WebTransport + WebCodecs

at W3C Games Workshop 6/19

Problem 1: WebRTC not great for cloud gaming

Problem 1: WebRTC not great for cloud gaming

Problem 2: MSE not great for cloud gaming

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Problem 3: WebSocket not great for gaming

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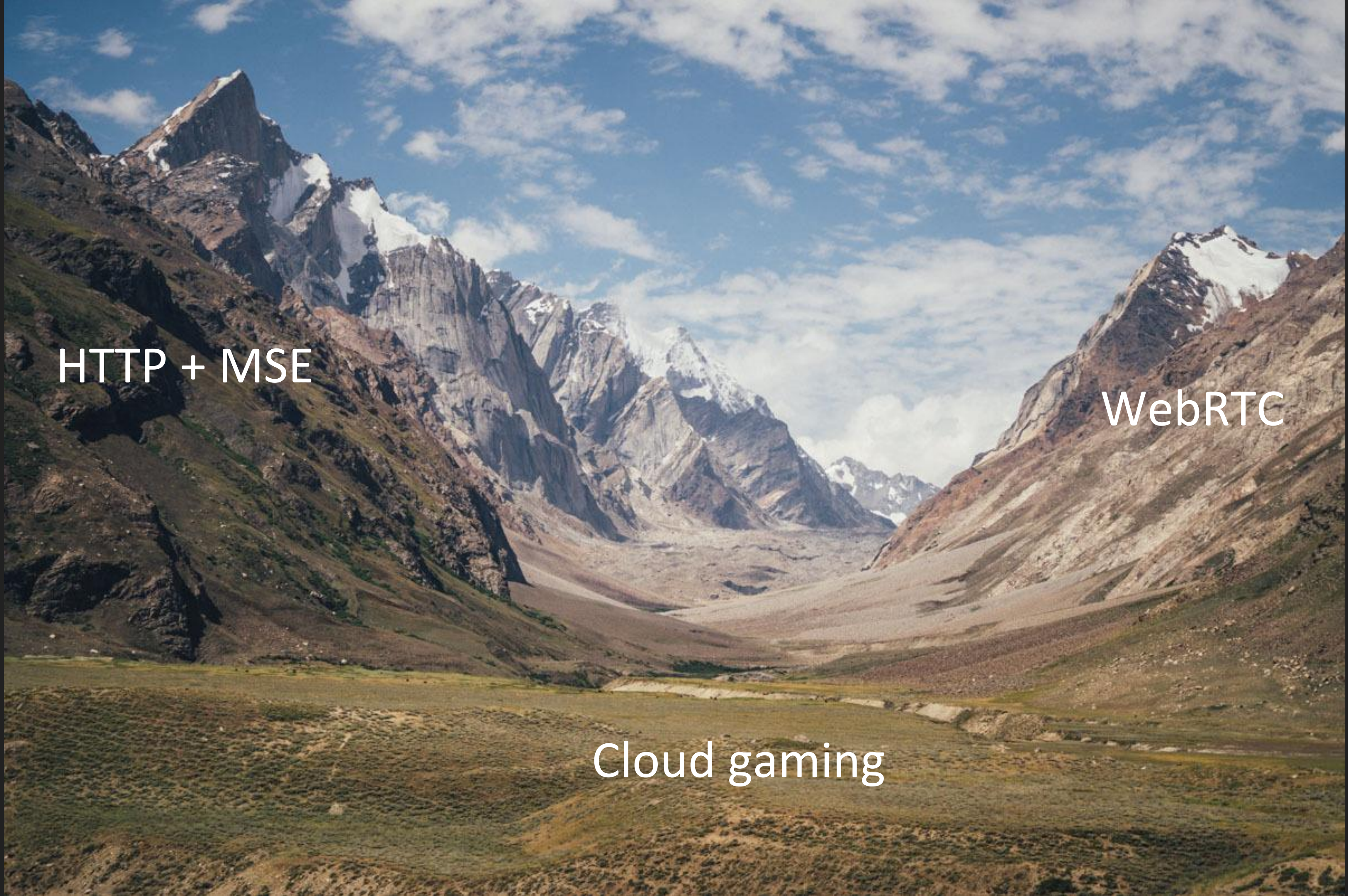
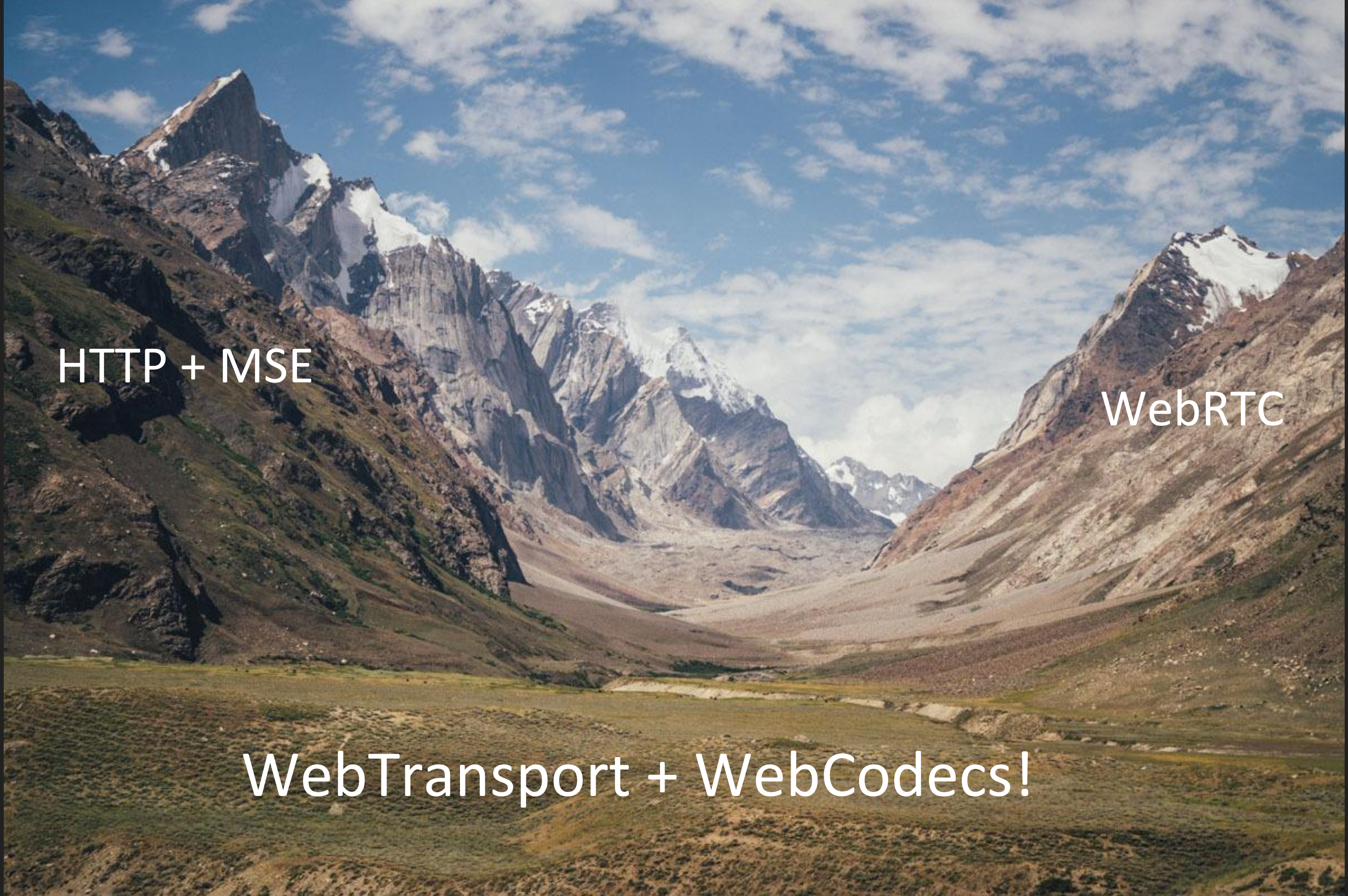
Problem 2: MSE not great for cloud gaming

Problem 3: WebSocket not great for gaming

Solution: WebTransport + WebCodecs


HTTP + MSE
WebRTC
Cloud gaming


HTTP + MSE
WebRTC
WebTransport + WebCodecs!

