OST
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Blockchain (BlCh)

WebRTC

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WebRTC – Introduction

- WebRTC for browser to browser communication
  - P2P, no server involved (~mostly)
  - Real time communication (RTC) via API
  - Main goal: eliminate plugins or native apps
  - Supported by Google, Microsoft, Mozilla, Opera, Apple
- Google bought in 2010 Global IP Solutions (GIPS) and open sourced WebRTC in 2011
- Protocol standardized by IETF (codec requirements, media protocol), JavaScript API by W3C
- Supported initially by Chrome, Firefox (now many others)
  - First cross-browser call in February 2013

WebRTC

- Compatibility, 93%
  - Microsoft Edge 15~
  - Google Chrome 56+
  - Mozilla Firefox 44+
  - Safari 11+
  - Opera 43+
- PeerJS
  - Not all browser work 100% compatbile with each other → PeerJS
  - «PeerJS provides a complete, configurable, and easy-to-use peer-to-peer API built on top of WebRTC»
- SimpleWebRTC – archived
  - Use webrtc directly?

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WebRTC – Introduction

- Filling gap in the Web-Experience
  - Video Chat → Google Hangouts Plugin, Flash, Java
  - Multimedia / Conferences → Expensive, proprietary 3rd party apps
  - Customer Service → Chat only, 3rd party plugins/apps
  - Online Games → Flash
  - Real-time Feeds → Proprietary software
  - File Sharing → Requires Server / BitTorrent
- WebRTC widely deployed, no client necessary!
- Used in WhatsApp, Facebook Messenger, whereby.com
- Standard still not finalized
  - W3C Candidate Recommendation 03 September 2020 (W3C process flow)
  - «The RTCPeerConnection API has endured three design iterations on this topic over the years. As a result, each browser today implements a snapshot from a different point in the timeline of an evolving spec.» (source) (other updates) (large messages, fixed)

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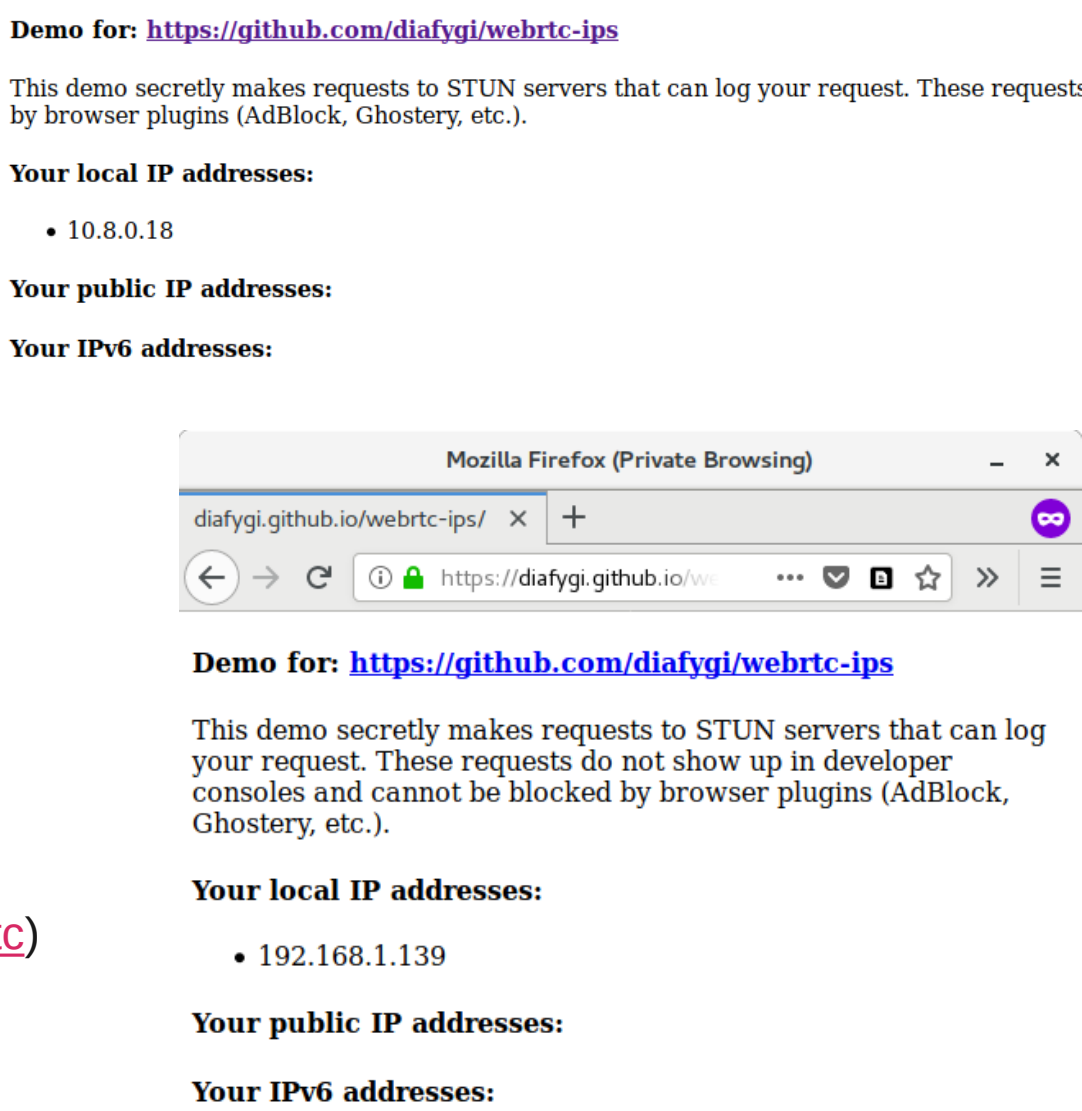
WebRTC – Concerns

