

Frame/Stream Transport

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What We'll Cover

- Tools
 - Next Generation Web Media APIs
 - WebCodecs
 - WebTransport
- Frame/Stream Transport
 - Observations
 - Concurrency
 - Partial reliability

Next Generation Web Media APIs

- Capture
 - [Media Capture and Streams Extensions](#)
 - [Mediacapture-transform](#)
- Discovery
 - [Media Capabilities](#)
- Encode/decode
 - [WebCodecs](#) (for raw media)
 - [MSEv2](#) (for containerized media)
 - [WebRTC-SVC](#) (scalable video coding support, shipping in M111)
- Transport
 - [WebTransport](#)
 - [WebRTC data channel in Workers](#)
- Performance
 - [Request VideoFrame Callback](#)
- Framework
 - [WHATWG Streams](#)
 - [Web Assembly](#)

The “Pipeline” Model (WHATWG Streams)

- **Send**



- **Receive**



The “Pipeline” Model (in code)



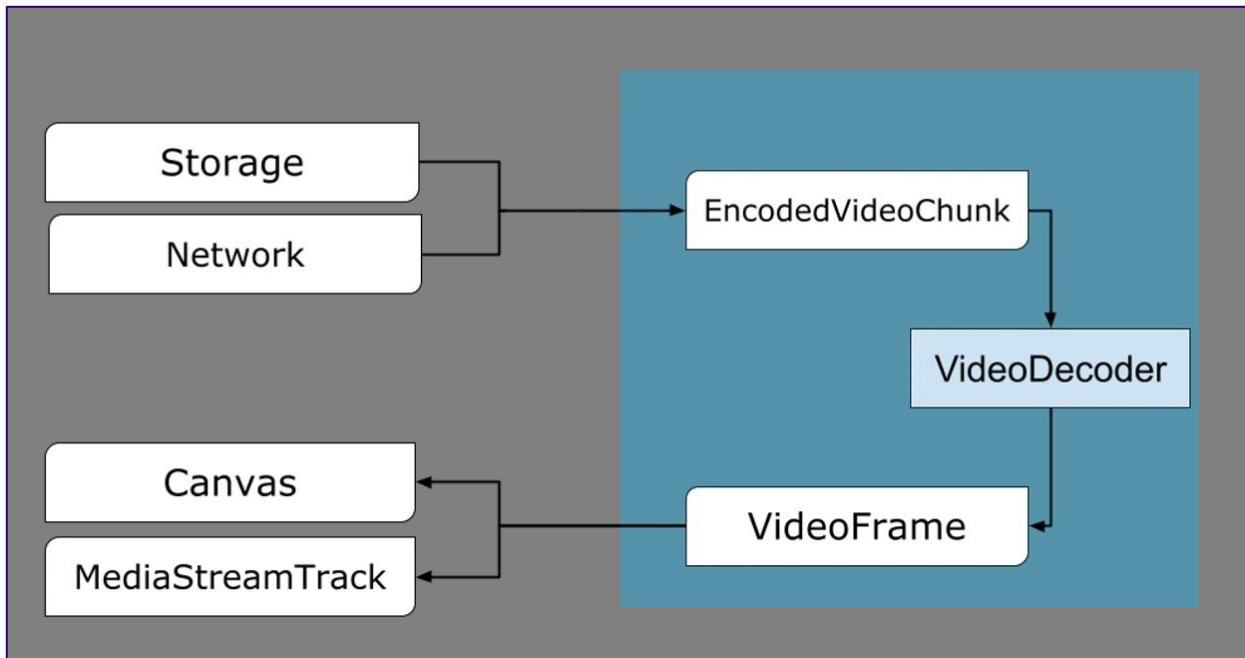
- Send

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

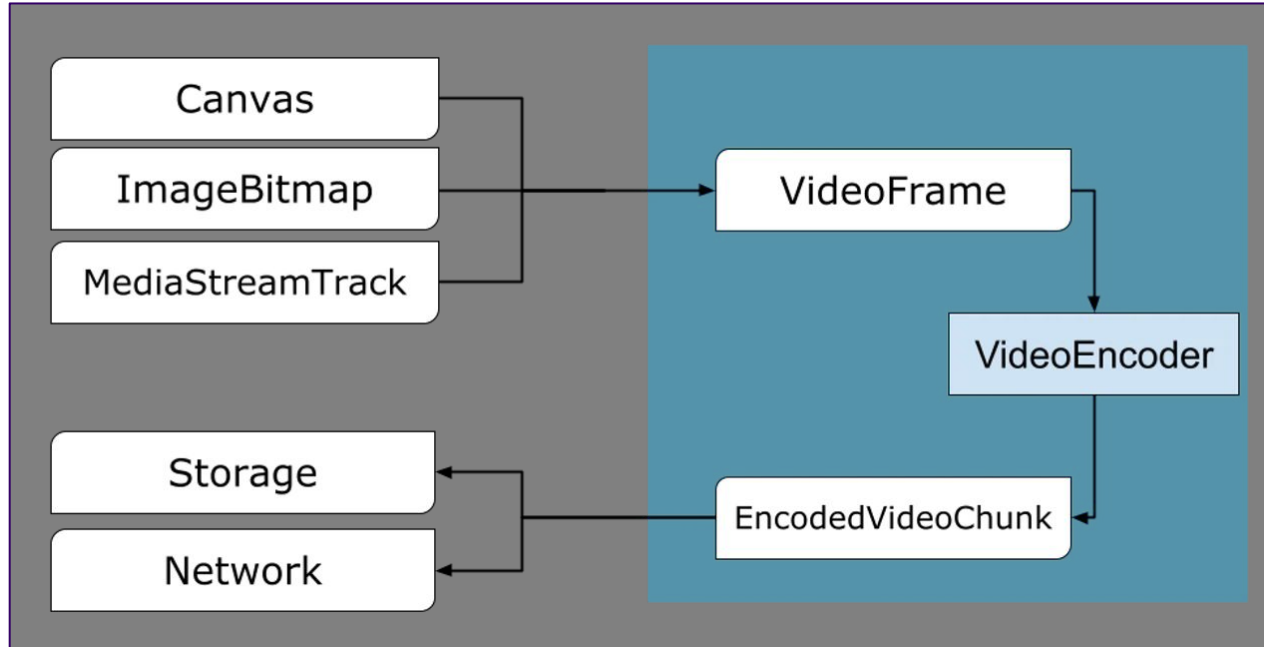
- Receive

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

Video decoding in WebCodecs



