

husseinnasser.com

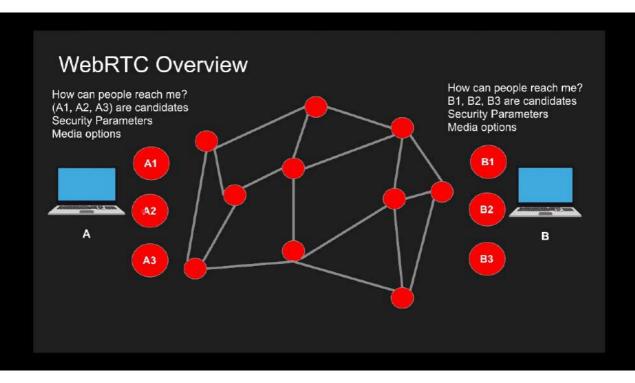
Web Real-Time Communication

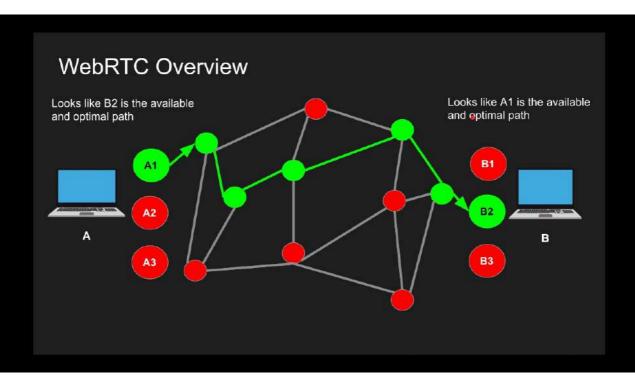
## WebRTC Overview

- Stands for Web Real-Time Communication
- Find a peer to peer path to exchange video and audio in an efficient and low latency manner
- Standardized API
- Enables rich communications browsers, mobile, IOT devices

#### WebRTC Overview

- A wants to connect to B
- A finds out all possible ways the public can connect to it
- B finds out all possible ways the public can connect to it
- A and B signal this session information via other means
  - WhatsApp, QR, Tweet, WebSockets, HTTP Fetch..
- A connects to B via the most optimal path
- A & B also exchanges their supported media and security



