



WebRTC mit PeerJS

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24.08.2023

Web  RTC

1.

Was ist WebRTC?

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Was ist WebRTC?

- Web Real-Time Communications
- Web-Standard
- Echtzeitkommunikation zwischen Browsern
- Audio / Video
- Peer-to-Peer



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2. Historie

Ein bisschen Geschichte

- 2011: W3C Arbeitsgruppe zu WebRTC beginnt
 - Peer-to-Peer-Verbindungen bei Skype und Co. im Einsatz
- November 2017: WebRTC 1.0 von “Draft” zu “Candidate Recommendation”
- Januar 2021: WebRTC 1.0 zu “Recommendation”
- Inzwischen stabile API und breiter Browser-Support

Web  RTC

3.

Wie kann ich es
benutzen?

Wie benutze ich WebRTC?

- ICE
- STUN
- TURN
- `RTCPeerConnection`
- `RTCDataChannel`
- `sendOffer`
- `oniceconnectionstatechange`

answer, leaving out the ICE layer for the moment:

1. The caller captures local Media via `MediaDevices.getUserMedia`
2. The caller creates `RTCPeerConnection` and calls `RTCPeerConnection.addTrack()` (Since `addStream` is deprecating)
3. The caller calls `RTCPeerConnection.createOffer()` to create an offer.
4. The caller calls `RTCPeerConnection.setLocalDescription()` to set that offer as the *local description* (that is, the description of the local end of the connection).
5. After `setLocalDescription()`, the caller asks STUN servers to generate the ice candidates
6. The caller uses the signaling server to transmit the offer to the intended receiver of the call.
7. The recipient receives the offer and calls `RTCPeerConnection.setRemoteDescription()` to record it as the *remote description* (the description of the other end of the connection).
8. The recipient does any setup it needs to do for its end of the call: capture its local media, and attach each media tracks into the peer connection via `RTCPeerConnection.addTrack()`.
9. The recipient then creates an answer by calling `RTCPeerConnection.createAnswer()`.
10. The recipient calls `RTCPeerConnection.setLocalDescription()`, passing in the created answer, to set the answer as its local description. The recipient now knows the configuration of both ends of the connection.
11. The recipient uses the signaling server to send the answer to the caller.
12. The caller receives the answer.
13. The caller calls `RTCPeerConnection.setRemoteDescription()` to set the answer as the remote description for its end of the call. It now knows the configuration of both peers. Media begins to flow as configured.

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3. The caller calls `RTCPeerConnection.createOffer()` to create an offer.
4. The caller calls `RTCPeerConnection.setLocalDescription()` to set that offer as the *local* end of the connection).
5. The caller uses a signaling server (or IN servers) to generate the ice candidates and send the offer to the intended receiver of the call.
6. The recipient calls `RTCPeerConnection.setRemoteDescription()` to record it (ie other end of the connection).
7. The recipient uses a signaling server to send the answer to the caller.
8. The recipient does any setup it needs to do for its end of the call: capture its local media, and attach each media tracks into the peer connection via `RTCPeerConnection.addTrack()`.
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Wie benutze ich PeerJS?

```
import { Peer } from 'peerjs';

const peer = new Peer();

const myStream = await navigator.mediaDevices.getUserMedia(
  { audio: true, video: true }
);

peer.on('open', (id) => console.log(`Meine ID ist: ${id}`));
peer.on('call', (call) => {
  call.answer(myStream);

  call.on('stream', (theirStream) => {
    const videoElement = document.querySelector("video");
    videoElement.srcObject = theirStream;
  });
});
```

```
import { Peer } from 'peerjs';

const peer = new Peer();

const myStream = await navigator.mediaDevices.getUserMedia(
  {audio: true, video: true}
);

const call = peer.call("<remote Id>", myStream);

call.on('stream', (theirStream) => {
  const videoElement = document.querySelector("video");
  videoElement.srcObject = theirStream;
});
```

