voxbone

Getting REAL with WebRTC

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ABOUT ME

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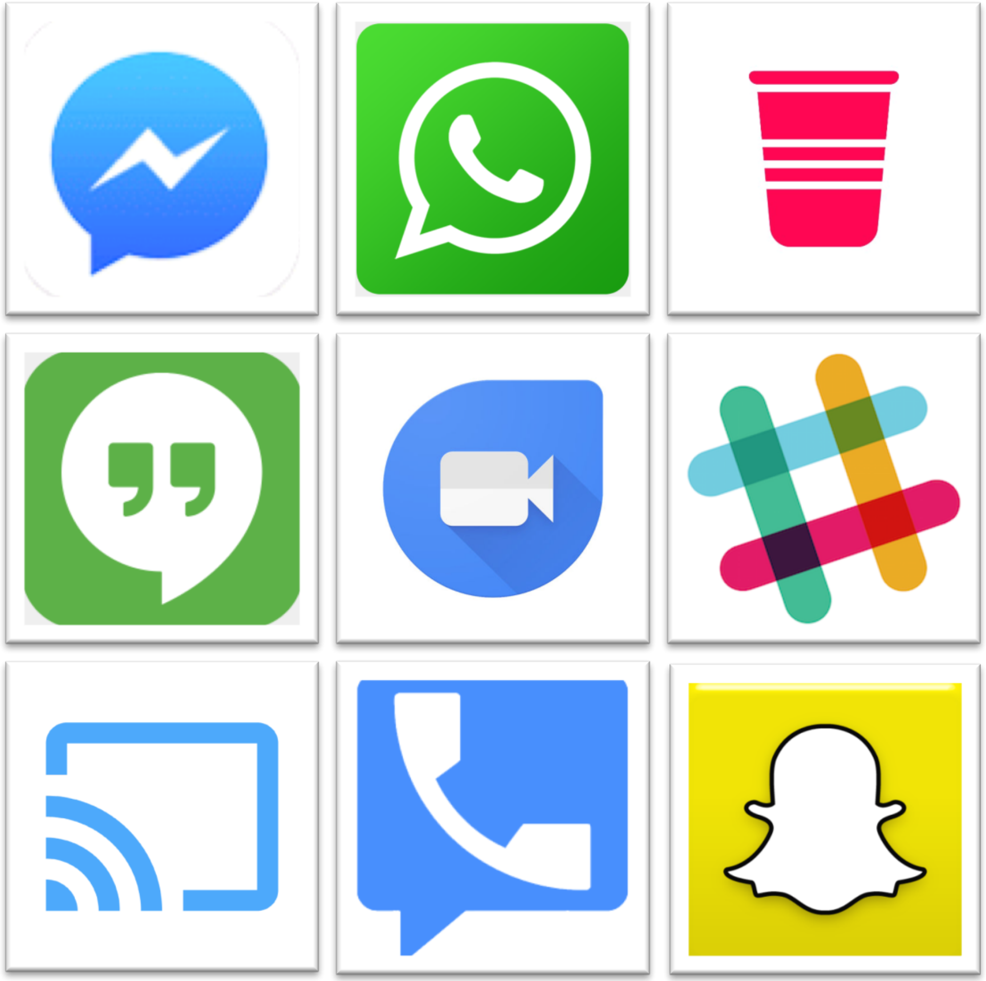
A blog for WebRTC developers
https://webrtcHacks.com
@webrtcHacks

WebRTC events & videos
https://www.krankygeek.com/
@webrtclive

SO WHAT IS WEBRTC?

HAVE YOU MADE A CALL WITH ONE OF THESE APPS?



IF NO, THEN YOU HAVE DEFINITELY WORKED WITH THESE

HOW DID WE GET HERE?

