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WebRTC Dependencies
draft-jennings-rtcweb-deps-00

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 W3C GetUserMedia specification normatively depends on [I-D.burnett-rtcweb-constraints-registry].

The W3C WebRTC specification normatively depended on [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] TODO I-D.ietf-rtcweb-video [RFC3264] and informatively depends on [I-D.ietf-rtcweb-overview] [I-D.ietf-rtcweb-security].

2. References

2.1. Normative References

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

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[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.

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2.2. Informative References

[I-D.ietf-rtcweb-overview]

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[I-D.ietf-rtcweb-security]

Rescorla, E., "Security Considerations for WebRTC", draft-ietf-rtcweb-security-06 (work in progress), January 2014.

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