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Contents

Intellectual Property Rights 2
Legal Notice 2
Modal verbs terminology..2
Foreword..6
1 Scope 7
2 References 7
3 Definitions and abbreviations..9
3.1 Definitions..9
3.2 Abbreviations 9
4 Overview of WebRTC access to IMS 10
4.1 General 10
5 Functional entities 10
5.1 General 10
5.2 WIC (WebRTC IMS Client) 10
5.3 WWSF (WebRTC Web Server Function) 11
5.4 WAF (WebRTC Authorisation Function) 11
5.5 eP-CSCF (P-CSCF enhanced for WebRTC) 11
5.6 eIMS-AGW (IMS Access Gateway enhanced for WebRTC) 11
5A Data transport 11
5A.1 General 11
5A.2 UE 11
5A.3 WWSF (WebRTC Web Server Function)..11
5A.4 eP-CSCF (P-CSCF enhanced for WebRTC) 12
5B Data framing and securing 12
5B.1 General 12
5B.2 UE 12
5B.3 WWSF (WebRTC Web Server Function)..12
5B.4 eP-CSCF (P-CSCF enhanced for WebRTC) 12
5C Data formats 13
5C.1 General 13
5C.2 UE 13
5C.3 WWSF (WebRTC Web Server Function)..13
5C.4 eP-CSCF (P-CSCF enhanced for WebRTC) 13
5D Connection management..14
5D.1 General 14
5D.2 UE 14
5D.3 WWSF (WebRTC Web Server Function)..14
5D.4 eP-CSCF (P-CSCF enhanced for WebRTC) 14
5E Presentation and control 14
5E.1 General 14
5E.2 UE 15
5E.3 WWSF (WebRTC Web Server Function)..15
5E.4 eP-CSCF (P-CSCF enhanced for WebRTC) 15
5F Local system support functions..15
5F.1 General 15
5F.2 UE 15
5F.3 WWSF (WebRTC Web Server Function)..15
5F.4 eP-CSCF (P-CSCF enhanced for WebRTC) 15
6 Registration and authentication..15

6.1 General 15
6.2 WIC (WebRTC IMS Client) 16
6.2.1 WIC registration of individual Public User Identity using IMS authentication 16
6.2.1.1 General 16
6.2.1.2 W2 using SIP Digest credentials 16
6.2.1.3 W2 using IMS-AKA 17
6.2.2 WIC registration of individual public user identity based on web authentication 17
6.2.3 WIC registration of individual public user identity from a pool of public user identities 17
6.3 WWSF (WebRTC Web Server Function) and WAF (WebRTC Authorisation Function) 18
6.3.1 WIC registration of individual public user identity using web credentials 18
6.3.2 WIC registration of individual public user identity from a pool of public user identities 18
6.4 eP-CSCF (P-CSCF enhanced for WebRTC) 18
6.4.1 WIC registration of individual Public User Identity using IMS authentication 18
6.4.1.1 Determination of IMS authentication mechanism 18
6.4.1.2 W2 using SIP Digest credentials 18
6.4.1.3 W2 using IMS-AKA 19
6.4.2 WIC registration of individual public user identity using web credentials 19
6.4.3 WIC registration of individual public user identity from a pool of public user identities 20

6A Deregistration 20
6A.1 General 20
6A.2 WIC (WebRTC IMS Client) 21
6A.3 eP-CSCF (P-CSCF enhanced for WebRTC) 21

7 Call origination and termination 21
7.1 General 21
7.2 WIC (WebRTC IMS Client) 21
7.2.1 General 21
7.2.2 WIC originating call 22
7.2.3 WIC terminating call 22
7.2.4 WIC emergency call 22
7.3 WWSF (WebRTC Web Server Function) 23
7.4 eP-CSCF (P-CSCF enhanced for WebRTC) 23
7.4.1 General 23
7.4.2 WIC originating call 23
7.4.3 WIC terminating call 24
7.4.4 WIC emergency call 25
7.4.5 Media optimization procedure 25
7.4.5.1 WIC originating call 25
7.4.5.2 WIC terminating call 27

