实时多媒体处理与传输

——Web下一代RTC技术实践

高纯

声网

Background

5G

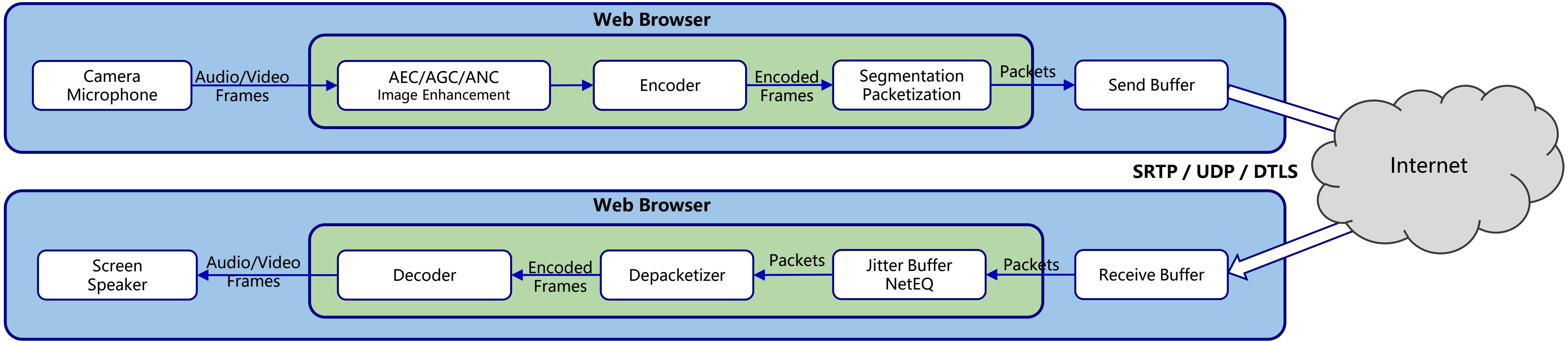
ARTIFICIAL INTELLIGENCE

CLOUDGAME

WebRTC

Capabilities:

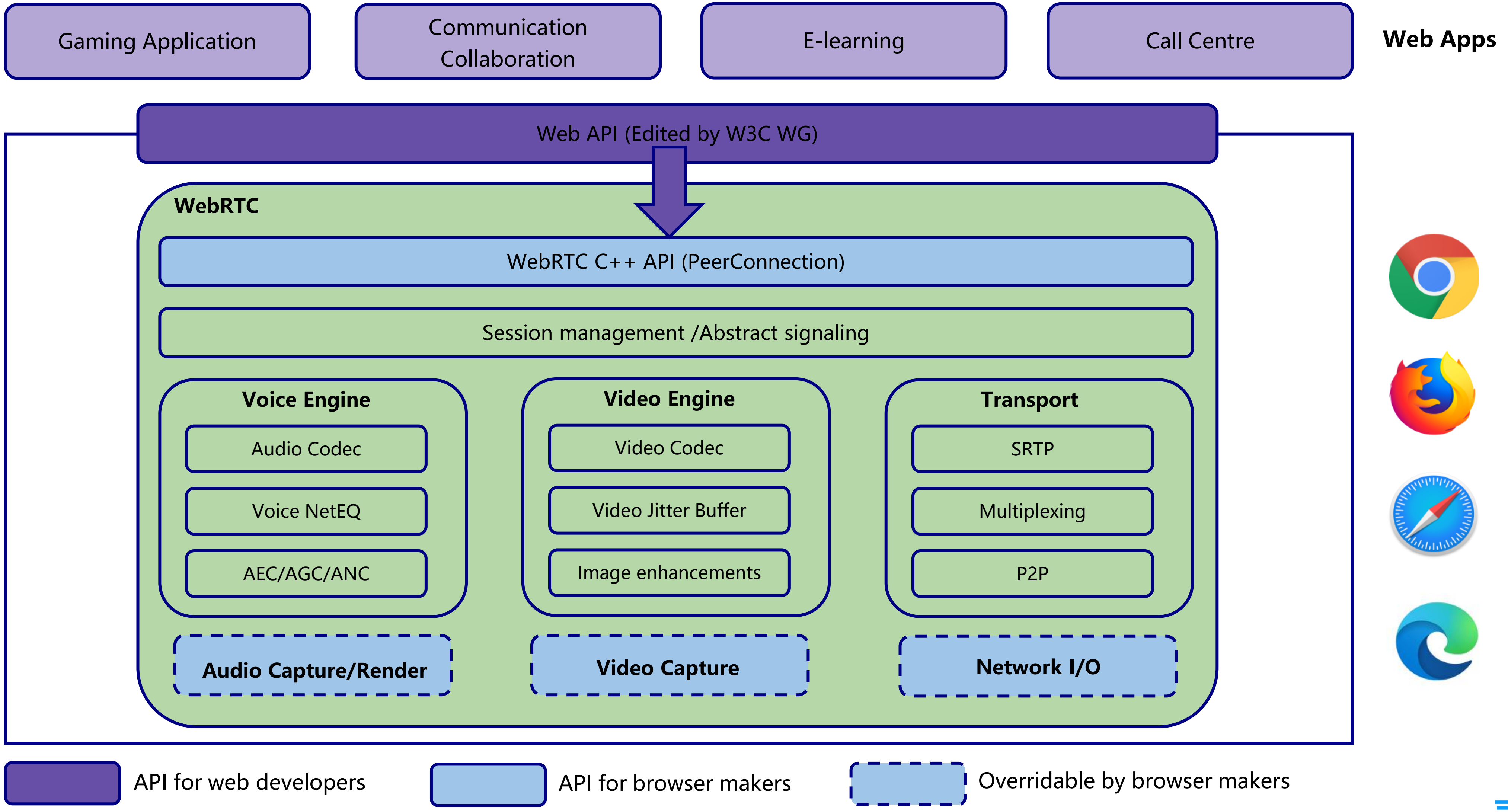
- Media capture
- Adaptive encoding (VP8, VP9, H264)
- Media Processing (AEC AGC ANC)
- Capability Negotiation (SDP)
- NAT traverse （TURN/STUN）
- Network QoS (NACK, FEC, TCC, GCC)
- Encryption (DTLS)



