Real Time Communications Conference and Expo at Illinois Tech

IEEE International Conference

WebCodecs

and the next generation of web media APIs

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Once Upon a Time… Before the Pandemic

Streaming and realtime communications technologies evolved in ***silos***, with distinct APIs and protocol stacks.

Streaming offered ***large scale*** but ***high delay***.

Realtime communications offered ***low delay***, but ***modest scale***.

The Pandemic: a Virtual Revolution

During the pandemic, Internet communications has become essential to survival for people and industries, compressing a decade of deployment and innovation into a few months. This revolution is ***all about*** virtualization.

Next generation Web media APIs

Overcome the “tyranny of OR”. Multi-threaded applications can now deliver ***both*** low-latency ***and*** large scale, through low-level access to building blocks:

- Capture
- Encode/Decode
- Transport
- Rendering

Next generation Web media APIs

- Capture
  - Media Capture and Streams Extensions
  - Mediacapture-transform
- Encode/decode
  - WebCodecs
  - MSEv2
- Transport
  - WebTransport (HTTP/3 over QUIC)
  - WebRTC data channel in Workers (SCTP/DTLS/UDP)
- Framework
  - WHATWG Streams
  - Web Assembly

The “Pipeline” Model

- Send

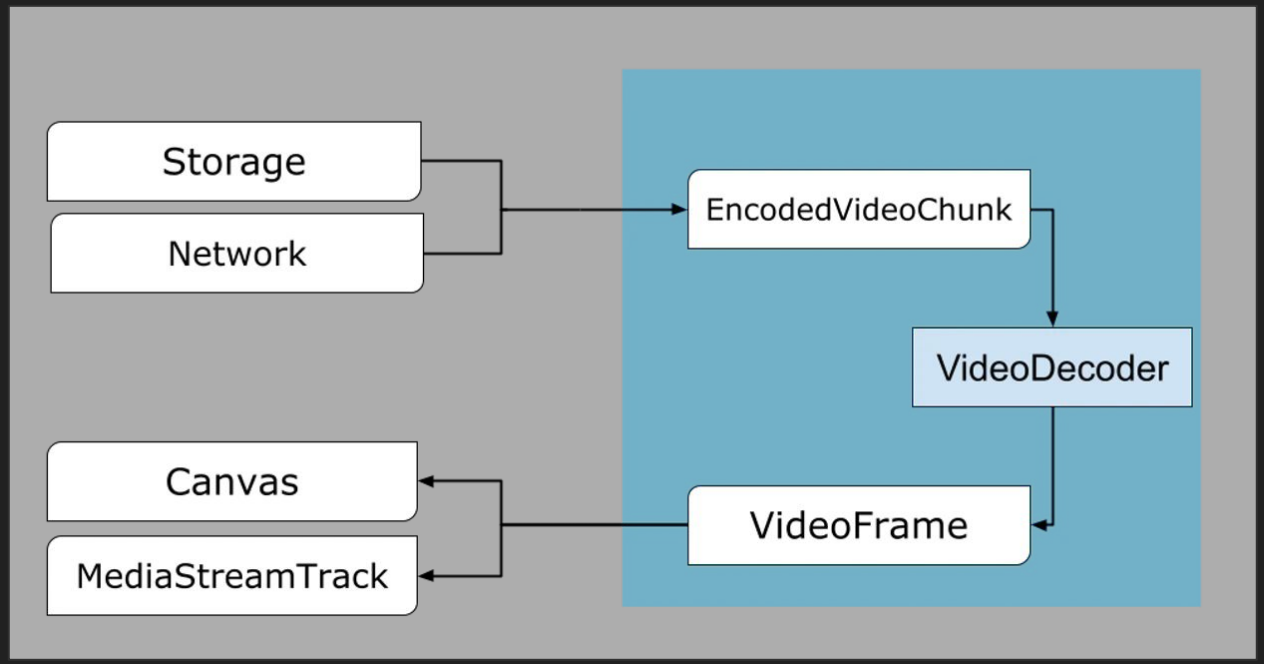
- Receive

Transport on the Web

<table>
<tr><th></th><th>Client-Server</th><th>Peer-to-peer</th></tr>
<tr><td>Reliable and ordered</td><td>WebSocket
(also WebTransport!)</td><td rowspan="3">RTCDataChannel
(SCTP/DTLS/UDP)</td></tr>
<tr><td>Reliable but unordered</td><td rowspan="2">WebTransport
(HTTP3/QUIC)</td></tr>
<tr><td>Unreliable and unordered</td></tr>
</table>