WORLD CLASS VOIP ENGINE FOR FREE

on2 technologies
VIDEO CODEC

GLOBAL IP SOLUTIONS
MEDIA ENGINE

opus
AUDIO CODEC

INTERNET & WEB STANDARD

3(+) WEB-FRIENDLY (/NATIVE) API'S

Stream Capture
Camera, Microphone, Screen

getUserMedia

Arbitrary p2p data transmission

dataChannel

getStats

Statistics

Peer-to-peer RTC connectivity

peerConnection

MASSIVE COMMUNITY

Too many projects to put on a slide

WebRTC.org

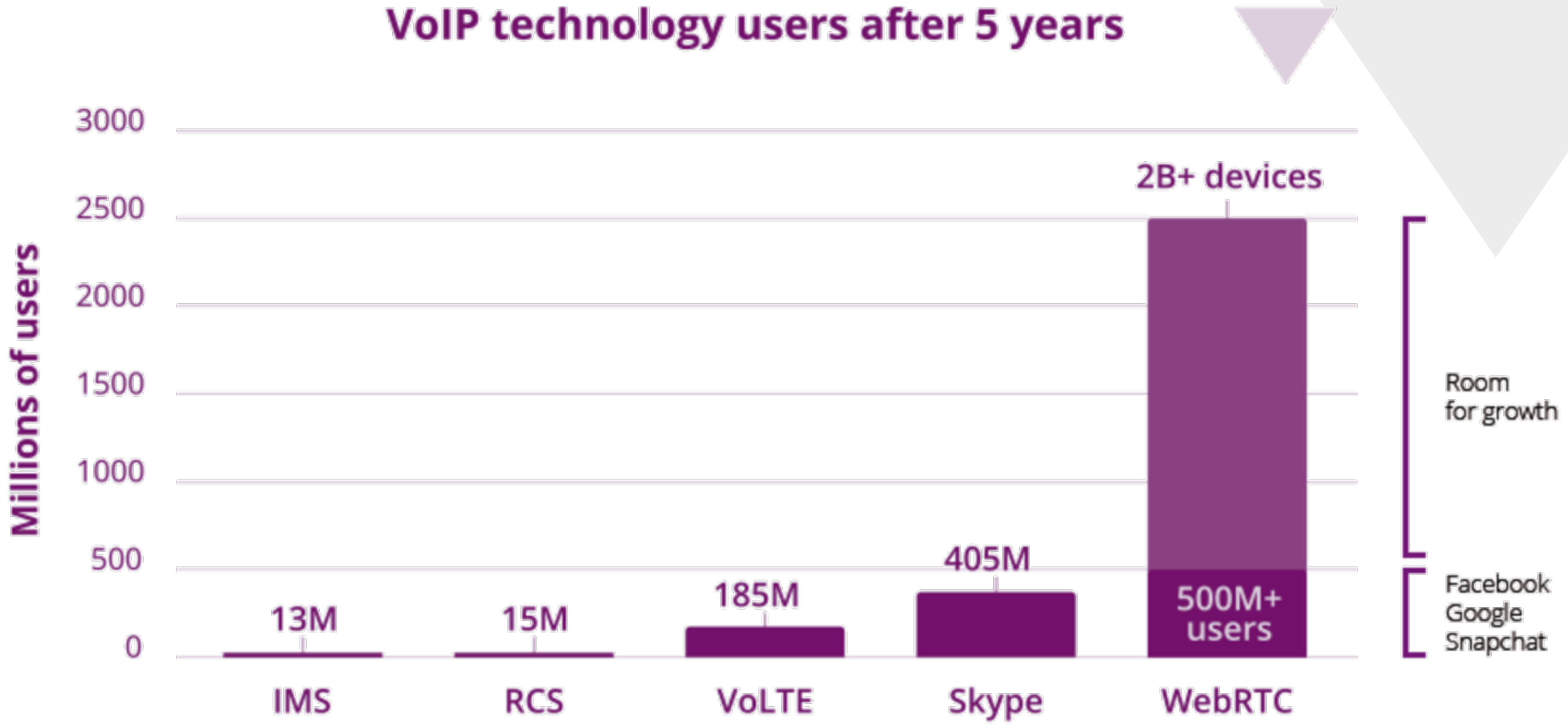
WEBRTC AFTER 5 YEARS

GOOGLE'S 5 YEAR STATS

- 2 Billion+ WebRTC Browsers
- 1 Billion+ minutes/week
- 950+ companies & projects
- 5 Billion+ WebRTC app downloads

https://groups.google.com/forum/#!topic/discuss-webrtc/I0GqzwfKJfQ
https://docs.google.com/presentation/d/1JwnW6v3OM0RfoDYrPPTJrDNeIpidgh7hF_k5E1j2oKM/edit?usp=sharing

WEBRTC IS WINNING



"Users" of various VoIP technologies (IMS, RCS, VoLTE) roughly 5 years after their introduction. Note "user" often means install or supported devices vs.WebRTC's 500M+ monthly active users.

Source: public figures, Chad Hart/Voxbone

400 million monthly active users
4 years since launch

WEBRTC WILL MAKE YOU RICH?

In 2016, the worldwide market for webRTC was valued at US$10.71 bn.

