**3GPP TSG Meeting #127-bis-e**  ***730***

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| *CR-Form-v12.2* |
| **Pseudo CHANGE REQUEST** |
|  |
|  |  | **CR** |  | **rev** |  | **Current version:** |  |  |
|  |
| *For* [***HE******LP***](http://www.3gpp.org/3G_Specs/CRs.htm#_blank)*on using this form: comprehensive instructions can be found at* [*http://www.3gpp.org/Change-Requests*](http://www.3gpp.org/Change-Requests)*.* |
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| --- | --- | --- | --- | --- | --- | --- | --- | --- |
| ***Proposed change affects:*** | UICC apps |  | ME | **X** | Radio Access Network |  | Core Network | **X** |

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|  |
| ***Title:***  | iRTCW Interoperability and codec requirements |
|  |  |
| ***Source to WG:*** | , Fraunhofer IIS, Qualcomm |
| ***Source to TSG:*** | SA4 |
|  |  |
| ***Work item code:*** | <Related\_WIs> |  | ***Date:*** | 3rd April 2024 |
|  |  |  |  |  |
| ***Category:*** | B |  | ***Release:*** |  Rel-18 |
|  | *Use one of the following categories:****F*** *(correction)****A*** *(mirror corresponding to a change in an earlier release)****B*** *(addition of feature),* ***C*** *(functional modification of feature)****D*** *(editorial modification)*Detailed explanations of the above categories canbe found in 3GPP [TR 21.900](http://www.3gpp.org/ftp/Specs/html-info/21900.htm). | *Use one of the following releases:Rel-8 (Release 8)Rel-9 (Release 9)Rel-10 (Release 10)Rel-11 (Release 11)…Rel-16 (Release 16)Rel-17 (Release 17)Rel-18 (Release 18)Rel-19 (Release 19)* |
|  |  |
| ***Reason for change:*** | Missing interoperability requirements for WebRTC clients |
|  |  |
| ***Summary of change:*** | Adding a reference to TS 26.114 for interoperability of audio and video support |
|  |  |
| ***Consequences if not approved:*** | Specification is not interoperable for media.  |
|  |  |
| ***Clauses affected:*** | New clause |
|  |  |
|  | **Y** | **N** |  |  |
| ***Other specs*** |  | **X** |  Other core specifications  | TS/TR ... CR ...  |
| ***affected:*** |  |  |  Test specifications | TS/TR ... CR ...  |
| ***(show related CRs)*** |  |  |  O&M Specifications | TS/TR ... CR ...  |
|  |  |
| ***Other comments:*** |  |
|  |  |
| ***This CR's revision history:*** |  |

## ===== CHANGE =====

# 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] 3GPP TS 26.506: "5G Real-time Media Communication Architecture (Stage 2)".

[3] 3GPP TS 26.510: "Media delivery; interactions and APIs for provisioning and media session handling".

[4] 3GPP TS 29.500: "5G System; Technical Realization of Service Based Architecture; Stage 3".

[5] IETF RFC 7231: "Hypertext Transfer Protocol (HTTP/1.1): Semantics and Content".

[6] 3GPP TS 26.512: "5G Media Streaming (5GMS); Protocols".

[7] IETF RFC 8834 (2021): "Media Transport and Use of RTP in WebRTC".

[8] IETF RFC 8835 (2021): "Transports for WebRTC".

[9] 3GPP TS 23.003: "Numbering, addressing and identification".

[10] IETF RFC 8829 (2021): "JavaScript Session Establishment Protocol (JSEP)".

[11] IETF RFC 7807 (2016): "Problem Details for HTTP APIs".

[12] IETF RFC 8825 (2021): "Overview: Real-Time Protocols for Browser-Based Applications".

[13] IETF RFC 5124 (2008): "Extended Secure RTP Profile for Real-time Transport Control Protocol (RTCP)-Based Feedback (RTP/SAVPF)".

[14] IETF RFC 7007 (2013): "Update to Remove DVI4 from the Recommended Codecs for the RTP Profile for Audio and Video Conferences with Minimal Control (RTP/AVP)".

[15] IETF RFC 3551 (2003): "RTP Profile for Audio and Video Conferences with Minimal Control".

[16] IETF RFC 4585 (2006): "Extended RTP Profile for Real-time Transport Control Protocol (RTCP)-Based Feedback (RTP/AVPF)".

[17] IETF RFC 3711 (2004): "The Secure Real-time Transport Protocol (SRTP)".

[18] IETF RFC 5104 (2008): "Codec Control Messages in the RTP Audio-Visual Profile with Feedback (AVPF)".

[19] IETF RFC 4588 (2006): "RTP Retransmission Payload Format".

[20] 3GPP TS 26.114: " IP Multimedia Subsystem (IMS); Multimedia Telephony; Media handling and interaction".

[21] IETF RFC 2616 (1999): "Hypertext Transfer Protocol -- HTTP/1.1".

[22] IETF RFC 7478 (2015): "Web Real-Time Communication Use Cases and Requirements".

[23] 3GPP TS 26.119: "Media Capabilities for Augmented Reality"

[24] 3GPP TS 38.331: "NR; Radio Resource Control (RRC); Protocol specification".

[25] Apple: "Getting Raw Accelerometer Events".

[26] Google: "Sensor Coordinate System".

[27] ITU-R Recommendation BT.601-7 (03/2011): "Studio encoding parameters of digital television for standard 4:3 and wide screen 16:9 aspect ratios".

[28] Microsoft: "Microphone Array Geometry Descriptor Format".

[29] IETF RFC 7874 (2016): "WebRTC Audio Codec and Processing Requirements"

[30] IETF RFC 7742 (2016): "WebRTC Video Processing and Codec Requirements"

1. End of Change

## ===== CHANGE =====

#### 4.3.1.3 Media transport (RTC-4m) procedures

This interface is used for transmission of media and other related data between two or more WebRTC endpoints. The WebRTC framework of the RTC endpoint send/receive the media data, application data and/or media related meta-data to/from RTC AS (e.g., MF) or other RTC endpoint based on the input from the RTC aware application (e.g., Native WebRTC app and Web app).

In the context of this specification for webRTC endpoints, neither the requirements for WebRTC endpoints for audio codecs as defined in IETF RFC RFC 7874 [29] nor the requirements for WebRTC endpoints for video codecs as defined in IETF RFC 7742 [30] apply. For codecs support in webRTC endpoints in the context of this specification, please refer to clause 16.

Media transport for RTC-4m is established based on the collaboration scenario defined in 3GPP TS 26.506 [2] and the signalling protocol applied for the media session establishment.

1. Start of Change 3 – New section

# 16 Media capabilities

This specification primarily specifies the protocols and APIs for real-time communication. The APIs and protocols defined in this specification are not restricted to specific codecs or media capabilities. In this specification, neither the requirements for WebRTC endpoints for audio codecs as defined in IETF RFC RFC 7874 [29], nor the requirements for WebRTC endpoints for video codecs as defined in IETF RFC 7742 [30] apply.

However, to support minimum service interoperability, a terminal implementing the protocols and APIs defined in the present document should implement

* The UE codec requirements for speech as specified in TS 26.114 [20], if speech/audio is supported.
* The UE codec requirements for video as specified in TS 26.114 [20], if video is supported.

Transcoding free operation to UEs implementing IMS-based codecs and media capabilities as defined in TS 26.114 [20] should be supported. If supported, a terminal shall implement the UE codec and media handling requirements as specified in TS 26.114 [20].

1. End of Change