WebRTC to complement IP Communication Services

Version 1.0

20 February 2016

This is a Non-binding White Paper of the GSMA

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1 Introduction

1.1 Executive Summary

Mobile operators can leverage WebRTC (Web Real-Time Communication) technology in several ways. This whitepaper considers the different options available from both a commercial and strategic perspective. Mobile operators could use WebRTC to extend the capabilities of their IMS (IP Multimedia Subsystem)-based infrastructures and expose their services to a larger number of customers and customer devices. Implemented through an internet browser, WebRTC has the advantage of not requiring a dedicated client software or a plug-in: Many browsers now have built-in support for peer-to-peer real-time communication.

In 2014, the GSMA Web Working Group published a whitepaper describing the technical attributes of WebRTC and identifying a number of opportunities at a high level for operators to benefit from this internet-based client technology. GSMA members and associate members can access that whitepaper on InfoCentre2 [1]. This new whitepaper builds on that paper by exploring the opportunities related to IMS implementation in greater detail.

1.2 Definitions

Term	Description
3GPP	3rd Generation Partnership Project
API	Application Programming Interface
B2B	Business to Business
C2C	Consumer to Consumer
C-RAN	Cloud Radio Access Network
DPI	Deep Packet Inspection
DSL	Digital Subscriber Line
DTLS	Datagram Transport Layer Security
E911	Enhanced 911
eIMS-AGW	Access Gateway Control enhanced for WebRTC
eP-CSCF	Enhanced Proxy - Call Session Control Function
FMC	Fixed Mobile Convergence
H2M	Human to Machine
ICE	Interactive Connectivity Establishment
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IoT	Internet of Things
IP-CAN	IP Connectivity Access Network
JSON	JavaScript Object Notation
LTE	Long Term Evolution
LTE-A	Long Term Evolution - Advanced
M2M	Machine to Machine

Term	Description
M2H	Machine to Human
MVNO	Mobile Virtual Network Operator
NAT	Network Address Translation
NFV	Network Function Virtualization
NNI	Network to Network Interface
ORTC	Object Real-Time Communication
OTT	Over the top
PCEF	Policy and Charging Enforcement Function
PCRF	Policy and Charging Rules Function
PS	Public Safety
QoS	Quality of Service
REST	Representational State Transfer
RESTful	A service based on Representational State Transfer architecture
RTC	Real-Time Communication
RCS	Rich Communication Suite
SDK	Software Development Kit
SDN	Software-Defined Networking
SIP	Session Initiation Protocol
SME	Small or Medium Enterprises
SRTP	Secure Real-time Transport Protocol
TAS	Telephony Application Server
TTS	Text to Speech
TURN	Traversal Using Relays around NAT
UC	Unified Communication
UE	User Equipment
UNI	User Network Interface
ViLTE	Video over Long Term Evolution
VoLTE	Voice over Long Term Evolution
W3C	World Wide Web Consortium
WAC	Web Application Controller
WAF	WebRTC Authorisation Function
WCDMA	Wideband Code Division Multiple Access
WebRTC	Web Real-Time Communications
WIC	WebRTC IMS Client
WWSF	WebRTC Web Server Function
XMPP	Extensible Messaging and Presence Protocol

1.3 References

Ref	Title / Link
[1]	WWG.01 - WebRTC Opportunities for Operators https://infocentre2.gsma.com/gp/wg/WWG/OfficialDocuments/WWG.01%20WebRTC%20Opportunities%20for%20Operators%20v1.0%20(Current)/WWG.01%20v1.0.pdf
[2]	RTCWeb Working Group https://datatracker.ietf.org/wg/rtcweb/charter
[3]	Web Real-Time Communications Group at W3C www.w3.org/Consortium/activities#Web_RealTime_Communications_Working_Group
[4]	2015 Q1 Update: WebRTC Market Status & Forecasts Report http://disruptivewireless.blogspot.co.uk/p/blog-page_30.html
[5]	WebRTC browser support scorecard http://iswebrtcreadyyet.com
[6]	Browser to browser using the RTCPeerConnection API http://caniuse.com/#search=webrtc
[7]	Enhanced 911 Wireless Services https://www.fcc.gov/encyclopedia/enhanced-9-1-1-wireless-services
[8]	Amazon Mayday is Making Us Rethink Customer Service http://www.sightcall.com/amazon-mayday-making-us-rethink-customer-service
[9]	How Extending Websites and Mobile Apps into the Contact Center Using WebRTC Enables Rich, In-Context Customer Interaction https://developer.att.com/static-assets/documents/enhanced-webrtc/transforming-customer-engagement-using-webrtc-v3.pdf
[10]	AT&T Enhanced WebRTC API https://developer.att.com/apis/enhanced-webrtc
[11]	Orange's Libon messaging app lets you chat with people who don't use Libon https://gigaom.com/2014/02/20/oranges-libon-messaging-app-lets-you-chat-with-people-who-dont-use-libon
[12]	Launching Today Libon 'Reach Me' http://www.prnewswire.co.uk/news-releases/launching-today-libon-reach-me---youll-never-miss-an-important-call-again-509794431.html
[13]	What should a telco really do (appear.in)? https://www.chriskranky.com/telco-really-appear
[14]	Get videochat on your blog or website in 5 minutes! http://blog.appear.in/post/108535587025/get-videochat-on-your-blog-or-website-in-5
[15]	NTT Communications Offers World's 1st Speech-recognition API for Multiple Browsers at No Charge http://www.eu.ntt.com/en/about-us/press-releases/news/article/NTT_Communications_Offers_Worlds_1st_Speech-recognition_API_for_Multiple_Browsers_at_No_Charge.html

1.4 Overview

Mobile operators have long offered customers reliable communications services that reach across networks. Now they are enriching their communications proposition by employing IP-based technologies that can support advanced features, such as file sharing and video calling. The GSMA uses the term “Green Button Promise” to refer to the provision of RCS (Rich Communication Suite) messaging, video-based calling and VoLTE (Voice over Long Term Evolution) calls with the same telecoms-grade experience associated with circuit-switched voice and SMS services. When consumers press the green button on a device to initiate a voice call, they expect the call to connect first time and stay connected.

However, communications always has a context. Increasingly, this context is provided by mobile or web apps, supporting online streaming platforms, social networks or individual web pages. WebRTC enables operators to extend their communication services into these contexts with the features and benefits they offer in the mobile or fixed-line space. Every communication context where the mobile operator does not offer a communication channel opens the door for third-party service providers to offer OTT (over the top) communication services, disintermediating the operators.

Both IP-based communications and WebRTC technology have relative strengths and weaknesses. WebRTC is a flexible web-based technology that provides many, but not all, of the key components required for an end-to-end communication service. Some of the gaps in WebRTC are addressed by telco IMS services. Combining the two to deliver complementary services enables a mobile operator to fully utilise its communication infrastructure.

At the same time, RCS, VoLTE, Enriched Calling and other IMS-based services, which are usually delivered either using traditional cellular networks or WebRTC facilities, need to be supported by Network APIs (Application Programming Interfaces) to provide a full solution.

This whitepaper aims to identify the market opportunities of WebRTC particularly when integrated with telco IMS services.

2 What is WebRTC?

2.1 WebRTC Standards

WebRTC is based on internet standards defined by the IETF [2] (for the protocols) and W3C [3] (for the client APIs). WebRTC can underpin peer-to-peer communication services, such as messaging, audio-video communication and file transfers, typically with internet browsers as endpoints. In some cases, browser-less WebRTC native endpoints can be used. WebRTC is a client technology and every device (mobile and fixed) running a browser capable of supporting this technology can establish a peer-to-peer communication over the web.

2.2 WebRTC Advantages

Thanks to WebRTC, providing audio-video applications on the web has become much easier: it enables developers to create a website to distribute a link to the individuals wishing to communicate.

The biggest difference between WebRTC and older web technologies is the user experience: WebRTC is able to access device hardware, such as microphones or cameras, without the need to install a plugin or preload a dedicated communication application, such as Skype or Viber.

The benefits for mobile operators to consider when implementing WebRTC based services can be grouped in five main categories:

a) Cost reduction
b) Ease of use
c) Security
d) Fast time to market
e) Simplified device integration

2.2.1 Cost Reduction

Licensing and integration costs for WebRTC-based solutions are often much cheaper than is the typically the case for licence-based vendor solutions.

For example, equipping call centres with WebRTC technology, rather than maintaining high cost toll-free numbers, with a guaranteed quality of service (operator values), can significantly reduce implementation, operation and maintenance costs for enterprises.

2.2.2 Ease of Use

WebRTC can be used to deliver video and audio calls to a range of web browsers on any device without the need to download and install any clients.

Moreover, WebRTC supports contextual communications that allow the end users to stay on the website, navigating and making calls or sending messages, without the need to use a separate device or to leave the page they are looking at.