- Outdated documentation (source):

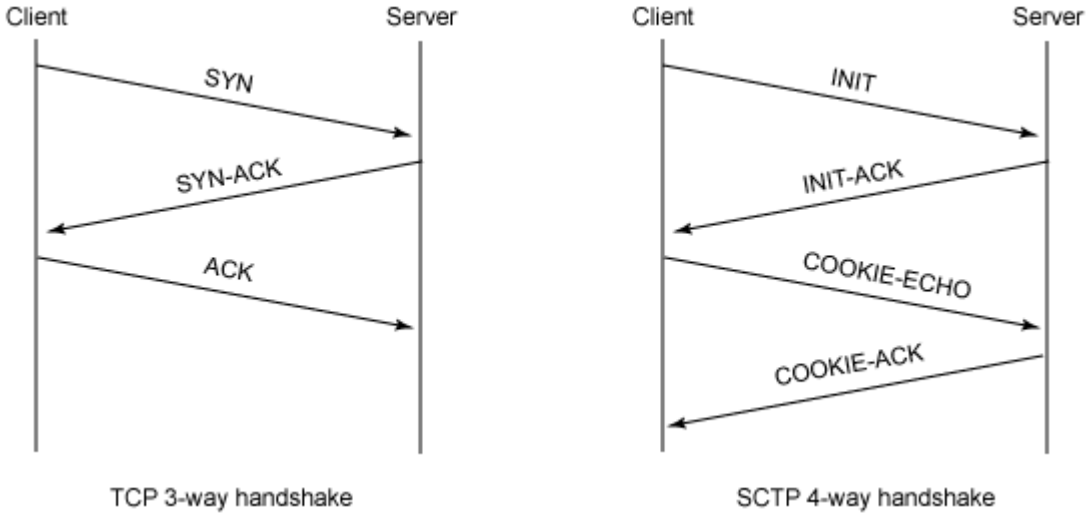
```
var dc = pc.createDataChannel("namedChannel", {reliable: false});
```

  - «reliable» is obsolete
- HTML browsers get bloated
  - Several GB RAM to open couple of tabs?
  - Hint: adblocker
- WebRTC API could be simplified
- Security Concerns
  - Private IP / IP behind VPN, Tor? (https://dsl.hsr.ch/lect/webrtc)
- But: WebRTC forbids unencrypted communication
  - DTLS, SRTP
  - Complexity - SCTP over DTLS over UDP



SCTP (Stream Control Transmission Protocol)

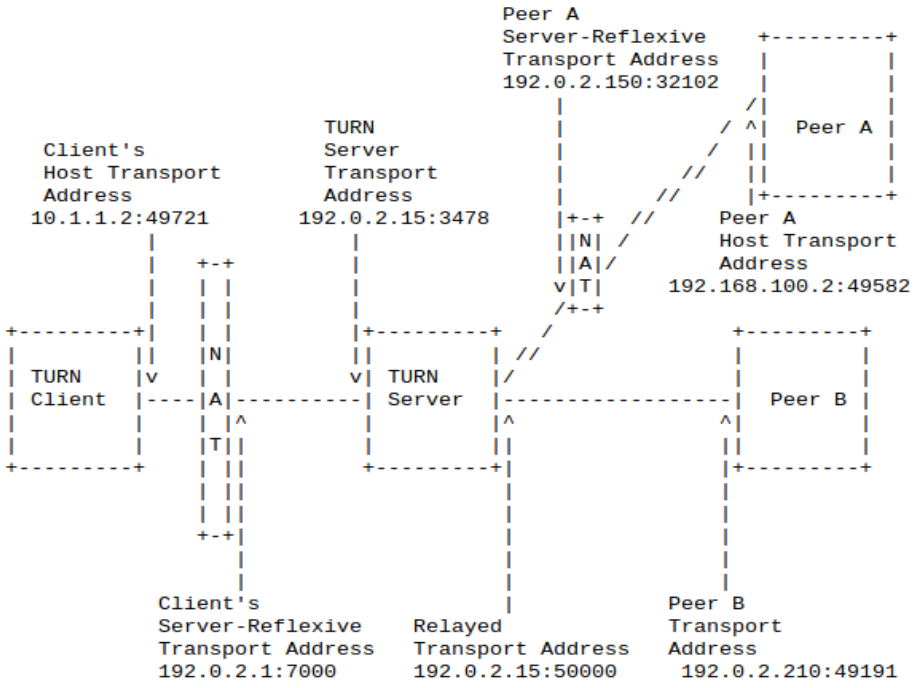
- SCTP Message-based (reliable) [rfc]
- Allows data to be divided into multiple streams
- Syn cookies - SCTP uses a four-way handshake with a signed cookie
- 32-bit end-to-end checksum (CRC32C)
- Multi-homing multiple IP addresses of endpoints
- Not widely used: "We have been deploying SCTP in several applications now, and encountered significant problem with SCTP support in various home routers."
  - E.g., OpenWRT – not enabled by default
  - E.g., UFW - Uncomplicated Firewall – not supported
- SCTP used by WebRTC

