WebRTC standards update (September 2014)

Victor Pascual Avila
Victor.pascual@quobis.com
@victorpascual

About Me

Technology, Innovation & Strategy Consultant

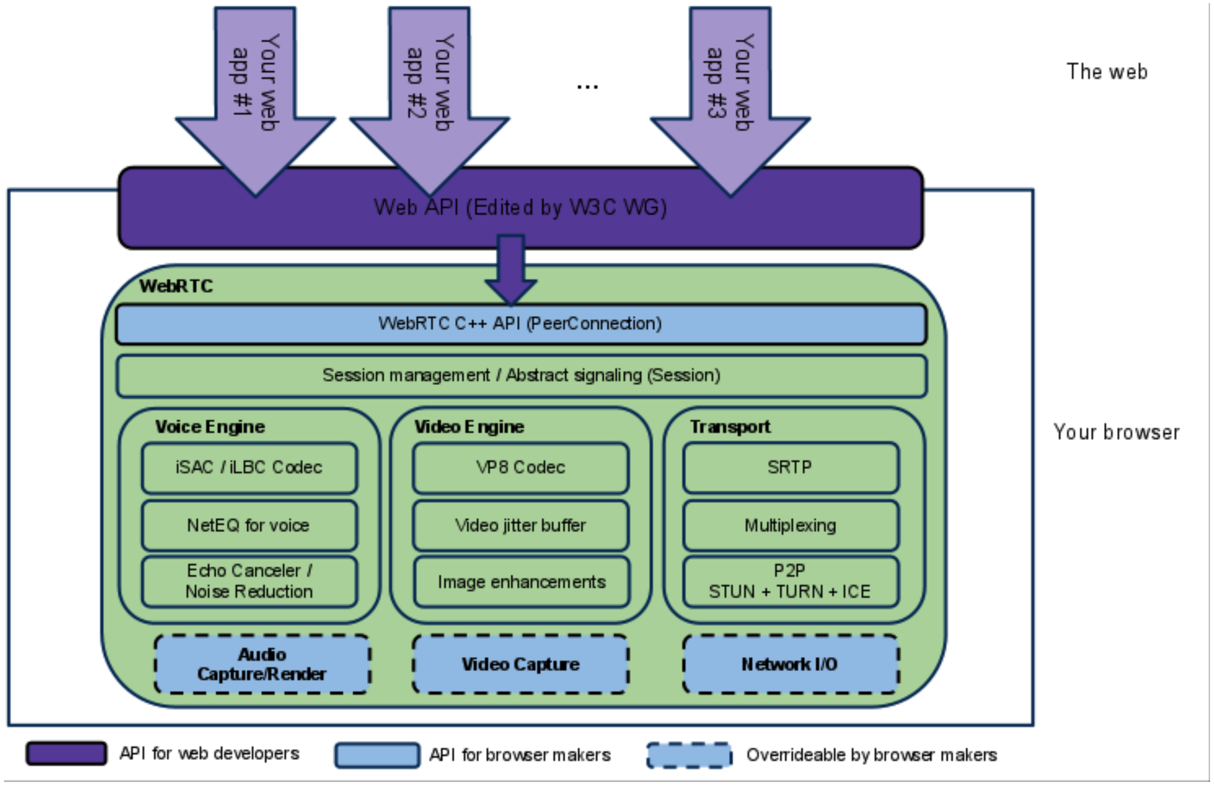
Main focus: help make WebRTC happen – involved in WebRTC standardization, development and first industry deployments (on-going RFX's, PoC's and field trials)

Other activities:

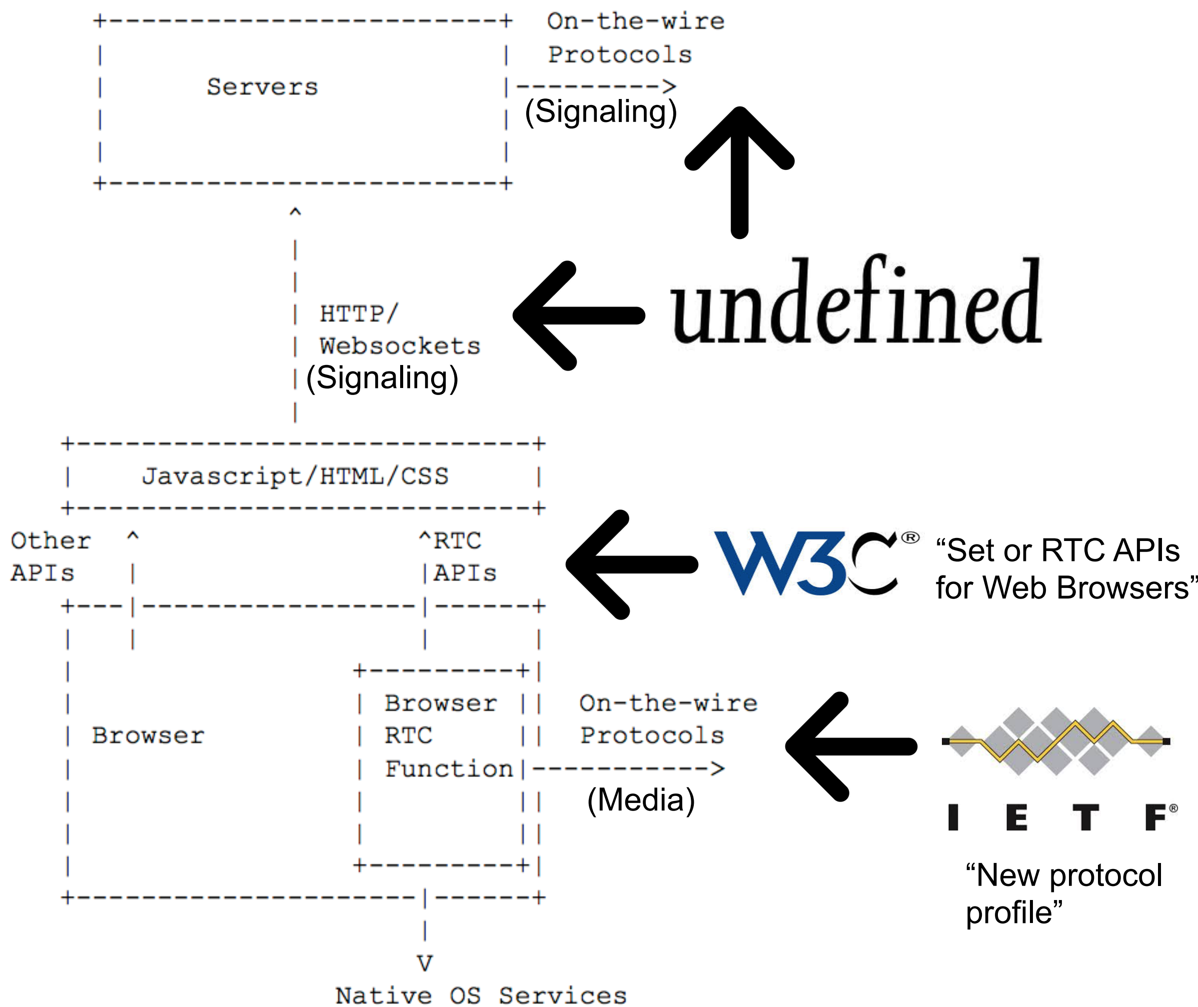
- Chief Strategy Officer (CSO)
- IETF contributor (SIP, Diameter and WebRTC areas)
- IETF STRAW WG co-chair
- SIP Forum WebRTC Task Group co-chair
- WebRTCHacks.com co-founder and blogger
- Independent Expert at European Commission
- Associate Professor at Universitat Pompeu Fabra