▲ WebRTC: the future of web games (getkey.eu)

293 points by getkey on Dec 27, 2016 | hide | past | web | favorite | 163 comments

▲ Matheus28 on Dec 27, 2016 [-]

I'm the guy that made Agar.io, Diep.io and a few smaller games. I analyzed the possibility of using WebRTC in my games several times so far, but it seems that right now, it's still hard to use in a server-client architecture. You need to bring this [1] behemoth and all of its dependencies to your project dependencies on the server side, even though you only care about a tiny bit of it (unreliable data channels). It's unlikely that people will start using it until there is an easy stripped-down version that only deals with data channels.

Why can't I send UDP packets from a browser?

A solution for enabling UDP in the web

Posted by Glenn Fiedler on Sunday, February 26, 2017

Premise

In 2017 the most popular web games like agar.io are networked via WebSockets over TCP. If a UDP equivalent of WebSockets could be incorporated into browsers, it would greatly improve the networking of these games.

[Proposal] hint attribute on HTMLMediaElement to configure rendering latency

Media and Real Time Communications

chcunningham 2 May 17

All user agents delay the start of playback until some minimum number of frames are decoded and ready for rendering. This buffer provides an important cushion against playback stalls that might otherwise be caused by intermittent decoder slowness.

It also adds a small amount of latency to start playback. For example, in Chromium this adds roughly 200 milliseconds for common video frame rates.

May 17

1 / 4
May 18

Media Source Extensions: Eviction Policies Explainer

Author: Matthew Wolenetz, Google Inc. - May 17, 2019. Last update May 17, 2019.

tl;dr

We propose adding new coded frame eviction policies to `SourceBuffer` objects, enabling web applications to have greater control over how the implementation manages buffered media. Over time, further eviction policies may be

Android Developers > Docs > Reference

Contents

Summary

- Public constructors
- Protected constructors
- Public methods
- •••

DatagramSocket

```
public class DatagramSocket
extends Object implements Closeable
```

Transport

Android Developers > Docs > Reference

Contents

- Data Types
- States
- Creation
- Initialization
- •••

MediaCodec

```
public final class MediaCodec
extends Object
```

Codecs

Android Developers > Docs > Reference

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DatagramSocket

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WebTransport

WebCodecs

UDP isn't secure!

- encryption
- congestion control
- CORS/consent

If only there were a way to add that to UDP but keep all the good parts of UDP...

QUIC

HTTP/3

What is QUIC?

- Fast connection setup (1-RTT, or sometimes 0-RTT)
- Secure
- Low-latency congestion control (pluggable)
- Many reliable streams (like TCP/TLS * N)
- Unreliable/unordered datagrams (like UDP)
- Can be p2p (with ICE)
- Basis of HTTP/3
- Widely deployed
- Many implementations coming

Benefits for games

- Faster game loading
  (particularly with many components).
- More network resilience: making bad networks usable
  (particularly for mobile network with high RTT/loss)

But what about cloud gaming?

Streaming with WebRTC stack

Encode/Forward, Packetize

Depacketize, Buffer, Decode, Render

Streaming with WebRTC stack

ICE, DTLS, SRTP

Server → Browser (WebRTC stack)

