
Google
Developers

WebRTC

Plugin-free realtime communication

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gowebrtc.appspot.com

github.com/samdutton/gowebrtc

“WebRTC is a new front in the long war for an open and unencumbered web”

Brendan Eich
– Mozilla CTO and inventor of JavaScript

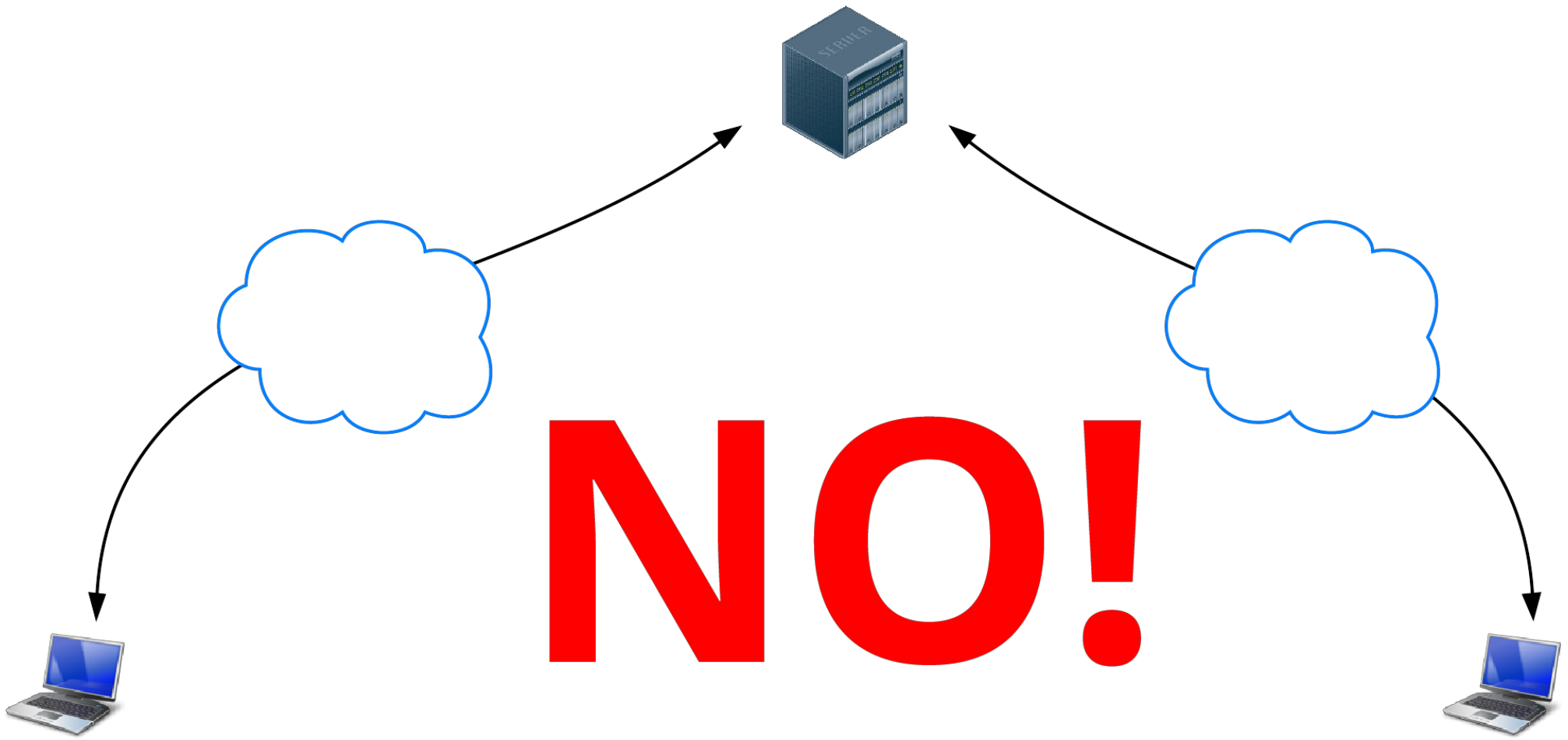
Real-time communication built into the browser

Open source, no plugins, free

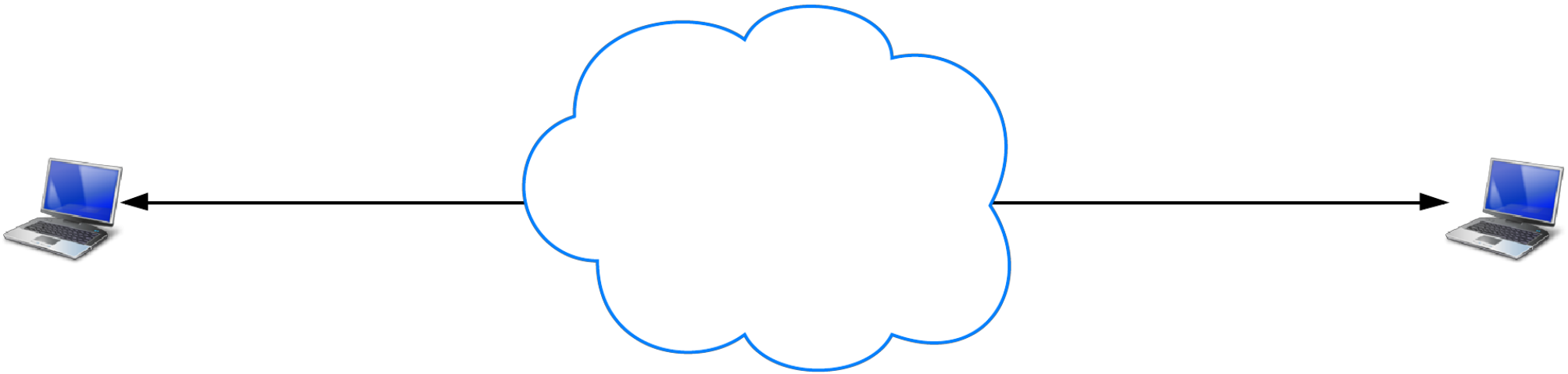
Video, audio, data

High quality, low cost

Peer to peer


NO!

JA!

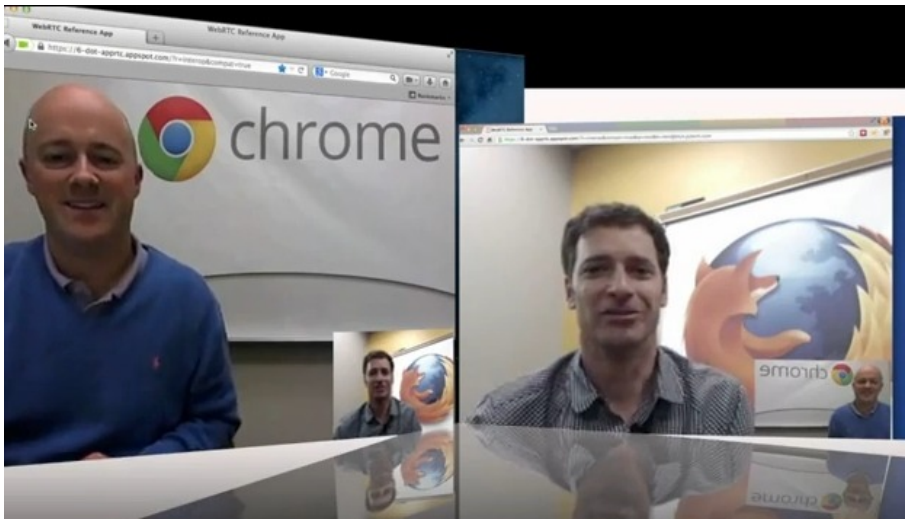


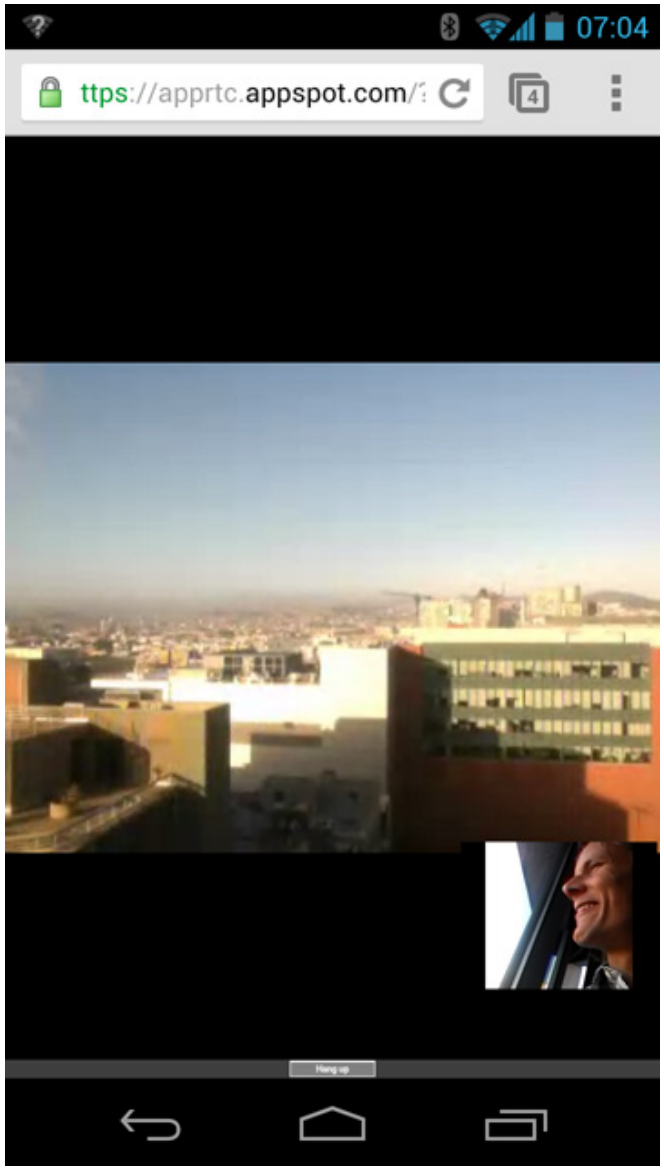
Shopping list

- IETF communication protocols: RTCWEB, JSEP…
- W3C API standards: WebRTC, Media Capture and Streams…
- Media and communication stack: libjingle, VP8, Opus…
- Design for a new communications ecosystem

