WebTransport over HTTP/2
draft-ietf-webtrans-http2-07

Abstract

WebTransport defines a set of low-level communications features designed for client-server interactions that are initiated by Web clients. This document describes a protocol that can provide many of the capabilities of WebTransport over HTTP/2. This protocol enables the use of WebTransport when a UDP-based protocol is not available.

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.

The repository tracking the issues for this draft can be found at https://github.com/ietf-wg-webtrans/draft-webtransport-http2. The web API draft corresponding to this document can be found at https://w3c.github.io/webtransport/.

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1. Introduction

WebTransport [OVERVIEW] is designed to provide generic communication capabilities to Web clients that use HTTP/3 [HTTP3]. The HTTP/3 WebTransport protocol [WEBTRANSPORT-H3] allows Web clients to use QUIC [QUIC] features such as streams or datagrams [DATAGRAM]. However, there are some environments where QUIC cannot be deployed.

This document defines a protocol that provides all of the core functions of WebTransport using HTTP semantics. This includes unidirectional streams, bidirectional streams, and datagrams.

By relying only on generic HTTP semantics, this protocol might allow deployment using any HTTP version. However, this document only defines negotiation for HTTP/2 [HTTP2] as the current most common TCP-based fallback to HTTP/3.

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/2 server. An HTTP/2 server is the server that terminates HTTP/2 connections; a WebTransport server is an application that accepts WebTransport sessions, which can be accessed using HTTP/2 and this protocol.
2. Protocol Overview

WebTransport servers are identified by an HTTPS URI as defined in Section 4.2.2 of [HTTP].

When an HTTP/2 connection is established, both the client and server have to send a SETTINGS_WEBTRANSPORT_MAX_SESSIONS setting with a value greater than "0" to indicate that they both support WebTransport over HTTP/2. For servers, the value of the setting is the number of concurrent sessions the server is willing to receive. Clients cannot receive incoming WebTransport sessions, so any value greater than "0" sent by a client simply indicates support for WebTransport over HTTP/2.

A client initiates a WebTransport session by sending an extended CONNECT request [RFC8441]. If the server accepts the request, a WebTransport session is established. The stream that carries the CONNECT request is used to exchange bidirectional data for the session. This stream will be referred to as a _CONNECT stream_. The stream ID of a CONNECT stream, which will be referred to as a _Session ID_, is used to uniquely identify a given WebTransport session within the connection. WebTransport using HTTP/2 uses extended CONNECT with the same webtransport HTTP Upgrade Token as [WEBTRANSPORT-H3]. This Upgrade Token uses the Capsule Protocol as defined in [HTTP-DATAGRAM].

After the session is established, endpoints exchange WebTransport messages using the Capsule Protocol on the bidirectional CONNECT stream, the "data stream" as defined in Section 3.1 of [HTTP-DATAGRAM].

Within this stream, _WebTransport streams_ and _WebTransport datagrams_ are multiplexed. In HTTP/2, WebTransport capsules are carried in HTTP/2 DATA frames. Multiple independent WebTransport sessions can share a connection if the HTTP version supports that, as HTTP/2 does.

WebTransport capsules closely mirror a subset of QUIC frames and provide the essential WebTransport features. Within a WebTransport session, endpoints can

* create and use bidirectional or unidirectional streams with no additional round trips using the WT_STREAM capsule

Stream creation and data flow on streams uses flow control mechanisms modeled on those in QUIC. Flow control is managed using the WebTransport capsules: WT_MAX_DATA, WT_MAX_STREAM_DATA, WT_MAX_STREAMS, WT_DATA_BLOCKED, WT_STREAM_DATA_BLOCKED, and
WT_STREAMS_BLOCKED. Flow control for the CONNECT stream as a whole, as provided by the HTTP version in use, applies in addition to any WebTransport-session-level flow control.

WebTransport streams can be aborted using a WT_RESET_STREAM capsule and a receiver can request that a sender stop sending with a WT_STOP_SENDING capsule.

A WebTransport session is terminated when the CONNECT stream that created it is closed. This implicitly closes all WebTransport streams that were multiplexed over that CONNECT stream.

3. Session Establishment

3.1. Establishing a Transport-Capable HTTP/2 Connection

In order to indicate support for WebTransport, both the client and the server MUST send a SETTINGS_WEBTRANSPORT_MAX_SESSIONS value greater than "0" in their SETTINGS frame. Endpoints MUST NOT use any WebTransport-related functionality unless the parameter has been negotiated.

3.2. Extended CONNECT in HTTP/2

[RFC8441] defines an extended CONNECT method in Section 4, enabled by the SETTINGS_ENABLE_CONNECT_PROTOCOL parameter. An endpoint needs to send both SETTINGS_ENABLE_CONNECT_PROTOCOL and SETTINGS_WEBTRANSPORT_MAX_SESSIONS for WebTransport to be enabled.

3.3. Creating a New Session

As WebTransport sessions are established over HTTP, 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 (Section 7.1 of [WEBTRANSPORT-H3]). The :scheme field MUST be https. Both the :authority and the :path value MUST be set; those fields indicate the desired WebTransport server. In a Web context, the request MUST include an Origin header field [ORIGIN] that includes the origin of the site that requested the creation of the session.

Upon receiving an extended CONNECT request with a :protocol field set to webtransport, the HTTP server checks if the identified resource supports WebTransport sessions. If the resource does not, the server SHOULD reply with status code 406 (Section 15.5.7 of [RFC9110]). 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.

A WebTransport session is established when the server sends a 2xx response. A server generates that response from the request header, not from the contents of the request. To enable clients to resend data when attempting to re-establish a session that was rejected by a server, a server MUST NOT process any capsules on the request stream unless it accepts the WebTransport session.

A client MAY optimistically send any WebTransport capsules associated with a CONNECT request, without waiting for a response, to the extent allowed by flow control. This can reduce latency for data sent by a client at the start of a WebTransport session. For example, a client might choose to send datagrams or flow control updates before receiving any response from the server.

3.4. Flow Control

Flow control governs the amount of resources that can be consumed or data that can be sent. WebTransport over HTTP/2 allows a server to limit the number of sessions that a client can create on a single connection; see Section 3.4.1.

For data, there are five applicable levels of flow control for data that is sent or received using WebTransport over HTTP/2:

1. TCP flow control.

2. HTTP/2 connection flow control, which governs the total amount of data in DATA frames for all HTTP/2 streams.

3. HTTP/2 stream flow control, which limits the data on a single HTTP/2 stream. For a WebTransport session, this includes all capsules, including those that are exempt from WebTransport session-level flow control.

4. WebTransport session-level flow control, which limits the total amount of stream data that can be sent or received on streams within the WebTransport session. Note that this does not limit other types of capsules within a WebTransport session, such as control messages or datagrams.

5. WebTransport stream flow control, which limits data on individual streams within a session.
TCP and HTTP/2 define the first three levels of flow control. This document defines the final two.

