WebRTC

Open Standardized Media Stack and its Implementations

above

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WebRTC is an API definition drafted by the World Wide Web Consortium (W3C) that supports applications for voice calling, video calling, and P2P file sharing without the need of either internal or external plugins.

WebRTC Players

Following players have been instrumental in standardization of protocols and API for browser based real time communication

W3C®

World Wide Web Consortium

Drafter the WebRTC API for browsers

I E T F®

Internet Engineering Task Force

Standardize protocol for WebRTC

Google

Google

Support and open sourcing WebRTC codecs

mozilla

Firefox

Support and open sourcing WebRTC codecs

OPERA™ software

Opera

Adopting WebRTC in browser

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Why is WebRTC important ?

- WebRTC video quality is noticeably better even in low bandwidth conditions.
- Up to 6x faster connection times than traditional centralized media architectures
- Reduced audio/video latency
- No need for Flash or other plugins

- Native HTML5 elements and Javascript

Background and adoption of WebRTC

2011

- Google bought On2 and GIPS
- Open source technologies for codec and echo cancellation.
- Ericsson builds 1st WebRTC

2013

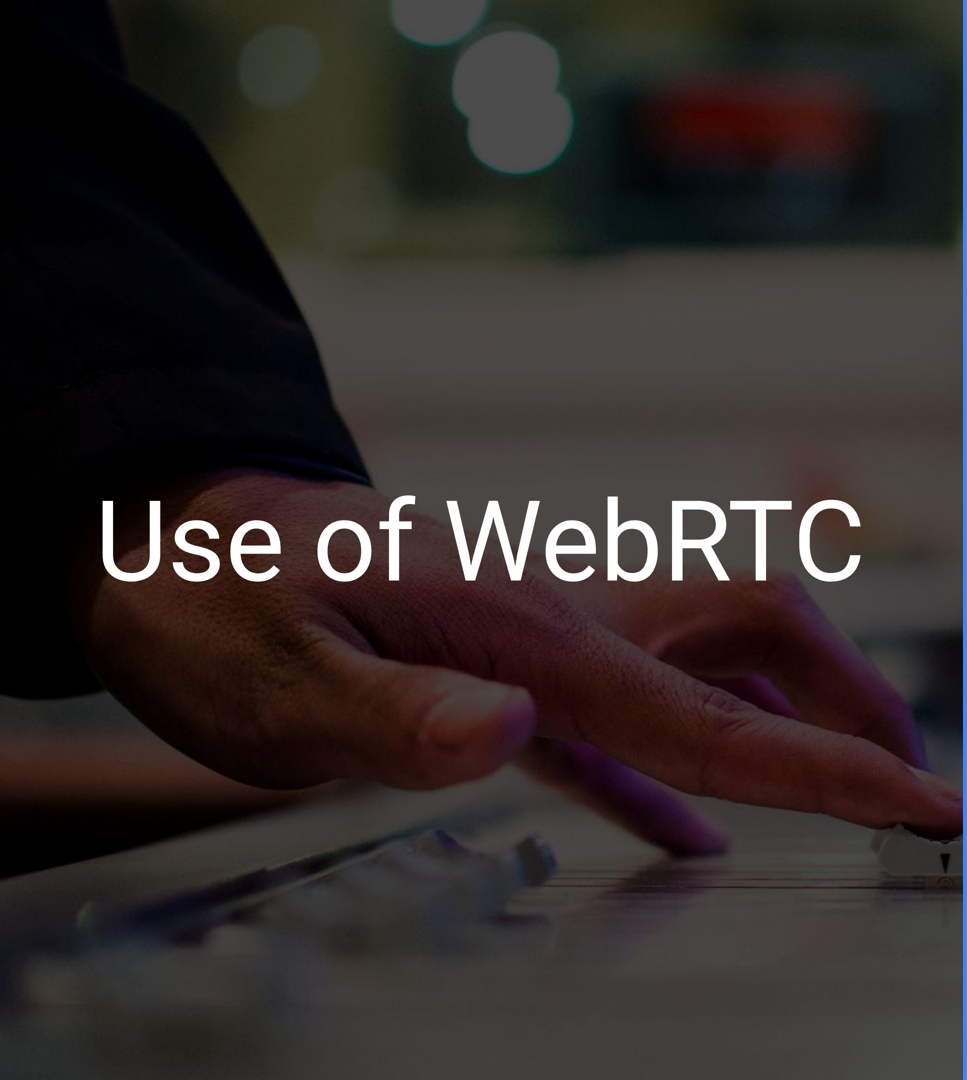
- Chrome 23 and Firefox 20 have WebRTC
- Chrome and Firefox for Android also support WebRTC

