Secure Application Development

DistriNet iMinds KU LEUVEN

# Real-time communications with WebRTC

Lieven Desmet – iMinds-DistriNet, KU Leuven
Lieven.Desmet@cs.kuleuven.be

# About me: Lieven Desmet

- Research manager at University of Leuven
  - (Web) Application Security


- Active participation in OWASP
  - Board member of the OWASP Belgium Chapter
  - Co-organizer of the OWASP AppSec EU 2015 Conference


- Program director at SecAppDev
  - Yearly week-long training on Secure Application Development

# Overview

- WebRTC in a nutshell
- Communication protocols
- WebRTC JavaScript APIs
- Security & Privacy in WebRTC
- Wrap-up

# WebRTC IN A NUTSHELL

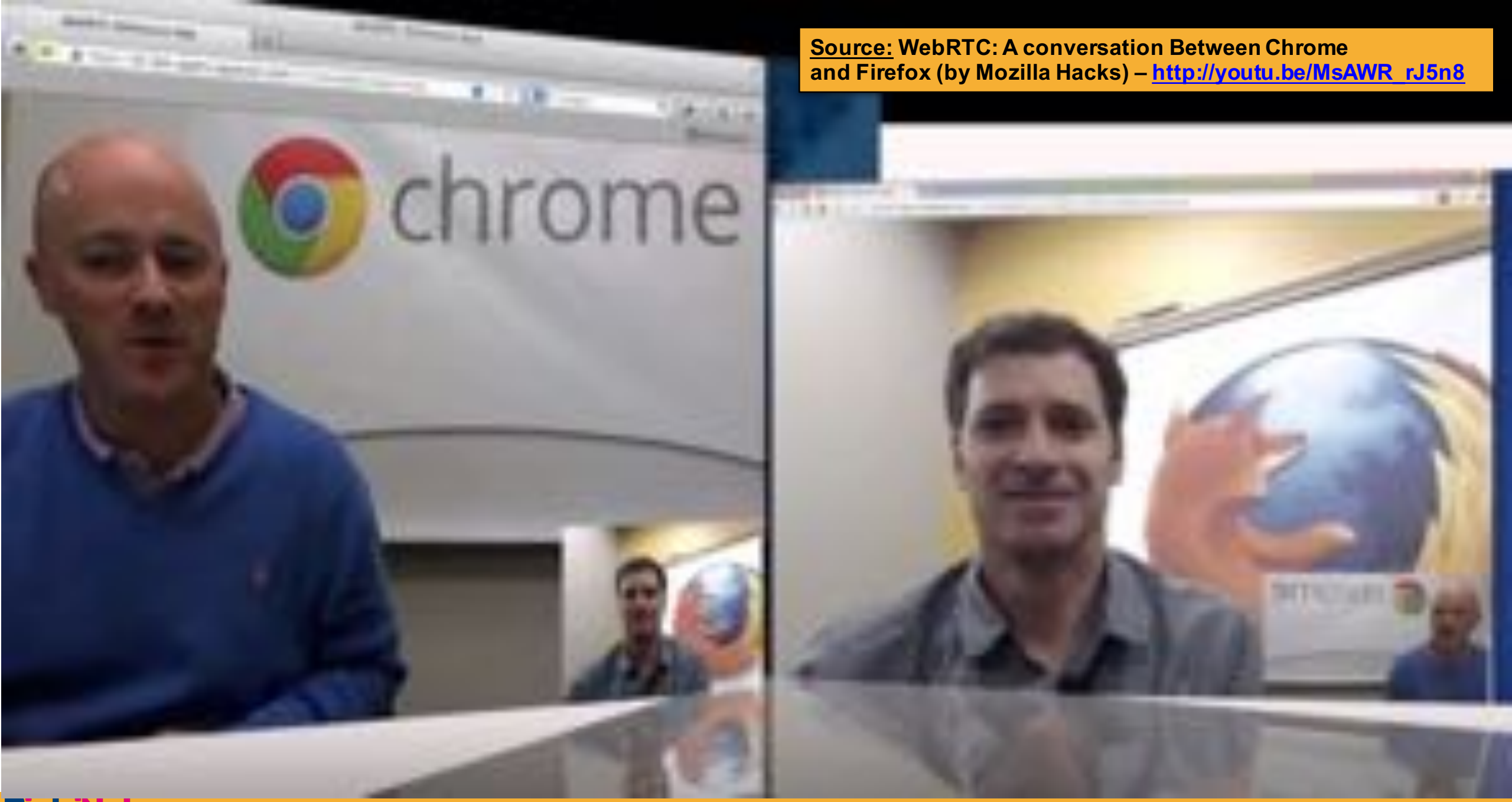
# Audio/video communication … in the browser

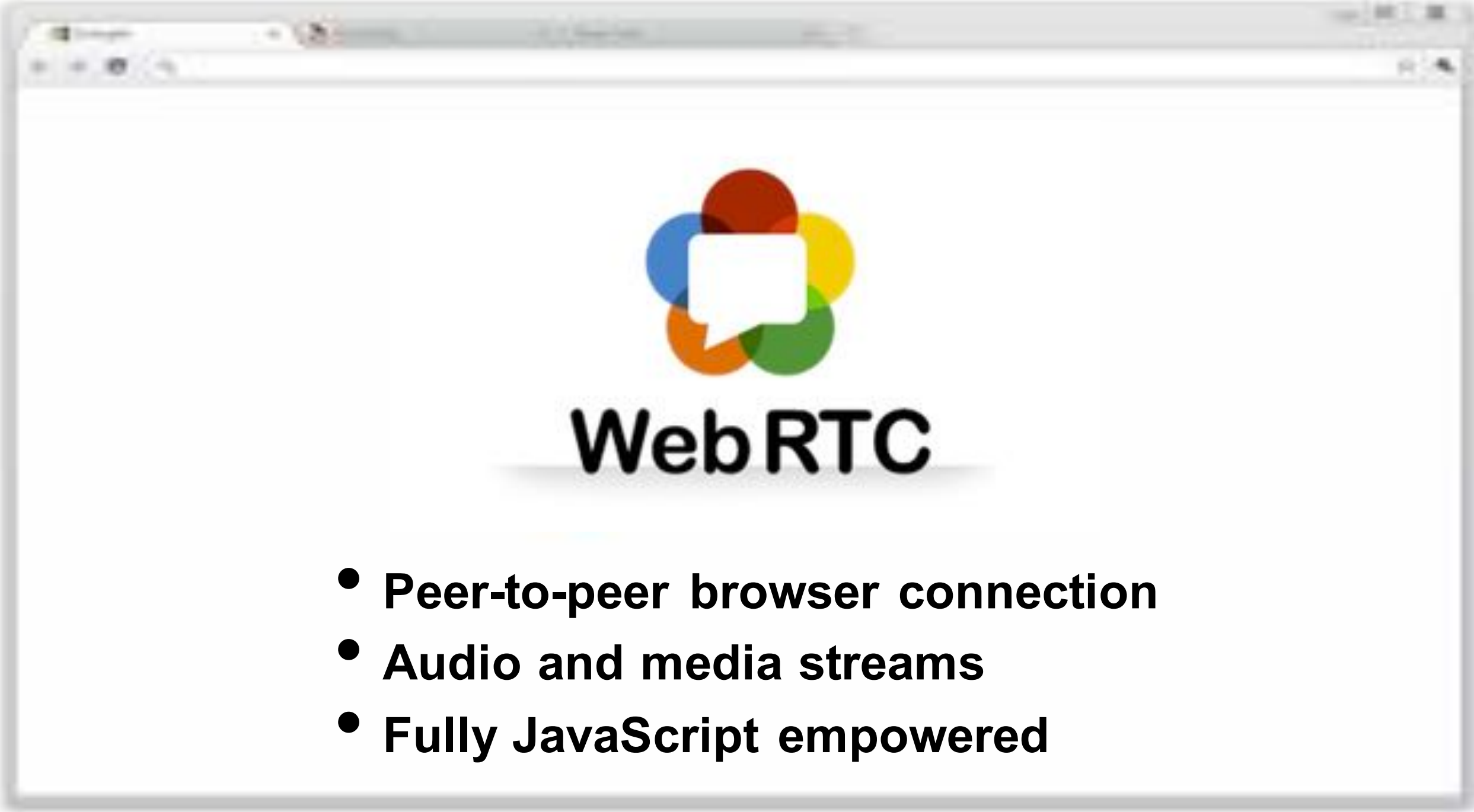





WhatsApp

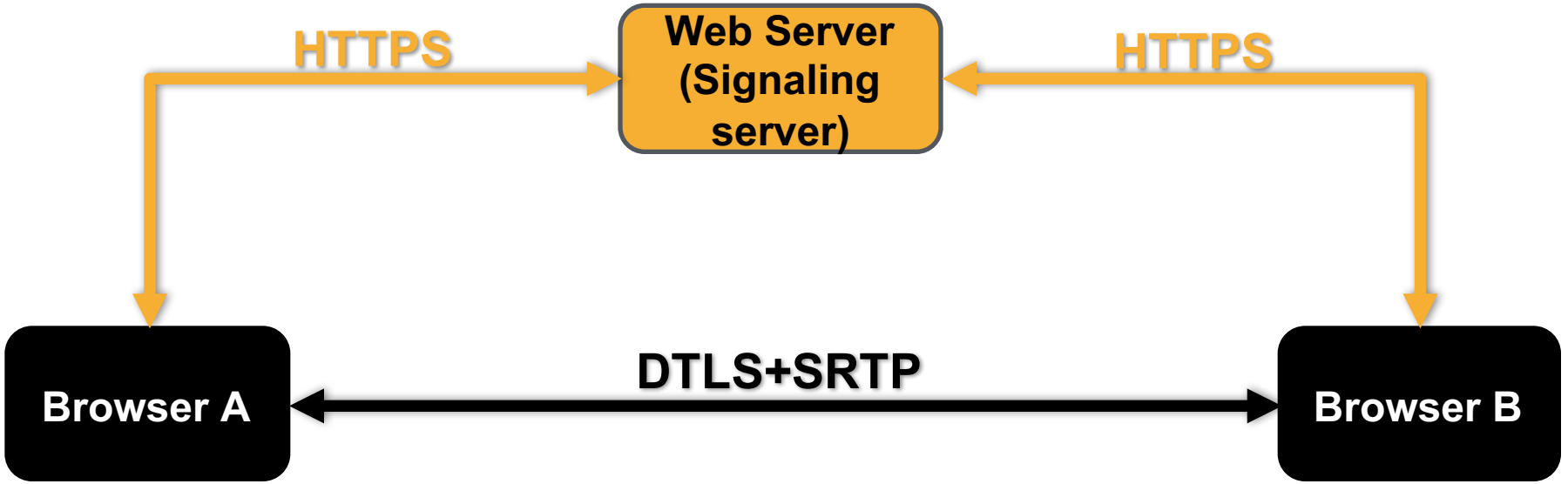

**Source:** WebRTC: A conversation Between Chrome and Firefox (by Mozilla Hacks) – http://youtu.be/MsAWR_rJ5n8



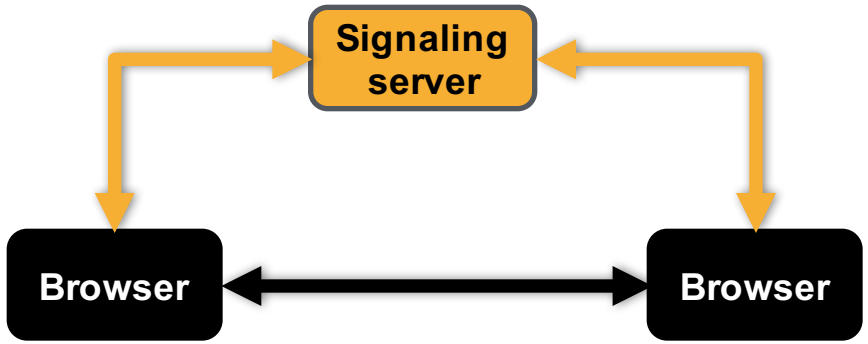



- Peer-to-peer browser connection
- Audio and media streams
- Fully JavaScript empowered