3.4.1. Limiting the Number of Simultaneous Sessions

This document defines a SETTINGS_WEBTRANSPORT_MAX_SESSIONS parameter that allows the server to limit the maximum number of concurrent WebTransport sessions on a single HTTP/2 connection. The client MUST NOT open more sessions than indicated in the server SETTINGS parameters. The server MUST NOT close the connection if the client opens sessions exceeding this limit, as the client and the server do not have a consistent view of how many sessions are open due to the asynchronous nature of the protocol; instead, it MUST reset all of the CONNECT streams it is not willing to process with the REFUSED_STREAM error code (Section 8.7 of [HTTP2]).

Just like other HTTP requests, WebTransport sessions, and data sent on those sessions, are counted against flow control limits. Servers that wish to limit the rate of incoming requests on any particular session have multiple mechanisms:

* The REFUSED_STREAM error code defined in Section 8.7 of [HTTP2] indicates to the receiving HTTP/2 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.

An endpoint that wishes to reduce the value of SETTINGS_WEBTRANSPORT_MAX_SESSIONS to a value that is below the current number of open sessions can either close sessions that exceed the new value or allow those sessions to complete. Endpoints MUST NOT reduce the value of SETTINGS_WEBTRANSPORT_MAX_SESSIONS to "0" after previously advertising a non-zero value.

3.4.2. Limiting the Number of Streams Within a Session

This document defines a WT_MAX_STREAMS capsule (Section 5.7) that allows each endpoint to limit the number of streams its peer is permitted to open as part of a WebTransport session. There is a separate limit for bidirectional streams and for unidirectional streams. Note that, unlike WebTransport over HTTP/3 [WEBTRANSPORT-H3], because the entire WebTransport session is contained within HTTP/2 DATA frames on a single HTTP/2 stream, this limit is the only mechanism for an endpoint to limit the number of WebTransport streams that its peer can open on a session.
3.4.3. Initial Flow Control Limits

To allow stream data to be exchanged in the same flight as the extended CONNECT request that establishes a WebTransport session, initial flow control limits can be exchanged via HTTP/2 SETTINGS (Section 3.4.3.1). Initial values for the flow control limits can also be exchanged via the WebTransport-Init header field on the extended CONNECT request (Section 3.4.3.2).

The limits communicated via HTTP/2 SETTINGS apply to all WebTransport sessions opened on that HTTP/2 connection. Limits communicated via the WebTransport-Init header field apply only to the WebTransport session established by the extended CONNECT request carrying that field.

If both the SETTINGS and the header field are present when a WebTransport session is established, the endpoint MUST use the greater of the two values for each corresponding initial flow control value. Endpoints sending the SETTINGS and also including the header field SHOULD ensure that the header field values are greater than or equal to the values provided in the SETTINGS.

3.4.3.1. Flow Control SETTINGS

Initial flow control limits can be exchanged via HTTP/2 SETTINGS (Section 9.1) by providing non-zero values for

* WT_MAX_DATA via SETTINGS_WEBTRANSPORT_INITIAL_MAX_DATA
* WT_MAX_STREAM_DATA via SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAM_DATA_UNI and SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAM_DATA_BIDI
* WT_MAX_STREAMS via SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAMS_UNI and SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAMS_BIDI

3.4.3.2. Flow Control Header Field

The WebTransport-Init HTTP header field can be used to communicate the initial values of the flow control windows, similar to how QUIC uses transport parameters. The WebTransport-Init is a Dictionary Structured Field (Section 3.2 of [RFC8941]). If any of the fields cannot be parsed correctly or do not have the correct type, the peer MUST reset the CONNECT stream. The following keys are defined for the WebTransport-Init header field:

u: The initial flow control limit for unidirectional streams opened by the recipient of this header field. MUST be an Integer.
bl: The initial flow control limit for the bidirectional streams opened by the sender of this header field. MUST be an Integer.

br: The initial flow control limit for the bidirectional streams opened by the recipient of this header field. MUST be an Integer.

3.4.4. Flow Control and Intermediaries

WebTransport over HTTP/2 uses several capsules for flow control, and all of these capsules define special intermediary handling as described in Section 3.2 of [HTTP-DATAGRAM]. These capsules, referred to as the "flow control capsules" are WT_MAX_DATA, WT_MAX_STREAM_DATA, WT_MAX_STREAMS, WT_DATA_BLOCKED, WT_STREAM_DATA_BLOCKED, and WT_STREAMS_BLOCKED.

Because flow control in WebTransport is hop-by-hop and does not provide an end-to-end signal, intermediaries MUST consume flow control signals and express their own flow control limits to the next hop. The intermediary can send these signals via HTTP/3 flow control messages, HTTP/2 flow control messages, or as WebTransport flow control capsules, where appropriate. Intermediaries are responsible for storing any data for which they advertise flow control credit if that data cannot be immediately forwarded to the next hop.

In practice, an intermediary that translates flow control signals between similar WebTransport protocols, such as between two HTTP/2 connections, can often simply reexpress the same limits received on one connection directly on the other connection.

An intermediary that does not want to be responsible for storing data that cannot be immediately sent on its translated connection would ensure that it does not advertise a higher flow control limit on one connection than the corresponding limit on the translated connection.

4. WebTransport Features

WebTransport over TCP-based HTTP semantics provides the following features described in [OVERVIEW]: unidirectional streams, bidirectional streams, and datagrams, initiated by either endpoint.

WebTransport streams and datagrams that belong to different WebTransport sessions are identified by the CONNECT stream on which they are transmitted, with one WebTransport session consuming one CONNECT stream.
4.1. Transport Properties

The WebTransport framework [OVERVIEW] defines a set of optional transport properties that clients can use to determine the presence of features which might allow additional optimizations beyond the common set of properties available via all WebTransport protocols.

Because WebTransport over TCP-based HTTP semantics relies on the underlying protocols to provide in order and reliable delivery, there are some notable differences from the way in which QUIC handles application data. For example, endpoints send stream data in order. As there is no ordering mechanism available for the receiver to reassemble incoming data, receivers assume that all data arriving in WT_STREAM capsules is contiguous and in order.

Below are details about support in WebTransport over HTTP/2 for the properties defined by the WebTransport framework.

Stream Independence: WebTransport over HTTP/2 does not support stream independence, as HTTP/2 inherently has head-of-line blocking.

Partial Reliability: WebTransport over HTTP/2 does not support partial reliability, as HTTP/2 retransmits any lost data. This means that any datagrams sent via WebTransport over HTTP/2 will be retransmitted regardless of the preference of the application. The receiver is permitted to drop them, however, if it is unable to buffer them.

Pooling Support: WebTransport over HTTP/2 supports pooling, as multiple transports using WebTransport over HTTP/2 may share the same underlying HTTP/2 connection and therefore share a congestion controller and other transport context. Note that WebTransport streams over HTTP/2 are contained within a single HTTP/2 stream and do not compete with other pooled WebTransport sessions for per-stream resources.

