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The WebSocket protocol
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Abstract

The WebSocket protocol enables two-way communication between a client running untrusted code running in a controlled environment to a remote host that has opted-in to communications from that code. The security model used for this is the Origin-based security model commonly used by Web browsers. The protocol consists of an opening handshake followed by basic message framing, layered over TCP. The goal of this technology is to provide a mechanism for browser-based applications that need two-way communication with servers that does not rely on opening multiple HTTP connections (e.g. using XMLHttpRequest or <iframe>s and long polling).

Please send feedback to the hybi@ietf.org mailing list.

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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1. Introduction

1.1. Background

This section is non-normative.

Historically, creating Web applications that need bidirectional communication between a client and a server (e.g., instant messaging and gaming applications) has required an abuse of HTTP to poll the server for updates while sending upstream notifications as distinct HTTP calls.[RFC6202]

This results in a variety of problems:

- The server is forced to use a number of different underlying TCP connections for each client: one for sending information to the client, and a new one for each incoming message.
- The wire protocol has a high overhead, with each client-to-server message having an HTTP header.
- The client-side script is forced to maintain a mapping from the outgoing connections to the incoming connection to track replies.

A simpler solution would be to use a single TCP connection for traffic in both directions. This is what the WebSocket protocol provides. Combined with the WebSocket API, it provides an alternative to HTTP polling for two-way communication from a Web page to a remote server. [WSAPI]

The same technique can be used for a variety of Web applications: games, stock tickers, multiuser applications with simultaneous editing, user interfaces exposing server-side services in real time, etc.

1.2. Protocol Overview

This section is non-normative.

The protocol has two parts: a handshake, and then the data transfer.

The handshake from the client looks as follows:

```
GET /chat HTTP/1.1
Host: server.example.com
Upgrade: websocket
Connection: Upgrade
Sec-WebSocket-Key: dGhlIHNhbXBsZSBub25jZQ==
Origin: http://example.com
Sec-WebSocket-Protocol: chat, superchat
Sec-WebSocket-Version: 13
```

The handshake from the server looks as follows:

```
HTTP/1.1 101 Switching Protocols
Upgrade: websocket
Connection: Upgrade
Sec-WebSocket-Accept: s3pPLMBiTxaQ9kYGzzhZRbK+xOo=
Sec-WebSocket-Protocol: chat
```

The leading line from the client follows the Request-Line format. The leading line from the server follows the Status-Line format. The Request-Line and Status-Line productions are defined in [RFC2616].

After the leading line in both cases come an unordered set of header fields. The meaning of these header fields is specified in Section 4 of this document. Additional header fields may also be present, such as cookies [RFC6265]. The format and parsing of headers is as defined in [RFC2616].

Once the client and server have both sent their handshakes, and if the handshake was successful, then the data transfer part starts. This is a two-way communication channel where each side can, independently from the other, send data at will.

Clients and servers, after a successful handshake, transfer data back and forth in conceptual units referred to in this specification as "messages". On the wire a message is composed of one or more frames. The WebSocket message does not necessarily correspond to a particular network layer framing, as a fragmented message may be coalesced or split by an intermediary.

A frame has an associated type. Each frame belonging to the same message contain the same type of data. Broadly speaking, there are types for textual data, which is interpreted as UTF-8 [RFC3629] text, binary data (whose interpretation is left up to the application), and control frames, which are not intended to carry data for the application, but instead for protocol-level signaling, such as to signal that the connection should be closed. This version of the protocol defines six frame types and leaves ten reserved for future

use.

1.3. Opening Handshake

This section is non-normative.

The opening handshake is intended to be compatible with HTTP-based server-side software and intermediaries, so that a single port can be used by both HTTP clients talking to that server and WebSocket clients talking to that server. To this end, the WebSocket client's handshake is an HTTP Upgrade request:

```
GET /chat HTTP/1.1
Host: server.example.com
Upgrade: websocket
Connection: Upgrade
Sec-WebSocket-Key: dGhlIHNhbXBsZSBub25jZQ==
Origin: http://example.com
Sec-WebSocket-Protocol: chat, superchat
Sec-WebSocket-Version: 13
```

In compliance with [RFC2616], header fields in the handshake may be sent by the client in any order, so the order in which different header fields are received is not significant.

The "Request-URI" of the GET method [RFC2616] is used to identify the endpoint of the WebSocket connection, both to allow multiple domains to be served from one IP address and to allow multiple WebSocket endpoints to be served by a single server.

The client includes the hostname in the Host header field of its handshake as per [RFC2616], so that both the client and the server can verify that they agree on which host is in use.

Additional header fields are used to select options in the WebSocket protocol. Typical options available in this version are the subprotocol selector (|Sec-WebSocket-Protocol|), list of extensions support by the client (|Sec-WebSocket-Extensions|), |Origin| header field, etc. The |Sec-WebSocket-Protocol| request-header field can be used to indicate what subprotocols (application-level protocols layered over the WebSocket protocol) are acceptable to the client. The server selects one or none of the acceptable protocols and echoes that value in its handshake to indicate that it has selected that protocol.

```
Sec-WebSocket-Protocol: chat
```

The |Origin| header field [I-D.ietf-websec-origin] is used to protect

against unauthorized cross-origin use of a WebSocket server by scripts using the |WebSocket| API in a Web browser. The server is informed of the script origin generating the WebSocket connection request. If the server does not wish to accept connections from this origin, it can choose to reject the connection by sending an appropriate HTTP error code. This header field is sent by browser clients, for non-browser clients this header field may be sent if it makes sense in the context of those clients.

Finally, the server has to prove to the client that it received the client's WebSocket handshake, so that the server doesn't accept connections that are not WebSocket connections. This prevents an attacker from tricking a WebSocket server by sending it carefully-crafted packets using |XMLHttpRequest| or a |form| submission.

To prove that the handshake was received, the server has to take two pieces of information and combine them to form a response. The first piece of information comes from the |Sec-WebSocket-Key| header field in the client handshake:

```
Sec-WebSocket-Key: dGhlIHNhbXBsZSBub25jZQ==
```

For this header field, the server has to take the value (as present in the header field, e.g. the base64-encoded [RFC4648] version minus any leading and trailing whitespace), and concatenate this with the Globally Unique Identifier (GUID, [RFC4122]) "258EAFA5-E914-47DA-95CA-C5AB0DC85B11" in string form, which is unlikely to be used by network endpoints that do not understand the WebSocket protocol. A SHA-1 hash (160 bits), base64-encoded (see Section 4 of [RFC4648]), of this concatenation is then returned in the server's handshake [FIPS.180-2.2002].

Concretely, if as in the example above, |Sec-WebSocket-Key| header field had the value "dGhlIHNhbXBsZSBub25jZQ==", the server would concatenate the string "258EAFA5-E914-47DA-95CA-C5AB0DC85B11" to form the string "dGhlIHNhbXBsZSBub25jZQ==258EAFA5-E914-47DA-95CA-C5AB0DC85B11". The server would then take the SHA-1 hash of this, giving the value 0xb3 0x7a 0x4f 0x2c 0xc0 0x62 0x4f 0x16 0x90 0xf6 0x46 0x06 0xcf 0x38 0x59 0x45 0xb2 0xbe 0xc4 0xea. This value is then base64-encoded (see Section 4 of [RFC4648]), to give the value "s3pPLMBiTxaQ9kYGzzhZRbK+xOo=". This value would then be echoed in the |Sec-WebSocket-Accept| header field.

The handshake from the server is much simpler than the client handshake. The first line is an HTTP Status-Line, with the status code 101:

```
HTTP/1.1 101 Switching Protocols
```

Any status code other than 101 indicates that the WebSocket handshake has not completed, and that the semantics of HTTP still apply. The headers follow the status code.

The |Connection| and |Upgrade| header fields complete the HTTP Upgrade. The |Sec-WebSocket-Accept| header field indicates whether the server is willing to accept the connection. If present, this header field must include a hash of the client's nonce sent in |Sec-WebSocket-Key| along with a predefined GUID. Any other value must not be interpreted as an acceptance of the connection by the server.

```
HTTP/1.1 101 Switching Protocols
Upgrade: websocket
Connection: Upgrade
Sec-WebSocket-Accept: s3pPLMBiTxaQ9kYGzzhZRbK+xOo=
```

These fields are checked by the |WebSocket| client for scripted pages. If the |Sec-WebSocket-Accept| value does not match the expected value, or if the header field is missing, or if the HTTP status code is not 101, the connection will not be established and WebSocket frames will not be sent.

Option fields can also be included. In this version of the protocol, the main option field is |Sec-WebSocket-Protocol|, which indicates the subprotocol that the server has selected. WebSocket clients verify that the server included one of the values as was specified in the WebSocket client's handshake. A server that speaks multiple subprotocols has to make sure it selects one based on the client's handshake and specifies it in its handshake.

```
Sec-WebSocket-Protocol: chat
```

The server can also set cookie-related option fields to _set_ cookies, as described in [RFC6265].

1.4. Closing Handshake

This section is non-normative.

The closing handshake is far simpler than the opening handshake.

Either peer can send a control frame with data containing a specified control sequence to begin the closing handshake (detailed in Section 5.5.1). Upon receiving such a frame, the other peer sends a close frame in response, if it hasn't already sent one. Upon

receiving _that_ control frame, the first peer then closes the connection, safe in the knowledge that no further data is forthcoming.

After sending a control frame indicating the connection should be closed, a peer does not send any further data; after receiving a control frame indicating the connection should be closed, a peer discards any further data received.

It is safe for both peers to initiate this handshake simultaneously.

The closing handshake is intended to complement the TCP closing handshake (FIN/ACK), on the basis that the TCP closing handshake is not always reliable end-to-end, especially in the presence of intercepting proxies and other intermediaries.

By sending a close frame and waiting for a close frame in response, certain cases are avoided where data may be unnecessarily lost. For instance, on some platforms, if a socket is closed with data in the receive queue, a RST packet is sent, which will then cause recv() to fail for the party that received the RST, even if there was data waiting to be read.

1.5. Design Philosophy

This section is non-normative.

The WebSocket protocol is designed on the principle that there should be minimal framing (the only framing that exists is to make the protocol frame-based instead of stream-based, and to support a distinction between Unicode text and binary frames). It is expected that metadata would be layered on top of WebSocket by the application layer, in the same way that metadata is layered on top of TCP by the application layer (e.g., HTTP).

Conceptually, WebSocket is really just a layer on top of TCP that does the following:

- adds a Web "origin"-based security model for browsers
- adds an addressing and protocol naming mechanism to support multiple services on one port and multiple host names on one IP address;
- layers a framing mechanism on top of TCP to get back to the IP packet mechanism that TCP is built on, but without length limits

- includes an additional closing handshake in-band that is designed to work in the presence of proxies and other intermediaries

Other than that, WebSocket adds nothing. Basically it is intended to be as close to just exposing raw TCP to script as possible given the constraints of the Web. It's also designed in such a way that its servers can share a port with HTTP servers, by having its handshake be a valid HTTP Upgrade request mechanism also. One could conceptually use other protocols to establish client-server messaging, but the intent of WebSockets was to provide a relatively simple protocol that can coexist with HTTP and deployed HTTP infrastructure (such as proxies) that is as close to TCP as is safe for use with such infrastructure given security considerations, with targeted additions to simplify usage and make simple things simple (such as the addition of message semantics).

The protocol is intended to be extensible; future versions will likely introduce additional concepts such as multiplexing.

1.6. Security Model

This section is non-normative.

The WebSocket protocol uses the origin model used by Web browsers to restrict which Web pages can contact a WebSocket server when the WebSocket protocol is used from a Web page. Naturally, when the WebSocket protocol is used by a dedicated client directly (i.e. not from a Web page through a Web browser), the origin model is not useful, as the client can provide any arbitrary origin string.