What is WebRTC?

A browser-embedded media engine



WebRTC standards



RTCWeb WG (and other)

```
+-----------+                              +-----------+
| Web App   |<--- App-Specific Signaling -->| Web App   |
+-----------+                              +-----------+
      ^                                          ^
      |  SDP                                     |  SDP
      V                                          V
+-----------+                                +-----------+
| Browser   |<----------- Media ------------>| Browser   |
+-----------+                                +-----------+
```

- Audio codecs – G.711, Opus
- Video codecs – H.264 vs. VP8
- Media codecs are negotiated with SDP (for now at least)
- Requires Secure RTP (SRTP) – DTLS-SRTP (SDES is prohibited)
- Requires Peer-2-peer NAT traversal tools (STUN, TURN, ICE) – trickle ICE
- Multiplexing: RTPs & RTP+RTCP
- Tools for firewall traversal
- DataChannel
- Etc.

NEW PROTOCOL PROFILE FOR MEDIA

RTCWeb WG

Working Group Documents:

Document collections: *epub* *mobi*

Draft name	Rev.	Dated	Status	Comments, Issues
Active:				
draft-ietf-rtcweb-alpn	-00	2014-07-23	Active	
draft-ietf-rtcweb-audio	-06	2014-09-05	Active	1/2
draft-ietf-rtcweb-constraints-registry	-00	2014-07-04	Active	
draft-ietf-rtcweb-jsep	-07	2014-07-04	Active	1/1
draft-ietf-rtcweb-overview	-11	2014-08-18	Active	
draft-ietf-rtcweb-rtp-usage	-17	2014-08-25	Active	11/17
draft-ietf-rtcweb-security	-07	2014-07-04	Active	7/7
draft-ietf-rtcweb-security-arch	-10	2014-07-04	Active	2/2
draft-ietf-rtcweb-stun-consent-freshness	-06	2014-08-14	Active	
draft-ietf-rtcweb-transports	-06	2014-08-13	Active	
draft-ietf-rtcweb-video	-00	2014-07-01	Active	
IESG Processing:				
draft-ietf-rtcweb-data-channel	-11	2014-07-04	AD Evaluation	1/1
draft-ietf-rtcweb-data-protocol	-07	2014-07-04	AD Evaluation	
draft-ietf-rtcweb-use-cases-and-requirements	-14	2014-02-12	IESG Evaluation::Revised I-D Needed	2/3
Replaced, Dead or Unknown:				
draft-ietf-rtcweb-qos	-00	2012-10-15	Replaced by draft-dhesikan-tsvwg-rtcweb-qos	

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WebRTC Doesn't Define Signaling

Don't panic, it's not a bad thing!

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Signaling Plane

- WebRTC has no defined signaling method. JavaScript app downloaded from web server. Popular choices are:
  - SIP over Websockets
    - Standard mechanism (RFC7118)
    - Extend SIP directly into the browser by embedding a SIP stack directly into the webpage – typically based on JavaScript
    - WebSocket create a full-duplex channel right from the web browser
    - Popular examples are jsSIP, sip-js, QoffeeSIP, or sipML5
  - Call Control API
    - proprietary signaling scheme based on more traditional web tools and techniques
    - “standard” APIs enhanced to include WebRTC support
  - Other alternatives based on XMPP, JSON or foobar

Some discussion on the topic: http://webrtchacks.com/signalling-options-for-webrtc-applications/

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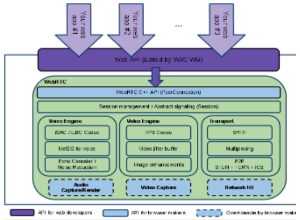
(1/3)

each deployment/vendor is implementing its own **proprietary** signaling mechanism

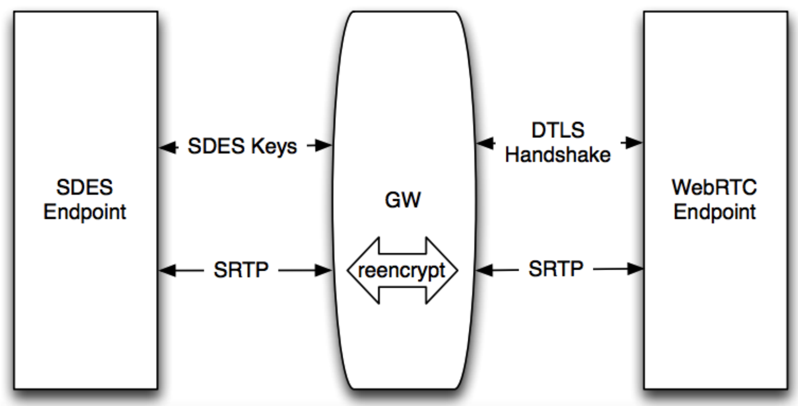
Interworking Towards Legacy?



