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Transports for RTCWEB
draft-ietf-rtcweb-transports-04

Abstract

This document describes the data transport protocols used by RTCWEB, including the protocols used for interaction with intermediate boxes such as firewalls, relays and NAT boxes.

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1. Introduction

The IETF RTCWEB effort, part of the WebRTC effort carried out in cooperation between the IETF and the W3C, is aimed at specifying a protocol suite that is useful for real time multimedia exchange between browsers.

The overall effort is described in the RTCWEB overview document, [[I-D.ietf-rtcweb-overview](#)]. This document focuses on the data transport protocols that are used by conforming implementations.

This protocol suite is designed for WebRTC, and intends to satisfy the security considerations described in the WebRTC security documents, [[I-D.ietf-rtcweb-security](#)] and [[I-D.ietf-rtcweb-security-arch](#)].

2. Requirements language

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 [RFC 2119](#) [[RFC2119](#)].

3. Transport and Middlebox specification

3.1. System-provided interfaces

The protocol specifications used here assume that the following protocols are available to the implementations of the RTCWEB protocols:

- o UDP. This is the protocol assumed by most protocol elements described.
- o TCP. This is used for HTTP/WebSockets, as well as for TURN/SSL and ICE-TCP.

For both protocols, IPv4 and IPv6 support is assumed.

For UDP, this specification assumes the ability to set the DSCP code point of the sockets opened on a per-packet basis, in order to achieve the prioritizations described in [[I-D.ietf-tsvwg-rtcweb-qos](#)] (see [Section 4.1](#)) when multiple media types are multiplexed. It does not assume that the DSCP codepoints will be honored, and does assume that they may be zeroed or changed, since this is a local configuration issue.

Platforms that do not give access to these interfaces will not be able to support a conforming RTCWEB implementation.

This specification does not assume that the implementation will have access to ICMP or raw IP.

3.2. Ability to use IPv4 and IPv6

Web applications running on top of the RTCWEB implementation MUST be able to utilize both IPv4 and IPv6 where available - that is, when two peers have only IPv4 connectivity to each other, or they have only IPv6 connectivity to each other, applications running on top of the RTCWEB implementation MUST be able to communicate.

When TURN is used, and the TURN server has IPv4 or IPv6 connectivity to the peer or its TURN server, candidates of the appropriate types MUST be supported. The "Happy Eyeballs" specification for ICE [[I-D.reddy-mmusic-ice-happy-eyeballs](#)] SHOULD be supported.

3.3. Usage of temporary IPv6 addresses

The IPv6 default address selection specification [[RFC6724](#)] specifies that temporary addresses [[RFC4941](#)] are to be preferred over permanent addresses. This is a change from the rules specified by [[RFC3484](#)]. For applications that select a single address, this is usually done by the IPV6_PREFER_SRC_TMP preference flag specified in [[RFC5014](#)]. However, this rule is not completely obvious in the ICE scope. This is therefore clarified as follows:

When a client gathers all IPv6 addresses on a host, and both temporary addresses and permanent addresses of the same scope are present, the client SHOULD discard the permanent addresses before forming pairs. This is consistent with the default policy described in [[RFC6724](#)].

3.4. Middle box related functions

The primary mechanism to deal with middle boxes is ICE, which is an appropriate way to deal with NAT boxes and firewalls that accept traffic from the inside, but only from the outside if it's in response to inside traffic (simple stateful firewalls).

ICE [[RFC5245](#)] MUST be supported. The implementation MUST be a full ICE implementation, not ICE-Lite; this allows interworking with both ICE and ICE-Lite implementations when they are deployed appropriately.

In order to deal with situations where both parties are behind NATs

which perform endpoint-dependent mapping (as defined in [\[RFC5128\] section 2.4](#)), TURN [\[RFC5766\]](#) MUST be supported.

Configuration of STUN and TURN servers, both from browser configuration and from an applicaiton, MUST be supported.

In order to deal with firewalls that block all UDP traffic, TURN using TCP between the client and the server MUST be supported, and TURN using TLS over TCP between the client and the server MUST be supported. See [\[RFC5766\] section 2.1](#) for details.

In order to deal with situations where one party is on an IPv4 network and the other party is on an IPv6 network, TURN extensions for IPv6 [\[RFC6156\]](#) MUST be supported.

TURN TCP candidates [\[RFC6062\]](#) MAY be supported.

