**3GPP TSG-SA WG4 Meeting #131-bis-eS4-251254**

**Online, 18 – 25 July 2025**

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| *CR-Form-v12.2* |
| **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:***  | [5G\_RTP\_Ph2] PDU handling and marking enhancements to Dynamic Policy API |
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| ***Source to WG:*** | Lenovo, Nokia, InterDigital Communications, BBC |
| ***Source to TSG:*** | S4 |
|  |  |
| ***Work item code:*** | 5G\_RTP\_Ph2 |  | ***Date:*** | 2025-07-14 |
|  |  |  |  |  |
| ***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-16 (Release 16)Rel-17 (Release 17)Rel-18 (Release 18)Rel-19 (Release 19)* |
|  |  |
| ***Reason for change:*** | Lack of support in the Real-Time Media Communications Dynamic Policy API for:* PDU Set handling of N6-unmarked PDUs (i.e., PDUs cannot/do not contain PDU Set infromation marking from the source)
* Dynamic traffic characteristics markings
* Media identification and multiplexing markings

***N6-unmarked PDUs enhancements***PDU Set and End of Data Burst marking only applies to RTP PDUs since marking is done via an RTP header extension. Hence, PDUs belonging to protocols such as RTCP, STUN, etc. cannot be marked i.e., they do not carry the PDU Set Information.In Rel-18, SA2 has agreed that the PSA UPF marks, in the downlink, each N6-unmarked PDU (“lone PDU”) with PDU Set Information into a PDU Set. If the UPF receives a PDU that does not belong to a PDU Set based on Protocol Description for PDU Set identification, the UPF still maps it to a PDU Set and determines the PDU Set Information by implementation-specific means.This means that for N6-unmarked PDUs, PDU Set Information must be determined by the UPF. For some elements of the PDU Set Information, this is straightforward, e.g., PSN=0 since the PDU Set has only one PDU, PSSize is equal to the size of the N6-unmarked PDU (since there is only one PDU in the PDU Set). However, for PSI, the UPF may only assign a preconfigured value (e.g. by the network operator) which may not reflect the application requirements.SA4 concluded in TR 26.822 that it would be beneficial for senders to signal application-defined PDU Set Importance (PSI) values to the 5GC for N6-unmarked PDUs. This signaling requires extensions to the Dynamic Policy API defined in TS 26.113.***Support for dynamic traffic characteristics marking***Dynamic traffic characteristics (i.e., data burst size, time to next burst, expedited transfer indication) have been defined in Rel-19 of TS 23.501 as downlink enhancements to support XR media services. Furthermore, TS 26.522 defined RTP header extensions to transport in user plane the dynamic traffic characteristics as required by TS 23.501. Yet, TS 26.113 lacks details about how dynamic traffic characteristics are applicable and usable in the context of RTC media delivery system protocols and Dynamic Policy API.***Support for media identification and multiplexing markings***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 an RTP receiver to associate each PDU or PDU Set to a media stream when the RTP PDUs carry the RTP SDES header extension for MID. To enable the traffic detection in 5G System, as defined in the IP Packet Filter Set of clause 5.7.3.2, TS 23.501. an *Application‌Flow‌Description* needs to 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:*** | ***N6-unmarked PDUs enhancements***The Media Session Handler in the Media Client includes a list of *unmarkedPDUInfo* objects in the *mediaTransportParameters* property of the *ApplicationFlowDescription* object when it invokes the Dynamic Policy API, if specific QoS with PDU Set parameters is desired for the application flows of an RTC session.If PDU Set marking is enabled, *unmarkedPDUInfoList* may be used to indicate the PSI values for N6-unmarked PDUs, i.e., PDUs of protocols that cannot be marked using the RTP HE for PDU Set marking such as RTCP or STUN packets.