**3GPP SA4 #128 S4-241099**

**Jeju, Korea, 20 May-24 May 2024**

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| *CR-Form-v12.0* |
| **PSEUDO CHANGE REQUEST** |
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|  | **26.822** | **CR** | pseudo | **rev** | **-** | **Current version:** | **0.0.1** |  |
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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 |  | Radio Access Network |  | Core Network |  |

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| ***Title:***  | **[FS\_5G\_RTP\_Ph2] Candidate RTCP messages and RTP header extensions to support XR services in 5G** |
|  |  |
| ***Source to WG:*** | Qualcomm Incorporated, Lenovo |
| ***Source to TSG:*** |  |
|  |  |
| ***Work item code:*** | FS\_5G\_RTP\_Ph2 |  | ***Date:*** | 05/20/2024 |
|  |  |  |  |  |
| ***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-10 (Release 10)Rel-11 (Release 11)Rel-12 (Release 12)**Rel-13 (Release 13)Rel-14 (Release 14)Rel-15 (Release 15)Rel-16 (Release 16)* *Rel-17 (Release 17)* *Rel-18 (Release 18)* |
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| ***Reason for change:*** | One aspect of Key issue #7: RTCP messages to better support XR services in 5GWe introduce potential RTCP messages as well as RTP header extensions to better support XR services in 5G  |
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| ***Summary of change:*** | Documentation of RTCP messages and RTP header extensions supported in the current WebRTC implementation  |
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| ***Consequences if not approved:*** | Risk of not having a discussion of focus and of commercial relevance |
|  |  |
| ***Clauses affected:*** |  |
|  |  |
|  | **Y** | **N** |  |  |
| ***Other specs*** |  |  |  Other core specifications  | TS/TR ... CR ...  |
| ***affected:*** |  |  |  Test specifications | TS/TR ... CR ...  |
| ***(show related CRs)*** |  |  |  O&M Specifications | TS/TR ... CR ...  |
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| ***Other comments:*** |  |
|  |  |
| ***This CR's revision history:*** |  |

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#  Proposed changes

Add the following to the References clause:

\* \* \* \* 1st change \* \* \* \*

[WebRTC-code] WebRTC source code: [https://source.chromium.org/chromium/chromium/src/+/main:third\_party/webrtc](https://source.chromium.org/chromium/chromium/src/%2B/main%3Athird_party/webrtc), retrieved May 1, 2024.

[TWCC] RTP Extensions for Transport-wide Congestion Control, draft-holmer-rmcat-transport-wide-cc-extensions-01 <https://datatracker.ietf.org/doc/html/draft-holmer-rmcat-transport-wide-cc-extensions-01>

[NADA] RFC 8698, Network-Assisted Dynamic Adaptation: A Unified Congestion Control Scheme for Real-Time Media, 2020.

[SCReAMv2] Self-Clocked Rate Adaptation for Multimedia, draft-johansson-ccwg-rfc8298bis-screamv2-00, 2024.

\* \* \* \* End of 1st change \* \* \* \*

Add the following to clause 6:

\* \* \* \* 2nd change \* \* \* \*

## 6.x Solution #x: Candidate RTCP messages and RTP header extensions to support XR services in 5G

### 6.x.1 Key Issue mapping

This maps to Key Issue #7.

### 6.x.2 Description

### 6.x.2.1 RTCP messages

To understand the RTCP messages that may be used for supporting XR applications, we need to know what RTCP messages have been used in commercial systems. For this purpose, we look at the WebRTC implementation [WebRTC-code].

RTCP messages defined in RFCs:

* **Receiver report (RR)** (RFC 3550): The packet type (PT) is 201. It provides reception quality feedback to the other RTP endpoint on a per source SSRC basis. Among the reported information is
	+ the fraction lost: The fraction of RTP data packets from a source SSRC lost since the previous SR or RR packet was sent
	+ cumulative number of packets lost: The total number of RTP data packets from a source SSRC that have been lost since the beginning of reception.
	+ interarrival jitter: An estimate of the statistical variance of the RTP data packet interarrival time
* **Sender report (SR)** (RFC 3550): The packet type (PT) is 200. It is the same as the RR except that it carries additional 20 bytes of information about the RTP endpoint that originates this report.
* **Application-Defined Packet** (APP) (RFC 3550): The PT is 204. It is is intended for experimental use as new applications and new features are developed, without requiring packet type value registration.
* **Generic NACK** (RFC 4585): The PT is 205 and the FMT is 1. The Generic NACK is used to indicate RTP packet losses, identified by the means of a packet identifier and a bit mask.
* **Picture Loss Indication** (PLI) (RFC 4585): The PT is 206 and the FMT is 1. It indicates the loss of an undefined amount of coded video data belonging to one or more pictures.

The following transport-wide RTCP feedback message is defined in an informal IETF document [TWCC].

* **Transport-wide feedback** [TWCC]: The PT is 205 and the FMT is 15. This feeds back information about each packet received with a transport-wide packet sequence number.

     0                   1                   2                   3

     0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1

    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+

    |V=2|P|  FMT=15 |    PT=205     |           length              |

    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+

  0 |                     SSRC of packet sender                     |

    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+

  4 |                      SSRC of media source                     |

    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+

  8 |      base sequence number     |      packet status count      |

    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+

 12 |                 reference time                | fb pkt. count |

    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+

 16 |          packet chunk         |         packet chunk          |

    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+

    .                                                               .

    .                                                               .

    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+

    |         packet chunk          |  recv delta   |  recv delta   |

    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+

    .                                                               .

    .                                                               .

    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+

    |           recv delta          |  recv delta   | zero padding  |

    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+

* Base sequence number: The transport-wide sequence number of the first packet in this feedback.
* Reference time: it indicates an absolute reference time in some (unknown) time base chosen by the sender of the feedback packets. The first recv delta in this packet is relative to the reference time.
* Packet chunk: A list of packet status chunks, indicating the status of one or more packets starting with the one identified by base sequence number.
* Recv delta: it represents a time interval in units of 0.25ms for a packet indicated in the packet chunk relative to the reference time.

### 6.x.2.2 RTP header extensions

For applicaiton bitrate adaptation and congestion control, it is important for the network to understand the state of the network, i.e., whether the network is in congestion or not. Many congestion control algorithms, e.g., Google congestion control algorithms [WebRTC-code], NADA [NADA] and SCReAMv2 [SCReAMv2], use the queueing delay as a signal of network congestion. Therefore, it is important to measure the delays and make the measurements available to the RTP sender in an efficient manner.

TS26.522 [2] defined two RTP header extensions for in-band end-to-end delay measurement. The first RTP header extension that carries only one timestamp, also known as the “Absolute Sender Time" RTP header extension, is already implemented in WebRTC [WebRTC-code]. The second RTP header extension that carries three timestamps returns the measured one-way delay in the direction from the sender to the receiver back to the sender. The current implementation in WebRTC uses RTCP messages to carry the one-way delay back to the sender and that may introduce large delay due to the RTCP bandwidth limitation or large overhead due to the additional IP/UDP packet headers for a separate packet.

\* \* \* \* End of 2nd change \* \* \* \*