**3GPP TSG-WG SA4 #127 meeting *-240319***

**Sophia-Antipolis, France, Jan. 29 – Feb. 2, 2024 Revision of S4-240146**

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| *CR-Form-v12.2* |
| **Pseudo CHANGE REQUEST** |
|  |
|  | **26.113** | **CR** |  | **rev** | **0** | **Current version:** | **1.0.0** |  |
|  |
| *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] pCR on 26113: API details |
|  |  |
| ***Source to WG:*** | Samsung Electronics, Co., LTD |
| ***Source to TSG:*** | S4 |
|  |  |
| ***Work item code:*** | iRTCW |  | ***Date:*** | 23 Jan. 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)* |
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| ***Reason for change:*** |  |
|  |  |
| ***Summary of change:*** |  |
|  |  |
| ***Consequences if not approved:*** |  |
|  |  |
| ***Clauses affected:*** |  |
|  |  |
|  | **Y** | **N** |  |  |
| ***Other specs*** |  |  |  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:*** |  |

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| **First Change** |

# 7 Media hosting interface (RTC-2)

Interfaces of this reference point are not specified in this release.

NOTE: The usage of content hosting at reference point RTC-2 is FFS.

# 8 RTC AS to RTC AF APIs (RTC-3)

Use of this reference point is outside the scope of RTC as described in TS 26.506 [2]

NOTE: APIs specified in clause 9 and clause X [Mas\_Configuration] of TS 26.510 [3] are applicable for this reference point.

# 9 Media-centric transport interface (RTC-4)

## 9.1 General

This clause deals with the interface to transport media over WebRTC session and signalling information at reference point RTC-4. TS 26.506 [2] specifies various collaboration scenario depending on the usable network entities in the trusted domain, leading to the different interactions and operations at RTC-4.

- Collaboration scenario 1: WebRTC session is completely managed over the top and no APIs at RTC-4 is specified.

- Collaboration scenario 2: ICE function is present in the trusted DN and only media transport is specified in clause 9.2 when TURN is involved for WebRTC session.

- Collaboration scenario 3 and 4: In addition to collaboration scenario 2, trusted signalling server and trusted media function is available. WebRTC framework communicates with RTC AS for both media transport and signalling exchange, as specified in clause 9.2 and 9.3 respectively.

## 9.2 Media transport (RTC-4m)

WebRTC framework in RTC endpoint may transport media data and/or other related data to RTC AS at reference point RTC-4m.

For the case of media data, RTC endpoint transmits any combination of video, audio, and speech using RTP for WebRTC (RFC 8834 [7]).

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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. However, in order to support minimum service interoperability, a terminal implementing the protocols and APIs defined in the present document [should|shall] implement the UE codec and media handling requirements as specified in TS 26.114 [xx].

NOTE: It is expected that terminals implementing this specification also implement TS 26.114 [xx] and hence the above recommendation is expected to be fulfilled.

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If RTC endpoint transports those media types, then it shall support the extended secure RTP profile for RTCP-based feedback (RTP/SAVPF) (RFC 5124 [13]), as extended by RFC 7007 [14]. Encoded media stream shall be encapsulated into the secure RTP packet as specified in RFC 3711 [17].

For the case of other related data such as application data or metadata, RTC endpoint shall use WebRTC Data Channel [29] and therefore support the encapsulation of SCTP over DTLS as defined in [30].

## 9.3 Signalling exchange (RTC-4s)

Signalling exchange refers a series of interactions to exchange the configuration information between two RTC endpoints (e.g., between applications (Native WebRTC Application/Web App) via WSF) to create and to manage RTCPeerConnection. It includes the available transport protocol, NAT traversal route, network addresses as well as the codecs and media types in common between two RTC endpoints or between the RTC endpoint and the trusted media function.

Those signalling information is exchanged based on the full-duplex reliable WebSocket connection, as specified in clause 13.2.

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| **Second Change** |

# 11 Media session handling client API (RTC-6, RTC-11)

Reference point RTC-6 is used to prepare consumption reporting parameters to be reported to RTC AF at reference point RTC-5. If consumption reporting for WebRTC session is configured, the RTC MSH shall regularly determine the consumption reporting parameters defined in clause 10.3.6 of TS 26.510 [3] shall report these values.

Reference point RTC-11 is used to

# 12 Client interface (RTC-7)

Reference point RTC-7 is used to communicate between Native WebRTC application and WebRTC framework for establishment and management of RTCPeerConnection, which is equivalent to WebRTC APIs specified by W3C [31].

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| **Third Change** |

# 2 References

[29] IETF RFC 8831 (2021): " WebRTC Data Channels".

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

[31] W3C Recommendation: WebRTC: Real-Time Communication in Browsers, March 2023