# FAC 9 WebRTC Workshop

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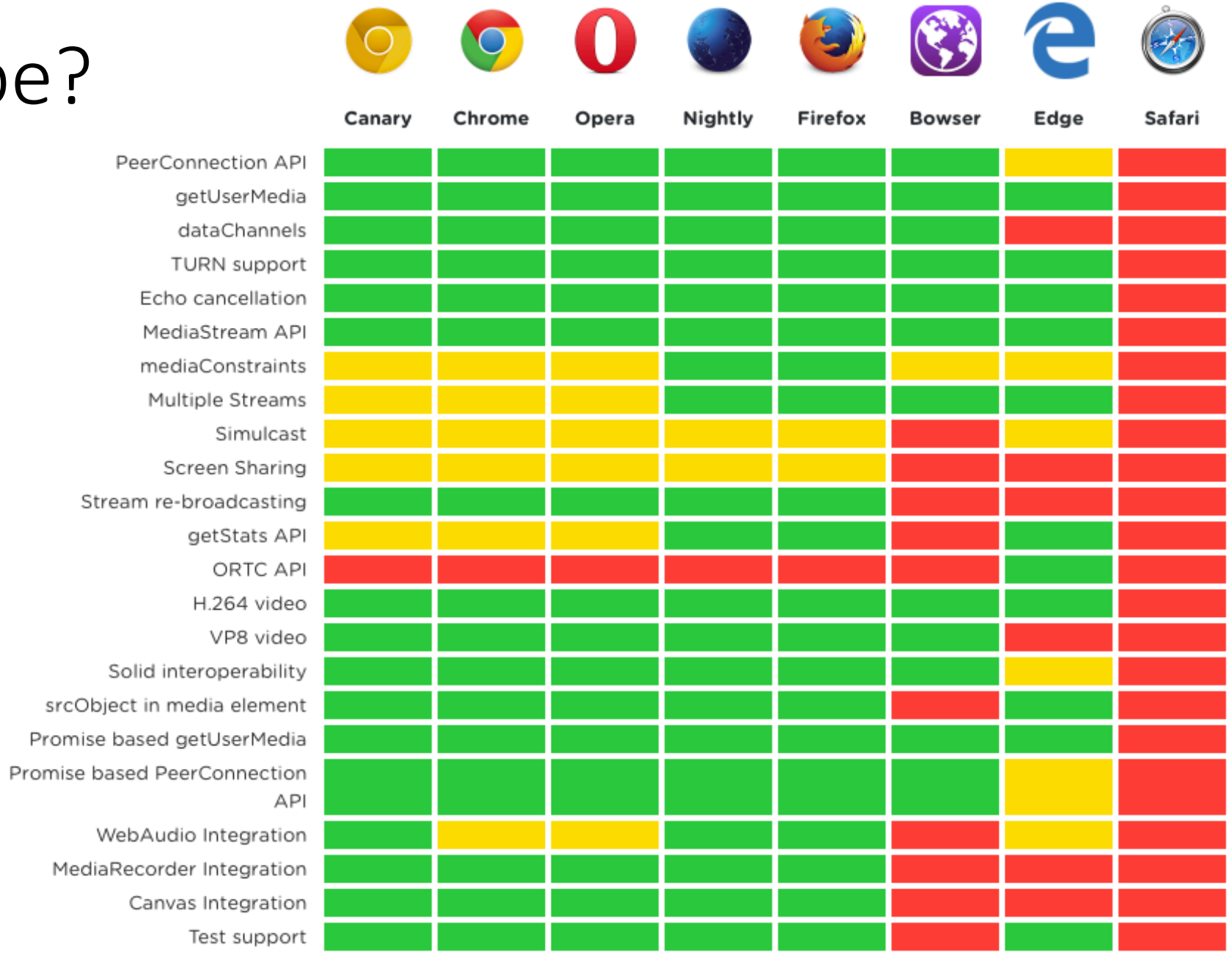
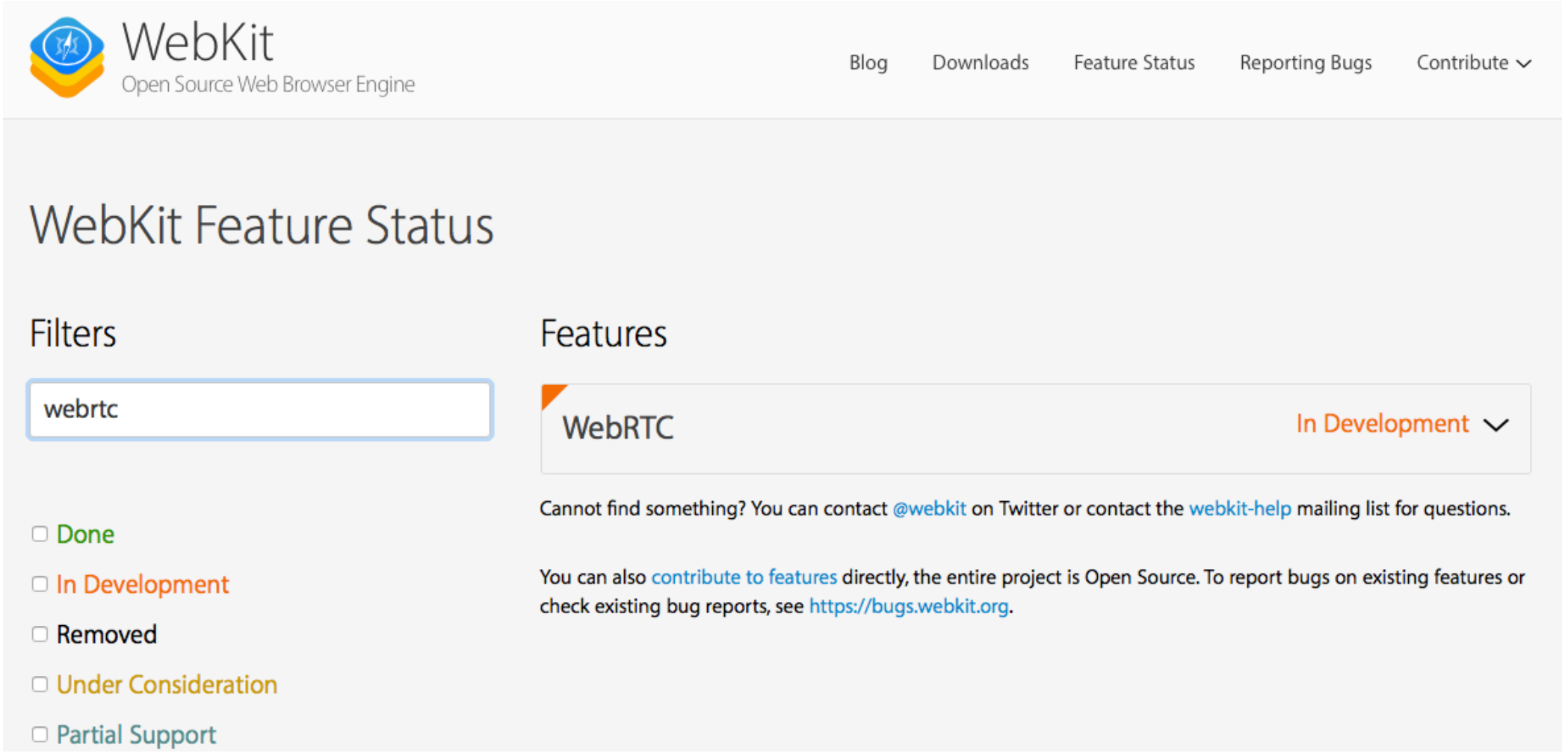
# Making the web ‘real-time’