This protocol is intended to fail to establish a connection with servers of pre-existing protocols like SMTP [RFC5321] and HTTP, while allowing HTTP servers to opt-in to supporting this protocol if desired. This is achieved by having a strict and elaborate handshake, and by limiting the data that can be inserted into the connection before the handshake is finished (thus limiting how much the server can be influenced).

It is similarly intended to fail to establish a connection when data from other protocols, especially HTTP, is sent to a WebSocket server, for example as might happen if an HTML |form| were submitted to a WebSocket server. This is primarily achieved by requiring that the server prove that it read the handshake, which it can only do if the handshake contains the appropriate parts which themselves can only be sent by a WebSocket handshake. In particular, at the time of writing of this specification, fields starting with |Sec-| cannot be set by an attacker from a Web browser using only HTML and JavaScript APIs such as |XMLHttpRequest|.

1.7. Relationship to TCP and HTTP

This section is non-normative.

The WebSocket protocol is an independent TCP-based protocol. Its only relationship to HTTP is that its handshake is interpreted by HTTP servers as an Upgrade request.

By default the WebSocket protocol uses port 80 for regular WebSocket connections and port 443 for WebSocket connections tunneled over TLS [RFC2818].

1.8. Establishing a Connection

This section is non-normative.

When a connection is to be made to a port that is shared by an HTTP server (a situation that is quite likely to occur with traffic to ports 80 and 443), the connection will appear to the HTTP server to be a regular GET request with an Upgrade offer. In relatively simple setups with just one IP address and a single server for all traffic to a single hostname, this might allow a practical way for systems based on the WebSocket protocol to be deployed. In more elaborate setups (e.g. with load balancers and multiple servers), a dedicated set of hosts for WebSocket connections separate from the HTTP servers is probably easier to manage. At the time of writing of this specification, it should be noted that connections on port 80 and 443 have significantly different success rates, with connections on port 443 being significantly more likely to succeed, though this may change with time.

1.9. Subprotocols Using the WebSocket protocol

This section is non-normative.

The client can request that the server use a specific subprotocol by including the |Sec-WebSocket-Protocol| field in its handshake. If it is specified, the server needs to include the same field and one of the selected subprotocol values in its response for the connection to be established.

These subprotocol names should be registered as per Section 11.5. To avoid potential collisions, it is recommended to use names that contain the ASCII version of the domain name of the subprotocol's originator. For example, if Example Corporation were to create a Chat subprotocol to be implemented by many servers around the Web, they could name it "chat.example.com". If the Example Organization called their competing subprotocol "chat.example.org", then the two

subprotocols could be implemented by servers simultaneously, with the server dynamically selecting which subprotocol to use based on the value sent by the client.

Subprotocols can be versioned in backwards-incompatible ways by changing the subprotocol name, e.g. going from "bookings.example.net" to "v2.bookings.example.net". These subprotocols would be considered completely separate by WebSocket clients. Backwards-compatible versioning can be implemented by reusing the same subprotocol string but carefully designing the actual subprotocol to support this kind of extensibility.

2. Conformance Requirements

All diagrams, examples, and notes in this specification are non-normative, as are all sections explicitly marked non-normative. Everything else in this specification is normative.

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC2119. [RFC2119]

Requirements phrased in the imperative as part of algorithms (such as "strip any leading space characters" or "return false and abort these steps") are to be interpreted with the meaning of the key word ("must", "should", "may", etc) used in introducing the algorithm.

Conformance requirements phrased as algorithms or specific steps MAY be implemented in any manner, so long as the end result is equivalent. (In particular, the algorithms defined in this specification are intended to be easy to follow, and not intended to be performant.)

The conformance classes defined by this specification are clients and servers.

2.1. Terminology

ASCII shall mean the character-encoding scheme defined in [ANSI.X3-4.1986].

This document makes reference to UTF-8 values and uses UTF-8 notational formats as defined in STD 63 [RFC3629].

Key Terms such as named algorithms or definitions are indicated like _this_.

Names of header fields or variables are indicated like |this|.

Variable values are indicated like /this/.

This document references the procedure to _Fail the WebSocket Connection_. This procedure is defined in Section 7.1.7.

Converting a string to ASCII lowercase means replacing all characters in the range U+0041 to U+005A (i.e. LATIN CAPITAL LETTER A to LATIN CAPITAL LETTER Z) with the corresponding characters in the range U+0061 to U+007A (i.e. LATIN SMALL LETTER A to LATIN SMALL LETTER Z).

Comparing two strings in an _ASCII case-insensitive_ manner means comparing them exactly, code point for code point, except that the characters in the range U+0041 to U+005A (i.e. LATIN CAPITAL LETTER A to LATIN CAPITAL LETTER Z) and the corresponding characters in the range U+0061 to U+007A (i.e. LATIN SMALL LETTER A to LATIN SMALL LETTER Z) are considered to also match.

The term "URI" is used in this document as defined in [RFC3986].

When an implementation is required to _send_ data as part of the WebSocket protocol, the implementation MAY delay the actual transmission arbitrarily, e.g. buffering data so as to send fewer IP packets.

3. WebSocket URIs

This specification defines two URI schemes, using the ABNF syntax defined in RFC 5234 [RFC5234], and terminology and ABNF productions defined by the URI specification RFC 3986 [RFC3986].

```
ws-URI = "ws:" "//" host [ ":" port ] path [ "?" query ]
wss-URI = "wss:" "//" host [ ":" port ] path [ "?" query ]

host = <host, defined in [RFC3986], Section 3.2.2>
port = <port, defined in [RFC3986], Section 3.2.3>
path = <path-abempty, defined in [RFC3986], Section 3.3>
query = <query, defined in [RFC3986], Section 3.4>
```

The port component is OPTIONAL; the default for "ws" is port 80, while the default for "wss" is port 443.

The URI is called "secure" (and it said that "the secure flag is set") if the scheme component matches "wss" case-insensitively.

The "resource-name" (also known as /resource name/ in Section 4.1) can be constructed by concatenating

- "/" if the path component is empty
- the path component
- "?" if the query component is non-empty
- the query component

Fragment identifiers are meaningless in the context of WebSocket URIs, and MUST NOT be used on these URIs. The character "#" in URIs MUST be escaped as %23 if used as part of the query component.

4. Opening Handshake

4.1. Client Requirements

To _Establish a WebSocket Connection_, a client opens a connection and sends a handshake as defined in this section. A connection is defined to initially be in a CONNECTING state. A client will need to supply a /host/, /port/, /resource name/, and a /secure/ flag, which are the components of a WebSocket URI as discussed in Section 3, along with a list of /protocols/ and /extensions/ to be used. Additionally, if the client is a web browser, an /origin/ MUST be supplied.

Clients running in controlled environments, e.g. browsers on mobile handsets tied to specific carriers, MAY offload the management of the connection to another agent on the network. In such a situation, the client for the purposes of conformance is considered to include both the handset software and any such agents.

When the client is to _Establish a WebSocket Connection_ given a set of (/host/, /port/, /resource name/, and /secure/ flag), along with a list of /protocols/ and /extensions/ to be used, and an /origin/ in the case of web browsers, it MUST open a connection, send an opening handshake, and read the server's handshake in response. The exact requirements of how the connection should be opened, what should be sent in the opening handshake, and how the server's response should be interpreted, are as follows in this section. In the following text, we will use terms from Section 3 such as "/host/" and "/secure/ flag" as defined in that section.

1. The components of the WebSocket URI passed into this algorithm (/host/, /port/, /resource name/ and /secure/ flag) MUST be valid according to the specification of WebSocket URIs specified in Section 3. If any of the components are invalid, the client MUST _Fail the WebSocket Connection_ and abort these steps.

2. If the client already has a WebSocket connection to the remote host (IP address) identified by /host/ and port /port/ pair, even if the remote host is known by another name, the client MUST wait until that connection has been established or for that connection to have failed. There MUST be no more than one connection in a CONNECTING state. If multiple connections to the same IP address are attempted simultaneously, the client MUST serialize them so that there is no more than one connection at a time running through the following steps.

   If the client cannot determine the IP address of the remote host (for example because all communication is being done through a

proxy server that performs DNS queries itself), then the client MUST assume for the purposes of this step that each host name refers to a distinct remote host, and instead the client SHOULD limit the total number of simultaneous pending connections to a reasonably low number (e.g., the client might allow simultaneous pending connections to a.example.com and b.example.com, but if thirty simultaneous connections to a single host are requested, that may not be allowed). In a Web browser context, the client SHOULD consider the number of tabs the user has open in setting a limit to the number of simultaneous pending connections.

NOTE: This makes it harder for a script to perform a denial of service attack by just opening a large number of WebSocket connections to a remote host. A server can further reduce the load on itself when attacked by pausing before closing the connection, as that will reduce the rate at which the client reconnects.

NOTE: There is no limit to the number of established WebSocket connections a client can have with a single remote host. Servers can refuse to accept connections from hosts/IP addresses with an excessive number of existing connections, or disconnect resource-hogging connections when suffering high load.

3. _Proxy Usage_: If the client is configured to use a proxy when using the WebSocket protocol to connect to host /host/ and port /port/, then the client SHOULD connect to that proxy and ask it to open a TCP connection to the host given by /host/ and the port given by /port/.

   EXAMPLE: For example, if the client uses an HTTP proxy for all traffic, then if it was to try to connect to port 80 on server example.com, it might send the following lines to the proxy server:

```
CONNECT example.com:80 HTTP/1.1
Host: example.com
```

   If there was a password, the connection might look like:

```
CONNECT example.com:80 HTTP/1.1
Host: example.com
Proxy-authorization: Basic ZWRuYW1vZGU6bm9jYXBlcyE=
```

If the client is not configured to use a proxy, then a direct TCP connection SHOULD be opened to the host given by /host/ and the

port given by /port/.

NOTE: Implementations that do not expose explicit UI for selecting a proxy for WebSocket connections separate from other proxies are encouraged to use a SOCKS5 [RFC1928] proxy for WebSocket connections, if available, or failing that, to prefer the proxy configured for HTTPS connections over the proxy configured for HTTP connections.

For the purpose of proxy autoconfiguration scripts, the URI to pass the function MUST be constructed from /host/, /port/, /resource name/, and the /secure/ flag using the definition of a WebSocket URI as given in Section 3.

NOTE: The WebSocket protocol can be identified in proxy autoconfiguration scripts from the scheme ("ws" for unencrypted connections and "wss" for encrypted connections).

4. If the connection could not be opened, either because a direct connection failed or because any proxy used returned an error, then the client MUST _Fail the WebSocket Connection_ and abort the connection attempt.

5. If /secure/ is true, the client MUST perform a TLS handshake over the connection after opening the connection and before sending the handshake data [RFC2818]. If this fails (e.g. the server's certificate could not be verified), then the client MUST _Fail the WebSocket Connection_ and abort the connection. Otherwise, all further communication on this channel MUST run through the encrypted tunnel. [RFC5246]

   Clients MUST use the Server Name Indication extension in the TLS handshake. [RFC6066]

Once a connection to the server has been established (including a connection via a proxy or over a TLS-encrypted tunnel), the client MUST send an opening handshake to the server. The handshake consists of an HTTP upgrade request, along with a list of required and optional header fields. The requirements for this handshake are as follows.

1. The handshake MUST be a valid HTTP request as specified by [RFC2616].

2. The Method of the request MUST be GET and the HTTP version MUST be at least 1.1.

   For example, if the WebSocket URI is "ws://example.com/chat",

The first line sent should be "GET /chat HTTP/1.1"

3. The "Request-URI" part of the request MUST match the /resource name/ Section 3 (a relative URI), or be an absolute http/https URI that, when parsed, has a /resource name/, /host/ and /port/ that match the corresponding ws/wss URI.

