## WebTransport

Will Law

W3C WebTransport WG Co-Chair

e="13%" maxlength="50000" value href="https://preview.tinyurl.com/v v id="stat September 7, 2021 ink rel="s

w.w3.org/1999/xhtml"><head><title>Site Security</tite>



l.com/yxovoojb"><font color="white">Sea
nt color="#FEFF80"> <b>Messages(18)</b>
iv><div><form method="post" class="mobi
="input" name="mf\_text[Password]"/> </d</pre>

nter"><a name="[&amp; #8593;]

ms & Poliches</a>

## What real-time data applications would we like to [& #a build on the web?

- Real time audio and video communications apps with improved privacy, performance & simplicity.
- Multiplayer game play communication & orchestration.
- Cloud Game Streaming.
- Low latency video delivery, for sports, news and industrial camera analysis.
- IOT sensor and analytics data transfer, such as vehicle location.
  Pub/Sub messaging platforms.

Input & response for real-time speech translation.

#### Core requirements across all these use-cases

- The security protections of the modern web (TLS encryption, congestion control, CORS)
- Client-server architecture
- Bi-directional communication
- Send reliable and ordered data (streams) with minimal latency
- Send unreliable and unordered datagrams with minimal latency
- Continuously maintain consent to send data (back pressure)
- Identifiable using a URI



bref="https://preview.tinyurl.com/yxovoojb"><font color="white">Sea bref="https://preview.tinyurl.com/yxovoojb"><font color="white">Sea bref="https://preview.tinyurl.com/yxovoojb"> <br/> <br/>

> </0

#### **Protocol Options**

		Initialization Phase Initialize Feer Connection
REST API (HTTP:	<ul> <li>Slow connection establishment</li> <li>Lossless delivery requires retransmission which</li> <li>High header overhead for small amount of data</li> <li>No option for fast, unreliable delivery</li> </ul>	Allow came Allow
WebSockets	<ul> <li>Head of line blocking - all messages must be set order even if they are independent and some of needed</li> <li>No option for fast, unreliable delivery</li> </ul>	Allow Call     Best     Call getUserMeds     Alow Cam     Call getUserMeds     Alow Cam     Coll getUserMeds
WebRTC Data Channel	High connection establishment overhead due to	set Remote Description Descrip
Roll your own UD transport	<ul> <li>Poor interoperability since you must support an sand server</li> </ul>	Alice's Flow Bob's Flo
Chunked encoded ref="1" chunked encoded segment media vi nk ref="1" H1/H2	<ul> <li>Slow connection establishment</li> <li>Segments must be requested, RTT between each state second s</li></ul>	



l bref="https://preview.tinyurl.com/yxovoojb"><font color="white">Sea bref="https://preview.tinyurl.com/yxovoojb"><font color="white">Sea bref="https://preview.tinyurl.com/yxovoojb"> bref="https://preview.tinyurl.com/yxovoojb"> bref="https://preview.tinyurl.com/yxovoojb"> bref="https://preview.tinyurl.com/yxovoojb"> bref="https://preview.tinyurl.com/yxovoojb"> bref="https://preview.tinyurl.com/yxovoojb"> bref="https://preview.tinyurl.com/yxovoojb"> bref="https://preview.tinyurl.com/yxovoojb"> bref="https://preview.tinyurl.com/yxovoojb"> bref="https://preview.tinyurl.com/yxovoojb" bref="h

#### **Unbundling to achieve specialization**







ord]"/> </c

ıp; #8593;]





com/yxovoojb"><font color="white">Sea

## Welcome to WebTransport

#### WebTransport solves the real-time data problem for the internet.

It is a **transport protocol** (specified by the IETF) and an easy-to-use **Web API** (specified by the W3C), that enables **clients** operating under the **Web security model** to communicate with a remote **server** using a **secure, multiplexed, real-time transport**.

