WebRTC (Web Real Time Communications) is a developing open standard for enabling web browsers to support multimedia audio, video chat and other data applications. The WebRTC approach to online real-time communications aims to remove the hassles and issues associated with the current practice of installing vendor-specific software plug-ins or browser extensions, and promises to vastly improve the user experience.

Interoperability complexities and challenges, as well as costs, are a major obstacle for current vendor solutions. Typically, the end-user experience suffers, which has led to low adoption and use in stark contradiction to the web browser-based application market.

WebRTC shows promise for not only improving the user experience and adoption of real-time communications, but also for unleashing a new wave of video, voice, and data web applications.

This paper explores some of the technical components for WebRTC to inform and aid institutions in decision-making for the provision of collaboration services now and into the future.

What is WebRTC?

WebRTC is a standard emerging out of the World Wide Web Consortium (W3C) and the Internet Engineering Task Force (IETF). This standard holds significant promise as a replacement for the current reliance on proprietary browser extensions, troublesome plugin downloads and for potentially eliminating the need for licensed vendor-supplied soft clients.

WebRTC is about putting real-time communications (RTC) capabilities into a standard browser, with no download, no plugins and no Flash. The standard includes echo-cancellation, packet loss concealment, or all functionality expected in a media stack, prevalent in most IP PBX/UC systems or multimedia endpoints.

The WebRTC standard stipulates a framework for browser-to-browser multimedia communications, independent of platform or operating system. For the purposes of interoperability it's vitally important to agree on standardised codecs.

The W3C has currently agreed to use the following codecs:

- G.711 - audio
- G.722 - audio
- iLBC - audio
- iSAC - audio
- VP8 - video

Key features:

- Built in NAT Traversal.
- Always on secure audio and video.
- Advanced audio and video quality.
- Reliable session establishment.
- Adaptive to network conditions.
- Platform and device independence.
- Standardised developer environment, built around existing web programmability.
What are the potential benefits of WebRTC?

The Developer community stands to benefit from the WebRTC framework. Web developers do not require detailed knowledge of audio, video or IP telephony concepts. They can create multimedia-based applications using HTML5 and JavaScript application programming interfaces (APIs).

Third party vendors have been quick to develop and market solutions that extend the implementation of the WebRTC framework into existing unified communications or collaboration environments, including contact centres.

These gateways can be used to extend WebRTC into existing SIP, Jingle and even PSTN environments.

The following diagrams illustrate examples of how this integration may be achieved; for a SIP environment (Fig1), a PSTN environment (Fig2) and a browser-to-browser environment (Fig3).

Note in Fig 3, that media is expected to travel peer to peer and not necessarily via a media gateway. WebRTC includes NAT Traversal using ICE (RFC 5245 and associated extensions), including consent to connect. Signalling between browsers occurs over HTTP or web sockets.

The built-in NAT Traversal techniques, together with the platform independence, are vital to empower users to use BYOD-type endpoints and enable mobility from nearly any given environment, (public WiFi, home/consumer networks, enterprise networks, etc)

As of this time of writing the WebRTC standard is available in the following browsers:

- Google Chrome
- Mozilla Firefox
- Opera

The W3C is currently debating the clarification of video codec use, to include VP9 and perhaps H.264.
Summary

WebRTC in its current form is considered an emerging standard. However, it does have the potential to significantly disrupt the entire industry controlling IP devices that support collaboration, from mobile devices to desktop and software-only phones, collaboration portals and telepresence suites, as well as all the interconnect and meet me servers and media gateways in the middle.

Institutions should consider planning their investment in collaboration-based equipment in stages, taking into account potential disruption by WebRTC. API software developers may increase the range of personalised and integrated services in unimaginable ways. To be assured of a level of future proofing it may be worthwhile seeking advice about WebRTC, tracking standards, stress testing standards compliance claims and taking a multi-vendor capable stance, without detracting from the goal of achieving optimal user experience.

About the Author

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