- Audio calls, Video calls, Phone calls (via gateway)
- Using ‘standard’ technologies
  - Vanilla browser
  - No proprietary plugins (Flash)
  - No proprietary protocols (Skype)
- The standardised framework is WebRTC
  - Web Real Time Communication
- Most browser are adopting this now

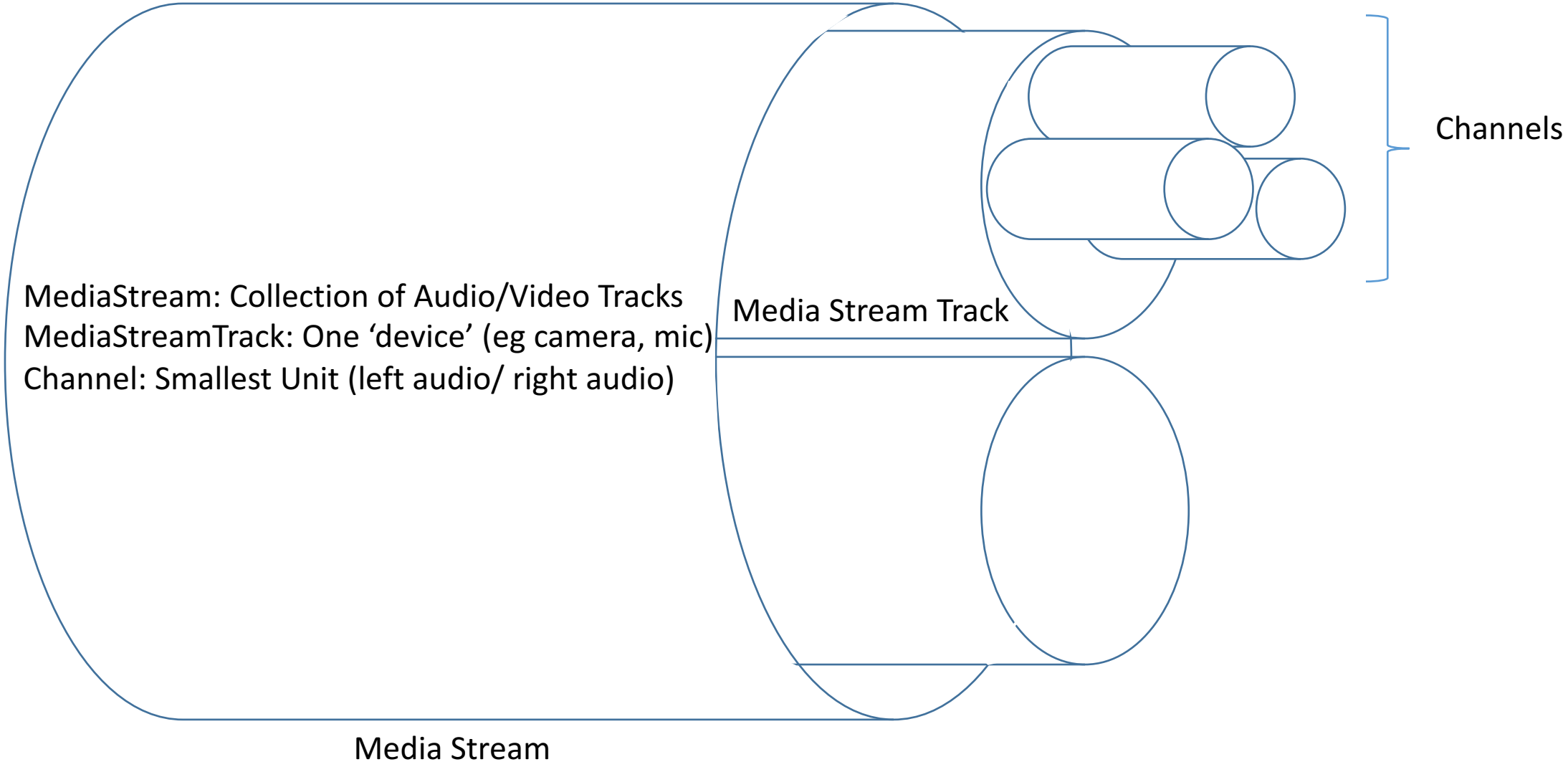
# Is it all hype?