WebRTC across platforms

- Chrome and Chrome for Android
- Firefox and Firefox for Android
- Opera
- Native Java and Objective-C bindings





Qt moving to Chromium

- Framework for cross-platform/device native and embedded apps
- Qt WebKit => Qt WebEngine
- Multimedia and new HTML5 features such as WebRTC working out-of the-box

1,000,000,000+

WebRTC endpoints

“Voice is just another JS application”

Henning Schulzrinne
– CTO, US FCC


sky NEWS FTSE 6457.78
SKY NEWS CENTRE LIVE SOUTH LONDON
15:37 TAKEN TO A MILITARY HOSPITAL
BREAKING NEWS
AP: POLICE SAY TWO ROCKETS FIRED FRO
0:03 / 4:22

en.wikipedia.org/wiki/File:BBC_HD_SNG.jpg

What do we need for RTC?

Three main tasks

- Acquiring audio and video
- Communicating audio and video
- Communicating arbitrary data

Three main JavaScript APIs

- MediaStream (aka getUserMedia)
- RTCPeerConnection
- RTCDataChannel

MediaStream

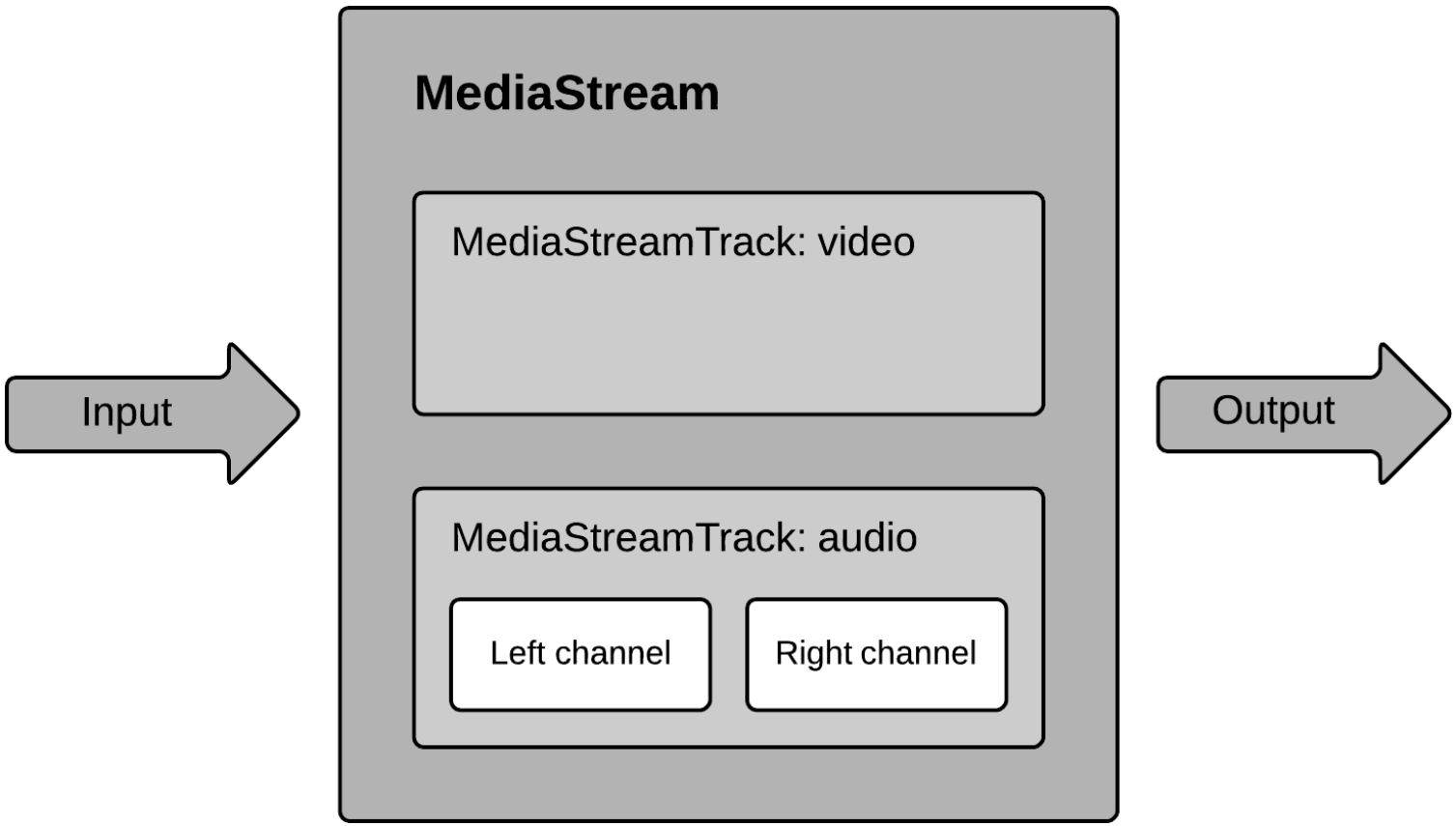
Acquiring audio and video

What do we need?

- Streaming media: `MediaStream`
- Get local media: `navigator.getUserMedia()`

MediaStream

- Represents a stream of synchronised media
- Can contain multiple audio and/or video MediaStreamTracks
- Obtain a MediaStream with `navigator.getUserMedia()`


MediaStream
MediaStreamTrack: video
MediaStreamTrack: audio
Left channel
Right channel
Input
Output

gUM

It's pretty simple.

JAVASCRIPT

```
var constraints = {video: true};

function successCallback(stream) {
  var video = document.querySelector("video");
  video.src = window.URL.createObjectURL(stream);
}

function errorCallback(error) {
  console.log("navigator.getUserMedia error: ", error);
}

navigator.getUserMedia(constraints, successCallback, errorCallback);
```

simpl.info/gum

gUM permissions

- HTTPS
- Chrome apps: `audioCapture` and `videoCapture` permissions
- Chrome flag: `--use-fake-ui-for-media-stream`
- UI settings can be changed afterwards.


file://

idevelop.github.com/ascii-camera

FaceKat

html5-demos.appspot.com/static/getusermedia/photobooth.html

webcamtoy.com

Constraints

- Mandatory or optional
- Resolution: width and height
  - from a fixed list
  - no cropping or scaling (yet)
- Frame rate
- Facing mode: front or back camera
- Source type: video camera, screen capture...
- Source id
- Volume

simpl.info/getusermedia/constraints

simpl.info/getusermedia/sources

Facing mode, applyConstraints()

- Choose source: spec
- Apply constraints from JavaScript: spec

getUserMedia + Web Audio

JAVASCRIPT

```
// Success callback when requesting audio input stream
function gotStream(stream) {
    var audioContext = new webkitAudioContext();

    // Create an AudioNode from the stream
    var mediaStreamSource = audioContext.createMediaStreamSource(stream);

    // Connect it to the destination or any other node for processing!
    mediaStreamSource.connect(audioContext.destination);
}

navigator.webkitGetUserMedia({audio:true}, gotStream);
```

Make sure to enable Web Audio Input in about:flags!

webaudiodemos.appspot.com/AudioRecorder

gUM screencapture

JAVASCRIPT

```
var constraints = {
  video: {
    mandatory: {
      chromeMediaSource: 'screen'
    }
  }
};

navigator.webkitGetUserMedia(constraints, gotStream);
```

Screen sharing

Tab capture: chrome.tabCapture

Media Stream Recording API

- Demo: simpl.info/mediarecorder
- Spec
- Chrome Intent to Implement
- Streams API