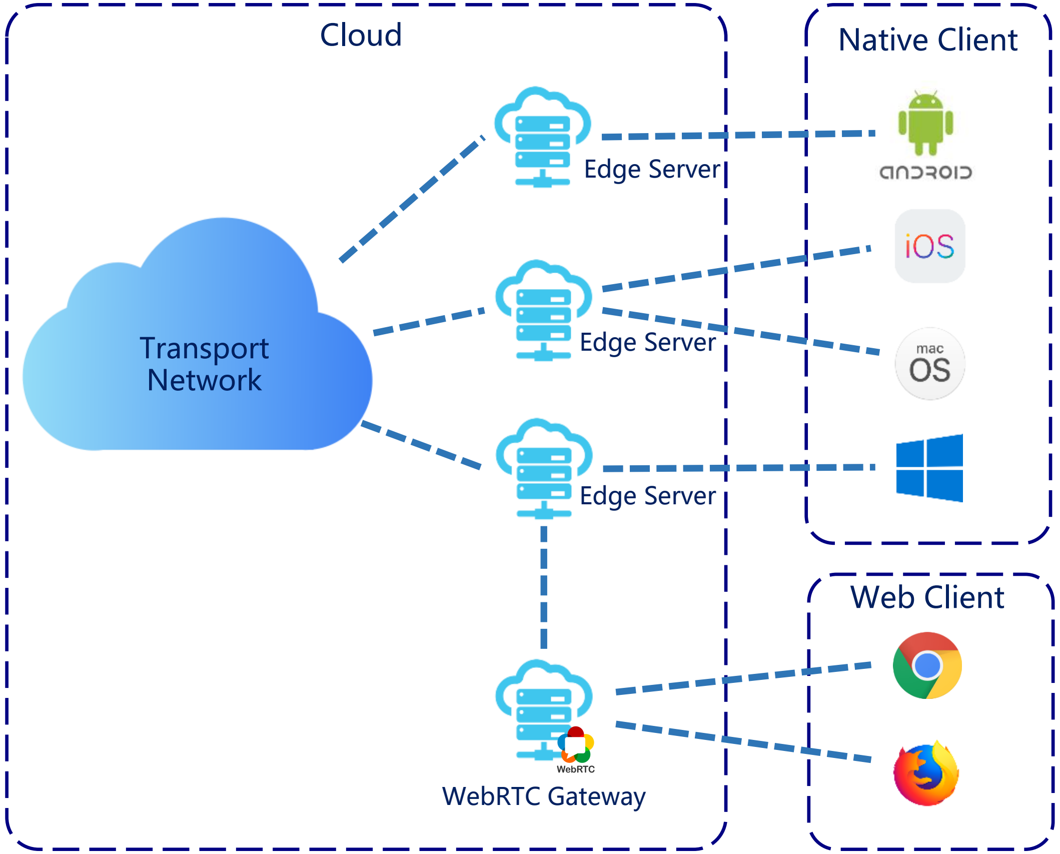
Specification:

- [WebRTC 1.0: Real-Time Communication Between Browsers]()
- [Media Capture and Streams]()

WebRTC In Browser



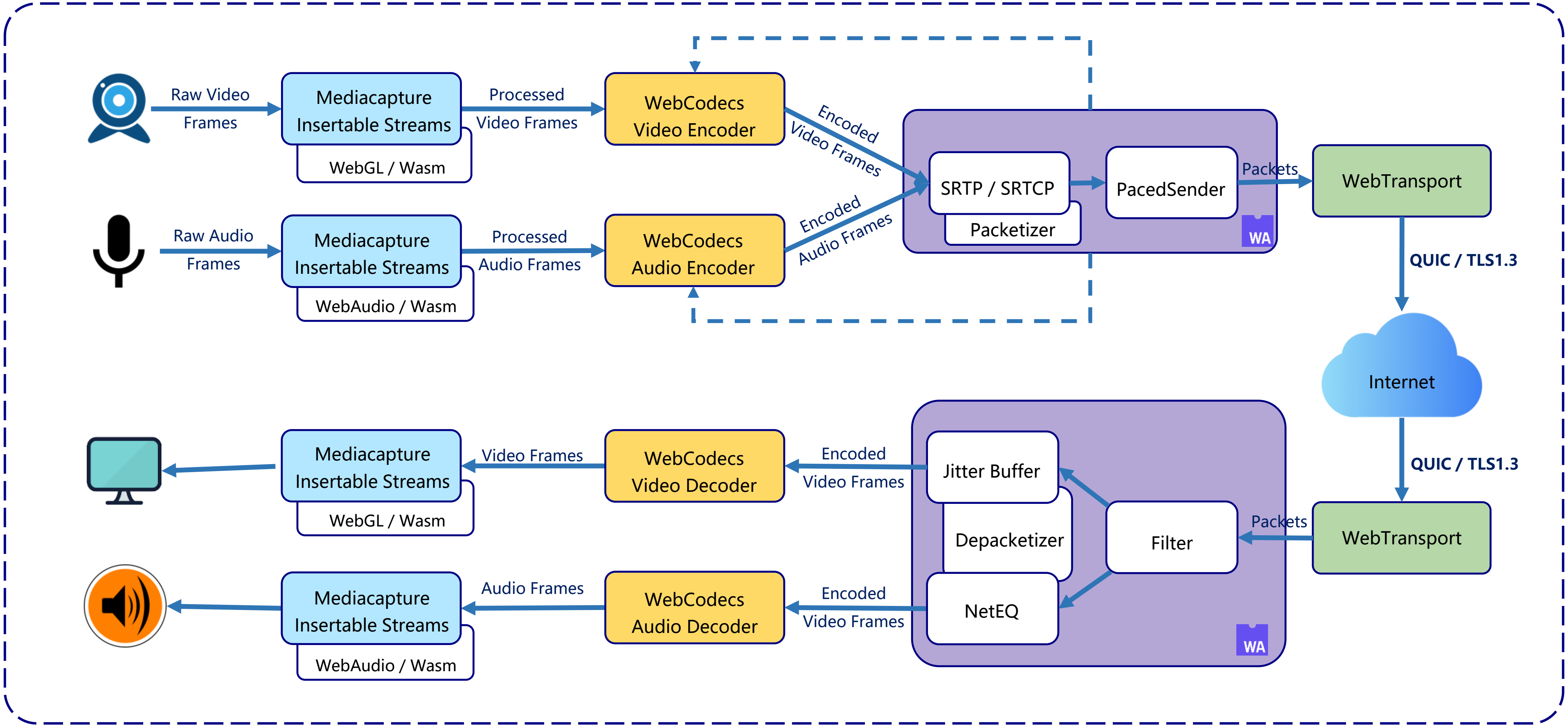
WebRTC based System



Capabilities Required:

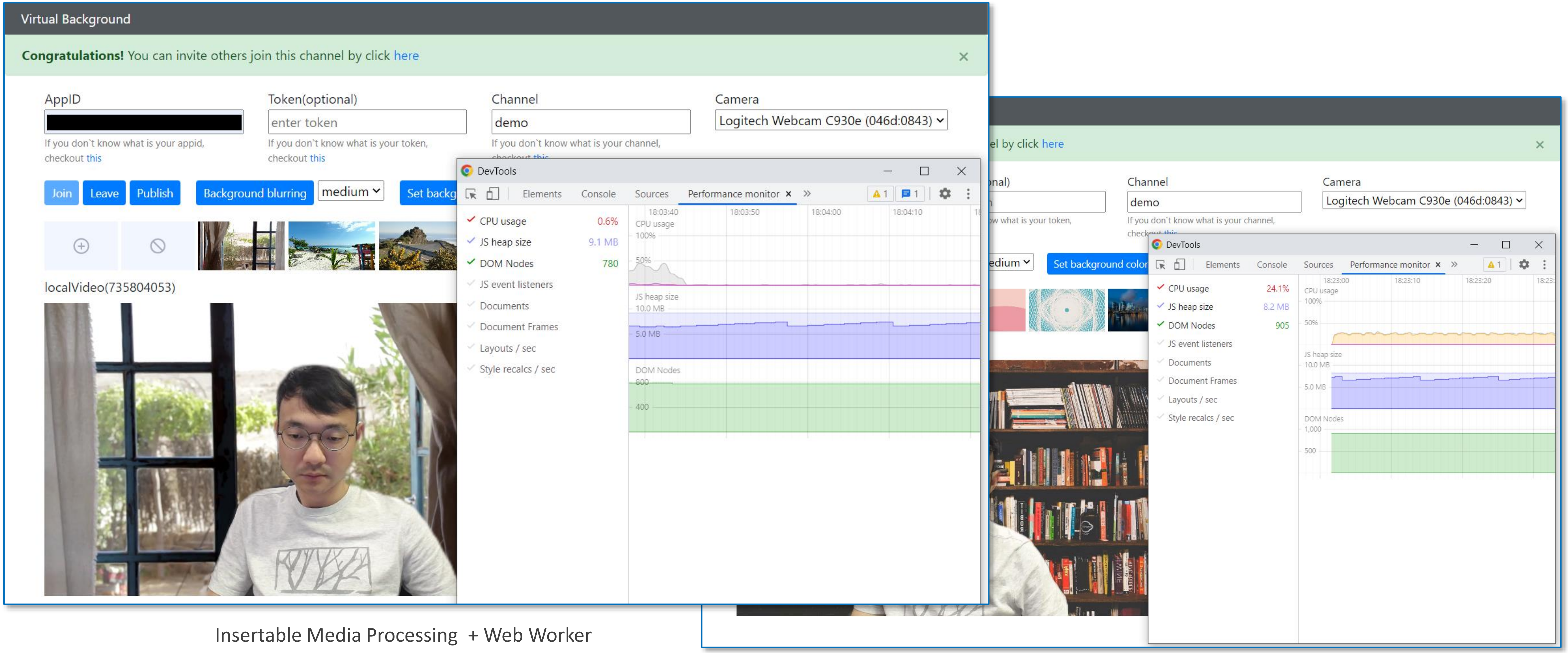
- Customized transporting protocols
- Customized codecs
- Data channel which is exactly synchronized with media data to support online gaming and metaverse
- Enhanced media processing algorithms (AI based 3A, image enhancement, etc.)
- Alpha channel transporting
- …

WebRTC NV based Solutions



Raw Data to Web Worker

- CPU usage of JS main thread



Insertable Media Processing + Web Worker

Processing in main thread

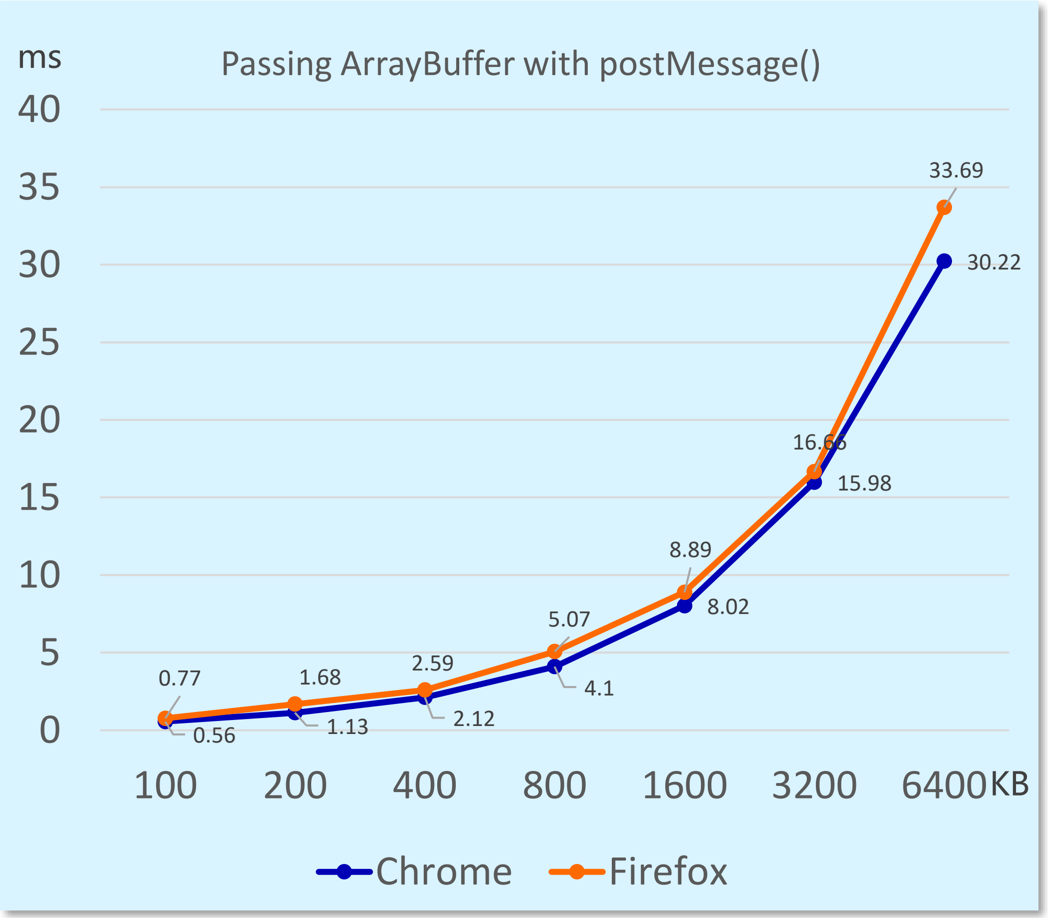
[Agora Virtual Background online Demo]