2.2.3 Security

All communications using WebRTC are fully encrypted between peers. Unlike legacy PSTN systems, with WebRTC technology, encryption is still a major contributor to end-to-end security for consumers and enterprises.

2.2.4 Fast Time to Market

Communication services can be developed quickly using WebRTC technology. Web developers can easily integrate communication features into their application and deliver enterprise business flows immediately on the web. Furthermore, worldwide, there are more web developers than any other set of developers, with a large supportive community and online documentation. Therefore, developing for the web is easier and faster than in other environments.

2.2.5 Simplified Device integration

WebRTC could simplify the integration of voice and video services for device manufacturers, as WebRTC services rely mainly on browsers and HTML/CSS/JavaScript programmes, which permit swift and frequent upgrades. Thus, mobile operators do not need to wait for, or stay synchronized with, the development cycle of devices.

2.3 WebRTC Limitations

Compared to traditional telco standards, internet standards are developed in a modular way. As a result, from a telecommunications perspective, WebRTC lacks several aspects. These gaps, together with others that might emerge as WebRTC gets deployed, provide opportunities for mobile operators to bring their requirements and expertise to build these additional building blocks into the new telecommunication standards.

2.3.1 Signalling

WebRTC accepts any signalling mechanism, such as SIP, XMPP and REST API, to avoid redundancy and maximise compatibility with existing technology, signalling and protocols that are not defined in the WebRTC standards. However, implementation providers, such as *Matrix*, *Orca.js and Sippo*, can overcome signalling interoperability issues.

2.3.2 Presence

WebRTC does not specify presence. Unless a service provider offers this capability, end users have no information whether the party they wish to communicate with is available on a WebRTC-enabled device.

2.3.3 Authentication

To maximise the interoperability with existing systems, the WebRTC standard doesn't specify authentication, so a user can be authenticated via any solution. WebRTC can be plugged into existing mechanisms via its peer identity API.

2.3.4 Address Book Integration

As with presence, any existing address book integration solution can be leveraged with WebRTC.

2.3.5 Ringing Notification

As the specification does not address notification of an incoming communication, the WebRTC endpoint needs to be connected to the internet and have its browser instance running to receive calls.

2.3.6 Native Push Notifications

The WebRTC standards don't define native push notifications. Users will need to have their WebRTC client instance (web application or browser) running in order to receive notifications.

2.4 WebRTC and IMS Integration

WebRTC does not prescribe a particular way to facilitate the discovery of the parties involved. This enables the application developer or service provider to choose from a variety of mechanisms. Depending on the use cases, interoperability with existing solutions may be a priority or not.

When interoperability with existing services is a requirement, and an IMS core has been deployed, this type of service architecture can offer significant benefits:

- Enhanced WebRTC capabilities, such as presence, can be leveraged at relatively low cost
- Bridging between WebRTC and non-WebRTC endpoints can be done seamlessly

For IMS-based deployments, WebRTC enables the expansion of communication services to any endpoint with a WebRTC compliant browser (or native client). The strengths of WebRTC and IMS-based services complement each other (see Table 1).

	WebRTC	**IMS-based services (such as RCS)**
Strengths	• Clientless environment • No plug-ins required • Secure peer-to-peer audio / video communications • Secure data transfer between peers • Codec support	• Provides end-to-end capability detection • No need for browser resources • Supports quality of service • Allow address book integration • Supports presence • Transcoding is done in IMS
Weaknesses	• Browser features dependency (fragmentation) • No presence feature • No address book integration • No quality of service • No native push notification	• Client is implemented on a per device basis • Limited support for interoperability • Deployment is slow moving

Table 1: WebRTC and IMS-based services weaknesses and strengths comparison

WebRTC can support any audio or video codec that has the session description defined by the IETF. The codecs that should be implemented include the pervasive H.264, VP8 for video and G.711 for audio (AMR-WB is optional). A common set of codes will ensure a call can always be completed.

2.5 WebRTC Alignment with Network Evolution

2.5.1 Alignment with 4G Evolution

The continuing evolution of mobile networks, such as the increase in bandwidth brought by LTE-A (Long Term Evolution – Advanced) compared to LTE, is helping to address the needs of the best effort real-time services, such as WebRTC. Where LTE has been deployed in lower frequencies (such as 700MHz or even 450MHz), mobile operators can provide widespread geographic coverage, making WebRTC services accessible for large numbers of customers.

The move towards outsourcing and virtualization in the core and radio networks, using technologies such as NFV, SDN and C-RAN, is also well-aligned with the deployment of WebRTC, which can run from any “cloud “environment. Moreover, the evolution of mobile networks to support a web-friendly service framework with increased availability of network APIs, fits natively into the scope of WebRTC.

2.5.2 Alignment with 5G

5G mobile network technologies are still in the very early phases of development. However, it is likely that WebRTC services delivered over 5G will benefit from improvements such as:

- Much lower latency
- Much greater bandwidth
- Ultra-high reliability
- Guaranteed security, trust, identity and privacy

For example, 5G could enable person-to-person or person-to-group UHD (Ultra High Definition) mobile video communication via WebRTC. Today, this use case would put a very high load on the network or probably would not work at all, but, with 5G, pervasive video, such as collaboration in 3D cyber-office, may well be a feasible use case.

Employing WebRTC-based applications for demanding use cases, such as gaming, remote control, tactile internet or e-health, will be highly feasible in the 5G environment.

2.6 Evolution of WebRTC NV (Next Version)

Work on the second version of the WebRTC API (currently known as WebRTC Next Version) will begin in the first half of 2016, incorporating some of the additional protocol flexibility developed as part of ORTC (Object Real-Time Communication). This presents an opportunity for telcos to bring their requirements to the WebRTC APIs, based on the specifics of supporting WebRTC communications in mobile networks.

Tackling the current WebRTC weaknesses of end point addressing and presence would make audio and video chat (and data exchange) seamlessly accessible for casual use. It would also pave the way for WebRTC to become an essential service option for communications between customers and businesses offering services over the internet. Table 2 shows the impact WebRTC could have on usage of telcos’ communications services, according to Disruptive Analysis.

End-year, million	**2011**	**2012**	**2013**	**2014**	**2015**	**2016**	**2017**	**2018**	**2019**
VoLTE subscribers	0	0	11	45	90	180	330	480	700
Of which IR94 ViLTE/RCS video users (active)	*0*	*0*	*0*	*1*	*5*	*15*	*30*	*50*	*70*
Residential VoIP lines (NGN / IMS)	150	180	205	240	275	310	355	395	440
Total NGN/IMS VoIP (exc UCaaS/SIP trunks)	**150**	**180**	**216**	**285**	**365**	**490**	**685**	**875**	**1140**
Carrier V/VoIP extension - active user-devices									
- Native OS / 3GPP WiFi-Calling	1	2	4	15	35	60	90	130	170
- Carrier-integrated app/application	0.5	1	2	3	8	20	40	65	120
- Browser	0	0	0	1	5	15	30	55	90
Total	**1.5**	**3**	**6**	**19**	**48**	**95**	**160**	**250**	**380**
Extensions as % Carrier VoIP subs	1%	2%	3%	7%	13%	19%	23%	29%	33%
- of which WebRTC	0%	0%	0%	1%	15%	25%	35%	45%	55%
WebRTC extensions	**0**	**0**	**0**	**0**	**7**	**24**	**56**	**113**	**209**
Non-WebRTC (3GPP WiFi-calling, SIP softphone etc)	1.5	3	6	19	41	71	104	138	171
Estimated % supporting/using video	0%	0%	0%	2%	6%	9%	11%	14%	18%
Carrier video-over-IP extensions	0	0	0	0	3	9	18	35	68
% WebRTC	0%	0%	0%	0%	50%	60%	70%	75%	80%
Carrier WebRTC Video extensions	**0**	**0**	**0**	**0**	**1**	**5**	**12**	**26**	**55**

Note: WebRTC extension use of video more likely to be residential/fixed rather than ViLTE/RC

Table 2: WebRTC extensions to VoLTE-VoIP lines (Disruptive Analysis, 2015 [4])

3 The RTC Vendor Landscape

The market can be segmented into two categories of players:

a) WebRTC Cloud Platform Providers
b) WebRTC Application and Gateway Vendors

3.1 WebRTC Cloud Platform Providers

Mostly start-ups, these companies develop platforms which provide RTC (Real-time Communication) solutions to web and mobile developers. These platforms employ RESTful (a service based on Representational State Transfer architecture) APIs to offer basic communications features, such as messaging, voice and video call capabilities, to web developers. The business models for these start-ups are typically based on a subscription or a monthly fee. These providers, which utilise open SDKs (software development kits) to react to market changes quickly, can typically afford to take risks, develop in an agile way and “fail early”.

Today, a number of start-ups offer real-time communication services, typically implementing the “Click to Call” use case scenario in which the end user clicks on a URL on a dedicated website to initiate an audio or video call to the customer care function of that service provider.

