audio latency in browser-based DAWs

w3c media production workshop, 2021

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Online collaborative DAW

Multi-track projects

Audio/Midi recording

Software instruments

Effects

...

WebAudio

WebMIDI

MediaRecorder

MediaStream

WebAssembly

WebRTC

...

Problem 1: Low roundtrip latency for ‘monitoring’

Ex. 1: guitarist using the DAW as effect pedal / amp simulation.

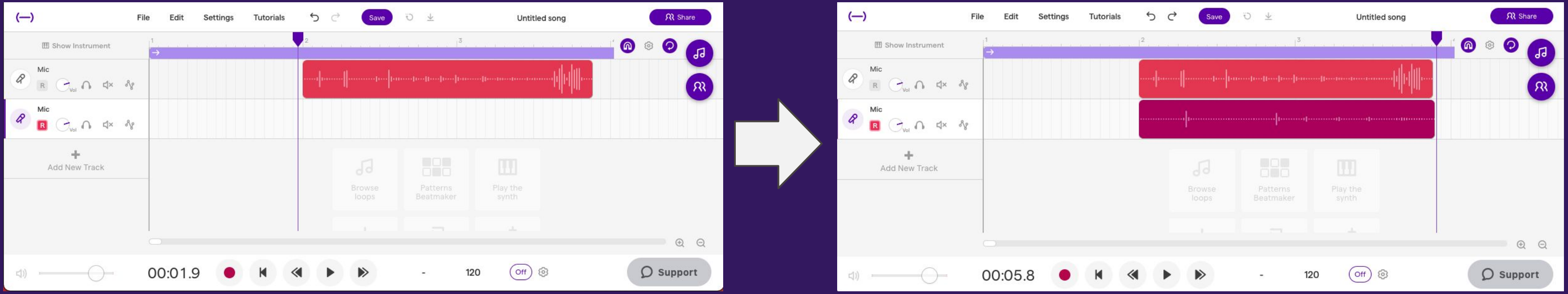
Ex. 2: keyboardist using the DAW as an instrument (feeding it MIDI notes).

Problem 1: Low latency for ‘monitoring’

Best case currently is <30ms, passable but not great.

Roundtrip latency <10ms is desirable to be competitive.

Problem 2: Recording latency compensation



The user will sing/play along to what they can hear.

Subproblems to solve:

a) Roundtrip latency information at the time of recording

b) Audio data ‘arrival time’ for buffers stored

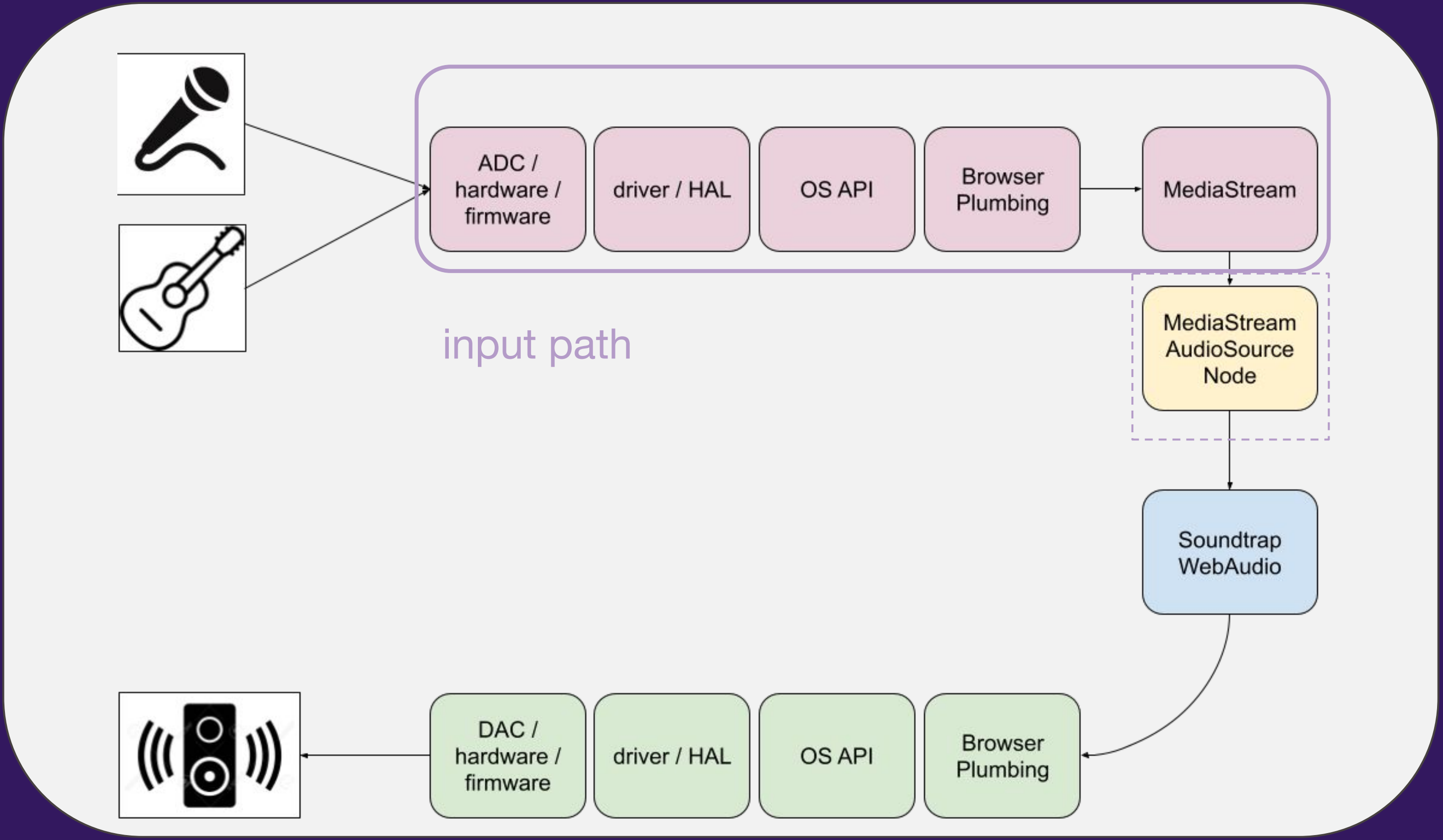
Problem 2a: Full round trip latency

round trip = input + processing + output

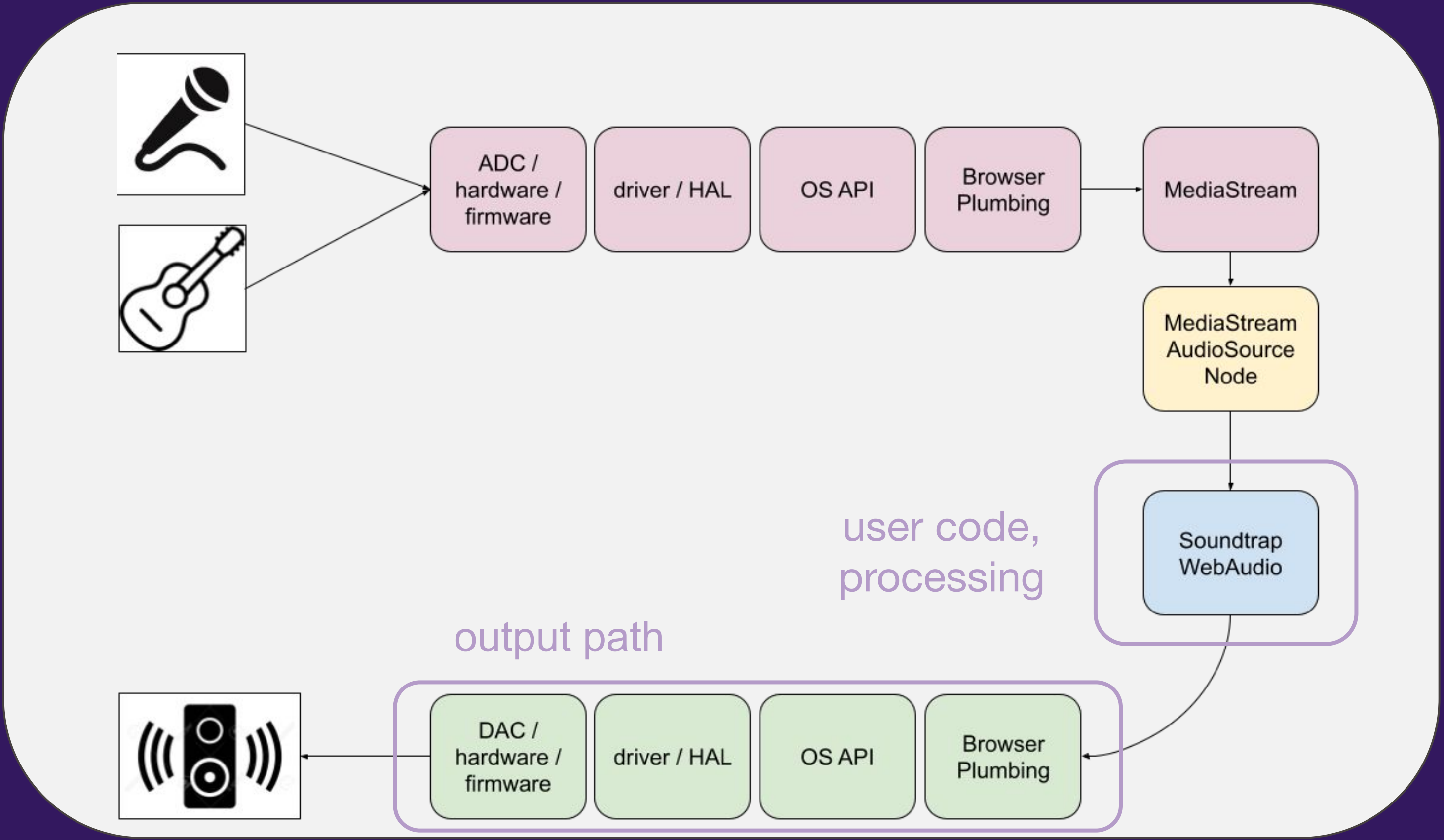
[latency]