Connection Mobility: WebTransport over HTTP/2 does not support connection mobility, unless an underlying transport protocol that supports multipath or migration, such as MPTCP [MPTCP], is used underneath HTTP/2 and TLS. Without such support, WebTransport over HTTP/2 connections cannot survive network transitions.
4.2. WebTransport Streams

WebTransport streams have identifiers and states that mirror the identifiers (Section 2.1 of [RFC9000]) and states (Section 3 of [RFC9000]) of QUIC streams as closely as possible to aid in ease of implementation.

WebTransport streams are identified by a numeric value, or stream ID. Stream IDs are only meaningful within the WebTransport session in which they were created. They share the same semantics as QUIC stream IDs, with client initiated streams having even-numbered stream IDs and server-initiated streams having odd-numbered stream IDs. Similarly, they can be bidirectional or unidirectional, indicated by the second least significant bit of the stream ID.

Because WebTransport does not provide an acknowledgement mechanism for WebTransport capsules, it relies on the underlying transport’s in order delivery to inform stream state transitions. Wherever QUIC relies on receiving an ack for a packet to transition between stream states, WebTransport performs that transition immediately.

5. WebTransport Capsules

WebTransport capsules mirror their QUIC frame counterparts as closely as possible to enable maximal reuse of any applicable QUIC infrastructure by implementors.

WebTransport capsules use the Capsule Protocol defined in Section 3.2 of [HTTP-DATAGRAM].

5.1. PADDING Capsule

A PADDING capsule is an HTTP capsule [HTTP-DATAGRAM] of type=0x190B4D38 and has no semantic value. PADDING capsules can be used to introduce additional data between other HTTP datagrams and can also be used to provide protection against traffic analysis or for other reasons.

Note that, when used with WebTransport over HTTP/2, the PADDING capsule exists alongside the ability to pad HTTP/2 frames (Section 10.7 of [RFC9113]). HTTP/2 padding is hop-by-hop and can be modified by intermediaries, while the PADDING capsule traverses intermediaries. The PADDING capsule is also constrained to be no smaller than the capsule overhead itself.
The Padding field MUST be set to an all-zero sequence of bytes of any length as specified by the Length field.

5.2. WT_RESET_STREAM Capsule

A WT_RESET_STREAM capsule is an HTTP capsule [HTTP-DATAGRAM] of type=0x190B4D39 and allows either endpoint to abruptly terminate the sending part of a WebTransport stream.

After sending a WT_RESET_STREAM capsule, an endpoint ceases transmission of WT_STREAM capsules on the identified stream. A receiver of a WT_RESET_STREAM capsule can discard any data that it already received on that stream.

WT_RESET_STREAM Capsule {
    Type (i) = 0x190B4D39,
    Length (i),
    Stream ID (i),
    Application Protocol Error Code (i),
}

The WT_RESET_STREAM capsule defines the following fields:

Stream ID: A variable-length integer encoding of the WebTransport stream ID of the stream being terminated.

Application Protocol Error Code: A variable-length integer containing the application protocol error code that indicates why the stream is being closed.

Unlike the equivalent QUIC frame, this capsule does not include a Final Size field. In-order delivery of WT_STREAM capsules ensures that the amount of session-level flow control consumed by a stream is always known by both endpoints.
5.3. WT_STOP_SENDING Capsule

An HTTP capsule [HTTP-DATAGRAM] called WT_STOP_SENDING (type=0x190B4D3A) is introduced to communicate that incoming data is being discarded on receipt per application request. WT_STOP_SENDING requests that a peer cease transmission on a WebTransport stream.

WT_STOP_SENDING Capsule {
  Type (i) = 0x190B4D3A,
  Length (i),
  Stream ID (i),
  Application Protocol Error Code (i),
}

Figure 3: WT_STOP_SENDING Capsule Format

The WT_STOP_SENDING capsule defines the following fields:

- **Stream ID**: A variable-length integer carrying the WebTransport stream ID of the stream being ignored.
- **Application Protocol Error Code**: A variable-length integer containing the application-specified reason the sender is ignoring the stream.

5.4. WT_STREAM Capsule

WT_STREAM capsules implicitly create a WebTransport stream and carry stream data.

The Type field in the WT_STREAM capsule is either 0x190B4D3B or 0x190B4D3C. The least significant bit in the capsule type is the FIN bit (0x01), indicating when set that the capsule marks the end of the stream in one direction. Stream data consists of any number of 0x190B4D3B capsules followed by a terminal 0x190B4D3C capsule.

WT_STREAM Capsule {
  Type (i) = 0x190B4D3B..0x190B4D3C,
  Length (i),
  Stream ID (i),
  Stream Data (..),
}

Figure 4: WT_STREAM Capsule Format

WT_STREAM capsules contain the following fields:

- **Stream ID**: The stream ID for the stream.
Stream Data: Zero or more bytes of data for the stream. Empty WT_STREAM capsules MUST NOT be used unless they open or close a stream; an endpoint MAY treat an empty WT_STREAM capsule that neither starts nor ends a stream as a session error.

5.5. WT_MAX_DATA Capsule

An HTTP capsule [HTTP-DATAGRAM] called WT_MAX_DATA (type=0x190B4D3D) is introduced to inform the peer of the maximum amount of data that can be sent on the WebTransport session as a whole.

WT_MAX_DATA Capsule {
  Type (i) = 0x190B4D3D,
  Length (i),
  Maximum Data (i),
}

Figure 5: WT_MAX_DATA Capsule Format

WT_MAX_DATA capsules contain the following field:

Maximum Data: A variable-length integer indicating the maximum amount of data that can be sent on the entire connection, in units of bytes.

All data sent in WT_STREAM capsules counts toward this limit. The sum of the lengths of Stream Data fields in WT_STREAM capsules MUST NOT exceed the value advertised by a receiver.

The WT_MAX_DATA capsule defines special intermediary handling, as described in Section 3.2 of [HTTP-DATAGRAM]. Intermedaries MUST consume WT_MAX_DATA capsules for flow control purposes and MUST generate and send appropriate flow control signals for their limits; see Section 3.4.4.

The initial value for this limit MAY be communicated by sending a non-zero value for SETTINGS_WEBTRANSPORT_INITIAL_MAX_DATA.

5.6. WT_MAX_STREAM_DATA Capsule

An HTTP capsule [HTTP-DATAGRAM] called WT_MAX_STREAM_DATA (type=0x190B4D3E) is introduced to inform a peer of the maximum amount of data that can be sent on a WebTransport stream.
WT_MAX_STREAM_DATA Capsule {
    Type (i) = 0x190B4D3E,
    Length (i),
    Stream ID (i),
    Maximum Stream Data (i),
}

Figure 6: WT_MAX_STREAM_DATA Capsule Format

WT_MAX_STREAM_DATA capsules contain the following fields:

Stream ID: The stream ID of the affected WebTransport stream, encoded as a variable-length integer.

