**3GPP TSG-SA4 Meeting #133-e *S4-251356***

**Online, , 18th Jul 2025 - 25th Jul 2025**

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| *CR-Form-v12.3* |
| **CHANGE REQUEST** |
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|  | **26.522** | **CR** | **0023** | **rev** | **-** | **Current version:** | **19.1.0** |  |
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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 |  |

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| ***Title:***  | Guidelines for RTP retransmission in multiplexed transmission scenarios |
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| ***Source to WG:*** | Huawei, HiSilicon, Nokia, and InterDigital Communications |
| ***Source to TSG:*** | S4 |
|  |  |
| ***Work item code:*** | 5G\_RTP\_Ph2 |  | ***Date:*** | 2025-07-15 |
|  |  |  |  |  |
| ***Category:*** | **B** |  | ***Release:*** | Rel-19 |
|  | *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)* |
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| ***Reason for change:*** | An RTP sender can multiplex retransmission streams with the source stream within the same RTP session (SSRC multiplexing and/or session- multiplexing with FID). According to the LS replies from SA2 [S4-251423] and RAN2 [S4-251404], no specific network treatment is necessary for retransmitted packets and the same QoS handling mechanisms apply. When these streams are mapped to different QoS flows, they might benefit from differentiated QoS handling, even if they are not using the RTP header extension for PDU Set marking. Currently, multiplexing scenarios in TS 26.522 do not address the case of multiplexed retranmission streams. |
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| ***Summary of change:*** | Add guidelines for multiplexing retransmitted packets within using SSRC mapping and FID based session-multiplexing mapping based on RFC 4588.  |
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| ***Consequences if not approved:*** | No guidelines provided for usage of RTP retransmission in the context of multiplexed streams.  |
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| ***Clauses affected:*** | 2, 4.2.6.4 and 4.6 |
|  |  |
|  | **Y** | **N** |  |  |
| ***Other specs*** |  | **x** |  Other core specifications  | TS/TR ... CR ...  |
| ***affected:*** |  | **x** |  Test specifications | TS/TR ... CR ...  |
| ***(show related CRs)*** |  | **x** |  O&M Specifications | TS/TR ... CR ...  |
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| ***Other comments:*** |  |
|  |  |
| ***This CR's revision history:*** | * Accepted update on mid usage
* Added sentence in 4.2.6.4 that multiplexed streams can benefit from differentiated QoS treatment.
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| \*\* CHANGE 1 \*\* |

# 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] ITU-T Rec H.264 (08/2021): "Advanced video coding for generic audiovisual services" | ISO/IEC 14496-10:2022: "Information technology – Coding of audio-visual objects – Part 10: Advanced Video Coding".

[3] ITU-T Rec H.265 (08/2021): "High efficiency video coding" | ISO/IEC 23008-2:2023: "High Efficiency Coding and Media Delivery in Heterogeneous Environments – Part 2: High Efficiency Video Coding".

[4] IETF RFC 3550 (2003): "RTP: A Transport Protocol for Real-Time Applications", H. Schulzrinne, S. Casner, R. Frederick and V. Jacobson.

[5] IETF RFC 6184 (2011): "RTP Payload Format for H.264 Video", Y.-K. Wang, R. Even, T. Kristensen, R. Jesup.

[6] IETF RFC 7798 (2016): "RTP Payload Format for High Efficiency Video Coding (HEVC)", Y.-K. Wang, Y. Sanchez, T. Schierl, S. Wenger, M. M. Hannuksela.

[7] 3GPP TR 26.928: "Extended Reality (XR) in 5G".

[8] 3GPP TR 26.998: "Support of 5G glass-type Augmented Reality / Mixed Reality (AR/MR) devices".

[9] IETF RFC 768 (1980): "User Datagram Protocol", J. Postel.

[10] IETF RFC 5761 (2010): "Multiplexing RTP Data and Control Packets on a Single Port", C. Perkins, M. Westerlund.

