Briefing Paper

WebRTC ASIA FORUMS

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What is WebRTC?

WebRTC stands for Real Time Communications over the Web – an emerging part of the HTML5 standard allowing interactive voice, video & data directly through a web-browser. It supports audio-visual and data communications such as file sharing and also screen-sharing on a peering basis. "It enables websites to become hubs for managing large amounts of data without actually having to store or provide transmission bandwidth for that information distribution."[1]

Technically, WebRTC is actually a combination of network protocols for voice/video "media engines" & IP network/firewall traversal, plus a set of simple Javascript APIs which can be used by developers or web designers to get access to those underlying "nuts and bolts". These two aspects are managed by IETF and W3C respectively.

Currently WebRTC is supported by

Desktop	○ Google Chrome 23+ ○ Mozilla Firefox 22+ ○ Opera 18+
Android	○ Google Chrome 28 (Enabled by default since 29) ○ Mozilla Firefox 24+ ○ Opera Mobile 12+
Google Chrome OS	

Notably absent for the moment are Microsoft's IE and Apple Safari, but for business rather than technology reasons – they may start to support WebRTC later this year. With WebRTC there is no need to download plug-ins or standalone applications such as Skype, Yahoo Messenger, Facebook, WeChat and others, but WebRTC can also be built into any apps and will as often as not be accessed in that way. An emerging trend is the relevance of "non-browser" forms of WebRTC, often provided via 3rd-party SDKs for mobile app developers, together with cloud platforms that automate certain of the more technical aspects.

WebRTC demo

WebRTC can enable the creation of completely open communications systems or adapted to work only with specific IDs or passwords for security. It has use-cases in enterprise communications, telecoms, the consumer web and beyond. Once the technology is embedded into a web page, which any web designer can do, visitors to the website can click on the icon and achieve real-time AV-D communications after allowing the browser to access camera and microphone. WebRTC can therefore be used for external

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[1] http://www.webrtcworld.com/topics/webrtc-world/articles/364011-yahoo-acquisition-peercdn-expands-webrtc-beyond-google.htm

communications, such as customers making enquiries and contact with suppliers, business partners and regulatory agencies; or for internal communications as part of an enterprise or agency's unified communications network. The following Mozilla video on YouTube provides an easy-to-follow introductory demo.

Watch: http://www.youtube.com/watch?v=S6-rAv6bU8Q

Beyond the enterprise, WebRTC can enhance consumer social networks, enable voice or video to be embedded in games or entertainment, enable video-consultations of various types, or even help create completely "non-phonecall" use cases for healthcare, finance or many other vertical sectors. Even M2M and consumer electronics can embed realtime communications.

WebRTC, the Cloud and the Challenge to Telecom Service Providers

Because WebRTC applications or services can work in the cloud (or even peer-to-peer), they can be designed to be network-independent and accessible through any type of connected device, for example a smartphone.

What Can WebRTC Be Used For?

Just about everything. On a peer-to-peer basis it can be used by the public to call and video chat or conference with each other from any device, such as a mobile phone, to any other device, such as a tablet or PC. For the enterprise sector is can be used to set up real-time conference conversations and exchange data files. For Call Centres it allows staff to know instantly the nature of the inquiry and of the inquirer, adding "context" to standalone communications. For telecom companies it offers a means to build inclusive and unified service platforms for customers – or create entirely new & innovative "digital services" outside traditional telephony, through the web & apps. For websites it improves the stickiness of visitors.[2] For web developers it offers opportunities to provide customized products and services to clients. As a communications technology it cuts horizontally across all sectors of the economy. The following video from Cisco illustrates many of the applications of WebRTC technology.