# Apple Safari



# Media Streams

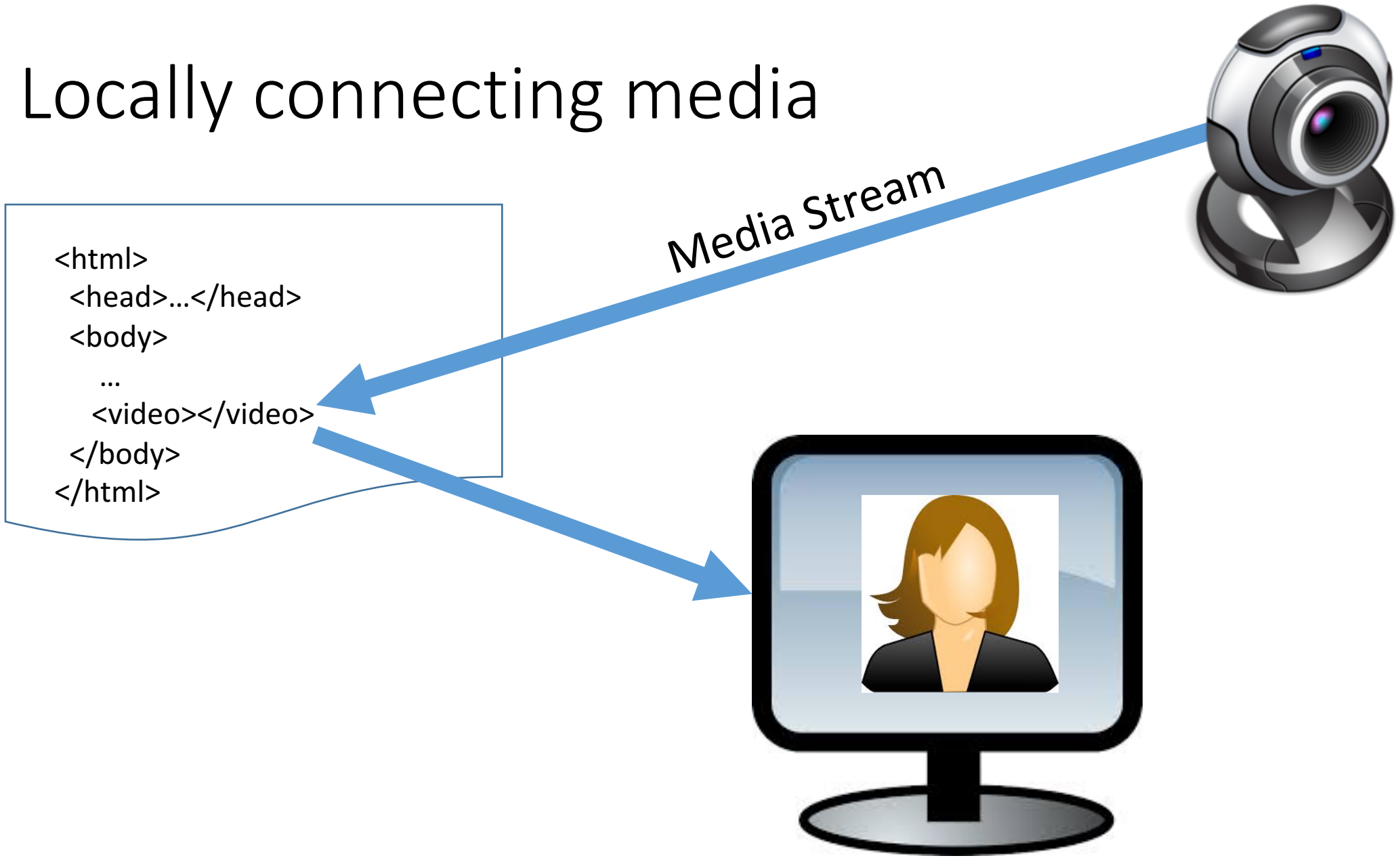



https://developer.mozilla.org/en-US/docs/Web/API/Media_Streams_API

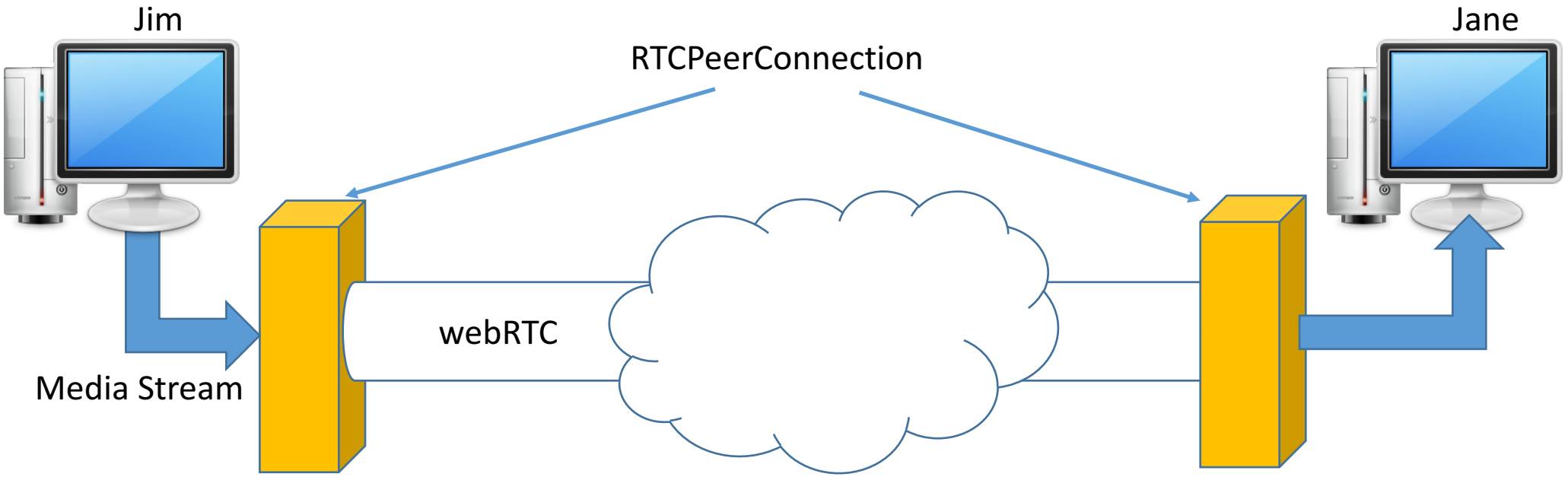
# Media Stream

- Unidirectional
- Has one input and out output
- Local inputs:
  - Microphone
  - Camera
  - *RTCPeerConnection*
- Outputs:
  - <video> tag
  - *RTCPeerConnection*

# Locally connecting media

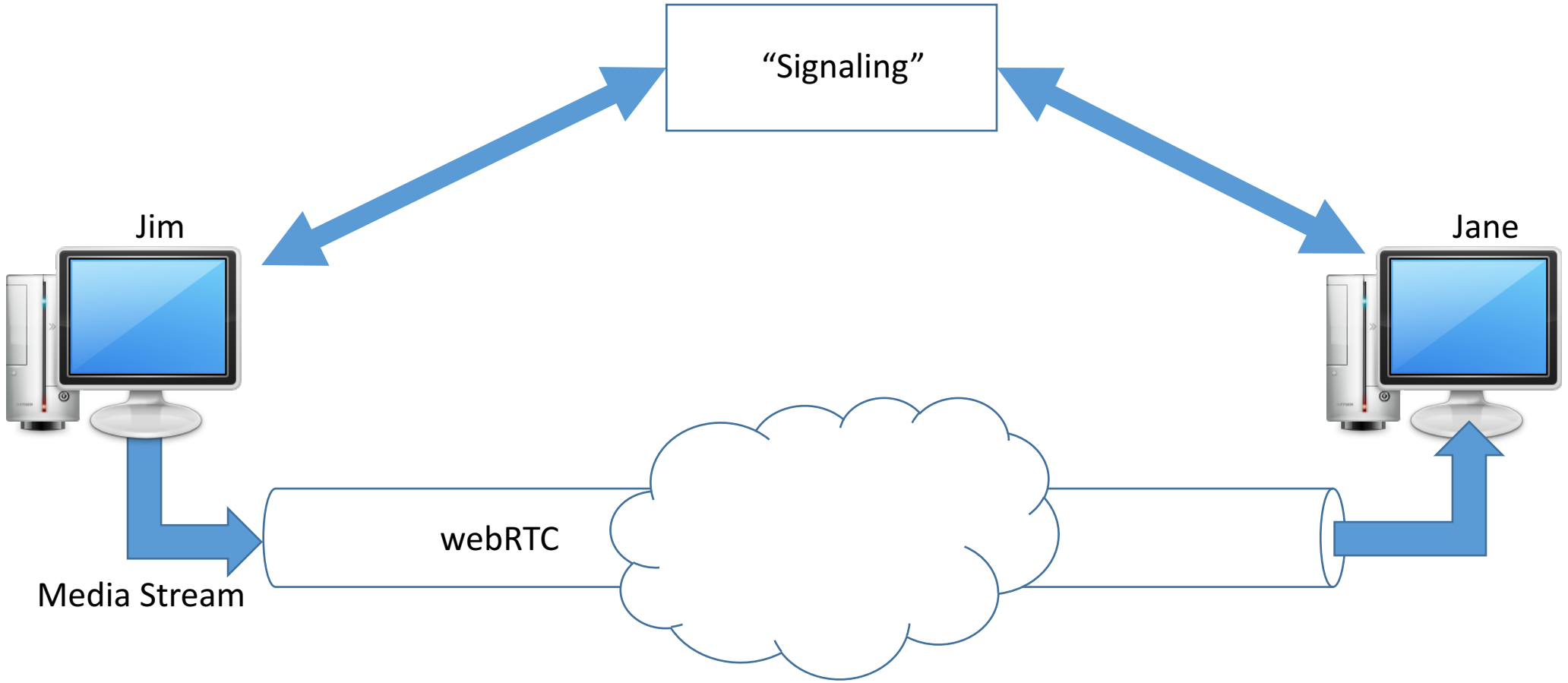


# Connecting between devices




- How do Jim and Jane find each other?
- How is their traffic routed across the Internet through corporate firewalls?
- How is permission asked to accept the call at the receiving end?