[11] IETF RFC 8285 (2017): "A General Mechanism for RTP Header Extensions", D. Singer, H. Desineni, R. Even.

[12] 3GPP TS 23.501: "System architecture for the 5G System (5GS)".

[13] IETF RFC 5905 (2010): "Network Time Protocol Version 4: Protocol and Algorithms Specification”, D. Mills, J. Martin, J. Burbank, W. Kasch.

[14] IEEE 1588-2019 – IEEE Standard for a Precision Clock Synchronization Protocol for Networked Measurement and Control Systems, June 2020.

[15] IETF RFC 4574 (2006): "The Session Description Protocol (SDP) Label Attribute", O. Levin, G. Camarillo.

[16] IETF RFC 3611 (2003): "RTP Control Protocol Extended Reports (RTCP XR)", T. Friedman, R. Caceres, A. Clark.

[17] 3GPP TS 26.119: "Media Capabilities for Augmented Reality".

[18] IETF RFC 7656 (2015): "A Taxonomy of Semantics and Mechanisms for Real-Time Transport Protocol (RTP) Sources ", J. Lennox, K. Gross, S. Nandakumar, G. Salgueiro, B. Burman.

[19] IETF RFC 5888 “The Session Description Protocol (SDP) Grouping Framework”, G. Camarillo et al.

[20] ISO/IEC 60559:2020: “Floating-point arithmetic”.

[21] 3GPP TR 26.114: "IP Multimedia Subsystem (IMS); Multimedia Telephony; Media handling and interaction".

[22] 3GPP TS 38.415 "NG-RAN; PDU Session User Plane Protocol".

[23] IETF RFC 7941 "RTP Header Extension for the RTP Control Protocol (RTCP) Source Description Items".

[24] IETF RFC 9143 "Negotiating Media Multiplexing Using the Session Description Protocol (SDP)".

[25] IETF RFC 4588: "RTP Retransmission Payload Format".

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| \*\* CHANGE 2 \*\* |

#### 4.2.6.4 Guidelines for multiplexed content

An RTP sender could also include RTP HE for PDU Set marking in case of multiplexed streams.

One possibility is RTP multiplexing when different RTP Streams exist (e.g. audio + video).

Another possibility is RTP multiplexing when different RTP Streams and RTCP packets are present.

Another common use case is to carry source and retransmitted streams using session-multiplexing or SSRC-multiplexing. In SSRC-multiplexing source and retransmission streams are transported in the same RTP session with a different SSRC see [25]. In session-multiplexing, source stream and retransmission streams are transported in two different streams and are grouped using the Flow Identification (FID) grouping mechanism using the MID values as described in RFC 4588 [25].

Another possibility is a multiplex in which RTP packets may contain different media types and in addition RTCP packets can be present (e.g. MPEG-2 TS over RTP and using RTCP).

Also cases may exist with multiple video streams.

To illustrate this, Table 4.2.6.4-1 provides some examples on different multiplexing scenarios and the corresponding guidelines for setting RTP HE are further given in Table 4.2.6.4-2.

The description of each scenario is given and the implication for RTP HE marking in the Tables.

NOTE: SSRC Multiplexed stream can benefit from differentiated QoS treatment in the 5G Core. This is optional, more details are provided in clause 4.6.

