WEB-BASED COMMUNICATIONS

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Communication is an intensely social activity. Not only do human beings engage in it, but all manner of species depend on communicating intent and information with each other. However, human beings are unique in that we have evolved communications to travel over very long distances. Technology has helped us, from communicating our thoughts on a written medium (papyrus, first manufactured in Egypt as far back as the third millenium BC) to scattering our voice and the written word through ether and light fibers. This, in turn, has enabled us to build special networks optimized for communications. Currently, we are in the throes of a monumental migration in the means of communications, as the incumbent technology (circuit-switched networks that formed the backbone of the public switched telephone network) gives way to a new contender (packet-switched network, or the Internet) [1]. Today, the Internet is the dominant technology over which we communicate our voices, our words, and our images.

Allowing us to communicate so richly over the Internet is the World Wide Web, which has progressively shifted from a document-based paradigm to a more distributed and collaborative form of communication. In the early days of the web, communications took a *one-to-many* trajectory. The content provider engaged in mostly one-way communication by disseminating web pages composed of text, images, and short movie clips. In today's web, the lines between a content provider and a content consumer are not so well defined. Starkly characterizing this are web-based audio and video communications. The web-based communication technologies of today empower users to engage in a rich communication experience composed of audio, video, and ancillary services such as content sharing and ad hoc conferencing. The underlying magic that makes this possible is alternatively referred to as WebRTC or RTCWeb (where RTC stands for real-time communications). This technology is poised to seamlessly integrate the web and communications.

Overseeing this move are two standards bodies: the Internet Engineering Task Force (IETF) and the World Wide Web Consortium (W3C). The IETF is responsible for the architecture and requirements for selection and profiling of the *on-the-wire* protocols that will comprise half of the equation in web-based communications. The other half — programmability of the components through simple easy-to-use, and easy-to-understand application programming interfaces (APIs) — will be the domain of the W3C. Collectively, these bodies aim to provide a seamless communication experience over the most widely deployed interface of our time — the web browser.

We are pleased to present this Feature Topic of *IEEE Communications Magazine* to track the latest advances in web-based communications. The Feature Topic starts with the article entitled "Real-Time Communications for the Web" by Jennings *et al.*. This article serves as an entry point and a tutorial on the technology that will enable web-based communications. It is written by the very principals who are involved in the standardization of web-based communications in the IETF and W3C. The authors introduce the core components of WebRTC, provide a reference architecture, take a look at deployment issues that will challenge WebRTC, and pose some relevant open research questions that will need answers as the technology evolves and gets deployed for consumption.

The article by Singh *et al.*, "A Case for SIP in JavaScript," takes the position that while WebRTC is being standardized, developed, and deployed ubiquitously, there may be some space for an interim solution to manifest itself and fill the gap in allowing users to communicate. The authors posit that a JavaScript implementation of the Session Initiation Protocol (SIP [2]) could serve as one such stopgap measure. We urge readers to read the article and see if they concur with such a measure.

In "Data Channel Considerations for RTCWeb," Becke *et al.* focus on *non-media* data, that is, the data transmitted between participants of RTCWeb sessions not characterized as a Real-Time Transfer Protocol (RTP) media stream. Often, non-media data can include an instant (or text) message session, a file sharing session, or an online gaming session. These sessions typically have requirements that are different than those related to a media stream: whereas a media stream is always considered delay sensitive, a non-media data channel may be asked to transport delay-sensitive as well as non-delay-sensitive content. In RTCWeb, the transport protocol for non-media data is the Stream Control Transmission Protocol (SCTP). The authors study SCTP's efficacy as a transport for non-media data in RTCWeb.

Romano *et al.* present an article entitled "On the Seamless Interaction between WebRTC Browsers and SIP-Based Conferencing Systems," which analyzes the main issues that need to be addressed in order to let legacy SIP-based systems interoperate with WebRTC applications. The authors also provide a real-world interoperability example presenting the engineering approach they have used to integrate WebRTC clients into their own conferencing architecture, which uses standard protocols (XMPP, SIP, BFCP, etc.) to provide collaboration features.

And finally, Johnston *et al.* present "Taking on WebRTC in an Enterprise," which provides an in-depth and informed view of what it will take to allow WebRTC media to flow across enterprise boundaries. This is more challenging than it may look because traditional enterprises' policies protect the enterprise network by restrictive firewalls and prohibitive network border elements. This view has to be balanced against WebRTC, which aims to establish an end-to-end secure media path between two communicating browsers with little, if any, regard for an intermediate security enforcement point.

REFERENCES

[1] H. Schulzrinne, "Transitioning the PSTN to IP," IETF 86 Technical Plenary, Mar. 2013, archived at http://www.ietf.org/proceedings/86/slides/slides-86-iab-techplenary-4.pptx.

[2] J. Rosenberg *et al.*, "SIP: Session Initiation Protocol," IETF RFC 3261, June 2002, http://www.ietf.org/rfc/rfc3261.txt.

BIOGRAPHIES

SALVATORE LORETO [M'01, SM'09] (salvatore.loreto@ieee.org) has 15 years of experience in a variety of information and communication industries, and has been working in networking and telecommunications since 1999. Currently, he works as a research scientist in the MultiMedia Technology section branch, which is part of the NomadicLab, at Ericsson Research Finland. He has made contributions in Internet transport protocols (e.g., TCP, SCTP), signal protocols (e.g., SIP, XMPP), VoIP, IP-telephony convergence, conferencing over IP, 3GPP IP Multimedia Subsystem (IMS), HTTP, and web technologies. He is also an active contributor to the IETF, where he has coauthored several RFCs and Internet drafts. Currently he is serving within the IETF as Co-Chair of the SIP Overload Control (soc), Application Area (appsawg), and BiDirectional or Server-Initiated HTTP (HyBi) Working Groups. For the IEEE Communications Society, he serves as a Design and Implementation Series Co-Editor and an Associate Technical Editor for *IEEE Communications Magazine*. He received an M.S. degree in engineer computer science and a Ph.D. degree in computer networking from Napoli University in 1999 and 2006, respectively.

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