[2] See Temasys video on YouTube at http://www.youtube.com/watch?v=dx_C4UA5tIg

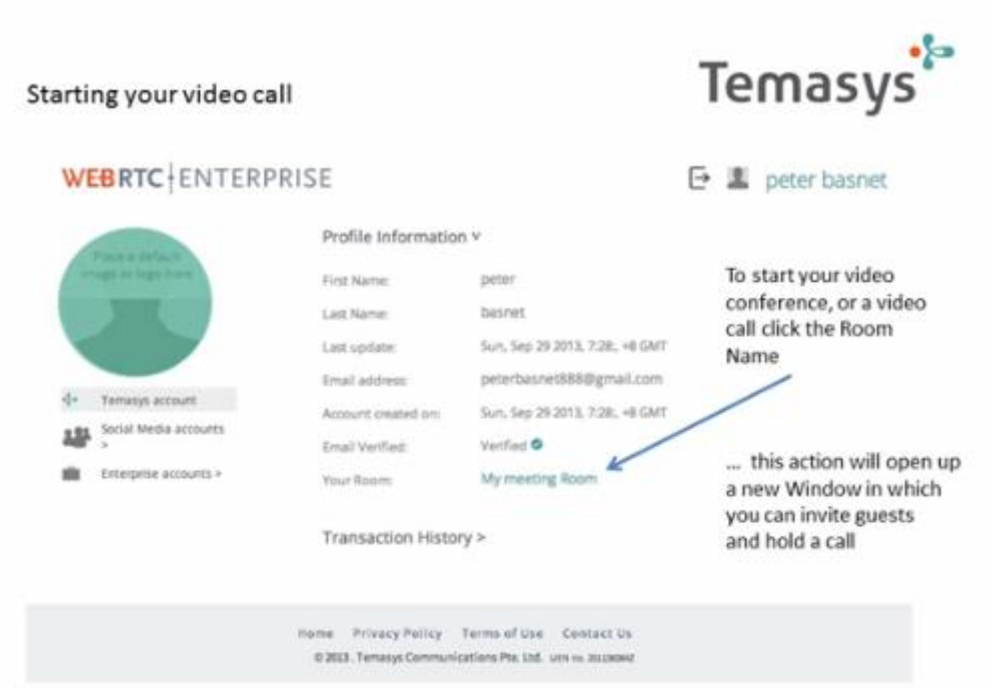
Watch: http://www.youtube.com/watch?v=Yf3eNciKddc

Mobile Cellular Operators

WebRTC could become a major challenge to mobile network operators (MNOs) which understandably want to offer AV-D services over their own networks and offer interconnection to the networks of other MNOs which are part of their peering "club". The slow-moving GSMA "Joyn" initiative is an example of this using the GSMA-supported Rich Communications Services (RCS) platform,[3] and the business models of coming 4G Voice-over-LTE (VoLTE) are being built to a large extent upon customers relying on this service for future AV-D communications, fast connections to social networking sites, downloads and uploads. This is an extension of the traditional "interoperable" and network-integrated telecom model, bringing ordinary phone calls and messaging to newer IP networks.

WebRTC continues and extends the existing risks from 3rd-party VoIP/video and messaging apps – the so-called "OTT" services like Skype & Whatsapp. It represents an alternative path to AV-D for any smartphone user if their web-browser is compatible, which is why, for the immediate future, iPhones and phones using Windows will not support WebRTC except where embedded in apps through 3rd-party APIs & SDKs. It should also be noted that although feature phones may not be able to offer video reception, WebRTC can still use a peer connection,[4] to provide audio services.[5] User ID (required for peers to find each other) will vary according to the medium being used. It could be an SNS identify on Facebook or Twitter, a visit to a virtual 'meeting room', a telephone number on a mobile phone, etc. See the video on YouTube from Temasys.

