**3GPP TSG- Meeting #**

**, , -** revision of S4-250725

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| *CR-Form-v12.3* |
| **CHANGE REQUEST** |
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|  |  | **CR** |  | **rev** |  | **Current version:** |  |  |
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| *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:***  |   |
|  |  |
| ***Source to WG:*** |  |
| ***Source to TSG:*** | S4 |
|  |  |
| ***Work item code:*** |  |  | ***Date:*** |  |
|  |  |  |  |  |
| ***Category:*** |  |  | ***Release:*** |  |
|  | *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-17 (Release 17)Rel-18 (Release 18)Rel-19 (Release 19) Rel-20 (Release 20)* |
|  |  |
| ***Reason for change:*** | The conclusion of KI#9 and Ki#14 (traffic detection of multiplexed media flows) from TR 26.822 are as below The following aspects are concluded as principles for normative work:- Based on response from SA2, normative work on multiplexed RTP streams may be needed. Furthermore, it is recommended to add guidelines to TS 26.522 [2] for RTP senders that use multiplexing. There may be potential normative aspects to be added to TS 26.510 [50].When multiple RTP media streams are multiplexed in an RTP session, each media stream can be identified using the identification-tag (the values of "mid" attribute) in the SDP information. The RTP SDES header extension for MID make it possible for a 5G System or an RTP receiver to associate each PDU or PDU Set to a media stream when the the PDUs in a PDU Set carry the RTP SDES header extension for MID. To enable the traffic detection in 5G System, the mediaTransportParameters paremetr in the Application‌Flow‌Description object shall be updated to include the information of the multiplexed media identification information which includes ssrc id, MID, RTP SDES header extension for MID, Payload type and RTCP packet type fields. |
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| ***Summary of change:*** | Updated the Dynamic Policy API to include the details of *multiplexed media identification information* in protocol description*.*  |
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| ***Consequences if not approved:*** | Recommendations from work item description are not met, key 5GA features are not supported |
|  |  |
| ***Clauses affected:*** | 10.3 |
|  |  |
|  | **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:*** | Rev1:Provided a NOTE on usage of the *IP Filter Set with (S)RTP Multiplexed Media Identification Information* for traffic detection by the 5G system. |

First 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 9110 (2022): "HTTP Semantics".

[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 9112 (2022): "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", <https://developer.apple.com/documentation/coremotion/getting_raw_accelerometer_events>.

[26] Google: "Sensor Coordinate System", <https://developer.android.com/develop/sensors-and-location/sensors/sensors_overview>.

[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", <https://learn.microsoft.com/en-us/windows-hardware/drivers/audio/microphone-array-geometry-descriptor-format>.

[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. <https://www.w3.org/TR/webrtc/>

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

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

[34] 3GPP TS 26.247: "Transparent end-to-end Packet-switched Streaming Services (PSS); Progressive Download and Dynamic Adaptive Streaming over HTTP (3GP-DASH)".

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

[36] 3GPP TS 26.571: "5G System; Common Data Types for Service Based Interfaces; Stage 3".

[37] 3GPP TS 26.522: "5G Real-time Media Transport Protocol Configurations".

[38] 3GPP TS 29.514: "5G System; Policy Authorization Service".

[39] 3GPP TS 29.244: "Interface between the Control Plane and the User Plane Nodes".

second change

## 10.3 Dynamic Policy API

### 10.3.1 Introduction

The Dynamic Policy API allows the RTC Media Session Handler of the RTC Client or the ICE Function of the RTC AS or the WebRTC Signalling Function of the RTC AS to request a specific QoS and/or charging policy to be applied to the application flows of an RTC session. The Dynamic Policy API is invoked as a result of SDP negotiation during the WebRTC signalling phase of the 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].

### 10.3.2 Enabling PDU Set handling in dynamic policies

If specific QoS with PDU Set parameters is desired, and PDU Set marking is not enabled for the selected Policy Template as specified in clause 5.3.3.2 of TS 26.510 [3], the Media Session Handler shall additionally populate the mediaTransportParameters property of the Application‌Flow‌Description object (see clause 5.5.4.13 of TS 29.571 [36]) as follows when creating or updating a Dynamic Policy Instance based on that Policy Template:

- The transportProto property shall be set to the value SRTP.

- The rtpHeaderExtInfo object (see clause 5.5.4.14 of TS 29.571 [36]) shall be omitted.

- The rtpPayloadInfoList property shall contain a single member populated as follows:

- rtpPayloadTypeList shall be set to the *RTP Payload Type* value(s) to be used by the RTC endpoint (e.g., the RTC Access Function of an RTC Client) for the negotiated SRTP session(s) to be carried by the application flow in question.

- rtpPayloadFormat shall be populated as appropriate in the absence of RTP header extensions.

If PDU Set marking is required by the selected Policy Template as specified in clause 5.3.3.2 of TS 26.510 [3], the Media Session Handler shall additionally populate the mediaTransportParameters property of the Application‌Flow‌Description object (see clause 5.5.4.13 of TS 29.571 [36]) as follows when creating or updating a Dynamic Policy Instance based on that Policy Template:

- The transportProto property shall be set to the value SRTP.

- The properties of the rtpHeaderExtInfo object (see clause 5.5.4.14 of TS 29.571 [36]) shall be populated as follows:

- rtpHeaderExtType shall be set to PDU\_SET\_MARKING.

- rtpHeaderExtId shall be set to the value of the *ID* field to be used by the RTC endpoint (e.g., the RTC Access Function of an RTC Client) in the *RTP Header Extension for PDU Set Marking* on the application flow in question, as specified in clause 4.2 of TS 26.522 [37]. The value of this parameter is negotiated via the SDP offer/answer procedure during the WebRTC signalling phase of the RTC session.