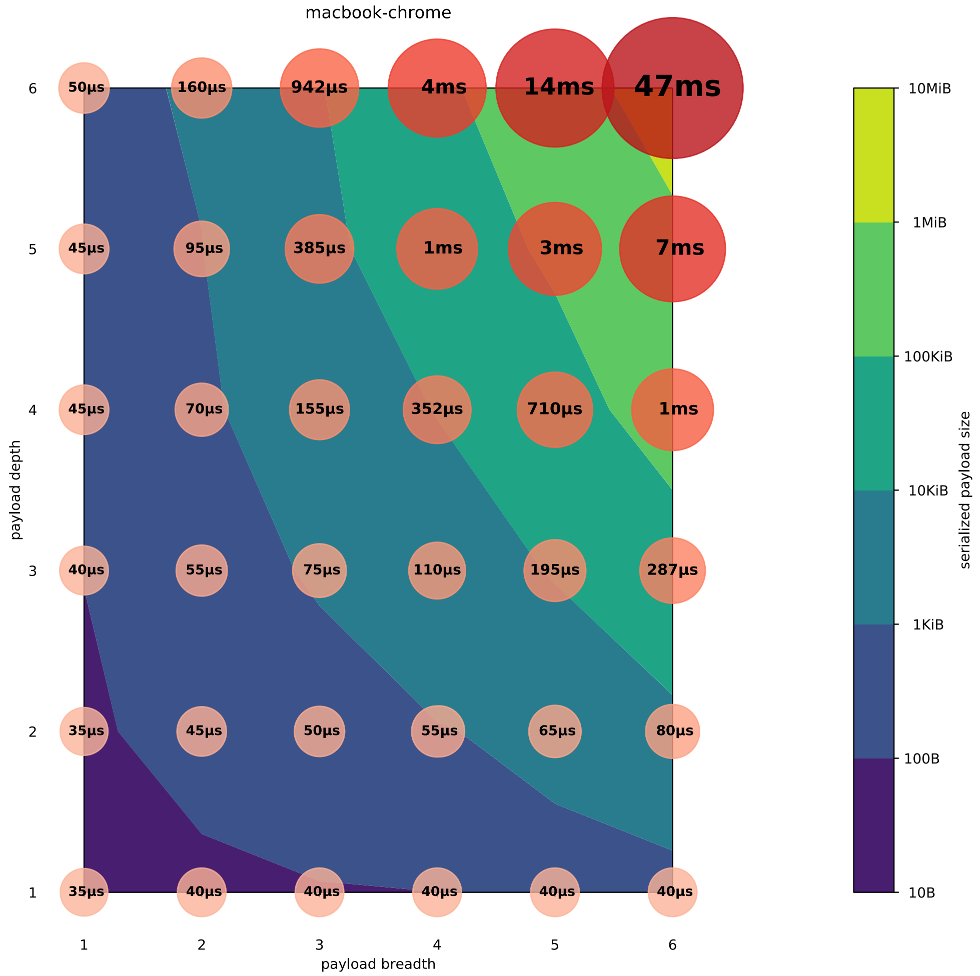
Raw Data to Web Worker

In general, media data should be processed in Web Worker to enhance performance of JS man thread.

- postMessage() with ArrayBuffer performance

[postMessage performance data from Surma]





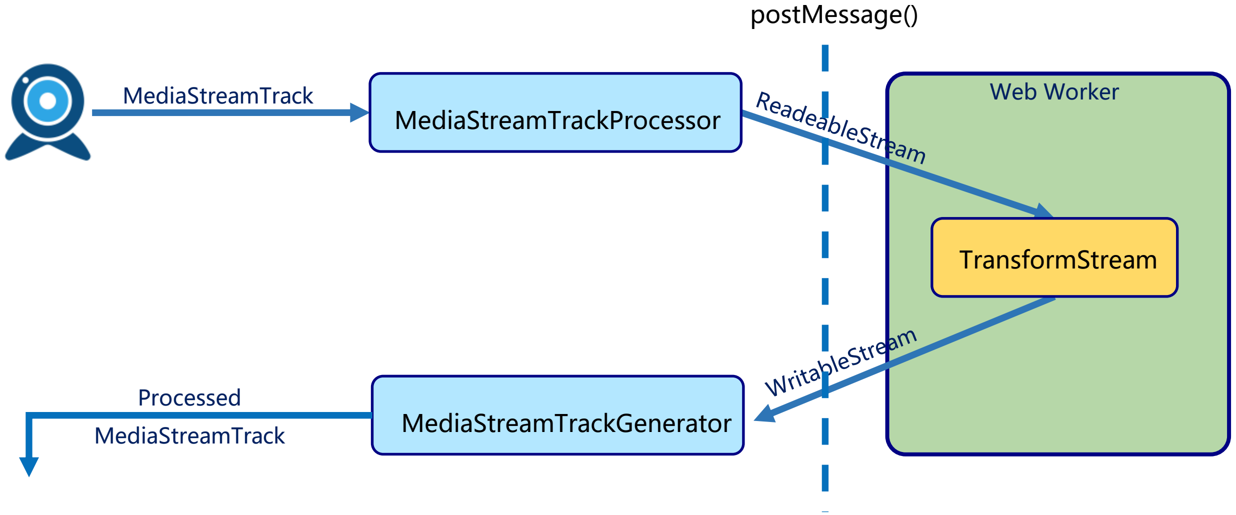
agora

Raw Data to Web Worker

Passing raw data to worker with transferable objects.

- MediaStreamTrack Insertable Media Processing using Streams

```
const transformer = new TransformStream({
  async transform(videoFrame, controller) {
    let bluredFrame = blurBackground(videoFrame);
    videoFrame.close();
    controller.enqueue(bluredFrame);
  }
});

readableStream.pipeThrough(transformer)
              .pipeTo(writableStream);
```