***Support for dynamic traffic characteristics marking***Complemented RTC Dynamic Policy API and the associated media delivery protocols details with dynamic traffic characteristics (incl. data burst size marking, time to next burst marking and expedited transfer indication marking).***Support for media identification and multiplexing markings***Updated the Dynamic Policy API to include the details of *multiplexed media identification information* in Application flow description |
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| ***Consequences if not approved:*** | No possibility to indicate and apply sender-defined PSI values to the 5GC for N6-unmarked PDUsLack of Stage 3 dynamic traffic characteristics and media identification/multiplexing support for RTC media delivery and misalignment with Stage 2 architecture and procedures. |
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| ***Clauses affected:*** | 10.3, 10.3.1 (re-sectioned), 10.3.2 (re-sectioned), 10.3.3 (new), 10.3.4 (new), B.1 |
|  |  |
|  | **Y** | **N** |  |  |
| ***Other specs*** | **X** |  |  Other core specifications  | TS26.510 CR0034TS26.522 CR0012TS26.506 CR0010 |
| ***affected:*** |  |  |  Test specifications |  |
| ***(show related CRs)*** |  |  |  O&M Specifications |  |
|  |  |
| ***Other comments:*** | This CR implements Stage-3 requirements in support of application-specific PDU handling in Dynamic Policies of RTC System. It specializes Stage-3 changes from TS 26.510 CR0034 to RTC system and implements Stage-2 of TS 26.506 CR0010.It depends on new RTP Header Extension for Expedited Transfer Indication, specified in TS26.522 CR0012 and already implemented in TS26.522 v19.1.0 after SA#108. |
|  |  |
| ***This CR's revision history:*** | Rev0* Merged contributions from CRs:
	+ **TS26.113 CR0005rev8 [**[**S4-251078**](https://apc01.safelinks.protection.outlook.com/?url=https%3A%2F%2Fwww.3gpp.org%2Fftp%2Ftsg_sa%2FWG4_CODEC%2FTSGS4_132_Fukuoka%2FDocs%2FS4-251078.zip&data=05%7C02%7Crstoica%40lenovo.com%7C3dfb2e887bcc446e641308ddbf800a7c%7C5c7d0b28bdf8410caa934df372b16203%7C0%7C0%7C638877279247567325%7CUnknown%7CTWFpbGZsb3d8eyJFbXB0eU1hcGkiOnRydWUsIlYiOiIwLjAuMDAwMCIsIlAiOiJXaW4zMiIsIkFOIjoiTWFpbCIsIldUIjoyfQ%3D%3D%7C0%7C%7C%7C&sdata=MqxhO4QD1hyOTdUcRePxt57%2FSlEwblsEOeLK4BuVpLQ%3D&reserved=0)**,** ***endorsed at SA4#132*]** – [5G\_RTP\_Ph2] Enhancements to RTC Dynamic Policy API for N6-unmarked PDUs.
	+ **TS 26.113 CR0008rev3 [**[**S4aR250120**](https://www.3gpp.org/ftp/TSG_SA/WG4_CODEC/3GPP_SA4_AHOC_MTGs/SA4_RTC/Docs/S4aR250120.zip)**, endorsed at SA4-(AH) RTC SWG post 132]** - [5G\_RTP\_Ph2] Enabling Dynamic Policy API with dynamic traffic characteristics markings.
	+ **TS 26.113 CR0010rev5 [**[**S4aR250120**](https://www.3gpp.org/ftp/TSG_SA/WG4_CODEC/3GPP_SA4_AHOC_MTGs/SA4_RTC/Docs/S4aR250120.zip)**, endorsed at SA4-(AH) RTC SWG post 132]** – [5G\_RTP\_Ph2] Enhancements to Dynamic Policy API for SDES RTP HE for MID.
* Removed square brackets around TTNB given LS out from RAN2 in [R2-2504812](https://www.3gpp.org/ftp/tsg_ran/WG2_RL2/TSGR2_130/Docs/R2-2504812.zip)
* Alignment with Stage-2 CR in TS26.506 CR0010rev2
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# Code changes

The code changes associated with this Change Request are available for review at the following URL on 3GPP Forge:

https://forge.3gpp.org/rep/sa4/amd-pro-med/-/merge\_requests/TBD/commits

The proposed changes are reproduced below for posterity.

\* \* \* \* 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".

[RFC8834] IETC RFC 8834: "Media Transport and Use of RTP in WebRTC", January 2021.

[29514] 3GPP TS 29.514: "5G System; Policy Authorization Service".

