WebRTC

Real Time Collaboration with JS & HTML5

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Amazon, @WalmartLabs, Adobe Systems

Loves the Web; believes moving it forward

JavaScript freak, HTML5 Aficionado

Usually, press right key. When bottom arrow glows, press down arrow key

I just want Internet, What the hell is HTML5

Web as an Application

Don't you want to Communicate?

Export, Share, Store, Edit?

Record & Play videos and songs?

Drag Drop, 3D, Flexible layouts, Create and Draw?

What are these?

You tell me! You tell me right now!!

HTML5 Features

Semantics, Offline & Storage, Multimedia, 3D & Graphics (WebGL, SVG), CSS3 (yes that's part of HTML5 Spec), Performance & Integration (CORS, AJAX2, WebWorkers), Connectivity (Real-time, WebSockets & Push), , Device Acces (Geolocation, Tilt, etc.)

What is WebRTC

Web Real Time Communication:

An HTML5 Featureset enabling Real Time P2P communication ...

natively in Browsers - without any plugins.
with support for data Sharing, Audio and Video Chat, etc.
API Standards and Specs being maintained by **W3C**.

Where is JavaScript?

- 3 main actions required for WebRTC on Client Side (Browser) are performed by JavaScript:

- Acquiring the Video and Audio Streams
  `getUserMedia` returning `MediaStream` *(JS Object pointing a live stream being fed to it)*
- Sharing Streams with peers - AKA Communicating !
  `RTCPeerConnection` `RTCIceCandidate`
- Sharing data with peers
  `RTCDataChannel`

Acquiring Video and Audio

`getUserMedia`

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

navigator.getUserMedia(constraints, successCallback, errorCallback);

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

function errorCallback(error) {
  alert("Error: ", error);
}
```

- `constraints` - Must for getusermedia, specifies Media type, resolution, etc.
- `video` - An HTML5 DOM element capable of playing live feed

Communicating Streams

`RTCPeerconnection`

```
var peerCon = new window.RTCPeerConnection(configuration);

peerCon.addStream(localStream); // got from getUserMedia
peerCon.onaddstream = function(e) {
...

peerCon.addIceCandidate(candidate);
peerCon.onicecandidate = function(e) {
...

peerCon.setLocalDescription(description);
peerCon.setRemoteDescription(new RTCSessionDescription(msg.data));

peerCon.createOffer(IceCandidate)
peerCon.createAnswer(...)
```

- `IceCandidate` - ICE - f/w to allow peers to communicate behind Firewalls, explained later.
- `RTCSessionDescription` - Description of multimedia content, following SDP, explained later.

Communicating Data

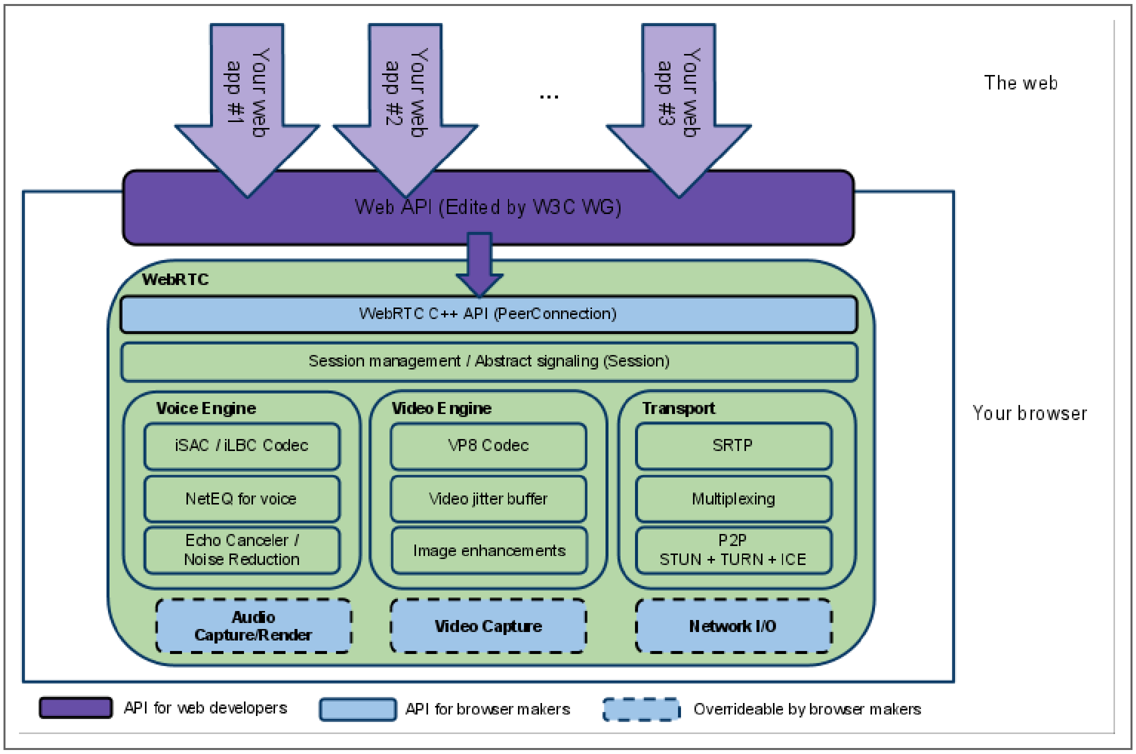
`RTCDataChannel`

