**3GPP TSG SA WG4 Meeting #128*****S4-240977***

**Jeju, Korea, 20th–24th May 2024**

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| *CR-Form-v12.0* | | | | | | | | |
| **Pseudo CHANGE REQUEST** | | | | | | | | |
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|  | **26.113** | **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] Updates on RTC-related API and reference point | | | | | | | | | |
|  |  | | | | | | | | | |
| ***Source to WG:*** | Samsung, | | | | | | | | | |
| ***Source to TSG:*** | S4 | | | | | | | | | |
|  |  | | | | | | | | | |
| ***Work item code:*** | iRTCW | | | | |  | ***Date:*** | | | 2024-05-20 |
|  |  | | | |  | |  | | |  |
| ***Category:*** |  |  | | | | | ***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 can be 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-15 (Release 15) Rel-16 (Release 16) Rel-17 (Release 17) Rel-18 (Release 18)* | |
|  |  | | | | | | | | | |
| ***Reason for change:*** | | Need alignments according to 26.510 progress | | | | | | | | |
|  | |  | | | | | | | | |
| ***Summary of change:*** | | * WebRTC endpoint -> RTC Access Function * Remove Configuration Provisioning API (replaced into Provisioning Sessions API) * New RTC-12 reference point to distinguish peer-to-peer connection * WebRTC session 🡪 RTC session | | | | | | | | |
|  | |  | | | | | | | | |
| ***Consequences if not approved:*** | |  | | | | | | | | |
| ***Q*** | |  | | | | | | | | |
| ***Clauses affected:*** | |  | | | | | | | | |
|  | |  | | | | | | | | |
|  | | **Y** | **N** |  | | | |  | | |
| ***Other specs*** | |  | **X** | Other core specifications | | | |  | | |
| ***affected:*** | |  | **X** | Test specifications | | | |  | | |
| ***(show related CRs)*** | |  | **X** | O&M Specifications | | | |  | | |
|  | |  | | | | | | | | |
| ***Other comments:*** | |  | | | | | | | | |
|  | |  | | | | | | | | |
| ***This CR's revision history:*** | |  | | | | | | | | |

First change

## 2 References

[34] OpenAPI: "OpenAPI 3.0.0 Specification", <https://github.com/OAI/OpenAPI-Specification/blob/master/versions/3.0.0.md>.

Next change

## 4.1 General

This clause defines all procedures for real-time media communication using the different RTC reference points. Table 4.1-1 summarises the APIs used to provision and use RTC features specified in TS 26.506 [2].

Table 4.1‑1: Summary of APIs relevant to RTC features

|  |  |  |  |  |
| --- | --- | --- | --- | --- |
| RTC  feature | Abstract | Relevant APIs | | |
| Interface | API name | Clause |
| Content configuration | Content delivery is configured according to Configuration Provisioning associated with a Provisioning Session. | RTC-1 | Provisioning Sessions API | 6.2 |
| RTC-5 | Service Access Information API | 10.2 |
| Metrics reporting | The RTC endpoint uploads metrics reports to the RTC AF according to a provisioned Metrics Reporting Configuration it obtains from the Service Access Information for its Provisioning Session. | RTC-1 | Provisioning Sessions API | 6.2 |
| Metrics Reporting Provisioning API | 6.6 |
| RTC-5 | Service Access Information API | 10.2 |
| Metrics Reporting API | 10.5 |
| Consumption reporting | The RTC endpoint provides feedback reports on currently consumed content according to a provisioned Consumption Reporting Configuration it obtains from the Service Access Information for its Provisioning Session. | RTC-1 | Provisioning Sessions API | 6.2 |
| Consumption Reporting Provisioning API | 6.3 |
| RTC-5 | Service Access Information API | 10.5 |
| Consumption Reporting API | 10.6 |
| Dynamic Policy invocation | The RTC endpoint activates different traffic treatment policies selected from a set of Policy Templates configured in its Provisioning Session. | RTC-1 | Provisioning Sessions API | 6.2 |
| Policy Templates Provisioning API | 6.5 |
| RTC-5 | Service Access Information API | 10.2 |
| Dynamic Policies API | 10.3 |
| Network Assistance | The RTC endpoint requests bit rate recommendations and delivery boosts from the RTC AF. | RTC-5 | Service Access Information API | 10.2 |
| Network Assistance API | 10.4 |
| Edge content processing | Edge resources are provisioned for processing content in RTC sessions. | RTC-1 | Provisioning Sessions API | 6.2 |
| Edge Resources Provisioning API | 6.4 |
| RTC-5 | Service Access Information API | 10.2 |