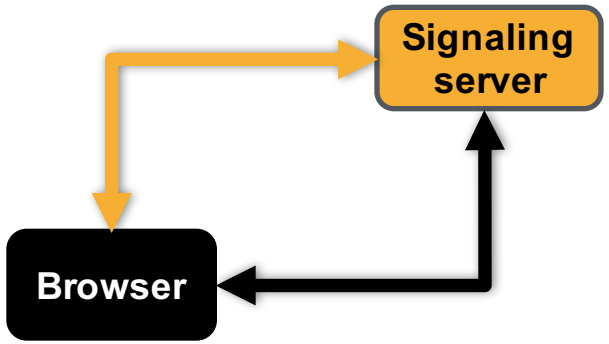
# WebRTC architecture (simplified)



# Various WebRTC deployments

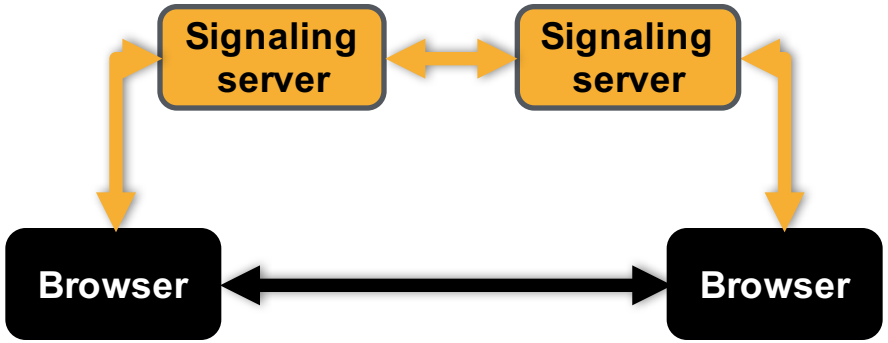



2-party video conferencing

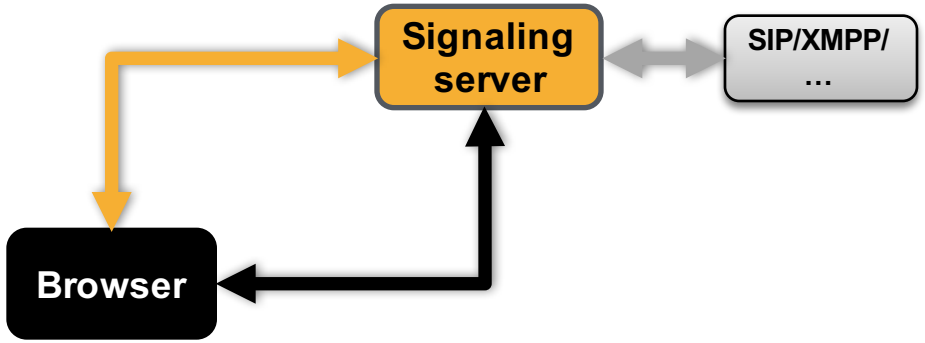



Helpdesk call

Federated signaling setup

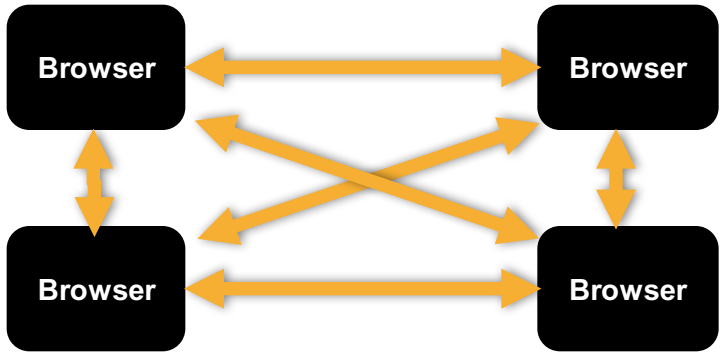



Bridged to SIP/Jingle/… infrastructure



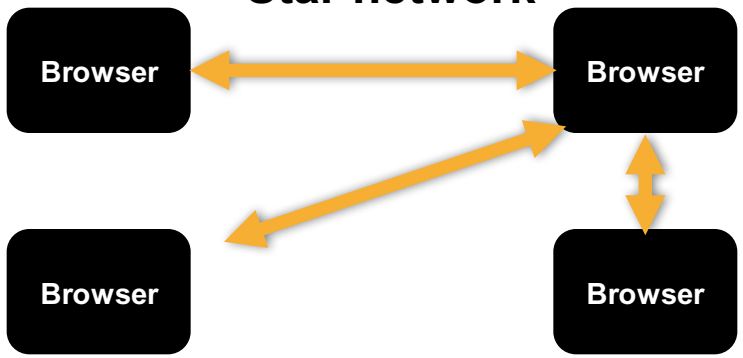
# Multiple peer topologies


Peer to Peer connection

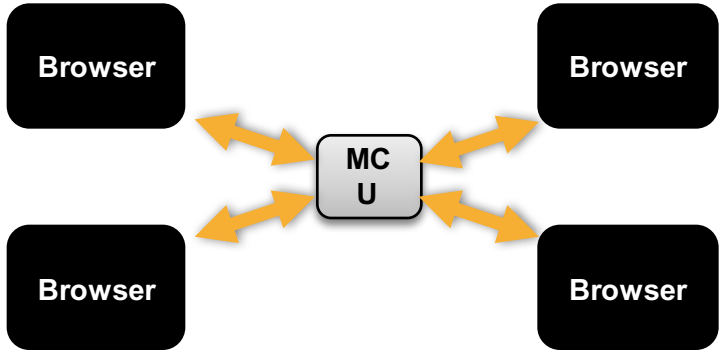



Mesh network

Star network




Multipoint Control Unit (MCU)



# COMMUNICATION PROTOCOLS

# Signaling path

- Signaling path between WebRTC end-points
- Signaling server(s)
  - Loads client-side context (JavaScript code)
  - Mediates control messages and meta-data between end-points
- Signaling protocol is undefined in WebRTC
  - Up to the application to deploy one !

# Media path

- Secure peer-to-peer connection between browsers
  - Media streams (video/audio)
  - Data channels

- DTLS: Datagram Transport Layer Security
- SRTP: Secure Real-time Transport Protocol
  - Encryption, message authentication and integrity

# Setting up the media path

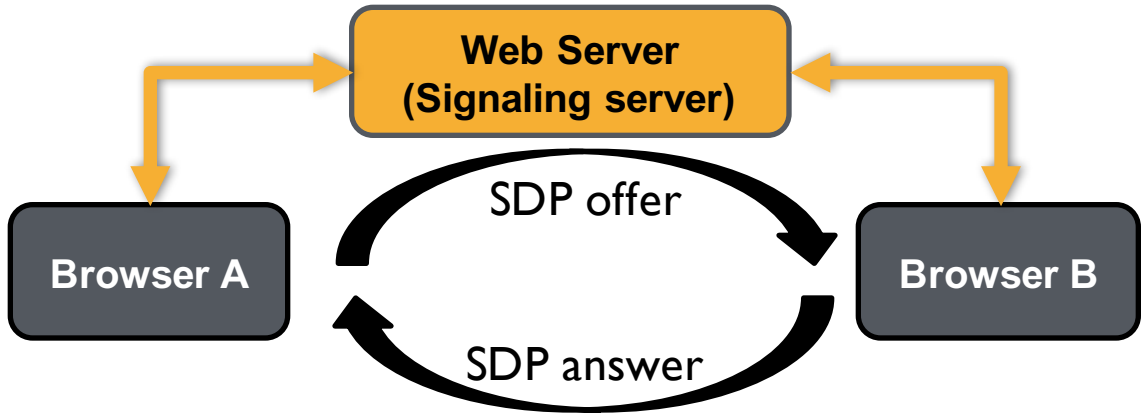
1. Exchange of media parameters
   - SDP: Session description protocol

2. Exchange of network parameters
   - UDP hole punching
   - STUN: Session description protocol
   - TURN: Traversal Using Relays around NAT
   - ICE: Interactive Connectivity Establishment

Technologies borrowed from SIP

# SDP: Session description protocol

- Initialization parameters for streaming media
  - Session announcement
  - Session invitation
  - Parameter negotiation (multimedia types, codecs, …)
- SDP offer and SDP answer