However, such candidates are not seen as providing any significant benefit. First, use of TURN TCP would only be relevant in cases which both peers are required to use TCP to establish a PeerConnection. Secondly, that use case is anyway supported by both sides establishing UDP relay candidates using TURN over TCP to connect to the relay server. Thirdly, using TCP only between the endpoint and its relay may result in less issues with TCP in regards to real-time constraints, e.g. due to head of line blocking.

ICE-TCP candidates [\[RFC6544\]](#) MUST be supported; this may allow applications to communicate to peers with public IP addresses across UDP-blocking firewalls without using a TURN server.

If TCP connections are used, RTP framing according to [\[RFC4571\]](#) MUST be used, both for the RTP packets and for the DTLS packets used to carry data channels.

The ALTERNATE-SERVER mechanism specified in [\[RFC5389\]](#) (STUN) [section 11](#) (300 Try Alternate) MUST be supported.

Further discussion of the interaction of RTCWEB with firewalls is contained in [\[I-D.hutton-rtcweb-nat-firewall-considerations\]](#). This document makes no requirements on interacting with HTTP proxies or HTTP proxy configuration methods.

NOTE IN DRAFT: This may be added.

[3.5. Transport protocols implemented](#)

For transport of media, secure RTP is used. The details of the profile of RTP used are described in "RTP Usage"

[[I-D.ietf-rtcweb-rtp-usage](#)].

For data transport over the RTCWEB data channel [[I-D.ietf-rtcweb-data-channel](#)], RTCWEB implementations MUST support SCTP over DTLS over ICE. This encapsulation is specified in [[I-D.ietf-tsvwg-sctp-dtls-encaps](#)]. Negotiation of this transport in SDP is defined in [[I-D.ietf-mmusic-sctp-sdp](#)]. The SCTP extension for NDATA, [[I-D.ietf-tsvwg-sctp-ndata](#)], MUST be supported.

The setup protocol for RTCWEB data channels is described in [[I-D.jesup-rtcweb-data-protocol](#)].

RTCWEB implementations MUST support multiplexing of DTLS and RTP over the same port pair, as described in the DTLS_SRTTP specification [[RFC5764](#)], [section 5.1.2](#). All application layer protocol payloads over this DTLS connection are SCTP packets.

4. Media Prioritization

The RTCWEB prioritization model is that the application tells the RTCWEB implementation about the priority of media and data flows through an API.

The priority associated with a media or data flow is classified as "normal", "below normal", "high" or "very high". There are only four priority levels at the API.

The priority settings affect two pieces of behavior: Packet markings and packet send sequence decisions. Each is described in its own section below.

4.1. Usage of Quality of Service - DSCP and Multiplexing

WebRTC implementations SHOULD attempt to set QoS on the packets sent, according to the guidelines in [[I-D.ietf-tsvwg-rtcweb-qos](#)]. It is appropriate to depart from this recommendation when running on platforms where QoS marking is not implemented.

The implementation MAY turn off use of DSCP markings if it detects symptoms of unexpected behaviour like priority inversion or blocking of packets with certain DSCP markings. The detection of these conditions is implementation dependent. (Question: Does there need to be an API knob to turn off DSCP markings?)

There exist a number of schemes for achieving quality of service that do not depend solely on DSCP code points. Some of these schemes depend on classifying the traffic into flows based on 5-tuple (source

address, source port, protocol, destination address, destination port) or 6-tuple (same as above + DSCP code point). Under differing conditions, it may therefore make sense for a sending application to choose any of the configurations:

- o Each media stream carried on its own 5-tuple
- o Media streams grouped by media type into 5-tuples (such as carrying all audio on one 5-tuple)
- o All media sent over a single 5-tuple, with or without differentiation into 6-tuples based on DSCP code points

In each of the configurations mentioned, data channels may be carried in its own 5-tuple, or multiplexed together with one of the media flows.

More complex configurations, such as sending a high priority video stream on one 5-tuple and sending all other video streams multiplexed together over another 5-tuple, can also be envisioned. More information on mapping media flows to 5-tuples can be found in [\[I-D.ietf-rtcweb-rtp-usage\]](#).

A sending implementation **MUST** be able to multiplex all media and data on a single 5-tuple (fully bundled), **MUST** be able to send each media stream on its own 5-tuple and data on its own 5-tuple (fully unbundled), and **MAY** choose to support other configurations.