WebCodecs: Encoders & Decoders in JS

Video decoding flow



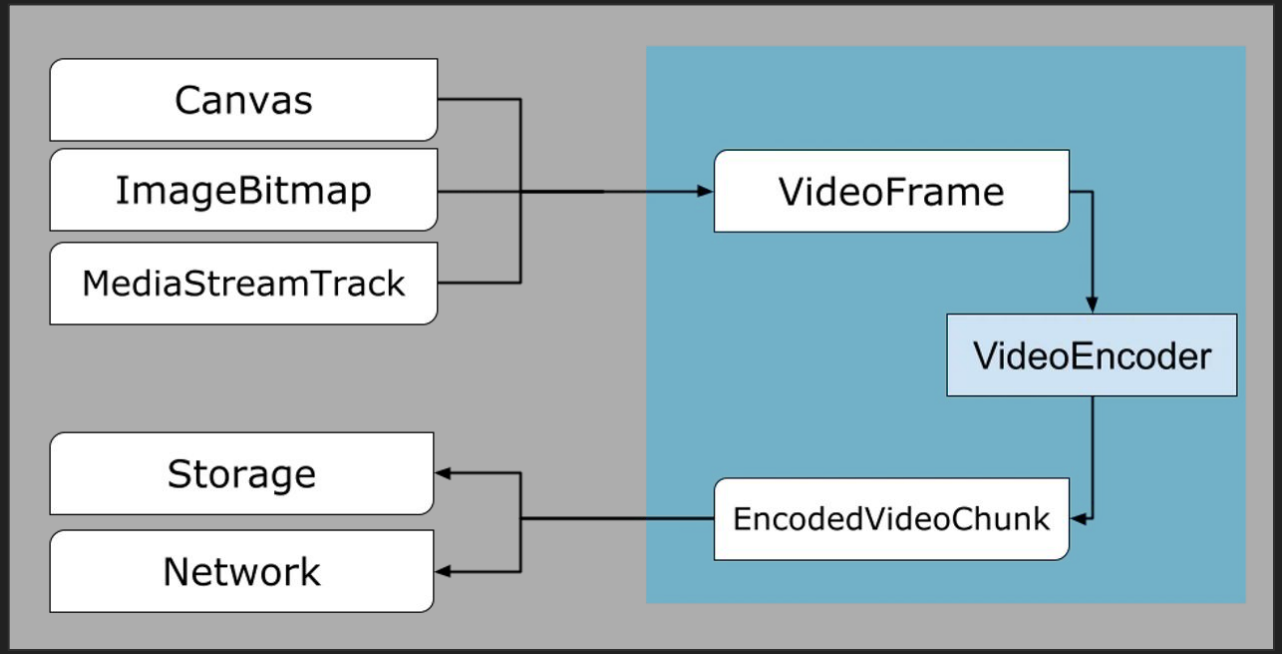
Build your own video renderer

https://chcunningham.github.io/wc-talk/rapid_video_painter.html

`AudioDecoder` looks very similar.

```
[Exposed=(Window,DedicatedWorker), SecureContext]
interface AudioDecoder {
  constructor(AudioDecoderInit init);

  readonly attribute CodecState state;
  readonly attribute long decodeQueueSize;

  undefined configure(AudioDecoderConfig config);
  undefined decode(EncodedAudioChunk chunk);
  Promise<undefined> flush();
  undefined reset();
  undefined close();

  static Promise<AudioDecoderSupport> isConfigSupported(AudioDecoderConfig config);
};

dictionary AudioDecoderInit {
  required AudioDataOutputCallback output;
  required WebCodecsErrorCallback error;
};

callback AudioDataOutputCallback = undefined(AudioData output);
```

Using Encoders looks very similar (just reversed).



