**WebRTC Conference, 12-14 November 2013**

*The first real-world deployments*

WebRTC has emerged as a potentially disruptive communication technology. The ongoing work being done by the IETF & W3C brings millions of web application developers and application users into the world of real-time IP communications.

Indeed, WebRTC enables web applications to leverage web browsers to make real-time voice and video connections to other WebRTC devices or traditional VoIP and video devices through mediation elements that can interface WebRTC with the existing infrastructure.

However, voice services have always been tricky to implement and deploy. Conference participants at the next WebRTC Conference in Paris will witness key items such as identity, quality of service of the media and regulation issues being deliberated during the conference.

The other important discussion would center on the critical points related to the interworking between WebRTC and the “PSTN” world (call-control protocol, encryption of the media, codecs, browser wake-up).

The 4G/LTE Promises

 Of interest to 4G network operators, WebRTC can extend the reach of their services to more end points.  Of equal interest, the mobile device application developer will find WebRTC as a means of bridging their app across mobiles and web browsers. Introduction of WebRTC capabilities to smartphones opens up great opportunities for new applications and services, but to get there, a number of technical challenges need to be overcome.

OTT’s and Enterprises

OTT VoIP players don‘t have to wait for standards and can implement new features quickly as they don't rely on any but the most basic functions of the underlying network. This is why they have been disrupting the Telco space for several years now. Question: has WebRTC, as a new standard, the potential of disrupting OTT VoIP players and becoming a real boom for telcos? OTT’s representatives will explain their strategy in this area and exchange with other service providers

Finally, many large enterprises consider WebRTC as a potential cost saving measure for their remote workers (be there home, mobile or other). However, existing enterprise telephony infrastructures do not use or conform to many of the of advanced standards being presented by the current implementations of WebRTC in the browser. ICE, STUN and SRTP are among many problems faced while integrating. Along with this, enterprise networks have existing security and compliance infrastructures that WebRTC solutions must take into account.

Does WebRTC constitute a threat or an opportunity for telco business models?

What are OTT VoIP players’ strategies?

How can WebRTC change enterprise communications?

Can WebRTC be considered as a plain telephony service?

What about scalability, security, confidentiality?

How do WebRTC and SIP-based solutions work together?

During the WebRTC conference, to be held in Paris-Roissy, from 12 to 14 November, 2013, experts will address all technical and standardization issues that still need to be solved, based on the first real-world deployments.

The following list of topics is not exhaustive and authors may propose other subjects in keeping within the thematic framework.

**Updates and Standards and Implementations**

o   Reports from W3C

o   Reports from IETF

o   Browser Implementation Experiences

**Interoperability (Browsers, Gateways)**

o   Cross-browser Interop

o   API Conformances

o   Browser-to-Media Server Interop

**WebRTC Enhancements for Mobile Devices and Networks**

o   IMS Support for WebRTC

o   WebRTC enabled Mobile Devices

o   External Devices for WebRTC (headsets…)

**WebRTC for Enterprise Users**

o   Gateways SIP and WebRTC

o   Call center applications

o   SIP vs. WebRTC

**WebRTC Application Contest**

o   Ideas from the WebRTC developer community

o   Reports from the edge

o   Contest for the most innovative, funny, useful and visionary WebRTC app

**Performance Assessments, Scalability and Quality**

o   Mass-scale deployment of WebRTC devices and services

o   Quality issues with WebRTC

o   QoS

**Identity Management in WebRTC**

o   Security

o   Identity Management

o   Silos and Interconnections

**WebRTC Development Support**

o   Libraries for WebRTC apps

o   Server Frameworks and Extensions

o   IDEs

Abstracts must not exceed one page. They may be submitted in PDF, HTML or Word format by email at: [info@upperside.fr](mailto:info@upperside.fr) or [remi.scavenius@wanadoo.fr](mailto:remi.scavenius@wanadoo.fr)   
  
DEADLINE   
  
Deadline for turning in abstracts: February 28, 2013  
Feedback from committee members: April 15, 2013