Table 4.2.6.4-1: Example of Multiplexing scenarios

|  |  |  |  |
| --- | --- | --- | --- |
| Scenario | Multiplex Type | Description | Implications for RTP HE for PDU Set Marking for sender |
| sc1 | audio + video RTP multiplex | Native audio and video streams are carried in separate RTP streams with different SSRC, and different PT Packets contain either audio or video. | Typically, RTP HE is used for the video stream, audio packets can be unmarked. If both audio and video RTP packets are marked, the RTP HE for PDU Set marking is usually applied to video and audio RTP streams separately. RTP video packets and audio packets are usually marked as separate PDU Sets, not as part of the same PDU Set. |
| sc2 | audio + video , RTCP  | Same as sc1, but in this case also RTCP packets exist. Packets contain audio, video or RTCP. | Same as sc1 for audio and video with the following addition. RTP HE cannot be used for RTCP packets, and these are handled as unmarked PDUs. (End of Data Burst signal cannot be used in case RTCP packet is the last one in a data burst). |
| sc3 | audio, video native multiplex  | Stream packets can contain both audio and video. In addition, packets can also contain other metadata related to the streams. | In this case, PDU Sets can contain different media types (e.g. MPEG-2 TS over RTP); additional guidance is provided to handle this case in Table 4.2.6.X-2.(In MPEG-2 TS over RTP packets usually 6-7 188 byte transport stream packets are carried in an RTP packet that can contain different media in each TS packet to be identified based on the packet identifier) |
| sc4 | audio, video native multiplex + RTCP  | same as sc3 adding RTCP | Same as sc3 including RTCP packets [4] that cannot carry RTP HE and are therefore left unmarked. |
| sc5 | video + video or audio + audio  | Similar to sc1, but multiple native audio or multiple native video streams are carried in separate RTP streams with different SSRC, either with different PT field or sharing same PT field. Packets contain content from a single SSRC. | Packets from different RTP streams are marked as separate PDU Sets, ensuring that each PDU Set contains packets only from one RTP stream (single SSRC). |
| sc6 | video + video or audio + audio + RTCP | Same as sc5 adding RTCP | Same as sc5 including RTCP packets [4] that cannot carry RTP Header Extension |
| sc7 | Retransmission stream | Can apply to any of sc1–sc6 above with multiplexing of one or more source and retransmission streams using SSRC-multiplexing or session-multiplexing  |  re-transmitted packets need not be marked with RTP Header Extension. Packets in retransmission streams may optionally benefit from differentiated QoS handling (see clause 4.6), |

Table 4.2.6.4-2: Guidelines for applying RTP HE in different example multiplexing scenarios

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| --- | --- | --- |
| Scenario | Guideline | Additional Comments |
| sc1 | Video PDU Sets should be assigned e.g. for video frames/slices and PSI may be set using the guidelines from clause 4.6.2.Audio packets may be unmarked, or in case audio frames consist of multiple packets they may also be marked using RTP HE. PSI of the unmarked packet is determined by the 5G System, based on a configuration, and this can also be based on the payload type. | Typically, RTP HE is used for the video stream, audio packets can be unmarked as frames are often a single packet and the marking is not beneficial. In such case only overhead is introduced (see the unmarked PDU case), or the RTP HE can also be used for the audio stream. |
| sc2 | Same as sc1. RTCP packets can not be marked using RTP HE (there is no RTP HE for RTCP) and are treated as unmarked packet in the 5G System. PSI should be determined by the 5G System. | Same as sc1 for audio and video. End of Data burst signal may not be valid if RTCP is the last packet in a burst as no RTP HE can be added to an RTCP packet.  |
| sc3 | PDU Sets may be identified by the RTP sender based on the presentation time and the RTP HE can be used to support the PDU Set based QoS handling. PSI may be set to a preconfigured value or the value corresponding to the importance of the most important part of the multiplexed stream using the guidelines from clause 4.6.2. | In this case, the grouping of PDU sets will contain different media types, and therefore the guidance cannot only be based on one specific media type. Therefore, PDU Sets could be identified and marked by the RTP sender based on other aspects such as the presentation time. The PSI can be set based on a configuration. |
| sc4 | Same as sc3RTCP packets can not be marked and are treated as unmarked packet in the 5G System. | Same as sc3 including RTCP packets that cannot carry the RTP Header Extension.Data burst signal cannot be used if RTCP is the last packet in a burst. |
| sc5 | Video PDU Sets may be assigned e.g. for video frames or slices and PSI may be set using the guidelines from clause 4.6.2 for video, separating RTP streams into separate PDU Sets.Audio Packets can be unmarked or in case audio frames consist of multiple packets they may be marked using RTP HE, separating RTP streams into separate PDU Sets. | Multiple PDU Sets can be "open" at the same time, i.e., some PDUs are received from multiple different SSRC (i.e. different RTP streams), which requires the marking to keep track of multiple simultaneous PDU Set contexts. |
| sc6 | Same as sc5.RTCP are treated as unmarked packet in the 5G System. PDU Set importance can be determined by the 5G system. | Same as sc5 including RTCP packets [4] that cannot carry the RTP Header Extension and need not be marked. |
| sc7 | SSRC-multiplexing or FID based session-multiplexing is used for source and retransmission streams. | Packets in retransmission streams may optionally benefit from differentiated QoS handling (see clause 4.6), even if they are not marked using the RTP HE for PDU Set marking. |