# Signaling



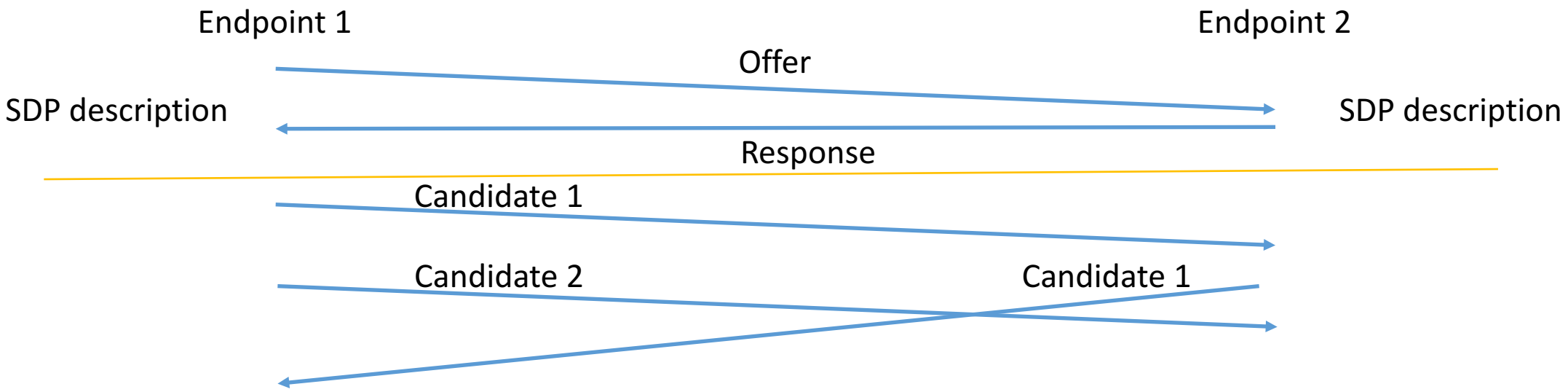
# Signaling

- Allows two ends of an RTCPeerConnections to find each other
- NOT standardised
  - Not necessary and much of this is application specific
- Examples
  - IpCortex PABX – signaling, directory, presence and PTSN gateway (phone calls)
  - webRTC.io – one of the first libraries
- Must be accessible to both parties
- Provide a means of relaying information between the two parties
- Should encrypt all communications
- *Usually* a separate server

# What is signaling used for?

- Very simple WebRTC requirements
  - Transfer Javascript Objects between endpionts (eg serialised as JSON)
  - Send media ‘offers’ and ‘responses’ between end points
  - Swap communication ‘candidates’ between end points
- Other signaling usually required for an application
  - End point discovery/name mapping (I’m Fred, I want to talk to Jane)
  - Presence (I’m Fred and I’m ‘online’)

# Typical signaling exchange




- Offers are send as Session Description Protocol messages
- Candidates give options for how to connect end points across a network
  - Accomplished via the  Interactive Connectivity Establishment (ICE) framework
  - Multiple ‘candidates’ are tested concurrently with the first (fastest) used

# Signaling Overhead

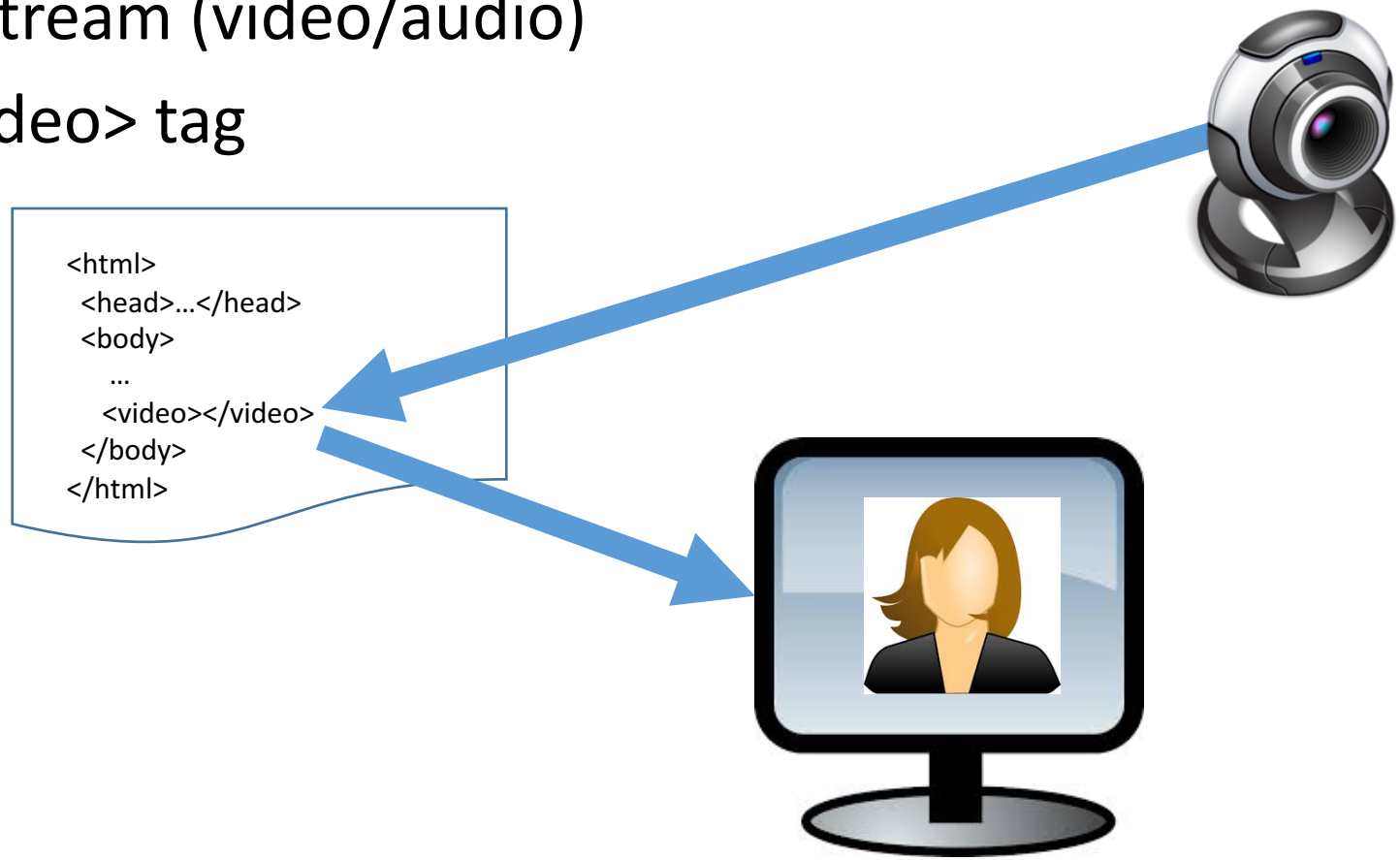
- Intentionally very low overhead
- Typically 24 exchanges per WebRTC session
- ~10K data exchanged
- Many techniques available
  - REST Polling
  - HTTP ‘Long Poll’
  - REST to the signaling server/ EventSource distribution to clients
  - WebSocket bi-directional ‘pipes’
- Only requires text transfer
- Not just for setup though – media can change during a call