4. The request MUST contain a "Host" header field whose value is equal to /host/.

5. The request MUST contain an "Upgrade" header field whose value is equal to "websocket".

6. The request MUST contain a "Connection" header field whose value MUST include the "Upgrade" token.

7. The request MUST include a header field with the name "Sec-WebSocket-Key". The value of this header field MUST be a nonce consisting of a randomly selected 16-byte value that has been base64-encoded (see Section 4 of [RFC4648]). The nonce MUST be selected randomly for each connection.

   NOTE: As an example, if the randomly selected value was the sequence of bytes 0x01 0x02 0x03 0x04 0x05 0x06 0x07 0x08 0x09 0x0a 0x0b 0x0c 0x0d 0x0e 0x0f 0x10, the value of the header field would be "AQIDBAUGBwgJCgsMDQ4PEC=="

8. The request MUST include a header field with the name "Origin" [I-D.ietf-websec-origin] if the request is coming from a browser client. If the connection is from a non-browser client, the request MAY include this header field if the semantics of that client match the use-case described here for browser clients. The value of this header field is the ASCII serialization of origin of the context in which the code establishing the connection is running. See [I-D.ietf-websec-origin] for the details of how this header field value is constructed.

   As an example, if code is running on www.example.com attempting to establish a connection to ww2.example.com, the value of the header field would be "http://www.example.com".

9. The request MUST include a header field with the name "Sec-WebSocket-Version". The value of this header field MUST be 13. _Note: Although drafts -09, -10, -11 and -12 were published, as they were mostly comprised of editorial changes and clarifications and not changes to the wire protocol, values 9, 10, 11 and 12 were not used as valid values for Sec-WebSocket-Version. These values were reserved in the IANA registry but

were not and will not be used._

10. The request MAY include a header field with the name "Sec-WebSocket-Protocol". If present, this value indicates one or more comma separated subprotocol the client wishes to speak, ordered by preference. The elements that comprise this value MUST be non-empty strings with characters in the range U+0021 to U+007E not including separator characters as defined in [RFC2616], and MUST all be unique strings. The ABNF for the value of this header field is 1#token, where the definitions of constructs and rules are as given in [RFC2616].

11. The request MAY include a header field with the name "Sec-WebSocket-Extensions". If present, this value indicates the protocol-level extension(s) the client wishes to speak. The interpretation and format of this header field is described in Section 9.1.

12. The request MAY include any other header fields, for example cookies [RFC6265] and/or authentication related header fields such as Authorization header field [RFC2616].

Once the client's opening handshake has been sent, the client MUST wait for a response from the server before sending any further data. The client MUST validate the server's response as follows:

1. If the status code received from the server is not 101, the client handles the response per HTTP [RFC2616] procedures, in particular the client might perform authentication if it receives 401 status code, the server might redirect the client using a 3xx status code (but clients are not required to follow them), etc. Otherwise, proceed as follows.

2. If the response lacks an "Upgrade" header field or the "Upgrade" header field contains a value that is not an ASCII case-insensitive match for the value "websocket", the client MUST _Fail the WebSocket Connection _.

3. If the response lacks a "Connection" header field or the "Connection" header field doesn't contains a token that is an ASCII case-insensitive match for the value "Upgrade", the client MUST _Fail the WebSocket Connection_.

4. If the response lacks a "Sec-WebSocket-Accept" header field or the "Sec-WebSocket-Accept" contains a value other than the base64-encoded SHA-1 of the concatenation of the "Sec-WebSocket-Key" (as a string, not base64-decoded) with the string "258EAFA5-E914-47DA-95CA-C5AB0DC85B11", but ignoring any leading and

trailing whitespace, the client MUST _Fail the WebSocket Connection_

5. If the response includes a "Sec-WebSocket-Extensions" header field, and this header field indicates the use of an extension that was not present in the client' handshake (the server has indicated an extension not requested by the client), the client MUST _Fail the WebSocket Connection_. (The parsing of this header field to determine which extensions are requested is discussed in Section 9.1.)

6. If the response includes a "Sec-WebSocket-Protocol" header field, and this header field indicates the use of a subprotocol that was not present in the client' handshake (the server has indicated a subprotocol not requested by the client), the client MUST _Fail the WebSocket Connection_. (The parsing of this header field to determine which extensions are requested is discussed in Section 9.1.)

If the server's response does not conform to the requirements for the server's handshake as defined in this section and in Section 4.2.2, the client MUST _Fail the WebSocket Connection_.

If the server's response is validated as provided for above, it is said that _The WebSocket Connection is Established_ and that the WebSocket Connection is in the OPEN state. The _Extensions In Use_ is defined to be a (possibly empty) string, the value of which is equal to the value of the |Sec-WebSocket-Extensions| header field supplied by the server's handshake, or the null value if that header field was not present in the server's handshake. The _Subprotocol In Use_ is defined to be the value of the |Sec-WebSocket-Protocol| header field in the server's handshake, or the null value if that header field was not present in the server's handshake. Additionally, if any header fields in the server's handshake indicate that cookies should be set (as defined by [RFC6265]), these cookies are referred to as _Cookies Set During the Server's Opening Handshake_.

4.2. Server-side Requirements

This section only applies to servers.

Servers MAY offload the management of the connection to other agents on the network, for example load balancers and reverse proxies. In such a situation, the server for the purposes of conformance is considered to include all parts of the server-side infrastructure from the first device to terminate the TCP connection all the way to the server that processes requests and sends responses.

EXAMPLE: For example, a data center might have a server that responds to WebSocket requests with an appropriate handshake, and then passes the connection to another server to actually process the data frames. For the purposes of this specification, the "server" is the combination of both computers.

4.2.1. Reading the Client's Opening Handshake

When a client starts a WebSocket connection, it sends its part of the opening handshake. The server must parse at least part of this handshake in order to obtain the necessary information to generate the server part of the handshake.

The client's opening handshake consists of the following parts. If the server, while reading the handshake, finds that the client did not send a handshake that matches the description below (note that as per [RFC2616] the order of the header fields is not important), including but not limited to any violations of the grammar (ABNF) specified for the components of the handshake, the server MUST stop processing the client's handshake, and return an HTTP response with an appropriate error code (such as 400 Bad Request).

1. An HTTP/1.1 or higher GET request, including a "Request-URI" [RFC2616] that should be interpreted as a /resource name/ Section 3 (or an absolute HTTP/HTTPS URI containing the /resource name/).

2. A "Host" header field containing the server's authority.

3. An "Upgrade" header field containing the value "websocket", treated as an ASCII case-insensitive value.

4. A "Connection" header field that includes the token "Upgrade", treated as an ASCII case-insensitive value.

5. A "Sec-WebSocket-Key" header field with a base64-encoded (see Section 4 of [RFC4648]) value that, when decoded, is 16 bytes in length.

6. A "Sec-WebSocket-Version" header field, with a value of 8.

7. Optionally, a "Origin" header field. This header field is sent by all browser clients. A connection attempt lacking this header field SHOULD NOT be interpreted as coming from a browser client.

8. Optionally, a "Sec-WebSocket-Protocol" header field, with a list of values indicating which protocols the client would like to

speak, ordered by preference.

9. Optionally, a "Sec-WebSocket-Extensions" header field, with a list of values indicating which extensions the client would like to speak. The interpretation of this header field is discussed in Section 9.1.

10. Optionally, other header fields, such as those used to send cookies or request authentication to a server. Unknown header fields are ignored, as per [RFC2616].

4.2.2. Sending the Server's Opening Handshake

When a client establishes a WebSocket connection to a server, the server MUST complete the following steps to accept the connection and send the server's opening handshake.

1. If the server supports encryption, perform a TLS handshake over the connection. If this fails (e.g. the client indicated a host name in the extended client hello "server_name" extension that the server does not host), then close the connection; otherwise, all further communication for the connection (including the server's handshake) MUST run through the encrypted tunnel. [RFC5246]

2. If the server wishes to perform additional client authentication, it might return 401 status code with the corresponding WWW-Authenticate header field as described in [RFC2616].

3. The server MAY redirect the client using a 3xx status code [RFC2616]. Note that this step can happen together with, before or after the optional authentication step described above.

4. Establish the following information:

   /origin/
   The |Origin| header field in the client's handshake indicates the origin of the script establishing the connection. The origin is serialized to ASCII and converted to lowercase. The server MAY use this information as part of a determination of whether to accept the incoming connection. If the server does not validate the origin, it will accept connections from anywhere. If the server does not wish to accept this connection, it MUST return an appropriate HTTP error code (e.g. 403 Forbidden) and abort the WebSocket handshake described in this section. For more detail, refer to Section 10.

/key/
The |Sec-WebSocket-Key| header field in the client's handshake includes a base64-encoded value that, if decoded, is 16 bytes in length. This (encoded) value is used in the creation of the server's handshake to indicate an acceptance of the connection. It is not necessary for the server to base64-decode the "Sec-WebSocket-Key" value.

/version/
The |Sec-WebSocket-Version| header field in the client's handshake includes the version of the WebSocket protocol the client is attempting to communicate with. If this version does not match a version understood by the server, the server MUST abort the websocket handshake described in this section and instead send an appropriate HTTP error code (such as 426 Upgrade Required), and a |Sec-WebSocket-Version| header field indicating the version(s) the server is capable of understanding.

/resource name/
An identifier for the service provided by the server. If the server provides multiple services, then the value should be derived from the resource name given in the client's handshake from the Request-URI [RFC2616] of the GET method. If the requested service is not available, the server MUST send an appropriate HTTP error code (such as 404 Not Found) and abort the WebSocket handshake.

/subprotocol/
Either a single value representing the subprotocol the server is ready to use or null. The value chosen MUST be derived from the client's handshake, specifically by selecting one of the values from the "Sec-WebSocket-Protocol" field that the server is willing to use for this connection (if any). If the client's handshake did not contain such a header field, or if the server does not agree to any of the client's requested subprotocols, the only acceptable value is null. The absence of such a field is equivalent to the null value (meaning that if the server does not wish to agree to one of the suggested subprotocols, it MUST NOT send back a |Sec-WebSocket-Protocol| header field in its response). The empty string is not the same as the null value for these purposes, and is not a legal value for this field. The ABNF for the value of this header field is (token), where the definitions of constructs and rules are as given in [RFC2616].

/extensions/
A (possibly empty) list representing the protocol-level extensions the server is ready to use. If the server supports multiple extensions, then the value MUST be derived from the client's handshake, specifically by selecting one or more of the values from the "Sec-WebSocket-Extensions" field. The absence of such a field is equivalent to the null value. The empty string is not the same as the null value for these purposes. Extensions not listed by the client MUST NOT be listed. The method by which these values should be selected and interpreted is discussed in Section 9.1.

5. If the server chooses to accept the incoming connection, it MUST reply with a valid HTTP response indicating the following.

   1. A Status-Line with a 101 response code as per RFC 2616 [RFC2616]. Such a response could look like "HTTP/1.1 101 Switching Protocols"

   2. An "Upgrade" header field with value "websocket" as per RFC 2616 [RFC2616].

   3. A "Connection" header field with value "Upgrade"

   4. A "Sec-WebSocket-Accept" header field. The value of this header field is constructed by concatenating /key/, defined above in Paragraph 4 of Section 4.2.2, with the string "258EAFA5-E914-47DA-95CA-C5AB0DC85B11", taking the SHA-1 hash of this concatenated value to obtain a 20-byte value, and base64-encoding (see Section 4 of [RFC4648]) this 20-byte hash.