Encode/Forward, Packetize

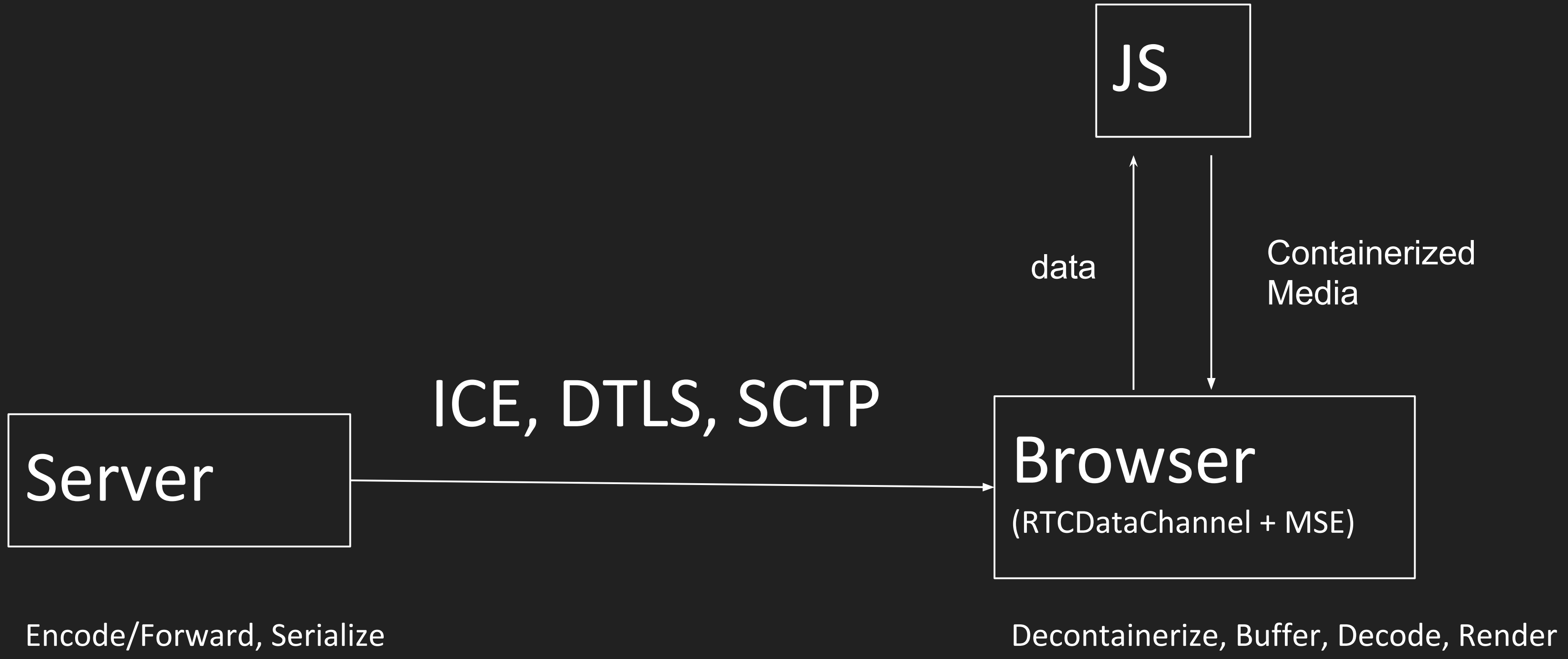
Depacketize, Buffer, Decode, Render

"Hard to use in a client-server architecture"

Not a lot of control in buffering, decoding, rendering. All controlled by browser.

Limited by RTP (no generic data)

Streaming with WebRTC Data Channel + MSE



Streaming with WebRTC Data Channel + MSE

"Hard to use in a client-server architecture"