source: http://www.transparencymarketresearch.com/webrtc-market.html

THE MATH

US$10.71 bn

source: TMR

÷

WebRTC
11,874

LinkedIn

source: BlogGeek.Me

=

$901,971 / person

÷

950+

WebRTC-based companies and projects

source: google

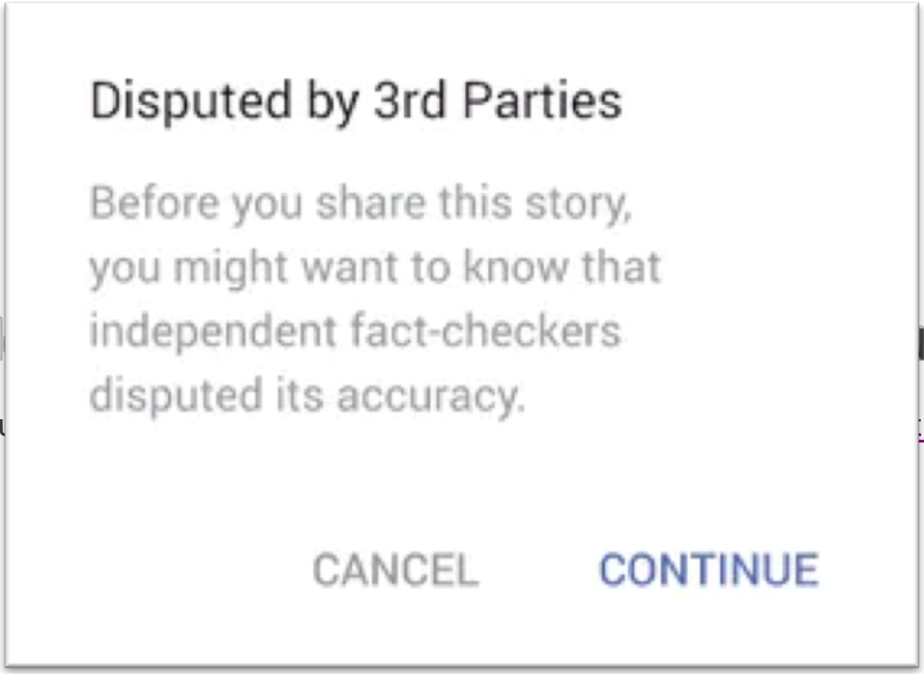
=

$ 11,273,684 / vendor

revenue:

$0B

Reality: the largest WebRTC user makes no money from WebRTC. Most of the largest WebRTC app monetize WebRTC indirectly at best.

In 2016, the worl 

ed at US$10.71 bn.

sou .html

Reality: Unfortunately WebRTC probably wont make you instantly rich by itself.

YOU NEED TO DO WEBRTC ANYWAY

WHY WEBRTC?
BETTER TECHNOLOGY

Media negotiation

Firewall/NAT Traversal

Audio codecs

Video codecs

Real-time flow controls

Encryption

WHY WEBRTC?
YOUR USERS DEMAND BETTER EXPERIENCES

DON'T DEFAULT TO THE SAME OLD USER EXPERIENCE

Source: http://www.publicdomainpictures.net/view-image.php?image=25044&picture=redial

WHERE TO START?

JavaScript is easy enough to start with

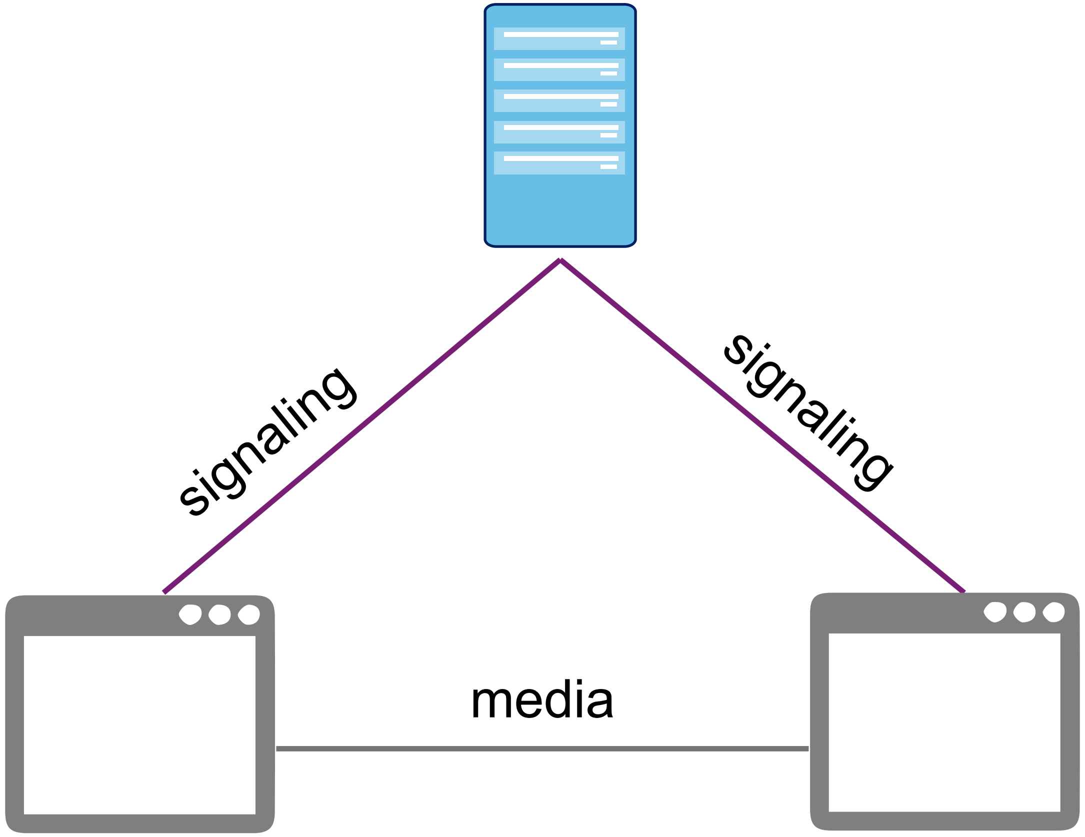
https://webrtc.github.io/samples/

MANY GUIDES AND SAMPLES

Blogs	Code	Videos	Forum
webrtcHacks BlogGeek.Me blog.mozilla.org/webrtc/ rtcbits.com	codelabs.developers.google.com/codelabs/webrtc-web webrtc.github.io/samples/ webrtc-experiment.com	youtube.com/krankygeek	groups.google.com/forum/#!forum/discuss-webrtc

and so many others…

Signaling: WebRTC is not 100% Peer-to-Peer



Global Lines

```
v=0
o=- 6801214714694628842 IN IP4 127.0.0.1
s=-
t=0 0
a=group:BUNDLE audio video
a=msid-semantic: WMS GUKF430Khp9jEQiPrdYe0LbTAALiNAKAIfl2
```

