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**WebRTC Data Channel Protocol**  
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**Abstract**

The Web Real-Time Communication (WebRTC) working group is charged to provide protocols to support for direct interactive rich communication using audio, video, and data between two peers' web-browsers. This document specifies an actual (minor) protocol for how the JS-layer dataChannel objects provide the data channels between the peers.

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## **1. Introduction**

The Data Channel Protocol is designed to provide, in the WebRTC context [[I-D.ietf-rtcweb-overview](#)], a generic transport service allowing Web Browser to exchange generic data in a bidirectional peer to peer fashion. As discussed in [[I-D.ietf-rtcweb-data-channel](#)] the protocol uses Stream Control Transmission Protocol (SCTP) [[RFC4960](#)] encapsulated on Datagram Transport Layer Security (DTLS) [[RFC6347](#)] as described in [[I-D.tuexen-tsvwg-sctp-dtls-encaps](#)] to benefit from their already standardized transport and security features.

## **2. Conventions**

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](#)].

## **3. Terminology**

This document uses the following terms:

Association: An SCTP association.

Stream: A unidirectional stream of an SCTP association. It is uniquely identified by a stream identifier.

Channel: A bidirectional channel consisting of two SCTP streams.

## **4. Opening Handshake**

The opening handshake is based on the multimedia session description exchange that happens between the browsers, typically through a Web Server acting as the signaling service.

[[I-D.ietf-mmusic-sctp-sdp](#)] defines the protocol identifier, 'SCTP/DTLS', and defines how to establish an SCTP association over DTLS using the Session Description Protocol (SDP).

The SCTP association is created with the number of streams specified by the application, and if not specified, then it SHOULD default to 16 streams.

It is recommended that additional streams be available dynamically based on [[RFC6525](#)].







DATA\_CHANNEL\_UNRELIABLE (0x02): TBD.

DATA\_CHANNEL\_PARTIAL\_RELIABLE\_REXMIT (0x03): The channel provides a partial reliable bi-directional Communication channel. User messages will not be retransmitted more times than specified in the Reliability Parameter.

**DATA\_CHANNEL\_PARTIAL\_RELIABLE\_TIMED (0x04):** The channel provides a partial reliable bi-directional Communication channel. User messages might not be transmitted or retransmitted after a specified life-time given in milli-seconds in the Reliability Parameter. This life-time starts when providing the user message to the Javascript engine.

Flags: 2 bytes (unsigned integer)

This field contains the bitwise OR of the following flags:

DATA\_CHANNEL\_FLAG\_OUT\_OF\_ORDER\_ALLOWED (0x0001): If this flag is set, the channel does not need to preserve the message sequencing.

Reliability Parameter: 2 bytes (unsigned integer)

This field is ignored if a reliable channel is used. If a partial reliable channel with limited number of retransmissions is used, this field specifies the number of retransmissions. If a partial reliable channel with limited lifetime is used, this field specifies the maximum life-time in milli seconds.

Priority: 2 bytes (integer)

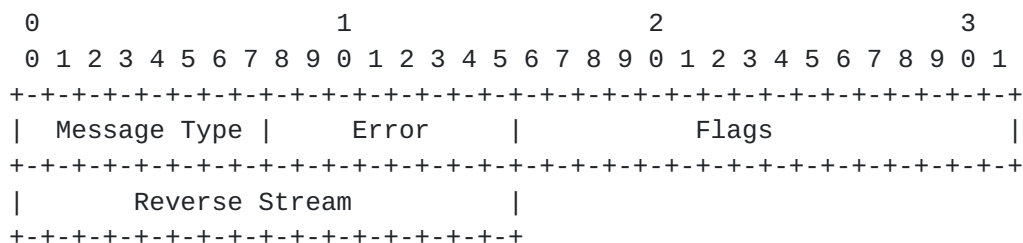
The priority of the channel.

Label: Variable Length (sequence of characters)

The name of the channel.

### 5.2. DATA\_CHANNEL\_OPEN\_RESPONSE Message

This message is sent in response to an `DATA_CHANNEL_OPEN_REQUEST` message on the stream used for user messages using the channel. Messages with the error field set to non-0 values can be sent on any stream; it is suggested that they be sent (if possible) on an unused stream.