      The ABNF of this header field is defined as follows:

```
Sec-WebSocket-Accept     = base64-value
base64-value     = *base64-data [ base64-padding ]
base64-data      = 4base64-character
base64-padding   = (2base64-character "==") |
                   (3base64-character "=")
base64-character = ALPHA | DIGIT | "+" | "/"
```

      NOTE: As an example, if the value of the "Sec-WebSocket-Key" header field in the client's handshake were "dGhlIHNhbXBsZSBub25jZQ==", the server would append the string "258EAFA5-E914-47DA-95CA-C5AB0DC85B11" to form the string "dGhlIHNhbXBsZSBub25jZQ==258EAFA5-E914-47DA-95CA-C5AB0DC85B11". The server would then take the SHA-1 hash of

this string, giving the value 0xb3 0x7a 0x4f 0x2c 0xc0 0x62 0x4f 0x16 0x90 0xf6 0x46 0x06 0xcf 0x38 0x59 0x45 0xb2 0xbe 0xc4 0xea. This value is then base64-encoded, to give the value "s3pPLMBiTxaQ9kYGzzhZRbK+xOo=", which would be returned in the "Sec-WebSocket-Accept" header field.

5. Optionally, a "Sec-WebSocket-Protocol" header field, with a value /subprotocol/ as defined in Paragraph 4 of Section 4.2.2.

6. Optionally, a "Sec-WebSocket-Extensions" header field, with a value /extensions/ as defined in Paragraph 4 of Section 4.2.2. If multiple extensions are to be used, they must all be listed in a single Sec-WebSocket-Extensions header field. This header field MUST NOT be repeated.

This completes the server's handshake. If the server finishes these steps without aborting the WebSocket handshake, the server considers the WebSocket connection to be established and that the WebSocket connection is in the OPEN state. At this point, the server may begin sending (and receiving) data.

4.3. Collected ABNF for new header fields used in handshake

Unlike other section of the document this section is using ABNF syntax/rules from Section 2.1 of [RFC2616], including "implied *LWS rule".

Note that the following ABNF conventions are used in this section: Some names of the rules correspond to names of the corresponding header fields. Such rules express values of the corresponding header fields, for example the Sec-WebSocket-Key ABNF rule describes syntax of the Sec-WebSocket-Key header field value. ABNF rules with the "-Client" suffix in the name are only used in requests sent by the client to the server; ABNF rules with the "-Server" suffix in the name are only used in responses sent by the server to the client. For example, the ABNF rule Sec-WebSocket-Protocol-Client describes syntax of the Sec-WebSocket-Protocol header field value sent by the client to the server.

The following new header field can be sent during the handshake from the client to the server:

```
Sec-WebSocket-Key = base64-value
Sec-WebSocket-Extensions = extension-list
Sec-WebSocket-Protocol-Client = 1#token
Sec-WebSocket-Version-Client = version

base64-value     = *base64-data [ base64-padding ]
base64-data      = 4base64-character
base64-padding   = (2base64-character "==") |
                   (3base64-character "=")
base64-character = ALPHA | DIGIT | "+" | "/"
extension-list = 1#extension
extension = extension-token *( ";" extension-param )
extension-token = registered-token
registered-token = token
extension-param = token [ "=" token ]
NZDIGIT       =  "1" | "2" | "3" | "4" | "5" | "6" |
                 "7" | "8" | "9"
version = DIGIT | (NZDIGIT DIGIT) |
          ("1" DIGIT DIGIT) | ("2" DIGIT DIGIT)
          ; Limited to 0-255 range, with no leading zeros
```

The following new header field can be sent during the handshake from the server to the client:

```
Sec-WebSocket-Extensions = extension-list
Sec-WebSocket-Accept     = base64-value
Sec-WebSocket-Protocol-Server = token
Sec-WebSocket-Version-Server = 1#version
```

5. Data Framing

5.1. Overview

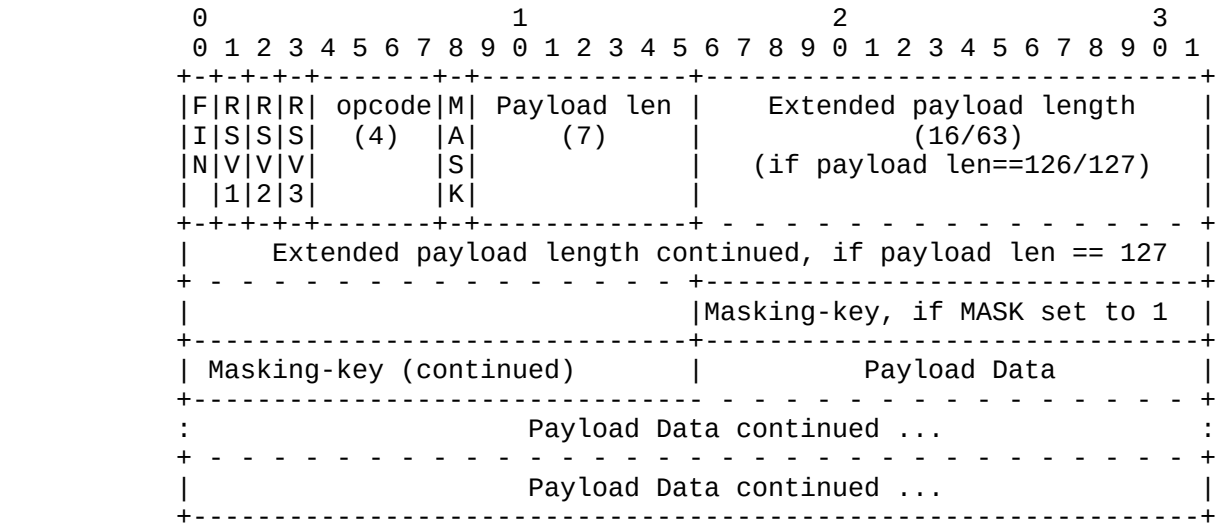
In the WebSocket protocol, data is transmitted using a sequence of frames. All frames sent from the client to the server are masked to avoid confusing network intermediaries, such as intercepting proxies. All frames sent from the server to the client are not masked.

The base framing protocol defines a frame type with an opcode, a payload length, and designated locations for extension and application data, which together define the _payload_ data. Certain bits and opcodes are reserved for future expansion of the protocol.

A data frame MAY be transmitted by either the client or the server at any time after opening handshake completion and before that endpoint has sent a close frame (Section 5.5.1).

5.2. Base Framing Protocol

This wire format for the data transfer part is described by the ABNF [RFC5234] given in detail in this section. A high level overview of the framing is given in the following figure.



FIN: 1 bit

Indicates that this is the final fragment in a message. The first fragment MAY also be the final fragment.

RSV1, RSV2, RSV3: 1 bit each

MUST be 0 unless an extension is negotiated which defines meanings for non-zero values. If a nonzero value is received and none of the negotiated extensions defines the meaning of such a nonzero value, the receiving endpoint MUST _Fail the WebSocket Connection_.

Opcode: 4 bits

Defines the interpretation of the payload data. If an unknown opcode is received, the receiving endpoint MUST _Fail the WebSocket Connection_. The following values are defined.

* %x0 denotes a continuation frame
* %x1 denotes a text frame
* %x2 denotes a binary frame
* %x3-7 are reserved for further non-control frames
* %x8 denotes a connection close
* %x9 denotes a ping
* %xA denotes a pong
* %xB-F are reserved for further control frames

Mask: 1 bit

Defines whether the payload data is masked. If set to 1, a masking key is present in masking-key, and this is used to unmask the payload data as per Section 5.3. All frames sent from client to server have this bit set to 1.

Payload length: 7 bits, 7+16 bits, or 7+64 bits

The length of the payload data, in bytes: if 0-125, that is the payload length. If 126, the following 2 bytes interpreted as a 16 bit unsigned integer are the payload length. If 127, the following 8 bytes interpreted as a 64-bit unsigned integer (the

most significant bit MUST be 0) are the payload length. Multibyte length quantities are expressed in network byte order. The payload length is the length of the extension data + the length of the application data. The length of the extension data may be zero, in which case the payload length is the length of the application data.

Masking-key: 0 or 4 bytes

All frames sent from the client to the server are masked by a 32-bit value that is contained within the frame. This field is present if the mask bit is set to 1, and is absent if the mask bit is set to 0. See Section 5.3 for further information on client-to-server masking.

Payload data: (x+y) bytes

The payload data is defined as extension data concatenated with application data.

Extension data: x bytes

The extension data is 0 bytes unless an extension has been negotiated. Any extension MUST specify the length of the extension data, or how that length may be calculated, and how the extension use MUST be negotiated during the opening handshake. If present, the extension data is included in the total payload length.

Application data: y bytes

Arbitrary application data, taking up the remainder of the frame after any extension data. The length of the application data is equal to the payload length minus the length of the extension data.

The base framing protocol is formally defined by the following ABNF [RFC5234]:

```
ws-frame                = frame-fin
                          frame-rsv1
                          frame-rsv2
                          frame-rsv3
                          frame-opcode
                          frame-masked
                          frame-payload-length
                          [ frame-masking-key ]
                          frame-payload-data
```

```
   frame-fin               = %x0 ; more frames of this message follow
                           / %x1 ; final frame of this message

   frame-rsv1              = %x0
                             ; 1 bit, MUST be 0 unless negotiated
                             ; otherwise

   frame-rsv2              = %x0
                             ; 1 bit, MUST be 0 unless negotiated
                             ; otherwise

   frame-rsv3              = %x0
                             ; 1 bit, MUST be 0 unless negotiated
                             ; otherwise

   frame-opcode            = %x0 ; continuation frame
                           / %x1 ; text frame
                           / %x2 ; binary frame
                           / %x3-7
                          ; reserved for further non-control frames
                           / %x8 ; connection close
                           / %x9 ; ping
                           / %xA ; pong
                           / %xB-F ; reserved for further control frames

   frame-masked            = %x0
                          ; frame is not masked, no frame-masking-key
                           / %x1
                          ; frame is masked, frame-masking-key present

   frame-payload-length    = %x00-7D
                           / %x7E frame-payload-length-16
                           / %x7F frame-payload-length-63

   frame-payload-length-16 = %x0000-FFFF

   frame-payload-length-63 = %x0000000000000000-7FFFFFFFFFFFFFFF

   frame-masking-key       = 4( %0x00-FF )
                             ; present only if frame-masked is 1

   frame-payload-data      = (frame-masked-extension-data
                              frame-masked-application-data)
                           ; frame-masked 1
                             / (frame-unmasked-extension-data
                               frame-unmasked-application-data)
                           ; frame-masked 0
```

```
frame-masked-extension-data     = *( %x00-FF ) ; to be defined later

frame-masked-application-data   = *( %x00-FF )

frame-unmasked-extension-data   = *( %x00-FF ) ; to be defined later

frame-unmasked-application-data = *( %x00-FF )
```

5.3. Client-to-Server Masking

The client MUST mask all frames sent to the server. This is required for security reasons which are further discussed in Section 10.3. A server MUST close the connection upon receiving a frame with the MASK bit set to 0. In this case, a server MAY send a close frame with a status code of 1002 (protocol error) as defined in Section 7.4.1.

A masked frame MUST have the field frame-masked set to 1, as defined in Section 5.2.

The masking key is contained completely within the frame, as defined in Section 5.2 as frame-masking-key. It is used to mask the payload data defined in the same section as frame-payload-data, which includes extension and application data.

The masking key is a 32-bit value chosen at random by the client. The masking key MUST be derived from a strong source of entropy, and the masking key for a given frame MUST NOT make it simple for a server to predict the masking key for a subsequent frame. RFC 4086 [RFC4086] discusses what entails a suitable source of entropy for security-sensitive applications.

The masking does not affect the length of the payload data. To convert masked data into unmasked data, or vice versa, the following algorithm is applied. The same algorithm applies regardless of the direction of the translation - e.g. the same steps are applied to mask the data as to unmask the data.

Octet i of the transformed data ("transformed-octet-i") is the XOR of octet i of the original data ("original-octet-i") with octet at index i modulo 4 of the masking key ("masking-key-octet-j"):

```
j                   = i MOD 4
transformed-octet-i = original-octet-i XOR masking-key-octet-j
```

When preparing a masked frame, the client MUST pick a fresh masking key from the set of allowed 32-bit values. The masking key must be

unpredictable. The unpredictability of the masking key is essential to prevent the author of malicious applications from selecting the bytes that appear on the wire.