Next change

## 4.3 Procedures for media content and signalling transport

### 4.3.1 Media transport (RTC-4m, RTC-12) procedures

#### 4.3.1.1 General

In the RTC System, real-time media shall be communicated at either reference point RTC-4m or RTC-12.

- RTC-12 shall be used for peer-to-peer communication between multiple RTC Access Functions in UEs where this is permitted by the underlying 5G System.

- RTC-4m shall be used for communication between the RTC Access Function in the UE and the RTC AS, and between multiple RTC Access Functions in UEs where peer-to-peer communication is not permitted by the underlying 5G System.

In addition, Rreference point RTC-4 interface may be further split into the *signalling part* (RTC-4s) and the *media transport part* (RTC-4m), depending on the collaboration scenario as specified in 3GPP TS 26.506 [2].

Table 4.3.1.1-1 describes the associated reference points for collaboration scenarios.

Table 4.3.1.1‑1: Associated reference point RTC-4 and RTC-12 for collaboration scenarios

|  |  |  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- |
| Type | | | Collaboration scenario 1 | | Collaboration scenario 2 | | Collaboration scenario 3 | | Collaboration scenario 4 | |
| Media,  metadata | UE-to-RTC AS | | X | | RTC-4m (NOTE 2) | | RTC-4m | | RTC-4m | |
| Peer-to-Peer | | X | | RTC-12 | | RTC-12 | | RTC-12 | |
| Signalling | | | X | | X | | RTC-4s | | RTC-4s | |
| NOTE 1: X denotes that the corresponding reference point is not the scope of the present document.  NOTE 2: For the case when TURN server within ICE Function is involved to the other RTC endpoint. | | | | | | | | | | |

#### 4.3.1.2 Media transport procedures at RTC-4m

This reference point is used for transmission of media and other related data between the RTC Access Function of the UE and the ICE Function and (in some RTC sessions) the Media Function of the RTC AS. The RTC Access Function sends/receives the media data, application data and/or media related metadata to/from the RTC AS (e.g., Media Function) or another RTC endpoint based on the input from the RTC application (i.e., *Native WebRTC App* or *Web app*).

In the context of the present document, neither the requirements for audio codecs and processing as defined in IETF RFC 7874 [32], nor the requirements for video codecs and processing as defined in IETF RFC 7742 [33] apply to RTC endpoints. The codecs that RTC endpoints are required to support are specified in clause 16.

#### 4.3.1.3 Media transport procedures at RTC-12

This reference point is used for direct peer-to-peer transmission of media and other related data between multiple RTC Access Functions in UEs. The procedures and protocols supported at this reference point shall be a subset of those at reference point RTC-4 (see clause 4.3.1.2), excluding the functionalities for media processing in the Media Function of the RTC AS (e.g., split rendering).

### 4.3.2 Signalling (RTC-4s) procedures

This reference point is used for the exchange of signalling messages related to the RTC session between the RTC Application of the UE and the WebRTC Signalling Function of the RTC AS. The RTC application (i.e., *Native WebRTC App* or *Web app*) sends/receives signalling message to/from RTC AS (i.e., WebRTC Signalling Function) via reference point RTC-4s and RTC-7. Signalling procedures for RTC-4s refer to the procedure specified in the signalling protocol for RTC System in clause 13.2.

If a WebRTC Signalling Function is provided in RTC AS, an RTC Application shall configure itself to use one of the WebRTC Signalling Function server (e.g., use the WebRTC Signalling Function server which supports the SWAP protocol listed in the swapEndpoints in the Service Access Information message obtained by the RTC Media Session Handler at reference point RTC-5. The configured signalling server information is sent to RTC Application via reference point RTC-6). Using this information, the RTC Application communicates with the configured signalling server for media session setup (e.g., SDP negotiation) at RTC-4s (throughout RTC-7).