- A browser-embedded media engine
  - Best-of-breed echo canceler
  - Video jitter buffer, image enhancer
  - Audio codecs – G.711, Opus are MTI
  - Video codecs – H.264 vs. VP8 (MTI TBD - IPR discussion)
  - Media codecs are negotiated with SDP (for now at least)
  - Requires Secure RTP (SRTP) – DTLS
  - Requires Peer-2-peer NAT traversal tools (STUN, TURN, ICE) – trickle ICE
  - Multiplexing: RTPs & RTP+RTCP
- Yes, your favorite SIP client implementation is compatible with most of this. But, the vast majority of deployments
  - Use plain RTP (and SDES if encrypted at all)
  - Do not support STUN/TURN/ICE
  - Do not support multiplexing (ok, not really an issue)
  - Use different codecs that might not be supported on the WebRTC side

TAKEAWAY (2/3)

WebRTC signaling and media is NOT compatible with **existing** VoIP/IMS deployments – gateways are required to bridge the two worlds



The Video Codec Battle

Some discussion on the topic: http://webrtchacks.com/cisco-openh264/

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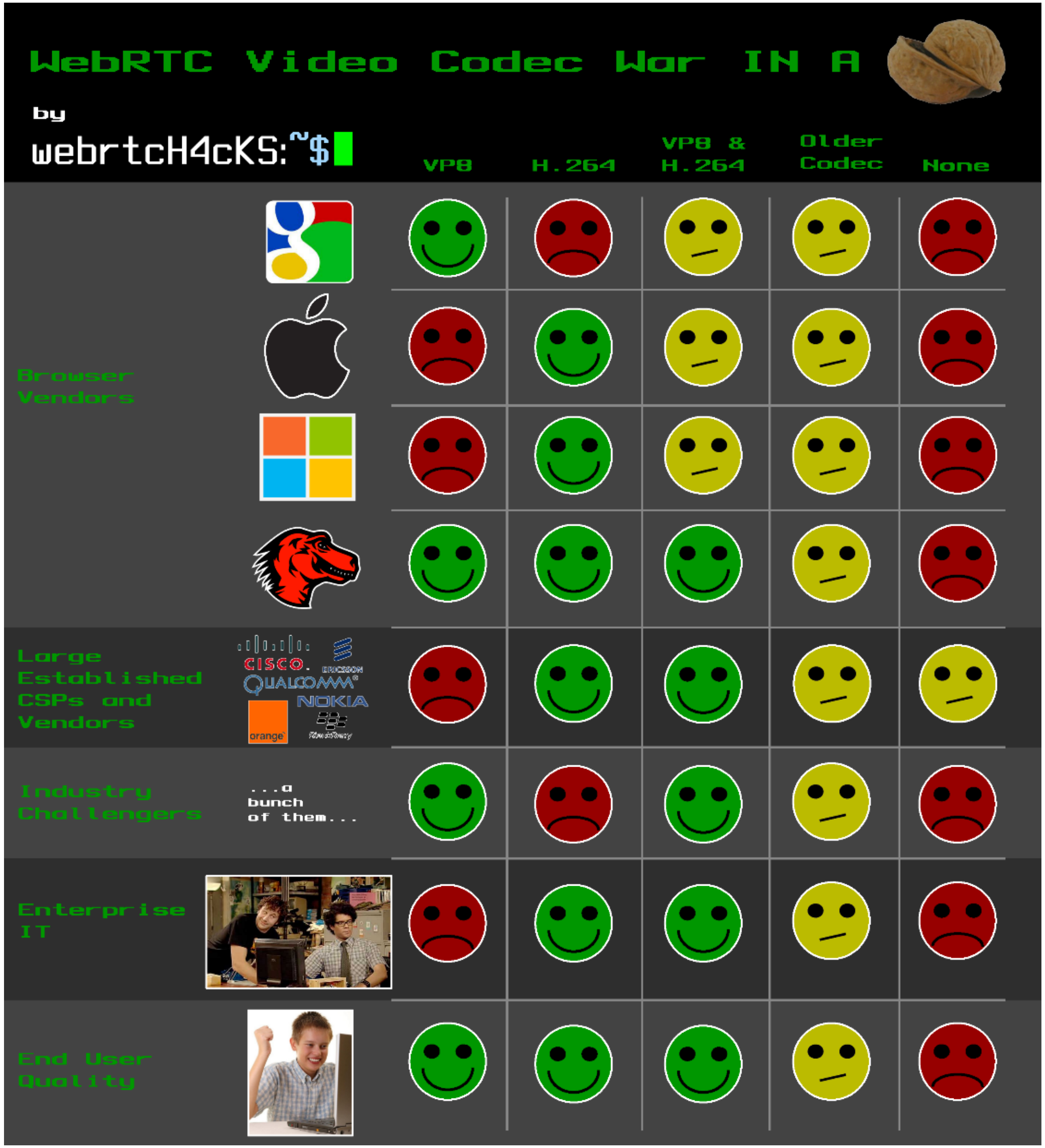
WebRTC Video Codec War IN A
by
webrtcH4cKS:~$
VP8
H.264
VP8 & H.264
Older Codec
None
Browser Vendors
Large Established CSPs and Vendors
CISCO
ERICSSON
QUALCOMM
NOKIA
orange
Industry Challengers
...a bunch of them...
Enterprise IT
End User Quality

Result of The Discussion?