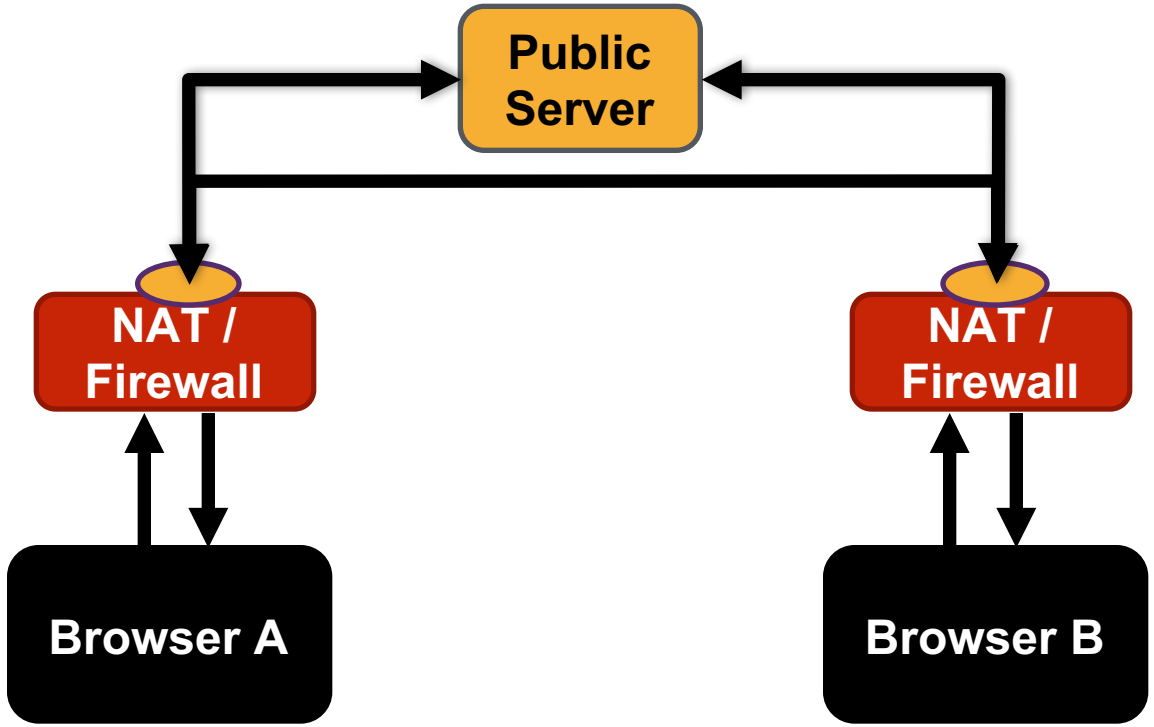
# SDP example

```
v=0
o=- 20518 0 IN IP4 0.0.0.0
t=0 0
a=msid-semantic:WMS ma
a=group:BUNDLE audio
m=audio 54609 UDP/TLS/RTP/SAVPF 109 0 8
c=IN IP4 24.23.204.141
a=mid:audio
a=msid:ma ta
a=rtcp-mux
a=rtcp:54609 IN IP4 24.23.204.141
a=rtpmap:109 opus/48000/2
a=ptime:60
a=rtpmap:0 PCMU/8000
a=extmap:1 urn:ietf:params:rtp-hdrext:ssrc-audio-level
a=sendrecv
a=setup:actpass
a=fingerprint:sha-1 99:41:49:83:4a:97:0e:1f:ef:6d:f7:c9:c7:70:9d: 1f:66:79:a8:07
a=ice-ufrag:074c6550
a=ice-pwd:a28a397a4c3f31747d1ee3474af08a068
a=candidate:0 1 UDP  2122194687 192.168.1.4 54609 typ host
a=candidate:0 2 UDP 2122194687 192.168.1.4 54609 typ host
a=candidate:1 1 UDP  1685987071 24.23.204.141 64678 typ srflx raddr 192.168.1.4 rport 54609
a=candidate:1 2 UDP  1685987071 24.23.204.141 64678 typ srflx raddr 192.168.1.4 rport 54609
a=rtcp-fb:109 nack
a=ssrc:12345 cname:EocUG1f0fcg/yvY7
a=rtcp-rsize
a=ice-options:trickle
```

Source: SDP Offer taken from "SDP for the WebRTC" (IETF Internet Draft)

# UDP hole punching

- Enables connectivity between peers across NAT(s)

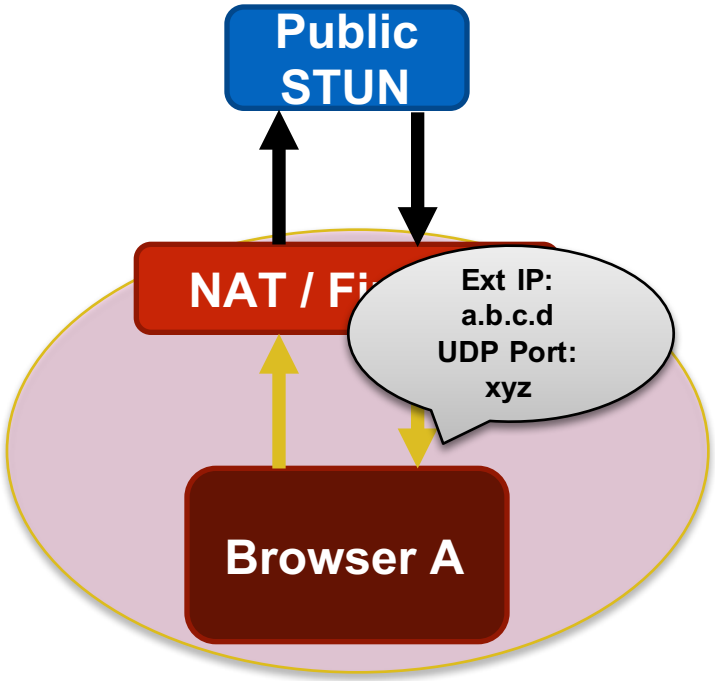


# When to use STUN/TURN/ICE?

- STUN
  - To collect your local network setup (local IPs, local subnets, NAT configuration, …)
- TURN
  - To relay your media connection if peer-to-peer fails
- ICE
  - Bundles all STUN/TURN information for exchange via the signaling channel

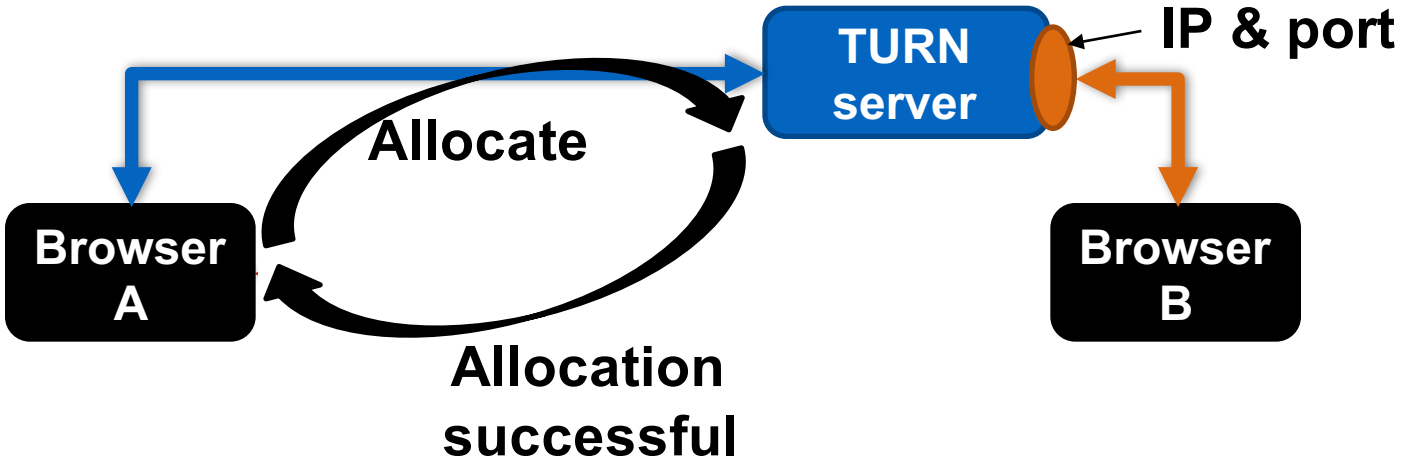
# STUN: Session Traversal Utilities for NAT

- Discover your public IP address
- Determine whether your browser sits behind a NAT
- Retrieve the UDP port that NAT has allocated for external communication



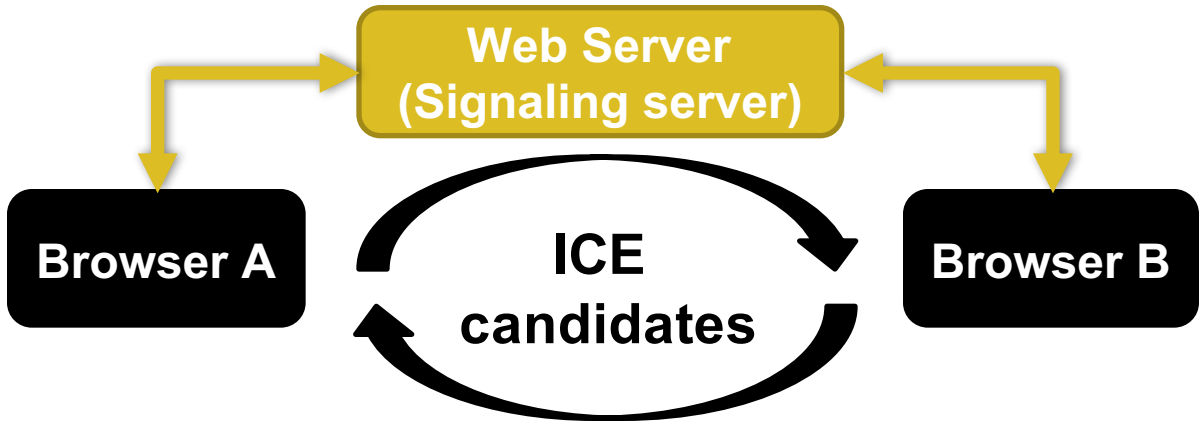
# TURN: Traversal Using Relays around NAT

- Used if STUN does not work
- TURN server relays traffic between 2 NAT'ed peers
- IP and port get allocated on STUN for sending or receiving a stream

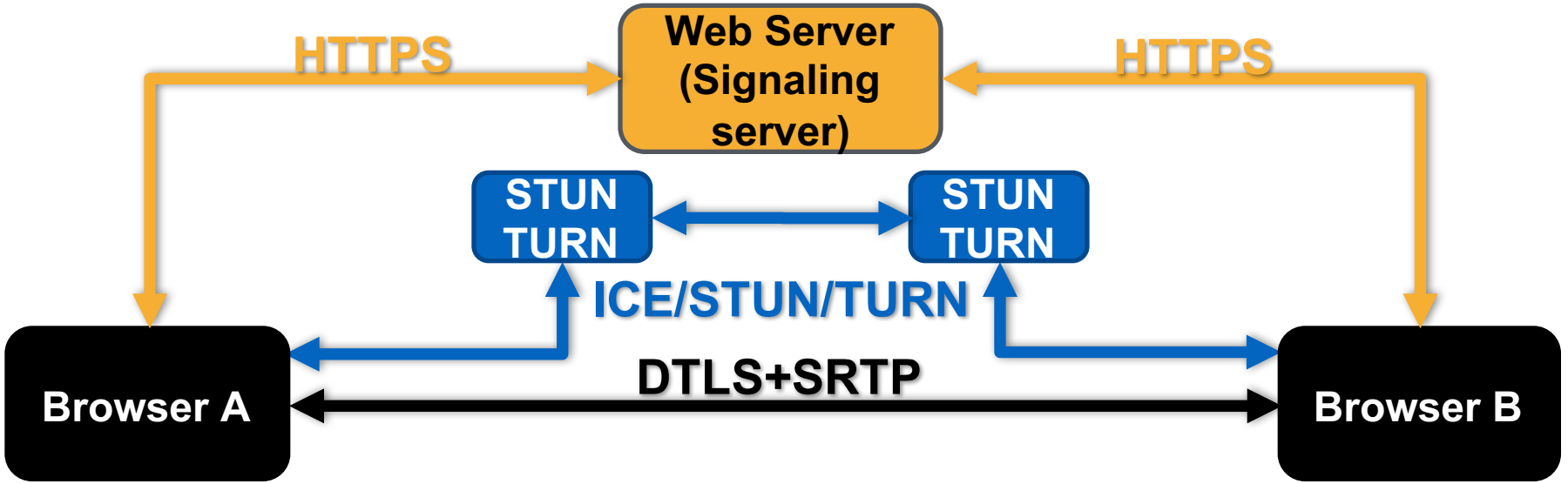


# ICE: Interactive Connectivity Establishment

- Gathering info (STUN, TURN, …)
- Offering and answering ICE candidates between peers
- Probe candidates in order of priority
  - Until ICE candidate pair works