2014

- Opera Adds WebRTC
- Security Issues are addressed with https

2015

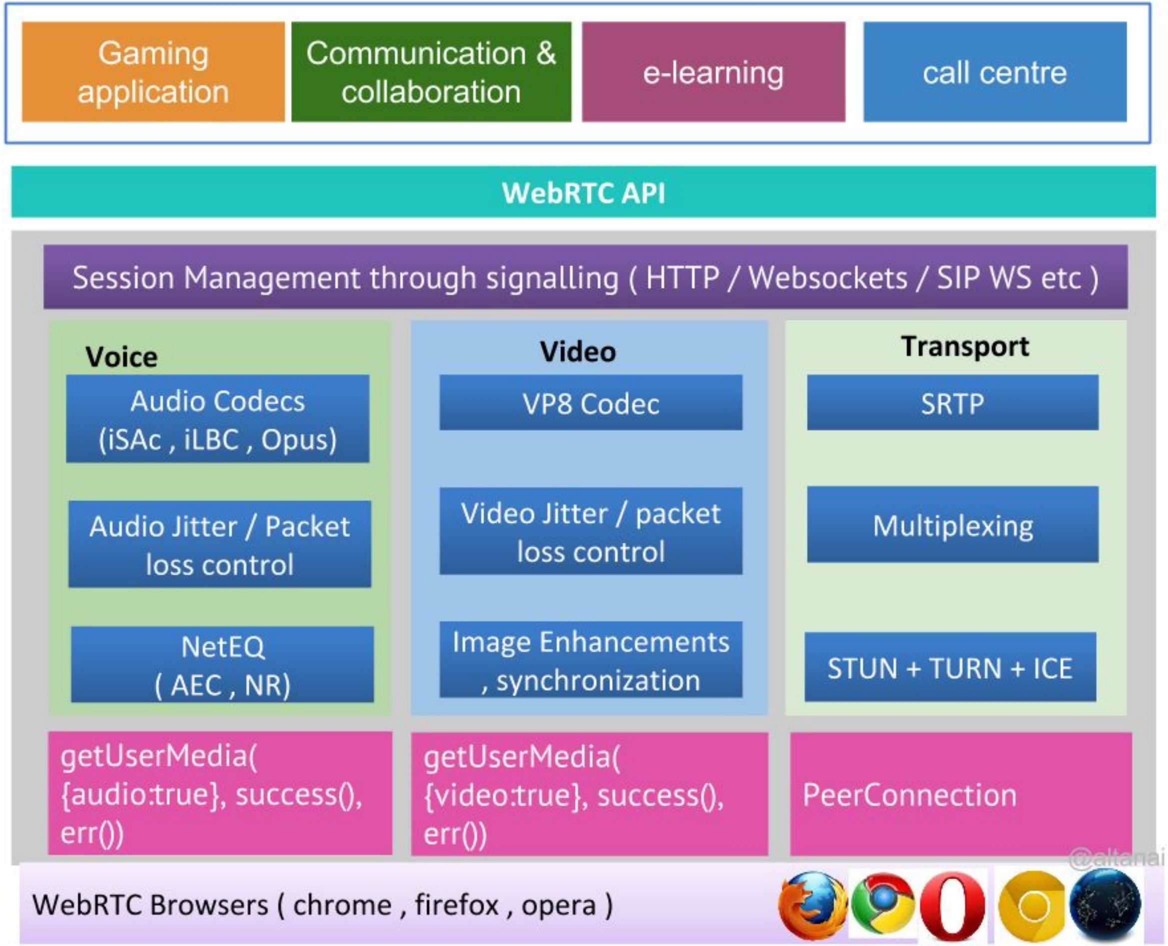
- Microsoft adds ORTC
- Facebook declares WebRTC adoption



Use of WebRTC

Enables modern web apps to capture or stream media streams and multimedia to/from its users through browser in p2p fashion.

WebRTC media Stack



WebRTC Media Stack

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WebRTC APIs

```
getUserMedia( {
      Audio: true
}, success(), err())
```

```
getUserMedia({
      Video : true
}, success(), err())
```

PeerConnection

WebRTC implements three APIs:

- getUserMedia : capture Media Stream
- RTCPeerConnection : peerconnection
- RTCDataChannel : arbitrary data exchange with low latency and high throughput