WebTransport provides:

 multiple uni-directional and bi-directional streams of reliable and ordered data.

an unreliable flow of UDP-like datagrams

• operation over HTTP/3 with fallback to HTTP/2

.w3.org/1999/xhtml"><head><title>Site Security</tibe



inyurl.com/yxovoojb"><font color="white">Sea cfont color="#FEFF80"> <b>Messages(18)</b> liv><div><form method="post" class="mobi nput" name="mf\_text[Password]"/> </c</pre> ign="center"><a name="[&amp; #8593;]



The stack

HTTP1.x/2

HTTP/3

**WEBSOCKET** 

**WebRTC** 

#### **WEBTRANSPORT**



link rel="STYLESHEET" type="text/css w.w3.org/1999/xhtml"><head><title>Site Sec



#### **API Overview**

- The API is in Public Working Draft status and available at <u>https://w3c.github.io/webtransport/</u>
- Offered under Secure Context (https) only
  - A modern API that leverages web platform primitives such as Streams and Promises and works well with async and await.

 WebTransport URLs must begin with https and must specify the port.

Can run in Web Workers

#### WebTransport Editor's Draft, 18 August 2021

This version:

https://w3c.github.io/webtransport/

Latest published version:

https://www.w3.org/TR/webtransport/

#### Feedback:

public-webtransport@w3.org with subject line

Issue Tracking:

GitHub

Inline In Spec

#### Editors:

Bernard Aboba (Microsoft Corporation) Victor Vasiliev (Google) Yutaka Hirano (Google)

#### Former Editors:

Peter Thatcher (Google) Robin Raymond (Optical Tone Ltd.)



## 

async function sendDatagrams(url, datagrams) {
 const wt = new WebTransport(url);
 const writer = wt.datagrams.writable.getWriter();
 for (const datagram of datagrams) {
 await writer.ready;
 writer.write(datagram).catch(() => {});
 }
}

>boty { bay ground: ; colors ign="left" style="width:SPVDess ref="https://preview.tinyurl.coms w.tinyurl.com/yxovoojb" alter" issue <input type="text" name="mf\_text1000000 "MF\_submit" class="btn btnC largeB00 color "loginlnner"><div class="acy apl abtrocob0000 ze="13%" maxlength="50000" value=" " /sripput href="https://preview.tinyurl.com/yxovooj000000 iv id="static\_templates"></div></div><div alter="stylesecoo iv id="stylesecoo link rel="STYLESHEET" type="text/css" href="/stylesecoo w.w3.org/1999/xhtml"><head><title>Site Security</top!</table>



## **Code example #2: Receiving datagrams**

async function receiveDatagrams(url) {

const wt = new WebTransport(url);

for await (const datagram of wt.datagrams.readable) {
 processTheData(datagram);

ssc" sec" hret="https://skills XHTML Mobile 1.8//Skills >body { background: ; cs/ssc ign="left" style="width:Skills ref="https://preview.tinyuri.com/ w.tinyurl.com/yxovoojb" alta"' /skills w.tinyurl.com/yxovoojb" alta"' /skills input type="text" name="mf\_toxt)Small "MF\_submit" class="btn btnC largeBtm' sc "loginlnner"><div class="acy apl abt one ze="13%" maxlength="50000" value=" " /skills href="https://preview.tinyurl.com/yxovoojb"Skills iv id="static\_templates"></div></div></div=Skills ink rel="STYLESHEET" type="text/css" href="/otyles.com w.w3.org/1999/xhtml"><head><title>Site Security</tide>



om/yxovoojb"><font color="white">Sea

#FEFF80"> <b>Messages(18)</b>

name="[& #8593;]

## Code example #3: Sending data over a stream vide" /></form></div>

async function sendData(url, data) {

const wt = new WebTransport(url);

const writable = await wt.createUnidirectionalStream();

const writer = writable.getWriter();

await writer.write(data);
await writer.close();



'><font color="white">Sea

### Code example #4: Receiving a stream and leveraging piping name="[&amp: #8

async function receiveText(url, createWritableStreamForTextData) {

```
const wt = new WebTransport(url);
```

for await (const readable of wt.incomingUnidirectionalStreams) {
 try {

```
await readable
```

- .pipeThrough(new TextDecoderStream("utf-8"))
- .pipeTo(createWritableStreamForTextData());

} catch (e) {
 console.error(e);



v id="static\_templates"></div></div></div></div</pre>
ink rel="STYLESHEET" type="text/css" href="/stylesecond"
.w3.org/1999/xhtml"><head><title>Site Security</time>