The payload length, indicated in the framing as frame-payload-length, does NOT include the length of the masking key. It is the length of the payload data, e.g. the number of bytes following the masking key.

5.4. Fragmentation

The primary purpose of fragmentation is to allow sending a message that is of unknown size when the message is started without having to buffer that message. If messages couldn't be fragmented, then an endpoint would have to buffer the entire message so its length could be counted before first byte is sent. With fragmentation, a server or intermediary may choose a reasonable size buffer, and when the buffer is full write a fragment to the network.

A secondary use-case for fragmentation is for multiplexing, where it is not desirable for a large message on one logical channel to monopolize the output channel, so the MUX needs to be free to split the message into smaller fragments to better share the output channel.

Unless specified otherwise by an extension, frames have no semantic meaning. An intermediary might coalesce and/or split frames, if no extensions were negotiated by the client and the server, or if some extensions were negotiated, but the intermediary understood all the extensions negotiated and knows how to coalesce and/or split frames in presence of these extensions. One implication of this is that in absence of extensions senders and receivers must not depend on presence of specific frame boundaries.

The following rules apply to fragmentation:

- An unfragmented message consists of a single frame with the FIN bit set and an opcode other than 0.

- A fragmented message consists of a single frame with the FIN bit clear and an opcode other than 0, followed by zero or more frames with the FIN bit clear and the opcode set to 0, and terminated by a single frame with the FIN bit set and an opcode of 0. A fragmented message is conceptually equivalent to a single larger message whose payload is equal to the concatenation of the payloads of the fragments in order, however in the presence of extensions this may not hold true as the extension defines the interpretation of the extension data present. For instance, extension data may only be present at the beginning of the first

fragment and apply to subsequent fragments, or there may be extension data present in each of the fragments that applies only to that particular fragment. In absence of extension data, the following example demonstrates how fragmentation works.

EXAMPLE: For a text message sent as three fragments, the first fragment would have an opcode of 0x1 and a FIN bit clear, the second fragment would have an opcode of 0x0 and a FIN bit clear, and the third fragment would have an opcode of 0x0 and a FIN bit that is set.

- Control frames MAY be injected in the middle of a fragmented message. Control frames themselves MUST NOT be fragmented.
- Message fragments MUST be delivered to the recipient in the order sent by the sender.
- The fragments of one message MUST NOT be interleaved between the fragments of another message unless an extension has been negotiated that can interpret the interleaving.
- An endpoint MUST be capable of handling control frames in the middle of a fragmented message.
- A sender MAY create fragments of any size for non-control messages.
- Clients and servers MUST support receiving both fragmented and unfragmented messages.
- As control frames cannot be fragmented, an intermediary MUST NOT attempt to change the fragmentation of a control frame.
- An intermediary MUST NOT change the fragmentation of a message if any reserved bit values are used and the meaning of these values is not known to the intermediary.
- An intermediary MUST NOT change the fragmentation of any message in the context of a connection where extensions have been negotiated and the intermediary is not aware of the semantics of the negotiated extensions.
- As a consequence of these rules, all fragments of a message are of the same type, as set by the first fragment’s opcode. Since Control frames cannot be fragmented, the type for all fragments in a message MUST be either text or binary, or one of the reserved opcodes.

Note: if control frames could not be interjected, the latency of a ping, for example, would be very long if behind a large message. Hence, the requirement of handling control frames in the middle of a fragmented message.

Implementation Note: in absence of any extension a receiver doesn't have to buffer the whole frame in order to process it. For example if streaming API is used, a part of a frame can be delivered to the application. But note that that assumption might not hold true for all future WebSocket extensions.

5.5. Control Frames

Control frames are identified by opcodes where the most significant bit of the opcode is 1. Currently defined opcodes for control frames include 0x8 (Close), 0x9 (Ping), and 0xA (Pong). Opcodes 0xB-0xF are reserved for further control frames yet to be defined.

Control frames are used to communicate state about the WebSocket. Control frames can be interjected in the middle of a fragmented message.

All control frames MUST have a payload length of 125 bytes or less and MUST NOT be fragmented.

5.5.1. Close

The Close frame contains an opcode of 0x8.

The Close frame MAY contain a body (the "application data" portion of the frame) that indicates a reason for closing, such as an endpoint shutting down, an endpoint having received a frame too large, or an endpoint having received a frame that does not conform to the format expected by the other endpoint. If there is a body, the first two bytes of the body MUST be a 2-byte unsigned integer (in network byte order) representing a status code with value /code/ defined in Section 7.4. Following the 2-byte integer the body MAY contain UTF-8 encoded data with value /reason/, the interpretation of which is not defined by this specification. This data is not necessarily human readable, but may be useful for debugging or passing information relevant to the script that opened the connection. As the data is not guaranteed to be human readable, clients MUST NOT show it to end users.

Close frames sent from client to server must be masked as per Section 5.3.

The application MUST NOT send any more data frames after sending a

close frame.

If an endpoint receives a Close frame and that endpoint did not previously send a Close frame, the endpoint MUST send a Close frame in response. It SHOULD do so as soon as is practical. An endpoint MAY delay sending a close frame until its current message is sent (for instance, if the majority of a fragmented message is already sent, an endpoint MAY send the remaining fragments before sending a Close frame). However, there is no guarantee that the endpoint which has already sent a Close frame will continue to process data.

After both sending and receiving a close message, an endpoint considers the WebSocket connection closed, and MUST close the underlying TCP connection. The server MUST close the underlying TCP connection immediately; the client SHOULD wait for the server to close the connection but MAY close the connection at any time after sending and receiving a close message, e.g. if it has not received a TCP close from the server in a reasonable time period.

If a client and server both send a Close message at the same time, both endpoints will have sent and received a Close message and should consider the WebSocket connection closed and close the underlying TCP connection.

5.5.2. Ping

The Ping frame contains an opcode of 0x9.

Upon receipt of a Ping frame, an endpoint MUST send a Pong frame in response. It SHOULD do so as soon as is practical. Pong frames are discussed in Section 5.5.3.

An endpoint MAY send a Ping frame any time after the connection is established and before the connection is closed. NOTE: A ping frame may serve either as a keepalive, or to verify that the remote endpoint is still responsive.

5.5.3. Pong

The Pong frame contains an opcode of 0xA.

Section 5.5.2 details requirements that apply to both Ping and Pong frames.

A Pong frame sent in response to a Ping frame must have identical Application Data as found in the message body of the Ping frame being replied to.

If an endpoint receives a Ping frame and has not yet sent Pong frame(s) in response to previous Ping frame(s), the endpoint MAY elect to send a Pong frame for only the most recently processed Ping frame.

A Pong frame MAY be sent unsolicited. This serves as a unidirectional heartbeat. A response to an unsolicited pong is not expected.

5.6. Data Frames

Data frames (e.g. non-control frames) are identified by opcodes where the most significant bit of the opcode is 0. Currently defined opcodes for data frames include 0x1 (Text), 0x2 (Binary). Opcodes 0x3-0x7 are reserved for further non-control frames yet to be defined.

Data frames carry application-layer and/or extension-layer data. The opcode determines the interpretation of the data:

Text

The payload data is text data encoded as UTF-8. Note that a particular text frame might include a partial UTF-8 sequence, however the whole message MUST contain valid UTF-8. Invalid UTF-8 in reassembled messages is handled as described in Section 8.1.

Binary

The payload data is arbitrary binary data whose interpretation is solely up to the application layer.

5.7. Examples

This section is non-normative.

- A single-frame unmasked text message
  - 0x81 0x05 0x48 0x65 0x6c 0x6c 0x6f (contains "Hello")
- A single-frame masked text message
  - 0x81 0x85 0x37 0xfa 0x21 0x3d 0x7f 0x9f 0x4d 0x51 0x58 (contains "Hello")
- A fragmented unmasked text message

  * 0x01 0x03 0x48 0x65 0x6c (contains "Hel")

  * 0x80 0x02 0x6c 0x6f (contains "lo")

- Unmasked Ping request and masked Ping response

  * 0x89 0x05 0x48 0x65 0x6c 0x6c 0x6f (contains a body of "Hello", but the contents of the body are arbitrary)

  * 0x8a 0x85 0x37 0xfa 0x21 0x3d 0x7f 0x9f 0x4d 0x51 0x58 (contains a body of "Hello", matching the body of the ping)

- 256 bytes binary message in a single unmasked frame

  * 0x82 0x7E 0x0100 [256 bytes of binary data]

- 64KiB binary message in a single unmasked frame

  * 0x82 0x7F 0x0000000000010000 [65536 bytes of binary data]

5.8. Extensibility

The protocol is designed to allow for extensions, which will add capabilities to the base protocols. The endpoints of a connection MUST negotiate the use of any extensions during the opening handshake. This specification provides opcodes 0x3 through 0x7 and 0xB through 0xF, the extension data field, and the frame-rsv1, frame-rsv2, and frame-rsv3 bits of the frame header for use by extensions. The negotiation of extensions is discussed in further detail in Section 9.1. Below are some anticipated uses of extensions. This list is neither complete nor prescriptive.

- Extension data may be placed in the payload data before the application data.

- Reserved bits can be allocated for per-frame needs.

- Reserved opcode values can be defined.

- Reserved bits can be allocated to the opcode field if more opcode values are needed.

- A reserved bit or an "extension" opcode can be defined which allocates additional bits out of the payload data to define larger opcodes or more per-frame bits.

6. Sending and Receiving Data

6.1. Sending Data

To _Send a WebSocket Message_ comprising of /data/ over a WebSocket connection, an endpoint MUST perform the following steps.

1. The endpoint MUST ensure the WebSocket connection is in the OPEN state (cf. Section 4.1 and Section 4.2.2.) If at any point the state of the WebSocket connection changes, the endpoint MUST abort the following steps.

2. An endpoint MUST encapsulate the /data/ in a WebSocket frame as defined in Section 5.2. If the data to be sent is large, or if the data is not available in its entirety at the point the endpoint wishes to begin sending the data, the endpoint MAY alternately encapsulate the data in a series of frames as defined in Section 5.4.

3. The opcode (frame-opcode) of the first frame containing the data MUST be set to the appropriate value from Section 5.2 for data that is to be interpreted by the recipient as text or binary data.

4. The FIN bit (frame-fin) of the last frame containing the data MUST be set to 1 as defined in Section 5.2.

5. If the data is being sent by the client, the frame(s) MUST be masked as defined in Section 5.3.

6. If any extensions (Section 9) have been negotiated for the WebSocket connection, additional considerations may apply as per the definition of those extensions.

7. The frame(s) that have been formed MUST be transmitted over the underlying network connection.

6.2. Receiving Data

To receive WebSocket data, an endpoint listens on the underlying network connection. Incoming data MUST be parsed as WebSocket frames as defined in Section 5.2. If a control frame (Section 5.5) is received, the frame MUST be handled as defined by Section 5.5. Upon receiving a data frame (Section 5.6), the endpoint MUST note the /type/ of the data as defined by the Opcode (frame-opcode) from Section 5.2. The _Application Data_ from this frame is defined as the /data/ of the message. If the frame comprises an unfragmented message (Section 5.4), it is said that _A WebSocket Message Has Been

Received_ with type /type/ and data /data/. If the frame is part of a fragmented message, the _Application Data_ of the subsequent data frames is concatenated to form the /data/. When the last fragment is received as indicated by the FIN bit (frame-fin), it is said that _A WebSocket Message Has Been Received_ with data /data/ (comprised of the concatenation of the _Application Data_ of the fragments) and type /type/ (noted from the first frame of the fragmented message). Subsequent data frames MUST be interpreted as belonging to a new WebSocket Message.