Voice engine

iSAC: wideband and super wideband audio codec for streaming audio

iLBC: narrowband speech codec for streaming audio

Opus: constant and variable bitrate encoding

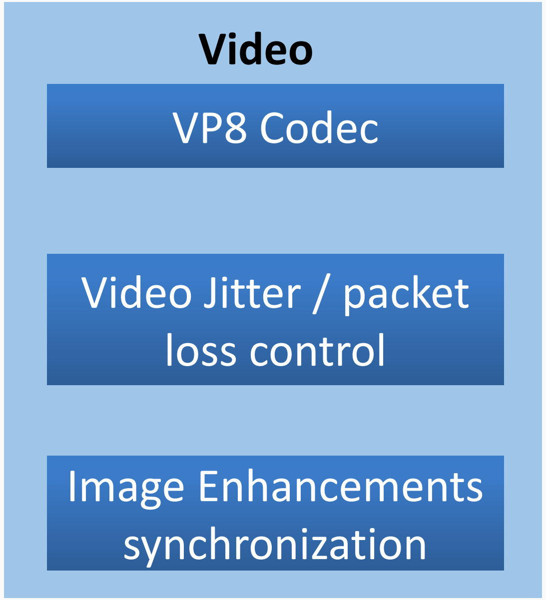
NetEQ: Net Equalizer

Dynamic jitter buffer + error concealment algorithm

Acoustic Echo Canceler (AEC) : remove acoustic echo

Noise Reduction (NR) : remove background noise

Video engine



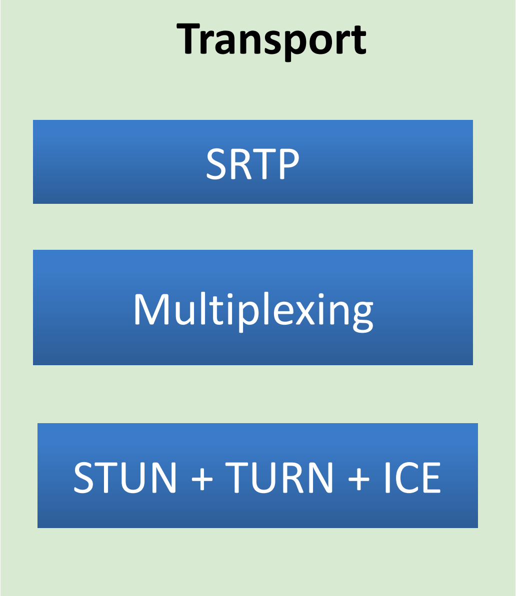
VideoEngine is a framework video media chain for video, from camera to the network, and from network to the screen.

VP8 : Video codec from the WebM Project. Designed for low latency Real time Comm.

Video Jitter Buffer: conceal the effects of jitter and packet loss on overall video quality.

Image enhancements : removes video noise

Transport



Transport / Session Layer of WebRTC stack provide Session Management for WebRTC media streams .

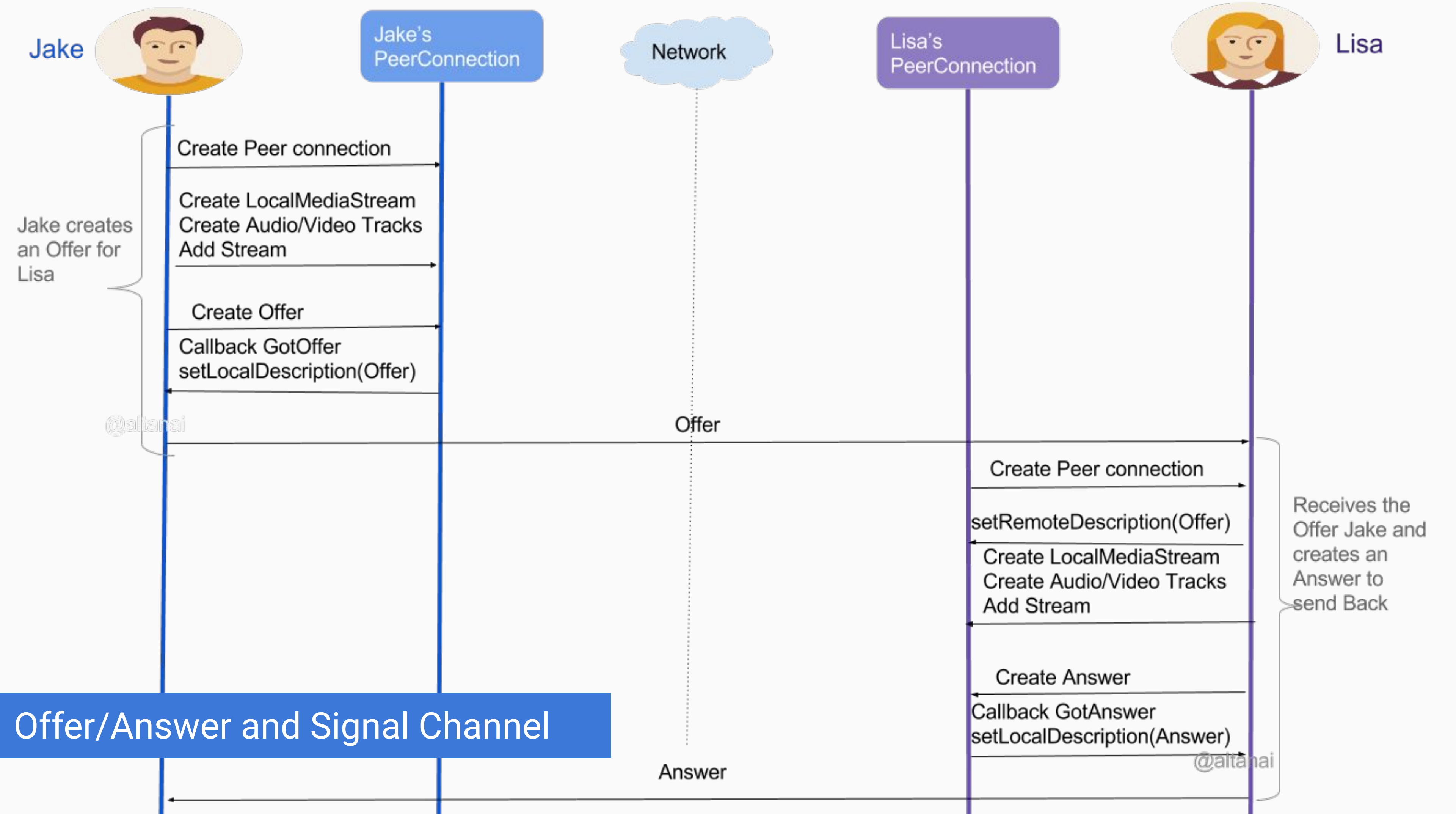
It consists of network stack for Secure RTP, the Real Time Protocol.