Room participants: 30/50 in favor of H.264
Remote participants (minority): 75/25 in favor of VP8
→ No clear consensus

No decision

Some discussion on the topic: http://webrtchacks.com/ietf-finally-made-decision-mandatory-implement-mti-video-codec-webrtc/

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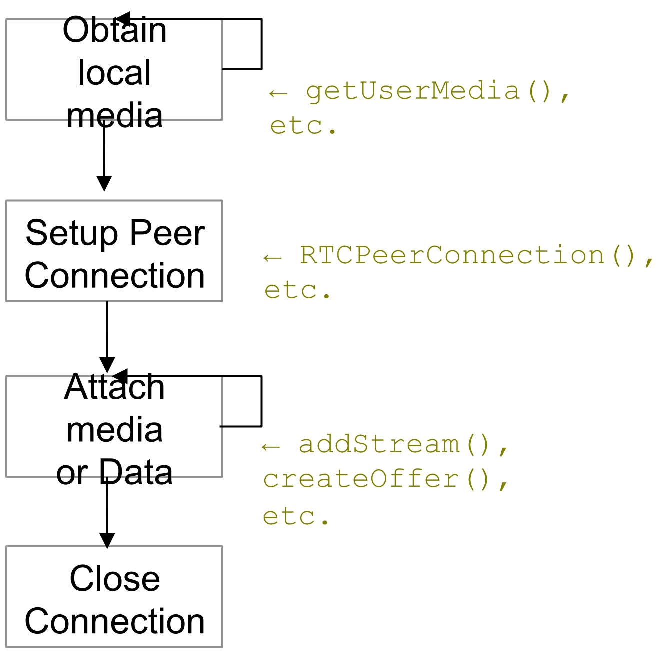
WebRTC WG

"The mission of the W3C WebRTC WG is to define client-side APIs to enable Real-Time Communications in Web-browsers. These APIs should enable building applications that can be run inside a browser, requiring no extra downloads or plugins, that allow communication between parties using audio, video and supplementary real-time communication, without having to use intervening servers (unless needed for firewall traversal)."



<u>Discussion:</u> provides the current API in its form (e.g. based on SDP O/A) the flexibility Web developers need?

<u>Answer:</u> well, not really but it's good enough for most of the use cases we have today

<u>Alternative proposals:</u> Microsoft's CU-RTC-WEB (Aug'12), WebRTC Object API (ORTC) (Aug'13)

<u>Next step:</u> "Done is better than perfect", Let's finish WebRTC 1.0, Let the industry adopt it

<u>Future work:</u> "fix/improve things in WebRTC 2.0", Backward interoperability?

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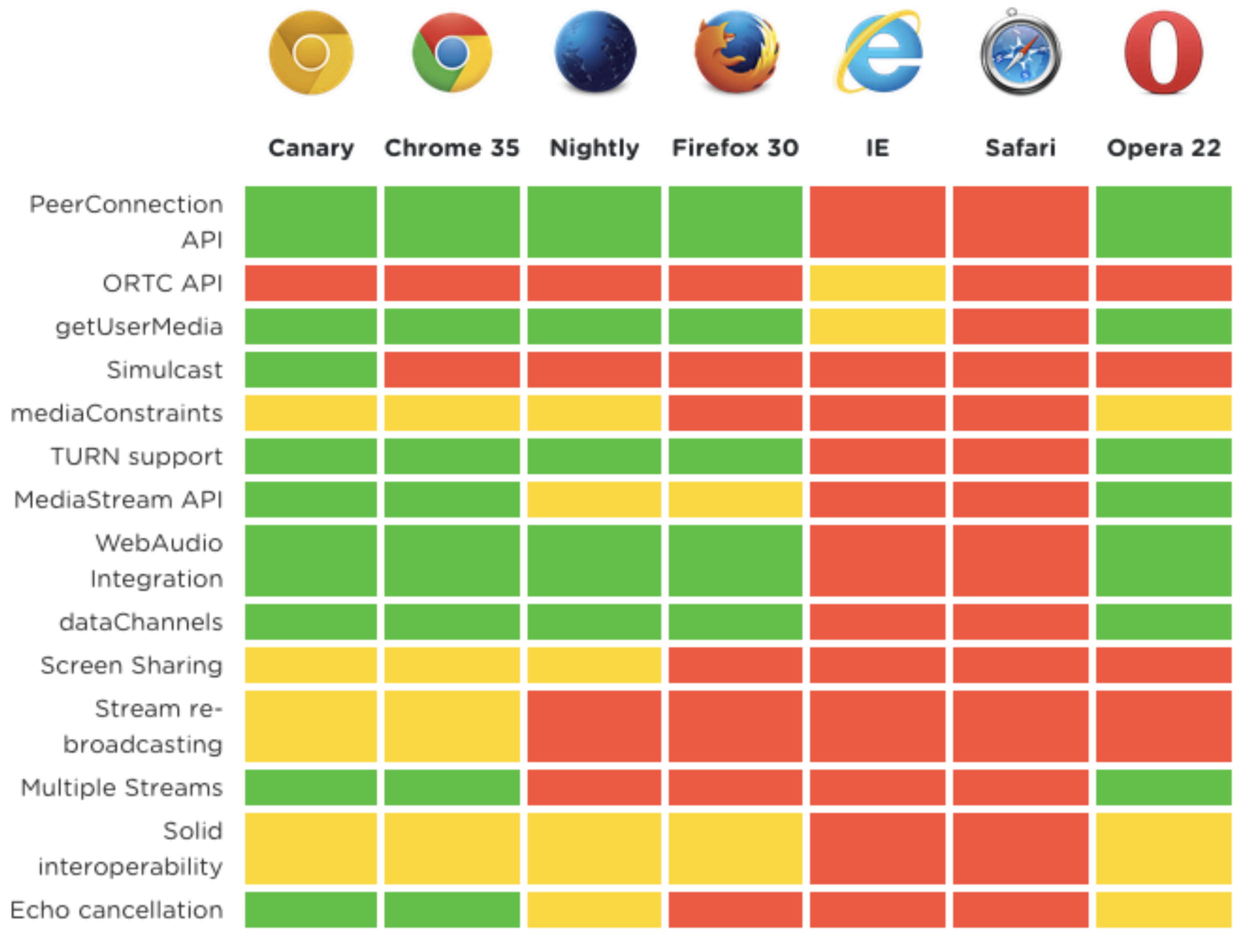
Browser Support



