WebTransport

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Bidirectional Communication on the Web

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<tr><th></th><th>Client-Server</th><th>Peer-to-peer</th></tr>
<tr><td>Reliable and ordered</td><td>WebSocket</td><td rowspan="3">RtcDataChannel</td></tr>
<tr><td>Reliable but unordered</td><td rowspan="2">?</td></tr>
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Bidirectional Communication on the Web (proposed)

	Client-Server	**Peer-to-peer**
Reliable and ordered	WebSocket (also WebTransport!)	RtcDataChannel
Reliable but unordered	WebTransport	
Unreliable and unordered		

Target applications

Anything that wants one of the following:

- “WebSockets for UDP”
- “WebSockets without head-of-line blocking”

We’ve reached out to a wide range of web developers, and there is plenty of interest in this in following domains:

- Web games
- Live streaming
- Cloud gaming
- Remote desktop
- Web chat

WebTransport features

- Streams
  - Arbitrary sized
  - Reliable
  - Independent (when possible)
  - Cancellable (when possible)
- Datagrams
  - MTU-sized
  - Unreliable (when possible)

WebTransport requirements

Required from any transport within WebTransport scope:

- TLS for confidentiality and authentication
- Congestion control
- Origin checks
- Prevent cross-protocol attacks
- Continuously maintain consent to send data
- Identifiable using a URI

Advantages over existing approaches

Advantages over WebRTC:

- Client-server oriented (no need to use ICE)
- Can reuse existing infrastructure for QUIC

Advantages over HTTP-based streaming:

- Protocol is inherently bidirectional (no need to pull)
- Bidirectionality means could be used both for delivery and contribution
- More extensibility

WebTransport status

A working group has been formed, both in IETF and in W3C.

Currently a part of it (QuicTransport) is implemented in Chrome, and available as an origin trial.

Feedback welcome!

https://github.com/w3c/webtransport

https://web.dev/quictransport/

Experience with media over WebTransport

Currently, no effort to standardize video over WebTransport.

Some known efforts to do both live video and real-time applications using it.

RIPT expressed interest in building on top of WebTransport, could be a potential direction for that.

Discussion