STUN/ICE for NAT , Network Address Traversal across various types of networks.

Session Management which is an abstracted session layer for call setup.

```
type: offer, sdp: v=0
m=audio 9 UDP/TLS/RTP/SAVPF 111 103 104 9 0 8 126
c=IN IP4 0.0.0.0
a=rtcp:9 IN IP4 0.0.0.0
a=sendrecv
a=rtcp-mux
a=rtpmap:111 opus/48000/2
a=fmtp:111 minptime=10;useinbandfec=1; stereo=1; sprop-stereo=1;
maxaveragebitrate=1048576; maxplaybackrate=1048576
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
...
m=video 9 UDP/TLS/RTP/SAVPF 101 100 107 116 117 96 97 99 98
c=IN IP4 0.0.0.0
a=rtcp:9 IN IP4 0.0.0.0
a=sendrecv
a=rtcp-mux
a=rtpmap:100 VP8/90000
a=rtpmap:101 VP9/90000
a=rtpmap:107 H264/90000
a=fmtp:107 level-asymmetry-allowed=1;packetization-mode=1;profile-level-id=42e01f
a=fmtp:100 x-google-min-bitrate=4194304; x-google-max-bitrate=4194304
...
m=application 9 DTLS/SCTP 5000
c=IN IP4 0.0.0.0
a=sctpmap:5000 webrtc-datachannel 1024
```

SDP exchange before streaming media

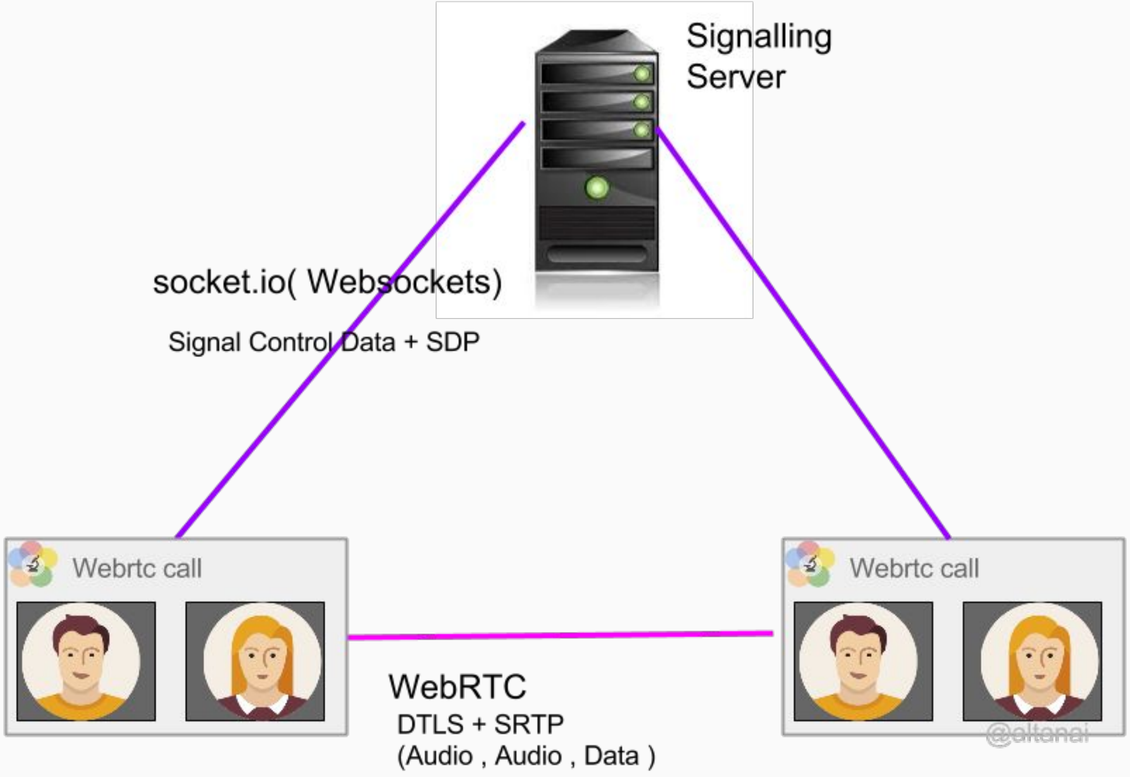


Offer/Answer and Signal Channel

Time	Event
24/08/2016, 11:08:44	▶ addStream
24/08/2016, 11:08:44	▶ createLocalDataChannel
24/08/2016, 11:08:44	▶ createOffer
24/08/2016, 11:08:44	onRenegotiationNeeded
24/08/2016, 11:08:44	onRenegotiationNeeded
24/08/2016, 11:08:44	▶ createOfferOnSuccess
24/08/2016, 11:08:44	▶ setLocalDescription
24/08/2016, 11:08:44	▶ signalingStateChange
24/08/2016, 11:08:44	setLocalDescriptionOnSuccess
24/08/2016, 11:08:44	▶ iceGatheringStateChange
24/08/2016, 11:08:44	▶ onIceCandidate
24/08/2016, 11:08:45	▶ setRemoteDescription
24/08/2016, 11:08:45	▶ signalingStateChange
24/08/2016, 11:08:45	▶ iceConnectionStateChange
24/08/2016, 11:08:45	▶ onAddStream
24/08/2016, 11:08:45	setRemoteDescriptionOnSuccess
24/08/2016, 11:08:45	▶ iceGatheringStateChange
24/08/2016, 11:08:45	▶ iceConnectionStateChange
24/08/2016, 11:08:45	▶ iceConnectionStateChange
24/08/2016, 11:08:46	▶ addIceCandidate
24/08/2016, 11:08:50	▶ iceConnectionStateChange