To support interoperability, a number of cloud WebRTC service providers have opened up their communication APIs and use open source server implementations of SIP over WebSockets, such as Asterisk, FreeSWITCH, Kamailio and OverSIP. These implementations provide interoperability with SIP endpoints, but don’t yet include integration with mobile operators’ IMS networks.

There are many Cloud API unified communications (UC) providers offering similar services. These include:

- 46elks
- Aculab
- Apidaze
- Layer
- Nexmo
- Plivo
- Telestax
- TenHands
- Xura Forge

Some of the more successful start-ups have achieved scale and/or have been acquired by large vendors:

- *Tropo* (recently acquired by Cisco) - A cloud-based UC service provider offering voice, video and messaging capabilities via cloud APIs. Many of the Tropo cloud-based APIs offer rich functionality, such as high definition call recording, speech recognition, TTS (text to speech) conversion, conferencing and HTTP live streaming.

Its business model, which is based on usage (such as call volumes and number of messages sent), is to charge developers directly. It also earns revenues by integrating WebRTC with telco operator's IMS services.

- *Twilio* - A cloud-based UC service provider offering voice (VOIP), video and messaging (SMS, MMS and IP based) capabilities for developers through cloud-based APIs. Twilio's products and services are focused on the integration of communications into developer applications. It covers many use cases, including: two-factor authentication, appointment reminders, call recording, masked phone numbers and call tracking.

  Twilio's business model is built around a pay-as-you-go model, with a specific subscription charge for a phone number, message sent or voice minute. Premium services, such as call recording and call transcription, are charged as additional extras.

- *Tokbox (Telefonica)* - Please see section 8.2 for details.

3.2 WebRTC Application and Gateway Vendors

This group includes vendors that develop WebRTC applications either in a browser or in dedicated native applications utilising WebRTC features.

Equipment vendors providing IMS infrastructure to operators, typically offer WebRTC capabilities in gateways designed to interconnect web and telco services. These gateways expose telco services to web developers via APIs (RESTful or JavaScript) and may provide transcoding functionality in case of an internet-based and telco-based codec mismatch. Their business models are typically based on the sale of equipment and additional royalties related to the number of users.

Gateway vendors typically fall into one of three main categories:

a) Traditional telco equipment vendors
b) Non-traditional equipment vendors
c) Start-ups and small or medium enterprises (SMEs)

3.2.1 Traditional Telco Equipment Vendors

These large companies usually provide telco network equipment for communication purposes. They include:

- Alcatel-Lucent
- Ericsson
- Huawei
- Nokia
- Xura

3.2.2 Non-traditional Equipment Vendors

These companies generally provide "middle boxes" for telecoms networks, such as firewalls, SBCs and NATs. Many provide their own WebRTC gateway for IMS interoperability. They include:

- Broadsoft
- Dialogic
- Genband
- Hewlett Packard
- Mitel (formerly Mavenir)
- Newnet
- Oracle (formerly Acme Packet)
- SONUS

3.2.3 Start-ups and SMEs

These companies usually have a gateway offer, supported (and often funded) by a collaborative larger enterprise. They include:

- Broadsoft
- Crocodile Communications (recently acquired by Acision, has been recently rebranded as Xura)
- Dialogic
- Solaiemes (recently acquired by Comverse, has been rebranded as Xura)

4 Growth Figures of WebRTC

Some analysts expect a rapid increase in the availability of the real-time communications services. There will be seven billion devices with WebRTC-compatible browsers in use by the end of 2019, according to Disruptive Analysis [4]) (see Figure 1).

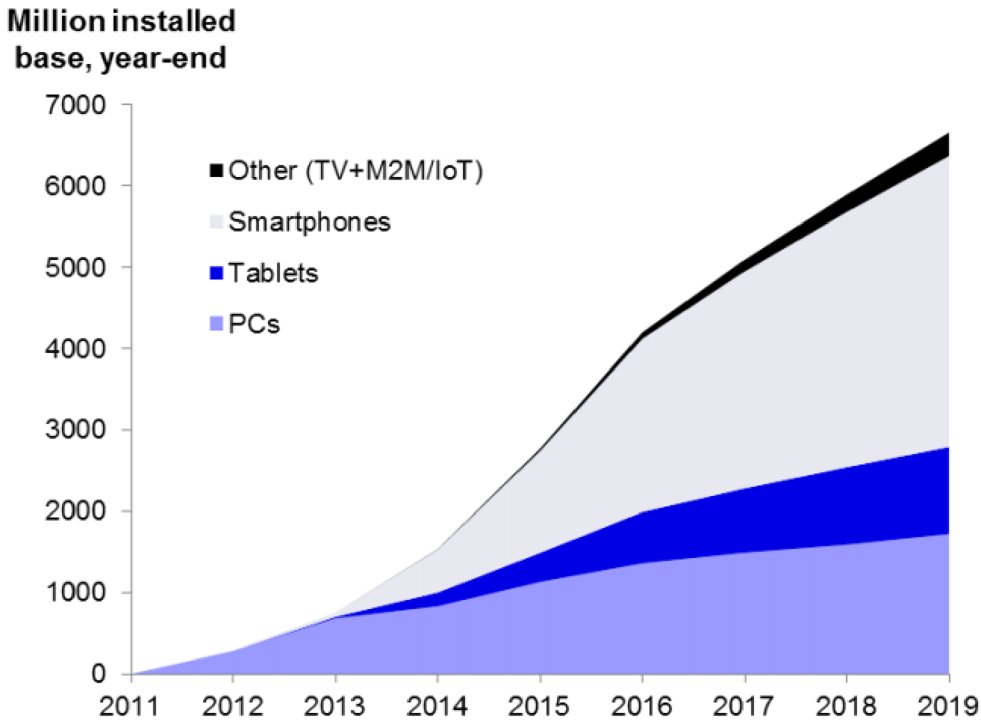

Figure 1: Overall WebRTC Device Support (Disruptive Analysis, 2015 [4])

Rising adoption of smartphones will be the key driver of this trend. The Android and iOS mobile platforms are used by more than 90% of the smartphones sold each year. Android supports WebRTC natively in the default browser Chrome, since the Marshmallow release in 2015. An initiative is underway to develop the code required to run WebRTC in the Safari browser, and it seems likely that Apple will include this code in a future version of its iOS. Moreover, a large number of desktop PCs, laptops, tablets and other computing devices already support real-time communication through WebRTC-capable browsers. Disruptive Analysis forecasts there will be nearly two billion WebRTC-enabled PCs in use by 2019 (see Figure 2).

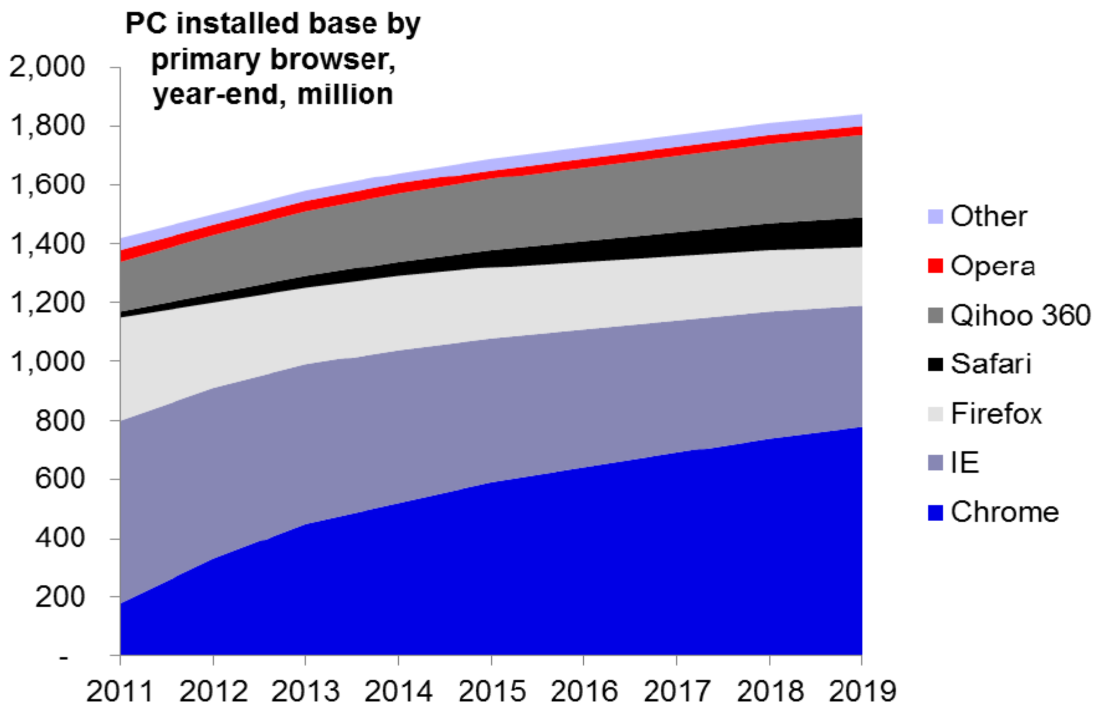

Figure 2: PC installed base by primary browser (Disruptive Analysis, 2015 [4])

Web browsers supporting WebRTC features include Mozilla Firefox, Google Chrome, Opera Turbo and lately Microsoft Edge / Internet Explorer (although the support in Microsoft Edge is currently limited to ORTC, which forms the basis for the next generation of the WebRTC APIs). iswebrtcreadyyet.com scores browsers' support for WebRTC today at almost 56% [5] [6]. Figure 3 shows the composition of this figure: A red square indicates the browser doesn't support that WebRTC feature, while an amber square indicate some support. Green indicates full support.