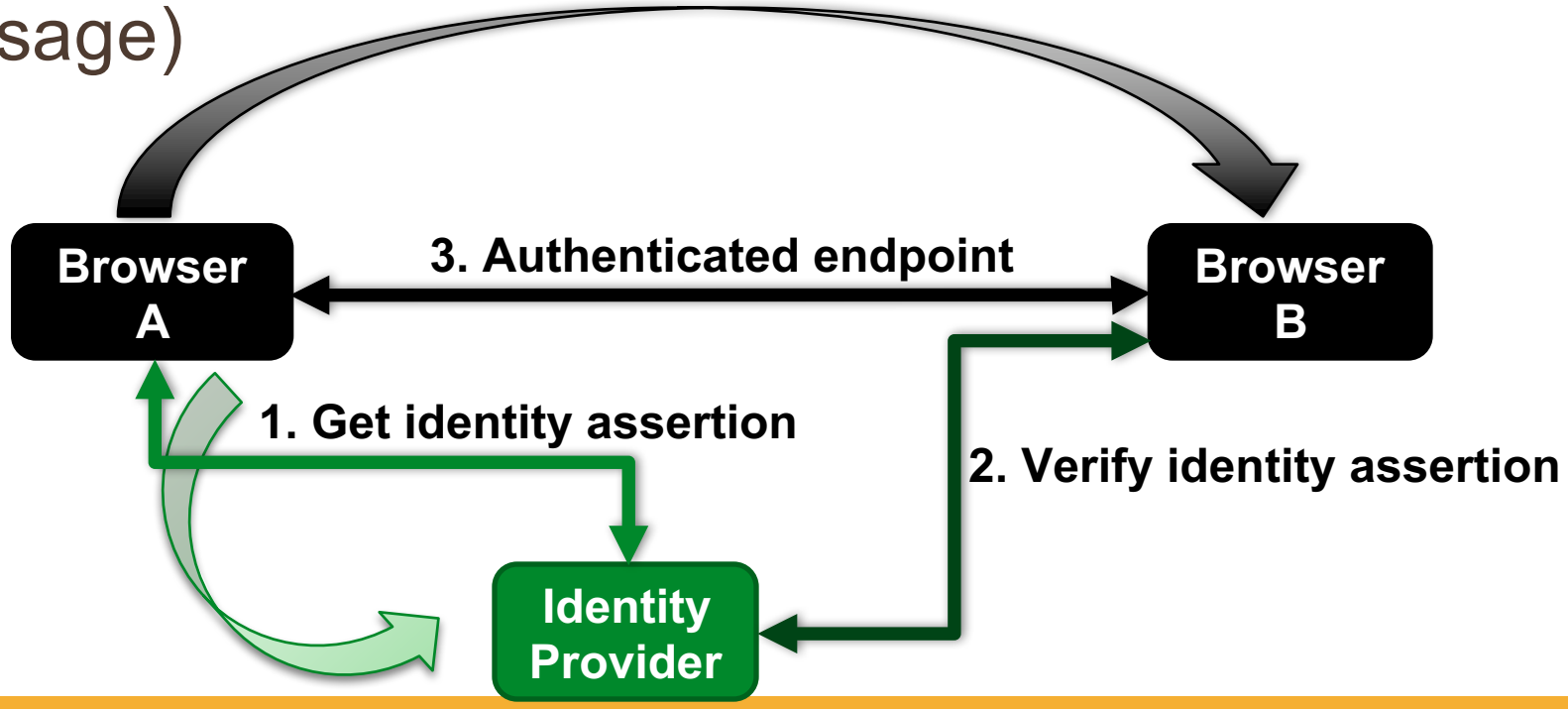


# WebRTC architecture (less simplified)

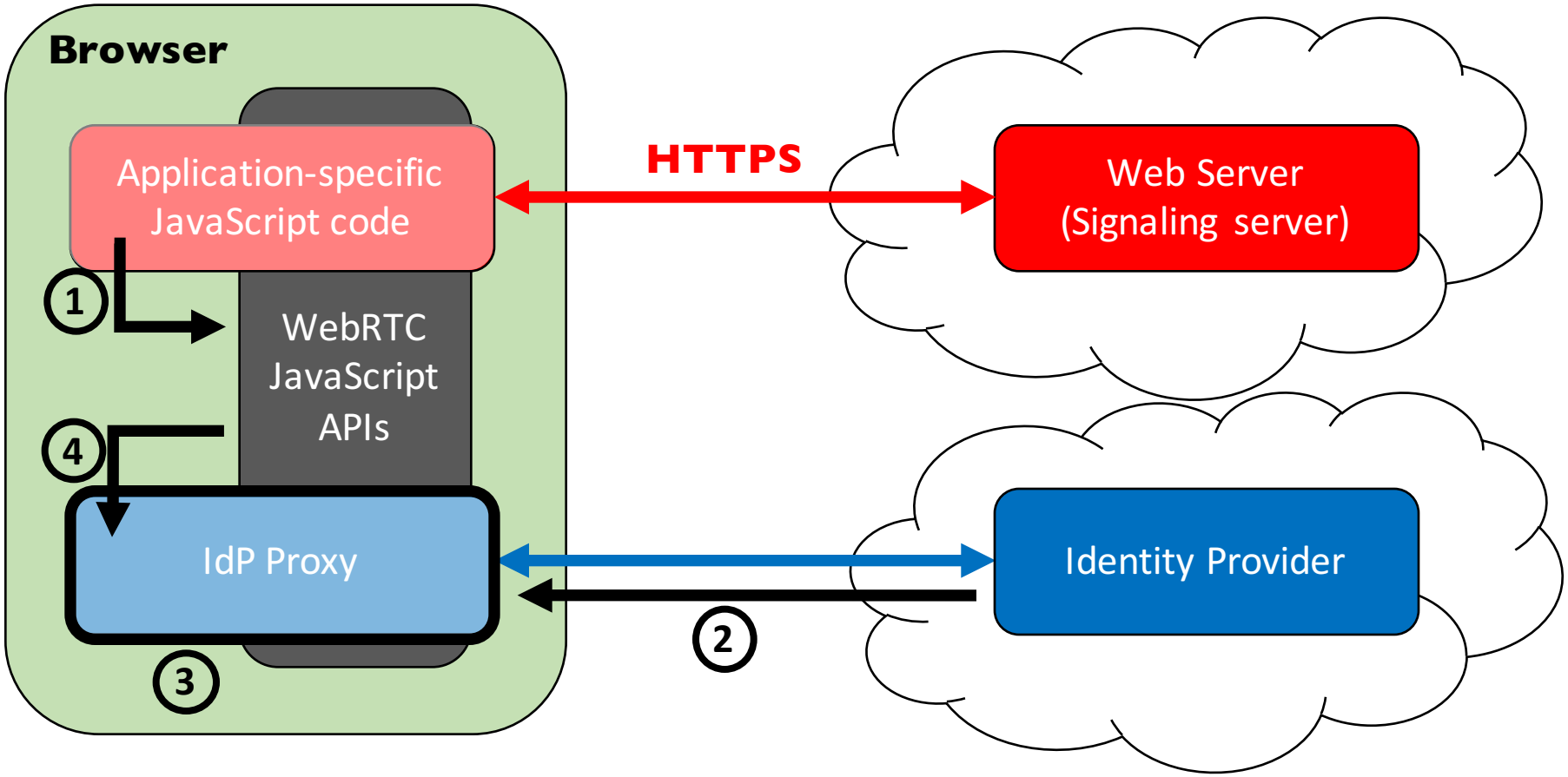


# Identity provision

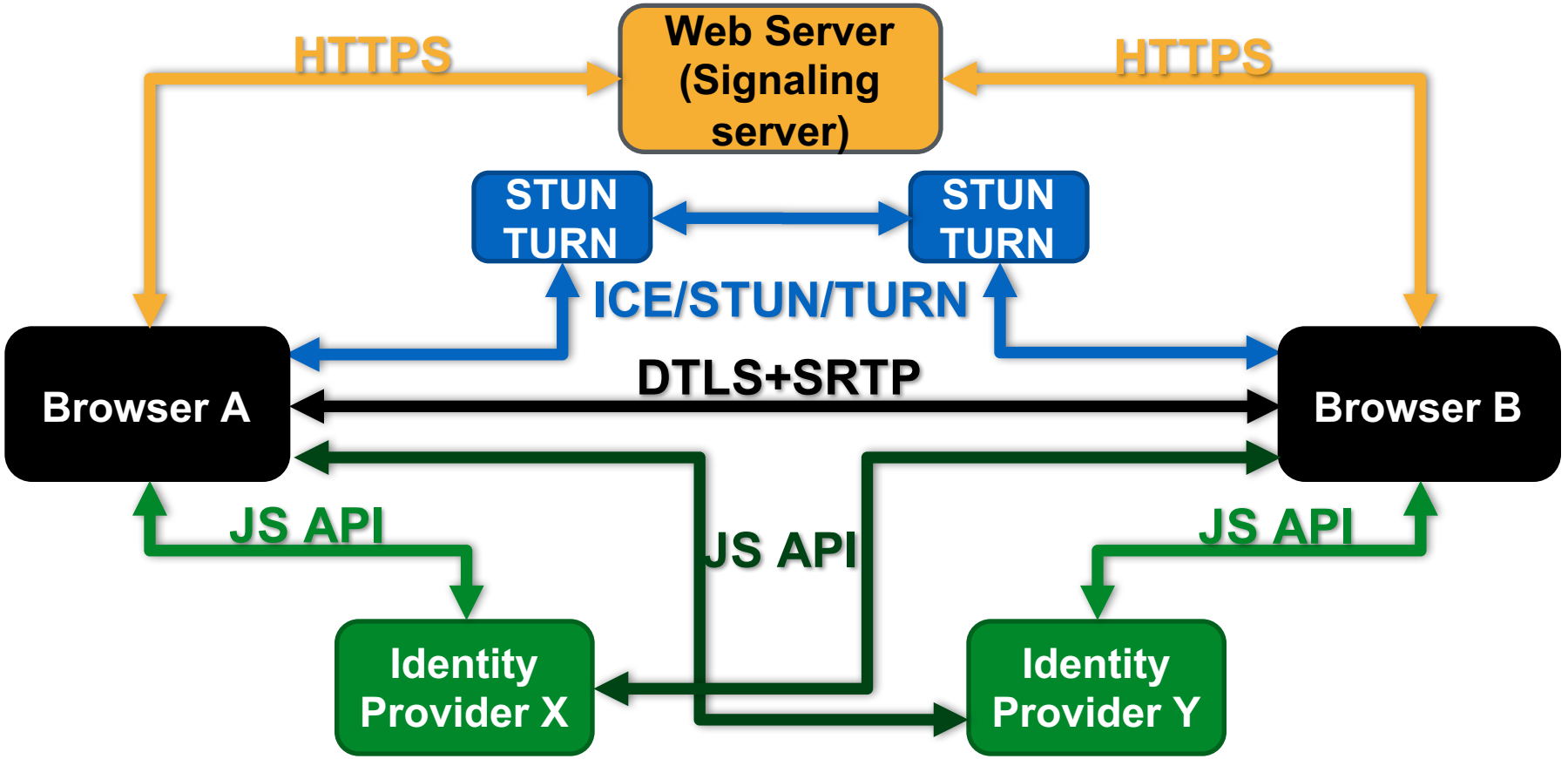
- To authenticate the endpoint with an Identity Provider (IdP)
  - Code of IdP gets loaded in a JavaScript realm
  - Interaction between client-side code and IdP via Web Messaging (aka postMessage)