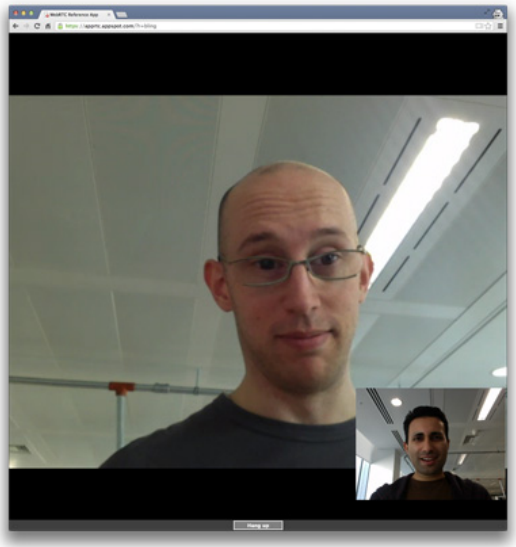
Media Stream Image Capture API

- ~~Demo~~
- Spec
- `getFrame()` creates an `ImageData` object available in `onframegrab`
- `takePhoto()` creates a Blob available in `onphoto`

RTCPeerConnection

Audio and video communication between peers

Communicate Media Streams



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getUserMedia

+

RTCPeerConnection

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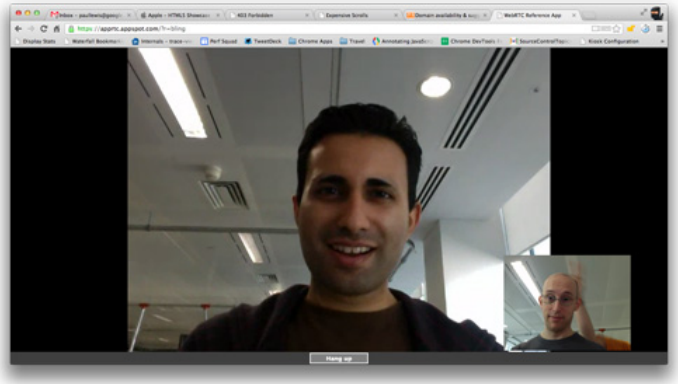
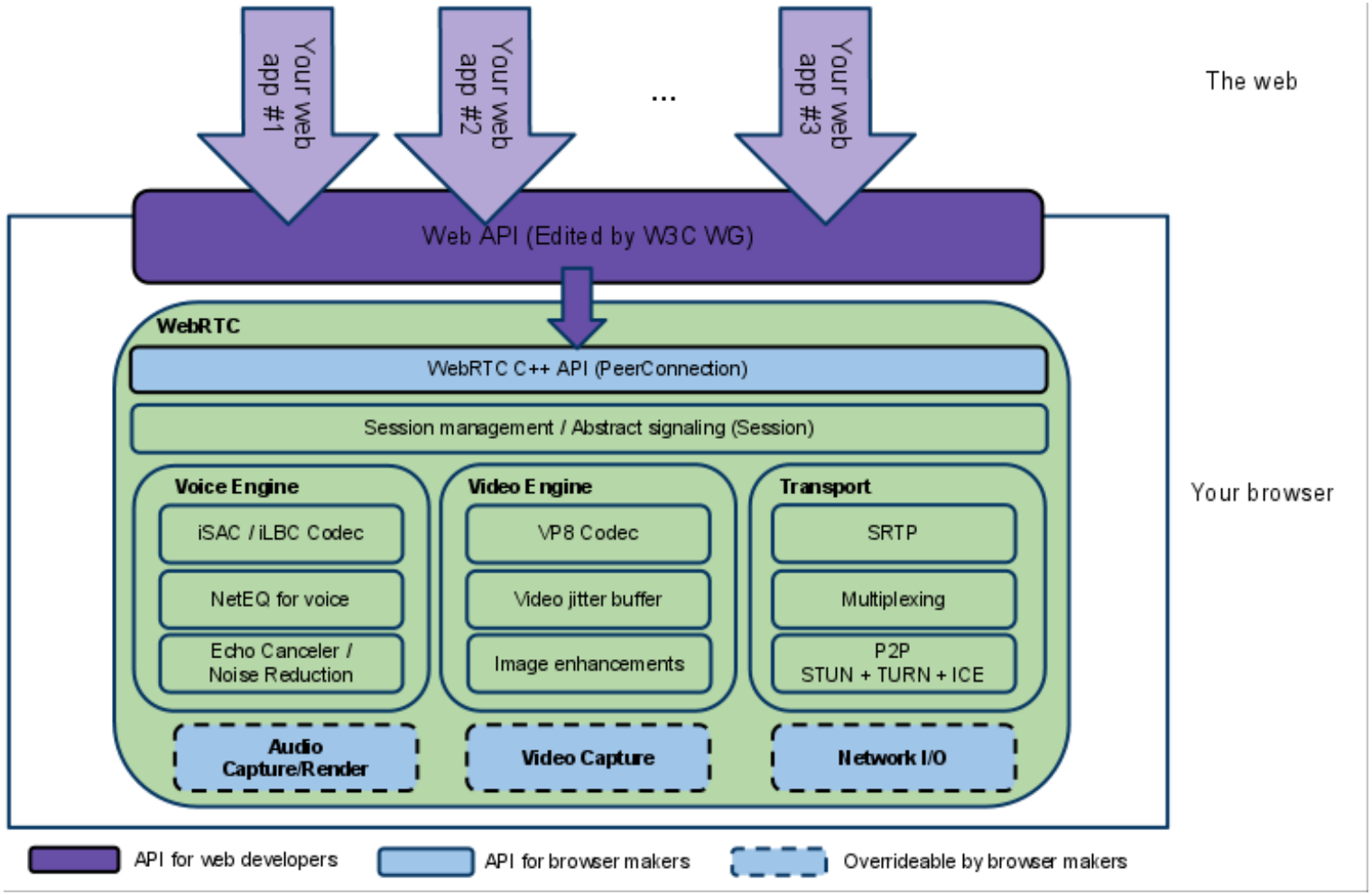


RTCPeerConnection does a lot

- Signal processing
- Codec handling
- Peer to peer communication
- Security
- Bandwidth management

...

WebRTC architecture

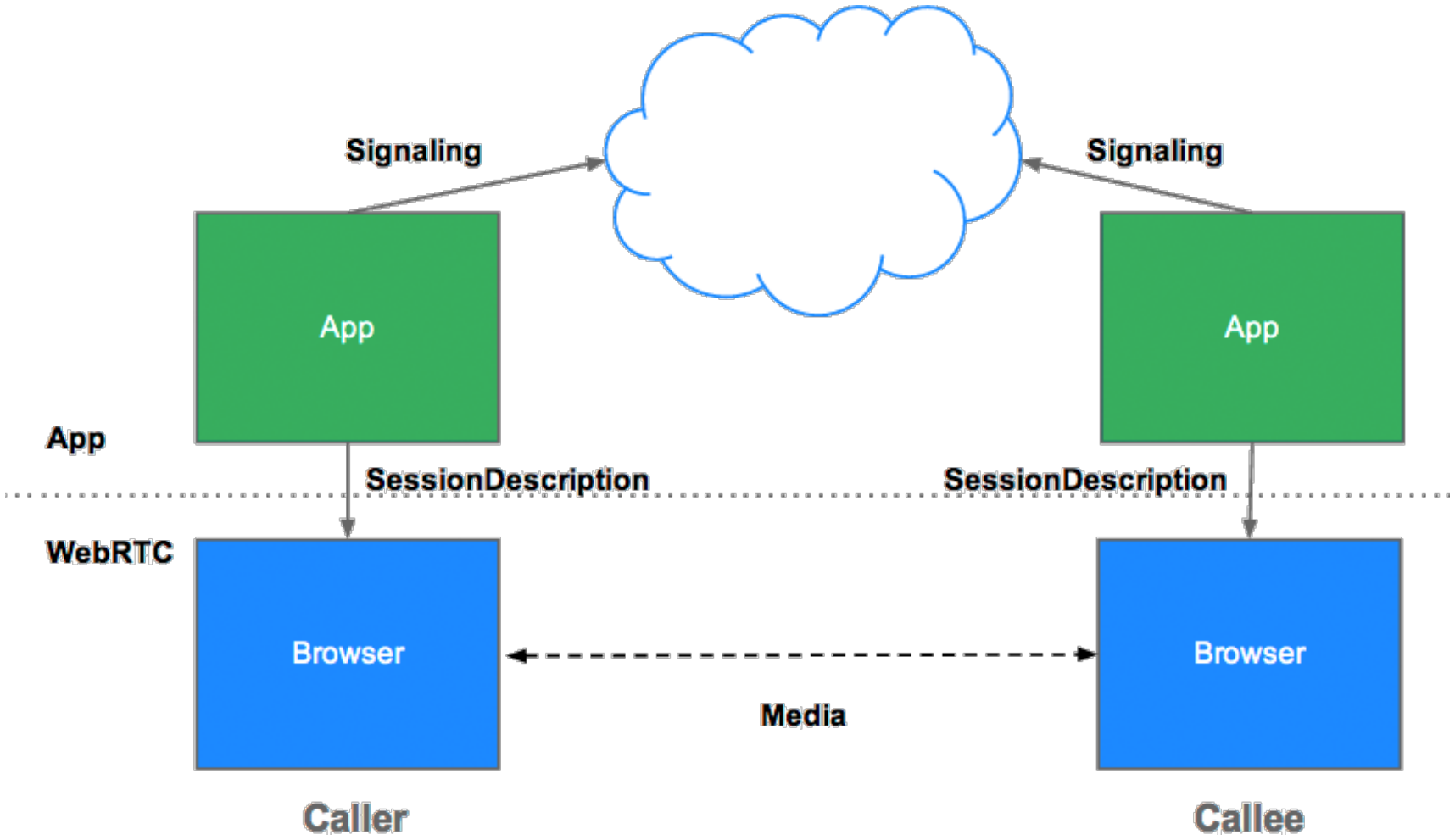


Peer to peer — but we need servers :^\

What does WebRTC need servers for?