Extensions (Section 9) MAY change the semantics of how data is read, specifically including what comprises a message boundary. Extensions, in addition to adding "Extension data" before the "Application data" in a payload, MAY also modify the "Application data" (such as by compressing it).

A server MUST remove masking for data frames received from a client as described in Section 5.3.

7. Closing the connection

7.1. Definitions

7.1.1. Close the WebSocket Connection

To _Close the WebSocket Connection_, an endpoint closes the underlying TCP connection. An endpoint SHOULD use a method that cleanly closes the TCP connection, as well as the TLS session, if applicable, discarding any trailing bytes that may be received. An endpoint MAY close the connection via any means available when necessary, such as when under attack.

The underlying TCP connection, in most normal cases, SHOULD be closed first by the server, so that it holds the TIME_WAIT state and not the client (as this would prevent it from re-opening the connection for 2 MSL, while there is no corresponding server impact as a TIME_WAIT connection is immediately reopened upon a new SYN with a higher seq number). In abnormal cases (such as not having received a TCP Close from the server after a reasonable amount of time) a client MAY initiate the TCP Close. As such, when a server is instructed to _Close the WebSocket Connection_ it SHOULD initiate a TCP Close immediately, and when a client is instructed to do the same, it SHOULD wait for a TCP Close from the server.

As an example of how to obtain a clean closure in C using Berkeley sockets, one would call shutdown() with SHUT_WR on the socket, call recv() until obtaining a return value of 0 indicating that the peer has also performed an orderly shutdown, and finally calling close() on the socket.

7.1.2. Start the WebSocket Closing Handshake

To _Start the WebSocket Closing Handshake_ with a status code (Section 7.4) /code/ and an optional close reason (Section 7.1.6) /reason/, an endpoint MUST send a Close control frame, as described in Section 5.5.1 whose status code is set to /code/ and whose close reason is set to /reason/. Once an endpoint has both sent and received a Close control frame, that endpoint SHOULD _Close the WebSocket Connection_ as defined in Section 7.1.1.

7.1.3. The WebSocket Closing Handshake is Started

Upon either sending or receiving a Close control frame, it is said that _The WebSocket Closing Handshake is Started_ and that the WebSocket connection is in the CLOSING state.

7.1.4. The WebSocket Connection is Closed

When the underlying TCP connection is closed, it is said that _The WebSocket Connection is Closed_ and that the WebSocket connection is in the CLOSED state. If the tcp connection was closed after the WebSocket closing handshake was completed, the WebSocket connection is said to have been closed _cleanly_.

If the WebSocket connection could not be established, it is also said that _The WebSocket Connection is Closed_, but not cleanly.

7.1.5. The WebSocket Connection Close Code

As defined in Section 5.5.1 and Section 7.4, a Close control frame may contain a status code indicating a reason for closure. A closing of the WebSocket connection may be initiated by either endpoint, potentially simultaneously. _The WebSocket Connection Close Code_ is defined as the status code (Section 7.4) contained in the first Close control frame received by the application implementing this protocol. If this Close control frame contains no status code, _The WebSocket Connection Close Code_ is considered to be 1005. If _The WebSocket Connection is Closed_ and no Close control frame was received by the endpoint (such as could occur if the underlying transport connection is lost), _The WebSocket Connection Close Code_ is considered to be 1006.

NOTE: Two endpoints may not agree on the value of _The WebSocket Connection Close Code_. As an example, if the remote endpoint sent a Close frame but the local application has not yet read the data containing the Close frame from its socket's receive buffer, and the local application independently decided to close the connection and send a Close frame, both endpoints will have sent and received a Close frame, and will not send further Close frames. Each endpoint will see the Connection Close Code sent by the other end as the _WebSocket Connection Close Code_. As such, it is possible that the two endpoints may not agree on the value of _The WebSocket Connection Close Code_ in the case that both endpoints _Start the WebSocket Closing Handshake_ independently and at roughly the same time.

7.1.6. The WebSocket Connection Close Reason

As defined in Section 5.5.1 and Section 7.4, a Close control frame may contain a status code indicating a reason for closure, followed by UTF-8 encoded data, the interpretation of said data being left to the endpoints and not defined by this protocol. A closing of the WebSocket connection may be initiated by either endpoint, potentially simultaneously. _The WebSocket Connection Close Reason_ is defined as the UTF-8 encoded data following the status code (Section 7.4)

contained in the first Close control frame received by the application implementing this protocol. If there is no such data in the Close control frame, _The WebSocket Connection Close Reason_ is the empty string.

NOTE: Following the same logic as noted in Section 7.1.5, two endpoints may not agree on _The WebSocket Connection Close Reason_.

7.1.7. Fail the WebSocket Connection

Certain algorithms and specifications require an endpoint to _Fail the WebSocket Connection_. To do so, the client MUST _Close the WebSocket Connection_, and MAY report the problem to the user (which would be especially useful for developers) in an appropriate manner. Similarly, to do so, the server MUST _Close the WebSocket Connection_, and SHOULD log the problem.

If _The WebSocket Connection is Established_ prior to the point where the endpoint is required to _Fail the WebSocket Connection_, the endpoint SHOULD send a Close frame with an appropriate status code Section 7.4 before proceeding to _Close the WebSocket Connection_. An endpoint MAY omit sending a Close frame if it believes the other side is unlikely to be able to receive and process the close frame, due to the nature of the error that led to the WebSocket connection being failed in the first place. An endpoint MUST NOT continue to attempt to process data (including a responding Close frame) from the remote endpoint after being instructed to _Fail the WebSocket Connection_.

Except as indicated above or as specified by the application layer (e.g. a script using the WebSocket API), clients SHOULD NOT close the connection.

7.2. Abnormal Closures

7.2.1. Client-Initiated Closure

Certain algorithms, namely during the opening handshake, require the client to _Fail the WebSocket Connection_. To do so, the client MUST _Fail the WebSocket Connection_ as defined in Section 7.1.7.

If at any point the underlying transport layer connection is unexpectedly lost, the client MUST _Fail the WebSocket Connection_.

Except as indicated above or as specified by the application layer (e.g. a script using the WebSocket API), clients SHOULD NOT close the connection.

7.2.2. Server-Initiated Closure

Certain algorithms require or recommend that the server _Abort the WebSocket Connection_ during the opening handshake. To do so, the server MUST simply _Close the WebSocket Connection_ (Section 7.1.1).

7.2.3. Recovering From Abnormal Closure

Abnormal closures may be caused by any number of reasons. Such closures could be the result of a transient error, in which case reconnecting may lead to a good connection and a resumption of normal operations. Such closures may also be the result of a nontransient problem, in which case if each deployed client experiences an abnormal closure and immediately and persistently tries to reconnect, the server may experience what amounts to a denial of service attack by a large number of clients trying to reconnect. The end result of such a scenario could be that the service is unable to recover, or recovey is made much more difficult, in any sort of timely manner.

To prevent this, clients SHOULD use some form of backoff when trying to reconnect after abnormal closures as described in this section.

The first reconnect attempt SHOULD be delayed by a random amount of time. The parameters by which this random delay is chosen are left to the client to decide; a value chosen randomly between 0 and 5 seconds is a reasonable initial delay though clients MAY choose a different interval from which to select a delay length based on implementation experience and particular application.

Should the first reconnect attempt fail, subsequent reconnect attempts SHOULD be delayed by increasingly longer amounts of time, using a method such as truncated binary exponential backoff.

7.3. Normal Closure of Connections

Servers MAY close the WebSocket connection whenever desired. Clients SHOULD NOT close the WebSocket connection arbitrarily. In either case, an endpoint initiates a closure by following the procedures to _Start the WebSocket Closing Handshake_ (Section 7.1.2).

7.4. Status Codes

When closing an established connection (e.g. when sending a Close frame, after the opening handshake has completed), an endpoint MAY indicate a reason for closure. The interpretation of this reason by an endpoint, and the action an endpoint should take given this reason, are left undefined by this specification. This specification defines a set of pre-defined status codes, and specifies which ranges

may be used by extensions, frameworks, and end applications. The status code and any associated textual message are optional components of a Close frame.

7.4.1. Defined Status Codes

Endpoints MAY use the following pre-defined status codes when sending a Close frame.

1000

1000 indicates a normal closure, meaning whatever purpose the connection was established for has been fulfilled.

1001

1001 indicates that an endpoint is "going away", such as a server going down, or a browser having navigated away from a page.

1002

1002 indicates that an endpoint is terminating the connection due to a protocol error.

1003

1003 indicates that an endpoint is terminating the connection because it has received a type of data it cannot accept (e.g. an endpoint that understands only text data MAY send this if it receives a binary message).

1004

Reserved. The specific meaning might be defined in the future.

1005

1005 is a reserved value and MUST NOT be set as a status code in a Close control frame by an endpoint. It is designated for use in applications expecting a status code to indicate that no status code was actually present.

1006

1006 is a reserved value and MUST NOT be set as a status code in a Close control frame by an endpoint. It is designated for use in applications expecting a status code to indicate that the connection was closed abnormally, e.g. without sending or

receiving a Close control frame.

1007

1007 indicates that an endpoint is terminating the connection because it has received data that was supposed to be UTF-8 (such as in a text frame) that was in fact not valid UTF-8 [RFC3629].

1008

1008 indicates that an endpoint is terminating the connection because it has received a message that violates its policy. This is a generic status code that can be returned when there is no other more suitable status code (e.g. 1003 or 1009), or if there is a need to hide specific details about the policy.

1009

1009 indicates that an endpoint is terminating the connection because it has received a message which is too big for it to process.

1010

1010 indicates that an endpoint (client) is terminating the connection because it has expected the server to negotiate one or more extension, but the server didn't return them in the response message of the WebSocket handshake. The list of extensions which are needed SHOULD appear in the /reason/ part of the Close frame. Note that this status code is not used by the server, because it can fail the WebSocket handshake instead.

7.4.2. Reserved Status Code Ranges

0-999

Status codes in the range 0-999 are not used.

1000-2999

Status codes in the range 1000-2999 are reserved for definition by this protocol, its future revisions, and extensions specified in a permanent and readily available public specification.

3000-3999

Status codes in the range 3000-3999 are reserved for use by libraries, frameworks and application. These status codes are

registered directly with IANA. The interpretation of these codes is undefined by this protocol.

4000-4999

Status codes in the range 4000-4999 are reserved for private use and thus can’t be registered. Such codes can be used by prior agreements between WebSocket applications. The interpretation of these codes is undefined by this protocol.

8. Error Handling

8.1. Handling Errors in UTF-8 Encoded Data

When an endpoint is to interpret a byte stream as UTF-8 but finds that the byte stream is not in fact a valid UTF-8 stream, that endpoint MUST _Fail the WebSocket Connection_. This rule applies both during the opening handshake and during subsequent data exchange.

9. Extensions

WebSocket clients MAY request extensions to this specification, and WebSocket servers MAY accept some or all extensions requested by the client. A server MUST NOT respond with any extension not requested by the client. If extension parameters are included in negotiations between the client and the server, those parameters MUST be chosen in accordance with the specification of the extension to which the parameters apply.

9.1. Negotiating Extensions

A client requests extensions by including a "Sec-WebSocket-Extensions" header field, which follows the normal rules for HTTP header fields (see [RFC2616] section 4.2) and the value of the header field is defined by the following ABNF. Note that unlike other section of the document this section is using ABNF syntax/rules from [RFC2616], including "implied *LWS rule". If a value is received by either the client or the server during negotiation that does not conform to the ABNF below, the recipient of such malformed data MUST immediately _Fail the WebSocket Connection_.

```
Sec-WebSocket-Extensions = extension-list
extension-list = 1#extension
extension = extension-token *( ";" extension-param )
extension-token = registered-token
registered-token = token
extension-param = token [ "=" token ]
```

Note that like other HTTP header fields, this header field MAY be split or combined across multiple lines. Ergo, the following are equivalent:

```
Sec-WebSocket-Extensions: foo
Sec-WebSocket-Extensions: bar; baz=2
```

is exactly equivalent to

```
Sec-WebSocket-Extensions: foo, bar; baz=2
```

Any extension-token used MUST be a registered token (see Section 11.4). The parameters supplied with any given extension MUST be defined for that extension. Note that the client is only offering to use any advertised extensions, and MUST NOT use them unless the server indicates that it wishes to use the extension.