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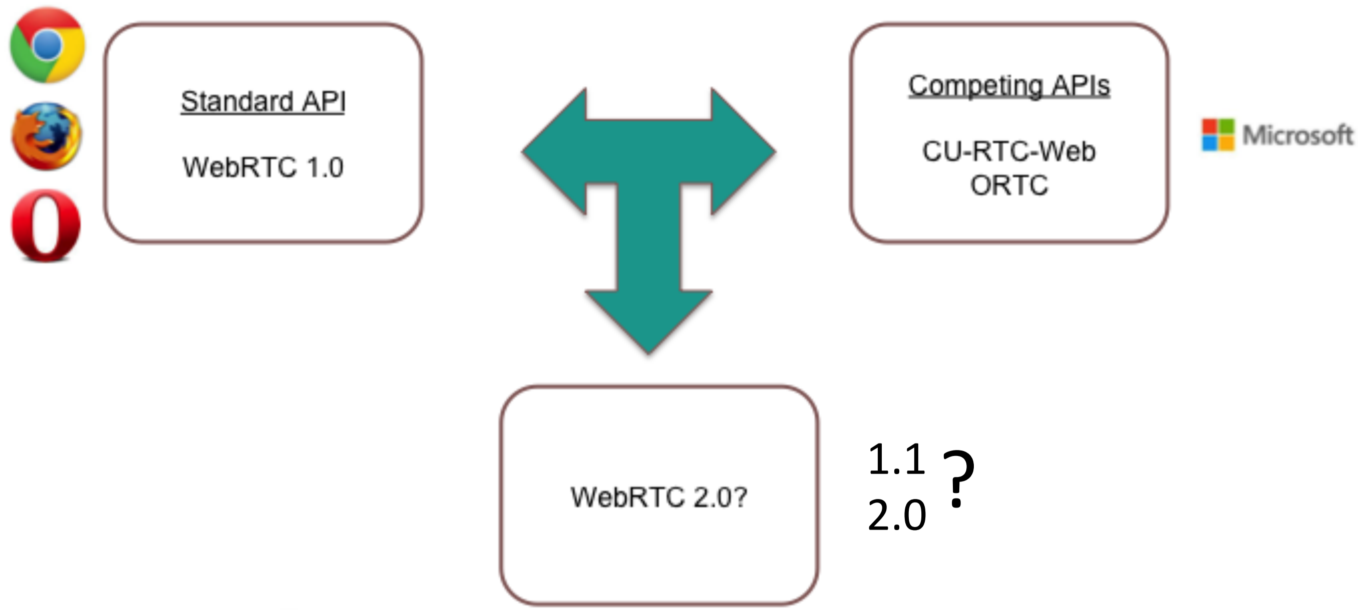
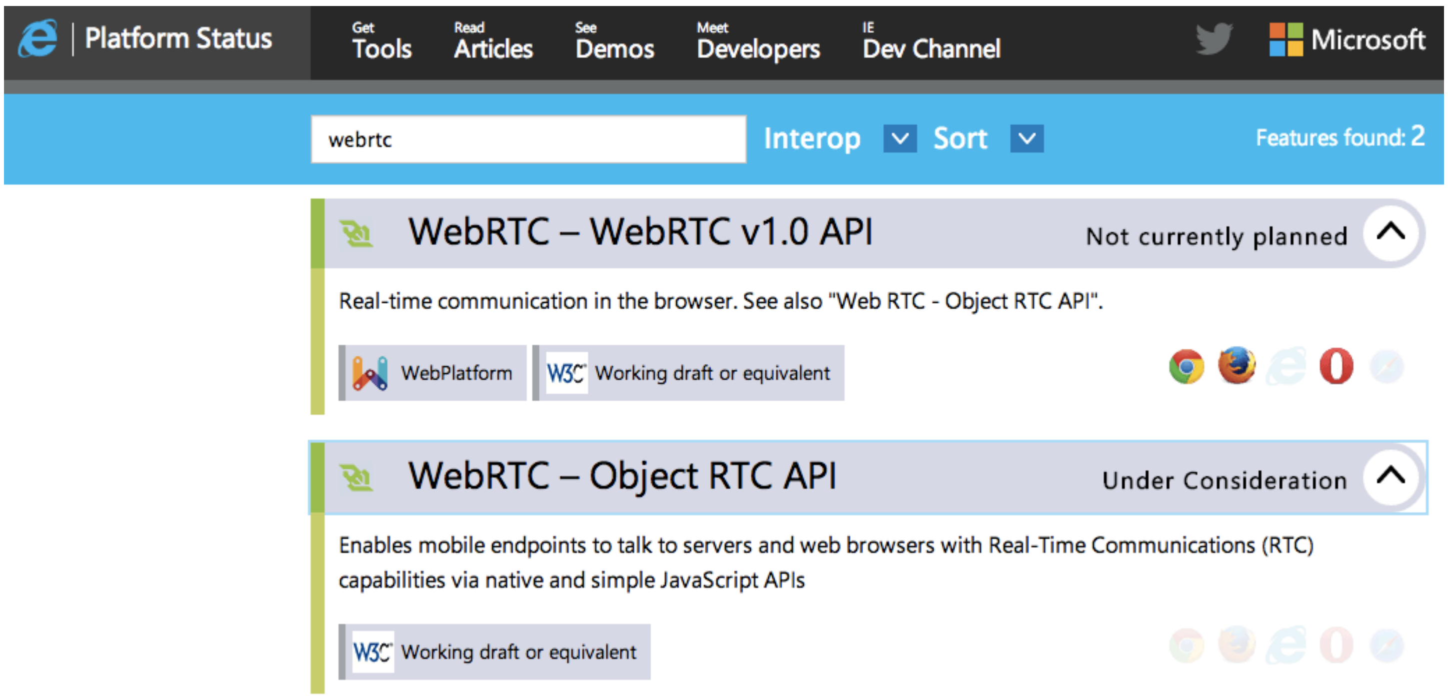
iswebrtcreadyyet.com