### 4.3.3 UE media delivery (RTC-7) procedures

Reference point RTC-7 is used for the following purposes:

- For the case when RTC Application is the *Native WebRTC App*: To enable to use the RTC Access Function for media handling (e.g., gathering media capability information of the UE, controlling media transport). It is also used for establishing and controlling RTC session by using API provided by the RTC Access Function.

- For the case when RTC Application is the *Web App*: To enable to use W3C-defined JavaScript APIs including WebRTC API exposed by RTC Access Function to Web App for establishing and controlling RTC session.

Next change

# 6 Provisioning interface (RTC-1)

## 6.1 General

This clause defines provisioning API used by the Application Provider to provision resources for their real-time communication sessions. The provisioning API is a subset of that specified in clause 8 of TS 26.510 [3].

Table 6.1-1 lists the subset of Maf\_Provisioning APIs specified in TS 26.510 [3] that are applicable to the RTC System. The OpenAPI specification for this subset is specified in clause B.2.

Table 6.1‑1: List of APIs relevant to RTC-1



|  |  |  |  |
| --- | --- | --- | --- |
| API name | Summary of usage by RTC Application Provider | TS 26.510 [3] clause | |
| Procedures specification | API specification |
| Provisioning Sessions | Instantiate and manipulate Provisioning Sessions in the RTC AF | 5.2.2 | 8.1 |
| Server Certificates Provisioning | Provision a set of Server Certificates associated with the parent Provisioning Session that the RTC AS may present at reference point RTC‑4. | 5.2.4 | 8.4 |
| Edge Resources Provisioning | Provision a set of configurations used to deploy the RTC AS associated with the parent Provisioning Session as a set of Edge Application Servers in the Edge Data Network. | 5.2.6 | 8.6 |
| Policy Templates Provisioning | Provision a set of Policy Templates within the scope of a parent Provisioning Session that can subsequently be applied to relevant RTC sessions. | 5.2.7 | 8.7 |
| Real-time Media Communication Provisioning | Provision an RTC configuration within the scope of a parent Provisioning Session for use by the RTC Access Function in facilitating RTC session. The configuration is included in Service Access Information retrieved from the RTC AF by the RTC Media Session Handler. | 5.2.10 | 8.10 |
| Metrics Reporting Provisioning | Provision the metrics collection and reporting procedure for a particular parent Provisioning Session at reference point RTC-1. | 5.2.11 | 8.11 |
| Consumption Reporting Provisioning | Provision the consumption reporting procedure for a particular parent Provisioning Session. | 5.2.12 | 8.12 |

## 6.2 Provisioning Sessions API

The Provisioning Sessions API is used by RTC Application Provider to instantiate and manipulate Provisioning Sessions in the RTC AF.

The relevant provisioning procedures are specified in clause 5.2.2 of TS 26.510 [3].

The resource structure and the data model are specified in clause 8.2 of TS 26.510 [3].



## 6.3 Real-time Media Communication provisioning API

The Real-time Media Communication provisioning API is used by the RTC Application Provider to provision configuration that will be relayed to the RTC Media Session Handler for usage with RTC sessions of that RTC Application Provider.

The relevant provisioning procedures are specified in clause 5.2.10 of TS 26.510 [3].

The resource structure and the data model of the RTCConfiguration resource are specified in clause 8.10 of TS 26.510 [3].

## 6.4 Server Certificates Provisioning

The Server Certificates Provisioning API is a RESTful API that is used by the RTC Application Provider to provision server certificates that the RTC AS may present at reference point RTC 4.

The relevant provisioning procedures are specified in clause 5.2.4 of TS 26.510 [3].

The API is specified in clause 8.4 of TS 26.510 [3].

## 6.5 Edge Resources Provisioning API

The Edge Resources Provisioning API is used by the RTC Application Provider to provision edge resource usage for RTC sessions associated with the parent Provisioning Session. The information serves as a template to select or instantiate the appropriate EAS instance that will serve the media session to the UE.

The relevant provisioning procedures are specified in clause 5.2.6 of TS 26.510 [3].

The resource structure and the data model are specified in clause 8.6 of TS 26.510 [3].

