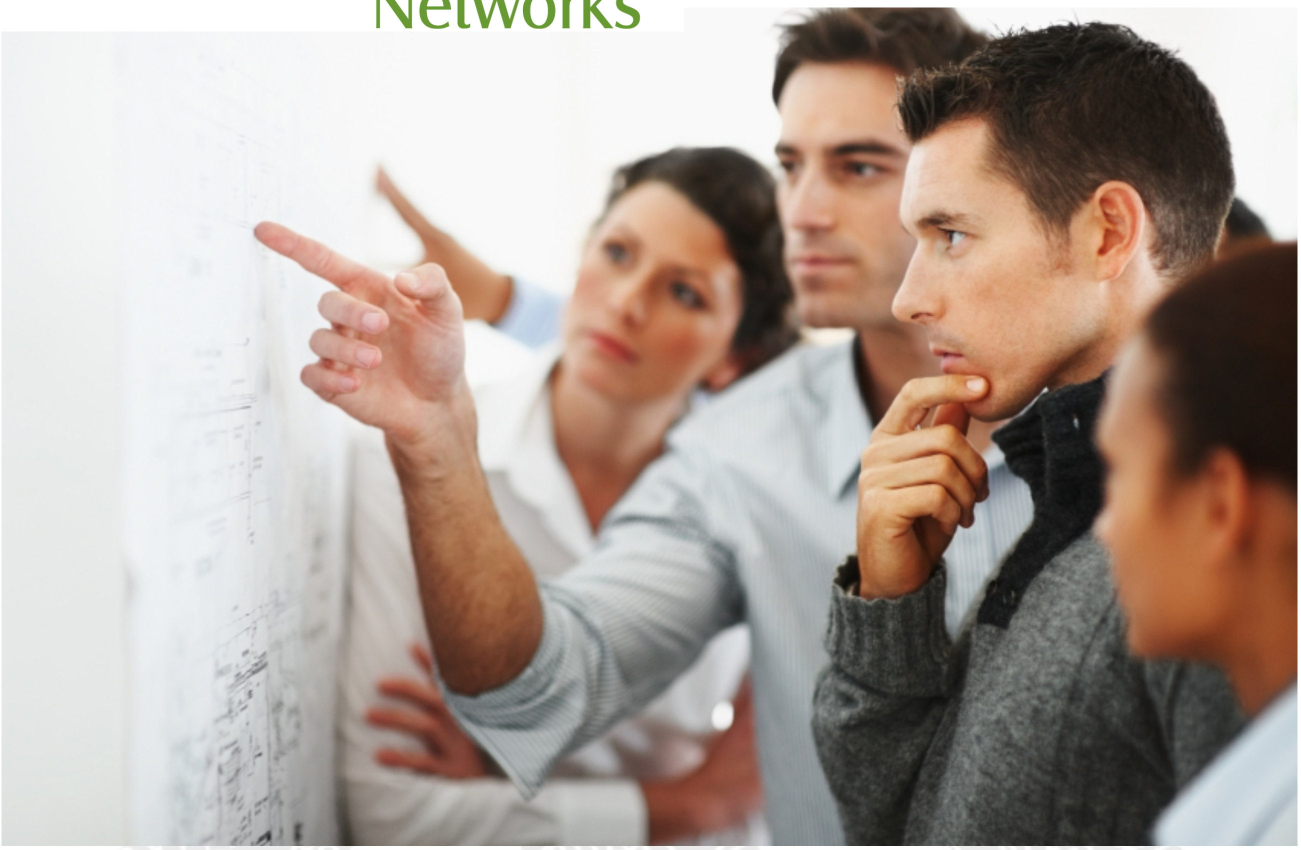
Whitepaper



Service Providers and WebRTC

New Product Opportunities

Over-the-Top (OTT) services are those that deliver communications features to customers but are apps running on the data network rather than native to the carrier. Voice over IP was one of the first, followed by text messaging which continues growing briskly. Now voice, video, presence and other features are appearing as stand-alone apps or integrated in social media. Customers use them, but the walled gardens set up by all these apps have resulted in a fragmented market with apps coming and going in fads. Few of them interconnect and none have the reliability of a telecom service provider.

Now, cable MSOs, wireless and wire line service providers are looking at ways to enrich the customer's experience and break down those walled gardens. One major technology getting traction in this space is WebRTC.

What is WebRTC?

WebRTC (Web Real Time Communication) is a free, open source project developed by Google that enables web browsers with Real-Time Communications capabilities via JavaScript APIs.

A packaged version of the Google Chrome browser called the "Chrome Embedded Framework" or CEF, can be used to create native desktop clients or mobile apps. Google-purchased technologies that are included in WebRTC including voice and video processing and telecom transport. This opens up a vast array of options to develop new, interesting services with embedded real-time communication and without license costs.

What WebRTC Can Do:

- Audio calls and multi-party conferencing:
  WebRTC includes both the common G.711 codec, and Opus, which is variable-rate. Opus is unique in that it samples and adjust bandwidth from very low to HD voice. It uses NetEQ for voice plus an echo canceller/noise reduction feature.

- Video calls and multi-party conferencing:
  Currently, WebRTC includes the VP8 codec and will soon add VP9. Combined with a video jitter buffer and image enhancing features, WebRTC can sample and adapt the resolution of a video connection to available network bandwidth. Most video conferencing products run the H.264 standard which requires a license. In late 2013, Cisco purchased a license that allows other products to use H.264 in WebRTC without extra costs, just by including their open source library. Interconnection with existing video conferencing equipment is now practical.