Audio Lines

```
m=audio 54278 RTP/SAVPF 111 103 104 0 8 106 105 13 126
c=IN IP4 180.6.6.6
a=rtcp:54278 IN IP4 180.6.6.6
```

ICE Candidates

```
a=candidate:4022866446 1 udp 2113937151 192.168.0.197 36768 typ host generation 0
a=candidate:4022866446 2 udp 2113937151 192.168.0.197 36768 typ host generation 0
a=candidate:2706108158 1 tcp 1509957375 192.168.0.197 0 typ host generation 0
a=candidate:2706108158 2 tcp 1509957375 192.168.0.197 0 typ host generation 0
a=candidate:1853887674 1 udp 1845501695 46.2.2.2 36768 typ srflx raddr 192.168.0.197 rport 36768 generation 0
a=candidate:1853887674 2 udp 1845501695 46.2.2.2 36768 typ srflx raddr 192.168.0.197 rport 36768 generation 0
a=candidate:2157334355 1 udp 33562367 180.6.6.6 54278 typ relay raddr 46.2.2.2 rport 38135 generation 0
a=candidate:2157334355 2 udp 33562367 180.6.6.6 54278 typ relay raddr 46.2.2.2 rport 38135 generation 0
```

ICE Parameters

```
a=ice-ufrag:kwlYyWNjhC9JBe/V
a=ice-pwd:AU/SQPupllyS0SDG/eRWDCfA
a=ice-options:google-ice
```

DTLS Parameters

```
a=fingerprint:sha-256 D1:2C:BE:AD:C4:F6:64:5C:25:16:11:9C:AF:E7:0F:73:79:36:4E:9C:1E:15:54:39:0C:06:8B:ED:96:86:00:39
a=setup:actpass
```

Session Description Protocol (SDP)

WebRTC uses Session Description Protocol (SDP) to negotiate media parameters

Source: webrtchacks.com/sdp-anatomy

SDP

There is a large continent in the WebRTC community that thinks SDP is evil. They have *very slowly* won, so expect more efforts to move away from SDP.

Many ORTC objects are already in WebRTC 1.0

ORTC was originally an alternative WebRTC standard without SDP but evolved into the future version of WebRTC. It is already built into mainline WebRTC standards and more ORTC concepts are coming.

```
PeerConnection
    .getSenders()
    .getReceivers()
    // Name TBD
    .addTransceiver(kind)
    .sctp
    ...

RtpSender
    .track
    .transport
    .getCapabilities()
    .getParameters()
    .setParameters(params)
    .replaceTrack(track)
    ...

RtpReceiver
    .track
    .transport
    .getCapabilities()

  .getContributingSources()
    ...
```

```
DtlsTransport
    .transport
    .state
    .getRemoteCertificates()
    .onstatechange
    ...
IceTransport
    .state
    .getLocalParameters(),
    .getRemoteParameters(),
    .getLocalCandidates()
    .getRemoteCandidates()
    .getSelectedCandidatePair()
    .onstatechange
    ...
SctpTransport
    .transport
DataChannel
    .transport
```

```
RtpParameters
    .codecs
    .encodings
    ...
  RtpCodecParameters
    (read only)
    .mimeType
    .payloadType
    ...
  RtpEncodingParameters
    .active
    .maxBandwidth
    (read only)
    .ssrc
    ...
  IceParameters
    (read only)
    .usernameFragment
    .password
  DtlsParameters
    ...
```

Source: https://docs.google.com/presentation/d/1oDVgPQtYpQNRkgiP51OQJS1sOW6mjWcGxpO29Bpdd28

SIGNALING

Often the first and fastest route to WebRTC for “telephony” people was to use SIP over WebSockets. There are several popular JavaScript SIP stacks that run in the browser.