- Exchange metadata to coordinate communication: signaling
- Cope with NATs and firewalls: STUN and TURN

JSEP architecture



Signaling

- Need to exchange 'session description' objects:
  - What media formats I support, what I want to send
  - Network information for peer-to-peer setup
- Can use any messaging mechanism
- Can use any messaging protocol
- ...and can be used for application data

Signaling: how?

- Needs to be bidirectional
- Repeated polling: inefficient and not scalable
- Long polling / Comet (as used by Google App Engine)
- XHR + EventSource (aka Server-sent events): demo
- WebSocket:
  - more natural solution — it's bidirectional!
  - supported by all browsers that support WebRTC, desktop and mobile
  - use TLS: for security and to avoid proxy problems
  - for more information: see Ilya Grigorik's forthcoming O'Reilly chapter
  - Peter Lubber's WebSocket Cheat Sheet

Signaling with Node and Socket.io

- Socket.io uses WebSocket with fallbacks
- Simple to exchange messages
- Built-in concept of 'rooms'

Codelab

Live example

Signaling gotchas

- RTCPeerConnection won't start gathering candidates until `setLocalDescription()` is called: mandated in JSEP IETF draft
- Take advantage of Trickle ICE: call addIceCandidate() as soon as candidates arrive
- Not bandwidth or CPU hungry, but could still be heavy load

RTCPeerConnection initialisation

- Ascertain local media conditions: resolution, codec capabilities...
- Get potential network addresses for the application's host: candidates

Make me an offer

1. Fred calls `createOffer()`
2. In the callback, Fred calls `setLocalDescription()`
3. Fred serialises the offer and sends it Wilma
4. Wilma calls `setRemoteDescription()` with the offer
5. Wilma calls `createAnswer()`
6. Wilma calls `setLocalDescription()` with the answer and sends it to Fred
7. Fred receives answer and calls `setRemoteDescription()`

Find me a candidate

1. Fred and Wilma create RTCPeerConnection objects
2. If success, `icecandidate` events start propagating
3. Fred starts serialises IceCandidates and sends them to Wilma
4. Wilma gets Fred's IceCandidates, calls `addIceCandidate()`
5. Wilma serialises IceCandidates and sends them to Fred
6. Fred gets Wilma's IceCandidates, calls `addIceCandidate()`
7. Ping!

An RTCSessionDescription

UGH

```
v=0
o=- 7614219274584779017 2 IN IP4 127.0.0.1
s=-
t=0 0
a=group:BUNDLE audio video
a=msid-semantic: WMS
m=audio 1 RTP/SAVPF 111 103 104 0 8 107 106 105 13 126
c=IN IP4 0.0.0.0
a=rtcp:1 IN IP4 0.0.0.0
a=ice-ufrag:W2TGCZw2NZHuwlnf
a=ice-pwd:xdQEccP40E+P0L5qTyzDgfmW
...
```

Want to know what all this SDP gobbledygook actually means?

Take a look at the IETF examples.

Candidate objects

NICE

```
{
  'type':'candidate',
  'label':1,
  'id':'video',
  'candidate':'a=candidate:1274936569 1 udp 1845501695 2.96.35.15 63579
              typ srflx raddr 192.168.0.3 rport 63579 generation 0\r\n'
}
...
{
  'type':'candidate',
  'label':0,
  'id':'audio',
  'candidate':'a=candidate:3802297132 1 udp 2113937151 192.168.0.3 63579
              typ host generation 0\r\n'
}
```

RTCPeerConnection + signaling

w3.org/TR/webrtc/#simple-peer-to-peer-example

simpl.info/rtcpeerconnection/munge

simpl.info/rtcpeerconnection/multi

simpl.info/pc

apprtc.appspot.com

RTCDataChannel

Bidirectional communication of arbitrary data between peers

Communicate arbitrary data


```
onreceivemessage = handle(data);
...
var myData = [
  {
    id: "ship1";
    x: 24,
    y: 11,
    velocity: 7
  },
  ....
]
send(myData);
```

→

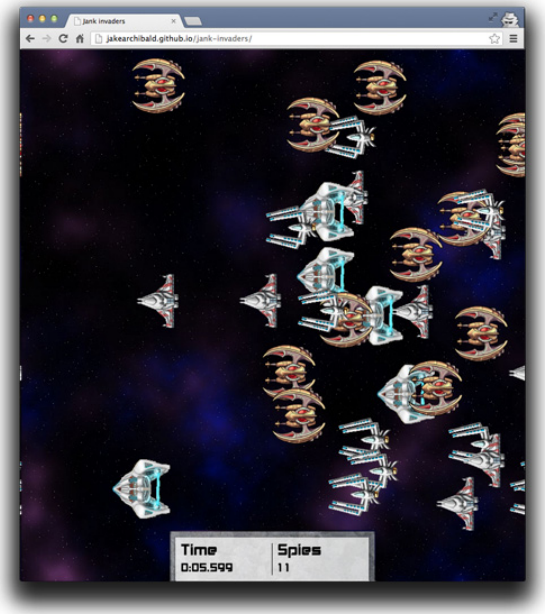
RTCDataChannel

+

RTCPeerConnection

←

```
onreceivemessage = handle(data);
...
var myData = [
  {
    id: "ship7";
    x: 19,
    y: 4,
    velocity: 18
  },
  ....
]
send(myData);
```



RTCDataChannel

- Same API as WebSockets
- Ultra-low latency
- Optionally unreliable or reliable:
  — Firefox and Chrome 31, Chrome 30 behind a flag
- Secure

RTCDataChannel API

JAVASCRIPT

```
var pc = new webkitRTCPeerConnection(servers,
  {optional: [{RtpDataChannels: true}]});

pc.ondatachannel = function(event) {
  receiveChannel = event.channel;
  receiveChannel.onmessage = function(event){
    document.querySelector("div#receive").innerHTML = event.data;
  };
};

sendChannel = pc.createDataChannel("sendDataChannel", {reliable: false});

document.querySelector("button#send").onclick = function (){
  var data = document.querySelector("textarea#send").value;
  sendChannel.send(data);
};
```

