Network Working Group Internet-Draft Intended status: Informational Expires: April 10, 2015 C. Jennings Cisco October 07, 2014

WebRTC Dependencies draft-jennings-rtcweb-deps-01

Abstract

This draft will never be published as an RFC and is meant purely to help track the IETF dependencies from the W3C WebRTC documents.

Status of This Memo

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1. Dependencies

The key IETF specifications that the W3C GetUserMedia specification normatively depends on is: [I-D.ietf-rtcweb-constraints-registry], [RFC2119].

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The key IETF specifications that the W3C WebRTC specification normatively depended on are: [RFC5245], [RFC2119], [RFC3388], [RFC7064], [RFC7065], [I-D.ietf-rtcweb-audio], [I-D.ietf-rtcweb-data-channel], [I-D.ietf-rtcweb-data-protocol], [I-D.ietf-rtcweb-jsep], [I-D.ietf-rtcweb-rtp-usage], [I-D.ietf-rtcweb-security-arch], [I-D.ietf-rtcweb-transports], [I-D.ietf-rtcweb-video], [RFC3264] and informatively depends on [I-D.ietf-rtcweb-overview], [I-D.ietf-rtcweb-security].
These IETF drafts in turn normatively depend on the following drafts:
[I-D.ietf-payload-rtp-opus].
[I-D.ietf-tsvwg-sctp-ndata], [I-D.ietf-rtcweb-data-protocol],
[I-D.ietf-tsvwg-sctp-dtls-encaps], [I-D.ietf-rtcweb-security],
[I-D.ietf-tsvwg-sctp-prpolicies], [I-D.ietf-mmusic-sctp-sdp],
[I-D.ietf-mmusic-msid], [I-D.ietf-mmusic-sctp-sdp],
[I-D.ietf-mmusic-sdp-búndle-negotiation],
[I-D.ietf-mmusic-sdp-mux-attributes],
[I-D.ietf-avtcore-multi-media-rtp-session],
[I-D.ietf-avtcore-rtp-circuit-breakers],
[I-D.ietf-avtcore-rtp-multi-stream_optimisation],
[I-D.ietf-avtcore-rtp-multi-stream],
[I-D.ietf-avtcore-6222bis] (now [RFC7022]),
[I-D.ietf-rtcweb-stun-consent-freshness],
[I-D.hutton-httpbis-connect-protocol], [I-D.patil-tram-alpn], [I-D.ietf-tls-applayerprotoneg] (now [RFC7301]), [I-D.ietf-httpbis-header-compression],
[I-D.petithuguenin-tram-turn-dtls], [I-D.ietf-tsvwg-rtcweb-qos], [I-D.reddy-mmusic-ice-happy-eyeballs], [I-D.ietf-rtcweb-alpn],
[I-D.ietf-payload-vp8].
Right now security normatively depends on [I-D.ietf-rtcweb-overview],
Right now video normatively depends on [I-D.grange-vp9-bitstream],
[I-D.ietf-payload-rtp-h265],
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The drafts webrtc currently normatively depends on that are not WG drafts are:

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[I-D.grange-vp9-bitstream], [I-D.hutton-httpbis-connect-protocol],
[I-D.patil-tram-alpn], [I-D.petithuguenin-tram-turn-dtls],
[I-D.reddy-mmusic-ice-happy-eyeballs],