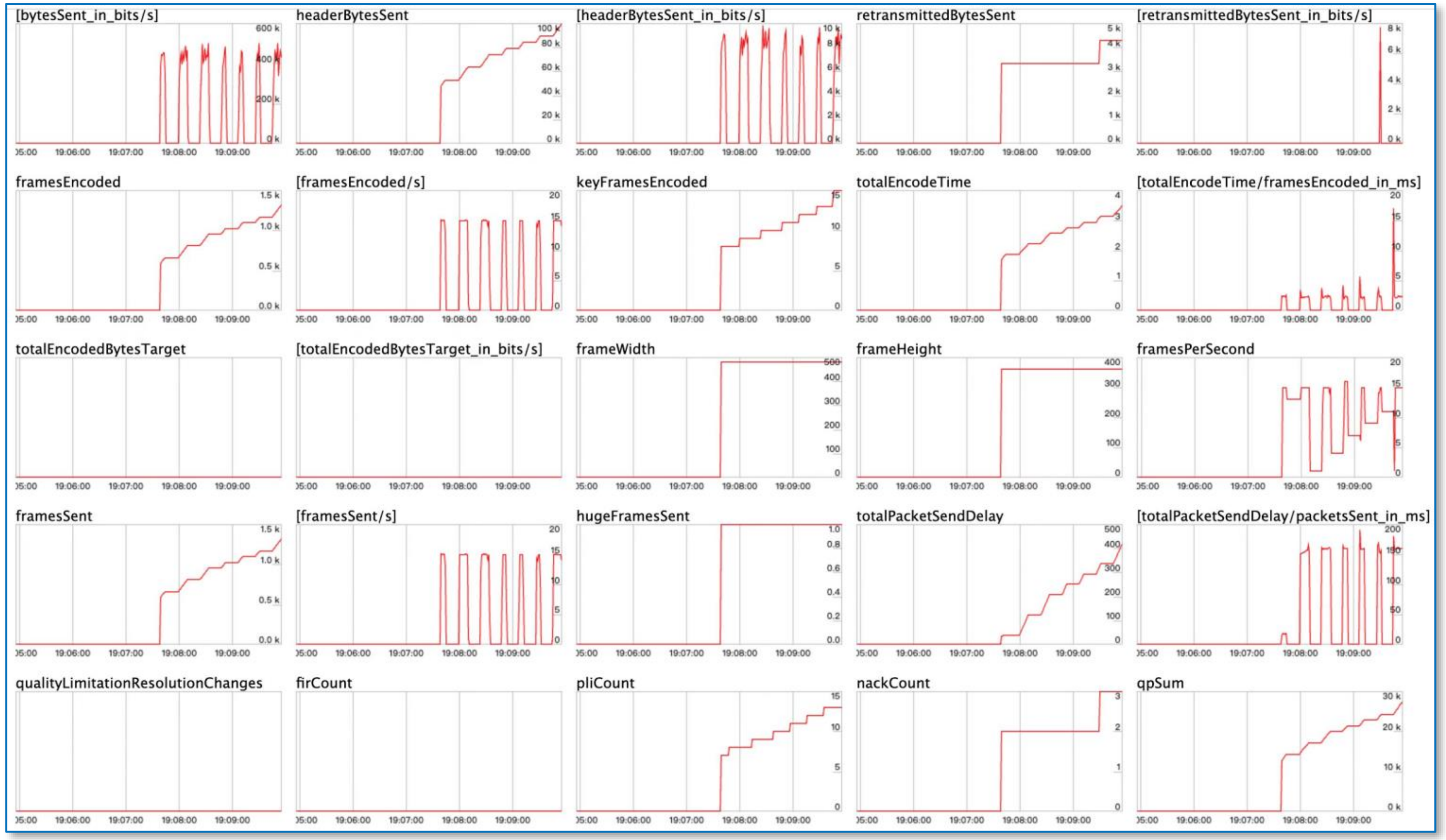
- mediacapture-extensions (Mozila Draft)

```
[Exposed=(Window,Worker), Transferable]
partial interface MediaStreamTrack {
};
```

声网 agora

Known Issues

- MediaStreamTrackGenerator h264 HW encoding compatibility issue



A same h264 HW encoding issue as MediaStreamTrack from HTMLMediaElement.captureStream() or from remote. Related chromium issues:

- Issue 1156408: If the stream obtained via HTMLVideoElement.captureStream() is sent on peer connection, it does not send any actual data.

- Issue 1132965: MediaStream when forwarded through WebRTC stops sending frames, tainted?

WebCodecs Issues

- 48bit image is not supported by WebCodecs

IDL

```
interface VideoFrame {
  constructor(CanvasImageSource image, optional VideoFrameInit init = {});
  ...
}
```

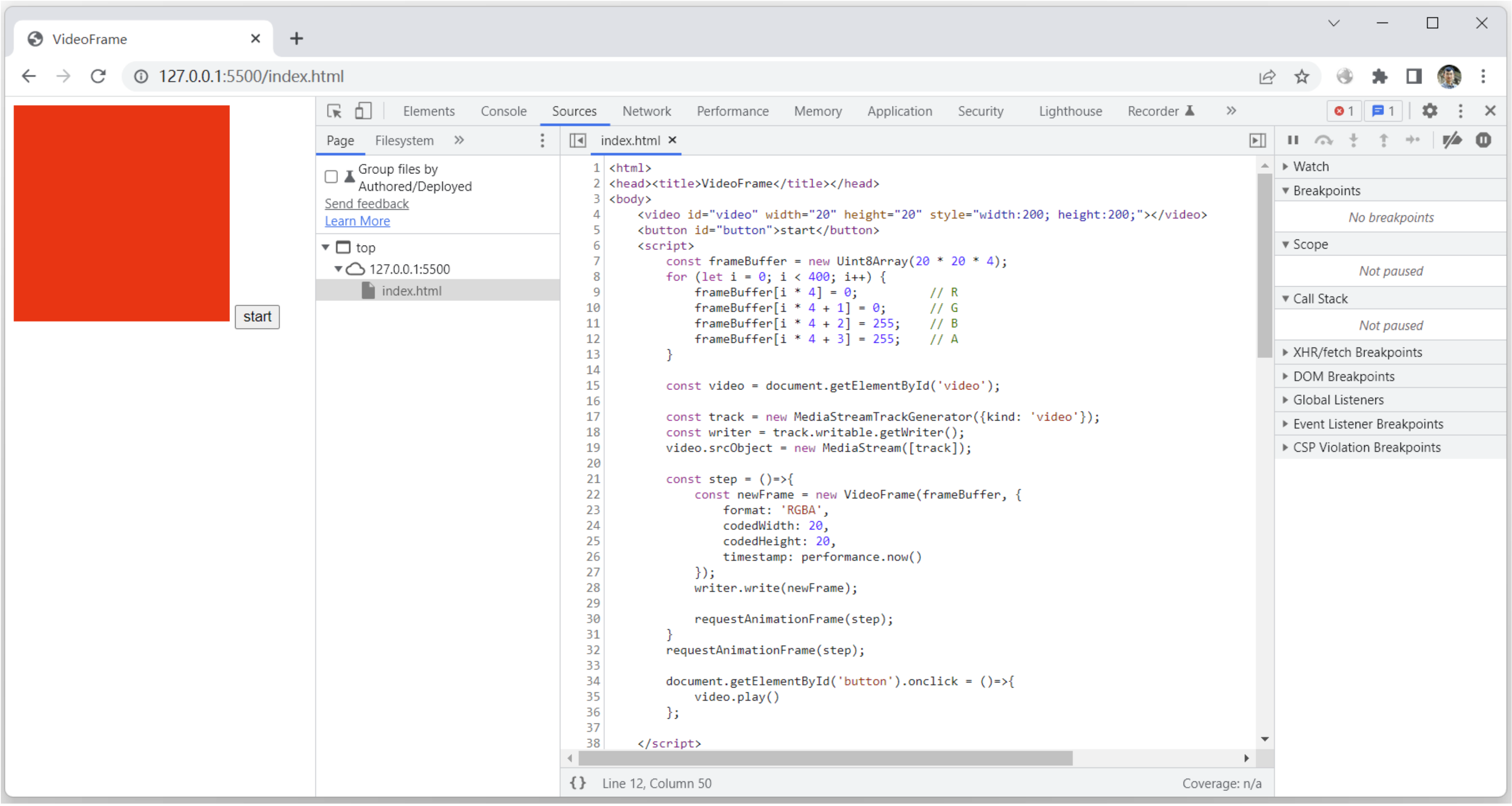
```
typedef (HTMLOrSVGImageElement or
         HTMLVideoElement or
         HTMLCanvasElement or
         ImageBitmap or
         OffscreenCanvas or
         VideoFrame) CanvasImageSource;
```