## 6.6 Policy Templates Provisioning API

The Policy Templates Provisioning API allow a RTC Application Provider to configure a set of Policy Templates within the scope of a Provisioning Session that can subsequently be applied to RTC sessions belonging to that Application Provider using the Dynamic Policies API specified in clause 10.3 of the present document.

The relevant provisioning procedures are specified in clause 5.2.7 of TS 26.510 [3].

The resource structure and the data model are specified in clause 8.7 of TS 26.510 [3].

## 6.7 Metrics Reporting Provisioning API

The Metrics Reporting Provisioning API allows a RTC Application Provider to configure the Metrics Collection and Reporting procedure for a particular RTC session at reference point RTC-1.

The relevant provisioning procedures are specified in clause 5.2.11 of TS 26.510 [3].

The resource structure and the data model are specified in clause 8.11 of TS 26.510 [3].

The metrics reporting scheme is signalled using in the **Scheme**@schemeIdUri property of the MetricsReportingConfiguration resource. The URN "urn:3GPP:ns:PSS:RTC:QM1" shall be indicated in this property.

## 8

The relevant provisioning procedures are specified in clause 5.2.12 of TS 26.510 [3].

 12

Next change

# 9 Media transport interface (RTC-4, RTC-12)

## 9.1 General

This clause deals with the interface to transport media data in RTC session at reference point RTC-4m or RTC-12 and signalling information at reference point RTC-4s. The different interactions and operations at RTC-4 and RTC-12 are specified in clause 4.3.1.

## 9.2 Media transport (RTC-4m, RTC-12)

The RTC Access Function of the RTC Client may transport media data and/or other related metadata either to the RTC AS at reference point RTC-4m or to an RTC Access Function in another RTC Client at reference point RTC‑12. The supported media capabilities of RTC endpoints are specified in clause 16.

For the case of media data, an RTC endpoint may transmit any combination of video, audio, and speech using RTP for WebRTC per RFC 8834 [7].

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

For the case of other related data such as application data or metadata, an RTC endpoint shall use the WebRTC Data Channel specified in RFC 8831 [29] and shall therefore support the encapsulation of SCTP over DTLS as specified in RFC 8261 [30].

## 9.3 Signalling exchange (RTC-4s)

Signalling exchange refers to a series of interactions to exchange configuration information between an RTC Application (i.e., a *Native WebRTC Application* or *Web App*) and the WebRTC Signalling Function of an RTC AS for the purpose of creating and managing an RTCPeerConnection. The exchange of signalling includes information about the available transport protocol, NAT traversal route, network addresses as well as the codecs and media types in common between the two RTC endpoints concerned.

This signalling information shall be exchanged over a full-duplex reliable WebSocket connection, as specified in clause 13.2.

NOTE: TS 26.119 [23] defines the device type and media capability identifiers specifically for UEs with immersive media capabilities. The use of these identifiers during the signalling exchange is for future study.

Next change

# 10 Media session handling interface (RTC-5, RTC-3)

## 10.1 General

This clause defines the media session handling API used by the RTC Media Session Handler to access resources exposed by the RTC AF at reference point RTC-5. The media session handling API is a subset of that specified in clause 9 of TS 26.510 [3].

NOTE: The APIs at reference point RTC‑3 are not specified in this release.

Table 10.1-1 lists the subset of Maf\_SessionHandling APIs specified in TS 26.510 [3] that are applicable to the RTC System. The OpenAPI specification for this subset is specified in clause B.4.

Table 10.1‑1: List of APIs relevant to RTC-5 and RTC‑3



|  |  |  |  |
| --- | --- | --- | --- |
| API name | Summary of usage by RTC Media Session Handler or RTC AS | TS 26.510 [3] clause | |
| Procedures specification | API specification |
| Service Access Information | Retrieve RTC configuration information. | 5.3.2 | 9.2 |
| Dynamic Policies | Request a specific QoS and charging policy to be applied to the data flows of an RTC session. | 5.3.3 | 9.3 |
| Network Assistance | Obtain bit rate recommendations and/or issue delivery boost requests during an ongoing RTC session. | 5.3.4 | 9.4 |
| Metrics Reporting | Report QoE metrics to the RTC AF. | 5.3.5 | 9.5 |
| Consumption Reporting | Report media consumption to the RTC AF. | 5.3.6 | 9.6 |

## 10.2 Service Access Information API

The Service Access Information API is used by the RTC Media Session Handler to acquire configuration information from the RTC AF that enables it to use the other media session handling APIs in clause 10.3 *et seq*.