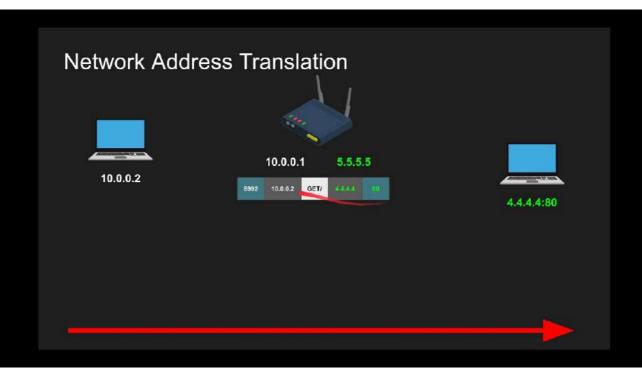


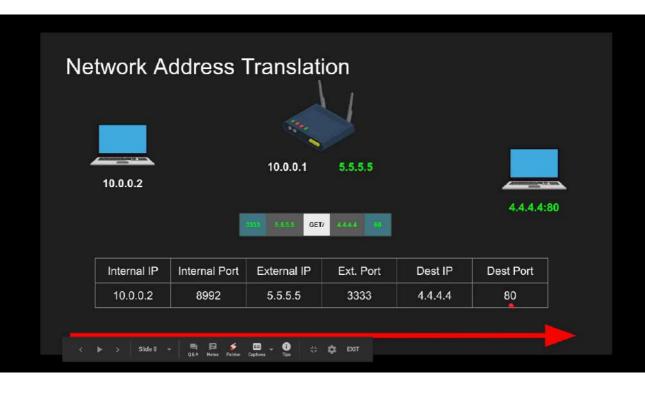
# WebRTC Demystified

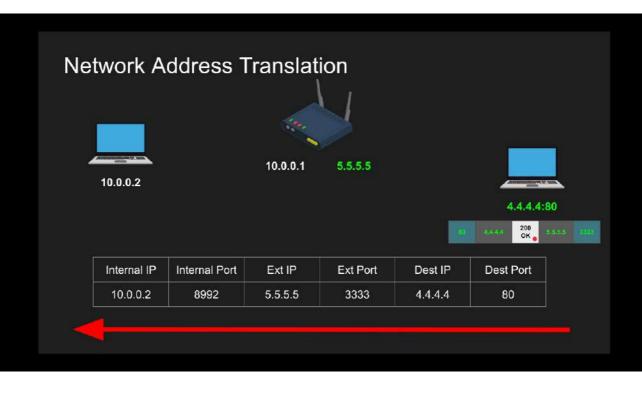
- NAT
- STUN, TURN
- ICE
- SDP
- Signaling the SDP

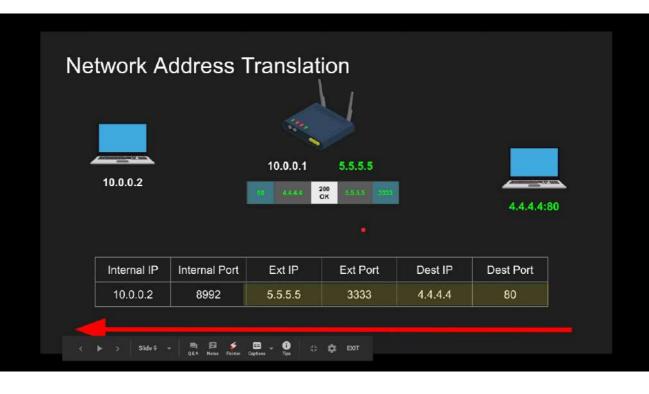


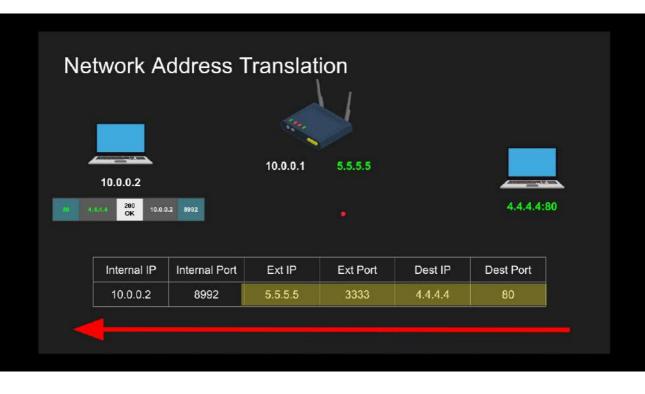












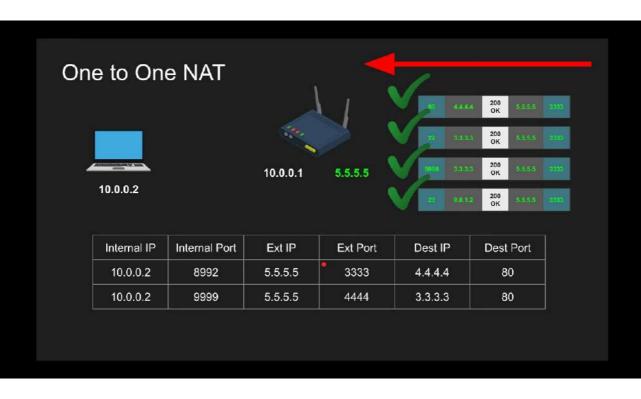
## **NAT Translations Method**

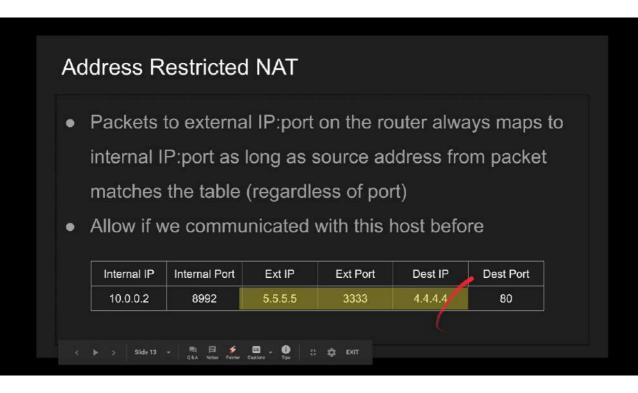
- One to One NAT (Full-cone NAT)
- Address restricted NAT
- Port restricted NAT
- Symmetric NAT