Browser API



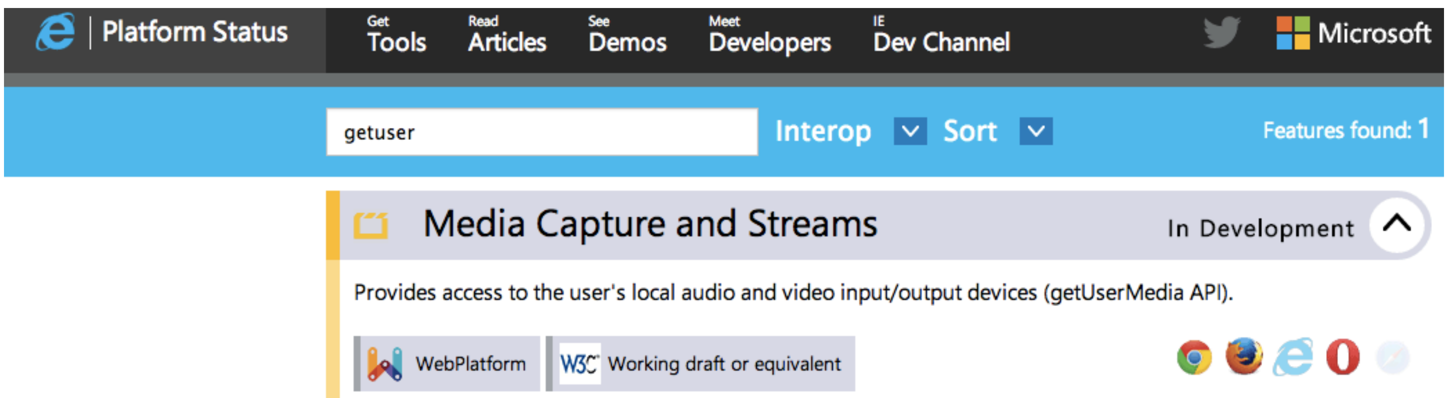
Some discussion on the topic: http://webrtchacks.com/why-the-webrtc-api-has-it-wrong-interview-with-webrtc-object-api-ortc-co-author-inaki-baz-3-2/

Browser API



http://status.modern.ie/

Browser API



http://status.modern.ie/

Browser API

Microsoft
Open Technologies

Building bridges between Microsoft and non-Microsoft technologies.
Powering interoperability through open standards and open source.

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Updated ORTC API Prototype Released

Posted on April 25, 2014 by Adalberto Foresti

MS Open Tech delivers updated ORTC API prototype, reflecting recent progress within the W3C ORTC Community Group

The W3C's ORTC Community Group (CG) recently published an updated editor's draft of the ORTC specification, available here. This version of the specification includes some significant enhancements compared with the original version supported in the original MS Open Tech prototype released last fall. The improvements include support for simulcast and scalable video coding, as well as enhancements to the object model such as control of senders/receivers as well as communications transport aspects (ICE, DTLS and data channel).

Plug-in free or free plug-in?

Jun 22, 2014 : Singapore

Temasys Plugin Supports WebRTC in Internet Explorer and Apple Safari, on Desktops

By: Nathaniel Currier

Temasys is pleased to announce that it now provides universal support for webRTC video and audio across all major desktop web browsers, including Microsoft's Internet Explorer and Apple's Safari web browsers.

The Temasys plugin will let any webRTC-based website work on Microsoft's Internet Explorer and Apple's Safari web browsers. The solution was designed and tested to work with Internet Explorer versions 9, 10, and 11, and current versions of Safari on Microsoft windows, and MacOS X.

easyRTC

Home Features Pricing Documentation About Priologic News

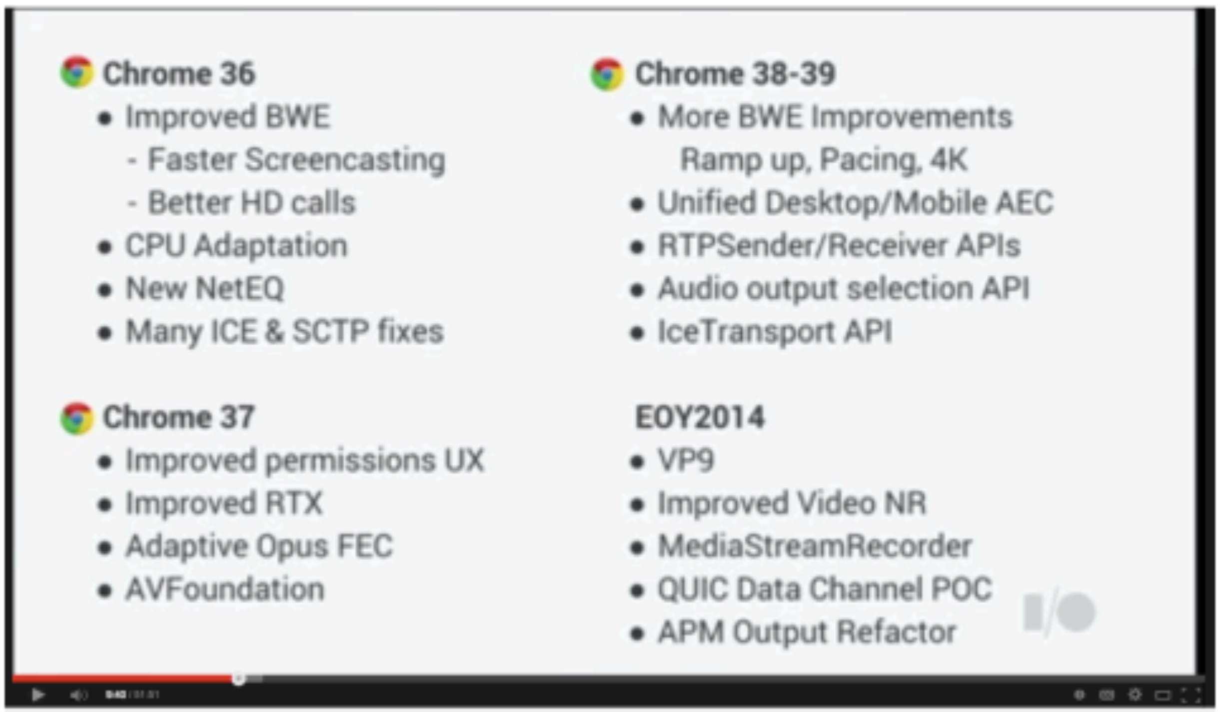
Priologic Releases First Open Source WebRTC Plugin for Internet Explorer