Message Type: 1 byte (unsigned integer)

This field holds the IANA defined message type for the the DATA\_CHANNEL\_OPEN\_RESPONSE message. The suggested value of this field for IANA is 0x01.





Error: 1 byte (unsigned integer)

TBD.

Flags: 2 bytes (unsigned integer)

TBD.

Reverse Stream: 2 bytes (unsigned integer)

The identifier for the incoming stream of the channel. The corresponding DATA\_CHANNEL\_OPEN\_REQUEST message was received on this stream.

### 5.3. DATA\_CHANNEL\_ACK Message

This message is sent in response to an DATA\_CHANNEL\_OPEN\_RESPONSE message on the stream used for user messages using the channel. Reception of this message tells the opener that the channel setup handshake is complete.

```

      0                   1                   2                   3
      0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
| Message Type |
+---+---+---+---+---+---+---+

```

Message Type: 1 byte (unsigned integer)

This field holds the IANA defined message type for the the DATA\_CHANNEL\_ACK message. The suggested value of this field for IANA is 0x02.

## 6. Procedures

### 6.1. Adding a Channel

When one side wants to add a channel, it picks an unused outgoing stream; if no unused streams are available a negotiation to increase the number is done. It then sends a DATA\_CHANNEL\_OPEN\_REQUEST control message on the outgoing stream.

When an DATA\_CHANNEL\_OPEN\_REQUEST is received on an incoming stream, an unused outgoing stream is picked; if no unused streams are available a negotiation to increase the number is done. A DATA\_CHANNEL\_OPEN\_RESPONSE message is sent on the outgoing stream, with the Reverse Stream field set to the incoming stream the DATA\_CHANNEL\_OPEN\_REQUEST message came in on.

When a DATA\_CHANNEL\_OPEN\_RESPONSE is received, the Reverse Stream value is matched against all pending DATA\_CHANNEL\_OPEN\_REQUEST messages. If no match can be found, the DATA\_CHANNEL\_OPEN\_RESPONSE message SHOULD be ignored. If a match is found, then if the error



value is 0 a DATA\_CHANNEL\_ACK message is sent on the originator's outgoing Stream for the channel. If the error value is non-zero, the open failed, and the originator SHOULD close down the originally-selected outgoing stream and notify the application.

The channel\_type and reliability\_parameters fields of the DATA\_CHANNEL\_OPEN\_REQUEST message MUST be used to set up the reverse side of the Data Channel so that both directions use the same options. So both directions are either reliable or use the PR-SCTP extension defined in [[RFC3758](#)] using the same policy and parameter.

## **6.2. Closing a Channel**

Data Channels shall be closed by resetting the outgoing stream. If an incoming stream is reset by the peer, a corresponding outgoing stream reset SHOULD be issued. If both streams of a channel are reset, the channel is closed and the streams are available for reuse for new channel opens.

## **6.3. Sending and Receiving Data**

Data shall be sent using PPID's other than the Data Channel Control PPID. These PPID's should be registered with IANA via (TBD). The meaning of these data PPIDs and the format of the data shall be specific to the usage of this protocol, and typically shall be provided to the higher layers to allow proper decoding of the data.

For WebRTC, data PPID's for DOMStrings and binary data blobs shall be created.

All data sent on a Data Channel in both directions MUST be sent over the underlying Stream using the reliability defined when the Data Channel was opened.

Data may be sent before the 3-way handshake is complete; if so it must be sent with in-order delivery set in order to avoid race conditions caused by a handshake message being lost. This is an exception to the requirement to send all data using the channel reliability settings.

It is recommended that message size be kept within certain bounds (TBD).

## **7. Security Considerations**

To be done.



## 8. IANA Considerations

This document also defines three new SCTP Payload Protocol Identifiers (PPIDs). [RFC 4960](#) [[RFC4960](#)] creates the registry from which these identifiers have been assigned. The following values have been reserved:

WebRTC Control - #To Be Assigned

DOMString - #To Be Assigned

Binary Data - #To Be Assigned

## 9. Acknowledgments

The authors wish to thank ... for there invaluable comments.

## 10. References

### 10.1. Normative References

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## **10.2. Informational References**

[I-D.ietf-rtcweb-overview]

Alvestrand, H., "Overview: Real Time Protocols for Browser-based Applications", [draft-ietf-rtcweb-overview-03](#) (work in progress), March 2012.

[I-D.ietf-rtcweb-data-channel]

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