## One to One NAT (Full cone NAT)

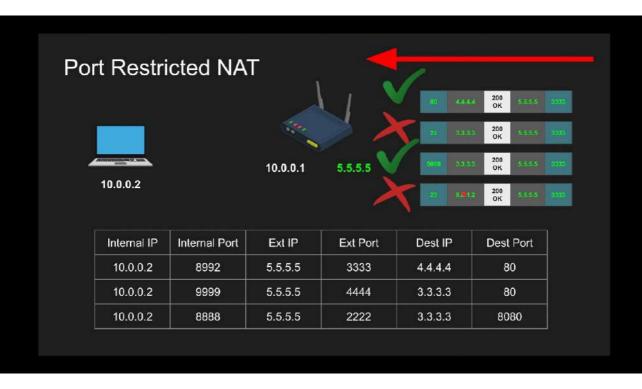
 Packets to external IP:port on the router always maps to internal IP:port without exceptions

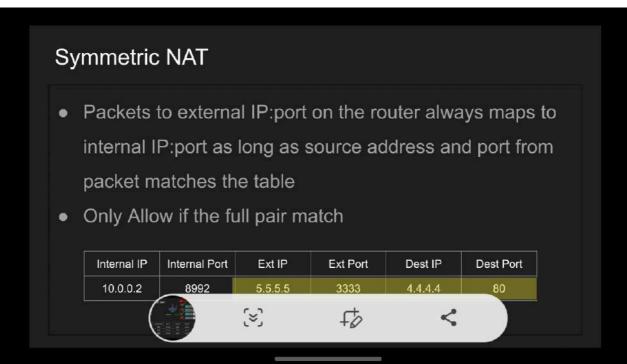
Internal IP	Internal Port	Ext IP	Ext Port	Dest IP	Dest Port
10.0.0.2	8992	5.5.5.5	3333	4.4.4.4	80