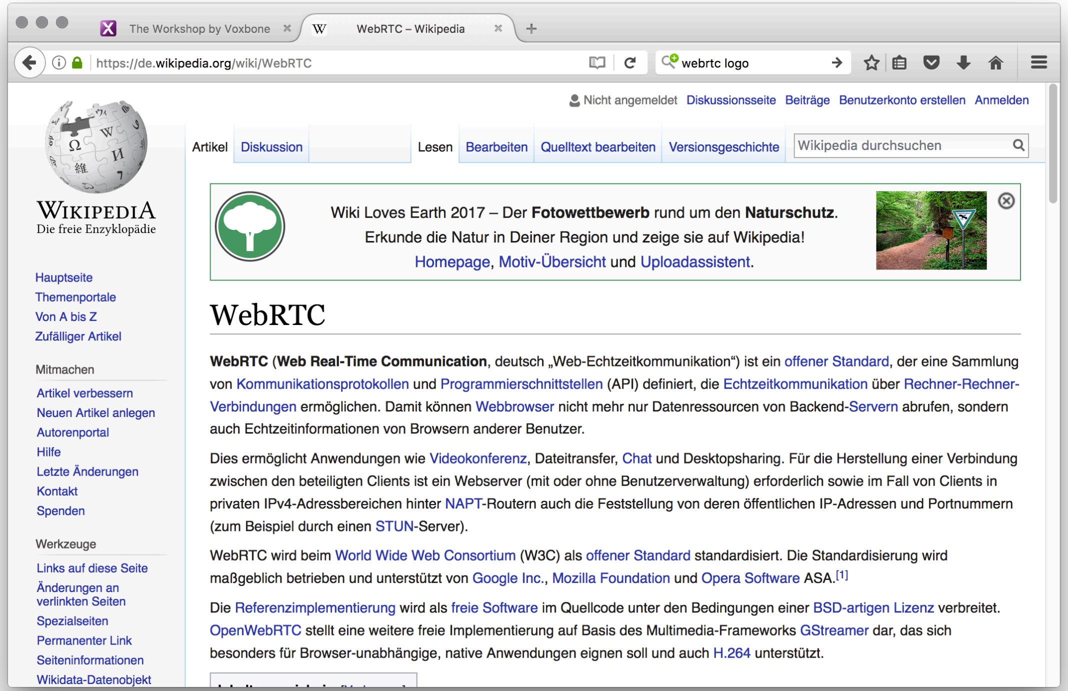
sipml5
The world's first HTML5 SIP client (WebRTC)

jsSIP
the javascript SIP library

SIP.js

Old-school Way

WHAT’S BEST FOR YOUR PHONE IS NOT WHAT’S BEST FOR YOUR BROWSER



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WHAT'S BEST FOR YOUR PHONE IS NOT WHAT'S BEST FOR YOUR MOBILE APP

Main concerns: load time, latency, background behavior, battery consumption

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JUST ASK ONE OF THE ORIGINAL SIP AUTHORS

Jonathan Rosenberg @jdrosen2 · 18 Mar 2016

Why no SIP in Spark softclients? 3/ webrtc often needs call state queries - doesn't exist in SIP @rowantrollope @DaveMichels

1 6 5

Jonathan Rosenberg @jdrosen2 · 18 Mar 2016

Why no SIP in Spark softclients? 2/ cross device media key, not well supported in SIP @DaveMichels @rowantrollope

2 5 3

Jonathan Rosenberg @jdrosen2 · 17 Mar 2016

Why no SIP in Spark softclients? 1/ sip does not work well with mobile push notifications @DaveMichels @rowantrollope

3 4 8

SIGNALING

The market has proven that a JSON signaling approach is both practical and more effective.

{JSON}

Modern Way

Make Your own Server

Simple Socket.IO server:

- Only 32 lines
- No logic, it only forwards messages

In many cases, making your own signaling server isn’t so hard or you can leverage existing web-oriented signaling systems.

```
var socketIO = require('socket.io');
var server = require('http').createServer().listen(7000, '0.0.0.0');
var io = socketIO.listen(server);

// Super simple server:
//  * One room only.
//  * We expect two people max.
//  * No error handling.

io.sockets.on('connection', function (client) {
    console.log('new connection: ' + client.id);

    client.on('offer', function (details) {
        client.broadcast.emit('offer', details);
        console.log('offer: ' + JSON.stringify(details));
    });

    client.on('answer', function (details) {
        client.broadcast.emit('answer', details);
        console.log('answer: ' + JSON.stringify(details));
    });

    client.on('candidate', function (details) {
        client.broadcast.emit('candidate', details);
        console.log('candidate: ' + JSON.stringify(details));
    });

    // Here starts evertyhing!
    // The first connection doesn't send anything (no other clients)
    // Second connection emits the message to start the SDP negotation
    client.broadcast.emit('createoffer', {});
});
```

See more of César’s talk here: https://youtu.be/HS1eKPL4f1o

HOW TO GET/USE A SIGNALING SERVER

Ask your CPaaS provider

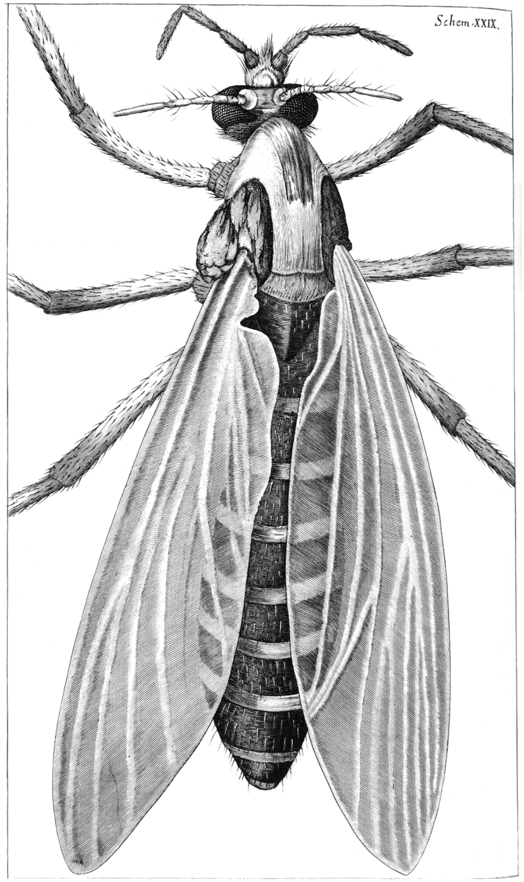
All include signaling

Run your own

Write your own in node.js
Matrix.org
EasyRTC
SimpleWebRTC
...
or search github

Use a messaging service

Firebase
PubNub
Pusher
GCM



NAT TRAVERSAL

Interactive Connectivity Establishment (ICE)

A protocol for establishing a peer-to-peer media connection between peers behind NAT and firewall devices.