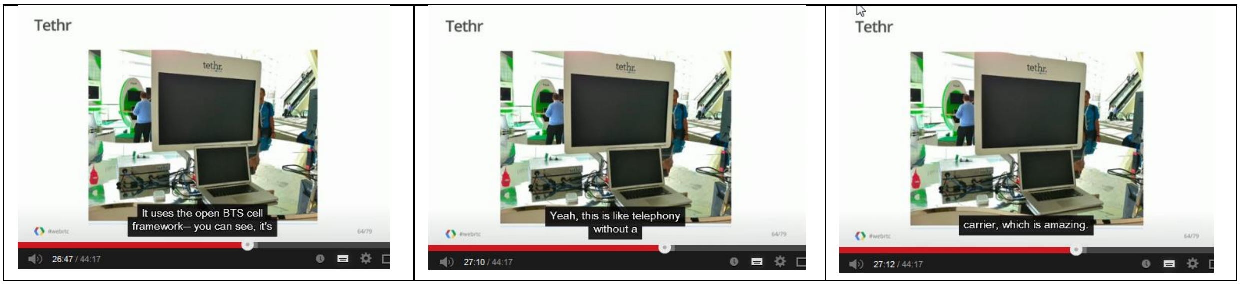
[3] http://www.gsma.com/futurecommunications/rcs/
[4] Introduced to provide Android phones access to radio stations, see https://play.google.com/store/apps/details?id=com.pure.purelounge&hl=en
[5] See a demo of this at http://www.youtube.com/watch?v=p2HzZkd2A40



Watch: http://www.youtube.com/watch?v=JwawofRPqz0

And from Google

A demo of basic telephony over WebRTC is given on YouTube by Google



Source: http://www.youtube.com/watch?v=p2HzZkd2A40

Perhaps the threat to MNOs will be even greater at the lower-end of the mobile market as low-cost Android smartphones begin replacing cheaper feature phones and as broadband 3G networks proliferate in developing economies. Just as paging and then SMS and later on chat services appealed to young and to low-income users, for example, in China, India, Indonesia, Malaysia, Thailand and the Philippines, so it could be that WebRTC services on low-end smartphones will be the device and platform- of-choice in these markets in the coming years. The big commercial issue here is whether customers will buy-into a AV-data package offered by MNOs or simply pay for the data connection and rely increasingly upon cloud-based and public WiFi-based services, as already occurs widely for messaging services like LINE and Whatsapp.

Fixed Line Operators

A similar challenge faces fixed line operators which are exploring ways of integrating their own versions of WebRTC into unified platforms that offer a wide range of digital services that can include web-based services of their own, home hubs that may soon interconnect household items in an Internet-of-things, and IPTV. If WebRTC, for example, is embedded into connected TVs it could enable users to watch a programme on one quarter screen, see the person they are talking to on another quarter screen, see a screen shot or a power-point or a photograph on yet another quarter screen, and see another person in another conversation on the fourth quarter screen. This could mean multi-tasking with a vengeance, but as an illustration of the possibilities it is worth imagining.

Given that the telecoms world and the Internet world have been travelling as uneasy companions for well over a decade, sometimes in parallel as alternative paths, sometimes in tandem, fixed line operators have been pondering the adoption of IMS for some years. Carriers must now consider how WebRTC, like other web-based OTT (Over-the-Top) technologies, can be incorporated into the carrier's service options. Or, take note of Dean Bubley's view that "There is not a single service that IMS can do that other platforms cannot - typically with greater flexibility, lower cost and much faster time-to-market."[6]

As part of a growing family of web-based OTT technologies, WebRTC is clearly part of a threat to the revenues on the services-side of the traditional telecom operator's business. In countries where the telco is a state-owned enterprise, OTT services are often strictly limited, as in China, or banned outright, such as in Vietnam, but this is swimming against the technological tide of history, at best a rearguard action to buy time. In the larger picture, OTT services, and WebRTC itself, represent an unavoidable progression from IP in the 1990s to web-based communications that threaten to detach themselves entirely from the underlying telecoms infrastructure.

Regulation

On the regulatory side one issue will be whether operators and service providers can block or throttle OTT services (the "net neutrality" issue) and even if they can – they will either appeal to "network management" and QoS issues or, more directly, to revenue and investment issues – will it be sufficient to prevent the by-pass of their networks? (See 'Networks' below). The answer most telecom companies would prefer is to introduce 'calling party pays' *and* 'receiving party pays' but in a broadband context of sending and receiving parties. Charging could be by bit-rate, file-size, bandwidth with capacity caps on flat-rate fees, etc.