Maximum Stream Data: A variable-length integer indicating the maximum amount of data that can be sent on the identified stream, in units of bytes.

All data sent in WT_STREAM capsules for the identified stream counts toward this limit. The sum of the lengths of Stream Data fields in WT_STREAM capsules on the identified stream MUST NOT exceed the value advertised by a receiver.

The WT_MAX_STREAM_DATA capsule defines special intermediary handling, as described in Section 3.2 of [HTTP-DATAGRAM]. Intermedaries MUST consume WT_MAX_STREAM_DATA capsules for flow control purposes and MUST generate and send appropriate flow control signals for their limits; see Section 3.4.4.

Initial values for this limit for unidirectional and bidirectional streams MAY be communicated by sending non-zero values for SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAM_DATA_UNI and SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAM_DATA_BIDI respectively.

5.7. WT_MAX_STREAMS Capsule

An HTTP capsule [HTTP-DATAGRAM] called WT_MAX_STREAMS is introduced to inform the peer of the cumulative number of streams of a given type it is permitted to open. A WT_MAX_STREAMS capsule with a type of 0x190B4D3F applies to bidirectional streams, and a WT_MAX_STREAMS capsule with a type of 0x190B4D40 applies to unidirectional streams.

Note that, because Maximum Streams is a cumulative value representing the total allowed number of streams, including previously closed streams, endpoints repeatedly send new WT_MAX_STREAMS capsules with increasing Maximum Streams values as streams are opened.
WT_MAX_STREAMS Capsule {
    Type (i) = 0x190B4D3F..0x190B4D40,
    Length (i),
    Maximum Streams (i),
}

Figure 7: WT_MAX_STREAMS Capsule Format

WT_MAX_STREAMS capsules contain the following field:

Maximum Streams: A count of the cumulative number of streams of the corresponding type that can be opened over the lifetime of the connection. This value cannot exceed $2^{60}$, as it is not possible to encode stream IDs larger than $2^{62}-1$.

An endpoint MUST NOT open more streams than permitted by the current stream limit set by its peer. For instance, a server that receives a unidirectional stream limit of 3 is permitted to open streams 3, 7, and 11, but not stream 15.

Note that this limit includes streams that have been closed as well as those that are open.

The WT_MAX_STREAMS capsule defines special intermediary handling, as described in Section 3.2 of [HTTP-DATAGRAM]. Intermedaries MUST consume WT_MAX_STREAMS capsules for flow control purposes and MUST generate and send appropriate flow control signals for their limits.

Initial values for these limits MAY be communicated by sending non-zero values for SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAMS_UNI and SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAMS_BIDI.

5.8. WT_DATA_BLOCKED Capsule

A sender SHOULD send a WT_DATA_BLOCKED capsule (type=0x190B4D41) when it wishes to send data but is unable to do so due to WebTransport session-level flow control. WT_DATA_BLOCKED capsules can be used as input to tuning of flow control algorithms.

WT_DATA_BLOCKED Capsule {
    Type (i) = 0x190B4D41,
    Length (i),
    Maximum Data (i),
}

Figure 8: WT_DATA_BLOCKED Capsule Format

WT_DATA_BLOCKED capsules contain the following field:
Maximum Data: A variable-length integer indicating the session-level limit at which blocking occurred.

The WT_DATA_BLOCKED capsule defines special intermediary handling, as described in Section 3.2 of [HTTP-DATAGRAM]. Intermedaries MUST consume WT_DATA_BLOCKED capsules for flow control purposes and MUST generate and send appropriate flow control signals for their limits; see Section 3.4.4.

5.9. WT_STREAM_DATA_BLOCKED Capsule

A sender SHOULD send a WT_STREAM_DATA_BLOCKED capsule (type=0x190B4D42) when it wishes to send data but is unable to do so due to stream-level flow control. This capsule is analogous to WT_DATA_BLOCKED.

WT_STREAM_DATA_BLOCKED Capsule {
  Type (i) = 0x190B4D42,
  Length (i),
  Stream ID (i),
  Maximum Stream Data (i),
}

Figure 9: WT_STREAM_DATA_BLOCKED Capsule Format

WT_STREAM_DATA_BLOCKED capsules contain the following fields:

Stream ID: A variable-length integer indicating the WebTransport stream that is blocked due to flow control.

Maximum Stream Data: A variable-length integer indicating the offset of the stream at which the blocking occurred.

The WT_STREAM_DATA_BLOCKED capsule defines special intermediary handling, as described in Section 3.2 of [HTTP-DATAGRAM]. Intermedaries MUST consume WT_STREAM_DATA_BLOCKED capsules for flow control purposes and MUST generate and send appropriate flow control signals for their limits; see Section 3.4.4.

5.10. WT_STREAMS_BLOCKED Capsule

A sender SHOULD send a WT_STREAMS_BLOCKED capsule (type=0x190B4D43 or 0x190B4D44) when it wishes to open a stream but is unable to do so due to the maximum stream limit set by its peer. A WT_STREAMS_BLOCKED capsule of type 0x190B4D43 is used to indicate reaching the bidirectional stream limit, and a STREAMS_BLOCKED capsule of type 0x190B4D44 is used to indicate reaching the unidirectional stream limit.
A WT_STREAMS_BLOCKED capsule does not open the stream, but informs the peer that a new stream was needed and the stream limit prevented the creation of the stream.

WT_STREAMS_BLOCKED Capsule {
    Type (i) = 0x190B4D43..0x190B4D44,
    Length (i),
    Maximum Streams (i),
}

Figure 10: WT_STREAMS_BLOCKED Capsule Format

WT_STREAMS_BLOCKED capsules contain the following field:

Maximum Streams: A variable-length integer indicating the maximum number of streams allowed at the time the capsule was sent. This value cannot exceed 2^60, as it is not possible to encode stream IDs larger than 2^62-1.

The WT_STREAMS_BLOCKED capsule defines special intermediary handling, as described in Section 3.2 of [HTTP-DATAGRAM]. Intermedaries MUST consume WT_STREAMS_BLOCKED capsules for flow control purposes and MUST generate and send appropriate flow control signals for their limits.

5.11. DATAGRAM Capsule

WebTransport over HTTP/2 uses the DATAGRAM capsule defined in Section 3.5 of [HTTP-DATAGRAM] to carry datagram traffic.

DATAGRAM Capsule {
    Type (i) = 0x00,
    Length (i),
    HTTP Datagram Payload (..),
}

Figure 11: DATAGRAM Capsule Format

When used in WebTransport over HTTP/2, DATAGRAM capsules contain the following fields:

HTTP Datagram Payload: The content of the datagram to be delivered.

The data in DATAGRAM capsules is not subject to flow control. The receiver MAY discard this data if it does not have sufficient space to buffer it.
An intermediary could forward the data in a DATAGRAM capsule over another protocol, such as WebTransport over HTTP/3. 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.