[29244] 3GPP TS 29.244: "Interface between the Control Plane and the User Plane nodes".

\* \* \* \* Next change \* \* \* \*

## 3.3 Abbreviations

For the purposes of the present document, the abbreviations given in 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 TR 21.905 [1].

API Application Programming Interface

AR Augmented Reality

DTLS Datagram Transport Layer Security

FFS For Further Study

FoV Field of View

HTTP Hyper-Text Transfer Protocol

ICE Interactive Connectivity Establishment

IMU Inertial Measurement Unit

MNO Mobile Network Operator

NAT Network Address Translation

OTT Over-The-Top

PSI PDU Set Importance

RGB Red-Green-Blue colour space

RTC Real-Time Communication

RTP Real-time Transport Protocol

RWT Response Wait Time

SCTP Stream Control Transmission Protocol

SRTCP Secure Real-time Transport Control Protocol

SRTP Secure Real-time Transport Protocol

SSE Server-Sent Events

STUN Session Traversal Utilities for NAT

SWAP Simple WebRTC Application Protocol

TLS Transport Layer Security

TURN Traversal Using Relays around NAT

WebRTC Web Real-Time Communication

XR Extended Reality

\* \* \* \* Next change \* \* \* \*

## 10.3 Dynamic Policy API

### 10.3.1 Introduction

The Dynamic Policy API allows a Dynamic Policy invoker (i.e., 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 Dynamic Policy invoker, i.e., the RTC Media Session Handler or the RTC AS, 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.

- When the unmarked-pdu-info attribute (as specified in clause 6.1 of TS 26.522 [37]) is present in the SDP offer/answer, the unmarkedPduInfoList property shall contain at least one unmarkedPduInfo member. The properties of the unmarkedPduInfo members of the unmarkedPduInfoList are negotiated by the RTC Access Function of the RTC Client via the SDP offer/answer procedure during the WebRTC signalling phase of the RTC session using the SDP attribute a=unmarked-pdu-info. The properties of each unmarkedPduInfo object (see clause 5.5.4.17 in TS 29.571 [36]) shall be populated as follows, in order of presence in the SDP offer/answer message:

- unmarkedProtocol shall indicate the application protocol used by N6-unmarked PDUs on the application flow in question*.*

- If the corresponding SDP media description includes an a=rtcp-mux or an a=rtcp-mux-only attribute, at least one unmarkedPduInfo member shall be present with its unmarkedProtocol property set to the value SRTCP.

- If the sender intends to indicate a default PDU Set Importance (PSI) value for N6-unmarked PDUs of non-video media types, an unmarkedPduInfo member shall be present with its unmarkedProtocol property set to the value SRTP. A default PSI value for N6-unmarked PDUs of non-video media types may only be indicated if RTP streams carrying such PDUs (e.g. audio streams) and video streams are multiplexed in the same RTP session.

NOTE 1: unmarkedProtocol may instead be set to the value STUN or OTHER depending on the value provided in the SDP negotiation during the WebRTC signalling phase.

- pduSetImportance shall be set to the desired PSI value for N6-unmarked PDUs on the application flow in question which uses the application protocol indicated by unmarkedProtocol. The setting shall follow the semantics defined for PSI in clause 4.2.4 of TS 26.522 [37], with a value in the range of 1 to 15 (inclusive).

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 Dynamic Policy invoker, i.e., the RTC Media Session Handler or the RTC AS, 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.

- pduSetPduCountActiveshall be set to reflect the presence of the *Number of PDUs in the PDU Set* 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.

- 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.

- When the unmarked-pdu-info attribute (as specified in clause 6.1 of TS 26.522 [37]) is present in the SDP offer/answer, the unmarkedPduInfoList property shall contain at least one unmarkedPduInfo member. The properties of the unmarkedPduInfo members of the unmarkedPduInfoList are negotiated by the RTC Access Function of the RTC Client via the SDP offer/answer procedure during the WebRTC signalling phase of the RTC session using the SDP attribute a=unmarked-pdu-info. The properties of each unmarkedPduInfo object (see clause 5.5.4.17 in TS 29.571 [36]) shall be populated as follows, in order of presence in the SDP offer/answer message:

- unmarkedProtocol shall indicate the application protocol used by N6-unmarked PDUs on the application flow in question*.*

- If the corresponding SDP media description includes an a=rtcp-mux or an a=rtcp-mux-only attribute, at least one unmarkedPduInfo member shall be present with its unmarkedProtocol property set to the value SRTCP.

