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WebTransport over HTTP/3  
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## Abstract

WebTransport [[OVERVIEW](#)] is a protocol framework that enables clients constrained by the Web security model to communicate with a remote server using a secure multiplexed transport. This document describes a WebTransport protocol that is based on HTTP/3 [[HTTP3](#)] and provides support for unidirectional streams, bidirectional streams and datagrams, all multiplexed within the same HTTP/3 connection.

## Note to Readers

Discussion of this draft takes place on the WebTransport mailing list ([webtransport@ietf.org](mailto:webtransport@ietf.org)), which is archived at [https://mailarchive.ietf.org/arch/search/?email\\_list=webtransport](https://mailarchive.ietf.org/arch/search/?email_list=webtransport).

The repository tracking the issues for this draft can be found at <https://github.com/ietf-wg-webtrans/draft-ietf-webtrans-http3/issues>. The web API draft corresponding to this document can be found at <https://w3c.github.io/webtransport/>.

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## [1.](#) Introduction

HTTP/3 [[HTTP3](#)] is a protocol defined on top of QUIC [[RFC9000](#)] that can multiplex HTTP requests over a QUIC connection. This document defines a mechanism for multiplexing non-HTTP data with HTTP/3 in a manner that conforms with the WebTransport protocol requirements and semantics [[OVERVIEW](#)]. Using the mechanism described here, multiple WebTransport instances can be multiplexed simultaneously with regular HTTP traffic on the same HTTP/3 connection.

### [1.1.](#) Terminology

The keywords "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [BCP 14](#) [[RFC2119](#)] [[RFC8174](#)] when, and only when, they appear in all capitals, as shown here.

This document follows terminology defined in Section 1.2 of [[OVERVIEW](#)]. Note that this document distinguishes between a WebTransport server and an HTTP/3 server. An HTTP/3 server is the server that terminates HTTP/3 connections; a WebTransport server is an application that accepts WebTransport sessions, which can be accessed via an HTTP/3 server.

## [2.](#) Protocol Overview

WebTransport servers in general are identified by a pair of authority value and path value (defined in [[RFC3986](#)] Sections [3.2](#) and [3.3](#) correspondingly).

When an HTTP/3 connection is established, both the client and server have to send a `SETTINGS_ENABLE_WEBTRANSPORT` setting in order to indicate that they both support WebTransport over HTTP/3.

WebTransport sessions are initiated inside a given HTTP/3 connection

by the client, who sends an extended CONNECT request [[RFC8441](#)]. If the server accepts the request, a WebTransport session is established. The resulting stream will be further referred to as a `_CONNECT stream_`, and its stream ID is used to uniquely identify a given WebTransport session within the connection. The ID of the CONNECT stream that established a given WebTransport session will be further referred to as a `_Session ID_`.

After the session is established, the peers can exchange data using the following mechanisms:

- \* A client can create a bidirectional stream using a special indefinite-length HTTP/3 frame that transfers ownership of the stream to WebTransport.
- \* A server can create a bidirectional stream, which is possible since HTTP/3 does not define any semantics for server-initiated bidirectional streams.
- \* Both client and server can create a unidirectional stream using a special stream type.
- \* A datagram can be sent using HTTP Datagrams [[HTTP-DATAGRAM](#)].

An WebTransport session is terminated when the CONNECT stream that created it is closed.

### [3.](#) Session Establishment

#### [3.1.](#) Establishing a Transport-Capable HTTP/3 Connection

In order to indicate support for WebTransport, both the client and the server MUST send a `SETTINGS_ENABLE_WEBTRANSPORT` value set to "1" in their `SETTINGS` frame. The `SETTINGS_ENABLE_WEBTRANSPORT` parameter value SHALL be either "0" or "1", with "0" being the default; an endpoint that receives a value other than "0" or "1" MUST close the connection with the `H3_SETTINGS_ERROR` error code.

The client MUST NOT send a WebTransport request until it has received the setting indicating WebTransport support from the server.

Similarly, the server MUST NOT process any incoming WebTransport requests until the client settings have been received, as the client may be using a version of WebTransport extension that is different from the one used by the server.

### [3.2.](#) Extended CONNECT in HTTP/3

[RFC8441] defines an extended CONNECT method in [Section 4](#), enabled by the SETTINGS\_ENABLE\_CONNECT\_PROTOCOL parameter. That parameter is only defined for HTTP/2. This document does not create a new multi-purpose parameter to indicate support for extended CONNECT in HTTP/3; instead, the SETTINGS\_ENABLE\_WEBTRANSPORT setting implies that an endpoint supports extended CONNECT.