Section 3.5 of [HTTP-DATAGRAM] indicates that intermediaries that forward DATAGRAM capsules where QUIC datagrams [DATAGRAM] are available forward the contents of the capsule as native QUIC datagrams, rather than as HTTP datagrams in a DATAGRAM capsule. Similarly, when forwarding DATAGRAM capsules used as part of a WebTransport over HTTP/2 session on a WebTransport session that natively supports QUIC datagrams, such as WebTransport over HTTP/3 [WEBTRANSPORT-H3], intermediaries follow the requirements in [WEBTRANSPORT-H3] to use native QUIC datagrams.

6. Examples

An example of negotiating a WebTransport Stream on an HTTP/2 connection follows. This example is intended to closely follow the example in Section 5.1 of [RFC8441] to help illustrate the differences defined in this document.
An example of the server initiating a WebTransport Stream follows. The only difference here is the endpoint that sends the first WT_STREAM capsule.
7. Session Termination

An WebTransport session over HTTP/2 is terminated when either endpoint closes the stream associated with the CONNECT request that initiated the session. Upon learning about the session being terminated, the endpoint MUST stop sending new datagrams and reset all of the streams associated with the session.

8. Security Considerations

WebTransport over HTTP/2 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/2 requires explicit opt-in through the use of HTTP SETTINGS; this avoids potential protocol confusion attacks by ensuring the HTTP/2 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/2, 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.

9. IANA Considerations

9.1. HTTP/2 SETTINGS Parameter Registration

The following entries are added to the "HTTP/2 Settings" registry established by [RFC7540):

The SETTINGS_WEBTRANSPORT_MAX_SESSIONS parameter indicates that the specified HTTP/2 connection is WebTransport-capable and the number of concurrent sessions an endpoint is willing to receive. The default value for the SETTINGS_WEBTRANSPORT_MAX_SESSIONS parameter is "0", meaning that the endpoint is not willing to receive any WebTransport sessions.

Setting Name: WEBTRANSPORT_MAX_SESSIONS

Value: 0x2b60

Default: 0

Specification: This document

The SETTINGS_WEBTRANSPORT_INITIAL_MAX_DATA parameter indicates the initial value for the session data limit, otherwise communicated by the WT_MAX_DATA capsule (see Section 5.5). The default value for the SETTINGS_WEBTRANSPORT_INITIAL_MAX_DATA parameter is "0", indicating that the endpoint needs to send a WT_MAX_DATA capsule within each session before its peer is allowed to send any stream data within that session.
Note that this limit applies to all WebTransport sessions that use the HTTP/2 connection on which this SETTING is sent.

Setting Name: SETTINGS_WEBTRANSPORT_INITIAL_MAX_DATA

Value: 0x2b61

Default: 0

Specification: This document

The SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAM_DATA_UNI parameter indicates the initial value for the stream data limit for incoming unidirectional streams, otherwise communicated by the WT_MAX_STREAM_DATA capsule (see Section 5.6). The default value for the SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAM_DATA_UNI parameter is "0", indicating that the endpoint needs to send WT_MAX_STREAM_DATA capsules for each stream within each individual WebTransport session before its peer is allowed to send any stream data on those streams.

Note that this limit applies to all WebTransport streams on all sessions that use the HTTP/2 connection on which this SETTING is sent.

Setting Name: SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAM_DATA_UNI

Value: 0x2b62

Default: 0

Specification: This document

The SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAM_DATA_BIDI parameter indicates the initial value for the stream data limit for incoming data on bidirectional streams, otherwise communicated by the WT_MAX_STREAM_DATA capsule (see Section 5.6). The default value for the SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAM_DATA_BIDI parameter is "0", indicating that the endpoint needs to send WT_MAX_STREAM_DATA capsules for each stream within each individual WebTransport session before its peer is allowed to send any stream data on those streams.

Note that this limit applies to all WebTransport streams on all sessions that use the HTTP/2 connection on which this SETTING is sent.

Setting Name: SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAM_DATA_BIDI

Value: 0x2b63
The SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAMS_UNI parameter indicates the initial value for the unidirectional max stream limit, otherwise communicated by the WT_MAX_STREAMS capsule (see Section 5.7). The default value for the SETTINGS_WEBTRANSPORTInicial_MAX_STREAMS_UNI parameter is "0", indicating that the endpoint needs to send WT_MAX_STREAMS capsules on each individual WebTransport session before its peer is allowed to create any unidirectional streams within that session.

Note that this limit applies to all WebTransport sessions that use the HTTP/2 connection on which this SETTING is sent.

Setting Name: SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAMS_UNI
Value: 0x2b64
Default: 0

The SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAMS_BIDI parameter indicates the initial value for the bidirectional max stream limit, otherwise communicated by the WT_MAX_STREAMS capsule (see Section 5.7). The default value for the SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAMS_BIDI parameter is "0", indicating that the endpoint needs to send WT_MAX_STREAMS capsules on each individual WebTransport session before its peer is allowed to create any bidirectional streams within that session.

Note that this limit applies to all WebTransport sessions that use the HTTP/2 connection on which this SETTING is sent.

Setting Name: SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAMS_BIDI
Value: 0x2b65
Default: 0

9.2. Capsule Types

The following entries are added to the "HTTP Capsule Types" registry established by [HTTP-DATAGRAM]:

The PADDING capsule.

Value: 0x190B4D38
Capsule Type: PADDING
Status: permanent
Specification: This document
Change Controller: IETF
Contact: WebTransport Working Group webtransport@ietf.org
        (mailto:webtransport@ietf.org)
Notes: None

The WT_RESET_STREAM capsule.

Value: 0x190B4D39
Capsule Type: WT_RESET_STREAM
Status: permanent
Specification: This document
Change Controller: IETF
Contact: WebTransport Working Group webtransport@ietf.org
        (mailto:webtransport@ietf.org)
Notes: None

The WT_STOP_SENDING capsule.

Value: 0x190B4D3A
Capsule Type: WT_STOP_SENDING
Status: permanent
Specification: This document
Change Controller: IETF
Contact: WebTransport Working Group webtransport@ietf.org
        (mailto:webtransport@ietf.org)
Notes: None

The WT_STREAM capsule.

Value: 0x190B4D3B..0x190B4D3C
Capsule Type: WT_STREAM
Status: permanent
Specification: This document
Change Controller: IETF
Contact: WebTransport Working Group webtransport@ietf.org
        (mailto:webtransport@ietf.org)
Notes: None

The WT_MAX_DATA capsule.

Value: 0x190B4D3D
Capsule Type: WT_MAX_DATA
The WT_MAX_STREAM_DATA capsule.

Value: 0x190B4D3E
Capsule Type: WT_MAX_STREAM_DATA
Status: permanent
Specification: This document
Change Controller: IETF
Contact: WebTransport Working Group webtransport@ietf.org
Notes: None

The WT_MAX_STREAMS capsule.