Video decoding (similar just reversed)



WebCodecs Codec Support



For Chrome:

- `VideoDecoder`: AVC (H.264), VP8, VP9, AV1 and HEVC (hardware only)
- `AudioDecoder`: AAC, FLAC, MP3, Opus, Vorbis, μ -law and A-law PCM formats.
- `VideoEncoder`: H264, VP8, VP9, AV1 and HEVC (hardware only)
- `AudioEncoder`: Opus and AAC.

Use `isConfigSupported()`

Details of support can be platform / device specific

WebTransport Protocol Mappings (Section 10)



| API Method | QUIC Protocol Action |
|---|---|
| <code>writable.abort(errorCode)</code> | sends RESET_STREAM with errorCode |
| <code>writable.close()</code> | sends STREAM_FINAL |
| <code>writable.getWriter().write()</code> | sends STREAM |
| <code>writable.getWriter().close()</code> | sends STREAM_FINAL |
| <code>writable.getWriter().abort(errorCode)</code> | sends RESET_STREAM with errorCode |
| <code>readable.cancel(errorCode)</code> | sends STOP_SENDING with errorCode |
| <code>readable.getReader().cancel(errorCode)</code> | sends STOP_SENDING with errorCode |
| <code>wt.close(closeInfo)</code> | terminates session with closeInfo |