Fail Case

```
const image = new Image();
image.src = './48bit.png';
image.onload = () => {
  // Uncaught (in promise) DOMException: Failed to construct
  // 'VideoFrame': Failed to create video frame
  const videoFrame = new VideoFrame(image, 0);
  videoFrame.copyTo(destination);
}
```

WebCodecs Issues

- Incorrect color space in VideoFrame local renderer

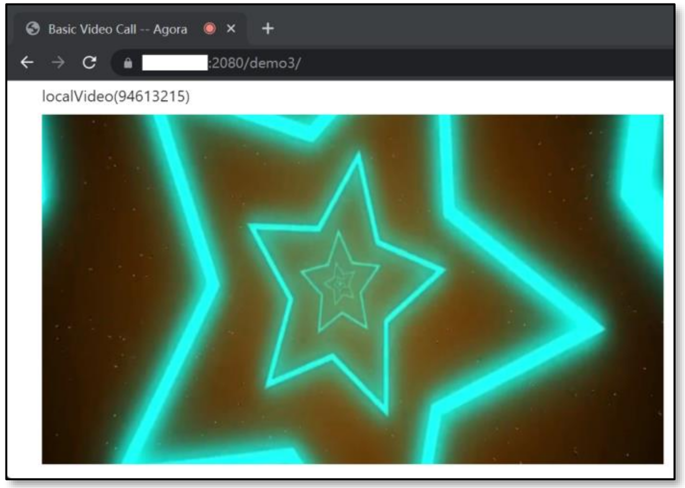


WebCodecs Issues

- Incorrect color space in VideoFrame local renderer

```
let frameBuffer = new Uint8Array(width * height * 4);

let transformer = new TransformStream({
  async transform(videoFrame, controller) {
    gl.texImage2D(gl.TEXTURE_2D, 0, gl.RGBA, width, height,
                  0, gl.RGBA, gl.UNSIGNED_BYTE, videoFrame);
    gl.bindFramebuffer(gl.FRAMEBUFFER, null);
    gl.drawArrays(gl.TRIANGLES, 0, 6);

    gl.readPixels(0, 0, width, height, gl.RGBA, gl.UNSIGNED_BYTE, frameBuffer);

    const newFrame = new VideoFrame(frameBuffer, {
      format: 'RGBA',
      codedWidth: width,
      codedHeight: height,
      timestamp: videoFrame.timestamp
    });
    videoFrame.close();
    controller.enqueue(newFrame);
  }
})

readableStream.pipeThrough(transformer).pipeTo(trackGenerator.writableStream);

playerElem.srcObject = trackGenerator;
playerElem.player();
```

