WebRTC Dependencies
draft-jennings-rtcweb-deps-03

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
This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

Internet-Drafts are working documents of the Internet Engineering Task Force (IETF). Note that other groups may also distribute working documents as Internet-Drafts. The list of current Internet-Drafts is at http://datatracker.ietf.org/drafts/current/.

Internet-Drafts are draft documents valid for a maximum of six months and may be updated, replaced, or obsoleted by other documents at any time. It is inappropriate to use Internet-Drafts as reference material or to cite them other than as "work in progress."

This Internet-Draft will expire on April 11, 2015.

Copyright Notice
Copyright (c) 2014 IETF Trust and the persons identified as the document authors. All rights reserved.

This document is subject to BCP 78 and the IETF Trust's Legal Provisions Relating to IETF Documents (http://trustee.ietf.org/license-info) in effect on the date of publication of this document. Please review these documents carefully, as they describe your rights and restrictions with respect to this document. Code Components extracted from this document must include Simplified BSD License text as described in Section 4.e of the Trust Legal Provisions and are provided without warranty as described in the Simplified BSD License.
1. Dependencies

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

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-transport], [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-nodata], [I-D.ietf-rtcweb-data-protocol], [I-D.ietf-tsvwg-sctp-dtls-encaps], [I-D.ietf-rtcweb-video], [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-bundle-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.ietf-tram-alpn], [I-D.ietf-tls-applayerprotoneg] (now [RFC7301]), [I-D.ietf-httpbis-http2], [I-D.ietf-httpbis-header-compression], [I-D.petithuguenin-tram-turn-dtls], [I-D.ietf-tsvwg-rtcweb-qos], [I-D.reddy-mmusic-ice-happy-eyes], [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].

The drafts webrtc currently normatively depends on that are not WG drafts are: [I-D.grange-vp9-bitstream], [I-D.hutton-httpbis-connect-protocol], [I-D.reddy-mmusic-ice-happy-eyes].

A few key drafts that the work 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],
Something audio should ref but does not yet:  
[I-D.ietf-rtcweb-audio-codecs-for-interop]

1.1. Time Estimates

The following table has some very rough estimates of when the draft will become an RFC. Historically these dates have often taken much longer than the estimates so take this with a large dose of salt.

<table>
<thead>
<tr>
<th>Draft Name</th>
<th>ETA</th>
</tr>
</thead>
<tbody>
<tr>
<td>[I-D.hutton-httpbis-connect-protocol]</td>
<td></td>
</tr>
<tr>
<td>[I-D.reddy-mmusic-ice-happy-eyeballs]</td>
<td></td>
</tr>
<tr>
<td>[I-D.ietf-avtcore-multi-media-rtp-session]</td>
<td></td>
</tr>
<tr>
<td>[I-D.ietf-avtcore-rtp-circuit-breakers]</td>
<td></td>
</tr>
<tr>
<td>[I-D.ietf-avtcore-rtp-multi-stream-optimisation]</td>
<td></td>
</tr>
<tr>
<td>[I-D.ietf-avtcore-rtp-multi-stream]</td>
<td></td>
</tr>
<tr>
<td>[I-D.ietf-httpbis-header-compression]</td>
<td></td>
</tr>
<tr>
<td>[I-D.ietf-httpbis-http2]</td>
<td></td>
</tr>
<tr>
<td>[I-D.ietf-mmusic-msid]</td>
<td></td>
</tr>
<tr>
<td>[I-D.ietf-mmusic-sctp-sdp]</td>
<td></td>
</tr>
<tr>
<td>[I-D.ietf-mmusic-sdp-bundle-negotiation]</td>
<td></td>
</tr>
<tr>
<td>[I-D.ietf-mmusic-sdp-mux-attributes]</td>
<td></td>
</tr>
<tr>
<td>[I-D.ietf-payload-rtp-h265]</td>
<td></td>
</tr>
</tbody>
</table>
[I-D.ietf-payload-rtp-opus]
[I-D.ietf-payload-vp8]
[I-D.ietf-rtcweb-alpn]
[I-D.ietf-rtcweb-audio]
[I-D.ietf-rtcweb-constraints-registry]
[I-D.ietf-rtcweb-data-channel]
[I-D.ietf-rtcweb-data-protocol]
[I-D.ietf-rtcweb-data-protocol]
[I-D.ietf-rtcweb-jsep] 2015 Oct
[I-D.ietf-rtcweb-overview]
[I-D.ietf-rtcweb-overview]
[I-D.ietf-rtcweb-rtp-usage]
[I-D.ietf-rtcweb-security-arch]
[I-D.ietf-rtcweb-security]
[I-D.ietf-rtcweb-security]
[I-D.ietf-rtcweb-stun-consent-freshness]
[I-D.ietf-rtcweb-transports]
[I-D.ietf-rtcweb-video]
[I-D.ietf-tsvwg-rtcweb-qos]
[I-D.ietf-tsvwg-sctp-dtls-encaps]
[I-D.ietf-tsvwg-sctp-ndata]
[I-D.ietf-tsvwg-sctp-prpolicies]
[I-D.grange-vp9-bitstream]
[I-D.ietf-tram-alpn] 2014 Nov
2. References