# Workshop – what we're going to do…

1. Configure simple HTTPS server to serve scripts
2. Using streams locally
   Create a <video> tag in a static page
   Request media (camera and microphone)
   Attach media to video tag
3. Local Peer Connection and Signaling
4. Using streams between devices over a network
5. *Building a simple remote 'Presentation' application using webRTC*

# 2. Local stream

- Request local media stream (video/audio)
- Attach to browser <video> tag



# 2. Local media streams

```
var promise = navigator.mediaDevices.getUserMedia({
    video: true,
    audio: true
});


promise.then((avSteam) => {
  // Find my video tag…
  video = document.createElement('video');

  attachMediaStream(video, stream);
  video.play();

  // Add video tag to DOM
  videoContainer.append(v);
}).catch(() => {…});
```

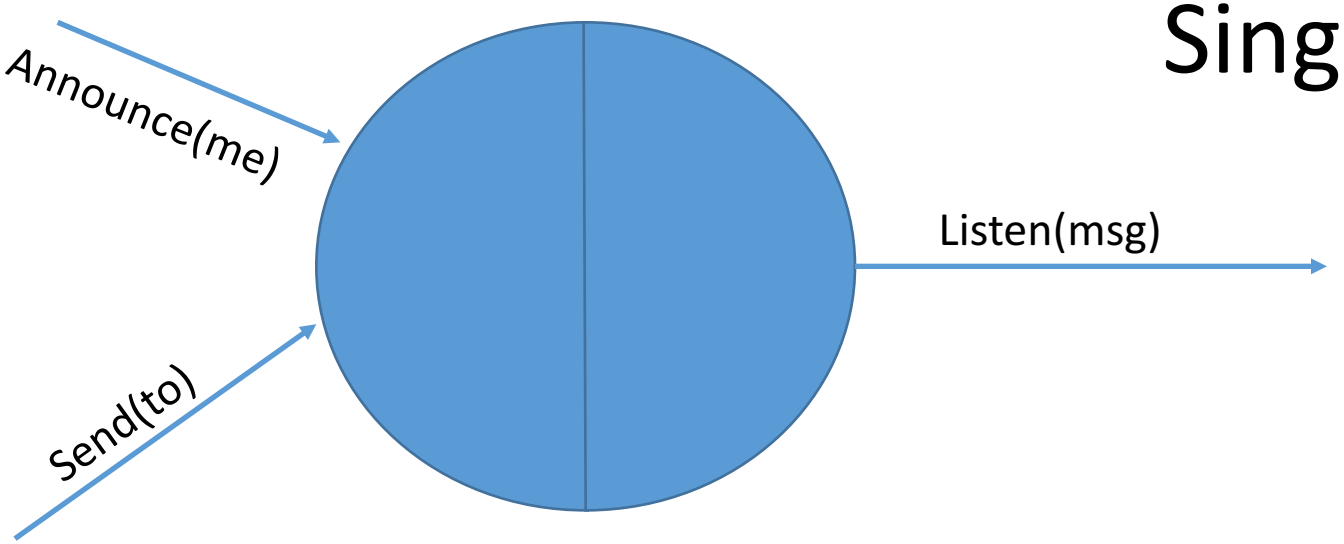
# 3. Local Peer Connection

```
<html>
  <head>…</head>
  <body>
    …
    <h2>Person 1</h2>
    <video id=”person_1”></video>
    <button id=”Start Call”>Call</button>

    <h2>Person 2</h2>
    <video id=”person_2”></video>

</body>
</html>
```