Value: 0x190B4D3F..0x190B4D40
Capsule Type: WT_MAX_STREAMS
Status: permanent
Specification: This document
Change Controller: IETF
Contact: WebTransport Working Group webtransport@ietf.org
Notes: None

The WT_DATA_BLOCKED capsule.

Value: 0x190B4D41
Capsule Type: WT_DATA_BLOCKED
Status: permanent
Specification: This document
Change Controller: IETF
Contact: WebTransport Working Group webtransport@ietf.org
Notes: None

The WT_STREAM_DATA_BLOCKED capsule.

Value: 0x190B4D42
Capsule Type: WT_STREAM_DATA_BLOCKED
Status: permanent
Specification: This document
Change Controller: IETF
Contact: WebTransport Working Group webtransport@ietf.org
The WT_STREAMS_BLOCKED capsule.

Value: 0x190B4D43..0x190B4D44
Capsule Type: WT_STREAMS_BLOCKED
Status: permanent
Specification: This document
Change Controller: IETF
Contact: WebTransport Working Group webtransport@ietf.org
Notes: None

9.3. HTTP Header Field Name

IANA will register the following entry in the "Hypertext Transfer Protocol (HTTP) Field Name Registry" maintained at
https://www.iana.org/assignments/http-fields
(https://www.iana.org/assignments/http-fields):

Field Name: WebTransport-Init
Template: None
Status: permanent
Reference: This document
Comments: None

10. References

10.1. Normative References


DOI 10.17487/RFC9113, June 2022,

[ORIGIN]  Barth, A., "The Web Origin Concept", RFC 6454,
DOI 10.17487/RFC6454, December 2011,

in Progress, Internet-Draft, draft-ietf-webtrans-overview-06, 6 September 2023,

[RFC2119]  Bradner, S., "Key words for use in RFCs to Indicate
Requirement Levels", BCP 14, RFC 2119,
DOI 10.17487/RFC2119, March 1997,

Codes", RFC 6585, DOI 10.17487/RFC6585, April 2012,

Protocol (HTTP/1.1): Message Syntax and Routing",
RFC 7230, DOI 10.17487/RFC7230, June 2014,

Transfer Protocol Version 2 (HTTP/2)", RFC 7540,
DOI 10.17487/RFC7540, May 2015,

[RFC8174]  Leiba, B., "Ambiguity of Uppercase vs Lowercase in RFC
2119 Key Words", BCP 14, RFC 8174, DOI 10.17487/RFC8174,

[RFC8441]  McManus, P., "Bootstrapping WebSockets with HTTP/2",
RFC 8441, DOI 10.17487/RFC8441, September 2018,

HTTP", RFC 8941, DOI 10.17487/RFC8941, February 2021,

10.2. Informative References


Acknowledgments

Thanks to Anthony Chivetta, Joshua Otto, and Valentin Pistol for their contributions in the design and implementation of this work.

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P S W

P

PADDING  Section 5.1, Paragraph 1; Section 5.1, Paragraph 1; Section 5.1, Paragraph 2; Section 5.1, Paragraph 2; Section 5.1, Paragraph 2; Section 9.2, Paragraph 3.4.1

S

SETTINGS_WEBTRANSPORT_INITIAL_MAX_DATA  Section 3.4.3.1, Paragraph 2.1.1; Section 5.5, Paragraph 7; Section 9.1, Paragraph 4; Section 9.1, Paragraph 4; Section 9.1, Paragraph 6.2.1

SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAM_DATA_BIDI  Section 3.4.3.1, Paragraph 2.2.1; Section 5.6, Paragraph 7; Section 9.1, Paragraph 10; Section 9.1, Paragraph 10; Section 9.1, Paragraph 12.2.1

SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAM_DATA_UNI  Section 3.4.3.1, Paragraph 2.2.1; Section 5.6, Paragraph 7; Section 9.1, Paragraph 7; Section 9.1, Paragraph 7; Section 9.1, Paragraph 9.2.1

SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAMS_BIDI  Section 3.4.3.1, Paragraph 2.3.1; Section 5.7, Paragraph 9; Section 9.1, Paragraph 16; Section 9.1, Paragraph 16; Section 9.1, Paragraph 18.2.1

SETTINGS_WEBTRANSPORT_INITIAL_MAX_STREAMS_UNI  Section 3.4.3.1, Paragraph 2.3.1; Section 5.7, Paragraph 9; Section 9.1, Paragraph 13; Section 9.1, Paragraph 13; Section 9.1, Paragraph 15.2.1

W

WT_DATA_BLOCKED  Section 2, Paragraph 8; Section 3.4.4,
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WebTransport over HTTP/3

draft-ietf-webtrans-http3-08

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), which is archived at <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/>.

Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on 25 April 2024.
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_WEBTRANSPORT_MAX_SESSIONS setting in order to indicate that they both support WebTransport over HTTP/3. This process also negotiates the use of additional HTTP/3 extensions.

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 and transfer its ownership to WebTransport by providing a special signal in the first bytes.

* A server can create a bidirectional stream and transfer its ownership to WebTransport by providing a special signal in the first bytes.

* Both client and server can create a unidirectional stream using a special stream type.

* A datagram can be sent using HTTP Datagrams [HTTP-DATAGRAM].

A 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_WEBTRANSPORT_MAX_SESSIONS value greater than "0" in their SETTINGS frame. The default value for the SETTINGS_WEBTRANSPORT_MAX_SESSIONS parameter is "0", meaning that the endpoint is not willing to receive any WebTransport sessions. Note that the client only needs to send a value greater than "0"; since clients initiate WebTransport sessions, the actual value is not significant.

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

[[RFC editor: please remove the following paragraph before publication.]]

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 the WebTransport extension that is different from the one used by the server.

Because WebTransport over HTTP/3 requires support for HTTP/3 datagrams and the Capsule Protocol, both the client and the server MUST indicate support for HTTP/3 datagrams by sending a SETTINGS_H3_DATAGRAM value set to 1 in their SETTINGS frame (see Section 2.1.1 of [HTTP-DATAGRAM]).
WebTransport over HTTP/3 also requires support for QUIC datagrams. To indicate support, both the client and the server MUST send a max_datagram_frame_size transport parameter with a value greater than 0 (see Section 3 of [QUIC-DATAGRAM]).

3.2. Extended CONNECT in HTTP/3

[RFC8441] defines an extended CONNECT method in Section 4, enabled by the SETTINGS_ENABLE_CONNECT_PROTOCOL setting. That setting is defined for HTTP/3 by [RFC9220]. A client supporting WebTransport over HTTP/3 MUST send the SETTINGS_WEBTRANSPORT_MAX_SESSIONS setting with a value greater than "0". A server supporting WebTransport over HTTP/3 MUST send both the SETTINGS_WEBTRANSPORT_MAX_SESSIONS setting with a value greater than "0" and the SETTINGS_ENABLE_CONNECT_PROTOCOL setting with a value of "1".