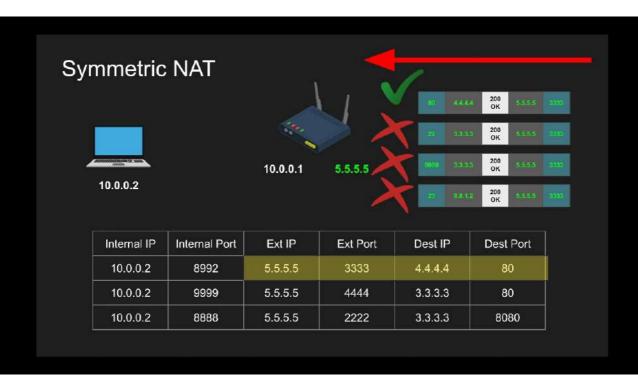






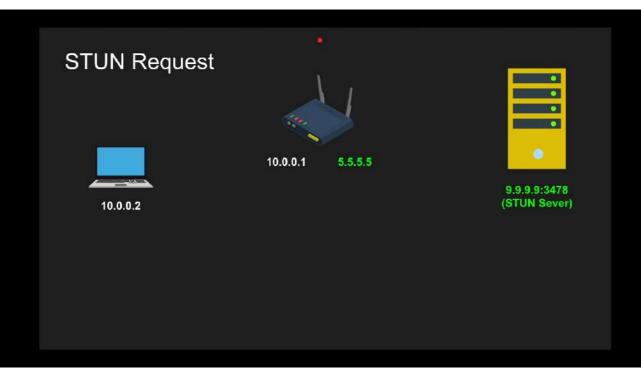


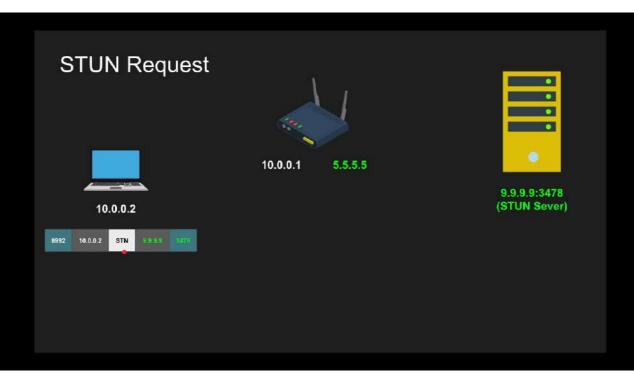


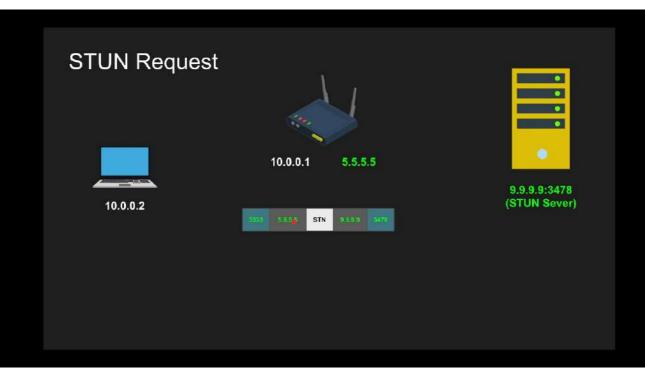


## STUN

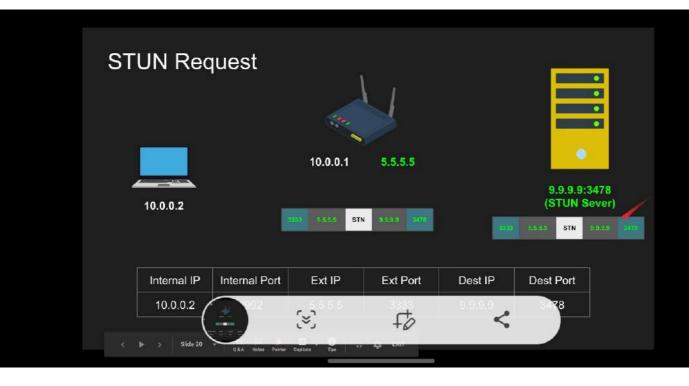
- Session Traversal Utilities for NAT
- Tell me my public ip address/port through NAT
- Works for Full-cone, Port/Address restricted NAT
- Doesn't work for symmetric NAT
- STUN server port 3478, 5349 for TLS
- Cheap to maintain