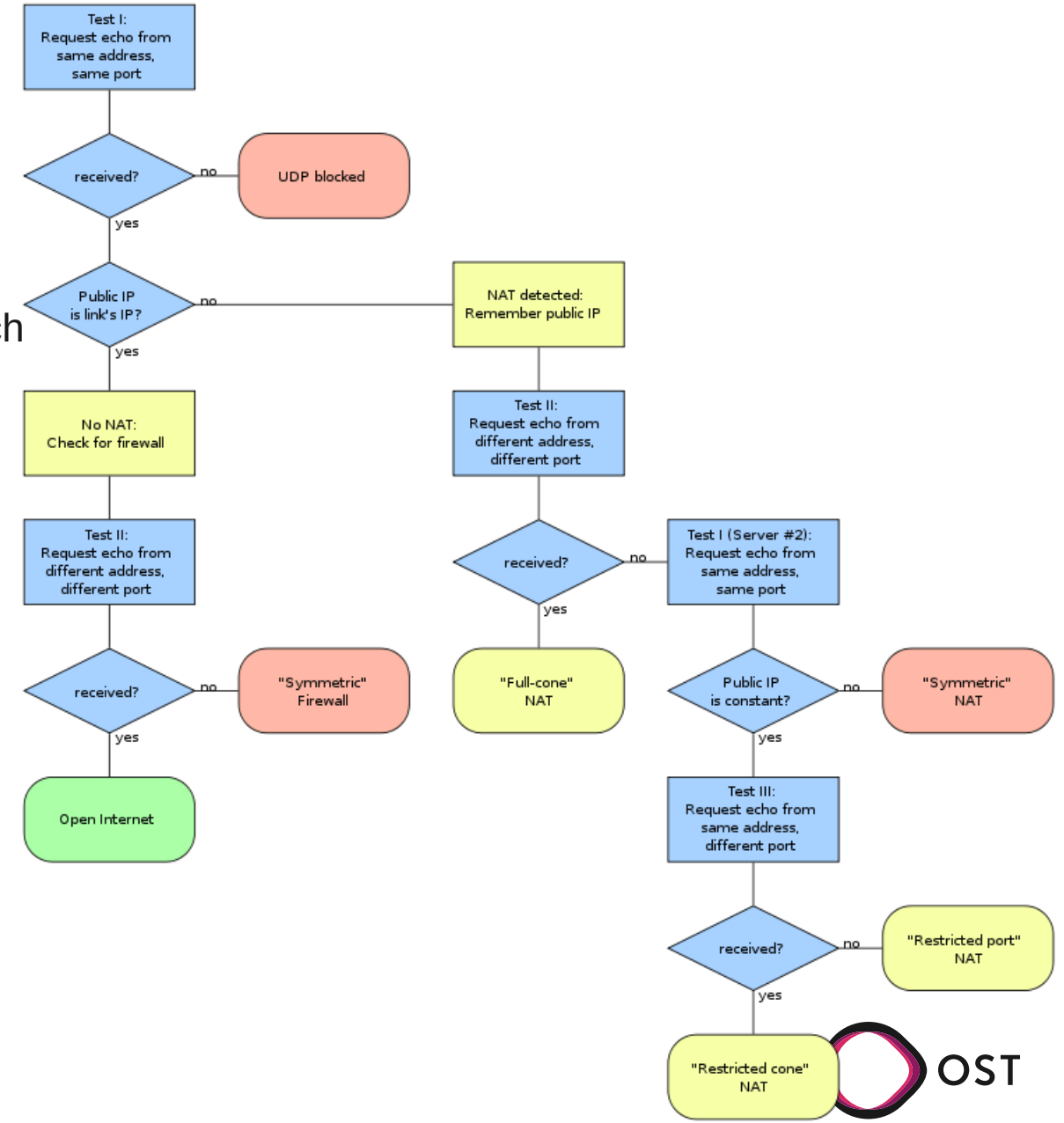


https://www.ibm.com/developerworks/library/l-sctp/

WebRTC – Introduction

- On the bright side: developer does not need to care about NAT
  - Abstraction using STUN, ICE, TURN
  - STUN: session traversal utilities for NAT (detect which kind of NAT, rfc5389)
    - “STUN is not a NAT traversal solution by itself”
  - TURN: **t**raversal **u**sing **r**elays around **N**AT





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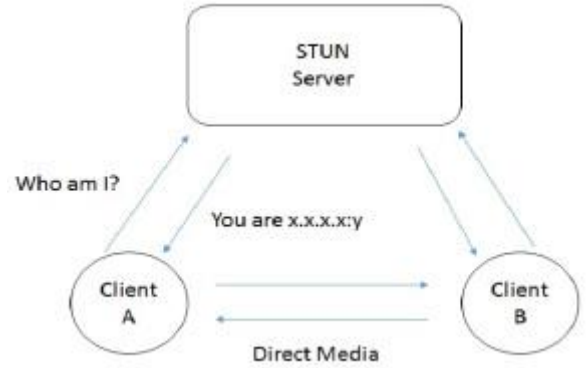
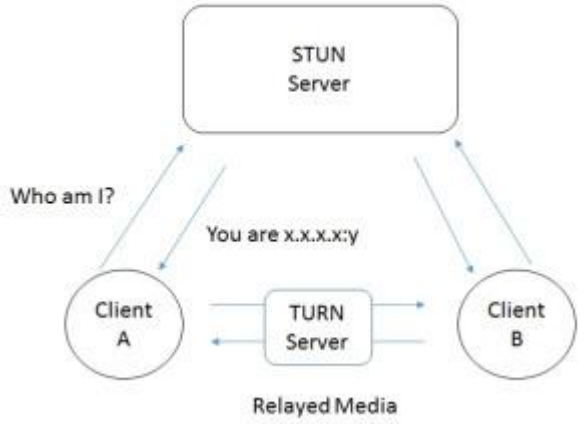
WebRTC – Introduction

- TURN
  - TURN always UDP server and the peer
  - TURN client/server UDP, TCP/TLS
    - Some firewalls block UDP entirely
  - UPnP / NAT-PMP setup by the browser optional?
    - Bugzilla@Mozilla – Bug 860045
  - TURN is a relay extension for STUN
- ICE - Interactive Connectivity Establishment
  - RFC 8445 a protocol for NAT traversal
  - “ICE works by exchanging a multiplicity of IP addresses and ports, which are then tested for connectivity by peer-to-peer connectivity checks.”