Get A STUN & TURN Server

Run your own

coturn
restund
Numb
Frozen Mountain
EyeBall Networks

Use a TURN service

Xirsys
Twilio
Bit6
Video Roaming

So WebRTC's perfect?

Let’s talk about browser support

Canary Chrome Opera Nightly Firefox Bowser Edge Safari

WEBRTC BROWSER SUPPORT

Vendor	Google	Microsoft	Mozilla	Microsoft	Apple
Browser	Chrome	Internet Explorer	Firefox	Edge	Safari
Desktop	💚	⛔	💚	💚	⛔
Android	💚	🥚	💚	🥚	🥚
iOS	⛔	🥚	⛔	🥚	⛔

Apple has been the worst for WebRTC

3 WAYS APPLE HURTS WEBRTC

Safari

No support for desktop or mobile

70% of usage on iOS

WebView for iOS

No support for easy native apps

20% of time share vs. Browser, Android, Desktop

Other Browser Apps on iOS

No one else is allowed to make a WebRTC Browser on iOS

30% of browser share on iOS

WEBRTC BROWSER SUPPORT
USAGE SHARE

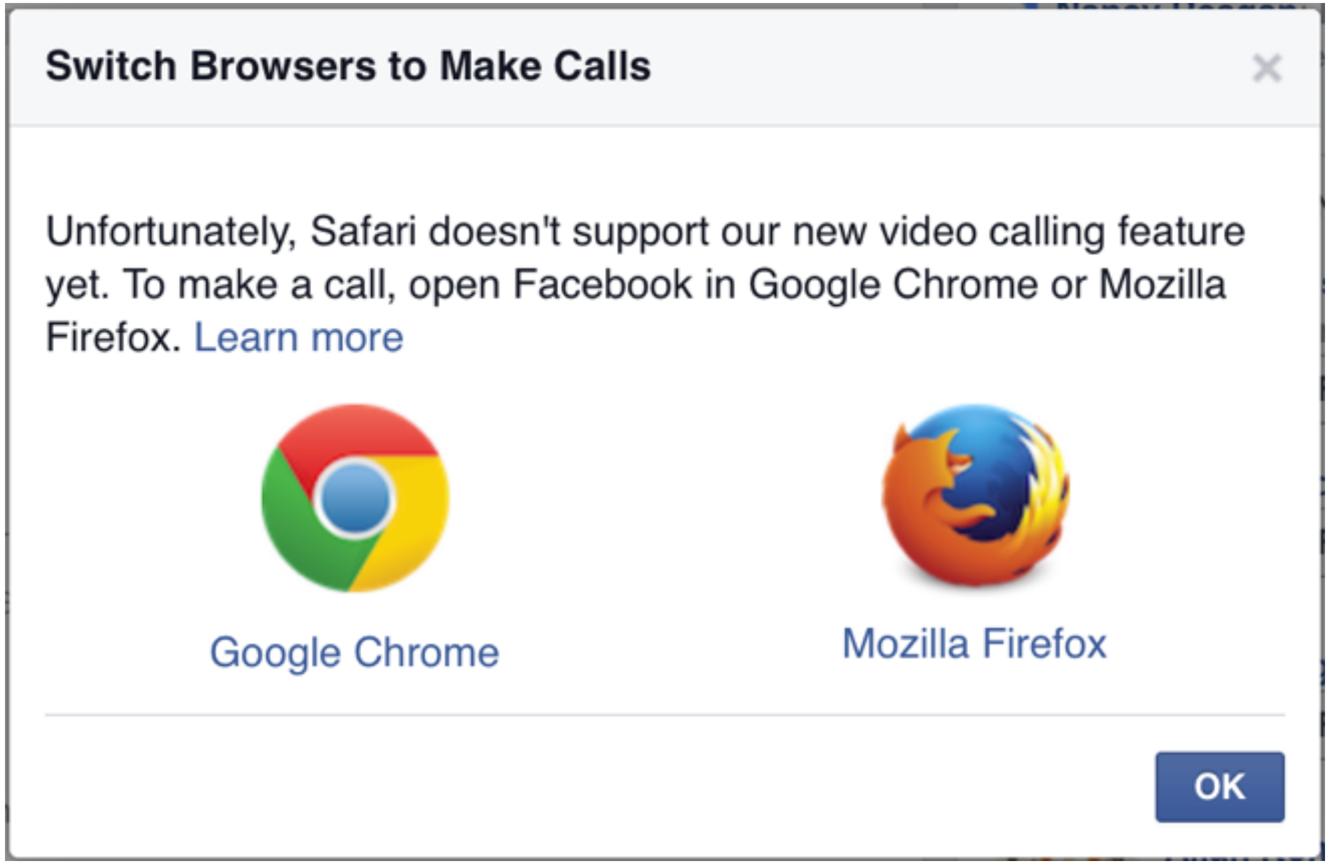
Usage patters vary considerably by application and country. Make sure you understand what your users are doing (and can do) before making any WebRTC decisions.