- Personal broadcasting:
  With the video capabilities in place, WebRTC enables multiple connection types including 1-on-1 or group video calls. Furthermore, it can provide a 1-to-many call starting to be referred to as "personal broadcasting." Imagine a child playing soccer while one parent is present but the other parent who is travelling and grandparents out of state could all be watching in real time.

- Instant Messaging
  SMS & MMS services to a landline or conversation threads that follow users across platforms are just two of the many options.

- Data transfer
  WebRTC data transfer abilities open up a whole new world of possibilities. Customers can use WebRTC to share a desktop or document, exchange files, record and replay videos or store and retrieve files at a later time. Furthermore, service providers could expose secure APIs for other uses such as collecting vital signs for an elderly patient at home, alarm system data, etc.

- Converged address book
  Tying a customer's service to their phone number is an obvious and easy way to move the customer from one communication platform to another, but why stop there? With a converged network address book that contains their registered devices, many can be configured, allowing the customer to choose, forward or turn off any of them.

- Presence
  Customers in the address book can be tied to other identifiers in addition to phone numbers. Then, similar to instant messaging services in use today, a presence indicator for each sub can be made available to those they choose to include.

Where it runs:

WebRTC has been incorporated into Google Chrome, Firefox, Opera and a few niche browsers. In early 2013, Google split from the WebKit standard governed by Apple and created a “fork” or variant called Blink. This is part of their Chromium project and is where they incorporated WebRTC into their browser. Firefox and Opera have done the same with more browsers following suit. However, WebRTC does NOT run in Microsoft’s Internet Explorer or Apple’s Safari browser. There is a wide expectation that these two big players will incorporate WebRTC support soon, but neither has announced a specific plan. It is likely that both will support WebRTC by the time service providers launch new services to the public.

CEF: The Chrome Embedded Framework is a stand-alone library for Chromium that can be included in other software to enable WebRTC features. Imagine native apps for the desktop, smartphone, tablet, TV, gaming console, car, refrigerator. The list is endless.

Session Management

Peer-to-Peer:

The basic method for WebRTC apps to communicate is peer-to-peer which is simple and quick. However, it’s common for public wifi networks and many corporate networks to block peer-to-peer traffic due to the security risk.

Server-based:

A more secure way to offer the service is to run the traffic through a gateway server. This is key for managing accounting and usage, but also has additional advantages. In this model, router or mux connections can be set up for multi-party calling and personal broadcasting. The number of total connections is vastly reduced as a result.

Signaling

WebRTC does not include the signaling technology that is used to set up, maintain and terminate call connections. In fact, it’s possible to establish basic implementations with no signaling at all. However, to incorporate WebRTC into the telecom network, it's essential, and there are a few ways it can be done.

WebRTC can be connected to SIP devices using either a commercial WebRTC to SIP gateway or by incorporating support for SIP right into the application’s business layer. Multiple vendors are now offering these devices. The gateway manages transcoding and encryption.

Authentication

Like signaling, WebRTC also does not have integrated security. This is intentional since as a browser technology, the two most common frameworks, OAuth and LDAP, are already robust and widely deployed. Furthermore, it is possible to integrate with existing telecom security such as using one of the WebRTC gateway devices mentioned earlier.

Business Opportunity for Service Providers

With all the features that are browser-based and peer-to-peer, the assumption may be that this will just be a new platform for OTT apps and there's no specific revenue opportunity for service providers.

There is nothing preventing new over-the-top apps from using WebRTC. At the same time, there are some compelling use cases that make WebRTC a valuable tool for service providers to gain customers and grow revenue.

- <u>Consumer client tied to phone service</u>:
  Subscribers have a native or browser-based client available on their desktop, laptop, mobile device or TV (or car or refrigerator). They can access it on the go or at work. They can call other subscribers who have the client or people off-net by connecting through the PSTN. Thanks to WebRTC basic browser support, the called party can simply be provided a web URL. Putting the brand in more places improves both retention and satisfaction. At the same time, it adds valuable new services with unified video, voice, messaging and other features.

- <u>Small/Medium business market</u>:
  Microsoft Lync and Cisco unified communications are complex and expensive for small businesses. Bringing instant, live video and messaging to an existing business's voice service is a real and measurable productivity boost. PBX service providers are beginning to incorporate WebRTC support and opening up advanced call features for find-me, assistant, and other business tools.

- <u>Customer contact center</u>:
  Amazon created positive buzz with their launch of "Mayday", a video customer service feature on their Kindle products which is powered by WebRTC. The opportunity to make stronger support and telesales connections with customers is tremendous.

- <u>Web services for app developers</u>:
  Expose secure APIs for application developers to use to generate new software products for subscribers that take advantage of the fact this service is connected to the PSTN and can exchange data. e-health, e-learning, remote monitoring and more uses not-yet-imagined will be developed by entrepreneurs.

Summary

WebRTC is gaining significant traction with service providers experimenting and creating new products. Several telecom equipment vendors now sell WebRTC gateway devices, including Oracle. Possibilities are endless, but one point is clear. WebRTC can be leveraged to deliver an array of value-added services from service providers that can increase revenue, improve retention and create a distinctive product in the marketplace. The challenge is to incorporate the technology into the network so that it is seamless for customers and can be effectively provisioned and managed.

About Mersoft Networks

We are the product development and system integration experts. Mersoft brings innovative ideas, compelling products and skilled professional services to cable MSOs, wireless and wire line service providers who demand innovation, agility and quality. We help companies achieve goals from reducing costs to improving efficiency to delivering profitable new services.

Mersoft Networks developed **Mersoft move™** which harnesses the power of WebRTC for amazing voice, messaging and video chat for business and consumer subscribers.

Contact Mersoft Networks for a demo and to learn more about WebRTC and Mersoft move™.

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