WebRTC and IMS in the Cloud
A NEW ERA IN REAL-TIME COMMUNICATIONS

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1 A NEW ERA IN REAL-TIME COMMUNICATIONS

It took us the entire 20th century to evolve PSTN technology and build an infrastructure that made a phone line available in virtually every home in the developed world.

In the mid-1980, we started building the first cellular networks, and it took only 25 years before cell phones reached near-universal coverage. As of 2013, there are about 6 billion cell phones in a world with 7 billion people. There are more people in the planet with access to cell phones than to clean water.

But as cell phones emerged towards universal coverage, use of the traditional PSTN is declining quickly. Today, 1/3 of the US households have dropped their phone landlines (source: CDC, 2012 NHIS Survey), and rely on mobile phones and the Internet for all connectivity needs.

In the early 2000’s we saw the first consumer Internet communication applications (like Skype). Originally these tools were primarily used to make free phone calls when connecting with family from a computer, but we now see increasing use in business and mobile applications, with additional rich media capabilities (voice, video, chat, screen share).

So, over the next few years, we can anticipate that IP-based communications will eventually reach that point of universal coverage and, if patterns repeat, use of cellular phones for real-time communication (RTC) will decline after that.
2 So, What is Missing for IP-based RTC to Take off?

If IP-based tools are less expensive to consumers and offer more capability and flexibility compared to traditional cell/SMS communication, what is keeping it from taking over the RTC market?

There are still two major obstacles IP-based technology has to overcome:

• Can you hear me? – The telecom industry has always focused on building infrastructures that could reliably deliver quality of service. IP-based tools primarily use the Internet, a best-effort network, providing unreliable end-to-end bandwidth and latency. So “can you hear me?” became part of the call protocol when using Internet tools. Perhaps we need quality of service capabilities over the Internet. Or, most likely, the solution might be just to make more bandwidth available to everyone.

• Pick up the phone and call anyone in the world – When I use the phone, I expect to be able to call anyone in the world, independent on the device, carrier or platform the other person uses. Universal Coverage demands interoperability. When we use Internet tools today, we often need to be aware what the other person is using. Some of my contacts might be on Skype and others are Facebook friends, while some can be found only in my Google address book. Until I can pick up my app and call anyone, we still will need a reliable phone for when the call really matters.

Only the future can tell how those challenges will be overcome. But reliable quality of service and universal interoperability are two subjects where the Internet generation can learn something from traditional telecom wizards.

3 What is WebRTC and Why is it Important?

WebRTC is an emerging standard to enable real-time communications (voice, text, video, 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 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 webpages with just a few lines of Javascript code. The developer doesn’t need telecommunications experience, since all the heavy lifting is done by the engine embedded in the browser.

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. So, we expect the first phase of WebRTC adoption to generate a wave of creative uses of communications in context of existing websites and business applications.

The existing model of communication based on proprietary applications shifted the control to the mobile (IOS, Android) and social platforms (Skype, Facebook, Google, etc), with all their inherent coverage limitations (there are 6 times more cell phones than Facebook users in the world today).
WebRTC frees communications from platforms and create the possibility of universal coverage for peer-to-peer communication, eventually replacing traditional and cell phones.

4 WHAT IS IMS AND WHY IS IT IMPORTANT?

IMS stands for “Internet Multimedia Subsystem” and is an architecture adopted by telecommunication operators to provide IP-based telecom services natively in an IP network environment (versus the circuit-switched network).

Most operators are deploying IMS frameworks using proprietary systems in their dedicated network infrastructure. We would argue that it is important for operators to embrace the cloud the same way as Internet companies do. The notion that to provide quality of service we need to use a dedicated infrastructure we can control is probably not viable in the long term. To compete in the market, telecom operators must be able to leverage the same technologies and economy of scale enjoyed by Internet players.

As an example of IMS system that is optimized for deployment in the cloud, we present Clearwater, an open-source project that follows IMS architectural principles and supports all of the key standardized interfaces expected of an IMS core network.

Clearwater provides SIP-based call control for real-time communications in an IP network. Clearwater can support VoIP services relying on its built-in set of basic calling features and subscriber database, or
work as an IMS core in conjunction with other existing elements such as Telephony Application Servers and a Home Subscriber Server.

As an IMS core, Clearwater incorporates a Proxy CSCF (Call Session Control Function), Interrogating CSCF and Serving CSCF, together with Breakout Gateway Control Function.

Interestingly, Clearwater includes a WebRTC gateway, and natively supports interworking between WebRTC clients and standard SIP-based clients, using SIP signaling over WebSockets.

The integration of WebRTC within an IMS framework make it possible to integrate browser-based communication with traditional telephony. Here are some examples of use cases:

- Access voice mail and other phone features from any browser
- Use of WebRTC to extend the reach of traditional telephony – For example, when travelling abroad, we can use WebRTC to connect into the phone system and make local calls
- Integrate WebRTC with existing call plans

5 USING WebRTC TO PROVIDE UNIVERSAL REAL-TIME COMMUNICATIONS

The WebRTC standard provides for peer-to-peer media connections between browsers at the clients machines once the connection is established. So, if two or more users access a common website, it is possible to connect them ad hoc for direct communication. An example would be a user accessing a customer service site and getting directly connected to an agent without having to leave the context of the website.

But WebRTC does not provide a signaling/control plane for one user to find another user and establish a peer-to-peer call. So there is a need for someone to provide services like presence detection, physical and logical location, directory services, establishment of calls, etc.

In a typical WebRTC connection, media flows directly from browser to browser which 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 (landlines and cell phones will exist for a long time).

With WebRTC integrated in a SIP environment, it becomes possible for WebRTC clients to leverage a signaling scheme that already supports universal coverage and interoperability. Here lies the opportunity for telecom operators to provide something that the WebRTC standards have not yet addressed.
**WEBRTC AND IMS IN THE CLOUD INTEGRATION DEMO**

Daitan leveraged its expertise in communications systems to integrate WebRTC applications to the Clearwater project 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 a SIP-based signaling plane in an IMS architecture. This demo shows the possibility to seamlessly integrate WebRTC to the existing telephony infrastructure and support the transition from a world using a mix of PSTN and cell phones to communicate worldwide in real time to a world that embraces Internet technologies.

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

**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, Mobile and Cloud-based Communications. We pioneered WebRTC implementation and some of our customers were first in the market with their business solutions supporting the technology.

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