3.3. Creating a New Session

As WebTransport sessions are established over HTTP/3, they are identified using the https URI scheme ([HTTP], Section 4.2.2).

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. If the WebTransport session is coming from a browser client, an Origin header [RFC6454] MUST be provided within the request; otherwise, the header is OPTIONAL.

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 15.5.5 of [HTTP]). When the request contains the Origin header, the WebTransport server MUST verify the Origin header to ensure that the specified origin is allowed to access the server in question. If the verification fails, the WebTransport server SHOULD reply with status code 403 (Section 15.5.4 of [HTTP]). If all checks pass, the WebTransport server MAY accept the session by replying with a 2xx series status code, as defined in Section 15.3 of [HTTP].

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.
The server may reply with a 3xx response, indicating a redirection (Section 15.4 of [HTTP]). The user agent MUST NOT automatically follow such redirects, as the client could potentially already have sent data for the WebTransport session in question; it MAY notify the client about the redirect.

Clients cannot initiate WebTransport in 0-RTT packets, as the CONNECT method is not considered safe; see Section 10.9 of [HTTP3]. However, WebTransport-related SETTINGS parameters may be retained from the previous session as described in Section 7.2.4.2 of [HTTP3]. If the server accepts 0-RTT, the server MUST NOT reduce the limit of maximum open WebTransport sessions from the one negotiated during the previous session; such change would be deemed incompatible, and MUST result in a H3_SETTINGS_ERROR connection error.

The webtransport HTTP Upgrade Token uses the Capsule Protocol as defined in [HTTP-DATAGRAM]. The Capsule Protocol is negotiated when the server sends a 2xx response. The capsule-protocol header field Section 3.4 of [HTTP-DATAGRAM] is not required by WebTransport and can safely be ignored by WebTransport endpoints.

3.4. Limiting the Number of Simultaneous Sessions

This document defines a SETTINGS_WEBTRANSPORT_MAX_SESSIONS parameter that allows the server to limit the maximum number of concurrent WebTransport sessions on a single HTTP/3 connection. The client MUST NOT open more sessions than indicated in the server SETTINGS parameters. The server MUST NOT close the connection if the client opens sessions exceeding this limit, as the client and the server do not have a consistent view of how many sessions are open due to the asynchronous nature of the protocol; instead, it MUST reset all of the CONNECT streams it is not willing to process with the HTTP_REQUEST_REJECTED status defined in [HTTP3].

Just like other HTTP requests, WebTransport sessions, and data sent on those sessions, are counted against flow control limits. This document does not introduce additional mechanisms for endpoints to limit the relative amount of flow control credit consumed by different WebTransport sessions, however servers that wish to limit the rate of incoming requests on any particular session have alternative mechanisms:

* The 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.

3.5. Prioritization

WebTransport sessions are initiated using extended CONNECT. While Section 11 of [RFC9218] describes how extensible priorities can be applied to data sent on a CONNECT stream, WebTransport extends the types of data that are exchanged in relation to the request and response, which requires additional considerations.

WebTransport CONNECT requests and responses MAY contain the Priority header field (Section 5 of [RFC9218]); clients MAY reprioritize by sending PRIORITY_UPDATE frames (Section 7 of [RFC9218]). In extension to [RFC9218], it is RECOMMENDED that clients and servers apply the scheduling guidance in both Section 9 of [RFC9218] and Section 10 of [RFC9218] for all data that they send in the enclosing WebTransport session, including Capsules, WebTransport streams and datagrams. WebTransport does not provide any priority signaling mechanism for streams and datagrams within a WebTransport session; such mechanisms can be defined by application protocols using WebTransport. It is RECOMMENDED that such mechanisms only affect scheduling within a session and not scheduling of other data on the same HTTP/3 connection.

The client/server priority merging guidance given in Section 8 of [RFC9218] also applies to WebTransport session. For example, a client that receives a response Priority header field could alter its view of a WebTransport session priority and alter the scheduling of outgoing data as a result.

Endpoints that prioritize WebTransport sessions need to consider how they interact with other sessions or requests on the same HTTP/3 connection.

4. WebTransport Features

WebTransport over HTTP/3 provides the following features described in [OVERVIEW]: unidirectional streams, bidirectional streams and datagrams, initiated by either endpoint. Protocols designed for use with WebTransport over HTTP/3 are constrained to these features. The Capsule Protocol is an implementation detail of WebTransport over HTTP/3 and is not a WebTransport feature.
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].

The client MAY optimistically open unidirectional and bidirectional streams, as well as send datagrams, for a session that it has sent the CONNECT request for, even if it has not yet received the server’s response to the request. On the server side, opening streams and sending datagrams is possible as soon as the CONNECT request has been received.

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

WebTransport endpoints can initiate 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 data (Figure 1).

Unidirectional Stream {
  Stream Type (i) = 0x54,
  Session ID (i),
  Stream Body (..)
}

Figure 1: Unidirectional WebTransport stream format

4.2. Bidirectional Streams

All client-initiated bidirectional streams are reserved by HTTP/3 as request streams, which are a sequence of HTTP/3 frames with a variety of rules; see Sections 4.1 and 6.1 of [HTTP3].

WebTransport extends HTTP/3 to allow clients to declare and use alternative request stream rules. Once a client receives settings indicating WebTransport support (Section 3.1), it can send a special signal value, encoded as a variable-length integer, as the first bytes of the stream in order to indicate how the remaining bytes on the stream are used.

WebTransport extends HTTP/3 by defining rules for all server-initiated bidirectional streams. Once a server receives settings indicating WebTransport support (Section 3.1), it can open a
bidirectional stream and SHALL send a special signal value, encoded as a variable-length integer, as the first bytes of the stream in order to indicate how the remaining bytes on the stream are used.

The signal value, 0x41, is used by clients and servers to open a bidirectional WebTransport stream. Following this is the associated session ID, encoded as a variable-length integer; the rest of the stream is the application payload of the WebTransport stream (Figure 2).

```
Bidirectional Stream {
  Signal Value (i) = 0x41,
  Session ID (i),
  Stream Body (..)
}
```

Figure 2: Bidirectional WebTransport stream format

This document reserves the special signal value 0x41 as a WEBTRANSPORT_STREAM frame type. While it is registered as an HTTP/3 frame type to avoid collisions, WEBTRANSPORT_STREAM is not a proper HTTP/3 frame, as it lacks length; it is an extension of HTTP/3 frame syntax that MUST be supported by any peer negotiating WebTransport. Endpoints that implement this extension are also subject to additional frame handling requirements.Endpoints MUST NOT send WEBTRANSPORT_STREAM as a frame type on HTTP/3 streams other than the very first bytes of a request stream. Receiving this frame type in any other circumstances MUST be treated as a connection error of type H3_FRAME_ERROR.