wrong info -> misaligned playback

Problem 2a: Full round trip latency



Problem 2a: Full round trip latency



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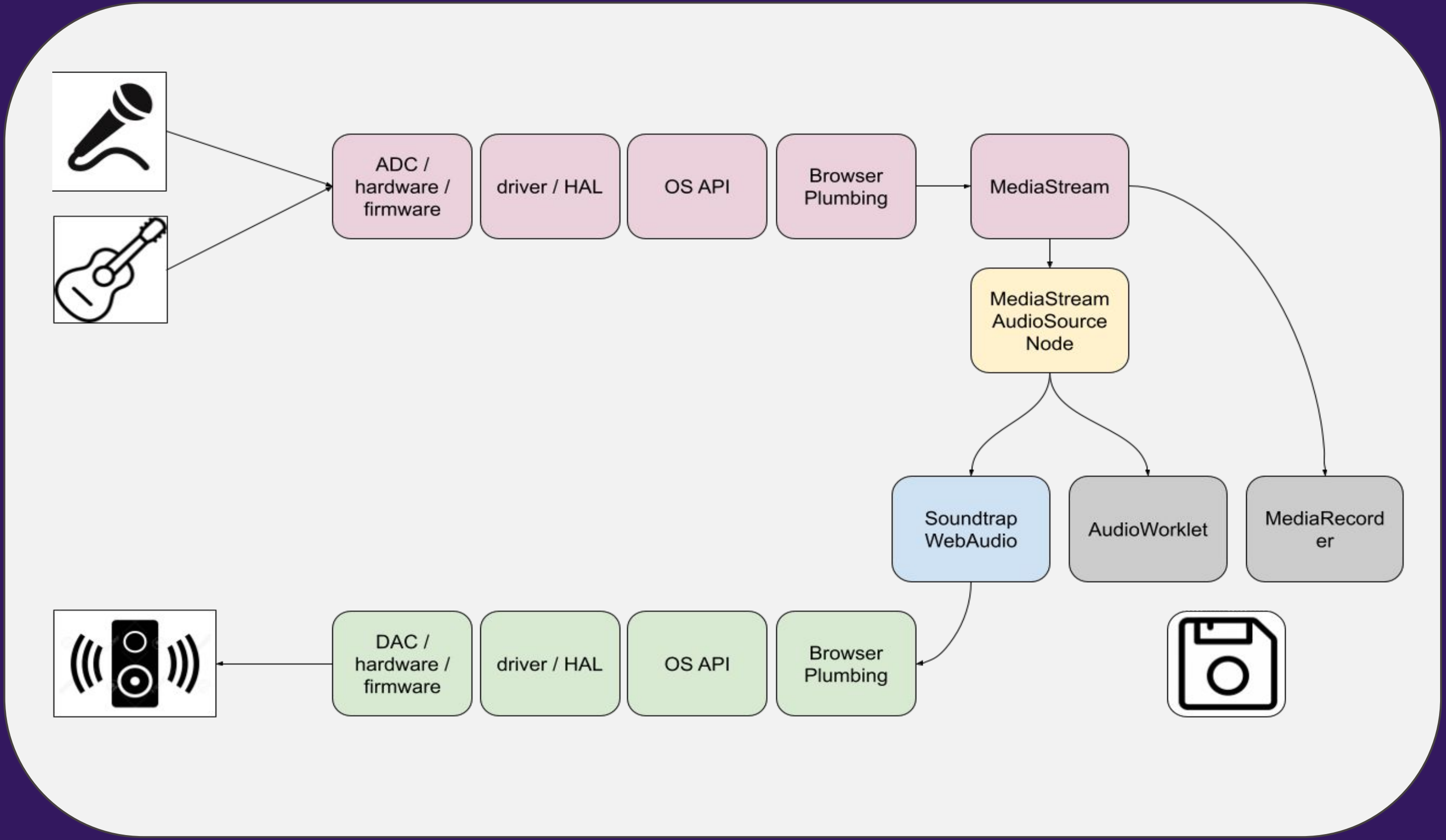
Problem 2b: Audio data 'arrival time'



Specced but not implemented across all browsers:

- Input latency info (MediaStreamTrackSetting)
- Output latency info (WebAudio)

Possible spec gaps:

- Specs on input and output latency - full paths?
- MediaStreamSourceNode - adds latency?
- MediaRecorder - how to reliably sync to audio clock?
- WebCodecs benefits from a packetization/container counterpart.

Encouragement to implementers:

- minimize both input and output latencies
- no hidden latencies

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