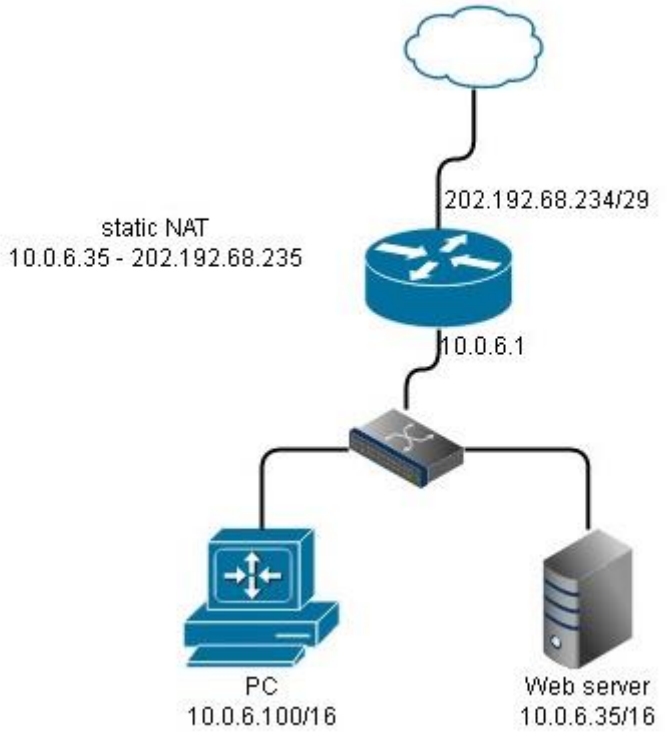




source

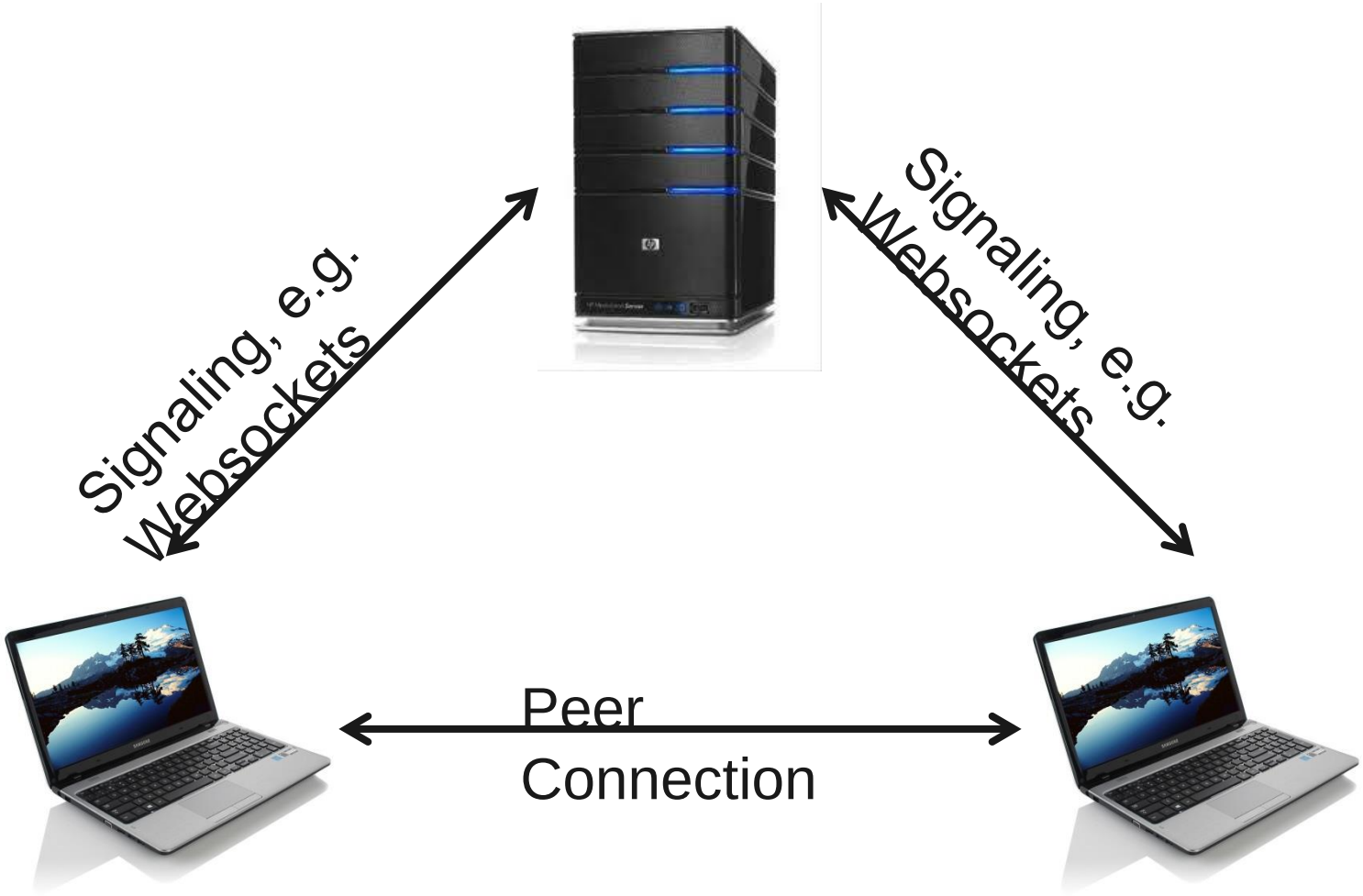
WebRTC – Connectivity

- NAT loopback: 2 devices behind same NAT
  - (peer-reflexive candidate)
- Encryption is mandatory for all WebRTC components
  - SRTP for Media, DTLS for Data, HTTPS for signaling (optional)
- WebRTC is not a plugin
  - Camera and microphone access must be granted explicitly
- Troubleshoot: test.webrtc.org
- Once connection is established – easy API
  - `RTCPeerConnection.send("hallo")`
  - `RTCPeerConnection.onmessage = function …`