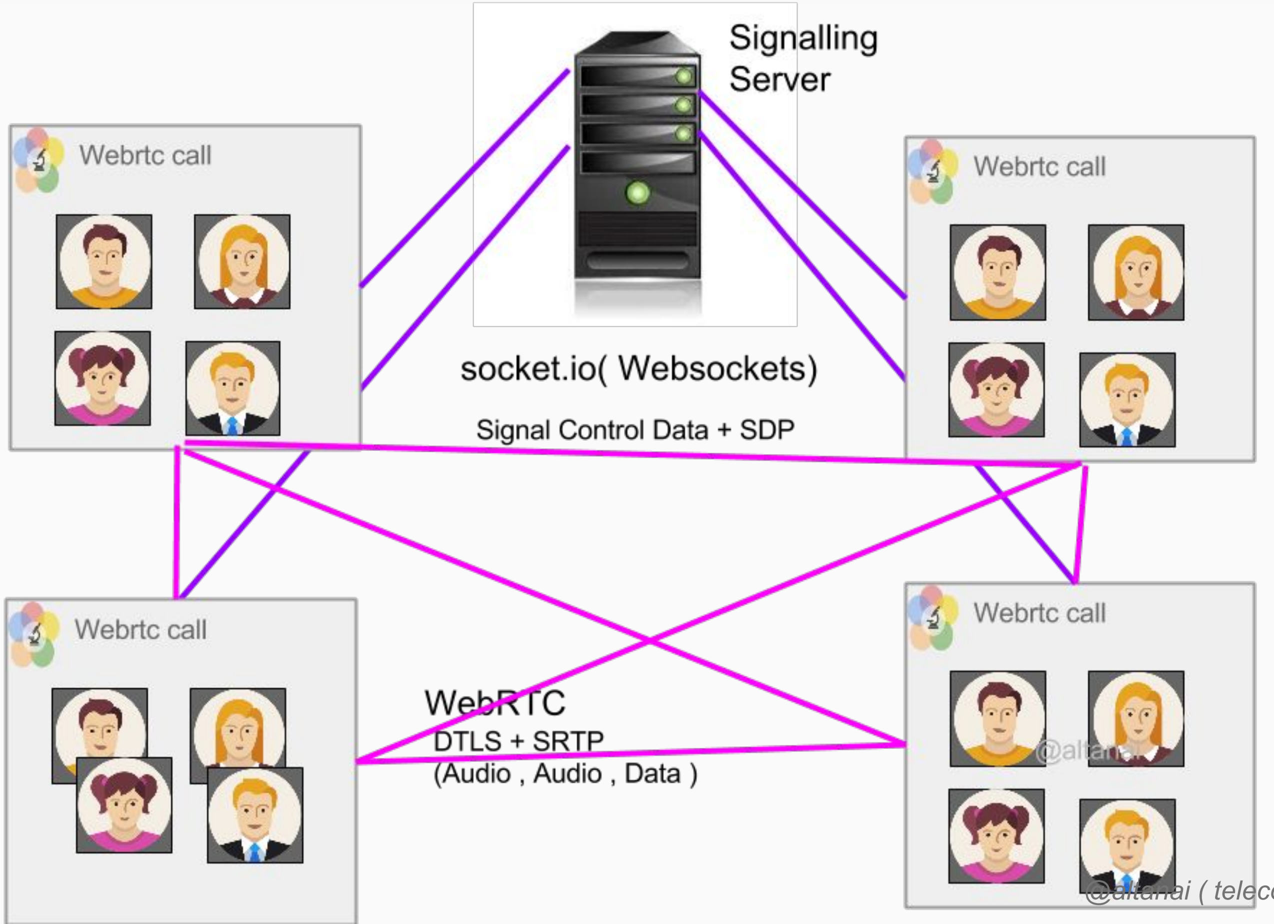
chrome://webrtc-internals/ traces

Basic Scheme for WebRTC based Media Streaming across Network



Mesh based Networking

WebRTC is designed for p2p communication. It makes a mesh network when peers are more in number



WebRTC Solutions

- Pluggable module or npm
- Collaboration as a Service ie CaaS
- Communication Platform

Tango FX

https://tfxserver.above-inc.com/

Basics for building a WebRTC based communication solution

Requirements :

- Websockets for signalling / Offer Answer
- TURN server like xirsys
- Js library for WebRTC wrappers
- Https served webpage
- WebRTC enabled Browser

Milestones in Building WebRTC solution

Mapping a 12 Week (3 months) timeline for development of full fledged WebRTC solution

1 2 3 4 5 6 7 8 9 10 11 12

POC stage

Define and experiment with available WebRTC

Set up Infrastructure & platform

Create nodejs based platform such as Express, Mongo , React etc for building codebase . TURN servers

