Can WebRTC revolutionize real-time communication?

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The impact of WebRTC technology is being felt throughout the entire ICT industry, but its future is hard to predict, given the immature standards in place and the mad scramble taking place between vendors.

WebRTC: A game-changer?

What is web real-time communication (WebRTC)? To put it simply, WebRTC allows browser-based video chat over any compatible terminal. With WebRTC, a browser is equipped with A/V codec capabilities, and with the RTC service control logic moved to the cloud, a standard server-client communication model is created.

WebRTC was created by Internet service providers (Google primarily). It is open and free and provides real-time communication through simple Javascript APIs, without the need for plug-ins or apps. With its basis in developer-friendly HTML5, WebRTC can evolve smoothly to support tomorrow’s web-based applications.

WebRTC has been a hot topic since its inception, with some analysts considering it a game-changer. However, its impact on the ICT sector, especially web conferencing and enterprise, is hard to predict. WebRTC supports video chat and data sharing, which will certainly revolutionize web conferencing, and its browser origins will make audio and video communication on enterprise websites much easier.

Is the industry ready?

WebRTC is a hot topic, it still has a long way to go.

Standards set by different organizations

The IETF is formulating WebRTC’s overall architecture, which is scheduled for release by the end of 2014. The W3C is responsible for WebRTC terminal-side API definition, with API 1.0 already defined, and a baseline version scheduled for release in February 2015 (after numerous delays). The 3GPP has defined the enhanced P-CSCF-based WebRTC gateway architecture, which means that WebRTC users can access an IP multimedia subsystem (IMS)-based network to use corresponding communication services.

Competition between browsers

The three browser giants are not equally geared for WebRTC; in fact, only one is presently ready. Google is the pioneer & champion of WebRTC, and has support from both Firefox and Opera. Microsoft tried and failed to come up with its own CU-RTC-Web standards through W3C, and has seemingly given up on WebRTC-IE integration. Apple seems equally disinterested, apparently quite content with its own Facetime...
service. Thus far it has sat on the sidelines of WebRTC development. All-in-all, this lack of mainstream acceptance will indeed slow down the WebRTC revolution, as will prolonged conflict between video codecs. Mass deployment is not expected until at least 2015.

**Opportunity or challenge?**

For traditional telcos, WebRTC is both a challenge and an opportunity. It further fragments real-time communication services, with anyone who installs a dedicated server able to provide web-based real-time communication (in theory). WebRTC also helps OTT companies further penetrate both the Internet itself and telco revenues, but it also transforms real-time communication from a standalone service into an embedded function, which means that it can actually help telcos open their capabilities to expand service channels and user groups. If all web-based applications, such as online games, telemedicine, and distance education, could use telcos’ real-time communication capabilities, all web terminal users could potentially be telco subscribers.

The future of WebRTC is bright, but during implementation, telcos must consider the relationship between WebRTC and existing VoLTE and rich communication suite (RCS) services.

**Complementary WebRTC and VoLTE**

Some see WebRTC as a threat to VoLTE, but in truth the two are more complementary, with VoLTE more suited to the individual consumer market and WebRTC more vertical-oriented.

**WebRTC and RCS can help each other**

Telcos have had a hard time promoting their RCS services because they have largely been presented thus far in app form, thus requiring promotion, download, and compatibility with the user’s terminal, but WebRTC-enabled browsers have fewer issues.

**WebRTC and Huawei CaaS: Open capabilities**

WebRTC may be revolutionary, but it cannot revolutionize the ICT industry on its own. In fact, WebRTC and Huawei’s CaaS solution have overlap. WebRTC opens communication capabilities to the terminal side. As WebRTC only defines the processing of the media plane, while the signaling plane is de-standardized, IMS telcos can provide SIP- and IMS-based WebRTC solutions to develop their IMS users in the field. Thus, telcos can further open real-time communication capabilities to third-party web apps (online games, telemedicine, and distance education, etc.), allowing themselves to break into the Internet and enterprise/vertical fields and create more flexible business models (such as revenue sharing and pay-as-you-use).

Future CaaS solutions will be entirely open in terms of architecture, with upstream capabilities opened to expand telco business channels and downstream capabilities that include software development kits (SDKs) and WebRTC (which can increase the user base). SDKs focus on H2M and M2M while WebRTC enables development of web-based applications for the Internet industry, especially enterprise/verticals.

How can telcos prevail over OTT players in the WebRTC race? Huawei’s Telco-WebRTC solution enables carrier-grade network communication and global interconnection, while differentiating service control through QoS control on demand. It also enables inheritance of traditional telecom subscriber identifiers and supplementary services.

Other benefits include session border controller-based (SBC-based) WebRTC gateway support of all forms of telco and Internet access, enabling carrier-grade network reliability. This gateway can be easily deployed by adding or upgrading a single network element, minimizing the impact on current network reconstruction. Cloud-based deployment also makes this solution flexible and scalable. Pre-integrated industry application cases can greatly shorten the service time-to-market.

As a founding member of the IETF RTCWeb work team and as a member of W3C, Huawei has led industry development of WebRTC since 2011. At GSMA’s Mobile Asia Expo held in June 2013, Huawei worked with GSMA to promote RCS capability opening through WebRTC. At MWC2014, Huawei and China Mobile jointly demonstrated for the first time the sharing of audio/video/files between WebRTC users and LTE users, with Huawei also demonstrating how telco-WebRTC enables hospitals to improve service quality and user experience for telemedicine through HD video chat, on-demand QoS, one-number-link-you (ONLY) service, and global interconnection, enabling in-home patient monitoring and care. At the IMS Forum held in April 2014, Huawei proposed the WebRTC-based RCS capability opening gateway and demonstrated an application involving car insurance. Since then, many other manufacturers have launched similar solutions, making web-based RCS capability opening an industry consensus.

WebRTC can satisfy communication needs in the time of ICT integration. However, the neutrality of its technology puts telcos and IT/OTT players at the same starting line. Telcos have to come up with strategic plans and execute those plans if WebRTC is to be successfully embraced. At the service level, telcos must focus on capability opening and the enterprise/vertical markets. At the operational level, they must select typical use cases for promotion and design the proper business models. [2]