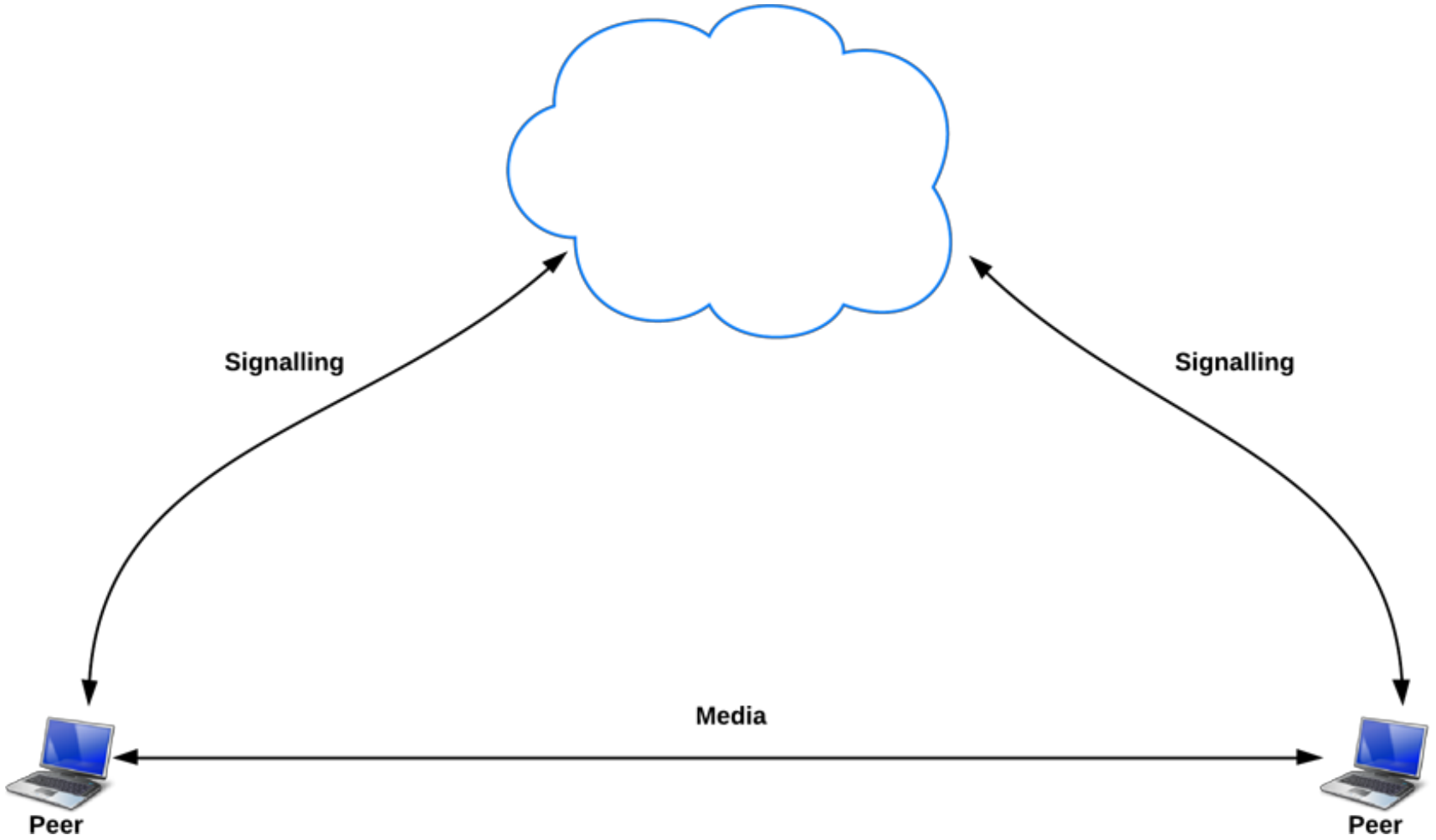
simpl.info/dc

Sharefest

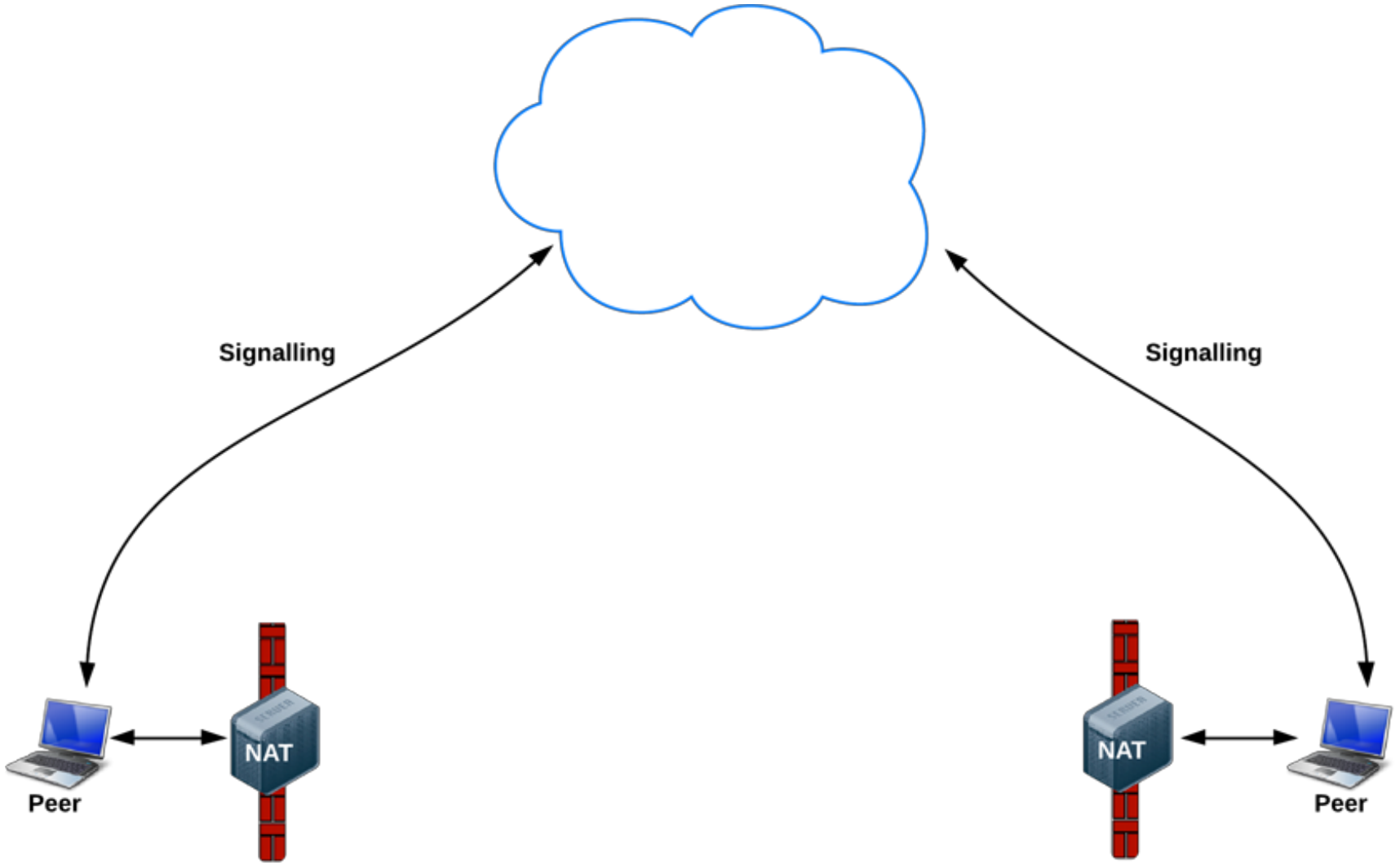
STUN and TURN

P2P in the age of firewalls and NATs

An ideal world



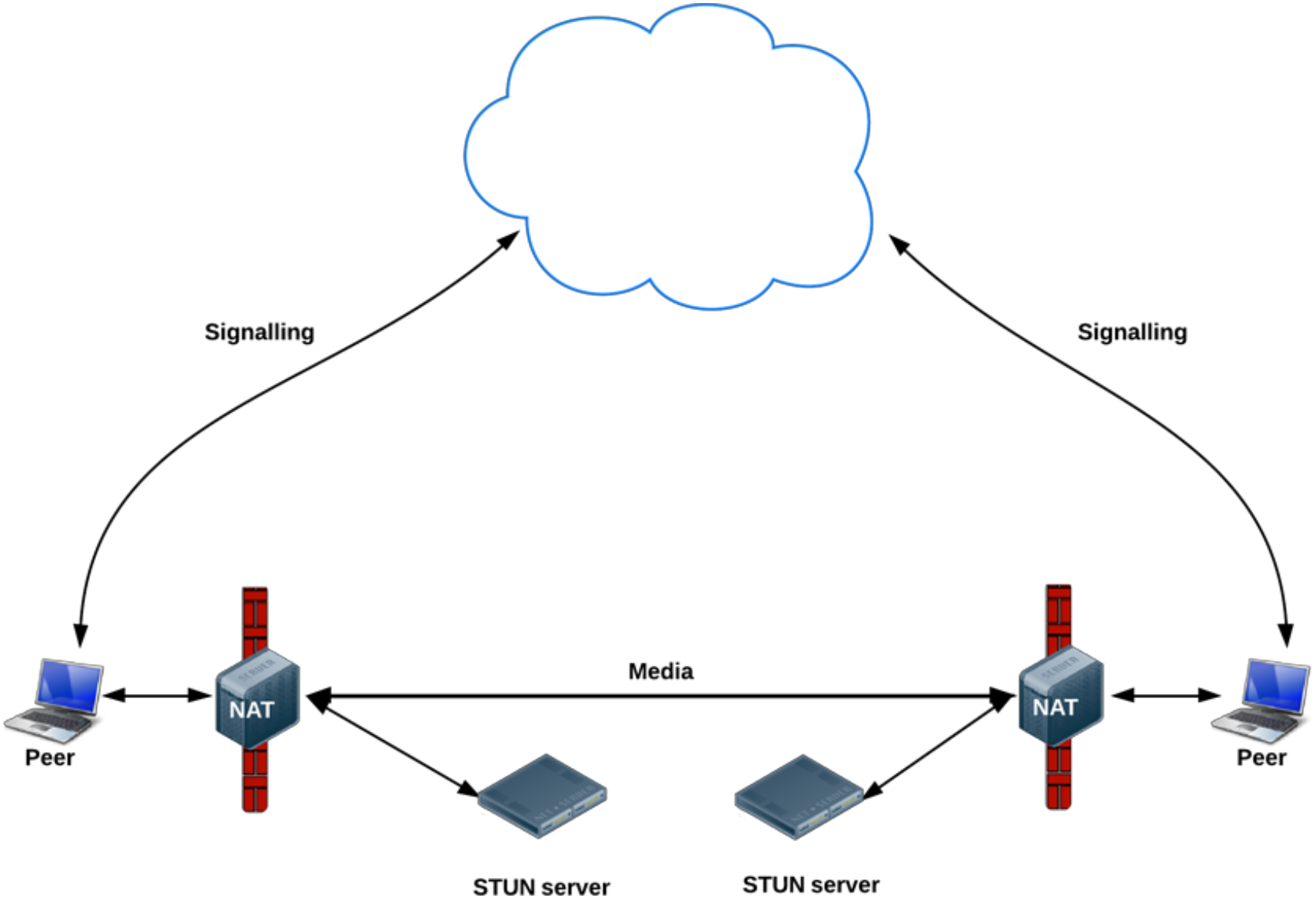
The real world



STUN

- Tell me what my public IP address is
- Simple server, cheap to run
- Data flows peer-to-peer