Vendor	Google	Microsoft	Mozilla	Microsoft	Apple
Browser	Chrome	Internet Explorer	Firefox	Edge	Safari
Desktop	25%	10%	5%	2%	2%
Android	28%	0%	2%	0%	0%
iOS	5%	0%	1%	0%	10%

**Chad's rough estimates based on public figures. Others such as Opera and UC browsers excluded (~10%)

HOW FACEBOOK HANDLES BROWSERS WITHOUT WEBRTC

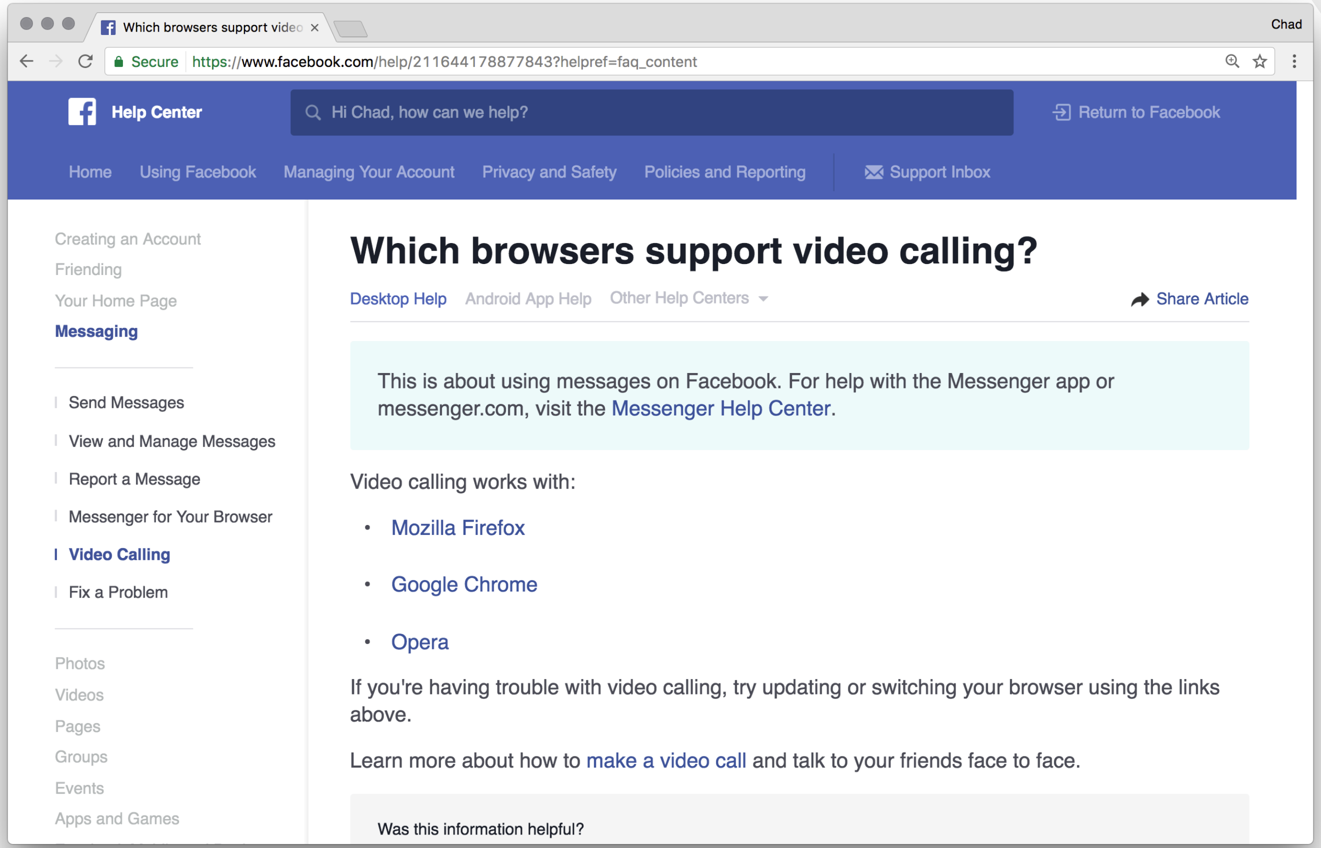
Facebook choose to tell its users to get a better browser if they tried calling with Safari.



This is what happened when I tried to make a Messenger call with Safari when they first launched the service. Facebook had no issues telling its users to get a better browser. Do you?

HOW FACEBOOK HANDLES BROWSERS WITHOUT WEBRTC

They have softened a bit and now hide WebRTC features on non-WebRTC browsers.



They have since hidden the feature on non-WebRTC browsers, but still recommend using a WebRTC browser.

So should we give up on Apple?

Image source: https://farm3.static.flickr.com/2540/4094662593_5e8e6d917b_b.jpg

APPLE SUPPORT IS COMING, EVENTUALLY…

No one outside of Apple knows exactly when or how they will introduce WebRTC, but it is definitely coming in the not too distant future.