QUIC Protocol -> API Effect (Section 10)



| QUIC Protocol Action | API Effect |
|---|--|
| received STOP_SENDING with errorCode | errors writable with streamErrorCode |
| received STREAM | (await readable .getReader()). read() .value |
| received STREAM_FINAL | (await readable .getReader()). read() .done |
| received RESET_STREAM with errorCode | errors readable with streamErrorCode |
| Session cleanly terminated with closeInfo | (await wt. closed).closeInfo, and errors open streams |
| Network error | (await wt. closed) rejects, and errors open streams |

RVFC Timing Model

§ 2. VideoFrameCallbackMetadata

```
dictionary VideoFrameCallbackMetadata {  
  required DOMHighResTimeStamp presentationTime;  
  required DOMHighResTimeStamp expectedDisplayTime;  
  
  required unsigned long width;  
  required unsigned long height;  
  required double mediaTime;  
  
  required unsigned long presentedFrames;  
  double processingDuration;  
  
  DOMHighResTimeStamp captureTime;  
  DOMHighResTimeStamp receiveTime;  
  unsigned long rtpTimestamp;  
};
```

For More Information



Tutorials

[WebCodecs](#)

[WebTransport](#)

[W3C Web Media Pipeline Architecture Repo](#)

Sample code

Architecture Issues

Links to specs

Two Samples



1. Sample #1 encodes and decodes video in a WHATWG Streams pipeline without transport.
 - a. Live site:
<https://webrtc.internaut.com/wc/wcWorker/>
 - b. Github repo:
<https://github.com/aboba/wc-demo/>
2. Sample #2 adds network transport to the sending and receiving pipelines, bouncing encoded frames off a relay in the cloud. Comparison with sample #1 can help isolate network effects.
 - a. Live site (Chrome Stable):
<https://webrtc.internaut.com/wc/wtSender10/>
 - b. GitHub repo: <https://github.com/aboba/wt-demo>

WebCodecs in Worker

```
log-info: DOM Content Loaded
log-info: Worker created.
log-info: Default (QVGA) selected
log-info: getMedia called
log-info: Worker msg: Stream event received.
log-info: Worker msg: Start method called.
log-info: Worker msg: Encoder successfully configured:
{"alpha":"discard",bitrate:3000000,"bitrateMode":"variable","codec":"vp8","framerate":30.000030517578125,"hardwareAcceleration":"no-preference","height":240,"latencyMode":"realtime","scalabilityMode":"L1T3","width":320}
log-info: Worker msg: Decoder successfully configured:
{"codec":"vp8","codedHeight":240,"codedWidth":320,"colorSpace":
{"fullRange":false,"matrix":"smpte170m","primaries":"smpte170m","transfer":"smpte170m"},"hardwareAcceleration":"no-preference"}
```



Start

Stop

What's in the Samples

- WHATWG Streams-based Receive and send pipelines.
 - Send and receive pipelines in a (single) worker.
 - Transferable streams used to tunnel video to/from the main thread.
 - Encode/Decode stages based on WebCodecs
 - Send/Receive transport based on WebTransport frame/stream transport (no datagrams).
 - Uncontainerized (raw) video.
 - Conversion from VideoFrames <=> MediaStreamTracks via Mediacapture-transform API.
 - Frames bounced off a store & forward relay.
 - To do: cut-through relay.
- Partial reliability.
 - Used along with Scalable Video Coding (temporal scalability).
 - 'RESET_FRAME timer' set lower for discardable (extension layer) frames. Base layer frames considered 'non-discardable'.
- Concurrency:
 - Send pipeline: multiple frames in transit
 - P-frames sent alongside (much larger) I-frames
 - To do: fully concurrent reading

Frame/Stream Transport



- Send pipeline
 - Sender opens a uni-directional stream.
 - Sender sets a timer with an expiration (RTO)
 - If the timer fires, sender resets the stream
 - RTO values can differ between discardable and non-discardable frames.
 - Sender writes the header + payload to the uni-directional stream
 - Sender closes the stream.
- Receive pipeline
 - Receiver is notified of an incoming uni-directional stream.
 - Receiver reads from incoming uni-directional streams until:
 - A stream is closed OR
 - The stream is reset.
 - Length field helpful for memory allocation as well as to check the frame was completely received.