Notes + FAQs

Close() your `VideoFrame` and `AudioData` objects ASAP!

Mandatory memory management

Use Workers

Keep your UI responsive and your codec well fed.

How to transfer streams to a worker? It looks like this…

```
streamWorker.postMessage(
  {
    type: "stream",
    config: config,
    url: url,
    streams: {input:  inputStream,
              output: outputStream
             }
  },
  [inputStream, outputStream]
);
```

What codecs are supported?

For Chrome:

- `VideoDecoder`: AV1, AVC (H.264), VP8, and VP9.
- `AudioDecoder`: AAC, FLAC, MP3, Opus, Vorbis, μ-law and A-law PCM formats.
- `VideoEncoder`: H264, VP8, VP9 (AV1 coming in Q4'21)
- `AudioEncoder`: Opus (AAC tentatively coming Q4'21)

Use `isConfigSupported()`

Details of support can be platform / device specific

What about WASM Codecs?

WASM codecs predate WebCodecs. The two can co-exist.

What containers are supported?

None. WebCodecs doesn't have a (de)muxing API.

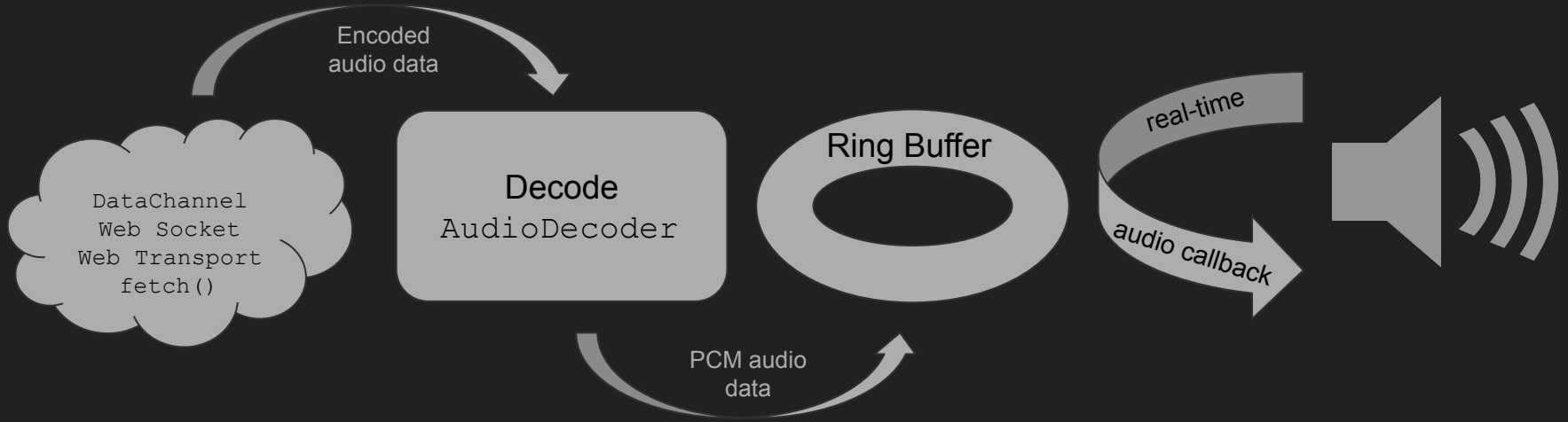
Build your own audio renderer

https://github.com/chcunningham/wc-talk/blob/main/simple_video_player.html

Build your own audio renderer

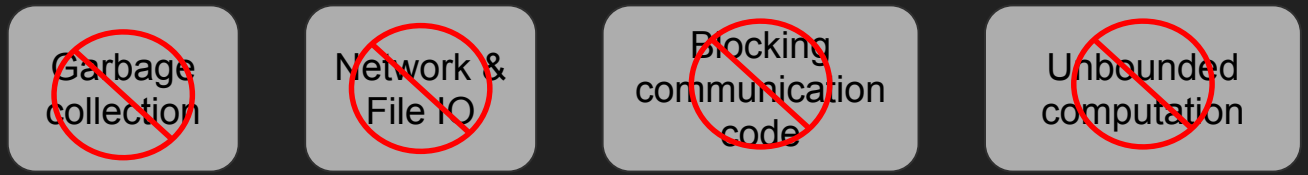
Three main steps, using an `AudioWorkletNode`:

1. Acquire and decode the encoded audio packets
2. Somehow send the decoded audio to the `AudioWorkletGlobalScope`
3. Render the audio frames



The `AudioWorklet` concept in two slides

Run arbitrary script on the high-priority real-time audio thread: the script needs to be **real-time safe**: only deterministic code is allowed, very few APIs are available



Works within an `AudioContext`, all the regular `AudioNode`s are available

Separate script file, then running in a separate global (similar to a `Worker`)

An AudioWorkletProcessor

Main thread: main.js

```
var ac = new AudioContext;
ac.audioWorklet.addModule('processo
r.js')
  .then(() => {
    var node =
      new AudioWorkletNode(ac,
'processor');
    node.port.onmessage = (m) => {
      console.log("received: " +
m);
    }
    node.port.postMessage("ping");
    node.connect(ac.destination);
  }
);
```

Real-time thread: processor.js

```
registerProcessor('processor',
  class Processor extends
AudioWorkletProcessor {
    constructor() {
      super();
      this.port.postMessage("ping");
      this.port.onmessage = () => {
        this.port.postMessage("pong");
      };
    }

    process(input, output, parameters) {
      /* White noise generator */
      for (let i = 0; i < 128; i++) {
        output[0][0][i] = Math.random()*2-1;
      }
      return true;
    }
  }
);
```

Real-time communication without generating garbage

`SharedArrayBuffer` is required, and allows writing lock-free data structures, such as an SPSC ring buffer, which is a data structure with just a few operations:

- `push(array)` enqueues PCM audio from a regular buffer into the ring buffer
- `pop(array)` dequeue PCM audio from the ring buffer into a regular buffer
- `available_write()` returns the numbers of free slots in the ring buffer
- `available_read()` return the number of elements readable from the buffer

Outline of an audio renderer: important concepts

1. The clock, for accurate audio/video synchronization
2. The pre-buffering, for a quick start
3. General operation: refilling the buffer during playback

Gettings accurate time for A/V synchronization

```
getMediaTime() {
    // Latency inherent to the OS and device in use
    let totalLatency = audioContext.outputLatency;
    // Latency of the Web Audio API implementation
    totalLatency += audioContext.baseLatency;
    return Math.max(audioContext.currentTime - totalLatency, 0.0);
 }
```

- Audio has the most robust clock: clocking the video to the audio clock is a good idea
- Range: As low as 5ms (macOS wired) up to 250-300ms (bluetooth device)
- https://blog.paul.cx/post/audio-video-synchronization-with-the-web-audio-api/

Planar vs. Interleaved, and playing the audio out

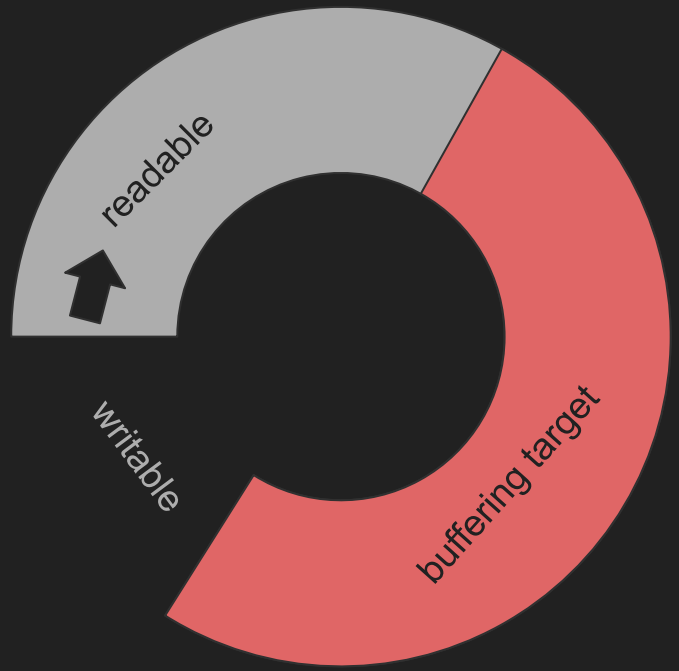
Planar audio, convenient for signal processing

Interleaved audio, convenient for IO

Filling the ring buffer

Goal: always have a few tens to a few hundreds of milliseconds of audio buffered.