### [3.3.](#) Creating a New Session

As WebTransport sessions are established over HTTP/3, they are identified using the https URI scheme [[RFC7230](#)].

In order to create a new WebTransport session, a client can send an HTTP CONNECT request. The :protocol pseudo-header field ([\[RFC8441\]](#)) MUST be set to webtransport. The :scheme field MUST be https. Both the :authority and the :path value MUST be set; those fields indicate the desired WebTransport server. An Origin header [[RFC6454](#)] MUST be provided within the request.

Upon receiving an extended CONNECT request with a :protocol field set to webtransport, the HTTP/3 server can check if it has a WebTransport server associated with the specified :authority and :path values. If it does not, it SHOULD reply with status code 404 ([Section 6.5.4](#), [[RFC7231](#)]). If it does, it MAY accept the session by replying with a 2xx series status code, as defined in Section 15.3 of [[SEMANTICS](#)]. The WebTransport server MUST verify the Origin header to ensure that the specified origin is allowed to access the server in question.

From the client's perspective, a WebTransport session is established when the client receives a 2xx response. From the server's perspective, a session is established once it sends a 2xx response. WebTransport over HTTP/3 does not support 0-RTT.

The webtransport HTTP Upgrade Token uses the Capsule Protocol as

defined in [[HTTP-DATAGRAM](#)].

### [3.4.](#) Limiting the Number of Simultaneous Sessions

From the flow control perspective, WebTransport sessions count against the stream flow control just like regular HTTP requests, since they are established via an HTTP CONNECT request. This document does not make any effort to introduce a separate flow control mechanism for sessions, nor to separate HTTP requests from WebTransport data streams. If the server needs to limit the rate of incoming requests, it has alternative mechanisms at its disposal:

- \* HTTP\_REQUEST\_REJECTED error code defined in [[HTTP3](#)] indicates to the receiving HTTP/3 stack that the request was not processed in any way.
- \* HTTP status code 429 indicates that the request was rejected due to rate limiting [[RFC6585](#)]. Unlike the previous method, this signal is directly propagated to the application.

## [4.](#) WebTransport Features

WebTransport over HTTP/3 provides the following features described in [[OVERVIEW](#)]: unidirectional streams, bidirectional streams and datagrams, initiated by either endpoint.

Session IDs are used to demultiplex streams and datagrams belonging to different WebTransport sessions. On the wire, session IDs are encoded using the QUIC variable length integer scheme described in [[RFC9000](#)].

If at any point a session ID is received that cannot a valid ID for a client-initiated bidirectional stream, the recipient MUST close the connection with an H3\_ID\_ERROR error code.

### [4.1.](#) Unidirectional streams

Once established, both endpoints can open unidirectional streams. The HTTP/3 unidirectional stream type SHALL be 0x54. The body of the stream SHALL be the stream type, followed by the session ID, encoded as a variable-length integer, followed by the user-specified stream



bidirectional streams is WEBTRANSPORT\_STREAM.

TODO: move the paragraph above into a separate draft; define what happens with already existing HTTP/3 frames on server-initiated bidirectional streams.

#### 4.3. Resetting Data Streams

A WebTransport endpoint may send a RESET\_STREAM or a STOP\_SENDING frame for a WebTransport data stream. Those signals are propagated by the WebTransport implementation to the application.

A WebTransport application SHALL provide an error code for those operations. Since WebTransport shares the error code space with HTTP/3, WebTransport application errors for streams are limited to an unsigned 8-bit integer, assuming values between 0x00 and 0xff. WebTransport implementations SHALL remap those error codes into an error range where 0x00 corresponds to 0x52e4a40fa8db, and 0xff corresponds to 0x52e4a40fa9e2. Note that there are code points inside that range of form "0x1f \* N + 0x21" that are reserved by Section 8.1 of [HTTP3]; those have to be accounted for when mapping the error codes by skipping them (i.e. the two HTTP/3 error codepoints adjacent to a GREASE codepoint would map to two adjacent WebTransport application error codepoints). An example pseudocode can be seen in Figure 3.

```
first = 0x52e4a40fa8db
last = 0x52e4a40fa9e2

def webtransport_code_to_http_code(n):
    return first + n + floor(n / 0x1e)

def http_code_to_webtransport_code(h):
    assert(first <= h <= last)
    assert((h - 0x21) % 0x1f != 0)
    shifted = h - first
    return shifted - shifted // 0x1f
```



application errors and HTTP/3 error codes; here, `/// is integer division`

WebTransport data streams are associated with sessions through a header at the beginning of the stream; resetting a stream may result in that data being discarded. Because of that, WebTransport application error codes are best effort, as the WebTransport stack is not always capable of associating the reset code with a session. The only exception is the situation where there is only one session on a given HTTP/3 connection, and no intermediaries between the client and the server.