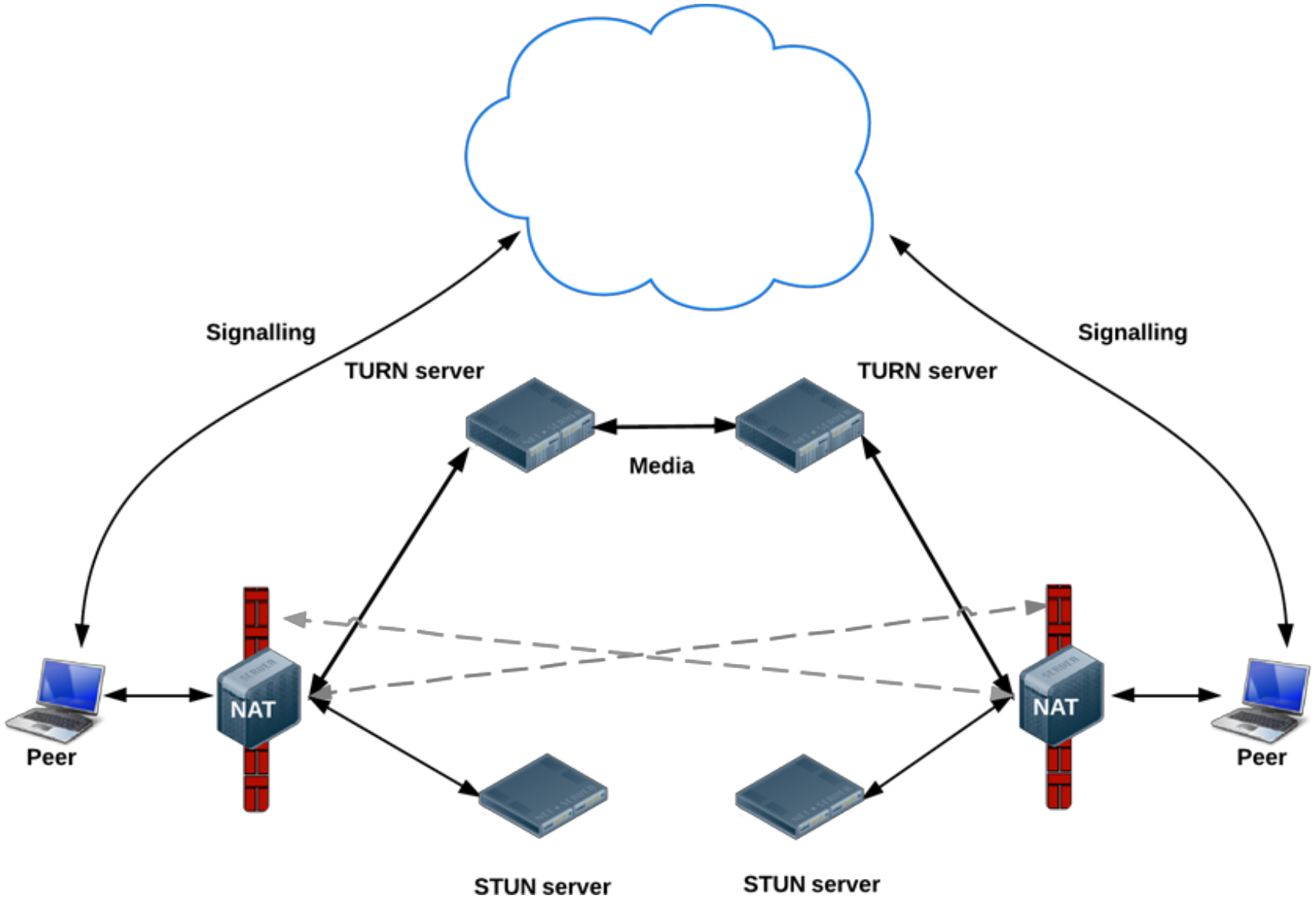
STUN


Signalling
Signalling
Media
NAT
NAT
Peer
Peer
STUN server
STUN server

TURN

- Provide a cloud fallback if peer-to-peer communication fails
- Data is sent through server, uses server bandwidth
- Ensures the call works in almost all environments

TURN


Signalling
TURN server
TURN server
Signalling
Media
NAT
NAT
Peer
Peer
STUN server
STUN server

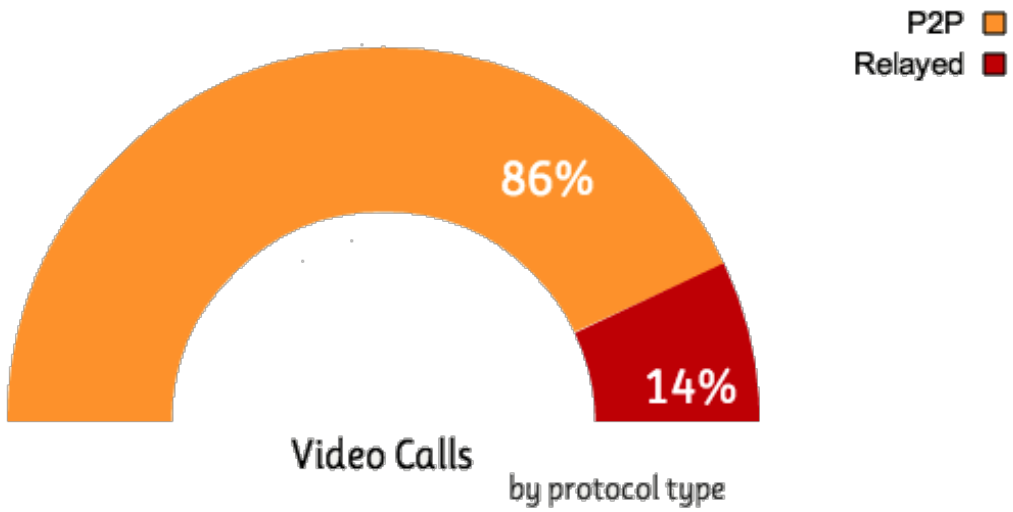
Selecting STUN and TURN servers

JAVASCRIPT

```
{
  "iceServers": [
    {
      "url": "stun:stun.l.google.com:19302"
    },
    {
      "url": "turn:192.158.29.39:3478?transport=udp",
      "credential": "JZEOEt2V3Qb0y27GRntt2u2PAYA=",
      "username": "28224511:1379330808"
    },
    {
      "url": "turn:192.158.29.39:3478?transport=tcp",
      "credential": "JZEOEt2V3Qb0y27GRntt2u2PAYA=",
      "username": "28224511:1379330808"
    }
  ]
}
```

ICE

- ICE: a framework for connecting peers
- Tries to find the best path for each call
- Vast majority of calls can use STUN (webrtcstats.com):



Deploying STUN/TURN

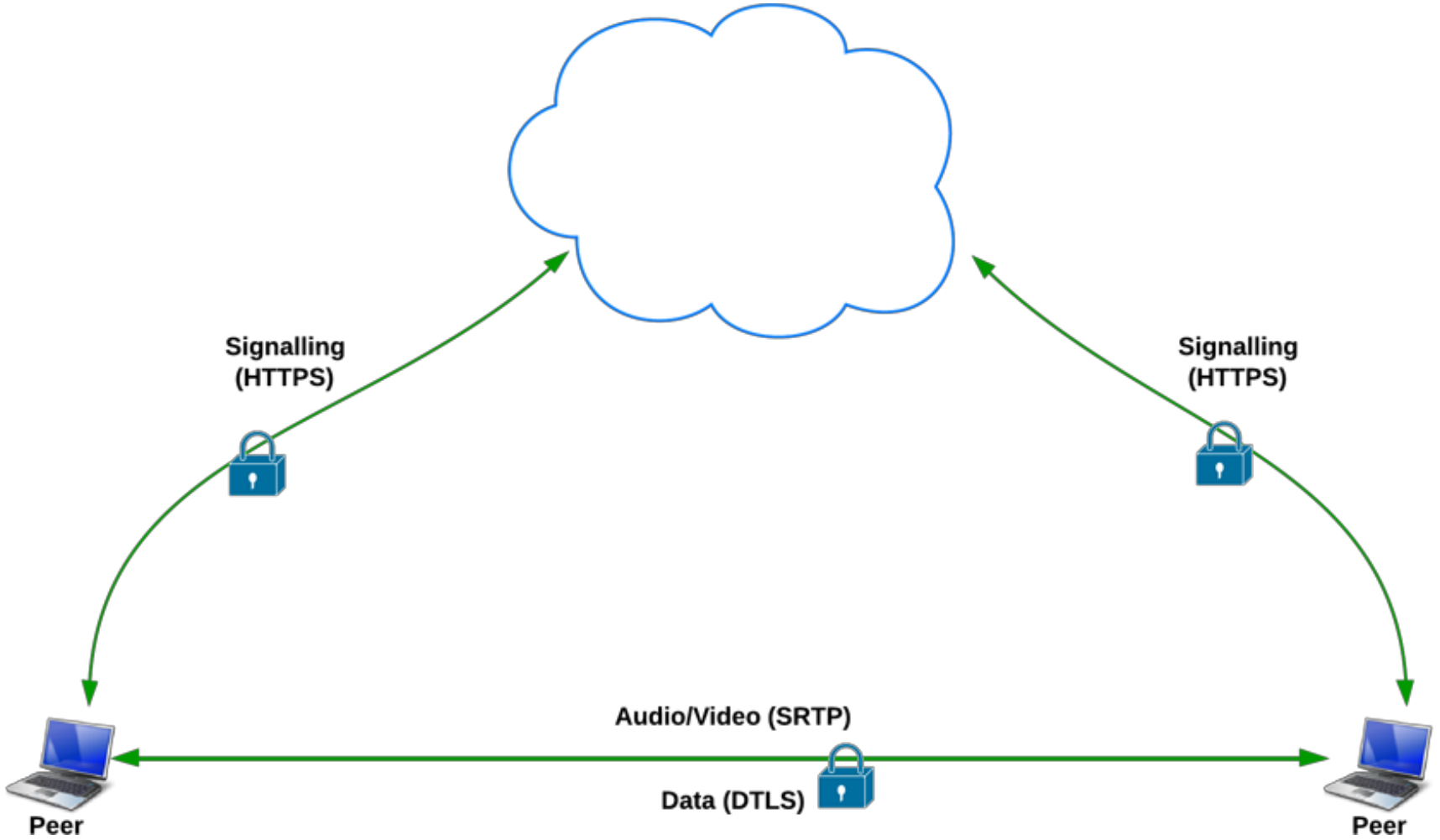
- stun.l.google.com:19302
- WebRTC stunserver, turnserver
- rfc5766-turn-server
- VM image for Amazon Web Services
- restund