Sending data over multiple 5-tuples is not supported.

NOTE IN DRAFT: is there a need to place the "group by media type, with data multiplexed on the video" as a **MUST** or **SHOULD** configuration? Are there other **MUST** configurations?

NOTE IN DRAFT: It's been suggested that at least one "MUST" configuration should be with data channels on its own 5-tuple, separate from the media. Opinions sought.

A receiving implementation **MUST** be able to receive media and data in all these configurations.

[4.2.](#) Local prioritization

When an RTCWEB implementation has packets to send on multiple streams that are congestion-controlled under the same congestion controller, the RTCWEB implementation **SHOULD** serve the streams in a weighted round-robin fashion, with each stream at each level of priority being given approximately twice the transmission capacity (measured in

payload bytes) of the level below.

Thus, when congestion occurs, a "very high" priority flow will have the ability to send 8 times as much data as a "below normal" flow if both have data to send. This prioritization is independent of the media type, but will lead to packet loss due to full send buffers occurring first on the highest volume flows at any given priority level. The details of which packet to send first are implementation defined.

For example: If there is a very high priority audio flow sending 100 byte packets, and a normal priority video flow sending 1000 byte packets, and outgoing capacity exists for sending >5000 payload bytes, it would be appropriate to send 4000 bytes (40 packets) of audio and 1000 bytes (one packet) of video as the result of a single pass of sending decisions.

Conversely, if the audio flow is marked normal priority and the video flow is marked very high priority, the scheduler may decide to send 2 video packets (2000 bytes) and 5 audio packets (500 bytes) when outgoing capacity exists for sending > 2500 payload bytes.

If there are two very high priority audio flows, each will be able to send 4000 bytes in the same period where a normal priority video flow is able to send 1000 bytes.

NOTE: The appropriate algorithm for deciding when to send SCTP data vs media data is not described yet.

5. IANA Considerations

This document makes no request of IANA.

Note to RFC Editor: this section may be removed on publication as an RFC.

6. Security Considerations

Security considerations are enumerated in [[I-D.ietf-rtcweb-security](#)].

7. Acknowledgements

This document is based on earlier versions embedded in [[I-D.ietf-rtcweb-overview](#)], which were the results of contributions from many RTCWEB WG members.

Special thanks for reviews of earlier versions of this draft go to Magnus Westerlund, Markus Isomaki and Dan Wing; the contributions from Andrew Hutton also deserve special mention.

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[Appendix A.](#) Change log

[A.1.](#) Changes from -00 to -01

- o Clarified DSCP requirements, with reference to -qos-
- o Clarified "symmetric NAT" -> "NATs which perform endpoint-dependent mapping"

- o Made support of TURN over TCP mandatory
- o Made support of TURN over TLS a MAY, and added open question
- o Added an informative reference to -firewalls-
- o Called out that we don't make requirements on HTTP proxy interaction (yet

A.2. Changes from -01 to -02

- o Required support for 300 Alternate Server from STUN.
- o Separated the ICE-TCP candidate requirement from the TURN-TCP requirement.
- o Added new sections on using QoS functions, and on multiplexing considerations.
- o Removed all mention of RTP profiles. Those are the business of the RTP usage draft, not this one.
- o Required support for TURN IPv6 extensions.
- o Removed reference to the TURN URI scheme, as it was unnecessary.
- o Made an explicit statement that multiplexing (or not) is an application matter.

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A.3. Changes from -02 to -03

- o Added required support for [draft-ietf-tsvwg-sctp-ndata](#)
- o Removed discussion of multiplexing, since this is present in rtp-usage.
- o Added [RFC 4571](#) reference for framing RTP packets over TCP.
- o Downgraded TURN TCP candidates from SHOULD to MAY, and added more language discussing TCP usage.
- o Added language on IPv6 temporary addresses.
- o Added language describing multiplexing choices.

- o Added a separate section detailing what it means when we say that an RTCWEB implementation MUST support both IPv4 and IPv6.

[A.4.](#) Changes from -03 to -04

- o Added a section on prioritization, moved the DSCP section into it, and added a section on local prioritization, giving a specific algorithm for interpreting "priority" in local prioritization.
- o ICE-TCP candidates was changed from MAY to MUST, in recognition of the sense of the room at the London IETF.

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