WebTransport implementations SHALL forward the error code for a stream associated with a known session to the application that owns that session; similarly, the intermediaries SHALL reset the streams with corresponding error code when receiving a reset from the peer. If a WebTransport implementation intentionally allows only one session over a given HTTP/3 connection, it SHALL forward the error codes within WebTransport application error code range to the application that owns the only session on that connection.

#### [4.4.](#) Datagrams

Datagrams can be sent using HTTP Datagrams, using the WEB\_TRANSPORT HTTP Datagram Format Type (see value in [Section 8.6](#)). When using the WEB\_TRANSPORT HTTP Datagram Format Type, the WebTransport datagram payload is sent unmodified in the "HTTP Datagram Payload" field of an HTTP Datagram. When sending a registration capsule using the "Datagram Format Type" set to WEB\_TRANSPORT, the "Datagram Format Additional Data" field SHALL be empty.

In QUIC, a datagram frame can span at most one packet. Because of that, the applications have to know the maximum size of the datagram they can send. However, when proxying the datagrams, the hop-by-hop MTUs can vary. TODO: Describe how the path MTU can be computed, specifically propagation across HTTP proxies.

#### [4.5.](#) Buffering Incoming Streams and Datagrams

In WebTransport over HTTP/3, the client MAY send its SETTINGS frame, as well as multiple WebTransport CONNECT requests, WebTransport data streams and WebTransport datagrams, all within a single flight. As those can arrive out of order, a WebTransport server could be put into a situation where it receives a stream or a datagram without a corresponding session. Similarly, a client may receive a server-initiated stream or a datagram before receiving the CONNECT response headers from the server.

To handle this case, WebTransport endpoints SHOULD buffer streams and datagrams until those can be associated with an established session. To avoid resource exhaustion, the endpoints MUST limit the number of buffered streams and datagrams. When the number of buffered streams is exceeded, a stream SHALL be closed by sending a RESET\_STREAM and/or STOP\_SENDING with the H3\_WEBTRANSPORT\_BUFFERED\_STREAM\_REJECTED error code. When the number of buffered datagrams is exceeded, a datagram SHALL be dropped. It is up to an implementation to choose what stream or datagram to discard.

## 5. Session Termination

A WebTransport session over HTTP/3 is considered terminated when either of the following conditions is met:

- \* the CONNECT stream is closed, either cleanly or abruptly, on either side; or
- \* a CLOSE\_WEBTRANSPORT\_SESSION capsule is either sent or received.

Upon learning that the session has been terminated, the endpoint MUST reset all of the streams associated with the session; it MUST NOT send any new datagrams or open any new streams.

To terminate a session with a detailed error message, an application MAY send an HTTP capsule [[HTTP-DATAGRAM](#)] of type CLOSE\_WEBTRANSPORT\_SESSION (0x2843). The format of the capsule SHALL be as follows:

```
CLOSE_WEBTRANSPORT_SESSION Capsule {  
  Type (i) = CLOSE_WEBTRANSPORT_SESSION,  
  Length (i),  
  Application Error Code (32),  
  Application Error Message (..8192),  
}
```

CLOSE\_WEBTRANSPORT\_SESSION has the following fields:

**Application Error Code:** A 32-bit error code provided by the application closing the connection.

**Application Error Message:** A UTF-8 encoded error message string provided by the application closing the connection. The message takes up the remainder of the capsule, and its length MUST NOT exceed 1024 bytes.

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A CLOSE\_WEBTRANSPORT\_SESSION capsule MUST be followed by a FIN on the sender side. If any additional stream data is received on the CONNECT stream after CLOSE\_WEBTRANSPORT\_SESSION, the stream MUST be reset with code H3\_MESSAGE\_ERROR. The recipient MUST close the stream upon receiving a FIN. If the sender of CLOSE\_WEBTRANSPORT\_SESSION does not receive a FIN after some time, it SHOULD send STOP\_SENDING on the CONNECT stream.

Cleanly terminating a CONNECT stream without a CLOSE\_WEBTRANSPORT\_SESSION capsule SHALL be semantically equivalent to terminating it with a CLOSE\_WEBTRANSPORT\_SESSION capsule that has an error code of 0 and an empty error string.

## 6. Negotiating the Draft Version

[[RFC editor: please remove this section before publication.]]

WebTransport over HTTP/3 uses two different mechanisms to negotiate versions for the different parts of the draft.

The hop-by-hop wire format aspects of the protocol are negotiated by changing the codepoint used for the SETTINGS\_ENABLE\_WEBTRANSPORT parameter. Because of that, any WebTransport endpoint MUST wait for the peer's SETTINGS frame before sending or processing any WebTransport traffic. When multiple versions are supported by both of the peers, the most recent version supported by both is selected.