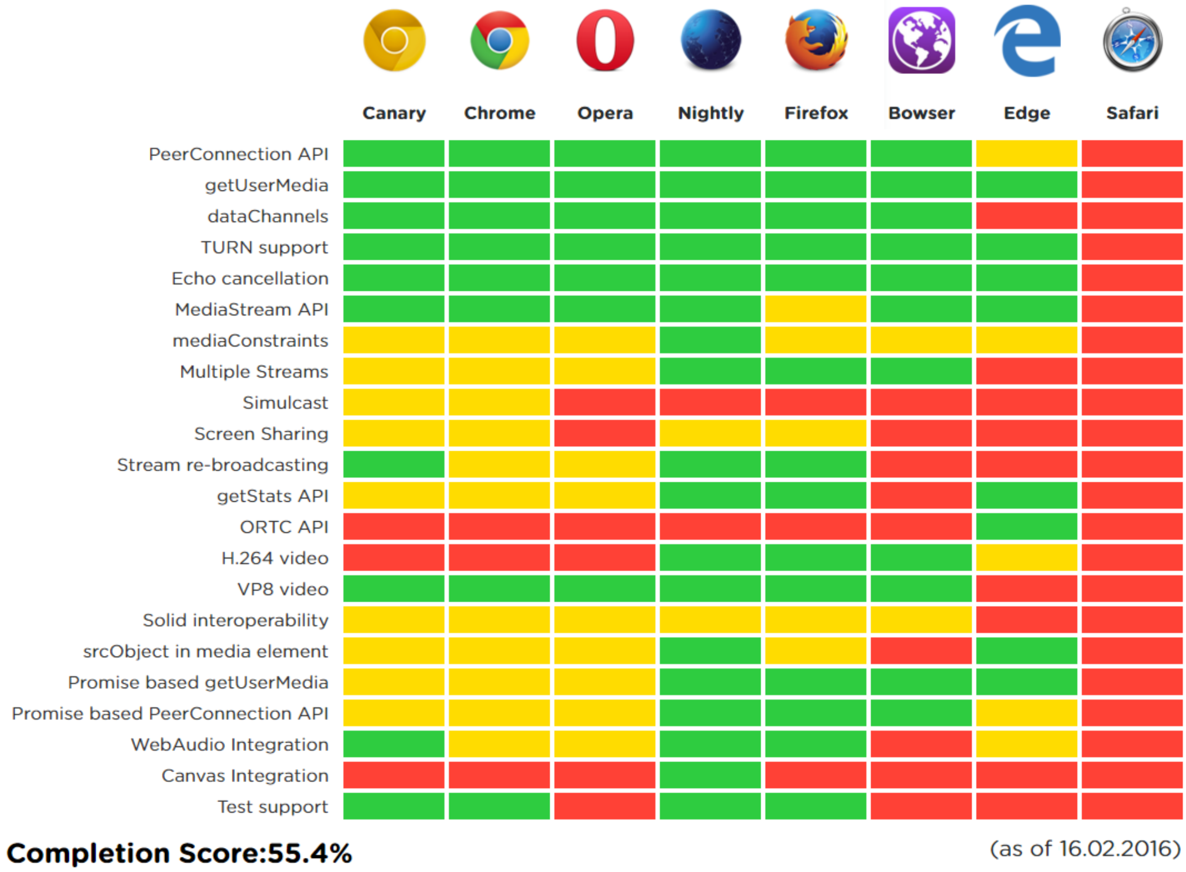

Figure 3: Browser WebRTC Support Scorecard [5]

5 WebRTC – IMS Technical Integration

Many telecoms operators are already experimenting with using WebRTC to extend their core voice and messaging communication services into the web domain.

From a telco point of view, interoperability between WebRTC and the telco domain is becoming important because of the declining market share of traditional operator communications services as alternative solutions, offered via social networks, cloud-based services and stand-alone over-the-top messaging apps, gain traction.

There are several use cases, independent of the application, that require such interoperability. These include:

- When a break out from WebRTC to existing applications / PBX platforms is needed
- When a break out from WebRTC to PSTN is needed
- When PSTN endpoints want to join WebRTC-based communications
- When endpoints on other PBX platforms want to join WebRTC-based communications
- When users of one silo want to communicate with users of another silo – user communications between two OTT solutions in which the mobile operator provides interoperability

There are two main ways to build WebRTC interoperability with telcos:

a. Interoperability between WebRTC and IMS Telcos:

WebRTC clients can be interconnected over telco IMS systems via IMS NNI (see Figure 4). This interconnection can be enabled via peer-to-peer IP connections and/or the use of exchanges and hubs (such as an IPX) (see PRD IR.34 [x] and PRD IR.65 [y]).

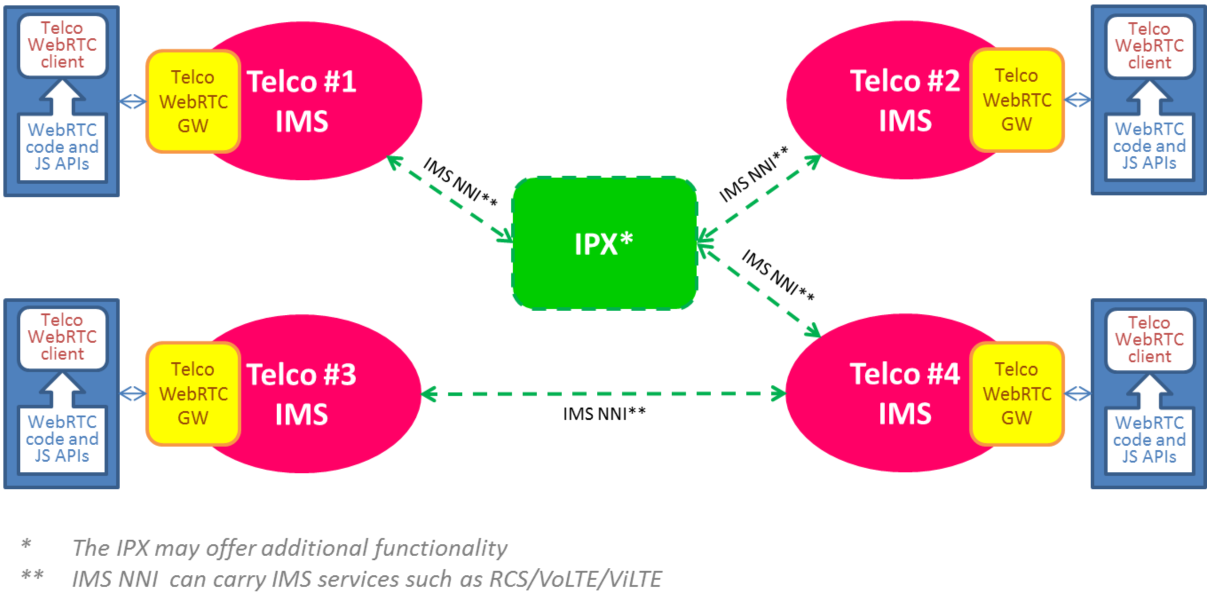

Figure 4: Interoperability between WebRTC and IMS Telcos

b. Interoperability between WebRTC-Enterprise PBX and IMS Telcos:

Enterprises who use WebRTC endpoints with their IP-PBXs will need to connect with telco endpoints (WebRTC or traditional) through gateways, as they do today for the PSTN. This next generation interoperability will also be enabled with gateways between the enterprise systems and IMS (see Figure 5).