While the media is not paused, periodically, check the status of the queue: if below the buffering target, kick off a few decoding operations to fill the ring buffer.



Final audio output

When the AudioWorkletProcessor's `process()` method is called :

- Dequeue just enough frames from the ring buffer (or else, an underrun happened, tune the ring buffer size and buffering target) to fulfill the request (for now 128 * number of channels)
- Deinterleave into the second parameter's planar buffers

The "Pipeline" Model

https://webrtc.internaut.com/wc/wtSender/

The “Pipeline” Model

- Send

```
inputStream
     .pipeThrough(SpecialEffects())
     .pipeThrough(EncodeVideoStream(config))
     .pipeThrough(Serialize())
     .pipeTo(Transport())
```

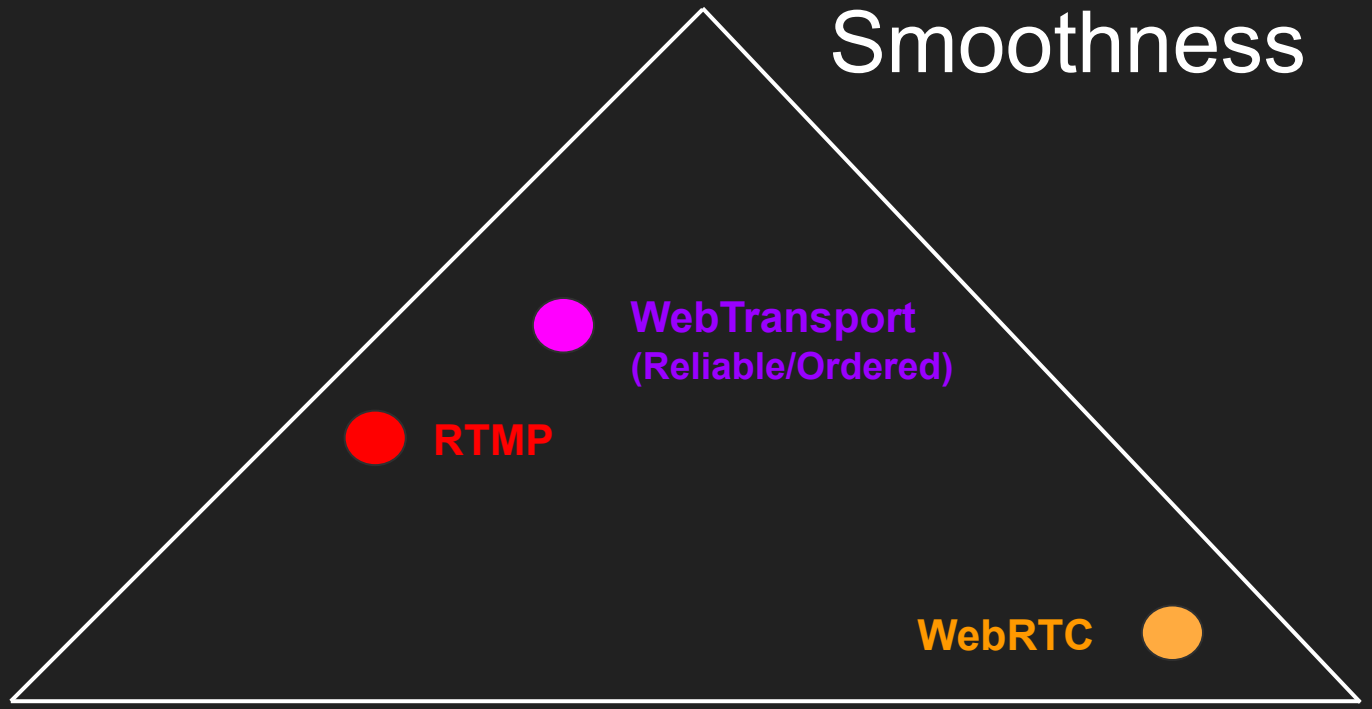
- Receive

```
receiveStream
     .pipeThrough(Deserialize())
     .pipeThrough(DecodeVideoStream())
     .pipeTo(outputStream)
```