The resource structure and the data model are specified in clause 9.2 of TS 26.510 [3].

When the Service Access Information API is used in RTC, the streamingAccess object shall not be present in the ServiceAccessInformation resource and the rtcClientConfiguration object shall be present as specified in table 9.2.3.1-1 of TS 26.510 [3].

## 10.3 Dynamic Policy API

The Dynamic Policy API allows the RTC Media Session Handler or the ICE Function of the RTC AS or the WebRTC Signalling Function of the RTC AS to request a specific QoS and charging policy to be applied to the data flows of an RTC session.

The relevant procedures are specified in clause 5.3.3 of TS 26.510 [3]

The resource structure and the data model are specified in clause 9.3 of TS 26.510 [3].

## 10.4 Network Assistance API

If AF-based Network Assistance is supported in the RTC System, then the Network Assistance API is first used to provision a Network Assistance Session resource. The Network Assistance Session resource can then be used to obtain bit rate recommendations and to issue delivery boost requests during the ongoing RTC session.

The relevant procedures are specified in clause 5.3.4 of TS 26.510 [3]

The resource structure and the data model are specified in clause 9.4 of TS 26.510 [3].

## 10.5 Metrics Reporting API

The Metrics Reporting API allows the RTC Media Session Handler to report QoE metrics to the RTC AF, as configured by the SerciveAccessInformation resource (see clause 10.2).

The relevant procedures are specified in clause 5.3.5 of TS 26.510 [3].

The reporting API is specified in clause 9.5 of TS 26.510 [3]

For RTC, clause 15.3.1 and clause 15.3.2 specify the required MIME content type and metrics report format for the 3GPP urn:‌3GPP:‌ns:‌PSS:‌RTC:‌QM1 metrics reporting scheme.

NOTE: When the WebRTC Signalling Function is used in an RTC session, QoE metrics may instead be reported to the WebRTC Signalling Function in the RTC AS.

## 10.6 Consumption Reporting API

The Consumption Reporting API allows the RTC Media Session Handler to report media consumption to the RTC AF, as configured by the SerciveAccessInformation resource (see clause 10.2).

The relevant procedures are specified in clause 5.3.6 of TS 26.510 [3].

The reporting API and report format are specified in clause 9.6 of TS 26.510 [3].

Next change

# 12 Client interface (RTC-7)

Reference point RTC-7 is used to communicate between RTC Application and RTC Access Function for establishment and management of an RTCPeerConnection. The procedures at this reference point are equivalent to those supported by the W3C-defined JavaScript APIs including WebRTC API.

Next change

# 14 Packet-loss handling

## 14.1 Packet-loss handling mechanisms in RTC endpoints

### 14.1.1 Video

#### 14.1.1.1 General

The following packet loss handling mechanisms are recommended in RFC 8834 [7] and RFC 8835 [8] for a RTC endpoint defined in RFC 8825 [12].

RTC endpoints offering video shall support extended secure RTP profile for RTCP-based feedback (RTP/SAVPF) (RFC 5124 [13]), as extended by RFC 7007 [14]. The RTP/SAVPF profile is the combination of the basic RTP/AVP profile in RFC 3551 [15], the RTP profile for RTCP-based feedback (RTP/AVPF) in RFC 4585 [16], and the secure RTP profile (RTP/SAVP) in RFC 3711 [17].

The RTC endpoints behaviour can be controlled by allocating enough RTCP bandwidth using "b=RR:" and "b=RS:" and setting the value of "trr-int". The attributes "b=RS:<bw>" and "b=RR:<bw>" as defined in RFC 4585 [16] may be used to assign a different bandwidth (measured in bits per second) for RTCP messages to RTP senders and receivers, respectively. The attribute "trr-int" in SDP is used to specify the minimum time interval between two Regular (full compound) RTCP packets in milliseconds for a media session.