What You Can Do

- Vary encoding parameters, codecs, bitrates, resolutions, etc.
 - Can visually compare local and remote video
 - Can (manually) measure glass-glass latency
 - Can find bugs and gasp in horror!
 - VP9 + SVC + VBR = Boom!
- Post-experiment diagnostics
 - Metrics: RTT stats, loss, reordering, etc.
 - Calculated at application layer
 - Chrome does not surface QUIC stack metrics yet
 - Graphs: RTT versus frame length
 - To do: latency breakdown in each pipeline stage

Parameters to Select



bitrate: 100000
keyframe interval: 3000

Codec:

- ☐ H.264
- ☐ H.265
- ☒ VP8
- ☐ VP9
- ☐ AV1

Hardware Acceleration Preference:

- ☐ Prefer Hardware
- ☐ Prefer Software
- ☒ No Preference

Latency goal:

- ☒ realtime
- ☐ quality

Bitrate mode:

- ☐ constant
- ☒ variable

Scalability Mode:

- ☐ L1T1
- ☐ L1T2
- ☒ L1T3

Resolution:

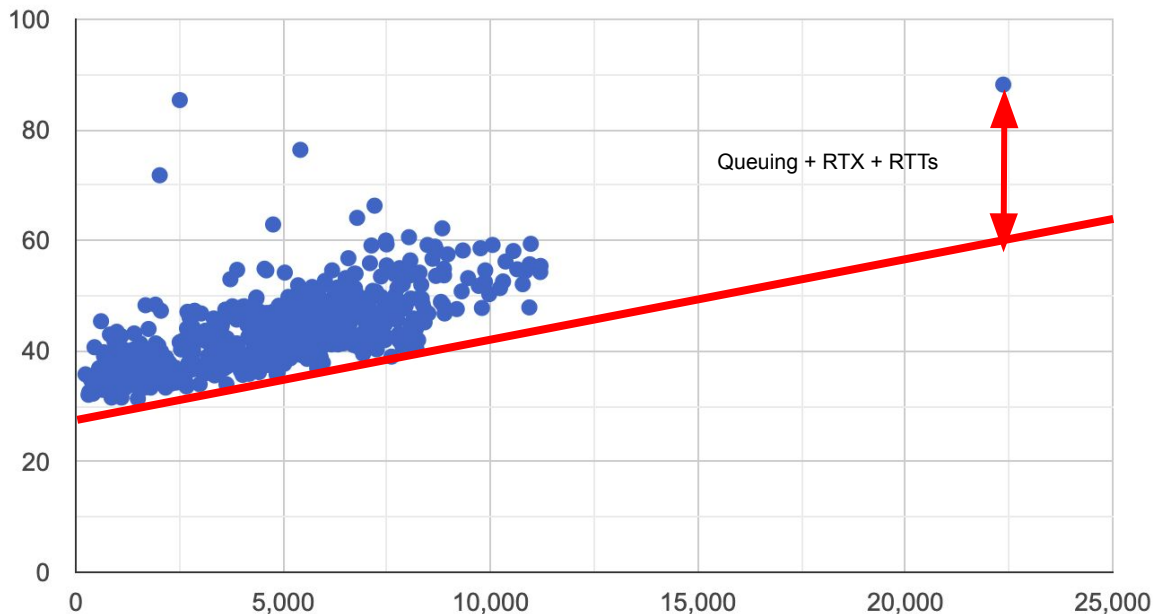
- ☒ QVGA
- ☐ VGA
- ☐ HD
- ☐ Full HD
- ☐ Television 4k (3840x2160)
- ☐ Cinema 4K (4096x2160)
- ☐ 8K

- Bitrate: “Average Target Bitrate” target provided to the encoder.
- Keyframe interval: number of frames between each keyframe.
- Codec: H.264, H.265, VP8, VP9 or AV1
 - H.265 support only where hw acceleration is available.
- Hardware Acceleration Preference: require hw acceleration, require sw only or “no preference”. Hw acceleration often not available.
- Latency goal: “quality” produces smaller frame sizes, but takes (marginally) longer than “realtime”.
- Bitrate mode: Constant Bitrate (CBR) or Variable Bitrate (VBR).
- Scalability mode: how many temporal layers to use. Enables differential protection for the base layer.
- Resolution: reflected in getUserMedia constraints. If your camera doesn't support the requested resolution, window will be blacked out.