Carriers will call this service differentiation, opponents will call it service discrimination, invoking concerns over "net neutrality." But the issue could become moot after a time *if* WebRTC is adopted *en masse*, for example by the business sector as a web-based

[6] http://disruptivewireless.blogspot.sg/2013/12/the-beginning-of-end-for-ims-webrtc-is.html

communications tool. Vendors of devices and carriers alike will have to bend to the pressure of market demand. What could delay or stifle widespread WebRTC adoption would be a lack of interconnectivity between different web-based systems. This is somewhat ironic given that the Internet is the ultimate any-to-any connectivity technology. See the standards discussion below.

There may also be regulatory concerns around the heavily-encrypted nature of WebRTC, and the implications of various types of interconnect/quality/competition rules on new communications modes that lie outside traditional telephony concepts.

Networks

Most devices (PCs, tablets, smartphones, connected TVs, etc.) are connected to proprietary networks of one kind or another, either public or private, but WebRTC using the cloud requires no proprietary network to function. What this means for the future is that if telecom companies lose their monopoly over networks, WebRTC can flourish using many alternative routes.

Alternatives have been around for quite some time, but in recent years their number has grown substantially. For example, public access to WiFi networks is becoming more common globally, and web-based companies are investing in networks to link their globally-distributed data centres. [7] Access to the Internet is now ubiquitous, either via traditional telco-owned broadband, or alternative access mechanisms

Interconnectivity and Standards

For any web-based technology to have maximum effect, and therefore take-up, a few basic things are necessary, and these include protocol standards and agreement on the use of the codec to be used to compress traffic, especially streamed video. The bottom line is interconnectivity through interoperability. A YouTube video from Oracle discusses this problem.

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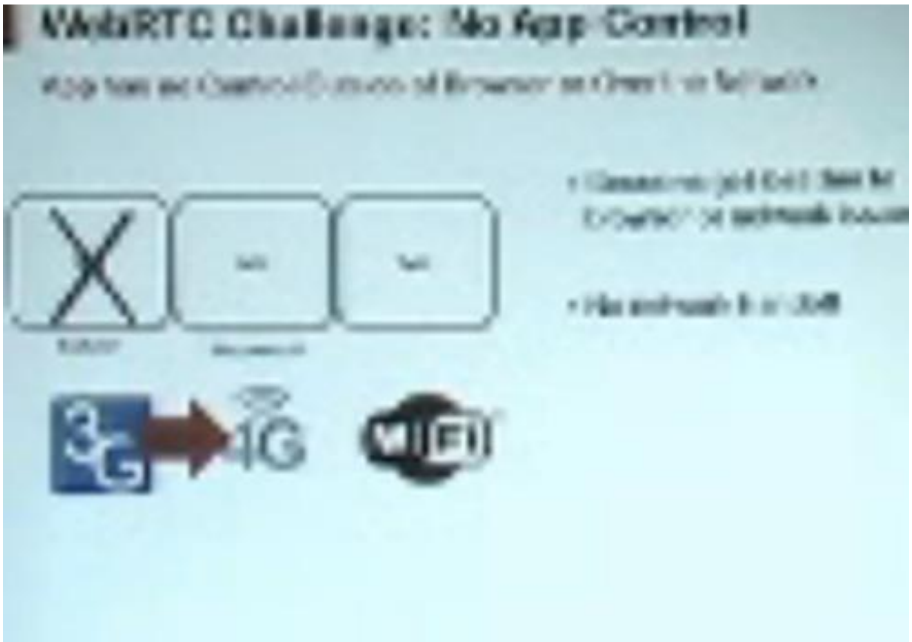
[7] Google is now the world's third largest carrier of Internet traffic. Facebook has become a member of the APG (Asia Pacific Gateway) submarine cable consortium. Yahoo has acquired PeerCDN, a content delivery network provider that claims it can reduce bandwidth costs by up to 90 percent by creating a peer-to-peer network using WebRTC's data channel protocol. (http://www.webrtcworld.com/topics/webrtc-world/articles/364011-yahoo-acquisition-peercdn-expands-webrtc-beyond-google.htm). Wikipedia provides a lengthy list of major CDNs as of 2013. Amazon Web Services (AWS) offers to carry data "that would have previously been transported over the Internet can now be delivered through a private network connection between AWS and your datacenter or corporate network." (http://aws.amazon.com/directconnect/faqs/) .