inputStream & outputStream (getUserMedia and MediaStreamTrackProcessor/Generator)

```
async function getMedia(constraints) {
    // Get a MediaStream from the webcam.
    const mediaStream = await navigator.mediaDevices.getUserMedia(constraints);

    // Connect the webcam stream to the video element.
    document.getElementById('inputVideo').srcObject = mediaStream;

    // Create a MediaStreamTrackProcessor, which exposes a ReadableStream of VideoFrames.
    let [track] = mediaStream.getVideoTracks();
    let ts = track.getSettings();
    const processor = new MediaStreamTrackProcessor(track);
    inputStream = processor.readable;

    // Create a MediaStreamTrackGenerator, which generates a track from a
    // WritableStream of VideoFrames.
    const generator = new MediaStreamTrackGenerator({kind: 'video'});
    outputStream = generator.writable;
    document.getElementById('outputVideo').srcObject = new MediaStream([generator]);


...
}
```

Transport Tradeoffs



WebCodecs Decode as a Transform Stream

```
DecodeVideoStream() {
  return new TransformStream({
    start(controller) {
      this.decoder = decoder = new VideoDecoder({
        output: frame => controller.enqueue(frame),
        error: (e) => {
         self.postMessage({severity: 'fatal', text: `Decoder error: ${e.message}`});
        }
      });
    },
    transform(chunk, controller) {
      if (this.decoder.state != "closed") {
        this.decoder.decode(chunk);
      }
    }
  });
}
```

WebCodecs Encode as a Transform Stream - 1/3

```
EncodeVideoStream(config) {
  return new TransformStream({
    start(controller) {
      this.frame_counter = 0;
      this.pending_outputs = 0;
      this.encoder = encoder = new VideoEncoder({
        output: (chunk, cfg) => {
          if (cfg.decoderConfig) {
            self.postMessage({text: 'Decoder reconfig!'});
            decoder.configure(cfg.decoderConfig);
          }
          chunk.temporalLayerId = 0;
          if (cfg.temporalLayerId) {
            chunk.temporalLayerId = cfg.temporalLayerId;
          }
```

WebCodecs Encode as a Transform Stream - 2/3

```
    this.pending_outputs--;
        controller.enqueue(chunk);
      },
      error: (e) => {
        self.postMessage({severity: 'fatal', text: `Encoder error: ${e.message}`});
      }
    });
  VideoEncoder.isConfigSupported(config).then((encoderSupport) => {
      if(encoderSupport.supported) {
        this.encoder.configure(encoderSupport.config);
      } else {
        self.postMessage({severity: 'fatal', text: 'Config not supported:\n' + JSON.stringify(encoderSupport.config)});
        this.stopped = true;
      }
    })
    .catch((e) => {
        self.postMessage({severity: 'fatal', text: `Encoder error: ${e.message}`});
    })
  },
```

WebCodecs Encode as a Transform Stream - 3/3

```
transform(frame, controller) {
   if (this.pending_outputs <= 30) {
     if (++this.frame_counter % 20 == 0) {
       self.postMessage({text: 'Encoded 20 frames'});
     }
     this.pending_outputs++;
    const insert_keyframe = (this.frame_counter % config.keyInterval) == 0;
     try {
       if (this.encoder.state != "closed") {
        this.encoder.encode(frame, { keyFrame: insert_keyframe });
       }
     } catch(e) {
      self.postMessage({severity: 'fatal', text: `Encoder error: ${e.message}`});
     }
   }
   frame.close();
      }
    });
```

Questions?