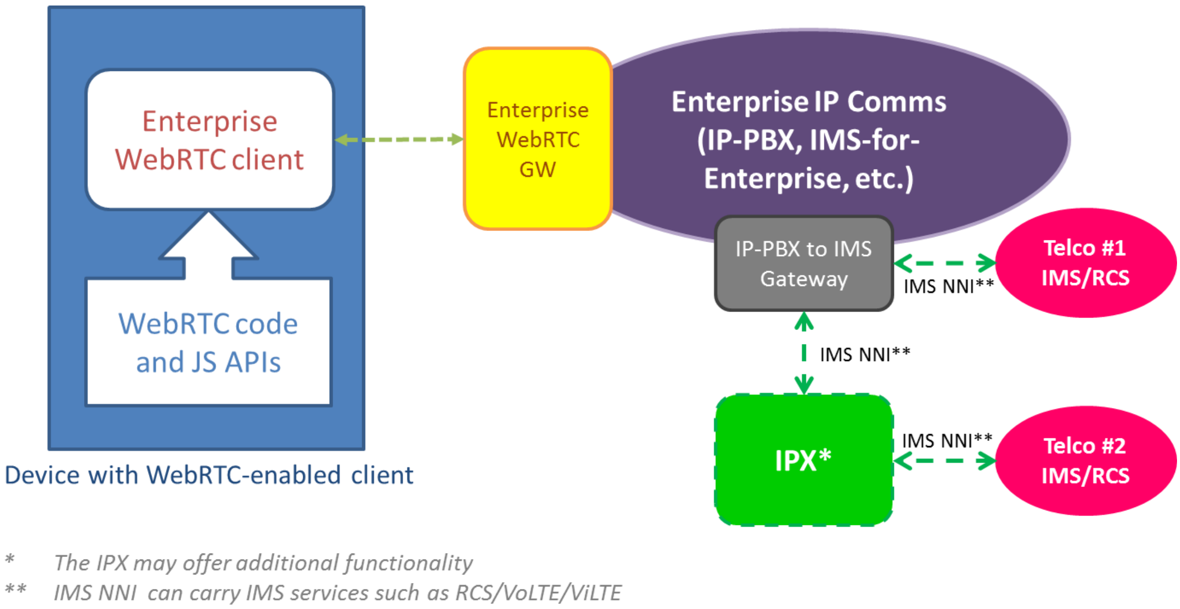

Figure 5: Interoperability between WebRTC-Enterprise PBX and IMS Telcos

There are several different mechanisms that can be used to enable interoperability. These include:

a) Vendor’s gateway
b) Dedicated TAS solution
c) Infrastructure-based interoperability

5.1.1 Vendor’s Gateway

A common mechanism to establish WebRTC interoperability with a telco domain is to use a gateway to handle protocol translation and transcoding. However, this approach can impact the customer experience when it comes to video transcoding between endpoints on the internet and in a mobile network. The cost of transcoding can also be expensive for mobile operators, depending on the gateway vendor’s business model.

5.1.2 Dedicated TAS Solution

Another approach is to have a dedicated TAS (Telephony Application Server) integrated into the existing IMS core implementation.

Operators can use a gateway to IMS to expose telco services to internet developers. There are some issues associated with this approach, such as the cost and the quality of the experience in cases where transcoding is required for video communication between a telco and internet-based customers.

5.1.3 Infrastructure-Based Interoperability

The 3GPP (3rd Generation Partnership Project) standards body has released the architecture design for WebRTC access to an IMS network, as part of Release 13 in 3GPP TR 23.228 technical report proposal. This specification is based on a study carried out by the 3GPP project for exploring different options for “WebRTC Access to IMS” as described in 3GPP TR 23.701. Several options have been proposed for WebRTC integration, with the latest approved technical recommendation (TR 23.228) describing the approved reference architecture, shown in Figure 6. This technical recommendation allows WebRTC to employ different signalling mechanisms, such as the use of JSON or SIP and XMPP over WebSockets, as well as a RESTful-based interface.

The reference architecture shown in Figure 6 describes the interfaces of WebRTC gateway integration to IMS.

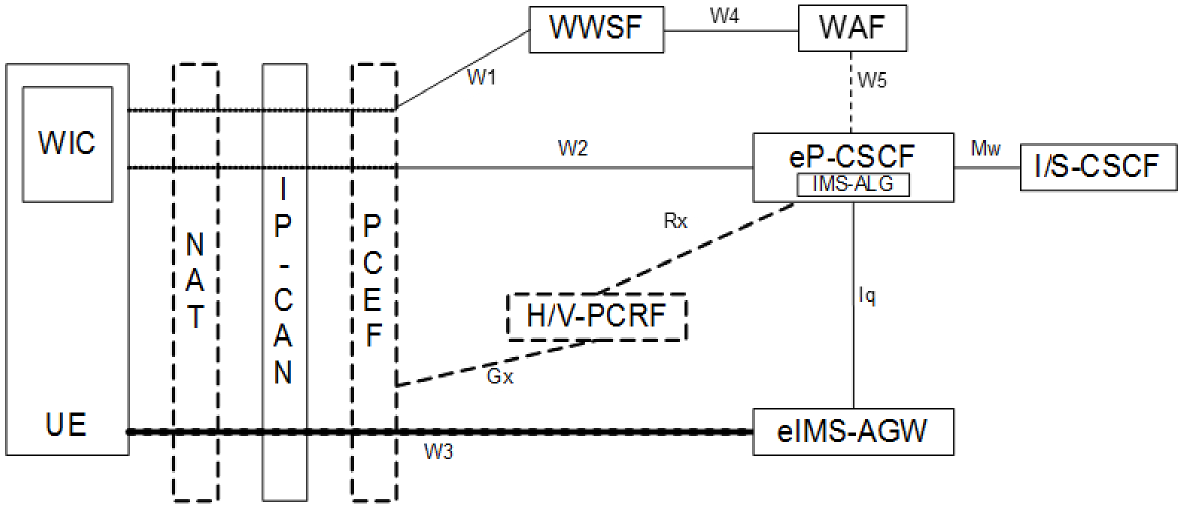

Figure 6: WebRTC reference architecture to IMS and reference model (3GPP TR 23.228)

In the reference model shown in Figure 6, the elements marked with dotted lines are configuration-dependent.

WIC (WebRTC IMS Client) is a JavaScript application including the extension for providing access to IMS and is downloaded from WWSF (WebRTC Web Server Function). WWSF is the initial contact point from the internet allowing and controlling communication access to the IMS platform.

The **NAT** (Network Address Translation) is also implementation-dependent, but the WIC is usually behind the NAT. WebRTC uses the ICE protocol to implement two way SRTP (Secure Real-Time Transport Protocol) communication between endpoints.

The **IP-CAN** is the typical IP access network where the **UE** (User Equipment) can access IMS access technologies, such as WCDMA, LTE, Wi-Fi and DSL.

The **WWSF** can be part of the operator network or a third party service provider network.

The **eP-CSCF** represents the WebRTC signalling endpoint supporting a variety of signalling protocols (SIP and XMPP over WebSockets, RESTful based interface etc.). It is located in the operator's network.

The **WAF** (WebRTC Authorisation Function) has to be on the same domain and it provides authorisation tokens to WWSF. It can be located in the operator network.

The **eIMS-AGW** (Access Gateway Control enhanced for WebRTC) supports the media plane interworking extension for WIC. It also supports the media security, media consent, DTLS-SRTP key exchange for media components using SRTP. Furthermore, it supports media transcoding and NAT traversals, including ICE.

The **PCRF** (Policy and Charging Rules Function) supports policy and charging control rules based on media and session-related information.

The **PCEF** (Policy and Charging Enforcement Function) uses deep packet inspection (DPI) technologies to determine, based on rules, whether the traffic is allowed or not. It is optional. As WebRTC traffic is fully-encrypted, DPI would not be possible on the user plane.

In summary, there is a well-defined implementation standard for operators to consider implementing WebRTC with IMS. Many gateway vendors offer seamless integration to IMS, either exposing operator's services via RESTful API or JavaScript libraries.