Watch http://www.youtube.com/watch?v=JEPAEV9SPVo

A key commercial advantage of telecommunications networks has been that they are connected to each other, offering any-to-any communications. That works well in a narrowband world where the traffic is mostly voice, text and data. In a broadband world the Internet itself offers global interconnectivity, but the Web requires standards that are interoperable to achieve any-to-any. In the case of proprietary standards, some are designed precisely to preclude that, the classic case being Apple's walled-garden approach.[8] Based on customer behaviour, it seems that there is appetite for both interconnected and "silo" communications for different purposes – WebRTC can enable both modes. (As an example, there is no relevance or benefit in interoperating a mobile karaoke service with online banking video-CRM).

In the case of WebRTC there are two alternative codec standards being pursued and under the sponsorship and funding of ISOC, two standards bodies, the IETF and the WC3. These bodies collaborate, the IETF focusing more on network protocols and the WC3 more on the APIs that go into browsers to enable secure interconnectivity between browser users.[9] Confusingly, IETF refers to RTCWeb and WC3 to WebRTC which has become the generally accepted name for the technology. Although the standards bodies agreed that WebRTC should employ a common codec, this is a continuing point of contention, principally between the video compression codec VP8 championed by Google and the ITUR's alternative codec H.264 which is favoured by Cisco among others.

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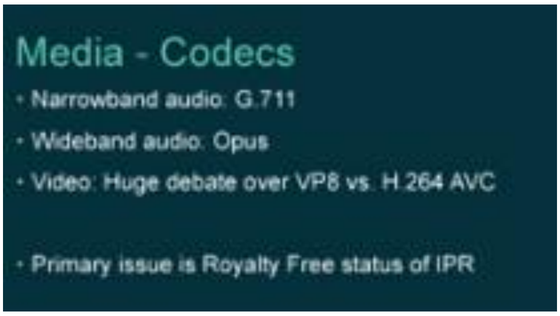
[8] It was two decades before the Mac and the PC could exchange files by which time the mutual advantage overtook the exclusive advantage.

[9] Previously browsers could only access their web server which offered protection from intrusive malware disguised as email, advertising, malicious websites, etc.; so freeing up browsers so they can connect directly with other browser users and servers creates security issues that WC3 is tackling.

A useful overview of the technical issues is provided by Dr. Cullen Jennings of Cisco, co-chair of the IETF RTCWeb working group, during a session recorded at the Internet Society's INET Bangkok on June 7, 2013.