8 Data channel open and close 30
8.1 General 30
8.2 WIC (WebRTC IMS Client) 30
8.2.1 General 30
8.2.2 WIC originating call 30
8.2.3 WIC terminating call 31
8.3 WWSF (WebRTC Web Server Function) 31
8.4 eP-CSCF (P-CSCF enhanced for WebRTC) 31
8.4.1 General 31
8.4.2 WIC originating call 31
8.4.3 WIC terminating call 32

9 Call modification 33

10 IP multimedia application support in the IM CN subsystem using webRTC 33
10.1 General 33
10.2 Access to MMTel and supplementary services using webRTC 33
10.2.1 General 33
10.2.2 WIC (WebRTC IMS Client) 33
10.2.2.1 SIP based protocol used by the WIC 33
10.2.2.2 non-SIP based protocol used by the WIC 33
10.2.3 WWSF (WebRTC Web Server Function) 33
10.2.4 eP-CSCF (P-CSCF enhanced for WebRTC) 33

Annex A (informative): Example signalling flows 34

A.1 Scope of signalling flows 34

A.2 Void 34

A.3 Signalling flows for registration 34
A.3.1 Void 34
A.3.2 WIC registration of individual public user identity based on web authentication 34
A.3.3 Void 36

A.4 Void 36

A.5 Void 36

Annex B (informative): Change history 37

History 39

Foreword

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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.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- 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".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[17] draft-ietf-rtcweb-data-protocol-09 (January 2015): "WebRTC Data Channel Establishment Protocol".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[18] draft-ietf-mmusic-sctp-sdp-25 (March 2017): "Stream Control Transmission Protocol (SCTP)-Based Media Transport in the Session Description Protocol (SDP)".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[19] RFC 3261 (June 2002): "SIP: Session Initiation Protocol".

[20] RFC 3264 (June 2002): "An Offer/Answer Model with the Session Description Protocol (SDP)".

[21] RFC 7675 (October 2015): "STUN Usage for Consent Freshness".

[22] RFC 5245 (April 2010): "Interactive Connectivity Establishment (ICE): A Protocol for Network Address Translator (NAT) Traversal for Offer/Answer Protocols".

[23] RFC 8261 (November 2017): "Datagram Transport Layer Security (DTLS) Encapsulation of SCTP Packets".

[24] RFC 6455 (December 2011): "The WebSocket Protocol".

[25] draft-ietf-mmusic-sdp-bundle-negotiation-29 (April 2016): "Negotiating Media Multiplexing Using the Session Description Protocol (SDP)".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[26] RFC 3581 (August 2003): "An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing".

[27] draft-ietf-sipcore-sip-token-authnz-02 (July 2019): "Third-Party Token-based Authentication and Authorization for Session Initiation Protocol (SIP)".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[28] RFC 6544 (March 2012): "TCP Candidates with Interactive Connectivity Establishment (ICE)".

[29] Void.

[30] draft-ietf-rtcweb-overview-18 (March 2017): "Overview: Real Time Protocols for Brower-based Applications".

Editor's note: The above document cannot be formally referenced until it is published as an RFC.

[31] Void.

[32] RFC 3310 (September 2002): "Hypertext Transfer Protocol (HTTP) Digest Authentication Using Authentication and Key Agreement (AKA)".

[33] RFC 4169 (November 2005): "Hypertext Transfer Protocol (HTTP) Digest Authentication Using Authentication and Key Agreement (AKA) Version-2".

[34] 3GPP TS 26.114: "IP multimedia subsystem (IMS); Multimedia telephony, Media handling and interaction".

[35] RFC 7519 (May 2015): "JSON Web Token (JWT)".

[36] draft-ietf-mmusic-data-channel-sdpneg-12 (March 2017): "SDP-based Data Channel Negotiation".