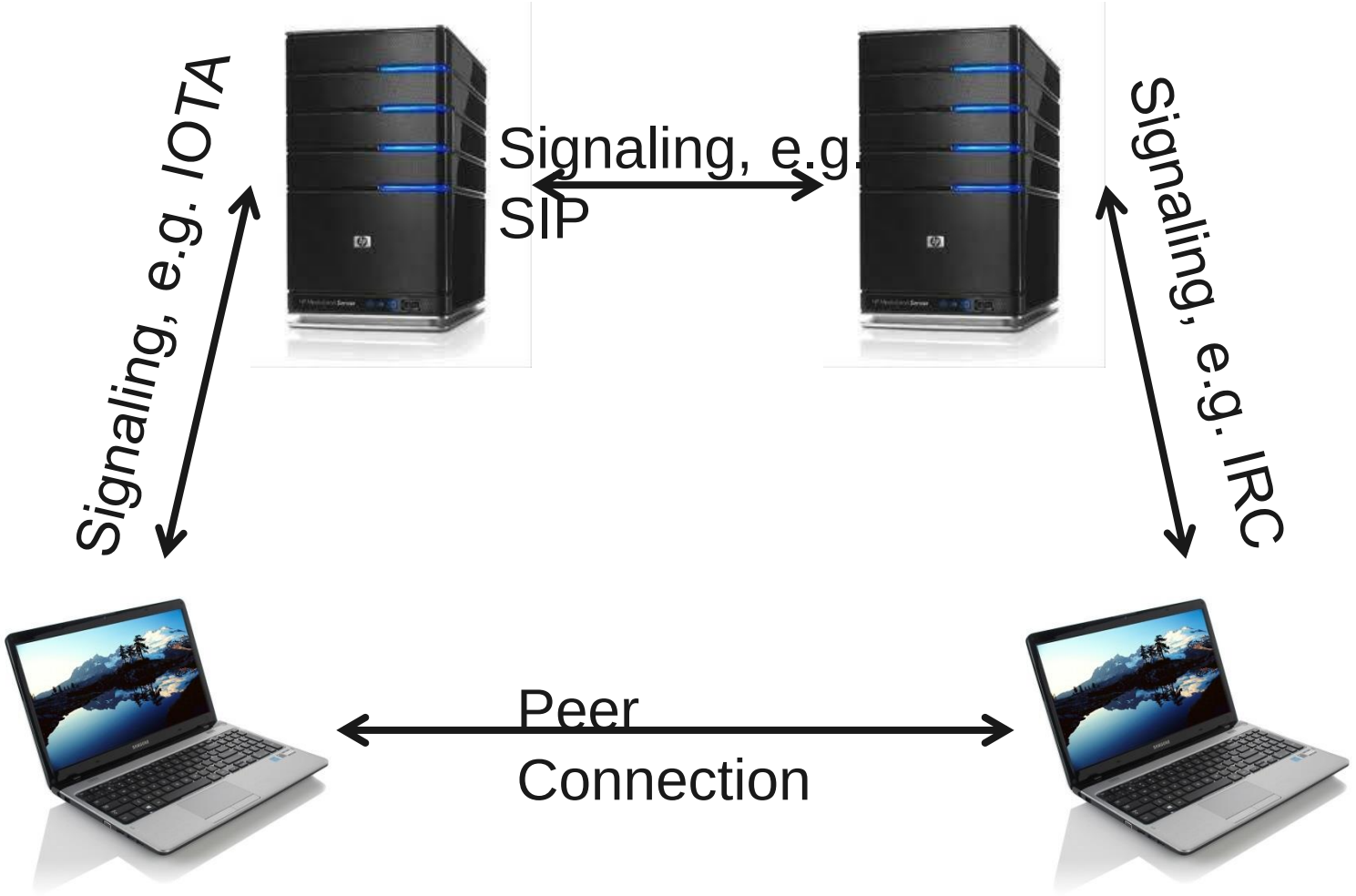


https://networkengineering.stackexchange.com/questions/25383/cisco-nat-hairpinning