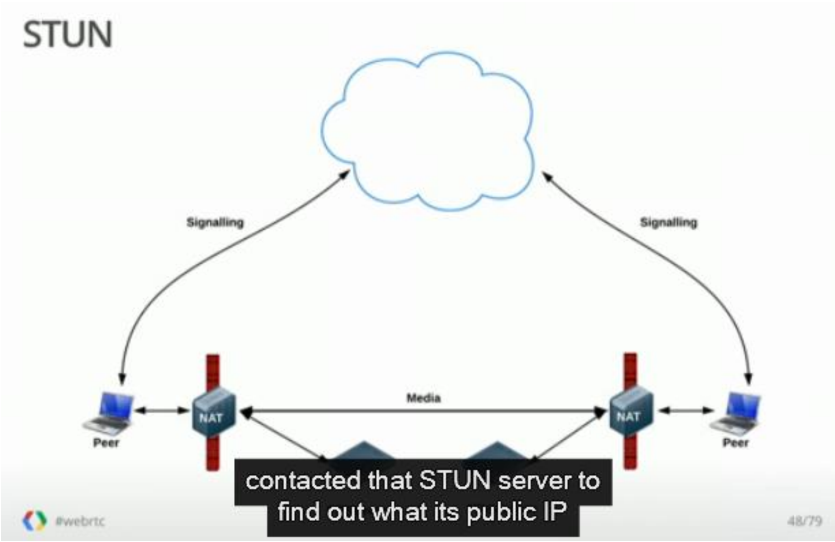


Watch: http://www.youtube.com/watch?v=Yf3eNciKddc

To make WebRTC work well there are inevitably a series of protocols that need to be developed to allow peers to find each other, to authenticate each other, to create session parameters, etc., and not all of these have yet been developed. For example, as the video above explains, communication errors need to be reported for resend and the appropriate protocol is still under discussion in the standards bodies working groups.

However, WebRTC is unlike most telecoms technologies in that there are many ways to achieve "workarounds" – development and deployment is occurring even though the standards aren't finalized yet. There is a plethora of tools available to help developers navigate the maze, create prototypes & launch commercial services. Ultimately, some of the "glitches" will be fixed in standards – but for many use-cases, the status quo is just a minor annoyance rather than a showstopper.

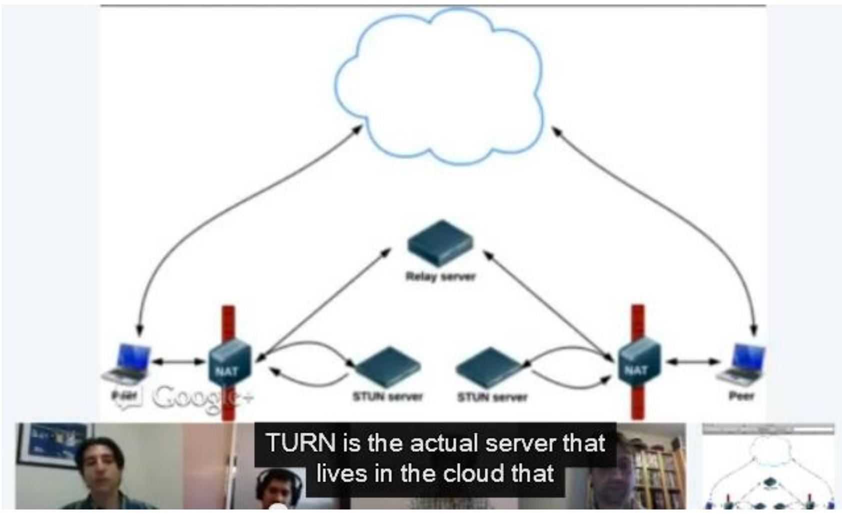
From an end-user perspective these are technical details of little interest or comprehension. But they mean that just as the Internet matured during the past twenty years to become a means of high quality and trusted communications (evidenced by its ubiquitous usage despite ever-present security challenges) so too will WebRTC communications. And just as no one needs to know what goes on under the hood of a car in order to drive it, it always helps to know when it won't start. For example, Box 1 lists the NAT, STUN and TURN server components of an architecture that supports WebRTC as partially illustrated in the screen shot below.



Watch: http://www.youtube.com/watch?v=p2HzZkd2A40

and

TURN



Watch: http://www.youtube.com/watch?v=dAhhniqwkp8

Box 1

Network Architectures that Support WebRTC

(Stuff only your IT and Web Design People Really Need to Know)

Network Address Translation (NAT) is a network protocol used in IPv4 networks that allows multiple devices to connect to a public network using the same public IPv4 address. NAT was originally designed in an attempt to help conserve IPv4 addresses.

NAT modifies the IP address information in IPv4header while in transit across a traffic routing device. This presents some drawback in terms of the quality of Internet connectivity and requires careful attention to the details of its implementation. In particular, all types of NAT break the originally envisioned model of IP end-to-end connectivity across the Internet and NAPT makes it difficult for systems behind a NAT to accept incoming communications. As a result, NAT traversal methods have been devised to alleviate the issues encountered. NAT has become a common, indispensable feature in routers for home and small-office Internet connections.

