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WebRTC Dependencies
draft-jennings-rtcweb-deps-08

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](#).

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1. Dependencies

The key IETF specifications that the W3C GetUserMedia specification normatively depends on is: [[I-D.ietf-rtcweb-security-arch](#)] and [[RFC2119](#)].

The key IETF specifications that the W3C WebRTC specification normatively depended on are: [[I-D.ietf-rtcweb-alpn](#)], [[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](#)], [[I-D.ietf-tram-turn-third-party-authz](#)], [[I-D.ietf-tsvwg-rtcweb-qos](#)], [[RFC2119](#)], [[RFC4566](#)], [[RFC5389](#)], [[RFC5888](#)], [[RFC6236](#)], [[RFC6464](#)], [[RFC6465](#)], [[RFC6544](#)], [[RFC7064](#)], [[RFC7065](#)] [[RFC3264](#)], [[RFC5245](#)], and informatively depends on [[I-D.ietf-rtcweb-overview](#)], [[I-D.ietf-rtcweb-security](#)], [[I-D.shieh-rtcweb-ip-handling](#)], and [[I-D.ietf-mmusic-trickle-ice](#)].

In addition 3GPP work normatively depends on [[I-D.ietf-rtcweb-gateways](#)].

These IETF drafts in turn normatively depend on the following drafts:
(Note this list is out of date)

[[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-ice-dualstack-fairness](#)], [[I-D.ietf-jose-json-web-algorithms](#)], [[I-D.ietf-mmusic-msid](#)], [[I-D.ietf-mmusic-proto-iana-registration](#)], [[I-D.ietf-mmusic-sctp-sdp](#)], [[I-D.ietf-mmusic-sdp-bundle-negotiation](#)], [[I-D.ietf-mmusic-sdp-mux-attributes](#)], [[I-D.ietf-mmusic-trickle-ice](#)], [[I-D.ietf-payload-flexible-fec-scheme](#)], [[I-D.ietf-payload-rtp-opus](#)], [[I-D.ietf-payload-vp8](#)], [[I-D.ietf-rtcweb-alpn](#)], [[I-D.ietf-rtcweb-data-protocol](#)], [[I-D.ietf-rtcweb-fec](#)], [[I-D.ietf-rtcweb-security](#)], [[I-D.ietf-rtcweb-stun-consent-freshness](#)], [[I-D.ietf-tls-applayerprotoneg](#)], [[I-D.ietf-tram-alpn](#)], [[I-D.ietf-tsvwg-rtcweb-qos](#)], [[I-D.ietf-tsvwg-sctp-dtls-encaps](#)], [[I-D.ietf-tsvwg-sctp-ndata](#)], and [[I-D.ietf-tsvwg-sctp-prpolicies](#)].

Right now security normatively depends on [[I-D.ietf-rtcweb-overview](#)].

The drafts webrtc currently normatively depends on that are not WG drafts are: [[I-D.shieh-rtcweb-ip-handling](#)]

A few key drafts that the work informatively depends on:
[[I-D.alvestrand-rtcweb-gateways](#)],

[[I-D.hutton-rtcweb-nat-firewall-considerations](#)],
[I-D.ietf-avtcore-multiplex-guidelines](#)],
[I-D.ietf-avtcore-rtp-topologies-update](#)],
[I-D.ietf-avtcore-srtp-ekt](#)], [[I-D.ietf-avtext-rtp-grouping-taxonomy](#)],
[I-D.ietf-dart-dscp-rtp](#)], [[I-D.ietf-mmusic-trickle-ice](#)],
[I-D.ietf-rmcat-cc-requirements](#)],
[I-D.ietf-rtcweb-use-cases-and-requirements](#)],
[I-D.kaufman-rtcweb-security-ui](#)], [[I-D.lennox-payload-ulp-ssrc-mux](#)],
[I-D.ietf-rtcweb-sdp](#)], [[I-D.roach-mmusic-unified-plan](#)],
[I-D.ietf-rtcweb-audio-codecs-for-interop](#)],
[I-D.westerlund-avtcore-multiplex-architecture](#)].

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.

Last updated Feb 25, 2016.

ETA	Draft Name
[RFC6904]	[I-D.ietf-avtcore-srtp-encrypted-header-ext]
[RFC7007]	[I-D.ietf-avtcore-avp-codecs]
[RFC7022]	[I-D.ietf-avtcore-6222bis]
[RFC7064]	[I-D.nandakumar-rtcweb-stun-uri]
[RFC7065]	[I-D.petithuguenin-behave-turn-uris]
[RFC7160]	[I-D.ietf-avtext-multiple-clock-rates]
[RFC7301]	[I-D.ietf-tls-applayerprotoneg]
[RFC7350]	[I-D.ietf-tram-stun-dtls]
[RFC7443]	[I-D.ietf-tram-alpn]
[RFC7496]	[I-D.ietf-tsvwg-sctp-prpolicies]
[RFC7518]	[I-D.ietf-jose-json-web-algorithms]
[RFC7587]	[I-D.ietf-payload-rtp-opus]
[RFC7635]	[I-D.ietf-tram-turn-third-party-authz]

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	[RFC7639]	[I-D.ietf-httpbis-tunnel-protocol]
	[RFC7675]	[I-D.ietf-rtcweb-stun-consent-freshness]
Auth 48		[I-D.ietf-payload-vp8]
Auth 48		[I-D.ietf-rtcweb-video]
Miss Ref		[I-D.ietf-rtcweb-rtp-usage]
Miss Ref		[I-D.ietf-tsvwg-sctp-dtls-encaps]
RFC Ed		[I-D.ietf-rtcweb-data-channel]
RFC Ed		[I-D.ietf-rtcweb-data-protocol]
RFC Ed		[I-D.ietf-avtcore-multi-media-rtp-session]
RFC Ed		[I-D.ietf-avtcore-rtp-multi-stream-optimisation]
RFC Ed		[I-D.ietf-avtcore-rtp-multi-stream]
IETF LC		[I-D.ietf-rtcweb-audio]
IETF LC		[I-D.ietf-avtcore-rtp-circuit-breakers]
IETF LC		[I-D.ietf-mmusic-proto-iana-registration]
AD Eval		[I-D.ietf-mmusic-sdp-mux-attributes]
PubReq		[I-D.ietf-rtcweb-security-arch]
PubReq		[I-D.ietf-rtcweb-security]
Write Up		[I-D.ietf-mmusic-sdp-bundle-negotiation]
Write Up		[I-D.ietf-rtcweb-alpn]
WGLC		[I-D.ietf-rtcweb-overview]
WGLC		[I-D.ietf-tsvwg-rtcweb-qos]
		[I-D.ietf-tsvwg-sctp-ndata]
		[I-D.ietf-rtcweb-transports]
		[I-D.ietf-mmusic-msid]

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