Security

Security throughout WebRTC

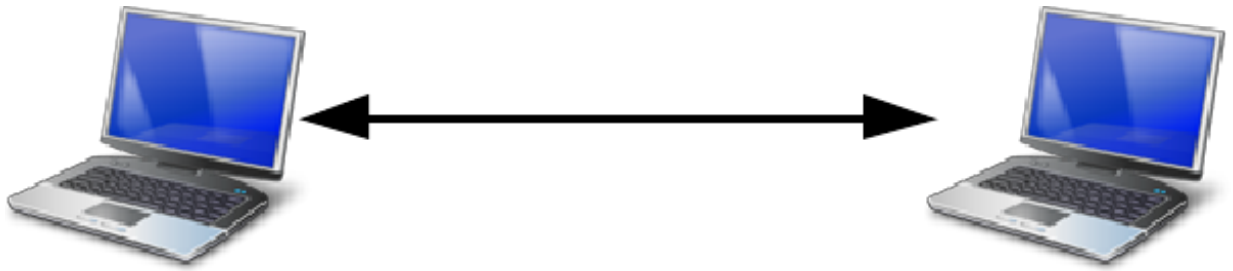
- Mandatory encryption for media and data
- Secure UI, explicit opt-in
- Sandboxed, no plugins
- WebRTC Security Architecture

