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1 Scope

The present document provides the details for allowing Web Real-Time Communication (WebRTC) IMS Clients (WIC) to access the IP Multimedia (IM) Core Network (CN) subsystem.

The present document is applicable to WebRTC IMS client (WIC), eP-CSCF, eIMS-AGW, WebRTC Web Server Function (WWSF) and WebRTC Authorization Function (WAF).

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

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- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document.*

[1] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

[2] IETF RFC 7118: "The WebSocket Protocol as a Transport for the Session Initiation Protocol (SIP)".

[3] 3GPP TS 24.229: "IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3".

[4] 3GPP TS 23.228: " IP Multimedia Subsystem (IMS); Stage 2".

[5] IETF RFC 5763: "Framework for Establishing a Secure Real-time Transport Protocol (SRTP) Security Context Using Datagram Transport Layer Security (DTLS)".

[6] IETF RFC 5764: "Datagram Transport Layer Security (DTLS) Extension to Establish Keys for the Secure Real-time Transport Protocol (SRTP)".

[7] 3GPP TS 22.173: "IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1".

[8] 3GPP TS 24.173: "IMS multimedia telephony communication service and supplementary services; Stage 3".

[9] 3GPP TS 33.203: "Access security for IP based services".

[10] RFC 6750 (October 2012): "The OAuth 2.0 Authorization Framework: Bearer Token Usage".

[11] 3GPP TS 23.292: "IP Multimedia Subsystem (IMS) Centralized Services; Stage 2".

[12] RFC 5009 (September 2007): "Private Header (P-Header) Extension to the Session Initiation Protocol (SIP) for Authorization of Early Media".

[13] 3GPP TS 23.334: "IMS Application Level Gateway (IMS-ALG) – IMS Access Gateway (IMS-AGW) interface".

[14] RFC 4145 (September 2005): "TCP-Based Media Transport in the Session Description Protocol (SDP)".

[15] RFC 8122 (March 2017): "Connection-Oriented Media Transport over the Transport Layer Security (TLS) Protocol in the Session Description Protocol (SDP)".

[16] draft-ietf-rtcweb-data-channel-13 (January 2015): "WebRTC Data Channels".