# Loading the Identity Provider



# WebRTC architecture (full)



# WEBRTC JAVASCRIPT APIS

# Fully JavaScript empowered…

- Purpose of the WebRTC JavaScript APIs
  - Handle Audio/Video streams
  - Both local and remote

- Initialize the browser's P2P capabilities
- Obtain all necessary information, so that the remote party can connect
  - SDP offer
  - ICE candidates

# Setting up a RTCPeerConnection

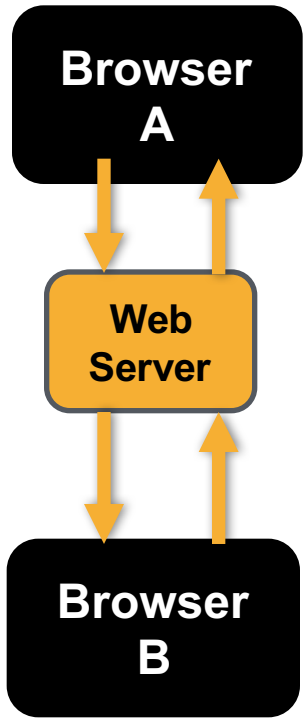
```
// overcome temporary browser differences ☺
RTCPeerConnection = window.RTCPeerConnection ||
window.mozRTCPeerConnection || window.webkitRTCPeerConnection;

// configuration of STUN, TURN, …
// can also be derived automatically by the browser
var configuration = {
    "iceServers": [{ "url": "stun:stun.example.org" }]
};

// Creating the Connection object and add a handler for incoming streams
peerConnection = new RTCPeerConnection(configuration);
```

# Handling SDP offers and answers

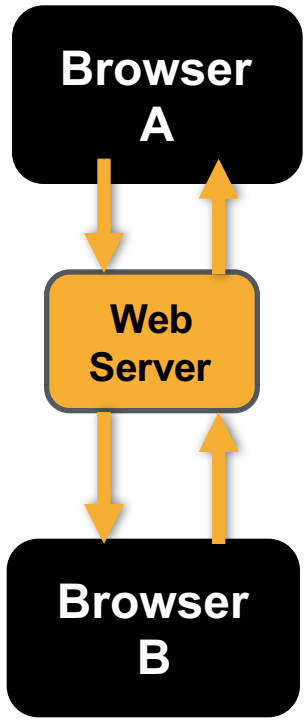
Application-specific signaling!




```
// create a SDP offer on negotiation
peerConnection.onnegotiationneeded = function () {
   peerConnection.createOffer(function (offer) {
      // set it as the Local SDP description and send the offer to the other peer
      return peerConnection.
         signaling
          peerCon
signalingChanne
   if(message.sdp
      var desc = n
      // if we get an
      peerConnecti
         return peerConnection.createAnswer(function (answer) {
            return peerConnection.setLocalDescription(answer, function () {
               signalingChannel.send(JSON.stringify({ "sdp":
                  peerConnection.localDescription }));   })  })  }) } });
```

Exchange of information that a connection shall happen.

# Handling ICE Candidates




```
// send any ice candidates to the other peer
peerConnection.onicecandidate = function (evt) {
    if (evt.candidate) {
        signalingChannel.send(JSON.stringify({ "candidate": evt.candidate }));
    }
};
```

Exchange of information "how" to connect.

```
// receive and process ice candidate
signalingChannel.onmessage = function (msg) {
    if(message.candidate){
        peerConnection.addIceCandidate(new RTCIceCandidate(msg.candidate));
    }
});
```

# Capturing a video stream

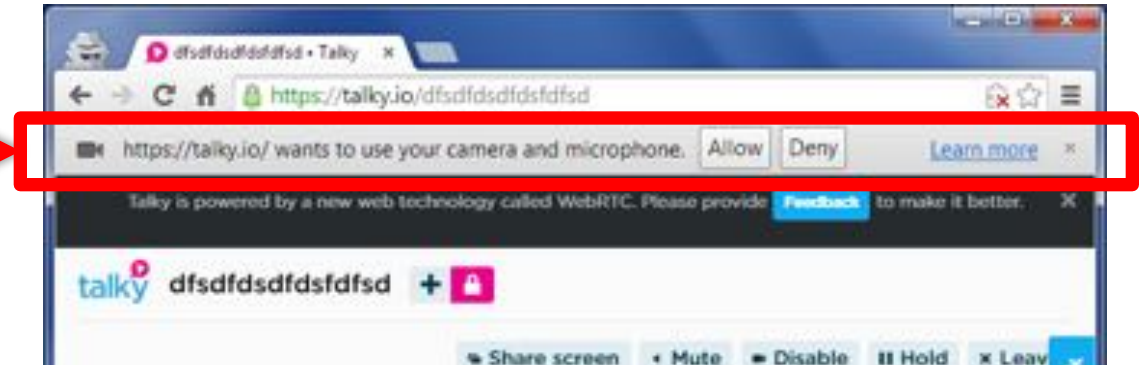
Browser A

Web Server

Browser B

```
// overcome temporary browser differences ☺
navigator.getUserMedia = navigator.getUserMedia ||
navigator.webkitGetUserMedia || navigator.mozGetUserMedia;

// request a UserMedia Stream and use it on the RTCPeerConnection
navigator.getUserMedia({ "audio": true, "video": true }, function (stream) {
    if(stream){
        video1.src = URL.createObjectURL(stream);
        peerConnection.addStream(stream);
    }
}, logError);
```




Asks the user for permission

# Setting up Identity provision

```
// setting up the identity provider
// [ this can also be done by the browser ]
// commented out example: also provide optional protocol and username
// peerConnection.setIdentityProvider("example.com", "default", "alice@example.com");

peerConnection.setIdentityProvider("example.com");

// possible interaction with the IdP proxy
// this is done implicitly by the PeerConnection
peerConnection.getIdentityAssertion ();

peerConnection.onpeeridentity = function(e) {
  console.log("IdP= " + e.target.peerIdentity.idp +
              " identity=" + e.target.peerIdentity.name);
};
```

Happens behind the scenes

WebRTC assessment by EU-funded project STREWS

Joint effort of SAP, W3C, TCD, and KU Leuven

# SECURITY & PRIVACY OF WEBRTC

# Attacker models in scope

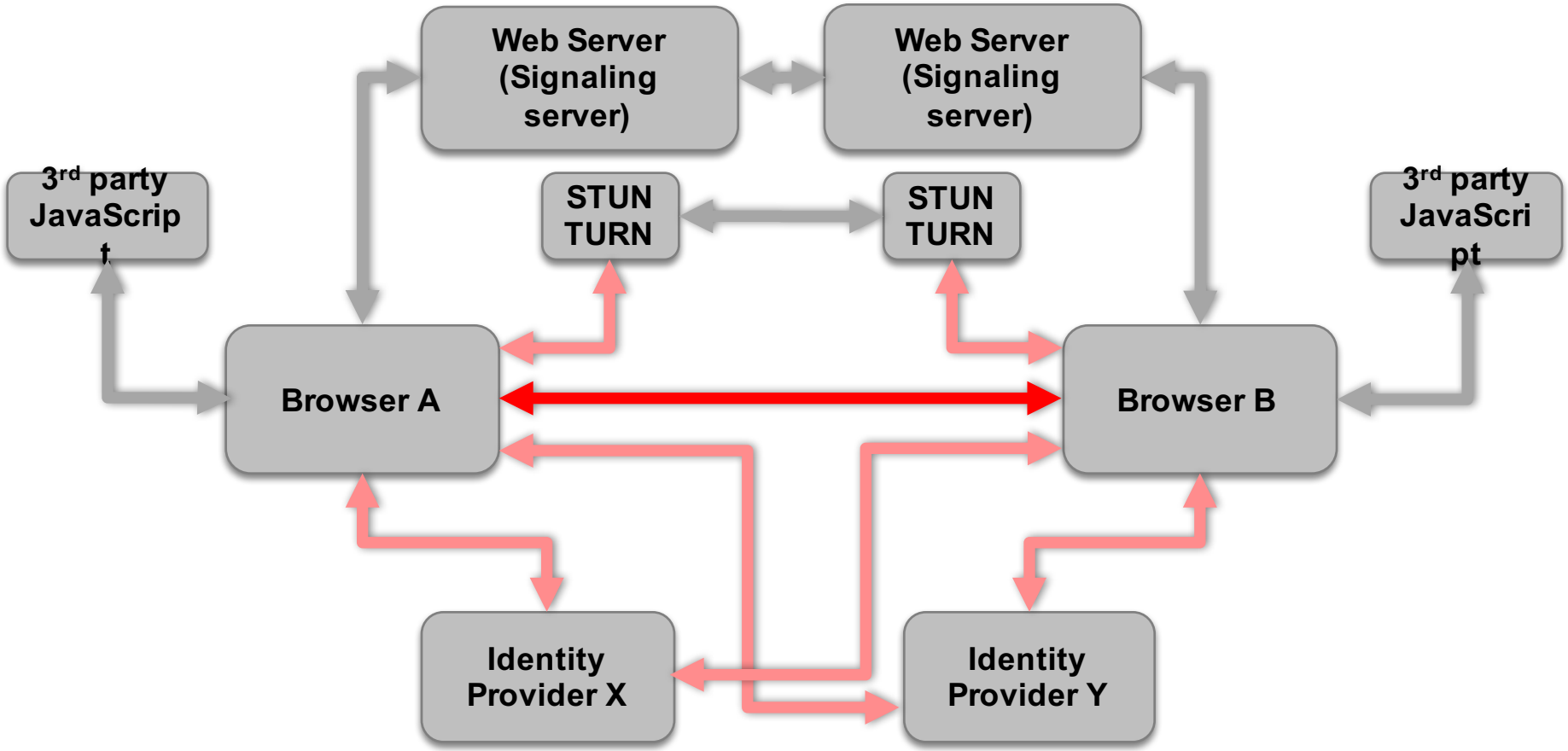
- Network attacker
  - Can observe (and tamper) network traffic
- Malicious web server
  - Sits inline on the signaling path (signaling server)
  - Controls the client-side JavaScript code
- Malicious 3rd party JavaScript
  - Can be injected (XSS) or included
  - Runs in same context as WebRTC JS code

# Overview of the security analysis

- General observations
  - Cross p2p attacker surface
  - WebRTC permission model
- Security impact on WebRTC-enabled websites
  - Potential meta-data leaks
  - Eavesdropping the connection
  - Endpoint authenticity
- Security impact on websites without WebRTC
  - Network info leaks
  - Giving up online privacy