A few key drafts that the informatively depends on:
[I-D.ietf-mmusic-trickle-ice], [I-D.nandakumar-rtcweb-sdp],
I-D.ietf-avtcore-multiplex-guidelines],
I-D.ietf-avtcore-rtp-topologies-update],
I-D.ietf-avtext-rtp-grouping-taxonomy],
I-D.ietf-rmcat-cc-requirements],
I-D.ietf-rtcweb-use-cases-and-requirements],
I-D.kaufman-rtcweb-security-ui], [I-D.alvestrand-rtcweb-gateways],
I-D.hutton-rtcweb-nat-firewall-considerations],
I-D.ietf-dart-dscp-rtp], [I-D.roach-mmusic-unified-plan],
I-D.westerlund-avtcore-multiplex-architecture],
I-D.lennox-payload-ulp-ssrc-mux],
I-D.lennox-payload-ulp-ssrc-mux],
I-D.ietf-avtcore-multiplex-guidelines], [I-D.ietf-avtcore-srtp-ekt],
I-D.ietf-rtcweb-use-cases-and-requirements],
Something audio should ref but does not yet:
[I-D.ietf-rtcweb-audio-codecs-for-interop]
```

2. References

2.1. Normative References

- [I-D.ietf-avtcore-6222bis]
 Begen, A., Perkins, C., Wing, D., and E. Rescorla,
 "Guidelines for Choosing RTP Control Protocol (RTCP)
 Canonical Names (CNAMEs)", draft-ietf-avtcore-6222bis-06
 (work in progress), July 2013.

- [I-D.ietf-avtcore-multi-media-rtp-session]
 Westerlund, M., Perkins, C., and J. Lennox, "Sending
 Multiple Types of Media in a Single RTP Session", draft ietf-avtcore-multi-media-rtp-session-05 (work in
 progress), February 2014.
- [I-D.ietf-avtcore-rtp-circuit-breakers]
 Perkins, C. and V. Singh, "Multimedia Congestion Control:
 Circuit Breakers for Unicast RTP Sessions", draft-ietf-avtcore-rtp-circuit-breakers-06 (work in progress), July 2014.
- [I-D.ietf-avtcore-rtp-multi-stream-optimisation]
 Lennox, J., Westerlund, M., Wu, W., and C. Perkins,
 "Sending Multiple Media Streams in a Single RTP Session:
 Grouping RTCP Reception Statistics and Other Feedback",
 draft-ietf-avtcore-rtp-multi-stream-optimisation-04 (work
 in progress), August 2014.
- [I-D.ietf-httpbis-header-compression]
 Peon, R. and H. Ruellan, "HPACK Header Compression for HTTP/2", draft-ietf-httpbis-header-compression-09 (work in progress), July 2014.

- [I-D.ietf-mmusic-sctp-sdp]
 Loreto, S. and G. Camarillo, "Stream Control Transmission
 Protocol (SCTP)-Based Media Transport in the Session
 Description Protocol (SDP)", draft-ietf-mmusic-sctp-sdp-07
 (work in progress), July 2014.

- [I-D.ietf-mmusic-sdp-mux-attributes]
 Nandakumar, S., "A Framework for SDP Attributes when
 Multiplexing", draft-ietf-mmusic-sdp-mux-attributes-02
 (work in progress), July 2014.
- [I-D.ietf-payload-rtp-h265]
 Wang, Y., Sanchez, Y., Schierl, T., Wenger, S., and M.
 Hannuksela, "RTP Payload Format for High Efficiency Video Coding", draft-ietf-payload-rtp-h265-02 (work in progress), February 2014.
- [I-D.ietf-payload-vp8]
 Westin, P., Lundin, H., Glover, M., Uberti, J., and F.
 Galligan, "RTP Payload Format for VP8 Video", draft-ietf-payload-vp8-11 (work in progress), February 2014.

- [I-D.ietf-rtcweb-constraints-registry]

 Burnett, D., "IANA Registry for RTCWeb Constrainable
 Properties", draft-ietf-rtcweb-constraints-registry-00
 (work in progress), July 2014.
- [I-D.ietf-rtcweb-data-channel]
 Jesup, R., Loreto, S., and M. Tuexen, "WebRTC Data
 Channels", draft-ietf-rtcweb-data-channel-08 (work in
 progress), April 2014.

- [I-D.ietf-rtcweb-data-protocol]
 Jesup, R., Loreto, S., and M. Tuexen, "WebRTC Data Channel Establishment Protocol", draft-ietf-rtcweb-data-protocol-04 (work in progress), April 2014.

- [I-D.ietf-rtcweb-rtp-usage]
 Perkins, C., Westerlund, M., and J. Ott, "Web Real-Time Communication (WebRTC): Media Transport and Use of RTP", draft-ietf-rtcweb-rtp-usage-06 (work in progress), February 2013.
- [I-D.ietf-rtcweb-security-arch]
 Rescorla, E., "WebRTC Security Architecture", draft-ietf rtcweb-security-arch-09 (work in progress), February 2014.
- [I-D.ietf-rtcweb-stun-consent-freshness]
 Perumal, M., Wing, D., R, R., Reddy, T., and M. Thomson,
 "STUN Usage for Consent Freshness", draft-ietf-rtcwebstun-consent-freshness-07 (work in progress), September 2014.
- [I-D.ietf-rtcweb-video]
 Roach, A., "WebRTC Video Processing and Codec