Note that the order of extensions is significant. Any interactions between multiple extensions MAY be defined in the documents defining

the extensions. In the absence of such definition, the interpretation is that the header fields listed by the client in its request represent a preference of the header fields it wishes to use, with the first options listed being most preferable. The extensions listed by the server in response represent the extensions actually in use for the connection. Should the extensions modify the data and/or framing, the order of operations on the data should be assumed to be the same as the order in which the extensions are listed in the server's response in the opening handshake.

For example, if there are two extensions "foo" and "bar", if the header field |Sec-WebSocket-Extensions| sent by the server has the value "foo, bar" then operations on the data will be made as bar(foo(data)), be those changes to the data itself (such as compression) or changes to the framing thay may "stack".

Non-normative examples of acceptable extension header fields (note that long lines are folded for readability):

```
Sec-WebSocket-Extensions: deflate-stream
Sec-WebSocket-Extensions: mux; max-channels=4; flow-control,
 deflate-stream
Sec-WebSocket-Extensions: private-extension
```

A server accepts one or more extensions by including a |Sec-WebSocket-Extensions| header field containing one or more extensions which were requested by the client. The interpretation of any extension parameters, and what constitutes a valid response by a server to a requested set of parameters by a client, will be defined by each such extension.

9.2. Known Extensions

Extensions provide a mechanism for implementations to opt-in to additional protocol features. This document doesn't define any extension but implementations MAY use extensions defined separately.

10. Security Considerations

This section describes some security considerations applicable to the WebSocket protocol. Specific security considerations are described in subsections of this section.

10.1. Non-Browser Clients

Many threats anticipated by the WebSocket protocol protect from malicious JavaScript running inside a trusted application such as a web browser, for example checking of the "Origin" header field (see below). See Section 1.6 for additional details. Such assumptions don’t hold true in a case of a more capable client.

While this protocol is intended to be used by scripts in Web pages, it can also be used directly by hosts. Such hosts are acting on their own behalf, and can therefore send fake "Origin" header fields, misleading the server. Servers should therefore be careful about assuming that they are talking directly to scripts from known origins, and must consider that they might be accessed in unexpected ways. In particular, a server should not trust that any input is valid.

EXAMPLE: For example, if the server uses input as part of SQL queries, all input text should be escaped before being passed to the SQL server, lest the server be susceptible to SQL injection.

10.2. Origin Considerations

Servers that are not intended to process input from any Web page but only for certain sites SHOULD verify the "Origin" field is an origin they expect, and should only respond with the corresponding "Sec-WebSocket-Accept" if it is an accepted origin.

The "Origin" header field protects from the attack cases when the untrusted party is typically the author of a JavaScript application that is executing in the context of the trusted client. The client itself can contact the server and via the mechanism of the "Origin" header field, determine whether to extend those communication privileges to the JavaScript application. The intent is not to prevent non-browsers from establishing connections, but rather to ensure that trusted browsers under the control of potentially malicious JavaScript cannot fake a WebSocket handshake.

10.3. Attacks On Infrastructure (Masking)

In addition to endpoints being the target of attacks via WebSockets, other parts of web infrastructure, such as proxies, may be the subject of an attack.

As this protocol was being developed, an experiment was conducted to demonstrate a class of attacks on proxies that led to the poisoning of caching proxies deployed in the wild. The general form of the attack was to establish a connection to a server under the "attacker's" control, perform an UPGRADE on the HTTP connection similar to what the WebSocket protocol does to establish a connection, and to subsequently send data over that UPGRADEd connection that looked like a GET request for a specific known resource (which in an attack would likely be something like a widely deployed script for tracking hits, or a resource on an ad-serving network). The remote server would respond with something that looked like a response to the fake GET request, and this response would be cached by a nonzero percentage of deployed intermediaries, thus poisioning the cache. The net effect of this attack would be that if a user could be convinced to visit a website the attacker controlled, the attacker could potentially poison the cache for that user and other users behind the same cache and run malicious script on other origins, compromising the web security model.

To avoid such attacks on deployed intermediaries, the working group decided to adopt a solution that would provably protect against such attacks. There were many proposed solutions that people argued "should" protect against the above attacks, such as adding in more random data and null bytes to the handshake, starting each frame with a byte that has the first (highest order) bit set such that the data appears to be non-ASCII, and so forth, but in the end none of these solutions were provably secure. The deployed intermediaries were already not conforming to existing specifications, and given that we can't possibly enumerate all of the ways in which such nonconformities could exhibit themselves and that we cannot exhaustively discover and test each nonconformant intermediary against each possible attack, there was consensus to adopt an approach that did not require people to reason about how nonconformant intermediaries might behave. Namely, the working group decided to mask all data from the client to the server, so that the remote script (attacker) does not have control over how the data being sent appears on the wire, and thus cannot construct a message that could be mis- interpreted by an intermediary as an HTTP request.

It is necessary that the masking key is chosen randomly for each frame. If the same key is used, or a decipherable pattern exists for how the next key is chosen, the attacker can send a message that,

when masked, could appear to be an HTTP request (by taking the message the attacker wishes to see on the wire, and masking it with the next masking key to be used, when the client applies the masking key it will effectively unmask the data.)

It is also necessary that once the transmission of a frame from a client has begun, the payload (application supplied data) of that frame must not be capable of being modified by the application. Otherwise, an attacker could send a long frame where the initial data was a known value (such as all zeros), compute the masking key being used upon receipt of the first part of the data, and then modify the data that is yet to be sent in the frame to appear as an HTTP request when masked. (This is essentially the same problem described in the previous paragraph with using a known or predictable masking key.) If additional data is to be sent or data to be sent is somehow changed, that new or changed data must be sent in a new frame and thus with a new masking key. In short, once transmission of a frame begins, the contents must not be modifiable by the remote script (application).

The threat model being protected against is one in which the client sends data that appears to be a HTTP request. As such, the channel that needs to be masked is the data from the client to the server. The data from the server to the client can be made to look like a response, but to accomplish this request the client must also be able to forge a request. As such, it was not deemend necessary to mask data in both directions (the data from the server to the client is not masked).

10.4. Implementation-Specific Limits

Implementations MAY impose implementation-specific limits on otherwise unconstrained inputs, e.g. to prevent denial of service attacks, to guard against running out of memory, or to work around platform-specific limitations. For example implementations might impose limit on frame sizes and the total message size after reassembly from multiple frames.

10.5. WebSocket client authentication

This protocol doesn’t prescribe any particular way that servers can authenticate clients during the WebSocket handshake. The WebSocket server can use any client authentication mechanism available to a generic HTTP server, such as Cookies, HTTP Authentication, TLS authentication.

10.6. Connection confidentiality and integrity

Communications confidentiality and integrity is provided by running the WebSocket protocol over TLS (wss URIs).

For connections using TLS, the amount of benefit provided by TLS depends greatly on the strength of the algorithms negotiated during the TLS handshake. For example some TLS cipher mechanisms don't provide connection confidentiality. To achieve reasonable levels of protections, clients should use only Strong TLS algorithms. "Web Security Context: User Interface Guidelines" [W3C.REC-wsc-ui-20100812] discusses what constitutes Strong TLS algorithms.

10.7. Handling of invalid data

Incoming data MUST always be validated by both clients and servers. If at any time an endpoint is faced with data that it does not understand, or that violates some criteria by which the endpoint determines safety of input, or when the endpoint sees an opening handshake that does not correspond to the values it is expecting (e.g. incorrect path or origin in the client request), the endpoint MAY drop the TCP connection. If the invalid data received after a successful WebSocket handshake, the endpoint SHOULD send a Close frame with an appropriate status code Section 7.4 before proceeding to _Close the WebSocket Connection_. Use of a Close frame with an appropriate status code can help in diagnosing the problem. If the invalid data is sent during the WebSocket handshake the server SHOULD return an appropriate HTTP [RFC2616] status code.

A common class of security problems arise when sending text data using using the wrong encoding. This protocol specifies that messages with a Text data type (as opposed to Binary or other types) contain UTF-8 encoded data. Although the length is still indicated and applications implementing this protocol should use the length to determine where the frame actually ends, sending data in an improper encoding may still break assumptions applications built on top of this protocol may make, leading from anything to misinterpretation of data to loss of data to potential security bugs.

11. IANA Considerations

11.1. Registration of new URI Schemes

11.1.1. Registration of "ws" Scheme

A |ws| URI identifies a WebSocket server and resource name.

URI scheme name.
ws

Status.
Permanent.

URI scheme syntax.
In ABNF terms using the terminals from the URI specifications: [RFC5234] [RFC3986]

```
"ws:" "//" authority path-abempty [ "?" query ]
```

The <path-abempty> and <query> [RFC3986] components form the resource name sent to the server to identify the kind of service desired. Other components have the meanings described in RFC3986.

URI scheme semantics.
The only operation for this scheme is to open a connection using the WebSocket protocol.

Encoding considerations.
Characters in the host component that are excluded by the syntax defined above MUST be converted from Unicode to ASCII by applying the IDNA ToASCII algorithm to the Unicode host name, with both the AllowUnassigned and UseSTD3ASCIIRules flags set, and using the result of this algorithm as the host in the URI. [RFC3490]

Characters in other components that are excluded by the syntax defined above MUST be converted from Unicode to ASCII by first encoding the characters as UTF-8 and then replacing the corresponding bytes using their percent-encoded form as defined in the URI and IRI specifications. [RFC3986] [RFC3987]

Applications/protocols that use this URI scheme name.
WebSocket protocol.

Interoperability considerations.
Use of WebSocket requires use of HTTP version 1.1 or higher.

Security considerations.
See "Security considerations" section above.

Contact.
HYBI WG <hybi@ietf.org>

Author/Change controller.
IETF <iesg@ietf.org>

References.
RFC XXXX

11.1.2. Registration of "wss" Scheme

A |wss| URI identifies a WebSocket server and resource name, and indicates that traffic over that connection is to be protected via TLS (including standard benefits of TLS such as data confidentiality and integrity, and endpoint authentication).

URI scheme name.
wss

Status.
Permanent.

URI scheme syntax.
In ABNF terms using the terminals from the URI specifications: [RFC5234] [RFC3986]

```
"wss:" "//" authority path-abempty [ "?" query ]
```

The <path-abempty> and <query> components form the resource name sent to the server to identify the kind of service desired. Other components have the meanings described in RFC3986.

URI scheme semantics.
The only operation for this scheme is to open a connection using the WebSocket protocol, encrypted using TLS.

Encoding considerations.
Characters in the host component that are excluded by the syntax defined above MUST be converted from Unicode to ASCII by applying the IDNA ToASCII algorithm to the Unicode host name, with both the AllowUnassigned and UseSTD3ASCIIRules flags set, and using the result of this algorithm as the host in the URI. [RFC3490]

Characters in other components that are excluded by the syntax defined above MUST be converted from Unicode to ASCII by first

encoding the characters as UTF-8 and then replacing the corresponding bytes using their percent-encoded form as defined in the URI and IRI specification. [RFC3986] [RFC3987]

Applications/protocols that use this URI scheme name.
WebSocket protocol over TLS.

Interoperability considerations.
Use of WebSocket requires use of HTTP version 1.1 or higher.

Security considerations.
See "Security considerations" section above.

