WebRTC and Universal Communications

1 ABSTRACT: A NEW ERA IN TELECOM

It took us the entire 20th century to evolve PSTN technology and build an infrastructure that made phone lines available in virtually every home in the developed world. It took only 25 years for cellular phones to reach near-universal coverage. As of 2014, more people have access to cell phones than to clean water.

As cell phones move toward universal coverage, use of the traditional PSTN is quickly declining. Today, more than 1/3 of the US households have dropped their landline phones (2012 NHIS Survey), relying on mobile phones and the Internet for all connectivity needs.

The first consumer Internet communication applications (like Skype) emerged in the early 2000s. Initially these tools were primarily used to make free phone calls when connecting with family from a computer, but they are increasingly used in business and mobile applications with additional rich media capabilities (voice, video, chat, screen share).

IP-based communications will eventually reach the point of universal coverage and, if patterns repeat, use of cellular phones for real-time communication (RTC) will decline. In order for that happen, however, IP-based solutions must overcome a number of challenges, including connection quality issues and the development of platform- and application-independent communications standards.
WebRTC, a browser-based standard for IP communications, could enable the creation of a new generation of peer-to-peer, non-proprietary communication capabilities and applications. This white paper will discuss both the possibilities and challenges of WebRTC adoption, the evolution of IP-based communications, and Daitan Group's experience in deploying WebRTC in real-world applications.

2  CHALLENGES WITH CURRENT IP-BASED RTC TECHNOLOGY

If IP-based tools are less expensive to consumers and offer more capability and flexibility compared to traditional cell/SMS communication, what is keeping them from taking over the RTC market? Why haven’t we completely switched to IP-based communications?

2.1  CAN YOU HEAR ME NOW?
The telecom industry has always focused on building infrastructure that could reliably deliver satisfactory service. IP-based tools, in contrast, primarily use the Internet, a best-effort network, which provides unreliable end-to-end bandwidth and latency.

So “Can you hear me?” became part of the call protocol when using Internet tools. That is not a technology challenge, as it would be possible to build a dedicated network to provide IP-based high-quality voice communications.

As consumers, we want real-time reliable communications that are available as part of our connectivity service, running over the same transport layer we use for everything else: the Internet.

We are at the threshold of accomplishing that. Improved codecs are making it possible to conduct reliable phone conversations even with less-than-ideal connections. Home Internet connections are on the path to provide sustained megabit-per-second bandwidth, which should enable multiple simultaneous voice/video connections.

2.2  UNIVERSAL ACCESS: PICK UP THE PHONE AND CALL ANYONE IN THE WORLD

When we use the phone, we expect to be able to call anyone in the world, independent of the device, carrier or platform used by the other person. Universal Coverage demands interoperability.

When we use Internet tools, however, we often need to be aware of which device or platform the other person is using. Some of our contacts might be on Skype, others on Facebook, and some might only be found in our Google address books. Until we can pick up our app and call anyone, we will still need a reliable phone when the call really matters.
Only the future will tell how those challenges will be overcome. But reliable quality of service and universal interoperability are two areas where the Internet generation can learn something from traditional telecom.

3 WHAT IS WebRTC AND WHY IS IT IMPORTANT?

WebRTC is an emerging standard that enables real-time communications (voice, text, video, and data). The real-time communications engine is embedded directly in the Web browser.

Because the browser is such a ubiquitous application, if WebRTC gains critical mass it will be available in any client device and will interoperate across platforms and operating systems without requiring the installation of proprietary applications.

WebRTC technology can also be embedded into mobile apps through a WebRTC Software Development Kit (SDK).

WebRTC enables application developers to incorporate rich, real-time communication capabilities (e.g., click-to-call buttons, chat rooms, video conferencing, screen share, etc.) to apps and Web pages with just a few lines of Java Script code. The developer doesn’t need telecommunications experience, since the engine embedded in the browser or SDK does all the heavy lifting.

For the first time, WebRTC provides a non-proprietary layer of functionality that developers can use without being encumbered by the limitations of one application or platform. We expect WebRTC adoption to generate a wave of creative uses of communications in the context of existing websites and business applications.

The existing model of communication based on proprietary applications shifted the control to mobile (IOS, Android) and social (Skype, Facebook, Google, etc.) platforms, with all of their inherent coverage limitations. (There are six times more cell phones than Facebook users.) WebRTC technology frees communications from platforms and creates the possibility of universal coverage for peer-to-peer communication, eventually replacing landline and cell phones.

Summarizing, WebRTC is relevant in the road to universal IP-based communications because:

- It potentially embeds a standard RTC engine in every computing platform using the browser as a vehicle.
- It creates the potential for innovation by making the integration of applications with communications much simpler.
- It shifts control from platform vendors to application developers.
4 CURRENT WEBRTC DEPLOYMENTS

There are already demonstration projects underway that allow users to make real calls from cloud-based WebRTC implementations.

4.1 WEBRTC AND IMS IN THE CLOUD (FOR TELECOM OPERATORS)
If IP-based communications are to be unavoidable, then telecom operators would like those communications to be independent of mobile and social platforms (Skype™, Google®, IOS, etc.). WebRTC offers an alternative that has potential for universal access integrated with traditional telephony in an IMS system.