Stack for Signalling

Signalling helps in offer answer model and policy checks

Third Party Integrations and APIs

WebRTC works best when integrated with 3rd party services for Calendar , Email Notification , Oautth etc

Testbeds

Automation tests help ensure the scalability and resillence of solution

Deploy

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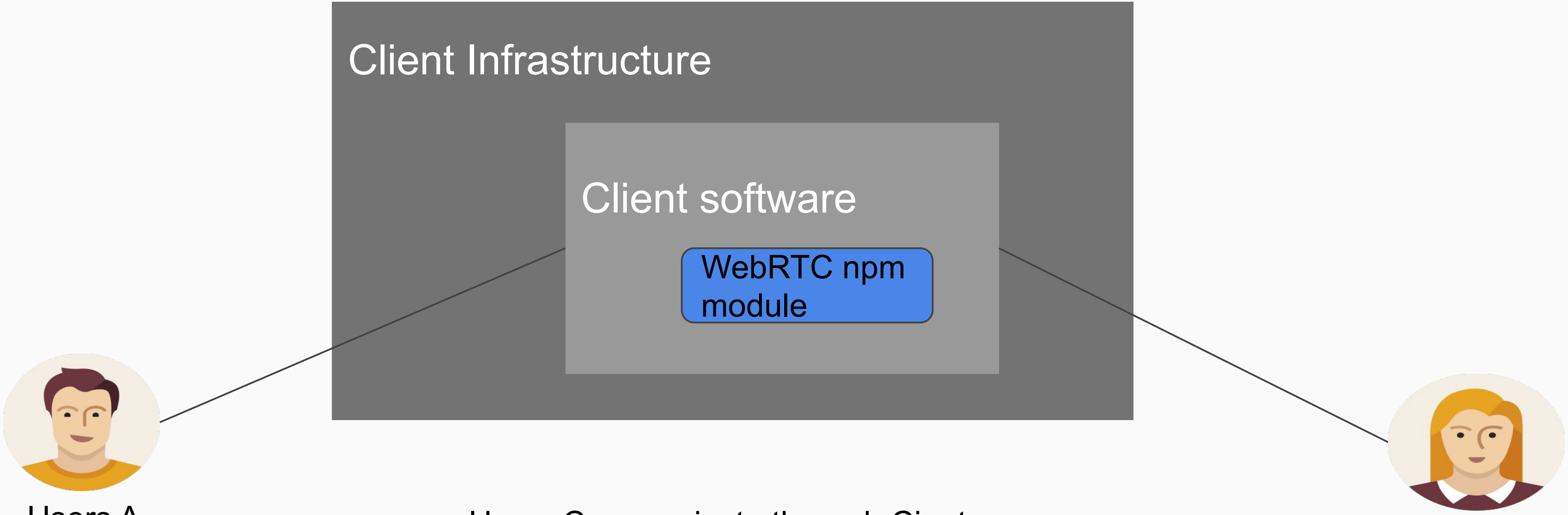
● Pluggable module or npm

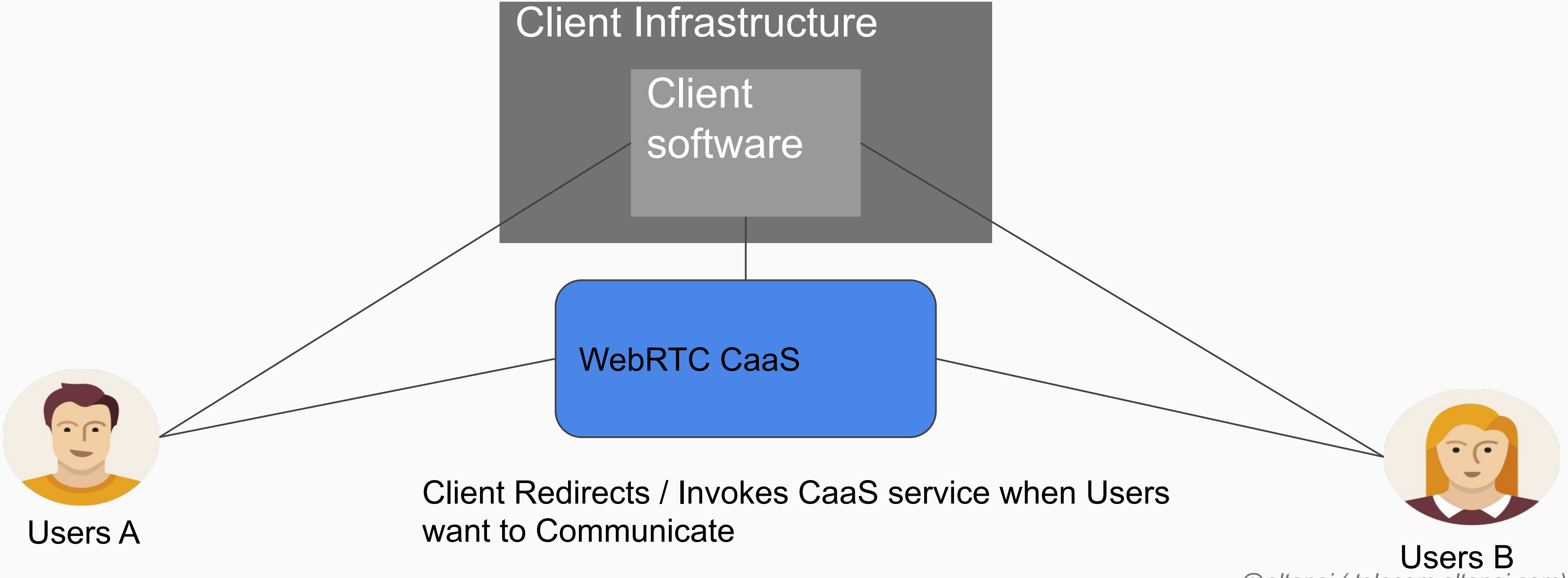
Source code for WebRTC project is shipped as a pluggable library or npm module.