- longFormat shall be set according to the use of the one- or two-byte *RTP Header Extension for PDU Set Marking*, as specified in clause 4.2.1 of TS 26.522 [37]. The value of this parameter is negotiated via the SDP offer/answer procedure during the WebRTC signalling phase of the RTC session.

- pduSetSizeActive shall be set to reflect the presence of the *PDU Set Size* field in the *RTP Header Extension for PDU Set Marking*, as specified in clause 4.2.4 of TS 26.522 [37]. The value of this parameter is negotiated via the SDP offer/answer procedure during the WebRTC signalling phase of the RTC session.

NOTE: The intention of the RTC Access Function of the RTC Client to include the optional NPDS (Number of PDUs in the PDU Set) field in the *RTP Header Extension for PDU Set Marking* is not yet signalled in advance to the 5G Core by means of a Boolean flag in the RtpHeaderExtInfo specified in clause 5.5.4.14 of TS 29.571 [36].

- The rtpPayloadInfoList property shall contain a single member populated as follows:

- rtpPayloadTypeList shall be set to the *RTP Payload Type* value(s) to be used by the RTC endpoint (e.g., the RTC Access Function of an RTC Client) for the negotiated SRTP session(s) to be carried by the application flow in question.

- rtpPayloadFormat shall be omitted because RTP header extensions are present.

In all PDUs it contributes at reference point RTC‑4m or RTC‑12 that fall within the scope of the application flow description, the RTC Access Function (Media Access Function) shall use the protocol indicated in transportProto; it shall set the SRTP header fields in accordance with rtpPayloadInfoList; and it shall include a one- or two- byte (consistent with the signalled length) *RTP Header Extension for PDU Set Marking* in the SRTP header with fields set according to the values declared in the rtpHeaderExtInfo property per above.

### 10.3.X Enabling multiplexed media flow handling in dynamic policies

If an RTC Session requires multiple media flows to be multiplexed into a single RTP Session, because the RTC endpoints involved have successfully negotiated media multiplexing as specified in clause 4.6 of TS 26.522 [37], and differentiated QoS handling is required for the multiplexed media streams by the dynamic policy invoker, the Media Session Handler or the RTC AS shall additionally populate the multiplexedMediaInfos property of the Application‌Flow‌Description object (see clause 5.5.4.13 of TS 29.571 [36]) as follows when creating or updating a Dynamic Policy Instance:

- The multiplexedMediaInfos property shall contain at least one MpxMediaInfoobject for each media stream in the multiplexed media application flow. The properties of the MpxMediaInfoobject are negotiated by the RTC Access Function of the RTC Client or the RTC AS using the BUNDLE group attribute in the SDP offer/answer procedure during the WebRTC signalling phase of the RTC Session. The properties of each MpxMediaInfo object (see clause 5.6.2.61 in TS 29.514 [38]) shall be populated as follows based on the RTP packet header values to be used by the sending RTC endpoint (i.e., the RTC Access Function of an RTC Client or the Media Function of the RTC AS) on the media stream in question:

- ssrcId may be set to the *synchronization source* value to be used by the sending RTC endpoint.

- payloadType shall be set to the *RTP Payload Type* value to be used by the sending RTC endpoint. The value of this parameter is negotiated via the SDP offer/answer procedure during the WebRTC signalling phase of the RTC Session. When this property is present all *RTP Payload Type* value(s) present in the rtpPayloadTypeList property of Protocol‌Description object shall be ignored.

- rsiMid shall be set to the value of the identification tag or media description identifier (MID) to be used by the sending RTC endpoint in the *SDES RTP Header Extension for MID* or the *RTCP MID SDES Item for MID*, as specified in clause 4.6 of TS 26.522 [37]*.* The value of this parameter is negotiated via the SDP offer/answer procedure during the WebRTC signalling phase of the RTC Session.

- rsheMid shall be set to the value of the local identifier or *ID* field to be used by the sending RTC endpoint in the *SDES RTP Header Extension for MID*, as specified in clause 4.6 of TS 26.522 [37]. The value of this parameter is negotiated via the SDP offer/answer procedure during the WebRTC signalling phase of the RTC Session.

- longFormat may be set according to the use of the one- or two-byte *RTP SDES Header Extension for MID*, as specified in clauses C.2.2 and C.2.3 of TS 26.522 [37].

- rtcpPt may be set to the RTCP Packet Type (PT) value to be used by the sending RTC endpoint.

NOTE: A combination of SSRC, Payload Type and/or MID values are required for multiplexed media identification.

In all PDUs it contributes at reference point RTC‑4m or RTC‑12 that fall within the scope of the application flow description, the sending RTC endpoint (i.e., the RTC Access Function of an RTC Client or the Media Function of the RTC AS) shall set the SRTP header fields in accordance with the MpxMediaInfoobject for the media stream in question; and it shall include a one- or two- byte *RTP SDES Header Extension for MID* in the SRTP header with fields set according to the values declared in the MpxMediaInfoobject per above to indicate the multiplexed media identification information.

NOTE: When multiplexed media identification marking is used in this way, multiplexed media traffic identification is performed by the 5G System for differentiated QoS treatment using the *IP Filter Set with (S)RTP Multiplexed Media Identification Information* feature defined in clause 8.2.5 of TS 29.244 [39], which inspects certain SRTP header fields −specifically the Synchronization Source identifier and the Payload Type(s) – in combination with the media description identifier information present in the relevant MpxMediaInfoobject.

End of changes