- If the sender intends to indicate a default PDU Set Importance (PSI) value for N6-unmarked PDUs of non-video media types, an unmarkedPduInfo member shall be present with its unmarkedProtocol property set to the value SRTP. A default PSI value for N6-unmarked PDUs of non-video media types may only be indicated if RTP streams carrying such PDUs (e.g. audio streams) and video streams are multiplexed in the same RTP session and PDU Set marking is used for the video stream(s).

NOTE 2: unmarkedProtocol may instead be set to the value STUN *or* OTHER depending on the value provided in the SDP negotiation during the WebRTC signalling phase*.*

- pduSetImportance shall be set to the desired PSI value for N6-unmarked PDUs on the application flow in question which uses the application protocol indicated by unmarkedProtocol. The setting shall follow the semantics defined for PSI in clause 4.2.4 of TS 26.522 [37], with a value in the range of 1 to 15 (inclusive).In all PDUs it contributes at reference point RTC‑4m or RTC‑12 that fall within the scope of the application flow description, an RTC endpoint sender as the RTC Access Function (Media Access Function) or the Media Function of the RTC AS shall use the protocol indicated in transportProto; the sender shall set the SRTP header fields in accordance with rtpPayloadInfoList; and the sender 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.

### 10.3.3 Enabling dynamically changing traffic characteristics marking in dynamic policies

#### 10.3.3.1 Dynamically changing traffic characteristics marking for data bursts

If any dynamically changing traffic characteristics marking for data bursts is required by the selected Policy Template, as specified in clause 5.3.3.2 of TS 26.510 [3] (i.e., downlinkData‌Burst‌Size‌Marking‌Required is present and set to true, and/or downlinkTime‌To‌Next‌Burst‌Marking‌Required is present and set to true in the policy binding of the Service Access Information), the Dynamic Policy invoker, i.e., the RTC Media Session Handler or the RTC AS, 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 type (see clause 5.5.4.14 of TS 29.571 [36]) as either a rtpHeaderExtInfo object or as an element of the addRtpHeaderExtInfo object (see clause 5.5.4.13 of TS 29.571 [36]) shall be populated as follows:

- rtpHeaderExtType shall be set to DYN\_CHANGING\_TRAFFIC\_CHAR.

- rtpHeaderExtId shall be set to the value of the *ID* field to be used by the Media Function of an RTC AS in the *RTP Header Extension for Dynamically Changing Traffic Characteristics Marking* on the application flow in question, as specified in clause 4.5 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 Dynamically Changing Traffic Characteristics Marking*, as specified in clause 4.5.1 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.

If any dynamically changing traffic characteristics marking for data bursts is required by the selected Policy Template (see clause 5.2.7.1 of TS 26.510 [3]), in all PDUs it contributes for media delivery at reference point RTC-4m that fall within the scope of the application flow description, the Media Function of the RTC AS shall use the protocol indicated in transportProto and in addition shall behave as follows:

- If data burst size marking is required (i.e., downlink‌Data‌Burst‌Size‌Marking‌Required is present and set to true in the Policy Template instantiated by the Dynamic Policy Instance), the Media Function of the RTC AS shall include in at least one SRTP header of each downlink data burst it transmits a one- or two-byte (consistent with the signalled length) *RTP Header Extension for Dynamically Changing Traffic Chacteristics Marking* with fields set according to the values declared in the rtpHeaderExtInfo property per above and a data burst size indication, BSize, per clause 4.5.4 of TS 26.522 [37].

- If time to next burst marking is required (i.e., downlink‌Time‌To‌Next‌Burst‌Marking‌Required is present and set to *t*rue in the Policy Template instantiated by the Dynamic Policy Instance), the Media Function of the RTC AS shall include in at least one SRTP header of each downlink data burst it transmits a one- or two-byte (consistent with the signalled length) *RTP Header Extension for Dynamically Changing Traffic Chacteristics Marking* with fields set according to the values declared in the rtpHeaderExtInfo property per above and a time to next burst indication, TTNB, per clause 4.5.4 of TS 26.522 [37].