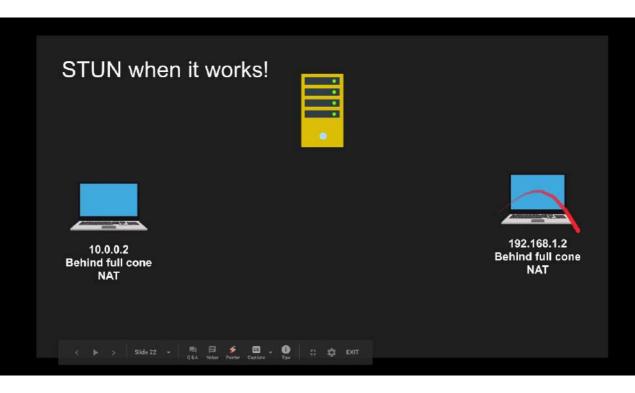


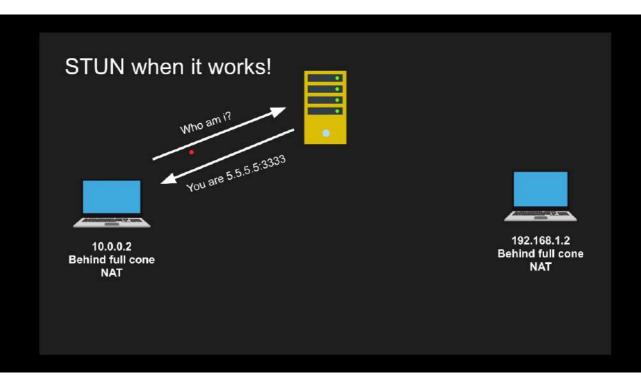


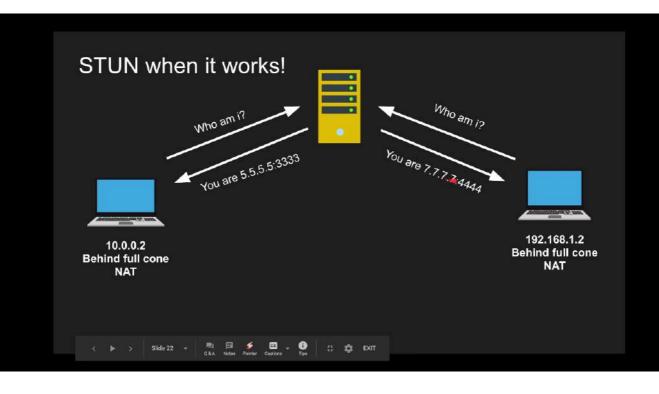


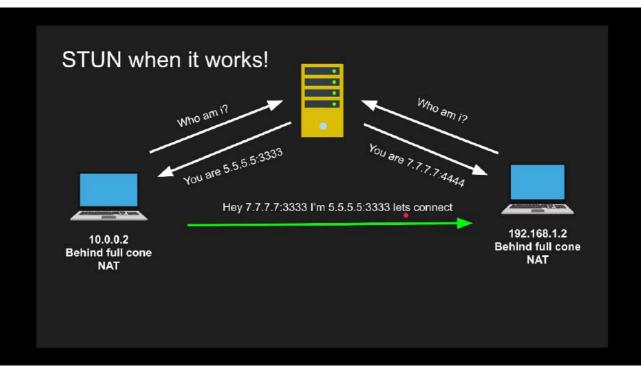


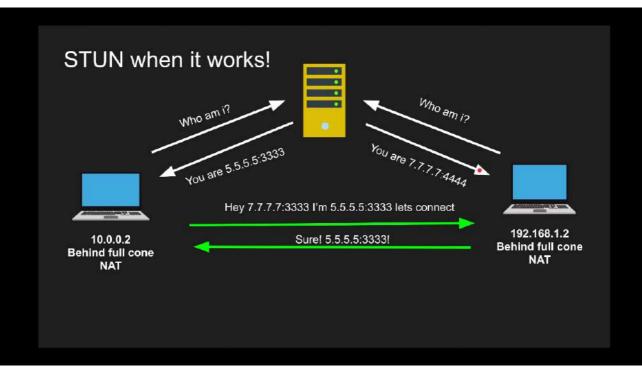


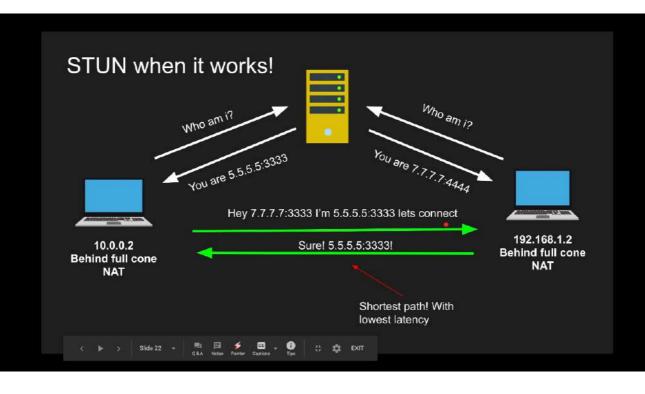




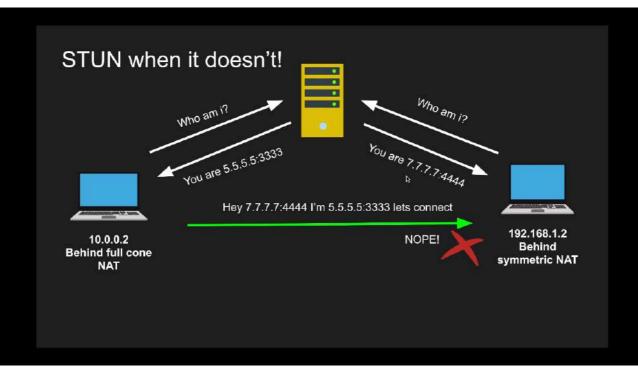






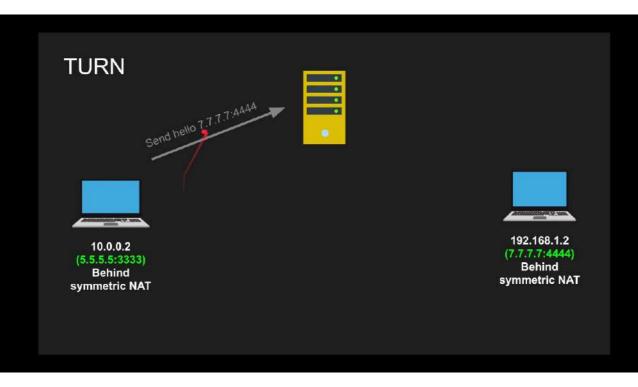


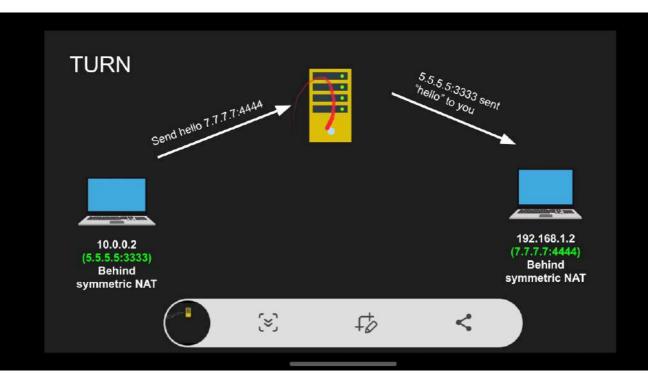


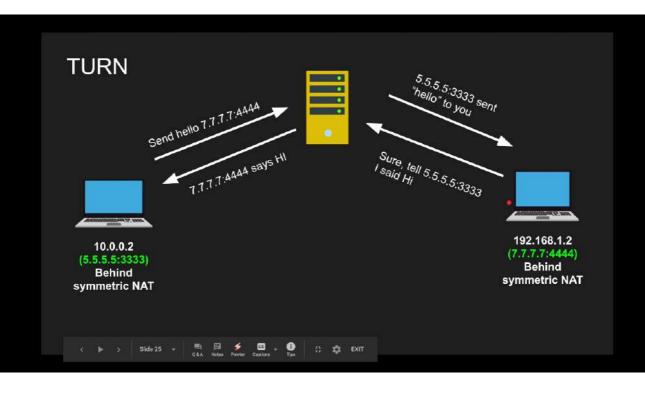


### **TURN**

- Traversal Using Relays around NAT
- In case of Symmetric NAT we use TURN
- It's just a server that relays packets
- TURN default server port 3478, 5349 for TLS
- Expensive to maintain and run







## **ICE**

- Interactive Connectivity Establishment
- ICE collects all available candidates (local IP addresses, reflexive addresses – STUN ones and relayed addresses – TURN ones)
- Called ice candidates
- All the collected addresses are then sent to the remote peer via SDP

#### SDP

- Session Description Protocol