WebRTC Architecture - Triangle

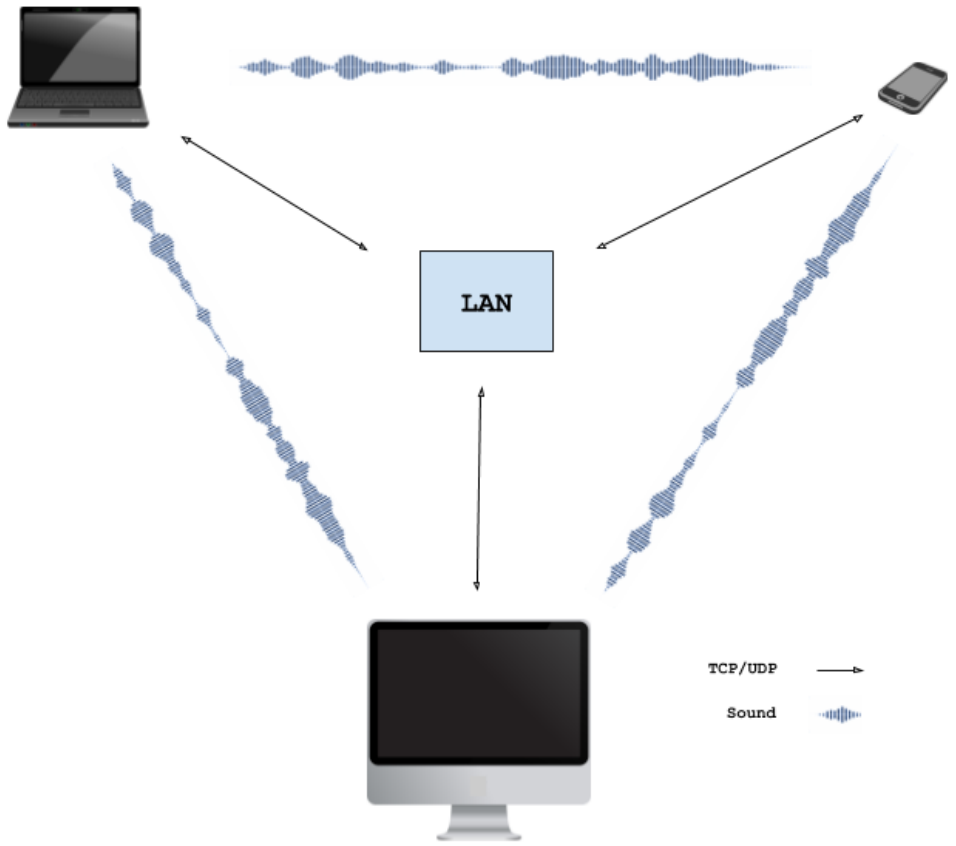


WebRTC Architecture - Trapezoid



WebRTC Signaling

- A proof-of-concept for WebRTC signaling using sound. Works with all devices that have microphone + speakers. Runs in the browser:
  - https://github.com/ggerganov/wave-share



WebRTC - Demo

- Server - server.js
  - Node.js server
  - Serves files (index.html / express)
  - Listens to incoming WS messages and broadcast messages to all connected clients
  - 19 loc
- Run
  - yarn install
  - node server.js
- Minimal example!

```
const express = require('express');
const WebSocket = require('ws');

let app = express();
// setup static files
app.use(express.static('.'));
// setup listening
let server = app.listen(4000, function () {
    console.log("listening on " + server.address().address + ":" +
server.address().port);
});

//WebSocket broadcast setup
const wss = new WebSocket.Server({ server });
wss.on('connection', function(ws) {
    ws.on('message', function(message) {
        // Broadcast any received message to all clients
        console.log('received: %s', message);
        wss.clients.forEach(function(client) {
            if(client.readyState === WebSocket.OPEN) {
                client.send(message);
            }
        });
    });
});

console.log('Server running.');
```

Example: https://github.com/tbocek/ADSB-webrtc

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WebRTC - Demo

- Client – index.html
  - 2 buttons, 2 text fields

- Client JavaScript
  - `peerConnection`, here we could add STUN/TURN
  - `createDataChannel`, create named channel. This needs to be done before offer/answer
  - `onicecandidate`, once offer was sent ICE candidates are sent

```
<body>

<button id="connectButton">Connect</button>
<label for="message_send">Enter a message:
    <input type="text" id="message_send" placeholder="Message
send" size=30 maxlength=30>
</label>
<button id="sendButton">Send</button>