NOTE 1: The frequency and occurrence of *RTP Header Extension for Dynamically Changing Traffic Characteristics* relative to associated dynamically changing traffic characteristics is left to sender implementation. For more details, see guidelines provided in clause 4.5 of TS 26.522 [37].

NOTE 2: Procedures to configure the required RTC AS behaviour via reference point RTC‑3 are not defined in this version of the present document

#### 10.3.3.2 Dynamically changing traffic characteristics marking for expedited data transfers

If dynamically changing traffic characteristics marking for expedited data transfers is required by the selected Policy Template, as specified in clause 5.3.3.2 of TS 26.510 [3] (i.e., downlink‌Expedited‌Transfer‌Indication‌Marking‌Required is present set to *true* in the policy binding of the Service Access Information), 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 type (see clause 5.5.4.14 of TS 29.571 [36]) as either a rtpHeaderExtInfo object or as an element of the addRtpHeaderExtInfo object (see clause 5.5.4.13 of TS 29.571 [36]) shall be populated as follows:

- rtpHeaderExtType shall be set to *DYN\_CHANGING\_TRAFFIC\_CHAR\_ETI*.

Editor’s Note: The addition of a new RtpHeaderExtType enumeration value (e.g., *DYN\_*‌*CHANGING\_*‌*TRAFFIC\_*‌*CHAR\_*‌*ETI)* as part of the Protocol Description corresponding to the *RTP Header Extension for Expedited Transfer Indication Marking* is up to CT4 TS 29.571 specification.

- rtpHeaderExtId shall be set to the value of the *ID* field to be used by the Media Function of an RTC AS in the *RTP Header Extension for Expedited Transfer Indication Marking* on the application flow in question, as specified in clause 4.7 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 Expedited Transfer Indication Marking*, as specified in clause 4.7.1 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.

If dynamically changing traffic characteristics marking for expedited data transfers is required by the selected Policy Template (see clause 5.2.7.1 of TS 26.510 [3]), in all PDUs it contributes for media delivery at reference point RTC-4m that fall within the scope of the application flow description, the Media Function of the RTC AS shall use the protocol indicated in transportProto and shall include in all SRTP headers of downlink packets a one- or two-byte (consistent with the signalled length) *RTP Header Extension for Dynamically Changing Traffic Chacteristics Marking* with fields set according to the values declared in the rtpHeaderExtInfo property per above.

NOTE 1: PDUs contributed within the scope of the application flow description at RTC-4m that cannot be marked (e.g., RTCP, STUN, see clause 4.7.6 of TS 26.522 [37]) are not expedited and can be handled by the 5G System on a default QoS flow depending on the User Plane Function configuration, see TS 29.244 [x1].

NOTE 2: Procedures to configure the required RTC AS behaviour via reference point RTC‑3 are not defined in this version of the present document.

### 10.3.4 Enabling multiplexed media flow handling in dynamic policies

If an RTC Session uses multiple media flows multiplexed into a single RTP Session as described in section 4.4 of RFC 8834 [RFC8834] (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 (i.e., the RTC Media Session Handler or the RTC AS) shall additionally populate the multiplexedMediaInfos property of the Application‌Flow‌Description object (see clause 7.3.3.2 of TS 26.510 [3]) 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 (e.g., 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 of TS 29.514 [29514]) shall be populated to aid traffic identification of each corresponding media stream in the 5G System, 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, as follows:

- 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(s) 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.

- identificationTag 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]*.*

- rtpSdesHdrExtId shall be present when the sending RTC endpoint includes the *SDES RTP Header Extension for MID* in SRTP packets, and it 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].

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

NOTE 1: A combination of SSRC, Payload Type and/or MID values is 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 MID fields set according to the values declared in the MpxMediaInfoobject per above to indicate the multiplexed media identification information.

NOTE 2: 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 [29244], 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.

Next change

# 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/TSG109-Rel19

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.

End of changes