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 32-bit integer, assuming values between 0x00000000 and 0xffffffff. WebTransport implementations SHALL remap those error codes into the error range reserved for WEBTRANSPORT_APPLICATION_ERROR, where 0x00000000 corresponds to 0x52e4a40fa8db, and 0xffffffff corresponds to 0x52e5ac983162. 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 skipped when mapping the error codes (i.e. the two HTTP/3 error codepoints adjacent to a reserved codepoint would map to two adjacent WebTransport application error codepoints). An example pseudocode can be seen in Figure 3.

```python
first = 0x52e4a40fa8db
last = 0x52e5ac983162

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 - floor(shifted / 0x1f)

Figure 3: Pseudocode for converting between WebTransport application errors and HTTP/3 error codes
```

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. The WebTransport datagram payload is sent unmodified in the "HTTP Datagram Payload" field of an HTTP Datagram (Section 2.1 of [HTTP-DATAGRAM]). Note that the payload field directly follows the Quarter Stream ID field, which is at the start of the QUIC DATAGRAM frame payload and refers to the CONNECT stream that established the WebTransport session.

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 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.

4.6. Interaction with HTTP/3 GOAWAY frame

HTTP/3 defines a graceful shutdown mechanism (Section 5.2 of [HTTP3]) that allows a peer to send a GOAWAY frame indicating that it will no longer accept any new incoming requests or pushes.

A client receiving GOAWAY cannot initiate CONNECT requests for new WebTransport sessions if the stream identifier is equal to or greater than the indicated stream ID.

An HTTP/3 GOAWAY frame is also a signal to applications to initiate shutdown for all WebTransport sessions. To shut down a single WebTransport session, either endpoint can send a DRAIN_WEBTRANSPORT_SESSION (0x78ae) capsule.
DRAIN_WEBTRANSPORT_SESSION Capsule {
    Type (i) = DRAIN_WEBTRANSPORT_SESSION,
    Length (i) = 0
}

After sending or receiving either a DRAIN_WEBTRANSPORT_SESSION capsule or a HTTP/3 GOAWAY frame, an endpoint MAY continue using the session and MAY open new streams. The signal is intended for the application using WebTransport, which is expected to attempt to gracefully terminate the session as soon as possible.

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 the send side and abort reading on the receive side of all of the streams associated with the session (see Section 2.4 of [RFC9000]) using the WEBTRANSPORT_SESSION_GONE error code; 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.
An endpoint that sends a CLOSE_WEBTRANSPORT_SESSION capsule MUST immediately send a FIN. The endpoint MAY send a STOP_SENDING to indicate it is no longer reading from the CONNECT stream. The recipient MUST close the stream upon receiving a FIN. If any additional stream data is received on the CONNECT stream after receiving a CLOSE_WEBTRANSPORT_SESSION capsule, the stream MUST be reset with code H3_MESSAGE_ERROR.

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.

In some scenarios, an endpoint might want to send a CLOSE_WEBTRANSPORT_SESSION with detailed close information and then immediately close the underlying QUIC connection. If the endpoint were to do both of those simultaneously, the peer could potentially receive the CONNECTION_CLOSE before receiving the CLOSE_WEBTRANSPORT_SESSION, thus never receiving the application error data contained in the latter. To avoid this, the endpoint SHOULD wait until all of the data on the CONNECT stream is acknowledged before sending the CONNECTION_CLOSE; this gives CLOSE_WEBTRANSPORT_SESSION properties similar to that of the QUIC CONNECTION_CLOSE mechanism as a best-effort mechanism of delivering application close metadata.

6. Negotiating the Draft Version

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

The wire format aspects of the protocol are negotiated by changing the codepoint used for the SETTINGS_WEBTRANSPORT_MAX_SESSIONS 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.

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 an HTTP/3 setting; 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.

A client could attempt to exhaust resources by opening too many WebTransport sessions at once. In cases when the client is untrusted, the user agent SHOULD limit the number of outgoing sessions the client can open.

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 Section 16.7 of [HTTP].

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_WEBTRANSPORT_MAX_SESSIONS parameter indicates that the specified HTTP/3 endpoint is WebTransport-capable and, for servers, the number of concurrent sessions it is willing to receive. The default value for the SETTINGS_WEBTRANSPORT_MAX_SESSIONS parameter is "0", meaning that the endpoint is not willing to receive any WebTransport sessions.

Setting Name: WEBTRANSPORT_MAX_SESSIONS

Value: 0xc671706a

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 is reserved for the purpose of avoiding collision with WebTransport HTTP/3 extensions:

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: WEBTRANSPORT_BUFFERED_STREAM_REJECTED
Value: 0x3994bd84
Description: WebTransport data stream rejected due to lack of associated session.
Specification: This document.

Name: WEBTRANSPORT_SESSION_GONE
Value: 0x170d7b68
Description: WebTransport data stream aborted because the associated WebTransport session has been closed.
Specification: This document.

In addition, the following range of entries is registered:

Name: WEBTRANSPORT_APPLICATION_ERROR
Value: 0x52e4a40fa8db to 0x52e5ac983162 inclusive, with the exception of the codepoints of form 0x1f * N + 0x21.
Description: WebTransport application error codes.
Specification: This document.

8.6. Capsule Types

The following entries are added to the "HTTP Capsule Types" registry established by [HTTP-DATAGRAM]:

The CLOSE_WEBTRANSPORT_SESSION capsule.

Value: 0x2843
Capsule Type: CLOSE_WEBTRANSPORT_SESSION
Status: permanent
Specification: This document
Change Controller: IETF
Contact: WebTransport Working Group webtransport@ietf.org (mailto:webtransport@ietf.org)
The DRAIN_WEBTRANSPORT_SESSION capsule.

Value: 0x78ae
Capsule Type: DRAIN_WEBTRANSPORT_SESSION
Status: provisional (when this document is approved this will become permanent)
Specification: This document
Change Controller: IETF
Contact: WebTransport Working Group webtransport@ietf.org
         (mailto:webtransport@ietf.org)
Notes: None

9. References

9.1. Normative References


9.2. Informative References

[I-D.ietf-webtrans-http2]

Appendix A. Changelog
A.1. Changes between draft versions 02 and 07

The following changes make the draft-02 and draft-07 versions of this protocol incompatible:

* draft-07 requires SETTINGS_WEBTRANSPORT_MAX_SESSIONS (#86) and uses it for version negotiation (#129)

* draft-07 explicitly requires SETTINGS_ENABLE_CONNECT_PROTOCOL to be enabled (#93)

* draft-07 explicitly requires SETTINGS_H3_DATAGRAM to be enabled (#106)

* draft-07 only allows WEBTRANSPORT_STREAM at the beginning of the stream

The following changes that are present in draft-07 can be also implemented by a draft-02 implementation safely:

* Expanding stream reset error code space from 8 to 32 bits (#115)

* WEBTRANSPORT_SESSION_GONE error code (#75)

* Handling for HTTP GOAWAY (#76)

* DRAIN_WEBTRANSPORT_SESSION capsule (#79)

* Disallowing following redirects automatically (#113)

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