 Requirements", draft-ietf-rtcweb-video-00 (work in
 progress), July 2014.

- [I-D.ietf-tsvwg-sctp-prpolicies]
 Tuexen, M., Seggelmann, R., Stewart, R., and S. Loreto,
 "Additional Policies for the Partial Reliability Extension
 of the Stream Control Transmission Protocol", draft-ietftsvwg-sctp-prpolicies-04 (work in progress), September
 2014.
- [I-D.petithuguenin-tram-turn-dtls]
 Petit-Huguenin, M. and G. Salgueiro, "Datagram Transport
 Layer Security (DTLS) as Transport for Traversal Using
 Relays around NAT (TURN)", draft-petithuguenin-tram-turndtls-00 (work in progress), January 2014.
- [I-D.reddy-mmusic-ice-happy-eyeballs]
 Reddy, T., Patil, P., and P. Martinsen, "Happy Eyeballs
 Extension for ICE", draft-reddy-mmusic-ice-happyeyeballs-07 (work in progress), June 2014.

- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.
- [RFC3264] Rosenberg, J. and H. Schulzrinne, "An Offer/Answer Model with Session Description Protocol (SDP)", RFC 3264, June 2002.
- [RFC3388] Camarillo, G., Eriksson, G., Holler, J., and H.
 Schulzrinne, "Grouping of Media Lines in the Session
 Description Protocol (SDP)", RFC 3388, December 2002.
- [RFC5245] Rosenberg, J., "Interactive Connectivity Establishment (ICE): A Protocol for Network Address Translator (NAT) Traversal for Offer/Answer Protocols", RFC 5245, April 2010.
- [RFC7022] Begen, A., Perkins, C., Wing, D., and E. Rescorla,
 "Guidelines for Choosing RTP Control Protocol (RTCP)
 Canonical Names (CNAMEs)", RFC 7022, September 2013.
- [RFC7064] Nandakumar, S., Salgueiro, G., Jones, P., and M. Petit-Huguenin, "URI Scheme for the Session Traversal Utilities for NAT (STUN) Protocol", RFC 7064, November 2013.
- [RFC7065] Petit-Huguenin, M., Nandakumar, S., Salgueiro, G., and P.
 Jones, "Traversal Using Relays around NAT (TURN) Uniform
 Resource Identifiers", RFC 7065, November 2013.
- [RFC7301] Friedl, S., Popov, A., Langley, A., and E. Stephan,
 "Transport Layer Security (TLS) Application-Layer Protocol
 Negotiation Extension", RFC 7301, July 2014.

2.2. Informative References

- [I-D.alvestrand-rtcweb-gateways]
 Alvestrand, H., "WebRTC Gateways", draft-alvestrand-rtcweb-gateways-00 (work in progress), August 2014.

- [I-D.ietf-avtcore-multiplex-guidelines]
 Westerlund, M., Perkins, C., and H. Alvestrand,
 "Guidelines for using the Multiplexing Features of RTP to
 Support Multiple Media Streams", draft-ietf-avtcore multiplex-guidelines-02 (work in progress), January 2014.
- [I-D.ietf-avtcore-rtp-topologies-update]
 Westerlund, M. and S. Wenger, "RTP Topologies", draft ietf-avtcore-rtp-topologies-update-04 (work in progress),
 August 2014.
- [I-D.ietf-avtext-rtp-grouping-taxonomy]
 Lennox, J., Gross, K., Nandakumar, S., and G. Salgueiro,
 "A Taxonomy of Grouping Semantics and Mechanisms for Real Time Transport Protocol (RTP) Sources", draft-ietf-avtext rtp-grouping-taxonomy-02 (work in progress), June 2014.

- [I-D.ietf-rmcat-cc-requirements]

 Jesup, R., "Congestion Control Requirements For RMCAT",

 draft-ietf-rmcat-cc-requirements-05 (work in progress),

 July 2014.

- [I-D.ietf-rtcweb-use-cases-and-requirements]

 Holmberg, C., Hakansson, S., and G. Eriksson, "Web Real-Time Communication Use-cases and Requirements", draftietf-rtcweb-use-cases-and-requirements-14 (work in progress), February 2014.
- [I-D.lennox-payload-ulp-ssrc-mux]
 Lennox, J., "Supporting Source-Multiplexing of the Real-Time Transport Protocol (RTP) Payload for Generic Forward Error Correction", draft-lennox-payload-ulp-ssrc-mux-00 (work in progress), February 2013.
- [I-D.roach-mmusic-unified-plan]
 Roach, A., Uberti, J., and M. Thomson, "A Unified Plan for Using SDP with Large Numbers of Media Flows", draft-roach-mmusic-unified-plan-00 (work in progress), July 2013.
- [I-D.westerlund-avtcore-multiplex-architecture]
 Westerlund, M., Perkins, C., and H. Alvestrand,
 "Guidelines for using the Multiplexing Features of RTP",
 draft-westerlund-avtcore-multiplex-architecture-03 (work
 in progress), February 2013.

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