<label for="message_rcv">Messages received:
    <input type="text" id="message_rcv" placeholder="Message
received" size=30 maxlength=30 disabled>
</label>

</body>
```

```
function startup() {
document.getElementById('connectButton').addEventListener('click
', connectPeers, false);
 document.getElementById('sendButton').addEventListener('click',
sendMessage, false);
peerConnection = new RTCPeerConnection();
peerConnection.onicecandidate = gotIceCandidate;
dataChannel = peerConnection.createDataChannel("test");
}
```

Example: https://github.com/tbocek/ADSB-webrtc

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WebRTC - Demo

- Client 1 JavaScript
  - `setLocalDescription()`
  - `serverConnection.send()`
- Server broadcasts local description to all
- Client 2
  - If SDP, set remote description
  - If offer, send answer

Example: https://github.com/tbocek/ADSB-webrtc

```
function connectPeers() {
 peerConnection.createOffer().then(createdDescription).catch(errorHandler)
}

function createdDescription(description) {
 console.log('Got description’);
 peerConnection.ondatachannel = gotDataChannel;
 peerConnection.setLocalDescription(description).then(function() {
 serverConnection.send(JSON.stringify({'sdp’:peerConnection.localDescription,
'uid': uid}));}).catch(errorHandler);
}
```

```
function gotMessageFromServer(message) {
 const signal = JSON.parse(message.data);
 if(signal.uid == uid) return; // Ignore messages from ourself
 console.log('Got message from signaling server: '+message.data);
 if(signal.sdp) {
  peerConnection.setRemoteDescription(new RTCSessionDescription(signal.sdp))
    .then(function() {
      // Only create answers in response to offers
      if(signal.sdp.type == 'offer') {
        peerConnection.createAnswer().then(createdDescription)
          .catch(errorHandler);
      }
    }).catch(errorHandler);
 } else if(signal.ice) {
  peerConnection.addIceCandidate(new RTCIceCandidate(signal.ice))
    .catch(errorHandler);
 }
}
```