Example

- AV1 @ full-Hd with 1152.4 Kbps average bitrate and 30 fps, GoP = 3000, L1T3 scalability mode
- Largest (I-)frame = 22361 octets, median (P-)frame size = 4894 octets
- RTTmin = 31.4 ms. RTTmax = 88.2 ms
- I-frame further from the transmission line than vast majority of P-frames.
 - I-frames most likely to experience loss, queuing delays, multiple RTTs if cwind < 22361

RTT (ms) versus Frame length



BWE report:

```
{"count":553,"loss":0,"reorder":1,bwu":1152379.52  
,"seqmin":0,"seqmax":552,"lenmin":234,"lenfquart"  
:2715.5,"lenmedian":4894,"lentquart":6605.5,"len  
max":22361,"recvsum":2657416}
```

RTT report:

```
{"count":553,"min":31.4,"fquart":38.6495,"avg":43.  
41247377938517,"median":42.1,"tquart":47.301,"  
max":88.199,"stdev":7.155866034173623,"srtt":40  
.33911687334871,"rttvar":2.802225680520845,"rt  
o":51.54801959543209}
```

High Level Observations

- Video quality
 - AV1 can encode/decode in full-HD on new hardware, producing passable video quality even at low bitrates (<300 Kbps)
- Resilience
 - WebTransport frame/stream + temporal scalability provides excellent resilience.
 - Most losses due to RESET_FRAME timeouts (e.g. discardable frames)
- RESET handling
 - RESET_FRAMEs SHOULD be surfaced immediately (but don't always appear to be).
 - Frames can be received intact, even after a RESET_FRAME is sent!

High Level Observations (cont'd)

- P-frames are typically small (a few packets) and exhibit frame RTT clustered around the “transmission line”.
- I-frames are ***much*** larger (10X or more) and often exhibit frame RTT considerably above the “transmission line”.
 - Effect most pronounced with high GoP sizes and low concurrency.
 - Effect seen even for low bandwidth utilization and low loss.
 - Suggests this is not just due to queuing delay or retransmissions
 - If frame size > cwind, requires multiple roundtrips.
 - Observation: increased concurrency lowers the gap between the frame RTT and the transmission line.

Thoughts on Concurrency

<https://webrtc.internaut.com/wc/wtSender10/>

- Concurrency desirable (lowers glass-glass latency) but implies more open streams.
 - For concurrency, both send and receive pipelines need to avoid blocking.
 - In Javascript, be wary of **await**!
 - `promise.then(f).catch()` is *not* the same as **await f**!
- A **good** sign: multiple P-frames arrive on the receiver prior to complete receipt of the initial I-frame.
 - Lack of re-ordering is a ***symptom of blocking!***
- Greater concurrency in frame/stream transport implies more re-ordering
 - More P-frames will be sent concurrently with initial (and subsequent) I-frames
 - Moves I-frame RTT closer to transmission line (cwind grows faster)
 - See `async writeChunk()` function implementing frame/stream sender
 - More work needed on read pipeline

Thoughts on Partial Reliability

- Temporal scalability enables “differential transport” on the sender.
 - Sender can set timer, send RESET_STREAM if timer expires.
 - Timer set lower for “discardable” frames (extension layers).
 - High timer set for “non-discardable” frames (base layer).
- Implications
 - Relay needs to forward RESET_STREAM frames.
 - Relay can receive RESET_STREAM before or after FIN.
 - Receiver needs to be prepared for loss of the RESET_STREAM frame.
 - Useful to have a Length field at the beginning of the packet.
 - Helps receiver to verify that it has received the complete frame.
 - Length field needs to be large enough for high resolution I-frames (could be 200KB+)