Users Communicate through Cient Environment

- collaboration as a Service ie CaaS

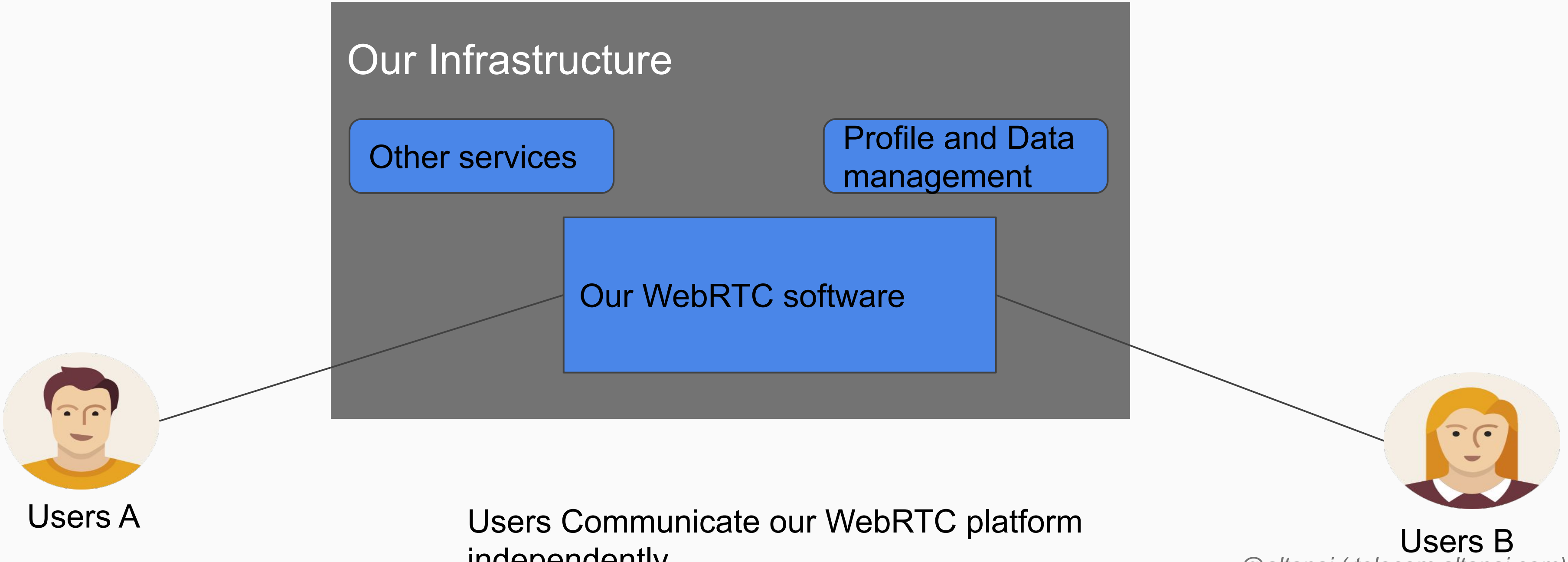
Clients redirect users to our WebRTC platform for communication