Secure pathways

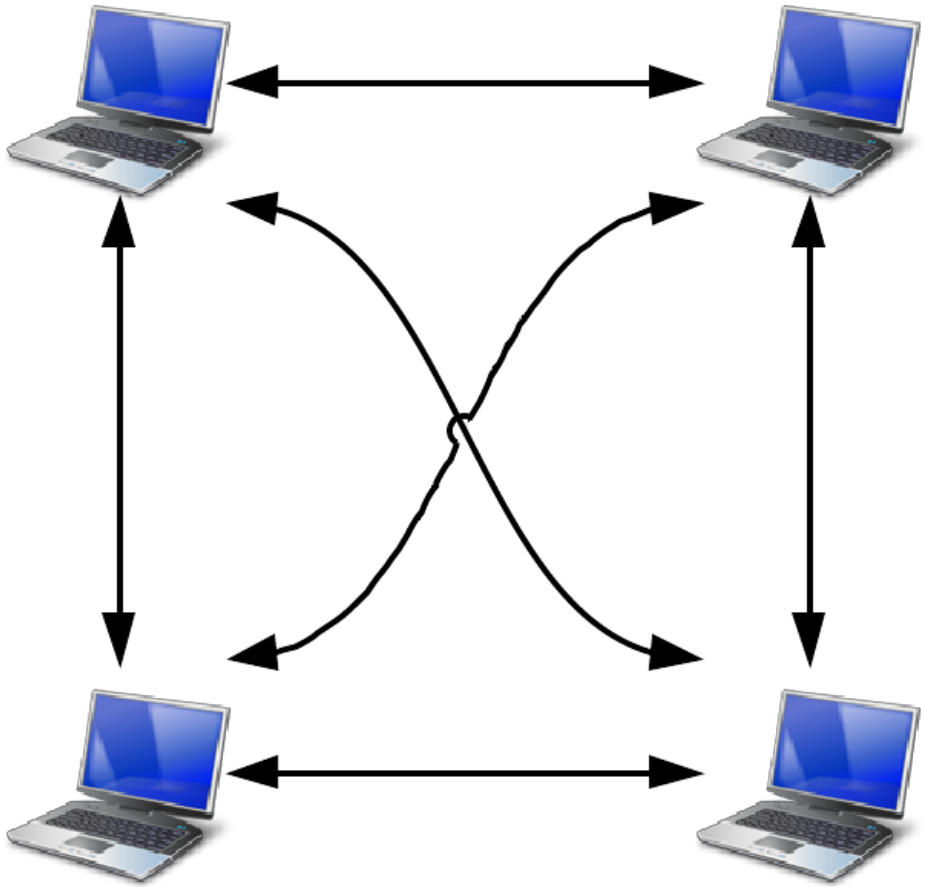


Architectures

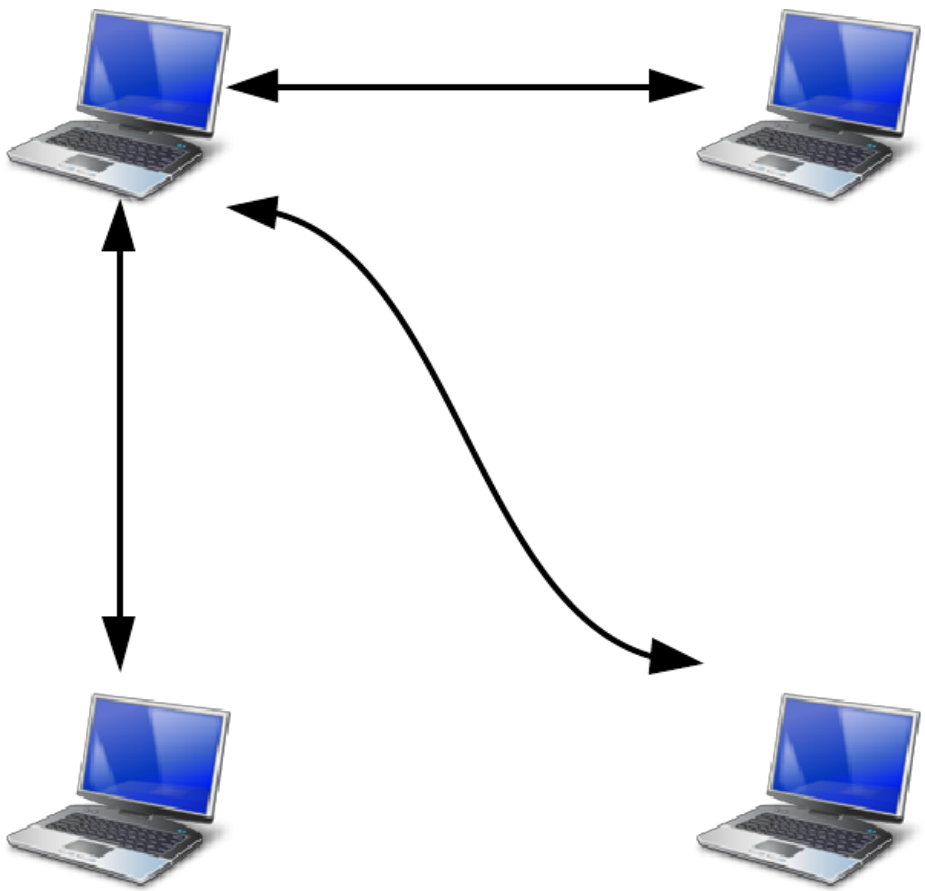
Peer to peer: one-to-one call



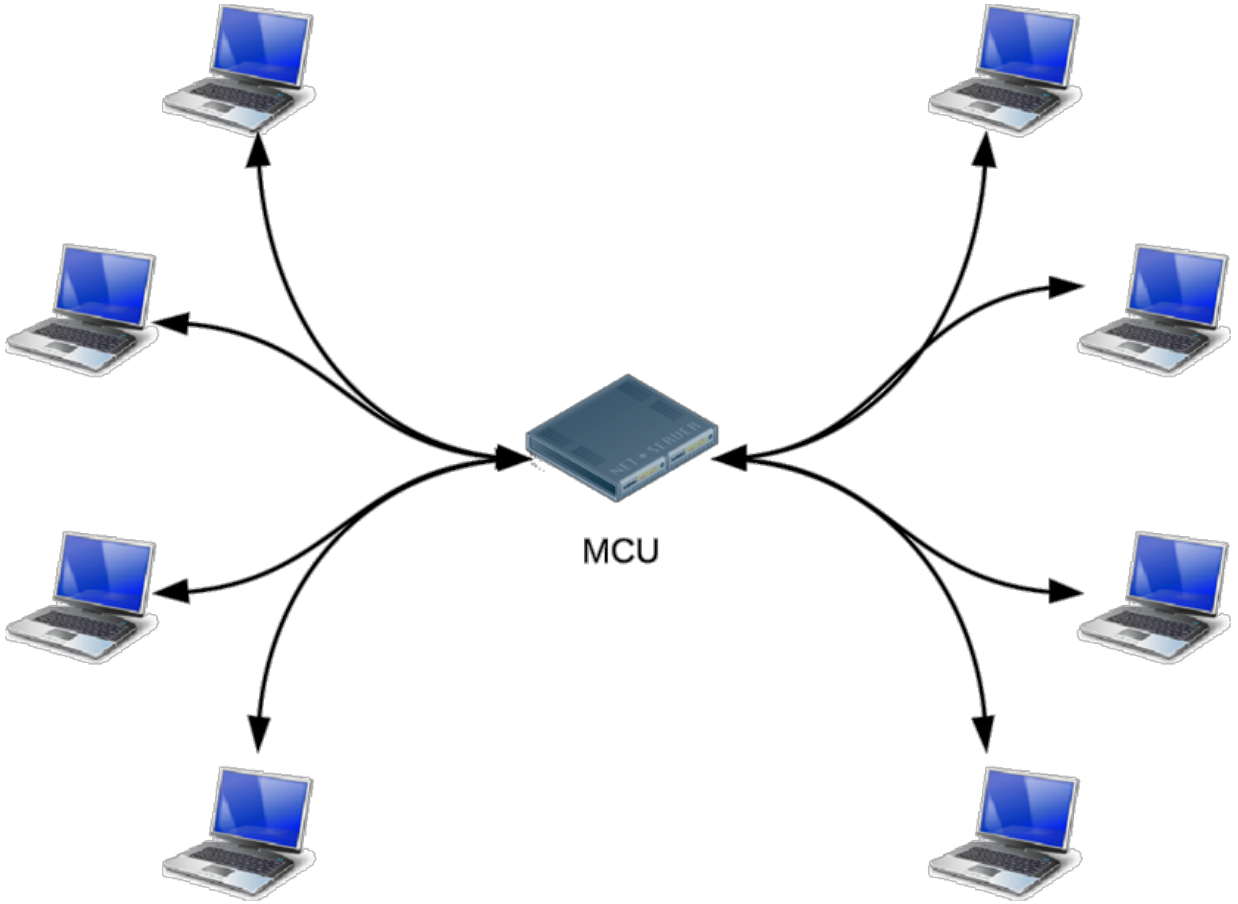
Mesh: small N-way call



Star: medium N-way call



MCU: large N-way call



Beyond browsers

Phones and more

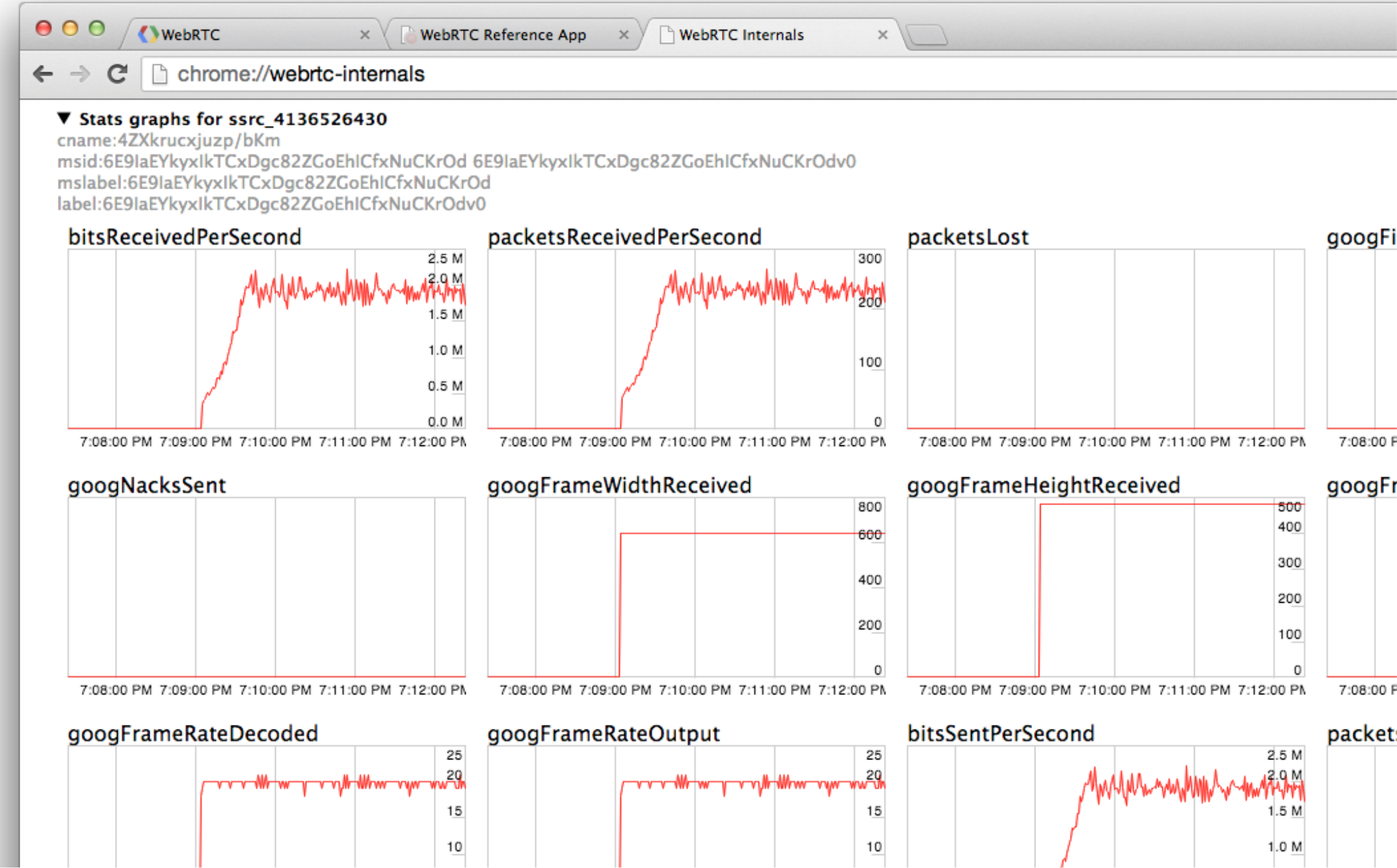
- Easy to interoperate with non-browser devices
  - sipML5 open source JavaScript SIP client
  - Phono open source JavaScript phone API
  - Zingaya embeddable phone widget

Zingaya PSTN

Tethr

Building a WebRTC app

chrome://webrtc-internals



adapter.js

Lets you use the same code in all browsers:

- Removes vendor prefixes
- Abstracts Chrome/Firefox differences
- Minimizes effects of spec churn

This is doing my head in.

JavaScript frameworks

- Video chat:
  - SimpleWebRTC
  - easyRTC
  - webRTC.io
- Peer-to-peer data:
  - PeerJS
  - Sharefest

SimpleWebRTC

Easy peer-to-peer video and audio

JAVASCRIPT

```
var webrtc = new WebRTC({
  localVideoEl: 'localVideo',
  remoteVideosEl: 'remoteVideos',
  autoRequestMedia: true
});

webrtc.on('readyToCall', function () {
    webrtc.joinRoom('My room name');
});
```

PeerJS

Easy peer-to-peer data

JAVASCRIPT

```
var peer = new Peer('someid', {key: 'apikey'});
peer.on('connection', function(conn) {
  conn.on('data', function(data){
    // Will print 'hi!'
    console.log(data);
  });
});

// Connecting peer
var peer = new Peer('anotherid', {key: 'apikey'});
var conn = peer.connect('someid');
conn.on('open', function(){
  conn.send('hi!');
});
```

Complete services

- OpenTok (acquired by Telefonica Digital)
- vLine

Don't forget the C++!

webrtc.org/webrtc-native-code-package

More Information

- WebRTC and Web Audio resources list: bit.ly/webrtcwebaudio
- Google I/O 2013 WebRTC presentation
- Justin Uberti: Google I/O 2012 presentation video
- Cullen Jennings video: HTML5 WebRTC
- HTML5 Rocks:
  - Capturing audio and video in HTML5
  - Getting Started With WebRTC
  - Updates
- ...and a book: webrtcbook.com

Contact Us

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- @webrtc
- crbug.com/new

“WebRTC and HTML5 could enable the same transformation for real-time communications that the original browser did for information.”

Phil Edholm
— NoJitter

talky.io/gowebrtc

gowebrtc.appspot.com

<Thank You!>

g+	plus.samdutton.com
twitter	@sw12
www	www.samdutton.com
github	github.com/samdutton


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