A typical vendor implementation is illustrated in Figure 7:

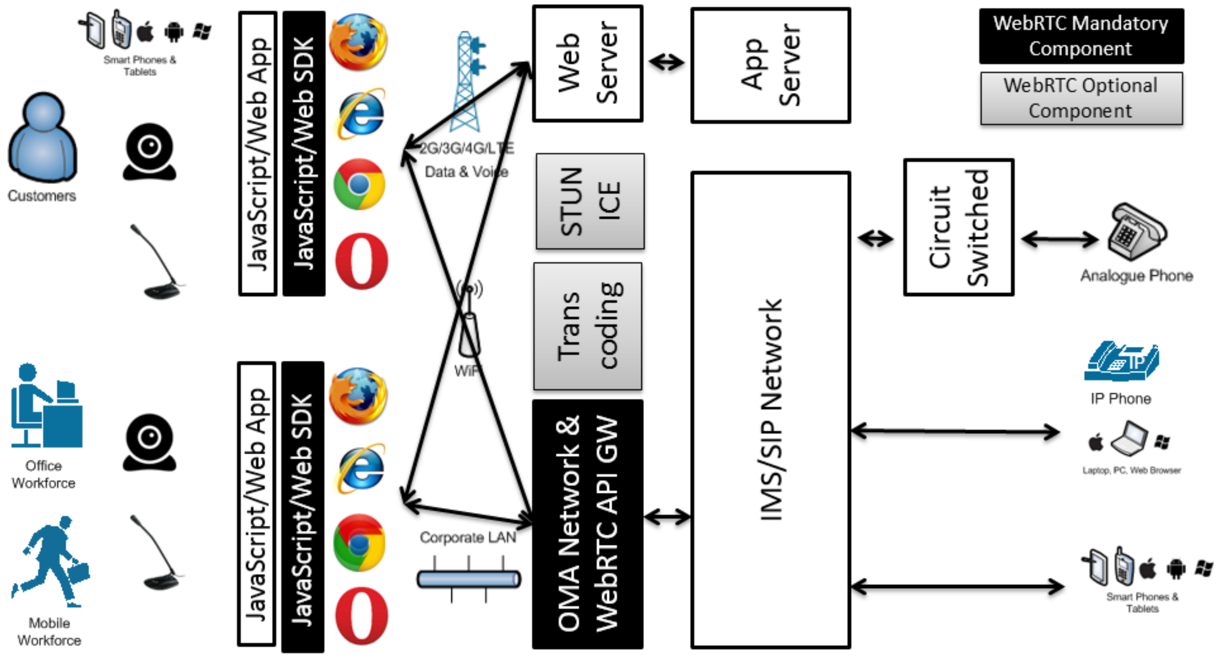

Figure 7. Typical vendor gateway implementation example

6 Use cases enabled by integration of WebRTC

6.1 Introduction

Operators can use WebRTC technology to extend their enterprise and consumer communications services beyond the handset. Extending the reach of these core services increases their utility and relevance to customers by making them available on more of the customer's devices. This offers convenience to customers, who can then manage their communications from multiple screens.

As WebRTC evolves, there are many opportunities and applications that will be supported over the WebRTC platform such as:

- Consumer-to-Consumer (C2C) communications
- Business-to-Consumer (B2C) communications
- Business-to-Business (B2B) communications
- Internet of Things (IOT)
- Interconnectivity Provider
- Public Safety (PS)

6.2 Consumer-to-Consumer communications

In general, the WebRTC-related opportunities for mobile operators in the consumer segment revolve around extending existing end-to-end voice and text services into the web domain or the creation of new end-to-end consumer services. Browser-based communications over LTE networks can enable a consumer to maintain a conversation over multiple devices and device-types, such as smartphones, PCs, tablets, televisions, in-car browsers and others.

As it is browser-based, WebRTC enables developers to mash-up IP communications with other applications, such as interactive gaming. This is of particular importance to help keep web-based contextual communications within the mobile operators' ecosystem. With WebRTC and internet app development kits, telcos have opportunities to redesign the user experience of unified communications (e.g. voice/video and text).

WebRTC can be used to deliver RCS via a web client, which supports partial or all RCS features. With an RCS web client (WebRTC-enabled browsers plus RCS-enabled web applications):

- Users can make voice or video RCS phone-calls, and the callees might be WebRTC users, RCS users or users of legacy telecommunication services.
- Callers and callees are able to employ existing RCS user-IDs.
- Users can employ RCS across multiple devices.
- WebRTC availability can be supported as RCS-capability detection: A person's WebRTC availability can be shown in the relevant sections of the RCS contacts/phonebook application.
- RCS official accounts may utilise WebRTC to offer web-accessible real-time audio/video services, such as (toll-free) hotlines, to their customers/subscribers.

The market segments of WebRTC are expanding. Telefonica OpenTok is the first commercial WebRTC-architecture service that enables video chatting on browsers across iOS and Android smartphones, and PCs.

6.3 Business-to-Consumer communications

In the B2C market, WebRTC is used to provide contextual communications in areas such as customer care (airlines, insurance, health care providers etc.). For example, educational institutions can use WebRTC to facilitate interactive parent-teacher conferences. Or businesses, such as airlines, insurance companies, healthcare providers, financial institutions and retail stores, can use WebRTC to enable service agents to engage with customers on specific questions and customer needs.

In this context, WebRTC offers several advantages, such as:

- No need for softphones
- The end user can stay on the website and can receive advertisements whilst waiting for the operator to connect
- No need to go through a complex IVR menu

Mobile operators could provide guaranteed quality of service for real-time communications, and leverage authentication and identity management capabilities from IMS to support B2C applications.

Other B2C applications include the extension of mobile operators' core services (e.g. Telefonica's TU Go proposition) and the creation of ad-hoc "chat rooms" (video and/or text), such as Telenor's "appear.in" WebRTC video conference service (see section 8.5 for more details).

6.4 Business-to-Business communications

In the enterprise market, mobile operators could leverage the multi-device capabilities of WebRTC to offer unified communications integrating audio, video or messaging service enhancements in specific vertical use cases, such as healthcare, enterprise communication, and education. As WebRTC is browser-based, the end user's device can be any hardware with a browser (e.g. doctors' tablet computers, handheld Wi-Fi devices, or a white-board with the ability for participants to remotely share a notepad). Operators can again enhance the service with authentication, identity management and quality of service guarantees.

Commercial call centres have long used WebRTC. Avaya, the leading vendor of traditional enterprise call centres, has employed WebRTC to set up commercial hybrid call centres, which are accessible from both traditional telecom networks and internet apps.

Other B2B applications include:

- Customer service and sales support in wholesale markets where WebRTC operators can provide audio and video contact services to third party enterprises wanting to enhance their current web-based customer experience.
- Integration within enterprise applications, such as CRM and ERP software.
- APIs to enable web developers to integrate with mobile operator services and service features from their websites (e.g. AT&T Enhanced WebRTC API).
- Provision of service components, such as example gateways, firewalls, multi-party calls, conferencing, messaging, back-up, authentication and identity, to developers.
- Fully-fledged video-conferencing services supporting booking, scheduling and other features.

6.5 Internet of Things

WebRTC defines a communications construct called DataChannel. It is a socket-like protocol (modelled after the WebSockets) between peers, via the RTCDataChannel API. Its key distinguishing feature, and its tremendous potential, lies in the peer-to-peer nature of WebRTC.

The WebRTC DataChannel allows for the creation of distributed solutions based on ad-hoc meshing, making it well-suited to support IoT (Internet of Things) scenarios, or more specifically WoT (Web of Things), assuming that the end points already implement the standard web stack.

WebRTC DataChannel's main functionality is to put communications in context. Examples of context include:

- a troubleshooting session with the customer service
- a tutoring session with a teacher
- a game played online

Moreover, WebRTC DataChannel provides developers with more flexible usage of the network (e.g. unordered packets, low-latency data exchanges), which is applicable beyond peer-to-peer usage, e.g. in multi-party gaming or remote control.

Web applications can leverage WebRTC communications without relying on browser instances to be open at all times. This is useful for both machine-to-machine (M2M) and machine-to-human (M2H) type scenarios in the IoT market. WebRTC could be used for carrying communications between sensors, such as industrial monitors or energy meters, as well as in the smart wearables segment. The technology could also be used to provide user control of (and data retrieval from) IoT devices utilising human communications interfaces, such as texting and talk.

Finally, the use of real-time communication APIs in a dedicated application is faster and more cost effective than developing a dedicated SIP client. The distribution of upgraded services will also be easier as operators do not need to install new applications on each terminal or require their users to do so. Instead, they can simply deploy the upgraded machine-to-human (M2H) version in the cloud.

6.6 Interconnectivity Provider

Operators could become interconnectivity providers for other communications service providers. WebRTC underpins a number of OTT services including social networks, such as Google Hangouts, Facebook WhatsApp calling features (voice and video) and Twitter (calling options) etc. These OTT services only provide connectivity within their own "*large island"*, but do not provide interoperability with one another. Operators could enable interconnectivity among these islands, providing both unified signalling and quality of service (which could be monetised) to the respective service providers.

6.7 Public Safety

In future, E911 [7] emergency calls will likely need to support video (in addition to voice) to help dispatchers see the emergency remotely and provide first-responders with more contextual information. WebRTC could also be used in other public safety scenarios. For example, the technology could enable insurance service agents and their customers to visually exchange details of emergency situations for quick assessment and approval of insurance for vehicle repairs or other property damages.

6.8 Use Cases for Media Processing by the Operator-Hosted Server

Mobile operators' media processing capabilities can support a range of use cases for WebRTC. These include:

- **Centralised Server for Multipart Conferencing:**
  Enable third party developers to create multi-party messaging, voice or video conversations or broadcasts
- **Media Transcoding such as VP8 vs H.264:**
  Convert media between WebRTC standards and common telco standards to enable interoperability
- **Interworking with IMS:**
  Connect to operator-hosted IMS functions, such as voice/messaging services

- **Centralised Server for Media Recording:**
  Enable recording of conversations and conferences set up through a server
- **Human to Machine / Machine to Human (IoT)**
- **Click to Call:**
  Simple mechanism to connect customers by proving a button in the web domain for the customer to connect directly to sales or care organisations
- **Voting:**
  Simple mechanisms for users to express preferences within the web environment. For example, an audience voting for favourite characters in TV shows may use the web channel in addition to SMS
- **File Exchange and Transfer:**
  File transfer within the context of a conversation between two parties. For example multimedia messaging in which the sender uses WebRTC and the receiver has an IP connected device.