Client Redirects / Invokes CaaS service when Users want to Communicate

Communication Platform

We provider All communication and related Services as a standalone platform



Users Communicate our WebRTC platform independently

Market Trends and Adoptions

Telecom Service Providers

VOIP Service Providers

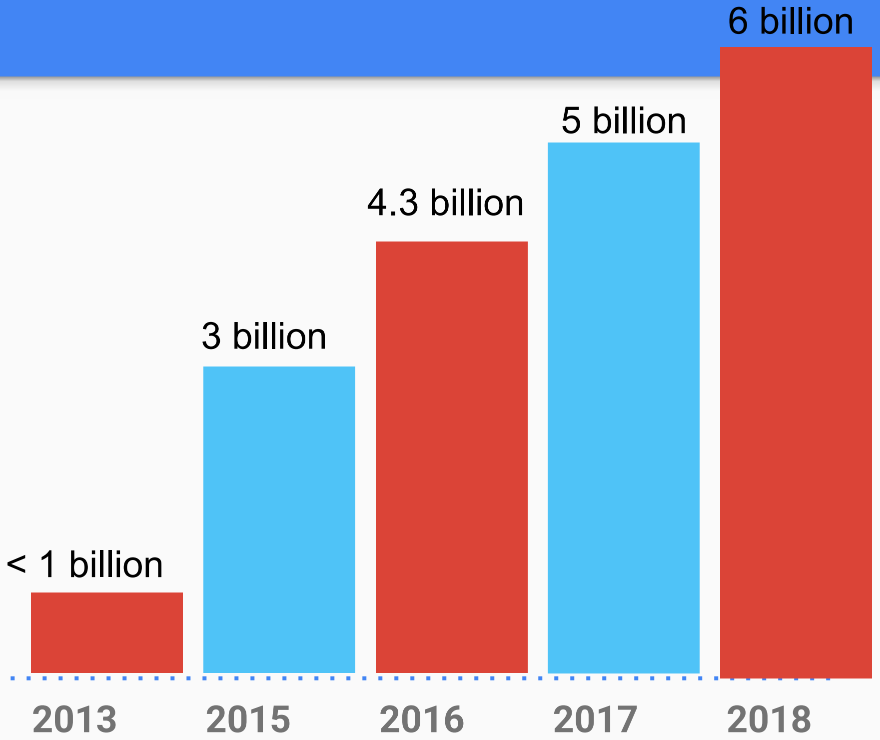
Genband

Huge opportunity for Small and Medium Enterprises to develop use cases around WebRTC media streams.

Market Trends for WebRTC supported devices (2013-2019)

WebRTC adoption by browsers , mobiles, TV , IOT , m2m etc is **2 billion today** .

It is expected to reach over 6 billion in 3 years.



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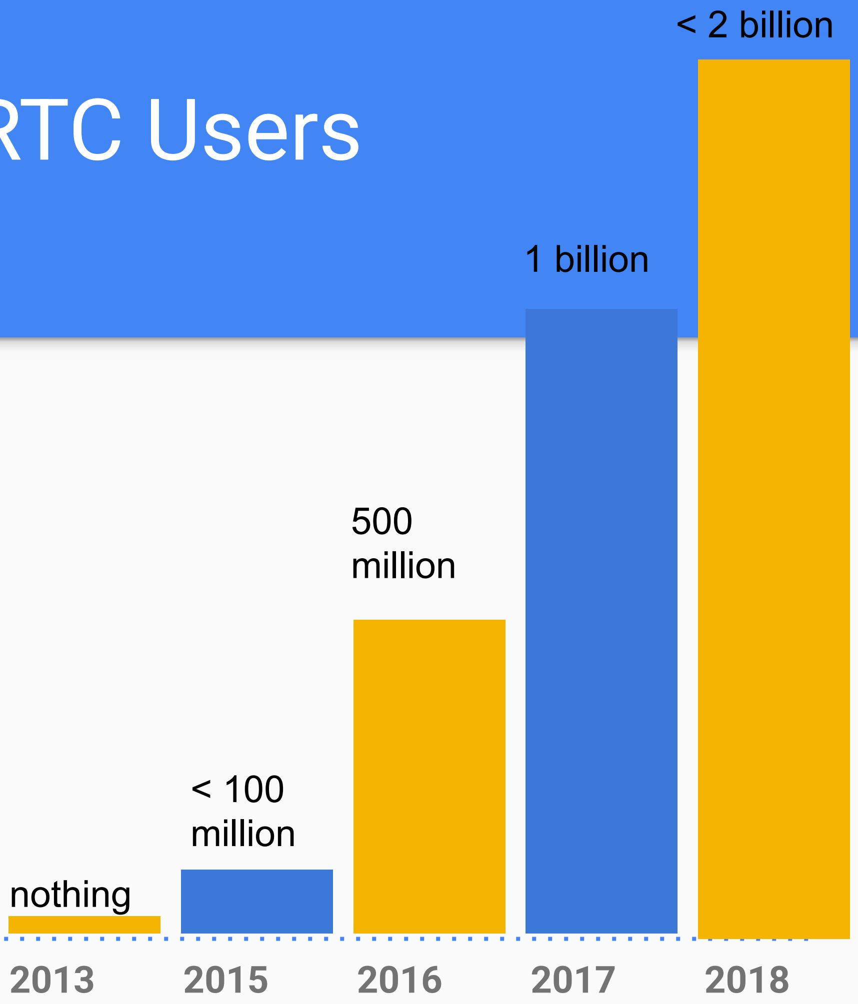
Market Trends for WebRTC Users (2013 -2019)

Business /enterprise and consumer web apps users for WebRTC is estimated to grow almost exponentially reaching almost 2 Billion by 2019



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Who is using WebRTC ?

Media Service providers

Enterprises / video conferencing

Telecom Service providers

Education, Healthcare, Transportation & Logistic

IOT and Device / surveillance Embedded like Rpi

Consumers / Web Sites

Customer Care Services

Concerns and bottlenecks around WebRTC

Although WebRTC is a great technology and holds very good potential it is not devoid of problems

TURN , NAT , ICE and Firewalls

Security in VPN and topology hiding

Cross platform concerns and codecs incompatible

Late adopters like Microsoft and Apple

References

- WebRTC 1.0: Real-time Communication Between Browsers

  W3C Editor's Draft 22 July 2016 http://w3c.github.io/webrtc-pc/
- https://webrtc.org/
- Getting Started with WebRTC

  http://www.html5rocks.com/en/tutorials/webrtc/basics/
- WebRTC : https://telecom.altanai.com/webrtc/
- Connectivity API

  https://developer.mozilla.org/en-US/docs/Web/API/WebRTC_API/Connectivity

Thank you

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