- A format that describes ice candidates, networking options, media options, security options and other stuff
- Not really a protocol its a format
- Most important concept in WebRTC
- The goal is to take the SDP generated by a user and send it "somehow" to the other party

# SDP Example

```
v=0
o=- 9148204791819634656 3 IN IP4 127.0.0.1
s=-
t=0 0
a=group:BUNDLE audio video data
a=msid-semantic: WMS kyaiqbOs7S2h3EoSHabQ3JIBqZ67cFqZmWFN
m=audio 50853 RTP/SAVPF 111 103 104 0 8 107 106 105 13 126
c=IN IP4 192.168.1.64
a=rtcp:50853 IN IP4 192.168.1.64
a=candidate:3460887983 1 udp 2113937151 192.168.1.64 50853 typ host generation 0 a=candidate:3460887983 2 udp 2113937151 192.168.1.64 50853 typ host generation 0
```

# Signaling

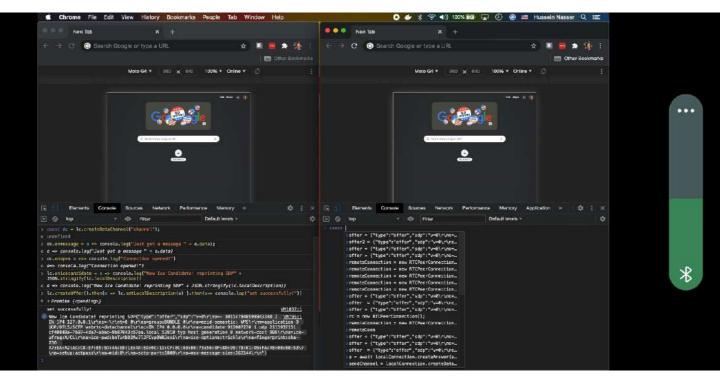
- SDP Signaling
- Send the SDP that we just generated somehow to the other party we wish to communicate with
- Signaling can be done via a tweet, QR code, Whatsapp,
   WebSockets, HTTP request DOESN'T MATTER! Just get
   that large string to the other party