6.9 Use Cases for Services Provided by the Operator

Mobile operators can support WebRTC with a range of value added services. These include:

- **Video conferencing:**
  Video conversation or broadcast between single or multiple parties
- **Developer (network and device) API driven by carrier:**
  Exposure of operator core network functionality through APIs. For example AT&T, please see section 8.1.
- **Enterprise services:**
  End-to-end B2B services, such as click to call-for-call centres, video customer care support, internal audio-video conference, etc.
- **Interworking with IMS:**
  Enabling extension of IMS functionality into the web domain for services such as voice/messaging services
- **Carrier as WebRTC Identity Provider:**
  Telco IDs such as MSISDN may be used for WebRTC-based authentications. For example AT&T, please see section 8.1.
- **Video Call Centre:**
  Video calling proposition with booking and service management wrap around on top of basic video functionality. For example, Amazon, please see the link in [8].

6.10 Future Opportunities

Operators deploying WebRTC technology, to offer rich real-time / context based communication services, will be able to benefit in several different ways:

- Operators can provide communication capabilities in the context of other services and where OTT competitors are already offering their services.
- Opportunities to provide identity, routing and authentication functionality, leveraging the SIM card etc.
- Opportunities to provide media processing, signalling transcoding-as-a-service and translation.
- Opportunities to provide media storage and big data services.

- Opportunities to provide developer APIs for small businesses.
- Can deploy a WebRTC Gateway as a part of NFV architecture.
- Can deploy WebRTC as part of a software-defined network.
- Can support Internet of Things use cases (surveillance, monitoring etc.)

7 Opportunities for Operators to Expand IP Communications with WebRTC Ecosystem

A service provider can leverage several different business models to monetise WebRTC services, regardless of whether they have employed cloud or gateway-based implementations. WebRTC offers a low cost solution to developers and service providers seeking to deploy communication services on the internet.

Examples of potential business models include revenue share, prepay services, subscription models and freemium models .

7.1 Revenue Share

In this model, a portion of the revenue is shared with the solution provider based on pre-agreed proportionate figures. As an example, when a customer calls a vendor from within an application, the call can be tracked (using call transaction records) and a revenue share paid back, based on the call record. It can include call termination to PSTN and GSM end users.

7.2 Prepay Service

The end user tops up the account for making calls and sending messages per subscription on a monthly basis. This option is very popular with cloud service providers, which may offer prepaid packages of 300 minutes of calls, 300 texts etc.

7.3 Service Based Subscription Model

This model provides an opportunity for service providers to buy a bulk volume of traffic “share” minutes and per text messages and pay in bulk each month. It can involve large volumes thus requiring a fixed capacity for partners.

7.4 Freemium Model

Certain basic functionality can be offered free of charge (either time or volume based) with some premium services, such as call recording, quality of services, storage and detailed call records, attracting a charge (after a limited trial period). This business model can attract a large number of customers in a short time period.

There will be additional business models that could be used to monetise WebRTC based solutions, but a detailed discussion of this topic is beyond the scope of this document.

8 Mobile Operator WebRTC deployment case studies

A number of operator services has been implemented based on WebRTC technology. These include telco OTT services, developer programmes and hybrid solutions combining the two. This section features case studies of existing implementations.

8.1 AT&T

AT&T has exposed telco services to web developers via JavaScript and RESTful APIs utilising WebRTC gateways to its IMS backend platforms [9]. AT&T offers complete calling solutions from basic to advanced level [10]. AT&T is also offering a WebRTC API to simplify VoIP application development, supported by a straightforward SDK. These tools are attracting a large number of web developers to use AT&T’s core services (voice, data and messaging).

As WebRTC does not define a unique identifier, AT&T has developed a variety of solutions for developers to choose from, such as:

- AT&T existing mobile number
- Virtual number
- Account ID (social networking, email address etc.)

The main advantages for developers are that the WebRTC API:

- is easy to integrate into applications
- has a simple and robust structure
- is supported by a full-featured SDK
- has reference server access for rapid development

The key features of AT&T’s WebRTC APIs are:

- **E911:**
  The API provides full support for E911 calls, overcoming one of the obstacles to wider deployment of WebRTC. For example, when an emergency call is made through the browser, it will be routed to 911 call centre and will include name, address, location etc.

- **SIP trunking:**
  It offers integration with SIP trunking for inbound and outbound call purposes dependent on the organisation’s setup, facilitating the click to call (web page to a call centre) scenario via IP-based PBXs.

- **Conferencing rooms:**
  Developers can expose conferencing features to enable users to hold conferences with up to 30 participants. The moderator can add new participants to the conference via WebRTC or dialling out to PSTN lines.

- **Directory services:**
  The WebRTC API supports a directory service where people can connect with one another by selecting their IDs from the list, as described above.

- **Mid-call controls:**
  Call features, such as call hold, mute, call transfer, resume and moving calls between endpoints (from a browser based call to a smartphone), are fully supported as a part of the enhanced call solution options.

- **Call states:**
  This feature provides the call–state information to developers when the calls are terminated, connected or in-progress.

AT&T offers the current WebRTC API to registered developers free of charge. (Developers should sign up for the platform for a nominal charge).

At the time of writing this whitepaper, AT&T had no plans to monetise its WebRTC API. However, when a developer decides to use a virtual number provided by AT&T, there is a small service charge for the number rental.

8.2 Telefónica

Telefónica acquired TokBox (https://tokbox.com) in 2012. TokBox uses WebRTC to provide live video communication services to customers of all sizes, from independent developers to global enterprises. TokBox owns, develops and operates the OpenTok Platform which is designed to make it quick and easy for customers to add live video, voice and messaging communications into their online and mobile websites, apps and services. TokBox says its scalable, customisable platform gives users the creative freedom to develop any video and voice interaction, from one-to-one chats to large-scale broadcasts.

OpenTok provides APIs and SDKs for native app development, supported by a scalable, global cloud infrastructure. TokBox says OpenTok has been used to add live video communications capabilities to more than 100,000 services by companies, such as Mozilla, Major League Baseball, Royal Bank of Scotland, Bridgestone Golf, Fox Sports and Double Robotics.

The OpenTok platform features the following capabilities:

- **Archiving:** Record live calls for later playback.
- **iOS and Android SDKs:** Fully interoperable mobile and web SDKs
- **Multi-party:** Optimised and reliable multi-party calls for large groups or broadcast
- **OpenTok Plugin for Internet Explorer:** Support across multiple end points
- **Text Chat:** Add native text chat without third party plugins
- **Security:** Encrypted media and signalling
- **Audio and Video Drivers:** Ability to fully customise audio and video streams
- **Firewall Traversal:** Connect across corporate and enterprise firewalls
- **Audio Detection:** Use audio activity to control stream layout and display
- **Audio Fall-back:** Dynamically prioritise audio in response to network quality
- **Video:** 1:1 or multi-party video calls
- **Voice:** Ability to build powerful audio-only embedded solutions
- **Messaging:** Ability to add text chat, moderator controls, etc.
- **Screen Sharing:** Share content for an enhanced communications experience

In addition to TokBox, Telefónica has used WebRTC in both its Tuenti and Tu-Go propositions:

- **Tuenti** is a Spanish social networking and Telefónica MVNO subsidiary. It launched a WebRTC / VoIP dedicated client app offering unified communication features to end users. The Tuenti solution is a proprietary solution using WebRTC technology.

- **Tu-Go** enables customers to use Wi-Fi to make and receive calls, texts and check voicemail when they do not have a mobile signal. The proposition also enables customers to use their mobile number across multiple devices (smartphones, tablets and laptops). WebRTC is used to extend the multi-device functionality beyond apps and into the browser domain so that customers can make and receive calls and texts directly via WebRTC capable browsers.

8.3 Orange Libon

Orange Libon (https://www.libon.com) is best described as a telco OTT service, incorporating WebRTC into an existing mobile VoIP service (similar to Viber or Skype) [11] [12]. Libon offers calling out to mobile numbers (E.164), IP calling, voicemail and messaging features on iPhone and Android clients, complemented by a fully-featured web-based communication interface based on WebRTC technology.

Libon has taken the approach of integrating WebRTC into the existing VoIP platform rather than deploying a more usual peer-to-peer topology. To do this, Orange has developed a new border controller component that enables routing of WebRTC traffic into the telco domain. This approach also allows monitoring of call quality, supports carrier obligations and has the ability to route traffic over telco-grade interconnections.