WebKit

Open Source Web Browser Engine

Features

WebRTC

In Development

An API to facilitate real-time communication for browser-to-browser applications.

Reference: http://www.w3.org/TR/webrtc/

Contact: @jonathandavis - Jon Davis

CHROME WILL BREAK YOUR IMPLEMENTATION

```
LOG: PC [__default] ERROR: OperationError: Failed to set remote offer sdp: Session error code:
ERROR_CONTENT. Session error description: rtcpMuxPolicy is 'require', but media description does not
contain 'a=rtcp-mux'..
```

Chrome likes to move fast and break things. If you don't move with them they will break you too.

TEST EARLY, TEST OFTEN, & FIX THINGS

Leverage the Beta, Developer, and "initial release" versions that the browser vendors provide to give you 2-6 months notice on coming browser changes.

Insider Preview (fast)

Insider Preview (slow)

YEAH, YOU NEED TO BE PREPARED

As long as you pay attention and are willing to do sustaining development, browser change issues are manageable.

DON'T IGNORE NATIVE MOBILE APPS

This is a huge topic. No time here, but don't disregard native mobile. The majority of WebRTC usage is on native mobile apps – not the web!

WEBRTC TO DO LIST

CLIENT SUPPORT	RTC Architecture	Front-end	Backend-end	Cloud Environment	Ops	Test & monitoring

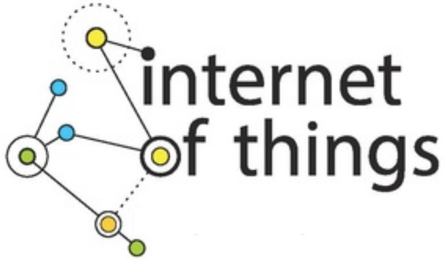
• WebRTC Browser support
• Non-WebRTC Browser support
• Native app support
	• Peer-peer
	• Peer-gateway
	• Multi-party
	  ◦ Mesh
	  ◦ MCU
	  ◦ SFU
	  ◦ Broadcast
	• Audio vs. video
	• Codec vs. data
	• Screenshare
	• Web vs. SIP
		• UI/UX
		• Framework
			• Web servers
			• Framework
			• DB
			• Client signaling
			• Media processing
			• STUN
			• TURN
				• Performance
				  ◦ CPU
				  ◦ Costs
				  ◦ Latency
				  ◦ Bandwidth
				  ◦ I/O
				• VM/Container strategy
				• Orchestration
				• CDN
				• DNS/IP Addressing
					• Infrastructure monitoring
					• Logging
						• Continuous Integration
						• Call monitoring
						• Load Test
						• Active Test
						• Browser test
						• Device test

Here is a quick check list of items to consider when deploying WebRTC.


Do you really want to make cheaper phone calls
for the rest of your career?
Me neither!

NEW FORMS OF REAL TIME COMMS ARE COMING

The good news is that WebRTC is a perfect gateway technology into many new exciting domains.
Leverage your RTC expertise in new ways!



Virtual Reality

Gaming

Real time streaming

Robotics

THANK YOU

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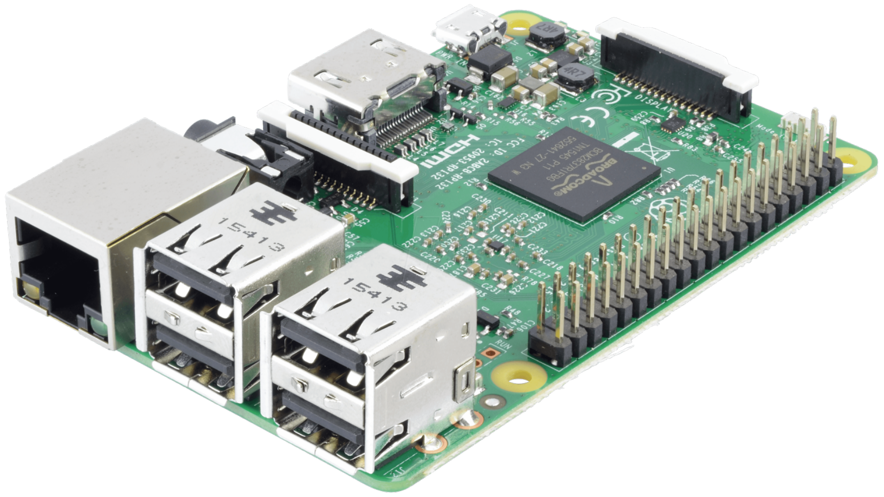
QUIZ TIME



Raspberry Pi 3

Belgian Chocolates

QUIZ & QUESTION TIME

1. What is your favorite colour?
2. What are the 2 primary standards bodies for WebRTC?
3. How many RTC subscribers does Facebook have?
4. What are some of the downsides to SIP for client signaling in WebRTC?
5. What is ORTC?
6. What does ICE stand for?
7. What does TURN do?
8. Which Microsoft Browser supports WebRTC?
9. What is the airspeed of an unladen swallow?
10. Do Android's native webview support WebRTC?

THANK YOU

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