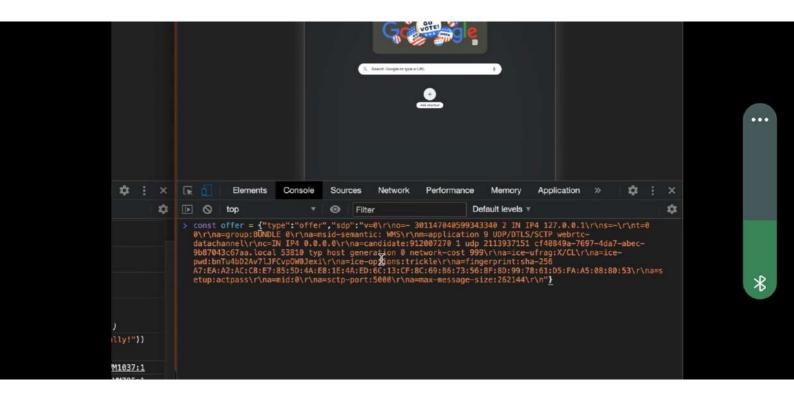
# WebRTC Demystified

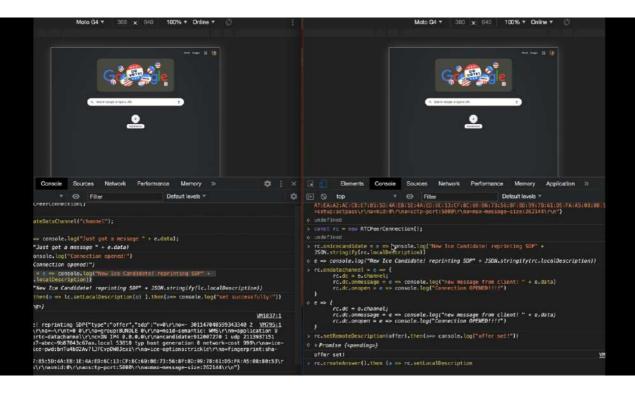
- 1. A wants to connect to B
- A creates an "offer", it finds all ICE candidates, security options, audio/video options and generates SDP, the offer is basically the SDP
- 3. A signals the offer somehow to B (whatsapp)
- 4. B creates the "answer" after setting A's offer
- 5. B signals the "answer" to A
- 6. Connection is created

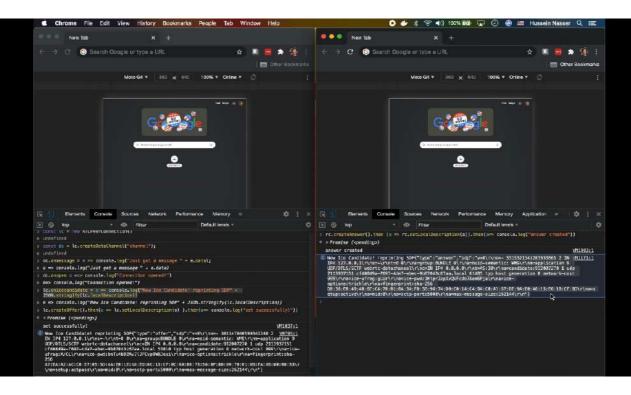
### WebRTC Demo

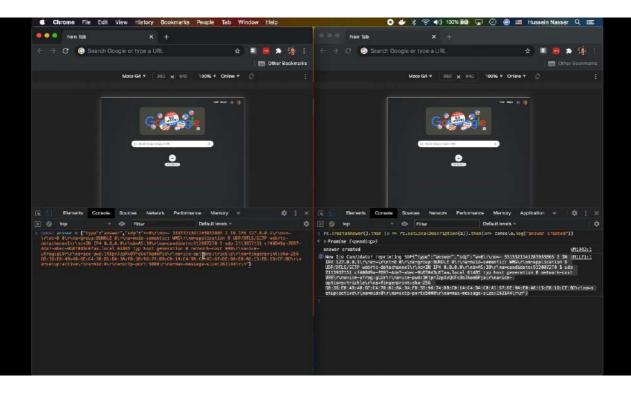
- We-will connect two browsers (Browser A & Browser B)
- A will create an offer (sdp) and set it as local description
- B will get the offer and set it as remote description
- B creates an answer sets it as its local description and signal the answer (sdp) to A
- A sets the answer as its remote description
- Connection established, exchange data channel

