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Internet-Draft Google
Intended status: Standards Track August 18, 2013
Expires: February 19, 2014

Transports for RTCWEB
draft-ietf-rtcweb-transports-00

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.

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].

Status of this Memo

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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 protocos 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. Transport and Middlebox specification

2.1. 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, this specification assumes the ability to set the DSCP code point of the sockets opened. 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.

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

2.2. 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).

In order to deal with symmetric NATs, TURN MUST be supported.

In order to deal with firewalls that block all UDP traffic, TURN over TCP MUST be supported. (QUESTION: What about ICE-TCP?)

The following specifications MUST be supported:

- ICE [RFC5245]
- TURN, including TURN over TCP [[QUESTION: and TURN over TLS]], [RFC5766].

For referring to STUN and TURN servers, this specification depends on the STUN URI, [I-D.nandakumar-rtcweb-stun-uri].

2.3. Transport protocols implemented

For data transport over the RTCWEB data channel [I-D.ietf-rtcweb-data-channel], RTCWEB implementations support SCTP over DTLS over ICE. This is specified in [I-D.ietf-tsvwg-sctp-dtls-encaps]. Negotiation of this transport in SCTP is defined in [I-D.ietf-mmusic-sctp-sdp].

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

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].

RTCWEB implementations MUST support multiplexing of SCTP/DTLS and RTP over the same port pair, as described in the DTLS_SRTP specification [RFC5764], section 5.1.2.

3. IANA Considerations

This document makes no request of IANA.

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

4. Security Considerations

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

5. 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.

6. References

6.1. Normative References

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[RFC5766] Mahy, R., Matthews, P., and J. Rosenberg, "Traversal Using

Relays around NAT (TURN): Relay Extensions to Session Traversal Utilities for NAT (STUN)", RFC 5766, April 2010.

6.2. Informative References

[I-D.ietf-rtcweb-overview]
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[I-D.jesup-rtcweb-data-protocol]
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