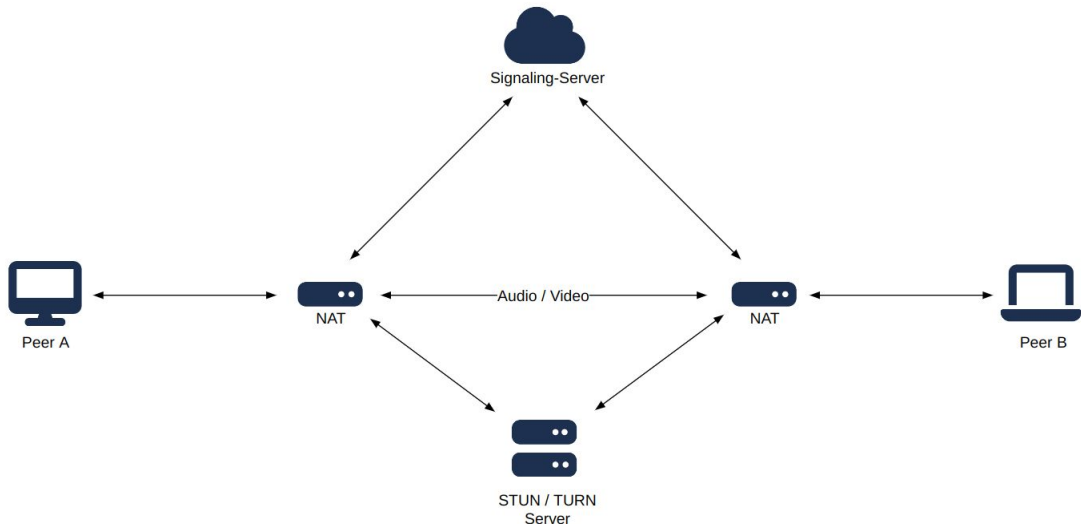
Web  RTC

4.

Was passiert da
(so in etwa)?

Wie funktioniert WebRTC?

- Signaling-Server:
Vermittelt
- STUN:
"wieistmeineip.de"
- TURN:
Weiterleitung statt P2P
- ICE:
Interactive
Connectivity
Establishment



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5.

Live Demo

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6.

Weitere Details

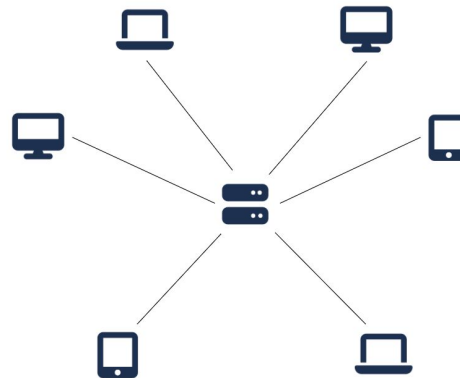
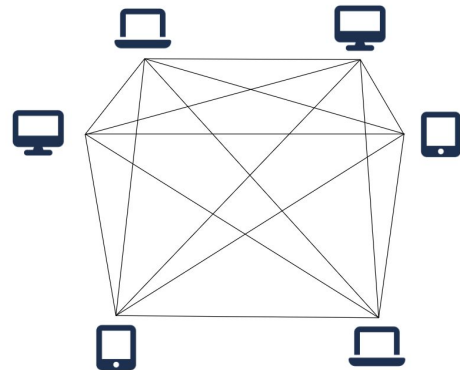
Weitere Details

Dev-Tools

- Chrome: `chrome://webrtc-internals`
- Firefox: `about:webrtc`

Performante Gruppen-Calls

- Multipoint Control Unit (MCU)
- Selective Forwarding Unit (SFU)



Security von WebRTC

Standard hat Security gleich mitgedacht

- Verschlüsselung ist Pflicht
- Sicherer Schlüsselaustausch
- Auch TURN-Server kann nicht mitlesen
- SFUs / MCUs hebeln Ende-zu-Ende-Verschlüsselung aus
- Signalling und Darstellung sind Eigenverantwortung

Weitere Use-Cases

Screen-Sharing

- `getDisplayMedia()`

Data-Channels

- Dateien schicken mit `sharedrop.io`
- Filesharing mit `webtorrent.io`

Überblick

- WebRTC ist mächtig
- WebRTC ist kompliziert
- PeerJS übernimmt Signalling und ICE-Austausch
- MediaStreams werden an HTML5 video-Elemente gehängt
- Eigenes Backend für non-triviale Use-Cases benötigt

```
navigator.mediaDevices.getUserMedia()
```

```
navigator.mediaDevices.getDisplayMedia()
```

```
<video autoplay></video>
```

```
peer.call()
```

```
peer.on("call", ...)
```

```
call.on("stream", (stream) => {  
    videoElement.objSource = stream  
});
```

Links

Code und Folien von heute:

- <https://github.com/sbungartz/frontend-freunde-webrtc-demo>

Weitere Links:

- <https://webrtc.org/getting-started/overview>
- https://developer.mozilla.org/en-US/docs/Web/API/WebRTC_API
- <https://github.com/peers/peerjs>

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Danke