new categories of the end-users, greatly extending enterprise's communication reach.

## Extending the reach of UC with WebRTC

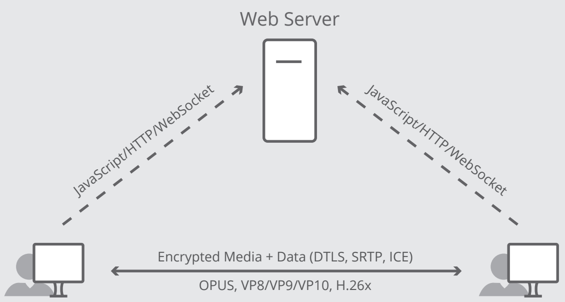
Google's WebRTC project, launched in 2011, created a media stack that can be built-in to a browser so that anyone can place a voice or video call from within that browser without needing to download any additional software.

Today, WebRTC technology is standardized in the W3C and IETF and software vendors are able to implement support for WebRTC within their UC products to better address several pain points for their business and public sector customers.

Adding support for WebRTC to a videoconferencing product, for example, could significantly reduce the hassle of setting up and joining conference calls, by removing the need for participants to download a separate client application or install a browser plug-in.

Contact center platform vendors could make it easier for businesses to offer their customers 'click to speak with an agent' functionality on their websites thereby helping them to provide a more joined-up and differentiated sales and customer service experience.

For vendors of SIP-based UC products, adding support for WebRTC to the enterprise communications facilitates creation of the flexible solutions allowing end-users to enjoy the simplicity of the browser-based calling from anywhere at any time while maintaining security and enabling integration with enterprise workflow.




WebRTC Triangle

## WebRTC Basics

WebRTC was initially developed by Google to enable browser-based real-time audio and video communications without the need to install any additional software on the computer.

Unlike traditional multimedia communication protocols, such as SIP and H.323, WebRTC doesn't define signaling, allowing web application developers to decide on the best way to exchange connection information for media exchange. Web application developers can use any browser-based communication mechanism, such as HTTP/HTTPS, JSON, Java Script, or any other to exchange codec options to be used to establish the media flow.

Next, WebRTC media flow is established directly between two web browser clients, as shown at the "WebRTC Triangle" diagram below. WebRTC media flow is aways secure, using DTLS protocol to exchange the encryption keys, and SRTP/SRTCP protocols to transmit the encrypted media. Such WebRTC media flow can also traverse firewalls and NATs with the help of traditional ICE/STUN/TURN protocols embedded in the same WebRTC media flow. All of individual protocol traffic – DTLS, ICE/STUN/TURN/SRTP/SRTTCP – is multiplexed on a single port as defined in IETF RFC 5761.

In WebRTC communications web browser play role of media subsystem, providing implementation of all audio and video codecs that can be used during WebRTC audio/video session. Thus, WebRTC communications have inherent dependency on media capabilities offered by each individual web browser vendor on each individual platform.

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## For Developers of:

- Enterprise PBX
- Contact Center
- IVR
- Audio Conferencing System
- Audio Endpoint
- Video Conferencing System
- Video Endpoint
- Audio Gateway
- Video Gateway
- Recording Servers

## Softil WebRTC Interconnect Framework Solution

Softil WebRTC Interconnect Framework Solution comprise SIP ToolKIt, A-RTP ToolKIt with SRTP add-on, and ICE/STUN/TURN ToolKit plus standard common core with two add-on modules to implement WebSocket and DTLS+Multiplexing as shown in the picture.
All combined, Softil WebRTC Interconnect Framework ensures support for the following mandatory WebRTC standards:

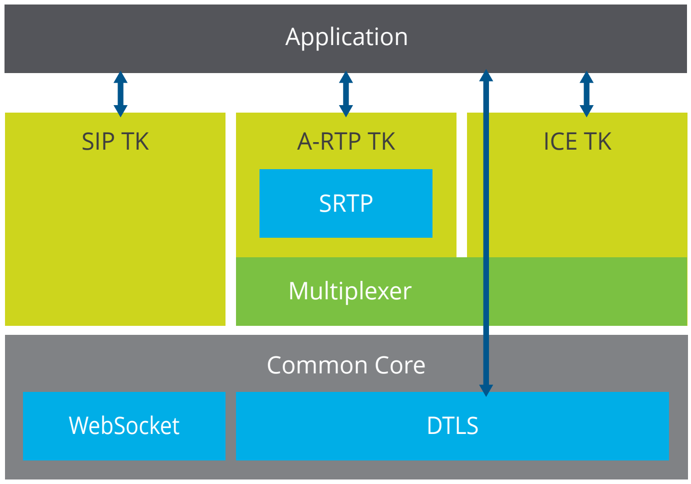
- WebSocket Transport (RFC 6455)
- DTLS Transport (RFC 6347)
- SIP over WebSocket (RFC 7118)
- RTP/SRTP with support for single port multiplexing (RFC 5761)
- ICE with support for single port multiplexing (RFC 5761)

## Three use cases for WebRTC Interconnect

Softil WebRTC Interconnect Framework components can be utilized in a number of ways, not requiring that all elements will be used at once.

### Standalone SIP over WebSocket

Web browsers support secure IP transport protocol, called WebSocket, which can be used by JavaScript applications for secure communication. RFC 7118 defines how WebSocket transport can be used in conjunction with SIP. Using SIP over WebSocket will allow standalone SIP applications, such as SIP Proxy, SIP Sever, and others, to connect to the SIP applications running in the browser implemented in Java Script, creating signaling interconnect solutions. To achieve this signaling interconnection, it is sufficient to use only SIP Toolkit with Common Core WebSocket transport add-on.




### WebRTC Media with Non-SIP Signaling

In certain use cases, HTTP or JSON might be sufficient to exchange all information needed for the media flow establishment. However, to exchange WebRTC media between web browser client and conventional UC endpoint, it is necessary to use multiplexed DTLS/ICE/STUN/TURN/SRTP media flow. IN such a scenario, only the DTLS/Multiplexing component of the WebRTC Interconnect Framework will be sufficient. In such a case A-RTP toolkit with SRTP add-on, ICE/STUN/TURN Toolkit and Common Core with DTLS/Multiplexing add-on will be sufficient to build a WebRTC media interconnect solution.

### Full WebRTC Interconnect

To achieve full interoperability of WebRTC web browser clients with enterprise grade UC endpoints, PBXs and contact centers, it is recommended to implement web browser client using JavaScript-based SIP implementation. SIP-based web browser client would allow users to experience a full set of enterprise UC telephony features, with ability to transfer and forward calls, exchange messages and access voice mail, from any place in the world without the need to install any software. To achieve interoperability with SIP-based web browser client, the full set of WebRTC Interconnect Framework components is required. SIP over WebSocket will allow signaling interconnection, were multiplexed DTLS/ICE/STUN/TURN/SRTP will be used for the media interconnection.


For more information, contact Softil at info@softil.com





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