Daitan leveraged its expertise in communications systems to integrate WebRTC applications to Project Clearwater and create a deployment in the cloud that can demonstrate end-to-end connections between any combination of WebRTC and traditional telephony end-points using an SIP-based signaling plane in an IMS architecture. Clearwater can be used as a stand-alone solution for mass-market VoIP services, or deployed as an IMS core in conjunction with telephony application servers or other elements.

This demo shows how WebRTC can be seamlessly integrated with the existing telephony infrastructure, and how it can support the transition from using a mix of PSTN and cell phones to real-time, IP-based communications.

4.2 WEBRTC AND FREESWITCH IN THE CLOUD (FOR COMMUNICATION SOLUTION PROVIDERS)
Solution providers offering applications such as video conferencing or customer service applications have always struggled with interoperability and support of end devices (traditional phones, soft phones, VoIP clients, etc.) using proprietary technologies. WebRTC promises to establish a standard client that will be available in every platform.

Daitan leveraged its expertise in communications systems to integrate WebRTC applications to FreeSWITCH software and create a deployment in the cloud that can demonstrate end-to-end connections between any combination of WebRTC and other VoIP endpoints using an SIP-based signaling plane in a cloud architecture. FreeSWITCH is a free, open-source software for creating voice and messaging products.

This demo shows the possibility of seamlessly integrating WebRTC to the existing commercial conferencing infrastructure and supporting the transition from proprietary endpoints to Internet technologies.

To access the demonstration and make real calls from WebRTC, please visit http://daitangroup.com/webrtc.

5 WEBRTC CHALLENGES

The road to WebRTC ubiquity will not be easy. It took 100 years for traditional telephony to reach universal access; it will take several more years for IP-based communications to get there.
5.1 **Browser Support**
As of today, Google Chrome™, Mozilla Firefox® and the Opera™ browsers have built-in support for WebRTC. That represents more than half of the browser installed base but does not include Apple Safari and Microsoft Internet Explorer.

Apple has not stated a position, but given the leveling effect of WebRTC the company is unlikely to join unless it is forced to do so. Microsoft has made moves in this direction but has a clear conflict of interest now that it owns Skype. Building a critical mass of applications will be a major challenge without early support from these two important platform/browser players.

5.2 **Video Codec Discussions**
The WebRTC standard will define the mandatory codecs to be supported by compliant implementations. The two main contenders are VP8 and H.264. As of today, there is still heated debate about these two options.

Google has acquired the technology behind the VP8 codec, released its implementation in open source form and promised never to demand royalty payments. H.264 is an existing standard that is available everywhere (including hardware-based acceleration in many platforms). The challenge with H.264 is that it is not royalty-free.

For most stand-alone applications, which video codec is ultimately supported is not very relevant. For business solutions that require integration with existing systems, the codec choice will affect the need for transcoding and royalty payments.

5.3 **Common Signaling for Universal Access**
Will WebRTC technology ease the integration of real-time communications to Web applications? Or will it take us one step forward on the road to universal IP-based communications?

The answer is the former. WebRTC was originally defined as a technology to facilitate the integration of communications to Web applications. Because it was not intended to serve as tool for universal communication, it does not include a signaling layer for establishment of ad hoc calls. For ad hoc calling, WebRTC still requires the intermediation of a Web server for call setup.

6 **Conclusion: A Critical Step Toward Universal Real-Time Communications**
The WebRTC standard provides for peer-to-peer media connections between browsers at the client machines once the connection is established. If two or more users access a common website, it is possible to connect them for direct communication. An example would be a user accessing a customer service site and being directly connected to an agent without leaving the context of the website.
But WebRTC does not provide a signaling/control plane for one user to find another and establish a peer-to-peer call. Someone still must provide presence detection, physical and logical location, directory services, establishment of calls, etc.

In a typical WebRTC connection, media flows directly from browser to browser. That is fine for two-party calls, but does not address the need for more sophisticated applications (such as multi-party conferencing) and interoperability with regular phones.

With WebRTC integrated in an SIP environment, it is possible for WebRTC clients to leverage a signaling scheme that already supports universal coverage and interoperability. Daitan Group and its partners have already demonstrated the viability of this model through their work on the Clearwater and FreeSWITCH initiatives.

Here lies an opportunity for telecom operators to provide something the WebRTC standards have not yet addressed. To integrate with business solutions using other VoIP technologies, WebRTC requires an MCU to do transcoding and call bridging.

WebRTC could prove to be a critical step toward universal coverage of IP-based communications, but it is not the complete solution. It was intended as a way to facilitate the integration of communications and Web applications, but if it gains critical mass it will create a wave of innovation and trigger other changes that will accelerate the decline of traditional and cellular telephony.
ABOUT DAITAN GROUP

Daitan provides highly reliable software development services. We partner with technology vendors to help them develop their next software solution in Telecom, Unified Communications and Cloud/Web Solutions. We pioneered WebRTC implementation and some of our customers were first in the market with their business solutions supporting the technology.

To learn more about Daitan, please visit us at http://daitangroup.com