Libon 3.0 allows non-Libon users to chat with Libon registered users via the web application. The service is simple: the non-Libon user receives a text message containing a link, enabling them to access the web app without registration and establish a rich communication session leveraging WebRTC technology.

The Libon “Reach-Me” feature was launched in June 2015. With this feature, users can receive calls in areas where mobile coverage is weak or non-existent. It works on iOS and Android smartphones and allows users to receive calls to their mobile number via Wi-Fi.

8.4 Deutsche Telekom

Deutsche Telekom discontinued the open development platform in its “Developer Garden” web portal (based on Tropo solutions, please see section 3.1 for details) in the summer of 2015. However, its open API approach will continue in the areas of localisation, authentication and payment, beyond communication services.

To date, Deutsche Telekom has not provided access-agnostic OTT communication services based on WebRTC technology. However, it sees WebRTC as a useful enabling technology, which can help save cost and time during the development process. Deutsche Telekom considers the technical outreach of WebRTC-enabled browsers (enabling a single development across all relevant devices, such as smartphones, tablets, PCs and Smart TVs) as the technology’s main advantage.

Various business segments of Deutsche Telekom use WebRTC-enabled communication tools. These include:

- **Fixed line / Convergence (FMC):**
  “HomeTalk” is a dedicated application, which provides full fixed-line telephony functions on a mobile device via Wi-Fi (proposition: “use your mobile as a fixed phone”). The frontend is a browser-based web application. The backend is a SIP-

WebRTC gateway, which is connected to the fixed line IMS platform to ensure the full QoS and connectivity.

- **Customer Care:**
  Deutsche Telekom's customer care service provides browser-based contact points for end users. Customers can establish a text-based chat, voice or video communication to a customer care agent directly from their devices or from a web site via WebRTC technology.

8.5 Telenor

In 2013, a small group of developers within Telenor Digital created a WebRTC-based video chat service called appear.in (http://appear.in) [13] [14]. The service offers dedicated chat rooms where up to eight people can join the video chat session at one time. There is no need for registration and anybody can use these chat rooms free of charge for a basic video chat service.

The service also allows users to register and claim rooms (making them private, where there are more features offered) and invite guests to join them. Registration can be done via a username, email or a mobile number. Having logged into the service, registered users can make WebRTC calls with each other.

Although the service supports teleconferencing, it doesn't currently enable file exchange, whiteboard and file sharing features. Appear.in has recently published a free Developer API and a JavaScript SDK for web developers, allowing them to build web-based services employing this service. There are also various plugins available for the Chrome browser and the blogging platform WordPress.

Appear.in has developed native applications for Android and iOS smartphone platforms which do not support WebRTC.

8.6 NTT Communications, Skyway

At the end of 2013, NTT Communications launched [15] a solution called Skyway, based on a WebRTC platform, which can be used by developers to enable standard features, such as voice, video calls and file transfers between endpoints.

Skyway also supports a new and enhanced feature called "subtitled-chat" which can be introduced during a voice call, where both parties can see the context printed in the browsers of their conversations.

In January 2015, NTT Comms announced a new feature called TURN (to relay audio and video data streaming between end points) to enable WebRTC peer-to-peer based communications in a highly secure environment. Furthermore, NTT Comms claims to have deployed the world's first voice recognition API for multiple browsers. It is available free to developers.

8.7 KDDI, ChatWork

KDDI introduced a WebRTC business-focused chat tool through a cooperation with ChatWork, a cloud-based provider located in San Francisco in 2013 (http://www.chatwork.com/product). The ChatWork solution offers audio, video and file management features combined with task management for business users.

8.8 Tata Communications, Jamvee

Tata communications has launched a business-focused, global video conferencing tool that enables any customer to join a conference from any device, anywhere at any time. Tata Communications is providing the connectivity for this cloud-based solution worldwide.

9 Conclusions

WebRTC is ushering in a new phase in the long term competition between OTT players and mobile operators in the core communications business (voice, video, file sharing, messaging).

To date, OTTs have built their position by developing services faster than telcos and through device agnosticism that enables multiple touchpoints with customers. The business rules of online content/service providing have challenged the traditional telco development process, requiring operators to consider communications services outside the traditional fixed and mobile device context.

However, operators have embraced the RCS initiative, which leverages their own strengths (QoS, charging and billing, identity, security and privacy). Now WebRTC provides an opportunity for operators to compete effectively with OTTs through free services, immediate user experience, and fast offer updates. Through integration with WebRTC, RCS can also become a multi-device communications service.

The combination of WebRTC and RCS can help to mitigate the decline of revenue in the traditional voice business. To recover revenues, operators could in future make some functionalities pay per use. That is, in the medium-term, operators will be able to earn revenues from data plans, advertising, premium services (such as file sharing, cloud storage and games offers) and perhaps subscription fees.

9.1 Strategic Options

There are several strategic options available for mobile operators:

- The Telco OTT play
- Telco IMS integration with WebRTC gateway
- Hybrid (a combination of the two above)
- Take no actions
- Interconnectivity provider

9.1.1 The Telco OTT Play

An operator could build their own OTT service, such as Telefonica Tokbox, Telenor "appear.in" and Orange Libon. This approach involves building a standalone service on the internet for end users associated with a separate brand name. These services can also be integrated with the operator's core services. For example, Telefónica Tokbox has aspirations to integrate Telefónica's core services with its OTT video conferencing service in the consumer market.

9.1.2 Telco IMS Integration with a WebRTC Gateway

An operator can integrate a WebRTC gateway into its IMS core. This approach, which has been well defined by 3GPP, supports the extension of the telco's VoIP or VoLTE service into the web, with the benefits outlined earlier in this document. Furthermore, this approach allows telcos to give their subscribers the ability to send a WebRTC URL to their friends and family for easy communication. It also enables the telco subscriber to use their VoLTE service on the web to make click-to-calls from any web page with a phone number on it, and it enables the telco subscriber to utilise a VoLTE service to talk to a friend while surfing and sharing web pages.

Although there are some concerns about the cost of transcoding, the impact on the end user experience and royalties, IMS integration with a WebRTC gateway is a natural step for any IP communications solution and an obvious way to extend the telco's IP communications services into web scenarios.

9.1.3 Hybrid (The combination of the two above)

The prime example of a hybrid approach is that taken by AT&T, which is using a WebRTC gateway to its IMS core, but the main focus is the WebRTC API exposed to developers to provide a number of telco-specific services, such as authentication and identity. The prime service offer is free of charge to web developers through the existing AT&T developer programme.

AT&T also offers a JavaScript SDK to simplify development. AT&T has no intention to monetise its WebRTC API at the time of writing this whitepaper (Jan 2016), but a hybrid approach could be underpinned by a freemium business model offering a basic free service with charges for additional extras through a subscription model. More details about AT&T's service solution can be found at section 8.1.

9.1.4 Take No Action

It is perfectly acceptable for operators to take no action and not implement any WebRTC-based services, therefore providing only the connectivity. This may be the approach taken by emerging market operators still earning major revenues from traditional voice and text services or no-frills operators. However, the risk of taking no action for the mobile operator is that it becomes a connectivity provider only to OTT players and is not serving end users.

9.2 How mobile operators can drive the evolution of WebRTC

Telecommunications standards are developed by telecom operators and their suppliers to achieve service interoperability and vendor solutions that are plug-and-play. The objective of W3C specifications is to achieve browser interoperability and a significant amount of the implementation work is carried out by browser vendors. If the use cases and features required by operators do not resonate with browser vendors or translate into a clear priority for them, it can be challenging to have operator requirements included, especially as browser vendors are building a free product. To influence the evolution of WebRTC, mobile operators must find compelling arguments that entice browser vendors to support those features, and must be ready to advocate these features in the standard bodies, such as IETF and W3C.

Secondly, WebRTC does not evolve according to a classic waterfall process: early versions of the technology were available and continued to be refined while the standard was being drafted. To drive the specification, operators must participate, supporting requirements with working solutions that demonstrate where specification evolution is necessary.

9.3 Summary

An important technology and standard in the communications industry, WebRTC can offer contextual communication. It is a baseline for platform and API providers, enables opportunities for mobile operators to extend their IMS-based solutions, such as RCS and VoLTE. It also creates a new ecosystem for communication service providers, web developers and cloud vendors. It presents several business opportunities and multiple different business models for mobile operators, as described in this document.

Annex A Document Management

A.1 Document History

Version	Date	Brief Description of Change	Approval Authority	Editor / Company
1.0	20 February 2016	Initial version	PSMC	Istvan Lajtos / GSMA David O'Byrne / GSMA Erdem Ersoz / GSMA

A.2 Other Information

Type	Description
Document Owner	Network 2020
Editor / Company	Erdem Ersoz / GSMA

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