Low-latency mode is implicit magic

Have to containerize media just to get it in

JS

data

Containerized
Media

ICE, DTLS, SCTP

Server

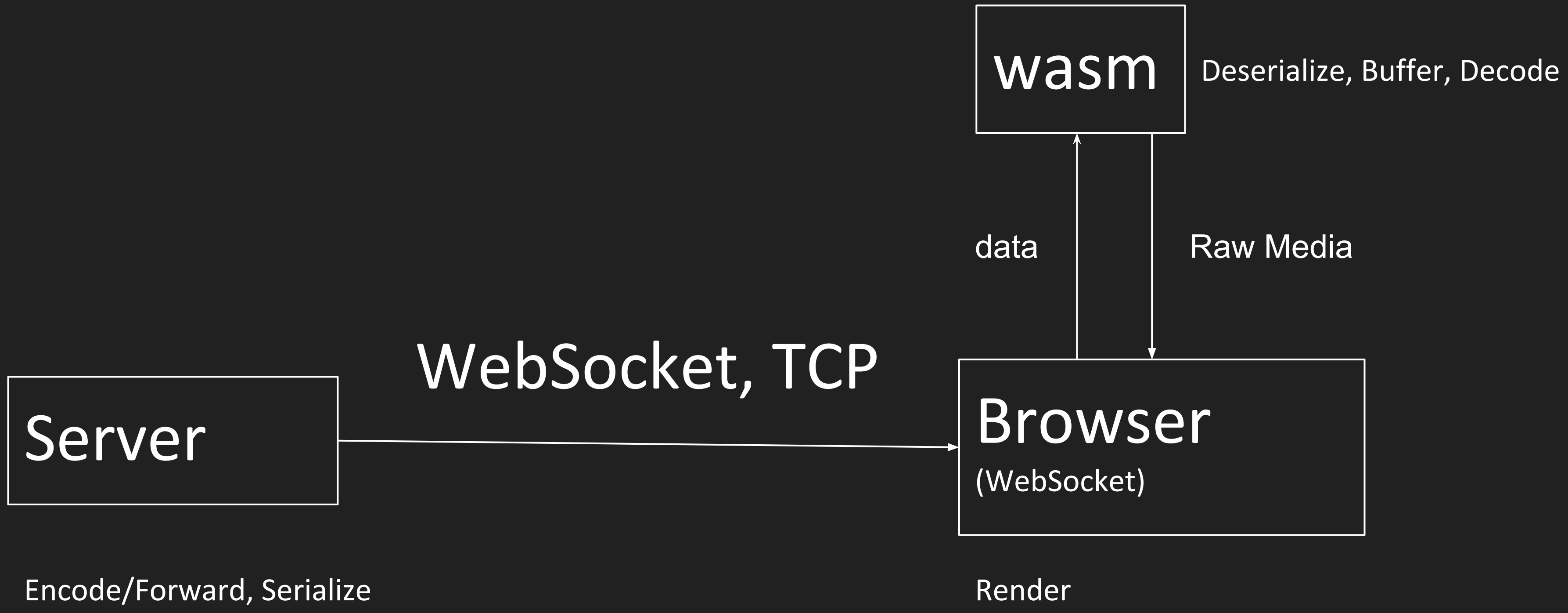
Browser

(RTCDataChannel + MSE)