The data exchanged over the CONNECT stream is transmitted across intermediaries, and thus cannot be versioned using a SETTINGS parameter. To indicate support for different versions of the protocol defined in this draft, the clients SHALL send a header for each version of the draft supported. The header corresponding to the version described in this draft is Sec-Webtransport-Http3-Draft02; its value SHALL be 1. The server SHALL reply with a Sec-Webtransport-Http3-Draft header indicating the selected version; its value SHALL be draft02 for the version described in this draft.

## 7. Security Considerations

WebTransport over HTTP/3 satisfies all of the security requirements imposed by [\[OVERVIEW\]](#) on WebTransport protocols, thus providing a secure framework for client-server communication in cases when the client is potentially untrusted.

WebTransport over HTTP/3 requires explicit opt-in through the use of a QUIC transport parameter; this avoids potential protocol confusion attacks by ensuring the HTTP/3 server explicitly supports it. It also requires the use of the Origin header, providing the server with the ability to deny access to Web-based clients that do not originate from a trusted origin.

Just like HTTP traffic going over HTTP/3, WebTransport pools traffic to different origins within a single connection. Different origins imply different trust domains, meaning that the implementations have to treat each transport as potentially hostile towards others on the same connection. One potential attack is a resource exhaustion attack: since all of the transports share both congestion control and flow control context, a single client aggressively using up those resources can cause other transports to stall. The user agent thus SHOULD implement a fairness scheme that ensures that each transport within connection gets a reasonable share of controlled resources; this applies both to sending data and to opening new streams.

## [8.](#) IANA Considerations

### [8.1.](#) Upgrade Token Registration

The following entry is added to the "Hypertext Transfer Protocol (HTTP) Upgrade Token Registry" registry established by [\[RFC7230\]](#):

The "webtransport" label identifies HTTP/3 used as a protocol for WebTransport:

Value: webtransport

Description: WebTransport over HTTP/3

Reference: This document and [[I-D.ietf-webtrans-http2](#)]

## [8.2.](#) HTTP/3 SETTINGS Parameter Registration

The following entry is added to the "HTTP/3 Settings" registry established by [[HTTP3](#)]:

The SETTINGS\_ENABLE\_WEBTRANSPORT parameter indicates that the specified HTTP/3 connection is WebTransport-capable.

Setting Name: ENABLE\_WEBTRANSPORT

Value: 0x2b603742

Default: 0

Specification: This document

## [8.3.](#) Frame Type Registration

The following entry is added to the "HTTP/3 Frame Type" registry established by [[HTTP3](#)]:

The WEBTRANSPORT\_STREAM frame allows HTTP/3 client-initiated bidirectional streams to be used by WebTransport:

Code: 0x41

Frame Type: WEBTRANSPORT\_STREAM

Specification: This document

## [8.4.](#) Stream Type Registration

The following entry is added to the "HTTP/3 Stream Type" registry established by [[HTTP3](#)]:

The "WebTransport stream" type allows unidirectional streams to be used by WebTransport:

Code: 0x54

Stream Type: WebTransport stream

Specification: This document

Sender: Both

### [8.5.](#) HTTP/3 Error Code Registration

The following entry is added to the "HTTP/3 Error Code" registry established by [[HTTP3](#)]:

Name: H3\_WEBTRANSPORT\_BUFFERED\_STREAM\_REJECTED

Value: 0x3994bd84

Description: WebTransport data stream rejected due to lack of associated session.

Specification: This document.

In addition, the following range of entries is registered:

Name: H3\_WEBTRANSPORT\_APPLICATION\_00 ...  
H3\_WEBTRANSPORT\_APPLICATION\_FF

Value: 0x52e4a40fa8db to 0x52e4a40fa9e2 inclusive, with the exception of 0x52e4a40fa8f9, 0x52e4a40fa918, 0x52e4a40fa937, 0x52e4a40fa956, 0x52e4a40fa975, 0x52e4a40fa994, 0x52e4a40fa9b3, and 0x52e4a40fa9d2.

Description: WebTransport application error codes.

Specification: This document.

### [8.6.](#) Datagram Format Type

This document will request IANA to register WEB\_TRANSPORT in the "HTTP Datagram Format Types" registry established by [[HTTP-DATAGRAM](#)].

+=====+=====+=====+

| Type          | Value    | Specification |
|---------------|----------|---------------|
| WEB_TRANSPORT | 0xff7c00 | This Document |

Table 1: Registered Datagram Format Type

## 9. References

### 9.1. Normative References

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