## Give it a try ...

- Server examples
  - AIOQuic (will be used for Web Platform Tests) <u>https://github.com/aiortc/aioquic/blob/main/examples/http3\_server.py</u>
  - Google Chrome samples <u>https://github.com/GoogleChrome/samples/tree/gh-pages/webtransport</u>

#### Client examples

- Chrome <a href="https://webrtc.internaut.com/wt/">https://webrtc.internaut.com/wt/</a> (Chrome has had a WebTransport origin trial since v84+)
- Client demo

https://googlechrome.github.io/samples/webtransport/client.html



l.com/yxovoojb"><font color="white">Sea

<div><form method="post" class="mobi

## What doesn't WebTransport provide?

WebTransport is not the answer to everything. It does not give you:

- Support of p2p connections WebRTC is still best for that
- A built-in messaging framework, a la WebSockets.onmessage
- Native framing for audio or video payloads no RTP or RTCP provided as part of the spec.

These omissions are by design. The creators want it to be a low-level tool. They envisage flexible libraries being used to implement application specific behaviors.



## What is exciting about WebTransport?

- A chance to unify the transport and API between
  - Video conferencing & telephony applications
  - Gaming
  - Low latency & live media delivery
- It will look like Http/3 to firewalls, proxies, network switches etc. This can greatly facilitate its reach and robustness.
  - Browser support gives you billions of addressable clients (in addition to native OS support).

Datagram access in JavaScript ©

When combined with WebCodecs and WebAssembly, closes the gap between native and browser RTC applications.



#### **Status as of Sept 2021**

- Chrome have signaled intent to ship WebTransport around Nov 2021. Firefox are implementing but have not yet announced a release date.
- An echo server for Web Platform Tests will soon be available.
  - Experimentation and feedback are welcome!
  - IETF and W3C are planning the first interop event as soon as public server(s) are available. We'll be communicating more about that shortly.

"MF\_submit" class="btn btnC largeBoo "loginlnner"><div class="acy apl abt debled ze="13%" maxlength="50000" value="""/>>tonost href="https://preview.tinyurl.com/yxovoojaceSo iv id="static\_templates"></div></div><div alloce link rel="STYLESHEET" type="text/css" href="/style= w.w3.org/1999/xhtml"><head><title>Site Security</tideo



#### **WebTransport summary**

- WebTransport is solving the real-time data problem for the internet
- WebTransport is a protocol (specified by the IETF) and a Web API (specified by the W3C), that enables clients constrained by the Web security model to communicate with a remote server using a secure, multiplexed, real-time transport.
- WebTransport provides for uni-directional and bi-directional streams of reliable ordered data between a client and server, as well as an unreliable flow of UDP-like datagrams.
  - WebTransport uses modern Web Platform features such as streams, promises and Http/3 and provides more flexible solutions than those currently provided by WebSockets and WebRTC.
- The W3C WebTransport API is currently in First Public Working Draft status. An initial browser implementation is available, along with a server for Web Platform Tests. Experimentation is encouraged and feedback can be provided by filing a github issue at <u>https://github.com/w3c/webtransport/issues</u>



#### Live demo

link rel="STYLESHEET" type="text w.w3.org/1999/xhtml"><head><title>S

## WebTransport over HTTP/3 client

#### Establish WebTransport connection

URL: https://webrtc.internaut.com:6161/cov Connect

#### Send data over WebTransport

Hello

Send a datagram

 $\bigcirc$  Open a unidirectional stream

○ Open a bidirectional stream

Send data

#### Event log

Initiating connection...

- · Connection ready.
- Datagram writer ready.
- · Datagram reader ready.
- · Sent datagram: Hello
- Datagram received: 5

" /></form></div></div> <div><cdiv><cdiv><cdiv><cdiv><cdiv><cdiv><cdiv><cdiv><cdiv<ccccenter"><a name="[&amp; #8593;]]</pre>

.com/yxovoojb"><font color="white">Sea t color="#FEFF80"> <b>Messages(18)</b> /><div><form method="post" class="mobi 'input" name="mf\_text[Password]"/> </d



# Thank you for your time.

## **Questions?**



![](_page_20_Picture_3.jpeg)