STUN (**Session Traversal Utilities for NAT**) is a standardized set of methods and a network protocol to allow an end host to discover its public IP address if it is located behind a NAT. It is used to permit NAT traversal for applications of real-time voice, video, messaging, and other interactive IP communications. STUN is intended to be a tool to be used by other protocols, such as ICE (see below).

The STUN protocol allows applications operating behind a network address translator (NAT) to discover the presence of the network address translator and to obtain the mapped (public) IP address (NAT address) and port number that the NAT has allocated for the application's User Datagram Protocol (UDP) connections to remote hosts. The protocol requires assistance from a third-party network server (STUN server) located on the opposing (public) side of the NAT, usually the public Internet.

Traversal Using Relays around NAT (**TURN**) is a protocol that allows for a client behind a network address translator (NAT) or firewall to receive incoming data over TCP or UDP connections. It is most useful for clients behind symmetric NATs or firewalls that wish to be on the receiving end of a connection to a single peer TURN does not allow for users to run servers on well-known ports if they are behind a NAT; it supports the connection of a user behind a NAT to only a single peer, as in video telephony, for example. In that regard, its role is to provide the same security functions provided by symmetric NATs and firewalls, but to *turn* the tables so that the client on the inside can be on the receiving end, rather than the sending end, of a connection that is requested by the client.

Interactive Connectivity Establishment (ICE) is a technique used in computer networking involving network address translators (NATs) in Internet applications of Voice over Internet Protocol (VoIP), peer-to-peer communications, video, instant messaging and other interactive media. In such applications, NAT traversal is an important component to facilitate communications involving hosts on private network installations, often located behind firewalls.

Source: Wikipedia

Conclusion

Why is WebRTC potentially so significant a part of this picture?

First, it offers cost-effective ways for web players, large enterprises (including call centres) and SMEs (including web developers) alike to reach out to customers, suppliers, partners; to connect staff across locations, locally and globally; and to develop new apps, using any type of device that can support voice, data and video. Minus the video, this also includes feature phones and even consumer electronics devices.

Second, it offers a remarkably efficient way to communicate real-time AV-D across platforms such as telecoms and computer networks and across devices. Because it can so be used directly with a browser or embedded into a website or within an app it is highly versatile. Efficiency + versatility = a compelling… so long as interconnectivity/interoperability across platforms can be assured wherever it is required. The growing WebRTC ecosystem, including cloud platforms, open-source tools and broad vendor/startup support, means that there is a low barrier to entry, experimentation and prototyping. Coupled with the blistering speed of WebRTC evolution, this is prompting many players to "try it out" even at this early stage, for fear of being left behind.

Third, as with services such as Skype, YM and SNS voice, text and video apps before it, it can directly eat away at traffic over the networks controlled by local telecom service providers. This is especially true of voice telephony traffic which has been the principle *raison d'être* for telecom companies for well over a century. But the challenge is also an opportunity for telecom companies so long as they maintain their dominance over investment in broadband networks.

Fourth, as part of a growing family of OTT technologies and services, WebRTC could take the convergence of the worlds of telecoms and IT to a new level by transforming telecommunications into a cloud-based service. It has taken most of two decades for NGNs to replace TDMA-PSTN telecom services and the process is mostly still very far from complete. Will it take the same time for telecom services in the cloud to become the NNGN?

Some useful references in addition to those cited above:

- http://www.html5rocks.com/en/tutorials/webrtc/basics/
- http://www.webrtcbook.com/
- http://www.webrtcworld.com/
- https://hacks.mozilla.org/2013/09/webrtc-update-and-workarounds/
- http://temasys.com.sg/category/news/
- https://search.oracle.com/search/search?start=1&search_p_main_operator=all&q=webRTC
- http://tools.cisco.com/search/results/en/us/get#q=webRTC

And some useful blogs:

- http://disruptivewireless.blogspot.sg/
- http://bloggeek.me/
- http://www.webrtc.org/home
- http://cp.wainhouse.com/blog/2013/11
- http://temasys.com.sg/blogs/