# GENERAL OBSERVATIONS
# #1 CROSS P2P ATTACK SURFACE

# Increased attack surface



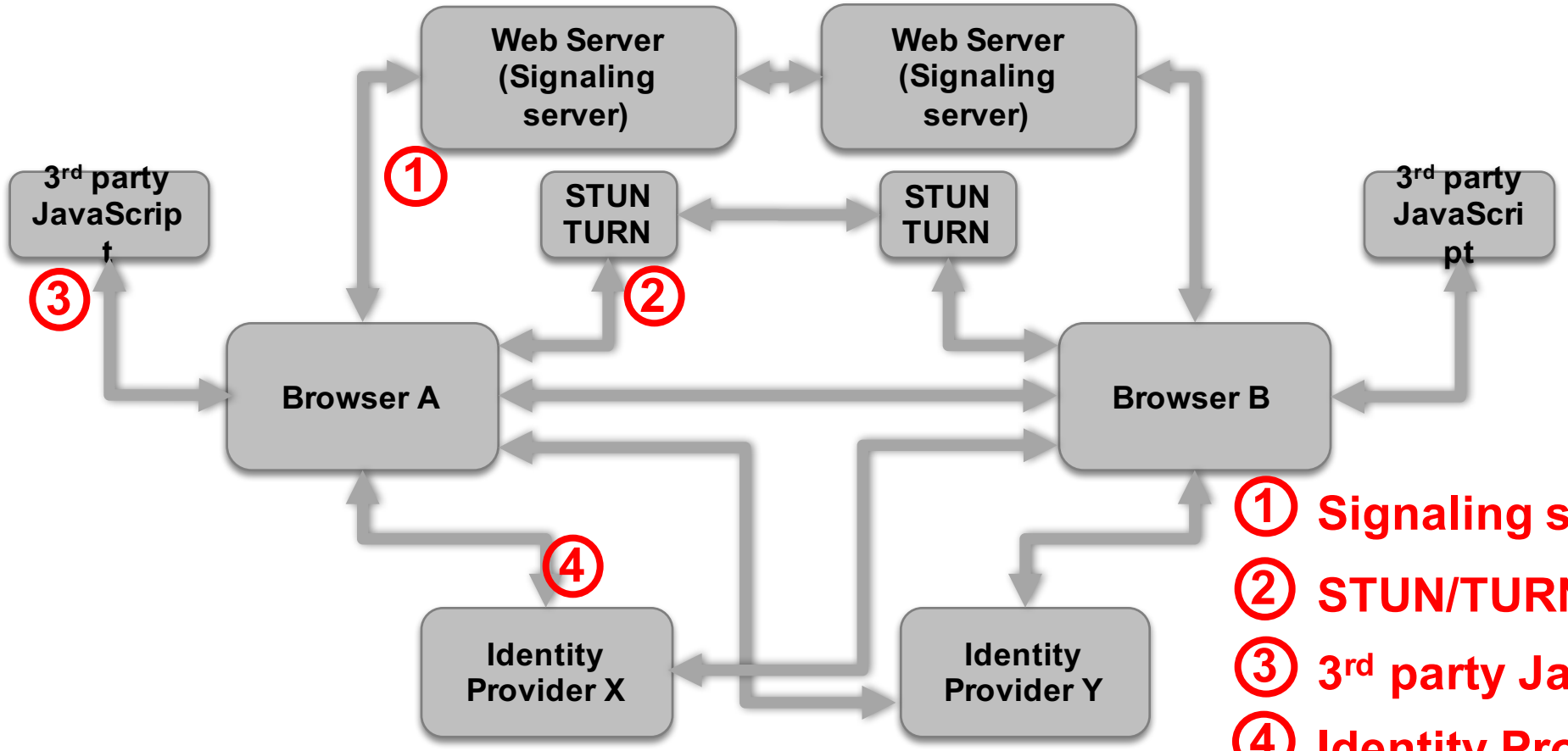
# GENERAL OBSERVATIONS
# #2 WEBRTC PERMISSION MODEL

# Permission model / UI feedback

- For which operation is user consent required?
  - Camera? ✓
  - Microphone? ✓
  - Getting network characteristics (ICE)? ✗
  - Setting up a peer-to-peer connection? ✗
  - Sending your audio/video to a remote peer? ✗
  - Sharing your screen? ✗ ✓
  - Selecting an appropriate Identity Provider? ✗
  - Veryfying your endpoint's identity? ✗

# WebRTC-ENABLED WEBSITES
# #3 POTENTIAL META-DATA LEAKS

# Meta-data leakage: Trace that communication has happened




1. Signaling server
2. STUN/TURN server
3. 3rd party JavaScript
4. Identity Provider

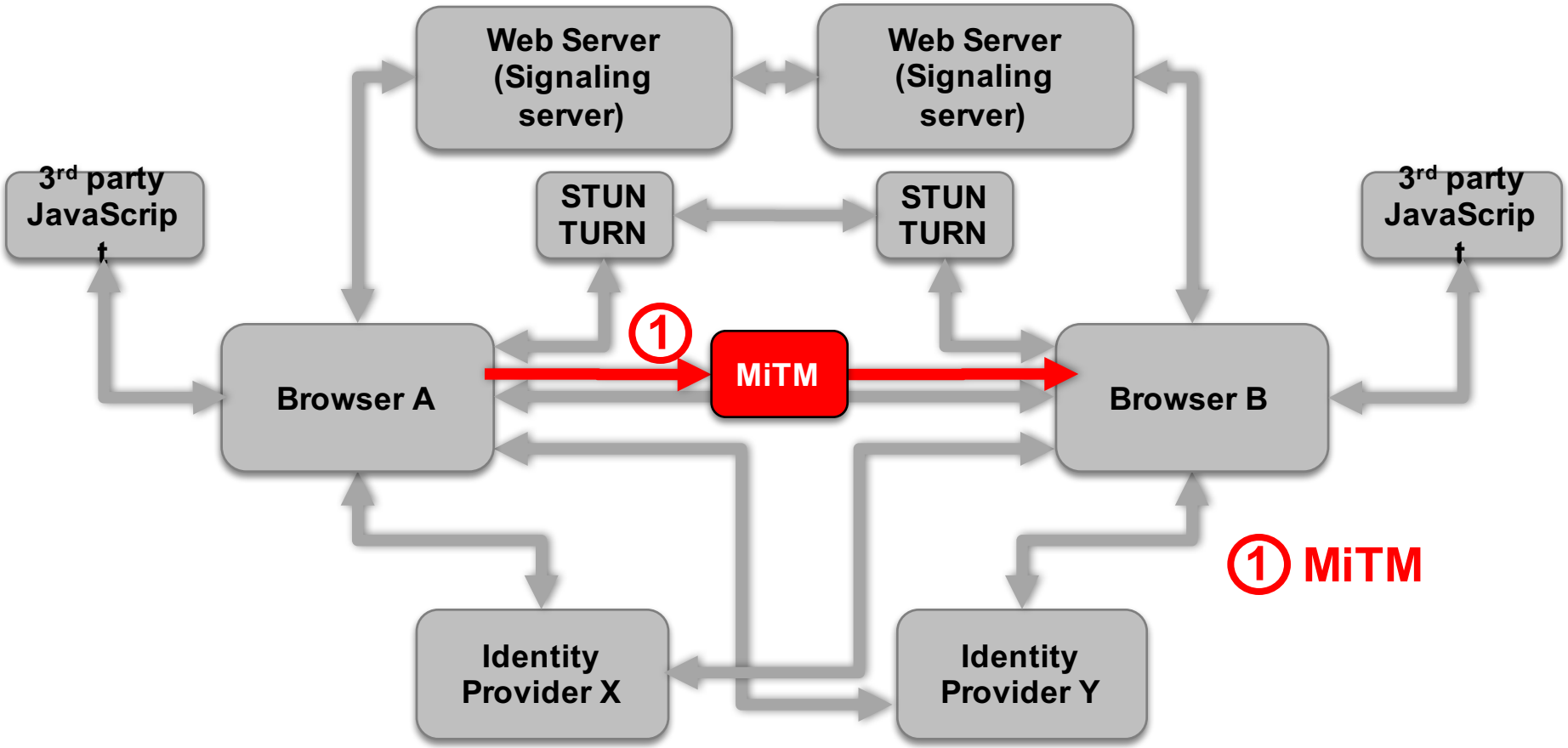
# Meta-data leakage

- Leaking the fact that communication has happened between entities:
  - Signaling server
  - STUN/TURN server (*)
  - IdP server (*)
  - 3rd party JavaScript provider