Encode/Forward, Serialize

Decontainerize, Buffer, Decode, Render

Streaming with WebSocket + WebAssembly



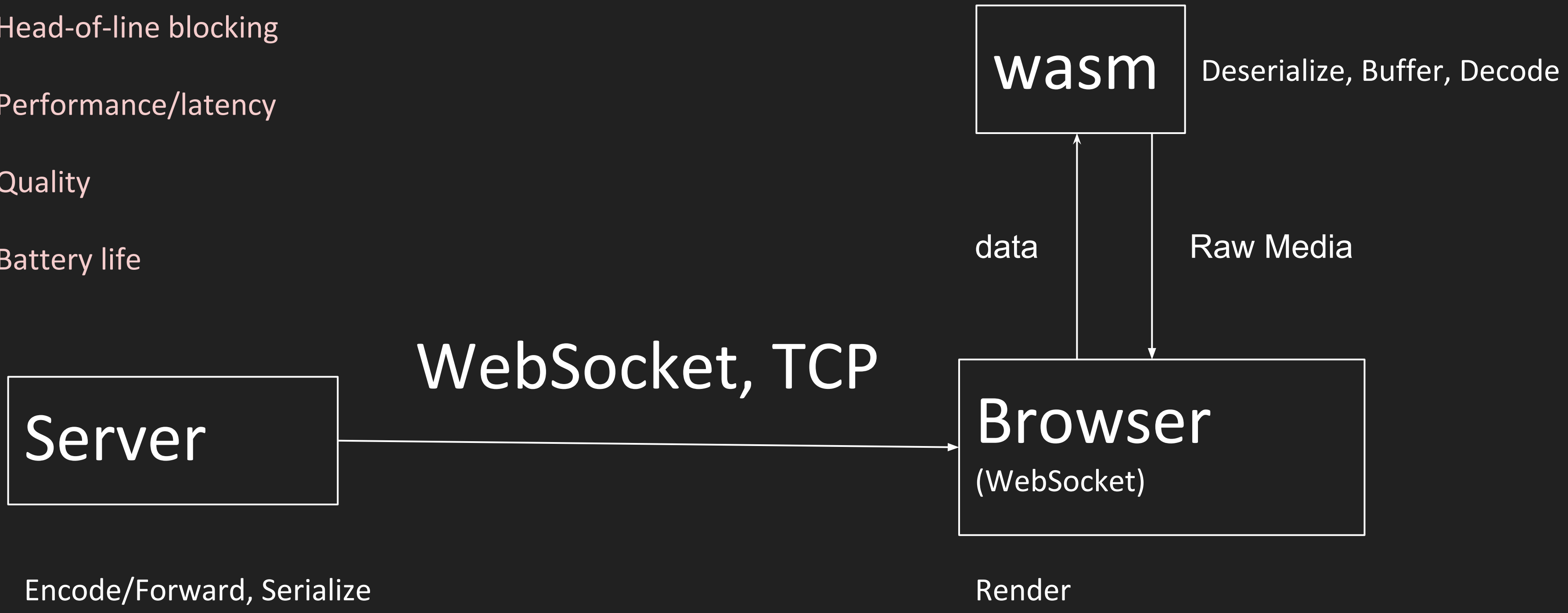
Streaming with WebSocket + WebAssembly

Head-of-line blocking

Performance/latency

Quality

Battery life



We can do better

We can do better

transport

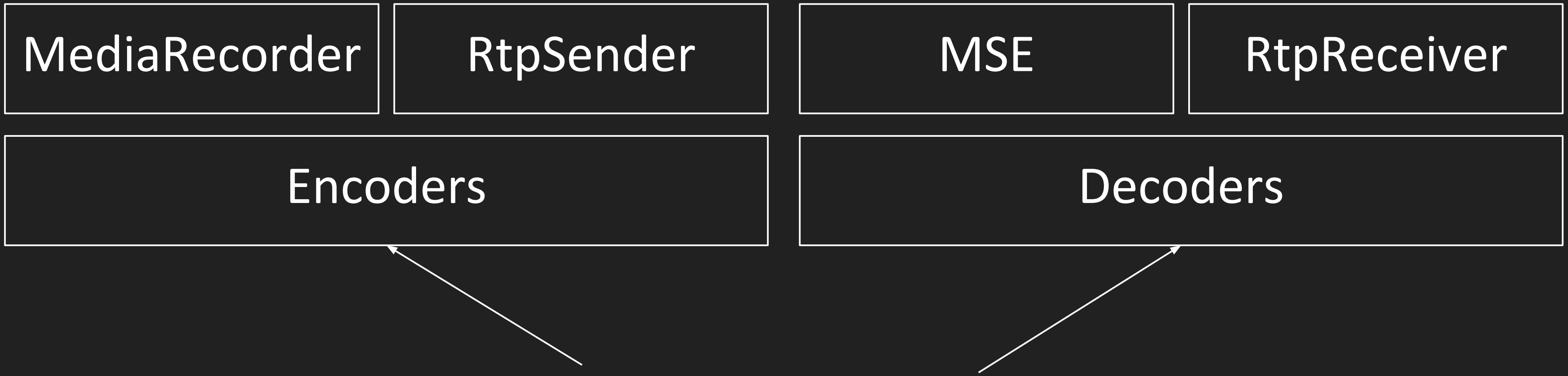
WebTransport is a better RTCDataChannel and a better WebSocket

- Reliable/ordered and unreliable/unordered
- Easy to use in a client/server architecture
- Client/server and p2p
- Provides datagram support
- Same security properties as RTCDataChannel and WebSockets (encryption, congestion control, CORS)
- Faster!
- Better API (support for back pressure)