Editor's note [WI: eWebRTCi_CT, CR#0044]: The above document cannot be formally referenced until it is published as an RFC.

[37] draft-ietf-mmusic-msrp-usage-data-channel-06 (October 2016): "MSRP over Data Channels".

Editor's note [WI: eWebRTCi_CT, CR#0043]: The above document cannot be formally referenced until it is published as an RFC.

[38] RFC 5761 (April 2010): "Multiplexing RTP Data and Control Packets on a Single Port".

[39] draft-ietf-mmusic-mux-exclusive-11 (February 2017): "Indicating Exclusive Support of RTP/RTCP Multiplexing using SDP".

Editor's note [WI: TEI14, CR#0050]: The above document cannot be formally referenced until it is published as an RFC.

[40] draft-schwarz-mmusic-t140-usage-data-channel-03 (April 2016): "T.140 Text Conversation over Data Channels".

Editor's note [WI: TEI14, CR#0051]: The above document cannot be formally referenced until it is published as an RFC.

[41] Void.

[42] RFC 8035 (November 2016): "Session Description Protocol (SDP) Offer/Answer Clarifications for RTP/RTCP Multiplexing".

[43] draft-ietf-ice-trickle-09 (April 2017): "Trickle ICE: Incremental Provisioning of Candidates for the Interactive Connectivity Establishment (ICE) Protocol".

[44] RFC 5766 (April 2010): "Traversal Using Relays around NAT (TURN): Relay Extensions to Session Traversal Utilities for NAT (STUN)".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in 3GPP TR 21.905 [1] and the following apply. A term defined in the present document takes precedence over the definition of the same term, if any, in 3GPP TR 21.905 [1].

For the purposes of the present document, the following terms and definitions given in 3GPP TS 23.228 [4] annex U apply:

P-CSCF enhanced for WebRTC (eP-CSCF)
WebRTC Authorization Function (WAF)
WebRTC IMS Client (WIC)
WebRTC Web Server Function (WWSF)

For the purposes of the present document, the following terms and definitions given in RFC 5245 [22] apply:

ICE Lite
Full ICE
Host ICE candidates

Editor's note: Terminology from draft-ietf-rtcweb-overview needs to be endorsed as part of the terminology of this document. This document uses the terms "WebRTC device" which it is understood will be changed to "non-WebRTC browser".

3.2 Abbreviations

For the purposes of the present document, the abbreviations given in 3GPP TR 21.905 [1] and the following apply. An abbreviation defined in the present document takes precedence over the definition of the same abbreviation, if any, in 3GPP TR 21.905 [1].

CN Core Network

CSCF	Call Session Control Function
DCEP	Data Channel Establishment Protocol
eP-CSCF	enhanced Proxy CSCF
IM	IP Multimedia
IP	Internet Protocol
WAF	WebRTC Authorization Function
WebRTC	Web Real-Time Communication
WWSF	WebRTC Web Server Function

4 Overview of WebRTC access to IMS

4.1 General

The relationship between functional entities for the interface at the W1 reference point, between the WWSF and the UE, the interface at the W2 reference point, between the eP-CSCF and the UE, the interface at the W3 reference point, between the UE and the eIMS-AGW, and the interface at the W4 reference point, between the WWSF and the WAF, are defined in annex U of 3GPP TS 23.228 [4].

The relationship between the functional entities for interface at the Mw reference point, between the eP-CSCF and the remainder of the IP multimedia core network subsystem, is defined in 3GPP TS 23.228 [4].

A number of appropriate mechanisms exist for signalling communication between the WIC and the eP-CSCF. Sucessful use of a mechanism other than those specified in this document will require some form of prior agreement between the operator of the WWSF and the operator of the eP-CSCF, as to the nature of the signalling mechanism that is to be adopted, and therefore the interworking required at the eP-CSCF. The mechanism of prior agreement and the nature of such agreement is not defined in this document.

A signalling transport mechanism for SIP is standardised in this release of this document, i.e. SIP over websockets (see RFC 7118 [2]), but this is not a mechanism that has to be supported by all eP-CSCFs.