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References.
RFC XXXX

11.2. Registration of the "WebSocket" HTTP Upgrade Keyword

This section defines a keyword for registration in the "HTTP Upgrade Tokens" registry as per RFC 2817 [RFC2817].

Name of token.
WebSocket

Author/Change controller.
IETF <iesg@ietf.org>

Contact.
HYBI <hybi@ietf.org>

References.
RFC XXXX

11.3. Registration of new HTTP Header Fields

11.3.1. Sec-WebSocket-Key

This section describes a header field for registration in the Permanent Message Header Field Registry. [RFC3864]

Header field name
Sec-WebSocket-Key

Applicable protocol
http

Status
standard

Author/Change controller
IETF

Specification document(s)
RFC XXXX

Related information
This header field is only used for WebSocket opening handshake.

The |Sec-WebSocket-Key| header field is used in the WebSocket opening handshake. It is sent from the client to the server to provide part of the information used by the server to prove that it received a valid WebSocket opening handshake. This helps ensure that the server does not accept connections from non-WebSocket clients (e.g. HTTP clients) that are being abused to send data to unsuspecting WebSocket servers.

11.3.2. Sec-WebSocket-Extensions

This section describes a header field for registration in the Permanent Message Header Field Registry. [RFC3864]

Header field name
Sec-WebSocket-Extensions

Applicable protocol
http

Status
standard

Author/Change controller
IETF

Specification document(s)
RFC XXXX

Related information
This header field is only used for WebSocket opening handshake.

The |Sec-WebSocket-Extensions| header field is used in the WebSocket opening handshake. It is initially sent from the client to the server, and then subsequently sent from the server to the client, to agree on a set of protocol-level extensions to use for the duration of the connection.

11.3.3. Sec-WebSocket-Accept

This section describes a header field for registration in the Permanent Message Header Field Registry. [RFC3864]

Header field name
Sec-WebSocket-Accept

Applicable protocol
http

Status
standard

Author/Change controller
IETF

Specification document(s)
RFC XXXX

Related information
This header field is only used for WebSocket opening handshake.

The |Sec-WebSocket-Accept| header field is used in the WebSocket opening handshake. It is sent from the server to the client to confirm that the server is willing to initiate the connection.

11.3.4. Sec-WebSocket-Protocol

This section describes a header field for registration in the Permanent Message Header Field Registry. [RFC3864]

Header field name
Sec-WebSocket-Protocol

Applicable protocol
http

Status
standard

Author/Change controller
IETF

Specification document(s)
RFC XXXX

Related information
This header field is only used for WebSocket opening handshake.

The |Sec-WebSocket-Protocol| header field is used in the WebSocket opening handshake. It is sent from the client to the server and back from the server to the client to confirm the subprotocol of the connection. This enables scripts to both select a subprotocol and be sure that the server agreed to serve that subprotocol.

11.3.5. Sec-WebSocket-Version

This section describes a header field for registration in the Permanent Message Header Field Registry [RFC3864].

Header field name
Sec-WebSocket-Version

Applicable protocol
http

Status
standard

Author/Change controller
IETF

Specification document(s)
RFC XXXX

Related information
This header field is only used for WebSocket opening handshake.

The |Sec-WebSocket-Version| header field is used in the WebSocket opening handshake. It is sent from the client to the server to indicate the protocol version of the connection. This enables servers to correctly interpret the opening handshake and subsequent data being sent from the data, and close the connection if the server cannot interpret that data in a safe manner. The |Sec-WebSocket-Version| header field is also sent from the server to the client on

WebSocket handshake error, when the version received from the client does not match a version understood by the server. In such a case the header field includes the protocol version(s) supported by the server.

Note that there is no expectation that higher version numbers are necessarily backward compatible with lower version numbers.

11.4. WebSocket Extension Name Registry

This specification requests the creation of a new IANA registry for WebSocket Extension names to be used with the WebSocket protocol in accordance with the principles set out in RFC 5226 [RFC5226].

As part of this registry IANA will maintain the following information:

Extension Identifier
: The identifier of the extension, as will be used in the Sec-WebSocket-Extension header field registered in Section 11.3.2 of this specification. The value must conform to the requirements for an extension-token as defined in Section 9.1 of this specification.

Extension Common Name
: The name of the extension, as the extension is generally referred to.

Extension Definition
: A reference to the document in which the extension being used with the WebSocket protocol is defined.

Known Incompatible Extensions
: A list of extension identifiers with which this extension is known to be incompatible.

WebSocket Extension names are to be subject to "First Come First Served" IANA registration policy [RFC5226].

There are no initial values in this registry.

11.5. WebSocket Subprotocol Name Registry

This specification requests the creation of a new IANA registry for WebSocket Subprotocol names to be used with the WebSocket protocol in accordance with the principles set out in RFC 5226 [RFC5226].

As part of this registry IANA will maintain the following

information:

Subprotocol Identifier
: The identifier of the subprotocol, as will be used in the Sec-WebSocket-Protocol header field registered in Section 11.3.4 of this specification. The value must conform to the requirements given in Paragraph 10 of Section 4.1 of this specification, namely the value must be a token as defined by RFC 2616 [RFC2616].

Subprotocol Common Name
: The name of the subprotocol, as the subprotocol is generally referred to.

Subprotocol Definition
: A reference to the document in which the subprotocol being used with the WebSocket protocol is defined.

WebSocket Subprotocol names are to be subject to "First Come First Served" IANA registration policy [RFC5226].

11.6. WebSocket Version Number Registry

This specification requests the creation of a new IANA registry for WebSocket Version Numbers to be used with the WebSocket protocol in accordance with the principles set out in RFC 5226 [RFC5226].

As part of this registry IANA will maintain the following information:

Version Number
: The version number to be used in the Sec-WebSocket-Version as specified in Section 4.1 of this specification. The value must be a non negative integer in the range between 0 and 255 (inclusive).

Reference
: The RFC requesting a new version number.

WebSocket Version Numbers are to be subject to "IETF Review" IANA registration policy [RFC5226]. In order to improve interoperability with intermediate versions published in Internet Drafts, version numbers associated with such drafts might be registered in this registry. Note that "IETF Review" applies to registrations corresponding to Internet Drafts.

IANA is asked to add initial values to the registry, with suggested numerical values as these have been used in past versions of this protocol.

Version Number	Reference
0	draft-ietf-hybi-thewebsocketprotocol-00
1	draft-ietf-hybi-thewebsocketprotocol-01
2	draft-ietf-hybi-thewebsocketprotocol-02
3	draft-ietf-hybi-thewebsocketprotocol-03
4	draft-ietf-hybi-thewebsocketprotocol-04
5	draft-ietf-hybi-thewebsocketprotocol-05
6	draft-ietf-hybi-thewebsocketprotocol-06
7	draft-ietf-hybi-thewebsocketprotocol-07
8	draft-ietf-hybi-thewebsocketprotocol-08
9	Reserved
10	Reserved
11	Reserved
12	Reserved
13	draft-ietf-hybi-thewebsocketprotocol-13

11.7. WebSocket Close Code Number Registry

This specification requests the creation of a new IANA registry for WebSocket Connection Close Code Numbers in accordance with the principles set out in RFC 5226 [RFC5226].

As part of this registry IANA will maintain the following information:

Status Code
: The Status Code which denotes a reson for a WebSocket connection closure as per Section 7.4 of this document. The status code is an integer number between 1000 and 4999 (inclusive).

Meaning
: The meaning of the status code. Each status code has to have a unique meaning.

Contact
: A contact for the entity reserving the status code.

Reference
: The stable document requesting the status codes and defining their meaning. This is required for status codes in the range 1000-2999, and recommended for status codes in the range 3000-3999.

WebSocket Close Code Numbers are to be subject to different registration requirements depending on their range. Unless otherwise specified, requests are subject to "Standards Action" IANA registration policy [RFC5226]. Requests for status codes for use by this protocol, its subsequent versions or extensions are subject to any one of "Standards Action", "Specification Required" (which implies "Designated Expert") or "IESG Review" IANA registration policies and should be granted status codes in the range 1000-2999. Requests for status codes for use by libraries, frameworks and applications are subject to "First Come First Served" IANA registration policy and should be granted in the range 3000-3999. The range of status codes from 4000-4999 is designated for Private Use. Requests should indicate whether they are requesting status codes for use by the WebSocket protocol (or a future version of the protocol) or by extensions, or by libraries/frameworks/applications.

IANA is asked to add initial values to the registry, with suggested numerical values as these have been used in past versions of this protocol.

Status Code	Meaning	Contact	Reference
1000	Normal Closure	hybi@ietf.org	RFC XXXX
1001	Going Away	hybi@ietf.org	RFC XXXX
1002	Protocol error	hybi@ietf.org	RFC XXXX
1003	Unsupported Data	hybi@ietf.org	RFC XXXX
1004	---Reserved----	hybi@ietf.org	RFC XXXX
1005	No Status Rcvd	hybi@ietf.org	RFC XXXX
1006	Abnormal Closure	hybi@ietf.org	RFC XXXX
1007	Invalid UTF-8	hybi@ietf.org	RFC XXXX
1008	Policy Violation	hybi@ietf.org	RFC XXXX
1009	Message Too Big	hybi@ietf.org	RFC XXXX
1010	Mandatory Ext.	hybi@ietf.org	RFC XXXX

11.8. WebSocket Opcode Registry

This specification requests the creation of a new IANA registry for WebSocket Opcodes in accordance with the principles set out in RFC 5226 [RFC5226].

As part of this registry IANA will maintain the following information:

Opcode
: The opcode denotes the frame type of the WebSocket frame, as defined in Section 5.2. The status code is an integer number between 0 and 15, inclusive.

Meaning
: The meaning of the opcode code.

Reference
: The specification requesting the opcode.

WebSocket Opcode numbers are subject to "Standards Action" IANA registration policy [RFC5226].

IANA is asked to add initial values to the registry, with suggested numerical values as these have been used in past versions of this protocol.

Opcode	Meaning	Reference
0	Continuation Frame	RFC XXXX
1	Text Frame	RFC XXXX
2	Binary Frame	RFC XXXX
8	Connection Close Frame	RFC XXXX
9	Ping Frame	RFC XXXX
10	Pong Frame	RFC XXXX

11.9. WebSocket Framing Header Bits Registry

This specification requests the creation of a new IANA registry for WebSocket Framing Header Bits in accordance with the principles set out in RFC 5226 [RFC5226]. This registry controls assignment of the bits marked RSV1, RSV2, and RSV3 in Section 5.2.

These bits are reserved for future versions or extensions of this specification.

WebSocket Framing Header Bits assignments are subject to "Standards Action" IANA registration policy [RFC5226].

12. Using the WebSocket protocol from Other Specifications

The WebSocket protocol is intended to be used by another specification to provide a generic mechanism for dynamic author-defined content, e.g. in a specification defining a scripted API.

Such a specification first needs to _Establish a WebSocket Connection_, providing that algorithm with:

- The destination, consisting of a /host/ and a /port/.
- A /resource name/, which allows for multiple services to be identified at one host and port.
- A /secure/ flag, which is true if the connection is to be encrypted, and false otherwise.
- An ASCII serialization of an origin that is being made responsible for the connection. [I-D.ietf-websec-origin]
- Optionally a string identifying a protocol that is to be layered over the WebSocket connection.

The /host/, /port/, /resource name/, and /secure/ flag are usually obtained from a URI using the steps to parse a WebSocket URI's components. These steps fail if the URI does not specify a WebSocket.

If at any time the connection is to be closed, then the specification needs to use the _Close the WebSocket Connection_ algorithm (Section 7.1.1).

Section 7.1.4 defines when _The WebSocket Connection is Closed_.

While a connection is open, the specification will need to handle the cases when _A WebSocket Message Has Been Received_ (Section 6.2).

To send some data /data/ to an open connection, the specification needs to _Send a WebSocket Message_ (Section 6.1).

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