(*) Possibly by aggregating data from both end-points

# WebRTC-ENABLED WEBSITES
# #4 NETWORK ATTACKERS EAVESDROPPING THE CONNECTION

# Eavesdropping on the connection



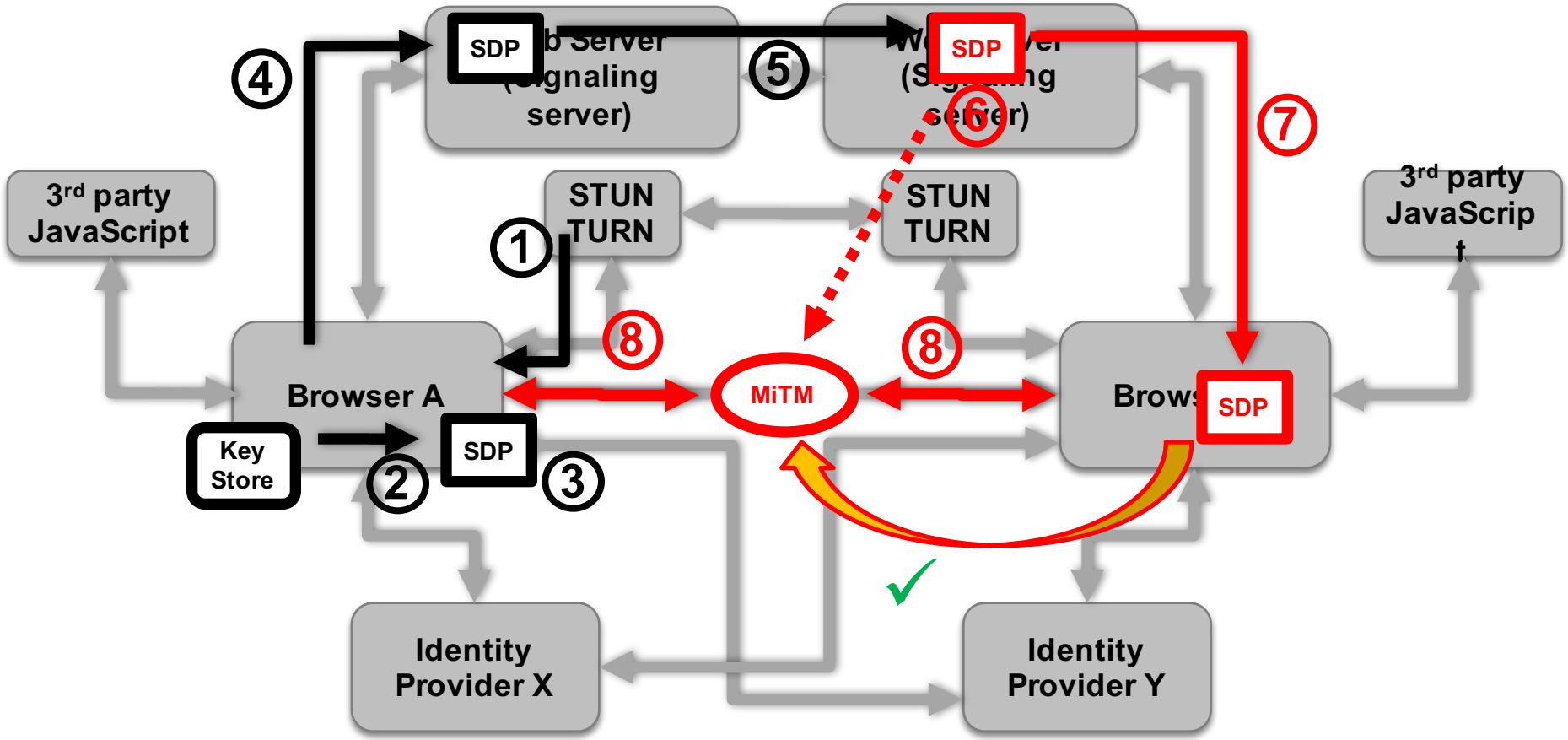
# DTLS-RSTP to the rescue

- DTLS provides
  - Encryption
  - Message authenticity
  - Message integrity

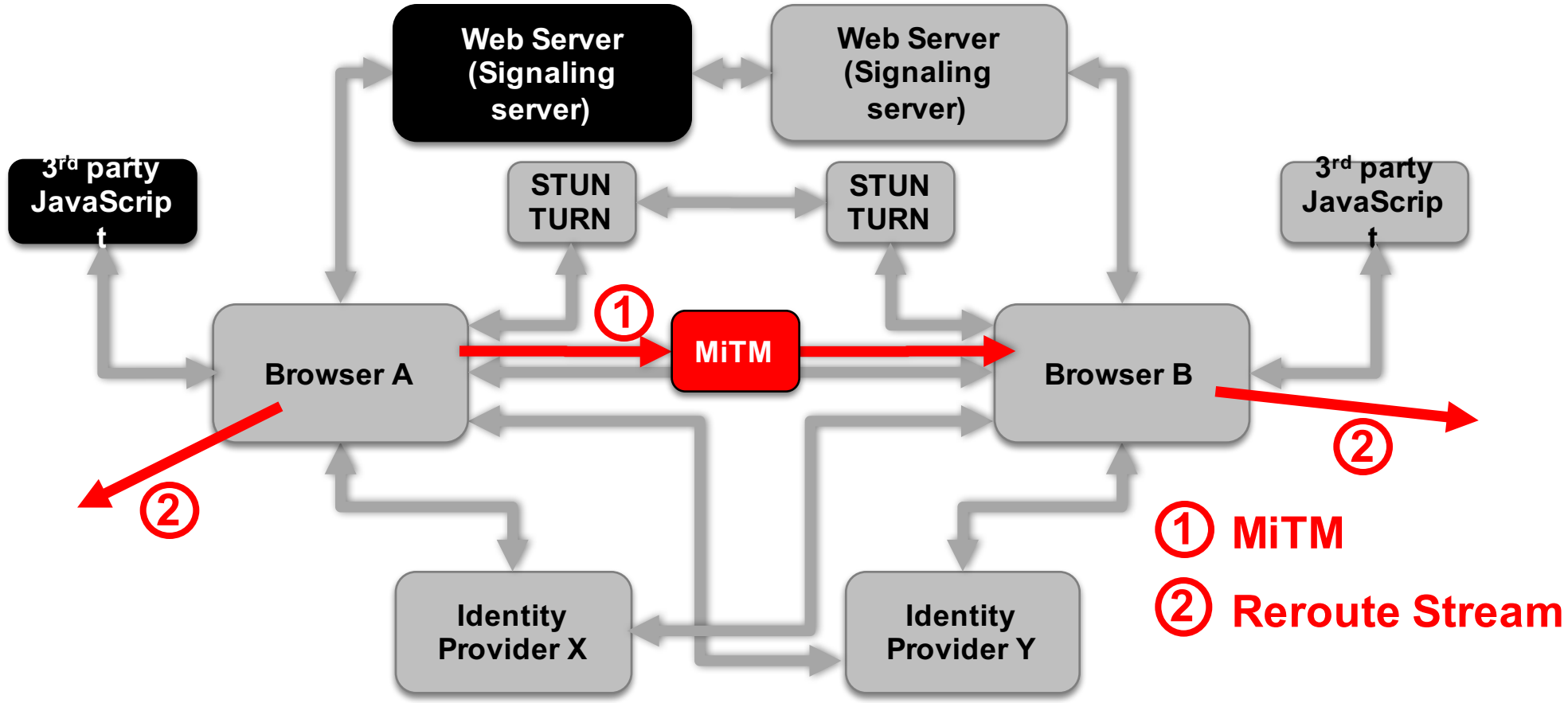
- Endpoint's certificate fingerprint is stored as part of the SDP

# WebRTC-ENABLED WEBSITES
# #5 SIGNALING COMPONENTS
# EAVESDROPPING THE CONNECTION

# Setting up the media channel



# Eavesdropping on the connection



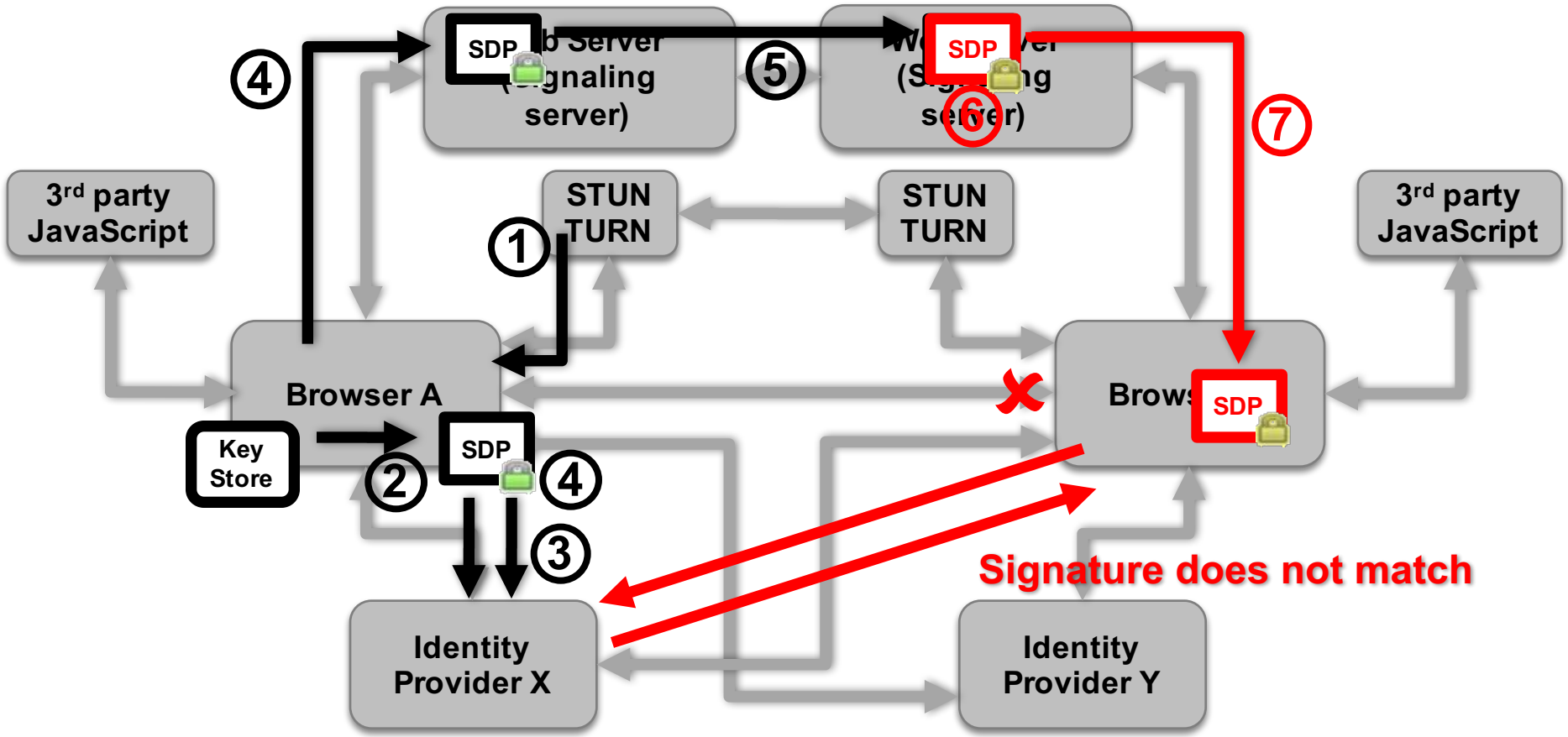
# Eavesdropping on the connection

- Set up the connection to a MiTM
  - By modifying the SDP information
- Reroute the stream
  - By cloning the media stream in JavaScript

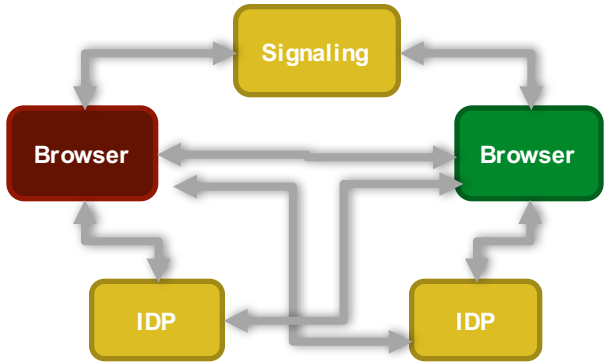
- Can be achieved by:
  - Malicious 3rd party JavaScript (included or via XSS)
  - Malicious signaling server

# WebRTC-ENABLED WEBSITES
# #6 ENDPOINT AUTHENTICITY

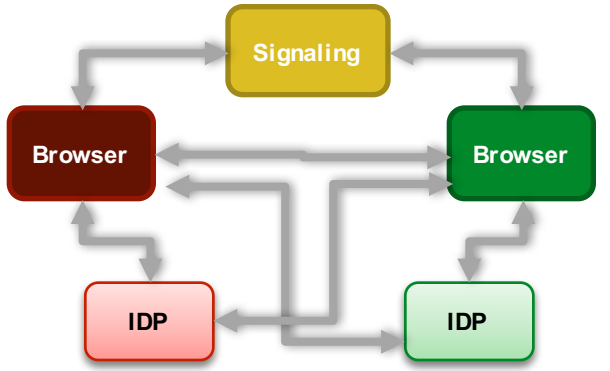
# Setting up the media channel with the IdP