- Incorrect color space in local player

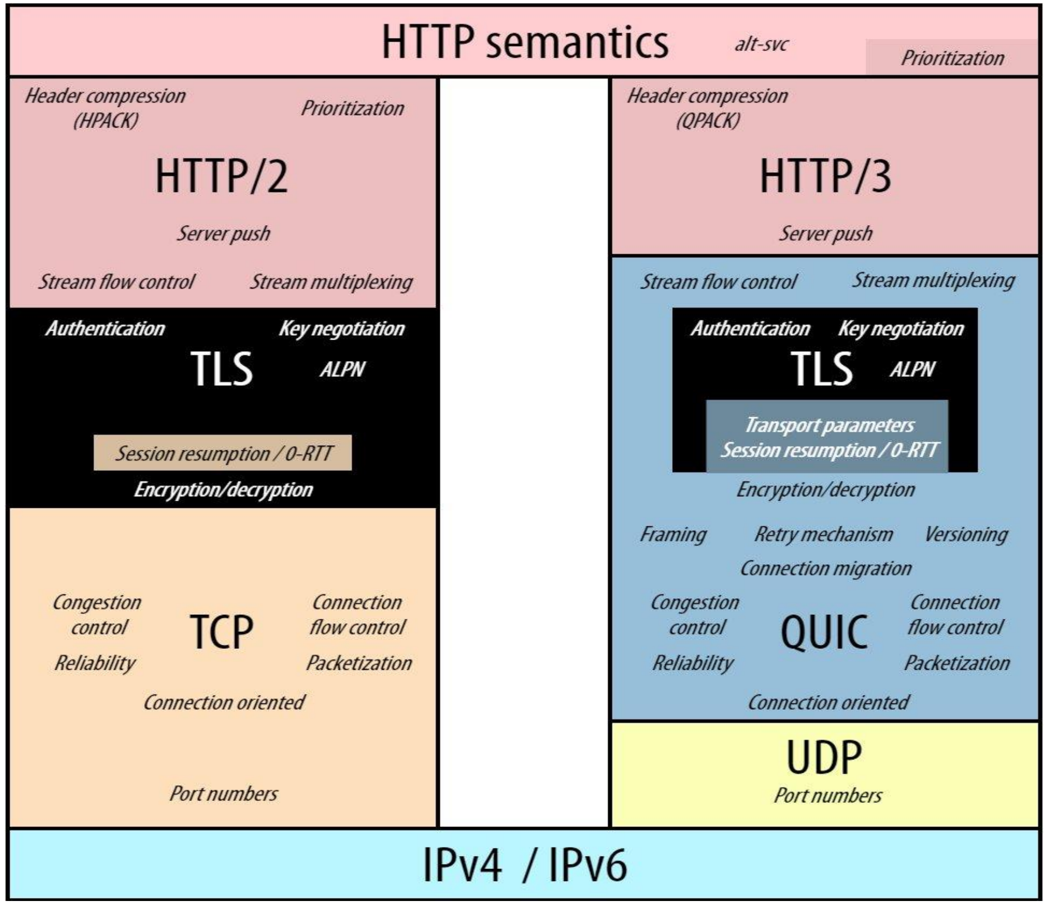


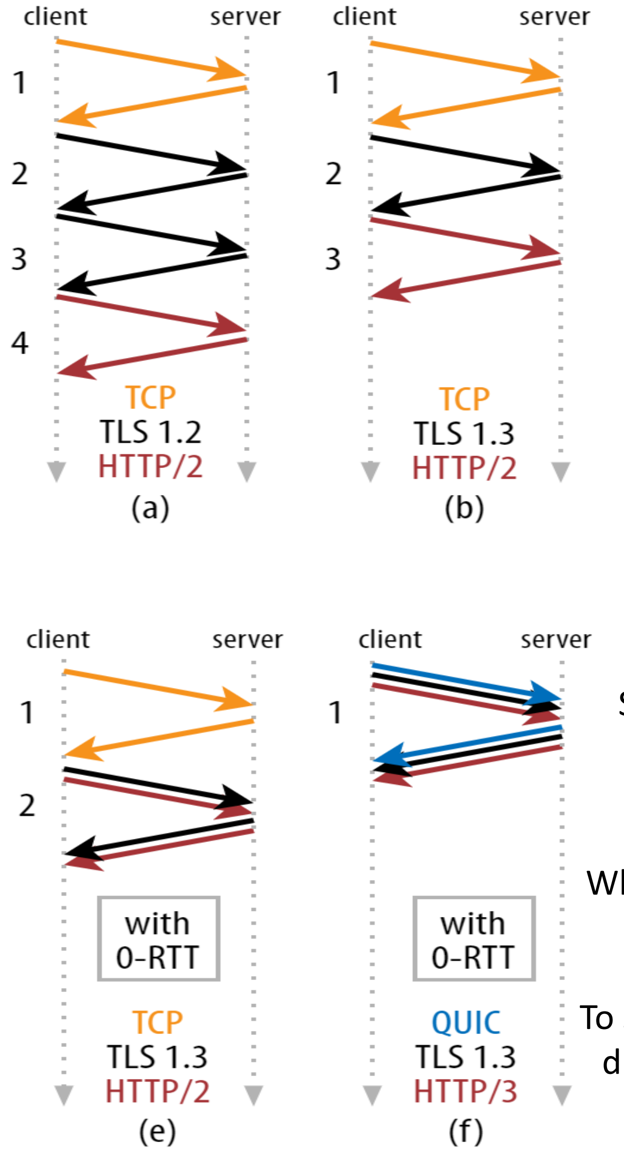
- Correct color space in receiver side player

声网 agora

WebTransport

WebTransport over HTTP/3 support for multiple streams, unidirectional streams, out-of-order delivery, and reliable as well as unreliable transport.





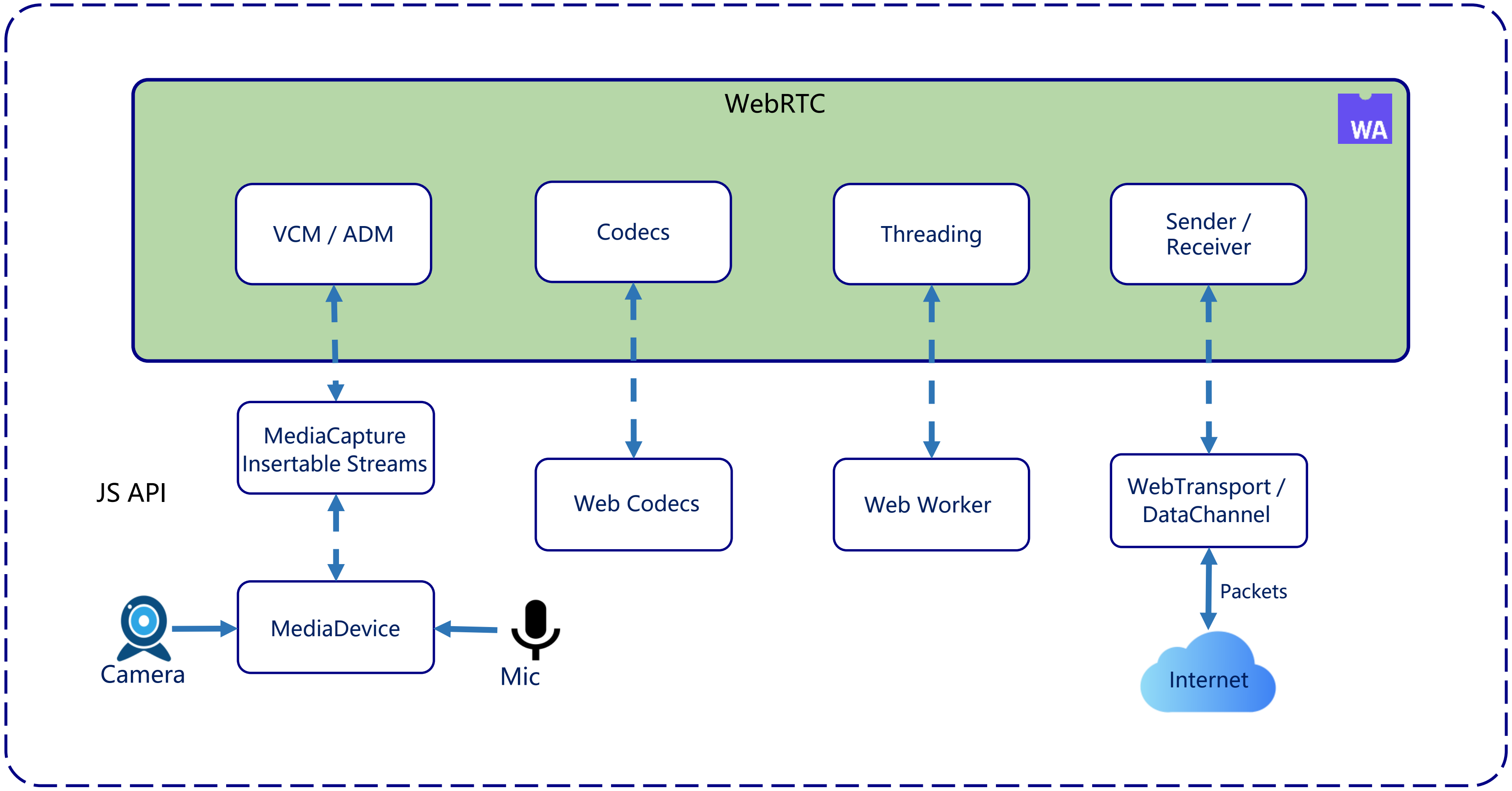
Session resumption and 0-RTT data are TLS features