We can do better

We can do better

codecs

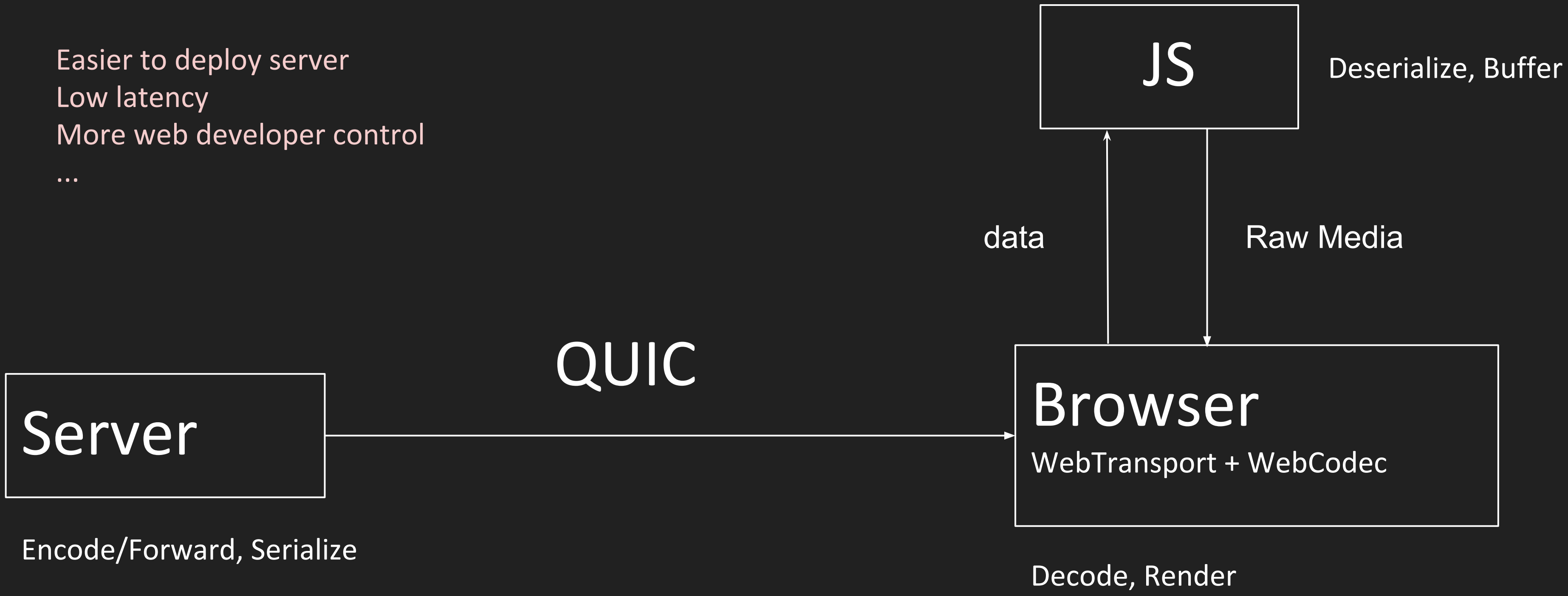


Like mobile APIs,
but hidden

"built-in encoder makes sense so we don't have to ship WebAssembly modules to do the same thing"

Streaming with WebTransport + WebCodec

Easier to deploy server
Low latency
More web developer control
...



Other use cases/benefits

- Push game state (with low latency)
- Stream game assets (with low latency)
- Communication during a game (with same APIs)
- Upload media to server for ML (with low latency)
- Transcoding (faster than real-time)
- Get new codec support faster (less for a browser to implement)
- Get new/wider container support (from JS library)
- Better support for spatial/temporal scalability
- e2e encrypted group communication

WebTransport Status:

- has origin trial (p2p version)
- looking for customers

WebCodecs Status:

- has proposal/explainer
- looking for interest

More Info

- RTCQuicTransport Origin trial
  - Announcement: https://developers.google.com/web/updates/2019/01/rtcquictransport-api
  - Documentation: https://github.com/shampson/RTCQuicTransport-Origin-Trial-Documentation
  - Sample code: https://webrtchacks.com/first-steps-with-quic-datachannel/
- WebTransport
  - Proposal/Explainer: https://discourse.wicg.io/t/webtransport-proposal/3508
  - Spec: https://wicg.github.io/web-transport/
- WebCodecs
  - Proposal/Explainer: https://discourse.wicg.io/t/webcodecs-proposal/3662
  - Repository for future spec: https://github.com/pthatcherg/web-codecs

Topics for discussion

We are *looking for customers*

- What use cases do you have for secure datagrams or low-level codecs?
- What would you like to see in these APIs?
- What do you care more about? client/server? p2p? cloud gaming? normal gaming? communication in games?
- **Use the RTCQuicTransport origin trial! (**If you run into issues with client/server, let me know... I've got a demo server almost working at https://github.com/pthatcherg/quic-go/tree/gquic/example/webtransport)

Other things I couldn't fit into the slides

- HouseParty/EPIC acquisition
- Data with audio/video at the same time
- Spatial audio
- Temporal/spatial scalability
- Study of origin trial vs SCTP:
  https://docs.google.com/document/d/1F0lfp62blYTiqDrbvcRewkKBr20a2RsnoFOwcexKUWk/edit#