```
var peerCon = new window.RTCPeerConnection(
  servers, {
    optional: [{ RtpDataChannels: true }]
  }
);

peerCon.ondatachannel = function(event) {
  receiverChannel = event.channel;
  receiverChannel.onmessage = function(event){
    receiveDOMElement.innerHTML = event.data;
  };
};

XmitterChannel = pc.createDataChannel("sendDataChannel", {reliable: false});
XmitterChannel.send('Hello WebRTC');
```

- `RtpDataChannels: true` - To enable arbitrary data via the same RTCPeerConnection.
- `onmessage`, etc. - Same API as WebSockets.

WebRTC & the Web

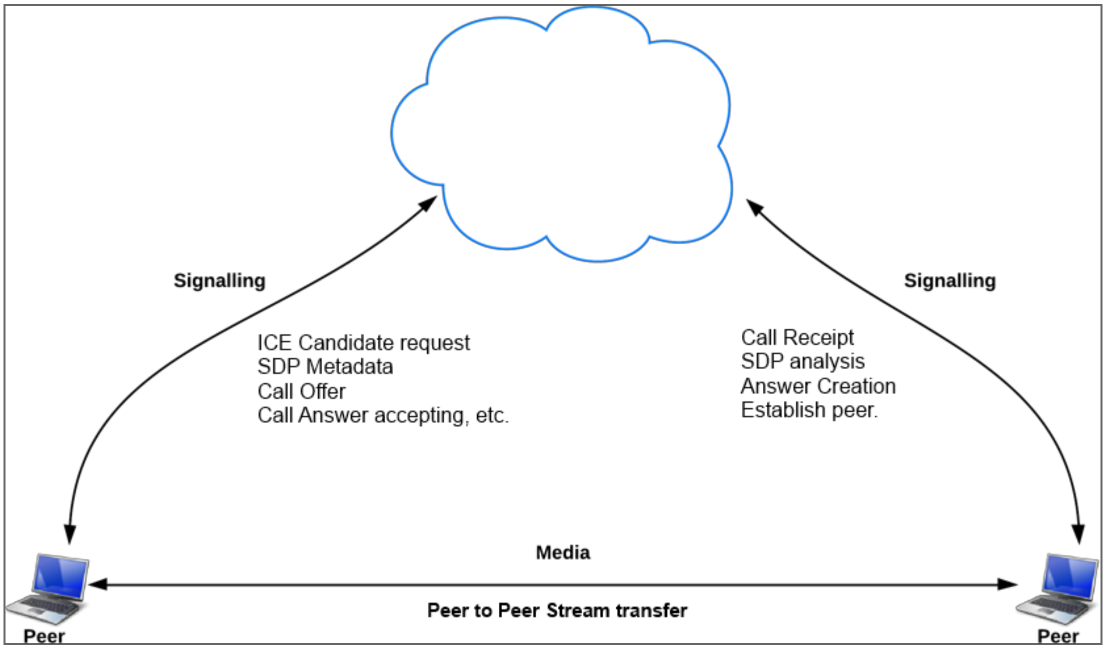
Architecture



I don't want to see that image again!

WebRTC: No Firewalls

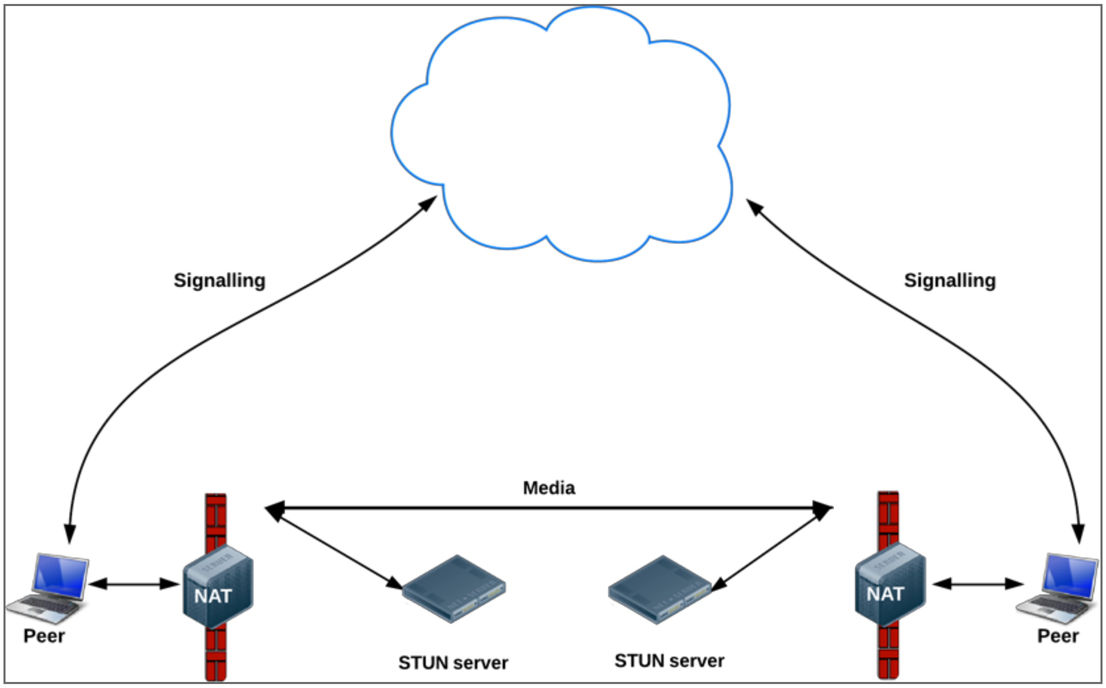
STUN and TURN not needed



Courtesy : WebRTC - Google IO; and Self Modified

WebRTC: Behind NAT Firewall

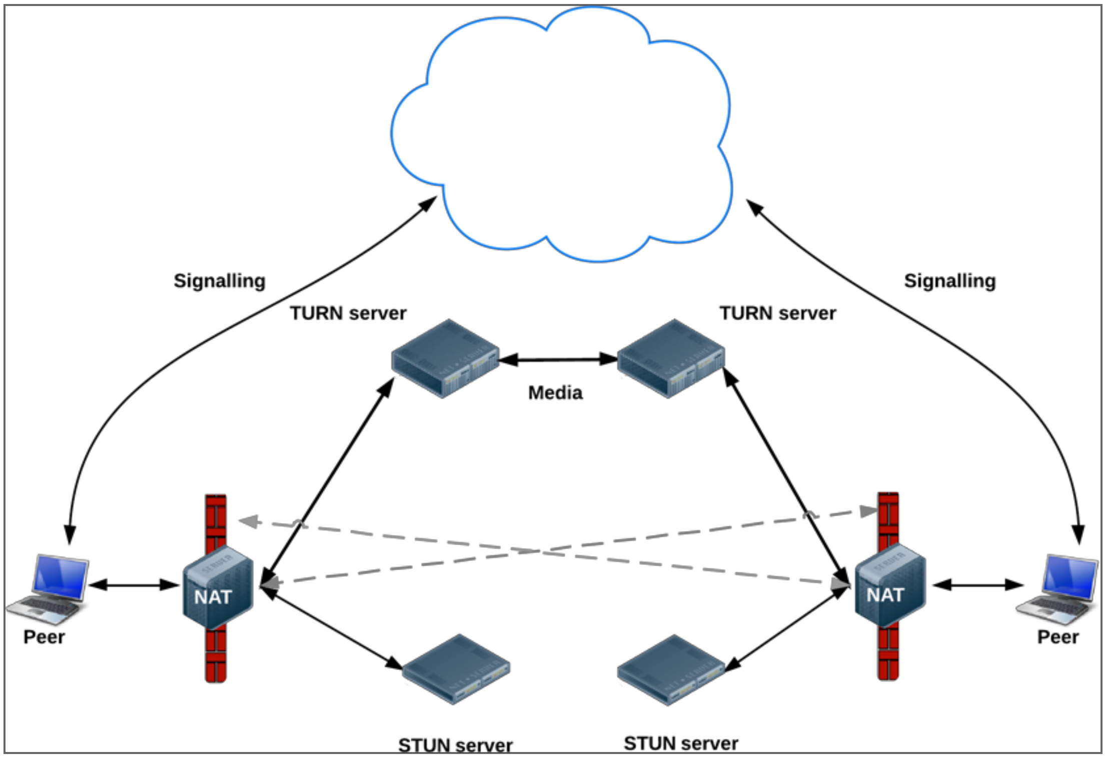
Solution 1: Use STUN



Courtesy : WebRTC - Google IO