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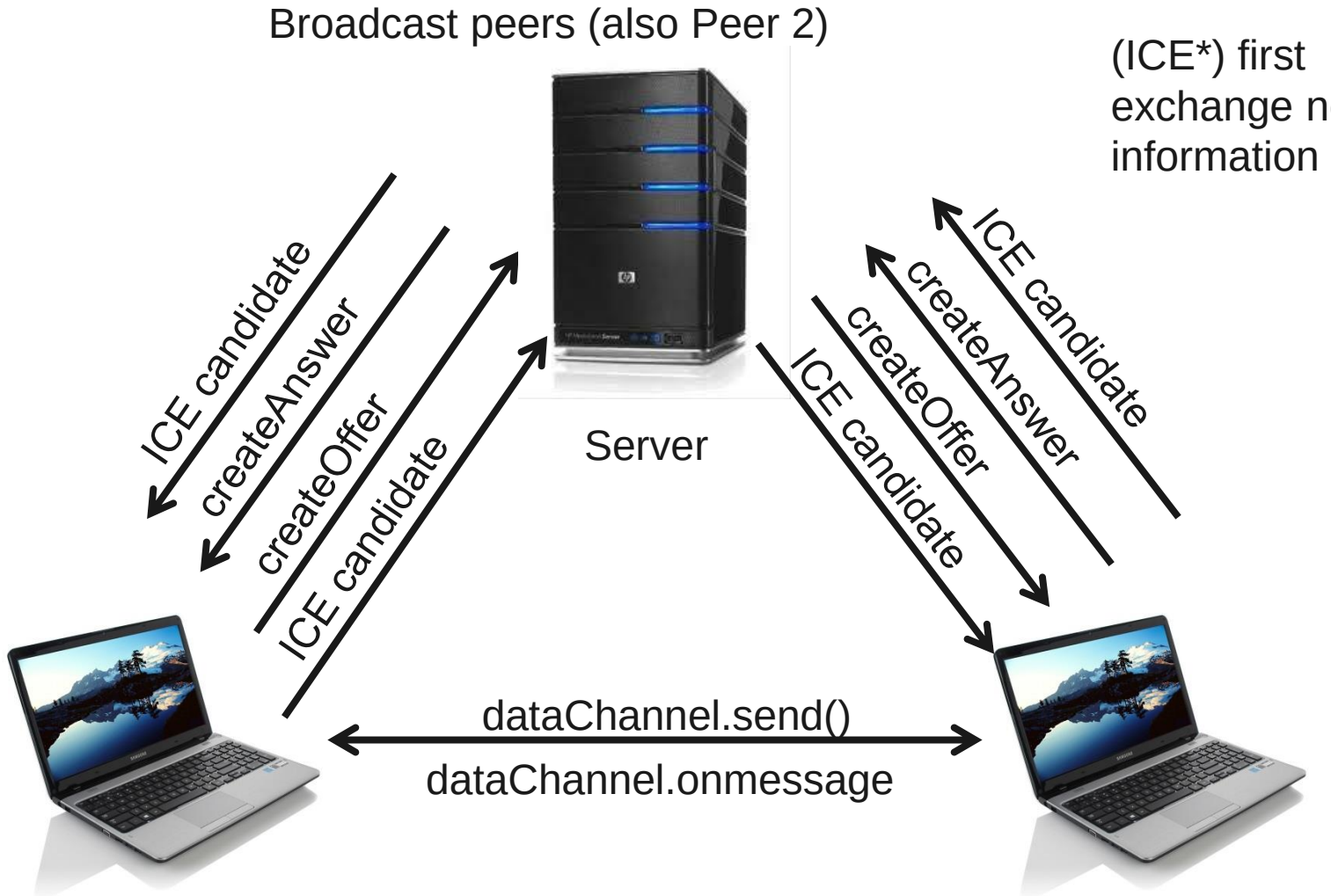
WebRTC Architecture - Demo



WebRTC – Outlook

- Strong focus on VoIP [demo]
  - Skype competitor? MS Teams?
  - Microsoft / IE / Edge and WebRTC?
  - SDP / signaling overhead if using with raw P2P data
- Fewer plugins (flash, java), fewer registrations
- Mandatory Codecs: VP8, H264
  - Video codec in JavaScript
- Web-based P2P frameworks
  - http://peerjs.com - make the API simpler, is it complex?
- New types of applications – Conferencing, gaming, P2P file sharing
  - PeerCDN, serve static content from browsers of other visitors (broken)
- ORTC, WebRTC 1.1?
  - Object Real-Time Communications (ORTC) instead Session Description Protocol (SDP)

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