- Connecting camera/mic to a local video tag THROUGH a peer connector
- Implement our own local signaling
- Shows the basic structure of how to connect streams to each other remotely without network complexity

# 3. Local signaling

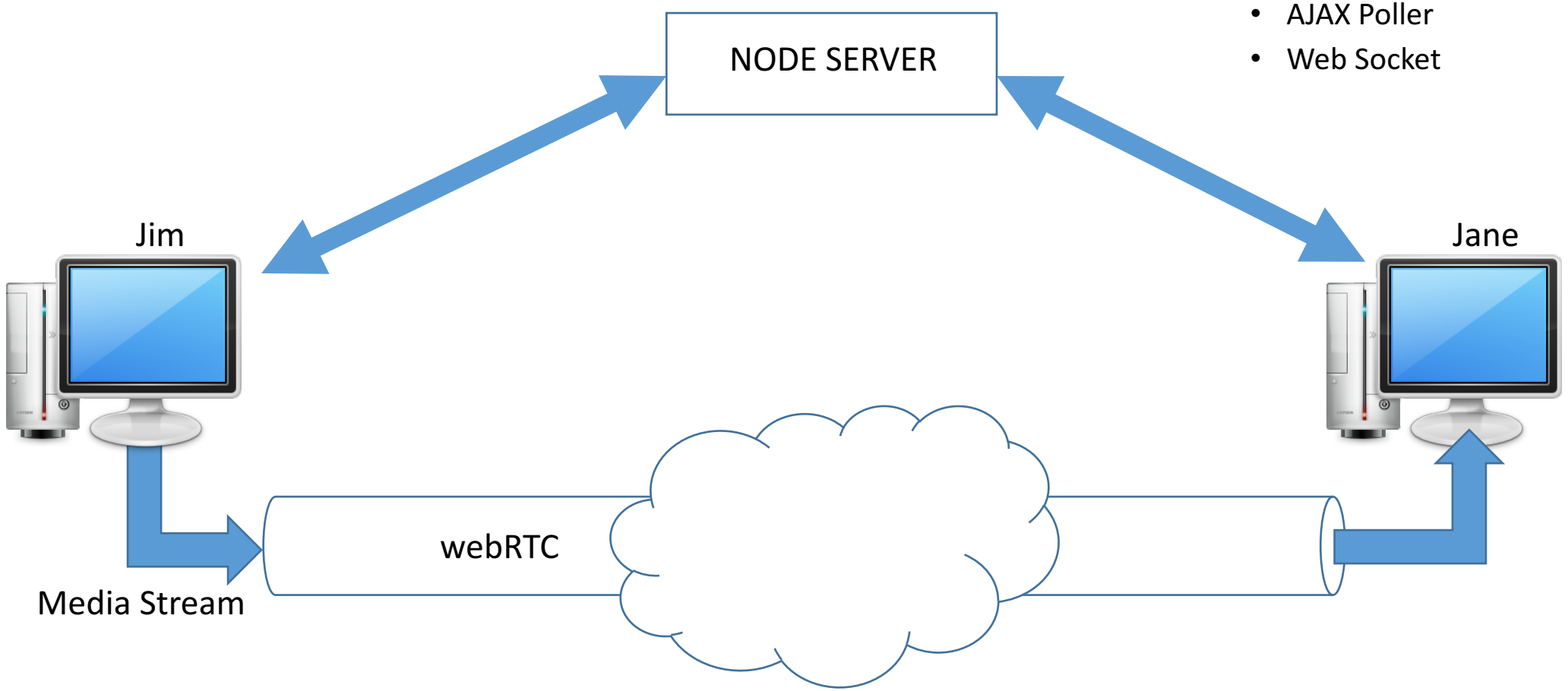
- Create a ‘signaling’ abstraction:
  - Announce(me), send(to), listen
  - Completely local




Single web page!

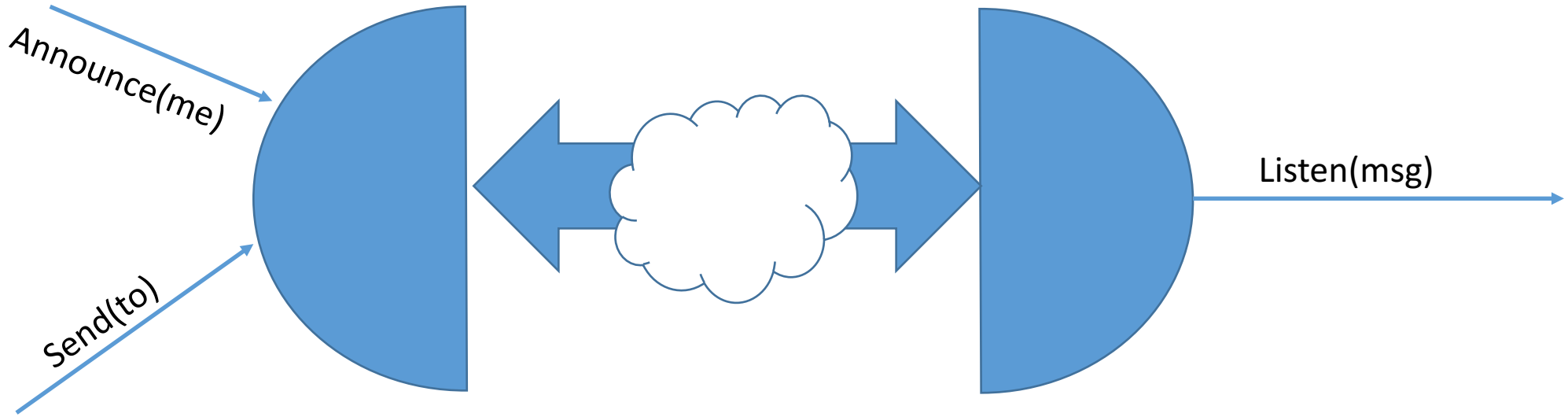
# 4. Remote Peer Connections

- Replace local signaling with node server proxy.
- Options for transferring signalling:
  - AJAX Poller
  - Web Socket