Which can be used with Both TCP and QUIC

To send an HTTP request during the handshake

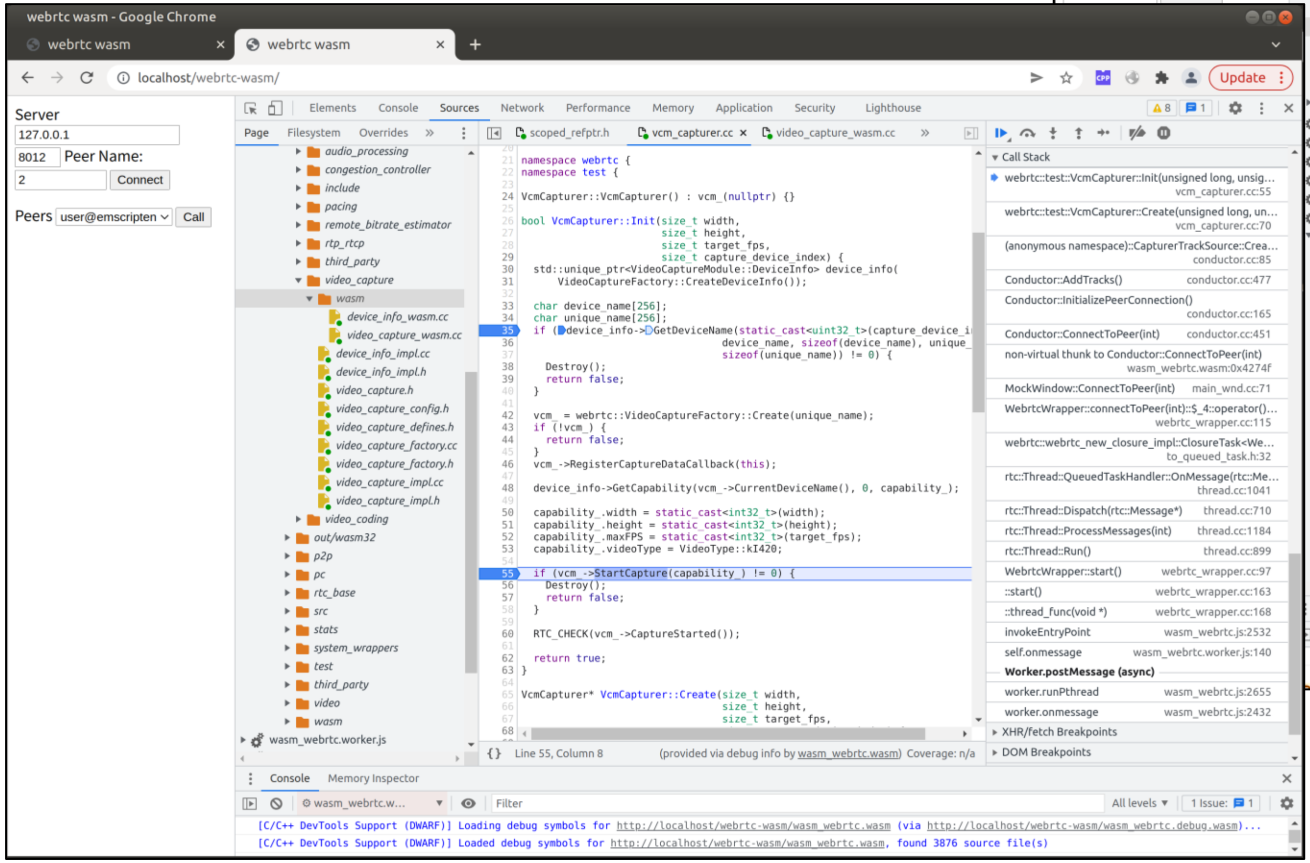
WebRTC-wasm

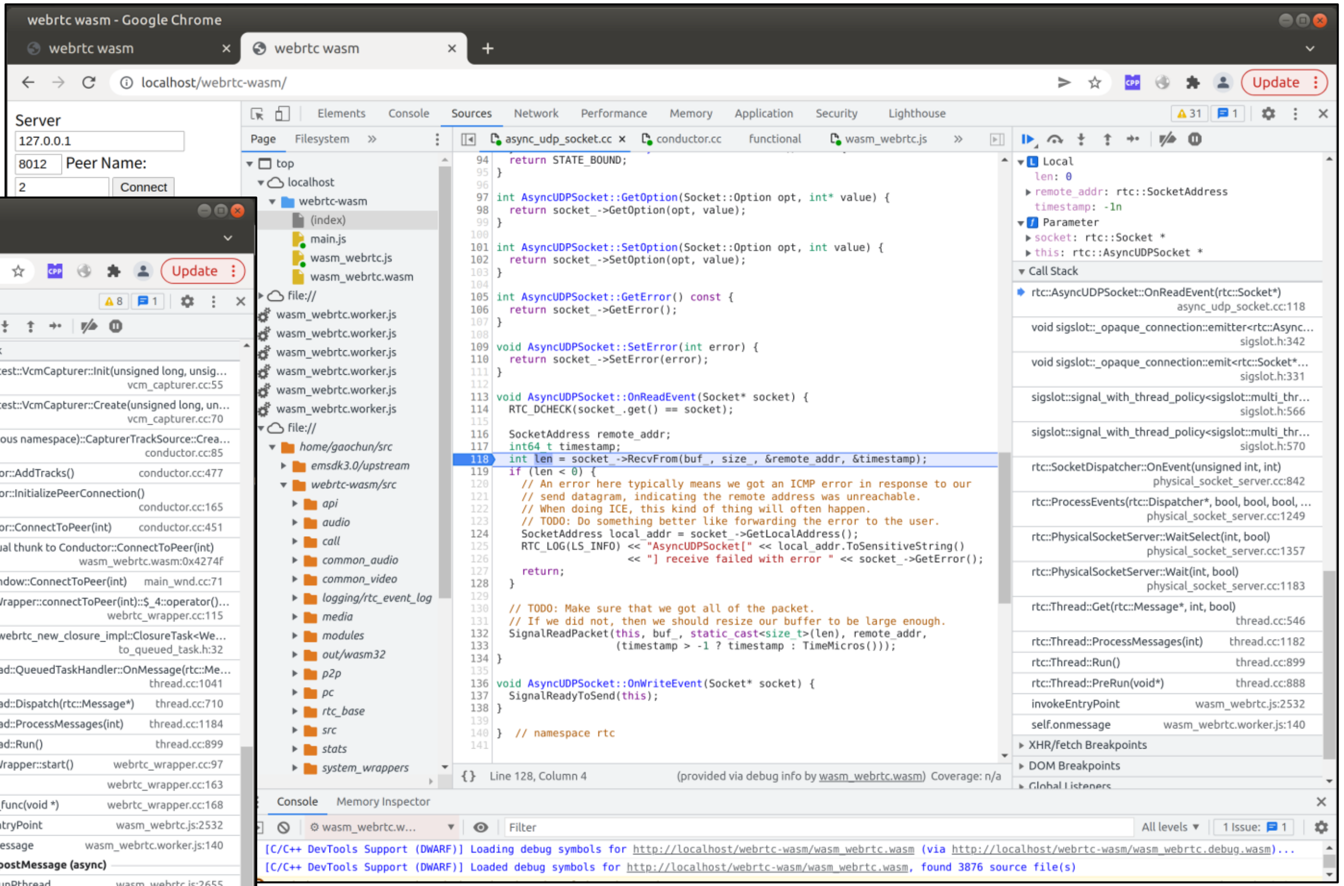
- Port WebRTC to WebAssembly



WebRTC-wasm

- VCM initialization





- Data receiving in network thread

Conclusion

- WebRTC is an excellent tool to build effective RTC apps on web.
- WebRTC can not meet all needs of build leading edge apps.
- WebRTC NV introduces more powerful standalone APIs and covers more RTC/media processing use cases.
- Agora has delivered many interesting features (AI noise cancelation, Virtual Background, Spatial Audio, Beauty effect, etc.) with WebRTC NV API, some of them has been integrated to "Agora Video Call".
- WebRTC-wasm is being developed by Agora WebRTC team, it combines the WebRTC pipeline/algorithms and WebRTC NV APIs to provide better user experience.

agora

400 632 6626

www.agora.io

THANKS