# IDP setups

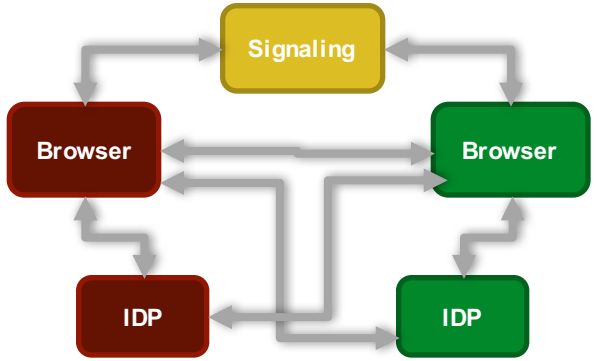



IDP = Signaling Server

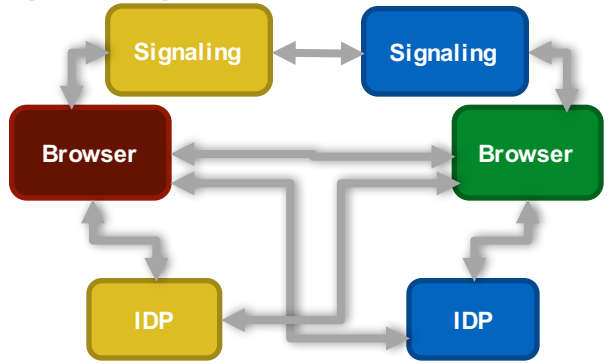



User chooses own IDP

Browser chooses IDP

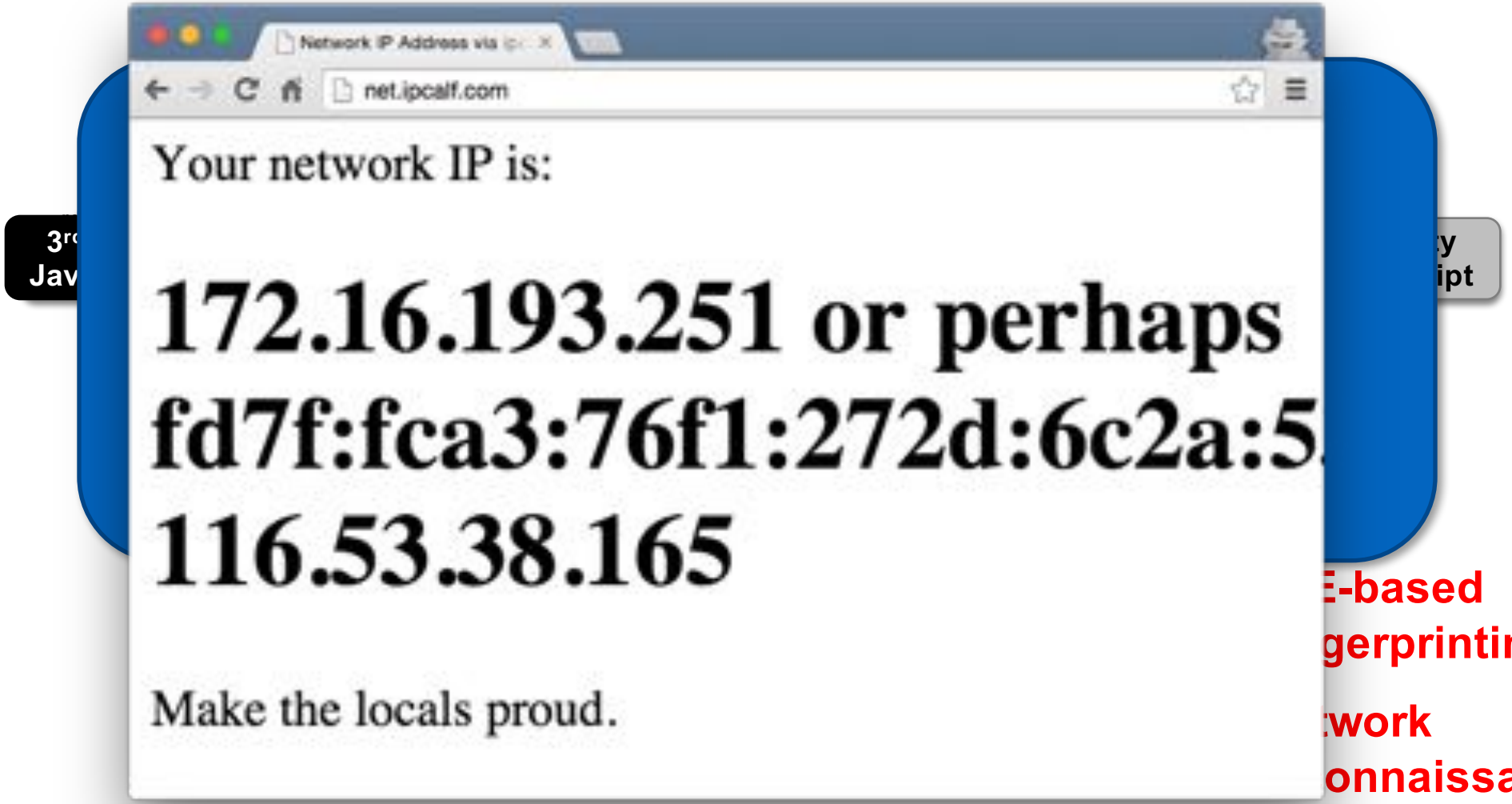



Signaling server chooses IDP



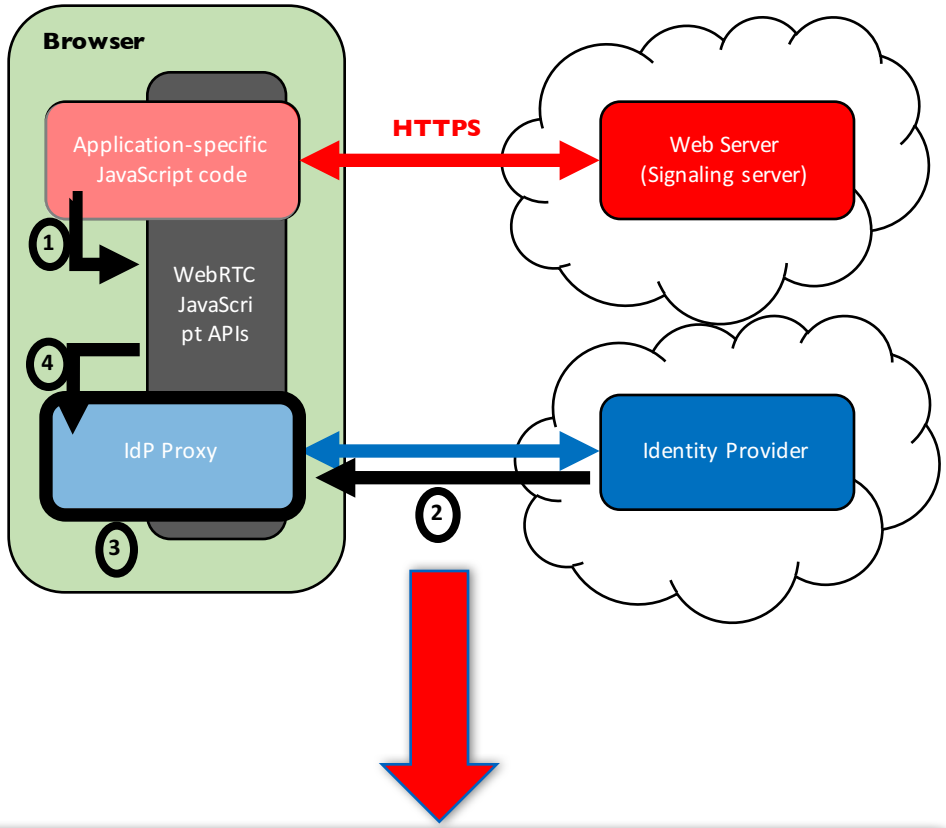
# WEBSITES WITHOUT WebRTC
# #7 LEAK OF LOCAL NETWORK INFO

# Leaking local network info (ICE)



# WEBSITES WITHOUT WebRTC
# #8 HOLY GRAIL OF …
# … GIVING UP ONLINE PRIVACY

# Maliciously triggering the IdP




IdPs will automatically provide identity assertions for <username>@<idp-domain>

- IdP is not allowed to have any user interaction
- If user is logged in:
  - Authenticate based on cookies/localstorage/…
- Else:
  - Fail authentication
  - Return authentication URL to JS signaling code
- Up to signaling code to load authentication URL

# Say goodbye to online privacy…

1. Malicious JS code can trigger automatic identity assertions
2. Online identity can be extracted from the identity assertion

3. Malicious JavaScript code can iterate over set of popular identity providers
4. All retrieved identities can linked to each other

# WRAP-UP

# Take home message

- WebRTC increases the attack surface
- WebRTC permission model is very liberal
  - Your browser has become a peer-to-peer tool without needing your consent
- DTLS-SRTP without an IdP does not authenticate endpoints
  - Use an identity provider to assert the identity of your remote party
  - The IdP integration may be subject to change in the near future

# Take home message (2)

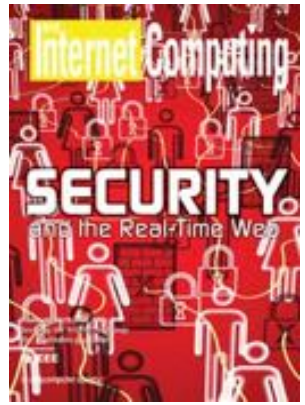
- JavaScript running in your application have full control over the WebRTC functionality
  - To eavesdrop on the media streams
  - To get network and identity information
- Limit trust in 3rd party JS running in your origin
- Use best-practices to protect against XSS

- Embrace the new browser capabilities!

# Relevant sources

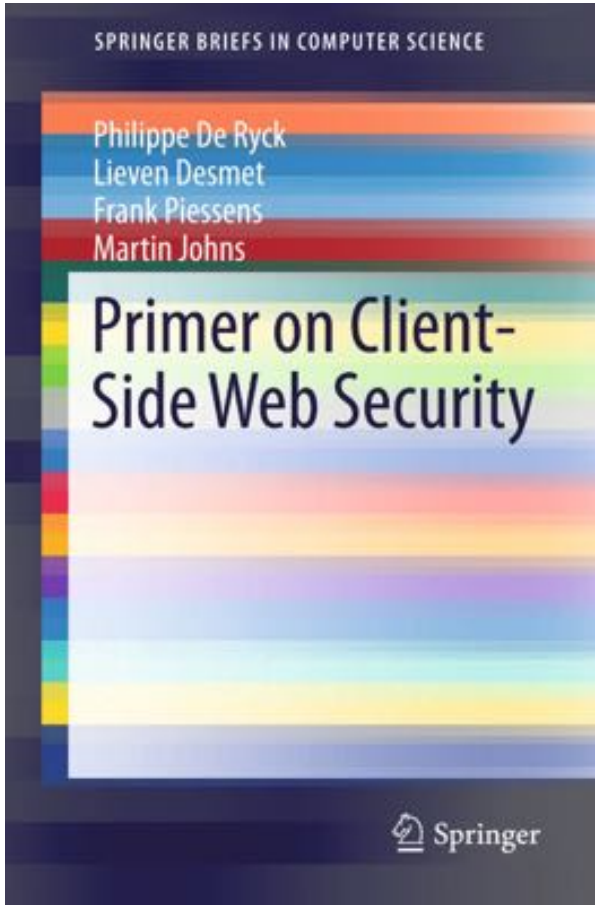
- Large security assessment of relevant specifications
  - Joint work with IETF, W3C and SAP
  - https://www.strews.eu/results/91-d12


- Identifying open issues and security challenges for WebRTC
  - Special Issue of IEEE Internet Computing, nov/dec 2014
  - http://www.computer.org/csdl/mags/ic/2014/06/index.html



# Client-side Web Security Handbook

- Provides an up-to-date overview of
  - State-of-the-art in web security
  - State-of-practice on the Web
  - Recent research and standardization activities
  - Security best practices per category
- http://www.springer.com/us/book/9783319122250




Q&A