Published on: June 10, 2014

Priologic will release the first-ever open source WebRTC plugin for IE prior to WebRTC IV in Atlanta on June 17-19, 2014. This new IE plugin will be released under a BSD2 license to allow any WebRTC team to use it and improve on itwithin their commercial products or as open source.

Google Chrome 38-39 to ship with ORTC / WebRTC 1.1 APIs

For those who missed it, Chrome 38-39 looks like it will be shipping with ORTC 1.1 RTPSender / Receiver APIs as announced by Justin Uberti at Google I/O 2014. This really should not come as any surprise as RTPSender / Receiver APIs are now on track for WebRTC 1.0 integration as well, as per the last W3C WebRTC WG Interim meeting.



http://blog.webrtc.is/2014/07/01/google-chrome-38-39-to-ship-with-ortc-webrtc-1-1-apis/

Posted by *Chad Hart* on *September 10, 2014* Edit This

webrtcH4cKS: ~ ORTC is not the "Other" RTC: Q&A with ORTC CG Chair Robin Raymond

Posted in: Standards. Tagged: CU-RTC-Web, Hookflash, ORTC, Q&A, robin raymond. 3 comments

Biggie vs. Tupac. Gates vs. Jobs. Apple vs. Samsung. Nothing catches people's attention for no legitimate reason like a feud. Unfortunately this isn't just a celebrity phenomenon. Feuds have been endemic even to real communications as well. From the very beginning, Elisha Gray's dispute with Alexander Graham Bell over the original telephone patent showed the industry has a propensity for squabbles. Unfortunately we have become so accustomed to feuds that we sometimes fabricate battles that do not really exist. I fear that this is often the case with one of the most important, but misunderstood efforts affecting WebRTC's future – Object Real Time Communications (ORTC).

http://webrtchacks.com/ortc-qa-robin-raymond/

(3/3)

the WebRTC API can have **different flavors**

WebRTC Access to IMS (r12)

3GPP TR 23.701 V12.0.0 (2013-12)

Technical Report

3rd Generation Partnership Project;
Technical Specification Group Services and System Aspects;
Study on Web Real Time Communication (WebRTC) access to
IP Multimedia Subsystem (IMS);
Stage 2
(Release 12)

Adding New Wheels to IMS with WebRTC

3GPP TS 23.228 V12.5.0 (2014-06)

Annex U (Normative): WebRTC access to IMS - network-based architecture

U.1 Overview

U.1.0 General

Web Real-Time Communication (WebRTC) is specified in IETF Draft, draft-ietf-rtcweb-overview-08 [84] and WebRTC 1.0 [85]. This Annex specifies a network-based architecture for the support of WebRTC client's access to IMS.

NOTE: The UE can also perform WebRTC access to IMS by implementation specific means in the UE in which it exposes a standard Gm interface towards IMS.

U.1.1 Assumptions

- This Release specifies an option to use a signalling interface from the UE to the network based on SIP over WebSocket (RFC 7118 [89]), which is used as the information model which may be used by other options. Options other than SIP over WebSocket, such as XMPP or other application protocols over WebSocket, a RESTful based interface, etc. are allowed but not described. Alternative message body formats such as JSON and alternative transport protocols are also not precluded. Any enhancements required to accommodate an unspecified signalling interface are considered compliant to the Release as long as other defined interfaces in the architecture are not impacted.
- SDP offer/answer exchange is the mechanism used for media plane feature negotiation.