To support multiplexed content in combination with PDU Set QoS based handling in the 5G System, groups of packets of different media types (audio, video) but same payload type (native multiplex) can also be grouped as a PDU Set (sc3, sc4). This enables groups of packets to benefit from transfer using PDU Set QoS parameters in NG-RAN (PSDB, PSER, PSIHI). In this case, each of the RTP packets can set the RTP HE for PDU Set Marking to enable 5G System to identify corresponding PDU Sets.

To summarize, different options exist when applying RTP HE for PDU Set marking for multiplexed content, for which some guidelines are defined as follows:

- When RTP multiplexing (sc1, sc2, sc5, sc6 and sc7) is used, it is possible to separately mark the PDU Sets in different streams.

- When packets combine different media types in a payload type such as in sc3 and sc4, PDU sets can be created around a common media presentation time grouping packets based on timestamps. In this case the PDU set importance can be set to a derived or default value or a value configured.

- In case only packets of single stream are marked (e.g. the video stream), and the packets of other streams are unmarked. In this case the 5G System may still identify PDU set information as detailed in Annex A based on the payload information for example or the payload type.

- In case packets cannot carry an RTP HE (e.g. RTCP packet), packets can be handled as unmarked PDU and PDU Set information may still be derived in the 5G system in some cases.

- In case of multiplexing of source and retransmission streams, packets in the retransmission stream might not be marked using the RTP HE for PDU Set marking.

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\*\* CHANGE 3 \*\*

## 4.6 RTP SDES Header Extension for MID

When an RTP sender transmits different media streams in a multiplexed data flow identified by an IP 5-tuple, the 5GS network needs to identify the PDU’s belonging to the respective media streams, for enabling differentiated QoS handling (i.e. mapping multiplexed streams to different QoS Flows). The RTP SDES header extension for MID defined in RFC 9143 [23], described in Annex C.2, enables an RTP receiver to associate each RTP stream with a specific identification-tag.

An RTP sender may use the BUNDLE attribute defined in RFC 9143 [23] in SDP negotiation to multiplex the media streams, particularly in case SSRC is not available before the RTP session is started. Endpoints that support the bundle mechanism for multiplexed RTP streams shall include the RTP SDES HE for MID for identifying the media streams. Endpoints that support the RTP SDES HE for MID shall support both RTP HE formats (i.e., the one-byte and the two-byte formats). Endpoints that support the bundle mechanism for multiplexing RTP and RTCP streams shall include the RTCP MID SDES Item as defined in RFC 9143 [23] in RTCP SDES packets for identifying the media streams

NOTE: Not every RTP packet is required to send MID information in the RTP SDES HE for MID.

 Not every RTCP packet is required to include MID SDES Item in the RTCP SDES packets.

If the RTP SDES HE for MID is the only RTP HE used, the endpoints shall use the 1-byte header format. If other 2-byte RTP HE elements are used in the same RTP stream, then the 2-byte header shall be used, unless the "a=extmap-allow-mixed" is successfully negotiated through SDP offer/answer, as described by RFC 8285 [11].

Multiplexing can also be used to carry retransmitted packets in a separate retransmission stream within the same RTP session using a different SSRC (SSRC multiplexing, see [25]). In this case, the method described above can be used to enable differentiated QoS handling by allowing the 5G Core to map the multiplexed retransmission streams to different QoS Flows.