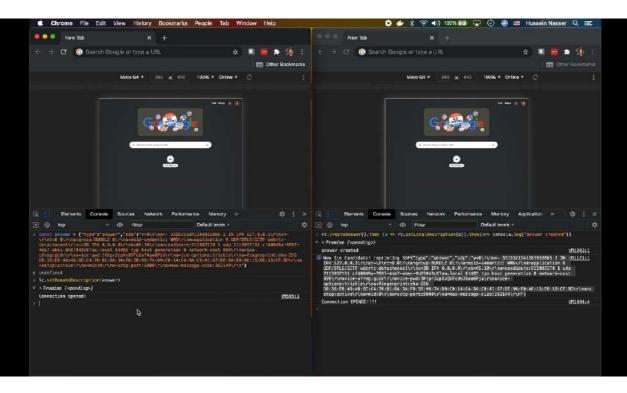


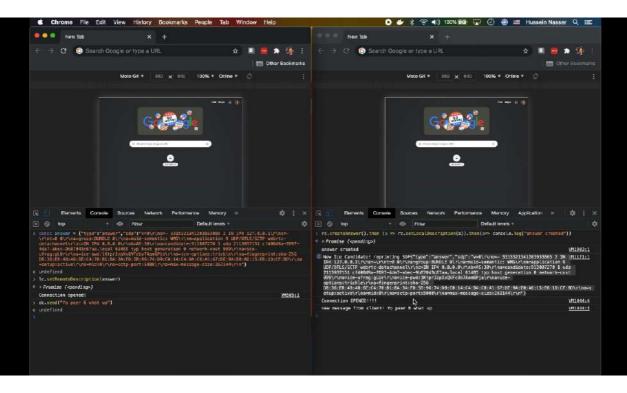


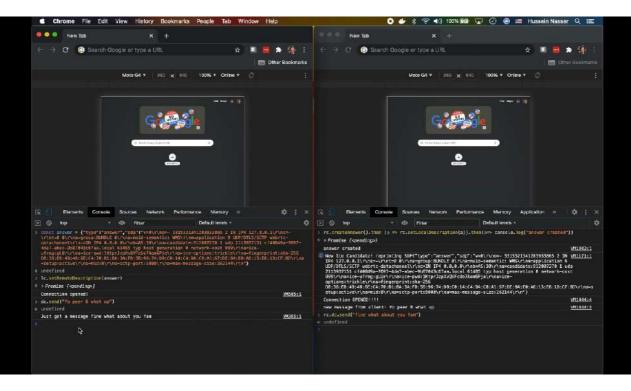












### WebRTC Pros & Cons

- Pros
  - o P2p is great! low latency for high bandwidth content
  - o Standardized API I don't have to build my own
- Cons
  - Maintaining STUN & TURN servers
  - Peer 2 Peer falls apart in case of multiple participants (discord case)

# More WebRTC stuff!

So more to discuss beyond this content

## Media API

- getUserMedia to access microphone, video camera
- RTCPConnection.addTrack(stream)
- https://www.html5rocks.com/en/tutorials/webrtc/basics/

#### onIceCandidate and addIceCandidate

- To maintain the connection as new candidates come and go
- onIceCandidate tells user there is a new candidate after the SDP has already been created
- The candidate is signaled and sent to the other party
- The other party uses addiceCandidate to add it to its
   SDP

### Set custom TURN and STUN Servers

# Create your own STUN & TURN server

- COTURN open source project
- https://github.com/coturn/coturn

## Public STUN servers

- stun1.1.google.com:19302
- stun2.1.google.com:19302
- stun3.1.google.com:19302
- stun4.1.google.com:19302
- stun.stunprotocol.org:3478

# Thank You

Subscribe for deep dive engineering discussions