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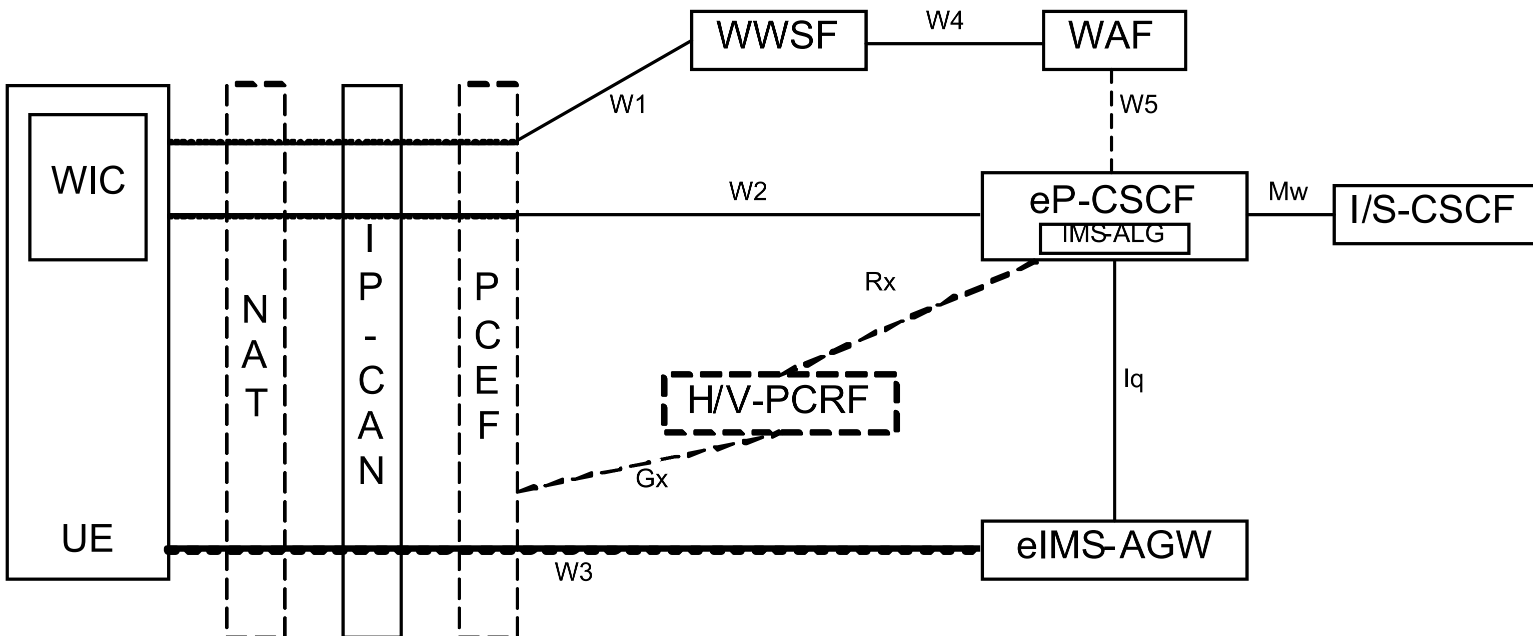
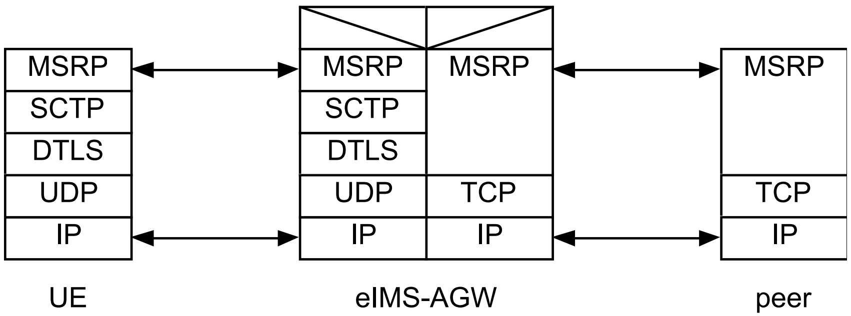
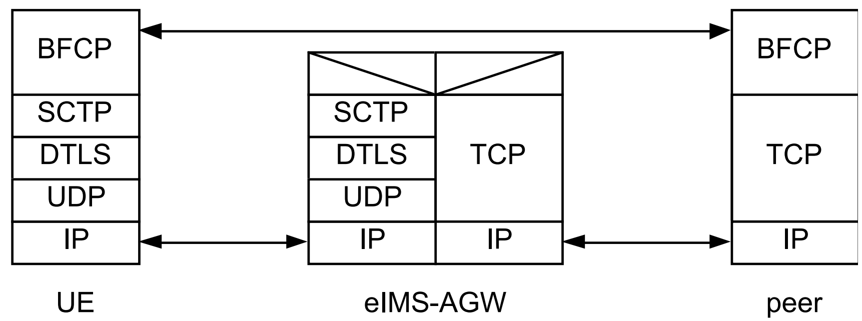
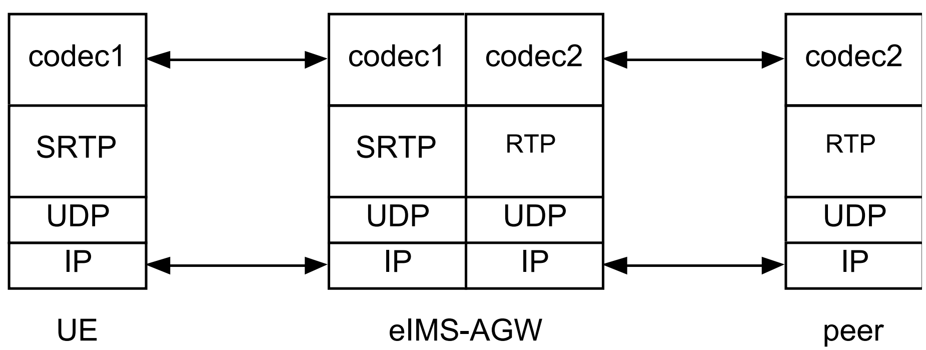
Reference Architecture



Interworking Towards Legacy IMS







SIP Forum WebRTC Task Group

SIP FORUM

"the initial focus of the Task Group is to determine what the needs are for successful interoperability of WebRTC-to-SIP deployments" covering both Enterprises and Service Providers

"recommendations, Reference Architecture Documents, Certifications, and/or White Papers"

Alliance for Telecom Solutions

atis

DEVICE SOLUTIONS INITIATIVE

Featured Project

ORCAjs

What is the DSI?

- The DSI develops solutions that expose the value of the network to devices and applications by reducing complexity in network APIs.
- These solutions provide application developers with network-agnostic, client-side APIs that simplify access to proprietary network APIs.
- DSI members identify and prioritize projects to bring together open source innovation and fully interoperable standards to enhance network value.

Participating Companies

Alcatel·Lucent, at&t, CenturyLink, ERICSSON, LG Electronics, Quobis, Sprint, verizon

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WebRTC Interop Activity Group

“focuses on interoperability issues relating to the use of WebRTC”

“the group is focused on enterprise WebRTC , interworking of WebRTC and other carrier technologies, and other existing videoconferencing systems”

“develop an interoperability test framework and prepare for IOT events”

GSMA

How does WebRTC relate to VoLTE and RCS?

Summary

- each deployment/vendor is implementing its own **proprietary** signaling mechanism
- WebRTC signaling and media is incompatible with **existing** VoIP deployments – gateways are required to bridge the two worlds
- the WebRTC API can have **different flavors**

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Victor.pascual@quobis.com
@victorpascual

Thank You!