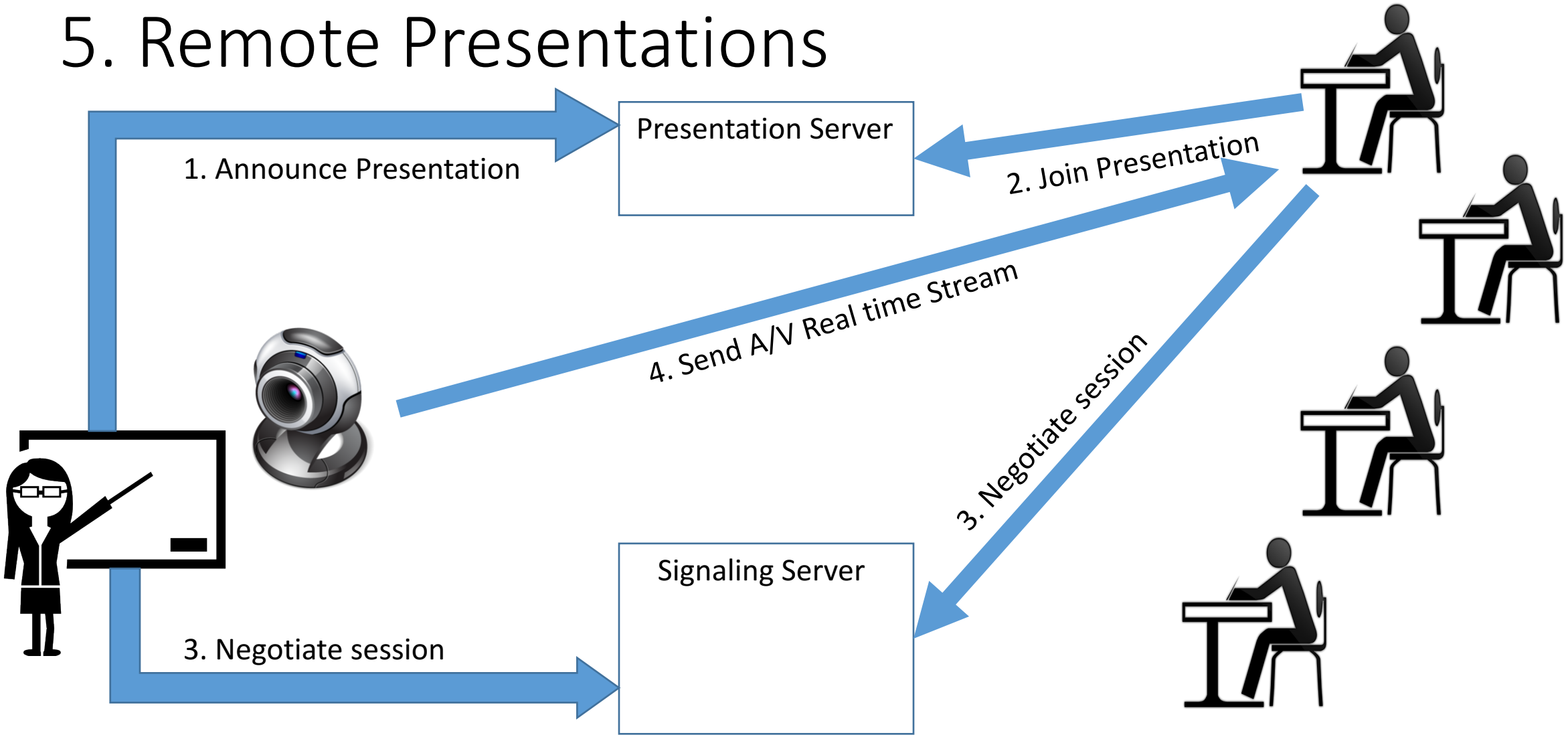


# 4. Signaling across a network

- Split your signaling into two parts:
  - Carry information across the local network
  - Modify the application to have one end point per browser



# 5. Remote Presentations




Signaling: EITHER using IpCortex api OR modified signaling from previous task

# Signaling for Remote Presentations: Two Options

- Evolution of simple signaling from previous example
  - Should work on a local LAN
  - Won’t work across the Internet without TURN/STUN servers (complexity)
- IPCortex API
  - Covers all the routing across the Internet
  - More complex to configure/run

# References

- MDN WebRTC
  - https://developer.mozilla.org/en-US/docs/Web/API/WebRTC_API
- WebRTC.org – Getting Started
  - https://webrtc.org/start/
- HTML 5 Rocks – Getting started with WebRTC (2012)
  - Good overview
  - Illustrates local signaling *but* not is a portable way
  - https://www.html5rocks.com/en/tutorials/webrtc/basics/
- HTML 5 Rocks – WebRTC Infrastructure
  - Great overview of signalling – everything you need to know!
  - https://www.html5rocks.com/en/tutorials/webrtc/infrastructure/
- adapter.js
  - Shim to isolate applications from browser incompatibilities
  - https://github.com/webrtc/adapter