WebRTC: STUN Fails

Solution 2: Use TURN



Courtesy : WebRTC - Google IO

What is STUN, TURN, SDP and ICE?

You tell me! You tell me right now!!

WebRTC: Stuff

Terminologies

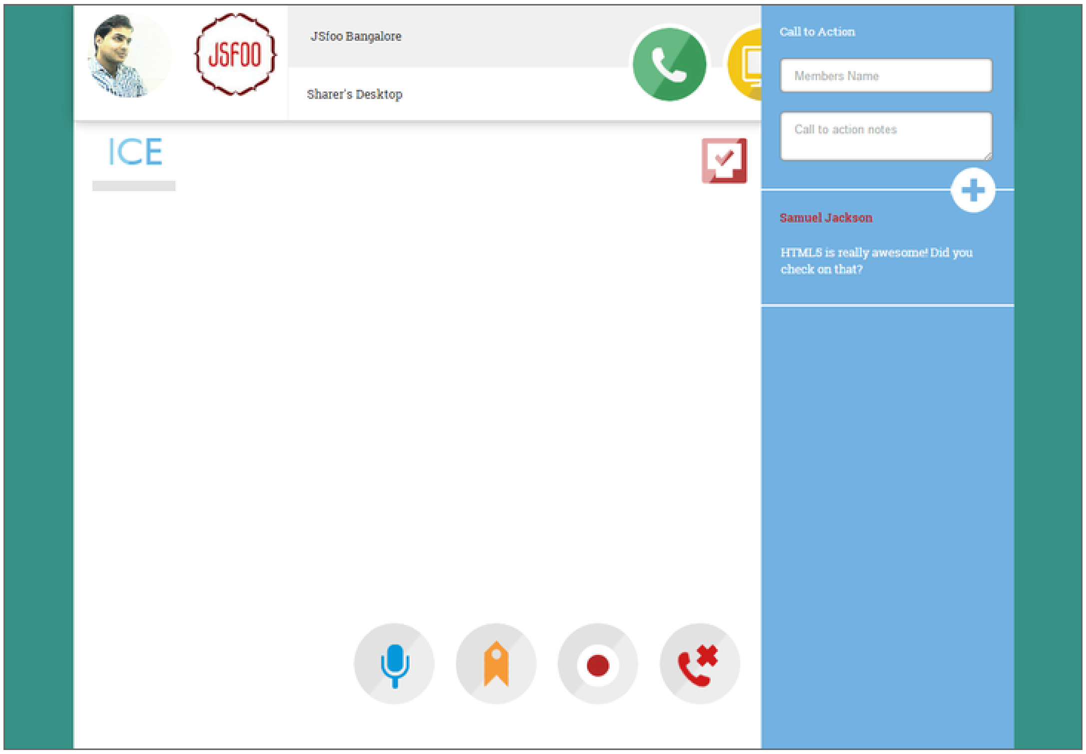
- `NAT` - Network Address Translation
  Is used to give the client a public address behind firewall.
- `ICE` - Interactive Connectivity Establishment
  Is a framework to allow your web browser to connect with peers behind NAT. How? Speaking...
- `STUN` - Session Traversal Utilities for NAT
  Protocol for Client Discovery.
- `TURN` - Traversal Using Relays around NAT
  ICE opens a connection with a TURN server and relays all information through that.
- `Signal Channel/Signaling` - Usually a PUSH enabled communication channel
  Required for exchanging Network info., Call Offer, Call Answer and SDP.
- `SDP` - Session Description Protocol
  Standard for describing the multimedia content of the connection such as resolution, formats, codecs, encryption, etc. and the connection itself.

WebRTC: Stuff

Over the Internet

- **ICE Breaker** - JSFoo Demo Application
- **WebRTC landing page**
- **WebRTC roadmap** in Chrome
- **Mozilla** - the HTML5 pillar
- **NVIDIA Tegra High def conferencing** using WebRTC
- **World's first WebRTC enabled mobile browser** - by Ericsson R&D labs
- **3D communication in browser** - by Ericsson R&D labs

ICE Breaker


JSfoo Bangalore
Sharer's Desktop
ICE
Call to Action
Members Name
Call to action notes
Samuel Jackson
HTML5 is really awesome! Did you check on that?

Cool Extras

- **ASCII Camera**
- **Audio Recorder**
- **ShareFest** - Send files Directly, P2P
- **Who's Talking** - Detecting who is speaking in a Conference
- **GIF image from Video** - Pure JavaScript and HTML5

`JavaScript` + `HTML5` = Awesomeness

Thank You

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