RTC endpoints are recommended to use the following mechanisms to recover from packet losses:

- AVPF Generic NACK

- Picture Loss Indication (PLI) feedback message

- Slice Loss Indication (SLI) feedback message

- Full Intra Request (FIR) feedback message

- Temporal-Spatial Trade-Off Request (TSTR)

- Temporary Maximum Media Stream Bit Rate Request (TMMBR)

- RTP Retransmission

These mechanisms offer different performance trade-offs according to channel conditions such as end-to-end delay, bandwidth, rate and packet loss profile.

#### 14.1.1.2 NACK messages

AVPF NACK messages are used by RTC endpoints to indicate non-received RTP packets for video. WebRTC receivers may send NACKs for missing RTP packets. RTP packet stream senders are required to understand the generic NACK message defined in RFC 4585 [16], but they can choose to ignore some or all of this feedback.

#### 14.1.1.3 PLI message

The Picture Loss Indication message is used by a receiver to tell the sending encoder that it lost the decoder context and would like to have it repaired. RTC endpoints that are sending media shall understand and react to PLI feedback messages as a loss-tolerance mechanism. Receivers can send PLI messages.

#### 14.1.1.4 SLI message

The Slice Loss Indication message as defined in RFC 4585 [16] is used by a WebRTC receiver to tell the encoder that it has detected the loss or corruption of one or more consecutive macro blocks and would like to have these repaired somehow. It should be that receivers generate SLI feedback messages if slices are lost when using a codec that supports the concept of macro blocks. A sender that receives an SLI feedback message should attempt to repair the lost slice(s).

#### 14.1.1.5 FIR message

The Full Intra Request message defined in RFC 5104 [18] is used to make a request by a WebRTC receiver for a new Intra picture from a WebRTC sender. RTC endpoints that are sending media shall understand and react to FIR feedback messages they receive. Support for sending FIR messages is optional.

#### 14.1.1.6 Temporal-Spatial Trade-Off Request (TSTR)

The temporal-spatial trade-off request and notification are defined in RFC 5104 [18]. This request can be used to ask the video encoder to change the trade-off it makes between temporal and spatial resolution -- for example, to prefer high spatial image quality but low frame rate. Support for TSTR requests and notifications in RTC endpoints is optional.

#### 14.1.1.7 Temporary Maximum Media Stream Bit Rate Request (TMMBR)

The Temporary Maximum Media Stream Bit Rate Request (TMMBR) feedback message is defined in RFC 5104 [18]. This request and its corresponding Temporary Maximum Media Stream Bit Rate Notification (TMMBN) message defined in RFC5104 are used by a WebRTC receiver to inform the sending party that there is a current limitation on the amount of bandwidth available to this receiver. RTC endpoints that are sending media are required to implement support for TMMBR messages and shall follow bandwidth limitations set by a TMMBR message received for their SSRC. The sending of TMMBR messages is optional.

Next change

Annex B (normative)  
OpenAPI representation of HTTP REST APIs

# B.1 General

The normative code specifying the APIs defined in clauses 6 and 10 of the present document, including JSON Schema representations of HTTP message bodies to be used with these APIs, is published on 3GPP Forge according to the OpenAPI 3.0.0 specification [34]. The YAML files corresponding to this version of the present document shall be published to the following location:

https://forge.3gpp.org/rep/all/5G\_APIs/-/tags/TSG104-Rel18

Informative copies of these YAML files shall be distributed with the present document for the convenience only. Where any discrepancy exisits, the version on 3GPP Forge shall be considered definitive.

# B.2 OpenAPI representation of RTC‑1 APIs

## B.2.0 Maf\_Provisioning API

The APIs used by the RTC Application Provider to provision the RTC AF are specified in the file named "TS26113\_Maf\_Provisioning.yaml".

# B.3 OpenAPI representation of RTC‑3 APIs

NOTE: The APIs used at reference point RTC‑3 are not specified in this release.

# B.4 OpenAPI representation of RTC‑5 APIs

## B.4.0 Maf\_SessionHandling API

The APIs used by the RTC Media Session Handler to handle media sessions in the RTC AF are specified in the file named "TS26113\_Maf\_SessionHandling.yaml".

End of changes