2.1. Normative References

[I-D.grange-vp9-bitstream]

[I-D.hutton-httpbis-connect-protocol]

[I-D.iweb-core-6222bis]

[I-D.iweb-core-avp-codecs]
Terriberry, T., "Update to Remove DVI4 from the Recommended Codecs for the RTP Profile for Audio and Video Conferences with Minimal Control (RTP/AVP)", draft-iweb-core-avp-codecs-03 (work in progress), July 2013.
[I-D.ietf-avtcore-multi-media-rtp-session]

[I-D.ietf-avtcore-rtp-circuit-breakers]

[I-D.ietf-avtcore-rtp-multi-stream]

[I-D.ietf-avtcore-rtp-multi-stream-optimisation]

[I-D.ietf-avtcore-srtp-encrypted-header-ext]

[I-D.ietf-avtext-multiple-clock-rates]

[I-D.ietf-httpbis-header-compression]

[I-D.ietf-httpbis-http2]
Internet-Draft             WebRTC Dependencies              October 2014

[I-D.ietf-mmusic-msid]

[I-D.ietf-mmusic-sctp-sdp]

[I-D.ietf-mmusic-sdp-bundle-negotiation]

[I-D.ietf-mmusic-sdp-mux-attributes]

[I-D.ietf-payload-rtp-h265]

[I-D.ietf-payload-rtp-opus]

[I-D.ietf-payload-vp8]

[I-D.ietf-rtcweb-alpn]

[I-D.ietf-rtcweb-audio]
Valin, J. and C. Bran, "WebRTC Audio Codec and Processing Requirements", draft-ietf-rtcweb-audio-05 (work in progress), February 2014.
[I-D.ietf-rtcweb-constraints-registry]

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

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

[I-D.ietf-rtcweb-jsep]

[I-D.ietf-rtcweb-overview]

[I-D.ietf-rtcweb-rtp-usage]

[I-D.ietf-rtcweb-security]

[I-D.ietf-rtcweb-security-arch]

[I-D.ietf-rtcweb-stun-consent-freshness]

[I-D.ietf-rtcweb-transports]
[I-D.ietf-rtcweb-video]

[I-D.ietf-tls-aplayerprotoneg]

[I-D.ietf-tram-alpn]

[I-D.ietf-tram-stun-dtls]

[I-D.ietf-tsvwg-rtcweb-qos]

[I-D.ietf-tsvwg-sctp-dtls-encaps]

[I-D.ietf-tsvwg-sctp-ndata]

[I-D.ietf-tsvwg-sctp-prpolicies]
[I-D.nandakumar-rtcweb-stun-uri]

[I-D.petithuguenin-behave-turn-uris]

[I-D.petithuguenin-tram-turn-dtls]

[I-D.reddy-mmusic-ice-happy-eyeballs]


[RFC7007] Terriberry, T., "Update to Remove DVI4 from the Recommended Codecs for the RTP Profile for Audio and Video Conferences with Minimal Control (RTP/AVP)", RFC 7007, August 2013.
2.2. Informative References


[I-D.ietf-avtcore-srtp-ekt]

[I-D.ietf-avtext-rtp-grouping-taxonomy]

[I-D.ietf-dart-dscp-rtp]
Black, D. and P. Jones, "Differentiated Services (DiffServ) and Real-time Communication", draft-ietf-dart-dscp-rtp-07 (work in progress), September 2014.

[I-D.ietf-mmusic-trickle-ice]

[I-D.ietf-rmcat-cc-requirements]
Jespup, R. "Congestion Control Requirements For RMCAT", draft-ietf-rmcat-cc-requirements-05 (work in progress), July 2014.

[I-D.ietf-rtcweb-audio-codecs-for-interop]
Proust, S., Berger, E., Feiten, B., Bogineni, K., Lei, M., and E. Marocco, "Additional WebRTC audio codecs for interoperability with legacy networks.", draft-ietf-rtcweb-audio-codecs-for-interop-00 (work in progress), September 2014.

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

[I-D.kaufman-rtcweb-security-ui]
[I-D.lennox-payload-ulp-ssrc-mux]

[I-D.nandakumar-rtcweb-sdp]

[I-D.roach-mmusic-unified-plan]

[I-D.westerlund-avtcore-multiplex-architecture]

Author's Address

Cullen Jennings
Cisco

Email: fluffy@iii.ca