When SIP over websockets is used, it can be appropriate for the SIP used to conform to the definitions for SIP on the Gm reference point as specified in 3GPP TS 24.229 [3]. Such a requirement is not mandatory, but where other SIP mechanisms are used:

a) the usage will require some form of prior agreement with the operator of the eP-CSCF, as to the nature of the signalling mechanism that is to be adopted; and

b) the SIP mechanisms will have to enable the eP-CSCF to conform to the SIP requirements over the Mw reference point to the remainder of the IP multimedia core network subsystem as specified in 3GPP TS 24.229 [3].

SDP is used for the signalling session information between the WIC and the eP-CSCF. Such SDP conforms to requirements for SDP on the Gm reference point.

5 Functional entities

5.1 General

5.2 WIC (WebRTC IMS Client)

A WebRTC IMS Client (WIC) establishing the service control signalling path over W2 interface, that is compliant with this specification shall implement the role of WIC capabilities defined in subclause 6.2, subclause 7.2 and subclause 8.2.

Where SIP over websockets is used, as specified in RFC 7118 [2], and no alternative SIP profiles have been agreed between the operator of the eP-CSCF and the operator of the WWSF, then the SIP used by the WIC over the W2 reference point shall conform to the requirements for UE over the Gm reference point as specified in 3GPP TS 24.229 [3].

When the WebSocket protocol is used, the WIC shall act as a WebSocket Client, as defined in RFC 6455 [24].

The SDP used shall conform to the requirements for UE over the Gm reference point as specified in 3GPP TS 24.229 [3] and further specified in the present document.

5.3 WWSF (WebRTC Web Server Function)

The WebRTC Web Server Function (WWSF) is the initial point of contact in the Web that controls access to the IMS communications services for the WIC as specified in 3GPP TS 23.228 [4].

5.4 WAF (WebRTC Authorisation Function)

The WebRTC Authorisation Function (WAF) issues authorization tokens that are provided to the WIC via the WWSF as specified in 3GPP TS 23.228 [4] and 3GPP TS 33.203 [9].

NOTE: The WWSF and the WAF realisations can be physically co-located or physically separate.

5.5 eP-CSCF (P-CSCF enhanced for WebRTC)

For the Mw reference point, the eP-CSCF shall conform to the requirements for the P-CSCF as specified in 3GPP TS 24.229 [3].

Where SIP over websockets is used, as specified in RFC 7118 [2], and no alternative SIP profile have been agreed between the operator of the eP-CSCF and the operator of the WWSF, then the SIP used by the eP-CSCF over the W2 reference point shall conform to the requirements for P-CSCF over the Gm reference point as specified in 3GPP TS 24.229 [3].

The SDP used by the eP-CSCF over the W2 reference point shall conform to the requirements for UE over the Gm reference point as specified in 3GPP TS 24.229 [3] and further specified in the present document.

5.6 eIMS-AGW (IMS Access Gateway enhanced for WebRTC)

The functionality of the eIMS-AGW is specified in 3GPP TS 23.228 [4] and in 3GPP TS 23.334 [13].

5A Data transport

5A.1 General

Data transport is the support of TCP, UDP and the means to securely set up connections between entities, as well as the functions for deciding when to send data: Congestion management, bandwidth estimation and so on.

5A.2 UE

A UE supporting WebRTC shall support the WebRTC device functionality as specified in draft-ietf-rtcweb-overview [30] clause 4, excluding requirements, if any, relating to specific audio and video codecs that are indirectly referenced within the draft-ietf-rtcweb-overview [30] clause 4.

Editor's note: This clause references draft-ietf-rtcweb-transports-06 which uses the terminology "WebRTC browser", "WebRTC endpoint" and "WebRTC device" for both ends of the transport. STUN and TURN introduce further "server" and "client" terminology that has to be allowed for.

5A.3 WWSF (WebRTC Web Server Function)

There are no data transport requirements for the WWSF.