**3GPP TSG-S4 Meeting #117-e *S4-220059***

**Online, , 14th–23rd February 2022** revision of S4aI221304

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| *CR-Form-v12.0* |
| **PSEUDO CHANGE REQUEST** |
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|  |  | **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:***  |  |
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| ***Source to WG:*** |  |
| ***Source to TSG:*** |  |
|  |  |
| ***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). |  |
|  |  |
| ***Reason for change:*** | Document potentially useful technology. |
|  |  |
| ***Summary of change:*** | * Brief description of QRT and QUIC over RTP.
* Usage of QRT in the potential solution as an alternative to RIST Main Profile.
* Usage of RTCP-based Congest Control algorithms to drive dynamic bit rate adaptation.
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|  |  |
| ***Consequences if not approved:*** | A potentially useful technolgoy will not be documented in the Feasibility Study. |
| ***Q*** |  |
| ***Clauses affected:*** | 2, 3.3, 4.6 (new), 6.3.4.3, 6.6. |
|  |  |
|  | **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:*** | S4aI221292->S4aI221304->S4-220059 |

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".

(SNIPPED)

[51] IETF RFC 9000: "QUIC: A UDP-Based Multiplexed and Secure Transport", May 2021.

[52] IETF Internet-Draft draft-ietf-quic-datagram-08: "An Unreliable Datagram Extension to QUIC", 14th January 2022, <https://www.ietf.org/archive/id/draft-ietf-quic-datagram-08.html>.

[53] IETF Internet-Draft draft-gruessing-moq-requirements-00: "QUIC Encapsulation for Media over RTP – Requirements and Use Cases", October 2021, https://www.ietf.org/id/draft-gruessing-moq-requirements-00.html.

[54] IETF Internet-Draft draft-hurst-quic-rtp-tunnelling-01: "QRT: QUIC RTP Tunnelling", 28th January 2021, <https://datatracker.ietf.org/doc/html/draft-hurst-quic-rtp-tunnelling-01>.

[55] IETF Internet-Draft draft-engelbart-rtp-over-quic-01: "RTP over QUIC", 25th October 2021, https://www.ietf.org/archive/id/draft-engelbart-rtp-over-quic-01.html.

[56] IETF Internet-Draft draft-ietf-quic-http-34: "Hypertext Transfer Protocol Version 3 (HTTP/3)", February 2021, https://www.ietf.org/archive/id/draft-ietf-quic-http-34.html.

[57] IETF RFC 2543: "SIP: Session Initiation Protocol", March 1999.

[58] IETF Internet-Draft draft-dawkins-avtcore-sdp-rtp-quic-00: "SDP Offer/Answer for RTP using QUIC as Transport", 28th January 2022, <https://www.ietf.org/archive/id/draft-dawkins-avtcore-sdp-rtp-quic-00.html>.

[59] IETF RFC 4588: "RTP Retransmission Payload Format", July 2006.

[60] IETF RFC 8888: "RTP Control Protocol (RTCP) Feedback for Congestion Control", January 2021.

[61] IETF RFC 8698: "Network-Assisted Dynamic Adaptation (NADA): A Unified Congestion Control Scheme for Real-Time Media", February 2020.

[62] IETF RFC 8298: "Self-Clocked Rate Adaptation for Multimedia", December 2017.

[63] IETF Internet-Draft draft-ietf-rmcat-gcc-02: "A Google Congestion Control Algorithm for Real-Time Communication", 8th July 2016, https://datatracker.ietf.org/doc/html/draft-ietf-rmcat-gcc-02.

[64] IETF RFC 8382: "Shared Bottleneck Detection for Coupled Congestion Control for RTP Media", June 2018.

[65] IETF Internet-Draft draft-ietf-quic-multipath-00: "Multipath Extension for QUIC", February 2022, https://www.ietf.org/archive/id/draft-ietf-quic-multipath-00.html.

NEXT CHANGE

## 3.3 Abbreviations

For the purposes of the present document, the abbreviations given in 3GPP 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 3GPP TR 21.905 [1].

(SNIPPED)

PTZ Pan, Tilt, Zoom

QRT QUIC RTP Tunnelling

RIST Reliable Internet Stream Transport

(SNIPPED)

NEXT CHANGE

## 4.6 Evolving Internet media transport protocols

### 4.6.1 General

Clause 4.2 describes existing media transport protocols used in media production operations. Because of the constantly evolving nature of the Internet, there exist new media transport protocols that are either incomplete or not yet adopted in the media production space. These may include support for new features or payload formats that are not included in existing solutions.

Table 4.6.1-1 below expands the feature comparison in table 4.2.7‑1 with the media transport protocols discussed in the following clauses.

Table 4.6.1-1: Comparison of future media transport protocols

| Parameter | QRT |
| --- | --- |
| Intended use | Contribution over unreliable links (e.g., public Internet) |
| Proprietary/‌Opensource | Open standards |
| Based on protocol | RTP, QUIC |
| Interoperability | Experimental |
| Latency | Configurable |
| Error correction | FEC/ARQ |
| Security | Transport encryption |
| Authentication | TLS client certificate |
| Multicast | Not yet specified |
| Multiple links | Supported, using multipath extension |
| Codec | Codec-agnostic |

### 4.6.2 Tunnelling RTP media sessions over QUIC

RTP media sessions [44] can be carried over the QUIC transport protocol specified in RFC 9000 [51] using the unreliable datagram extension specified in [52]. A survey of some recent proposals to standardise this usage are found in [53], along with relevant use cases and requirements. Of particular interest in the present document, the QUIC RTP Tunnelling (QRT) [54] and RTP over QUIC [55] proposals both specify a means to multiplex RTP media over a QUIC transport connection, allowing for secure transmission of media flows over lossy IP networks (including the Internet) with tuneable latency and quality parameters.

QRT [54] and RTP over QUIC [55] specify a mutually interoperable lightweight multiplexing layer on top of the QUIC unreliable datagram extension [52], allowing multiple RTP sessions to be multiplexed together into a single encrypted packet flow.

In addition, reliable streams may be multiplexed into the same QUIC connection as the RTP media flows to exchange data using application protocols requiring reliability, such as HTTP/3 [56] and/or the Session Initiation Protocol (SIP) specified in RFC 2543 [57] used in combination with appropriate session announcements [58]. The former may be used, for example, to convey NMOS configuration and control messages as introduced in clause 4.5.2.

NOTE: The use of SIP or NMOS may be convenient for in-band call control in certain scenarios, such as remote production contribution links.

Using RTP as the basis for media transport, QRT and RTP over QUIC can leverage the substantial existing feature set of already-deployed RTP solutions, including:

- Support for any codec and packaging format that has an associated RTP payload format.

- Support for **Forward Erasure Correction (FEC)**, including SMPTE 2022-1 [23].

**- Automatic Repeat Query (ARQ)** requests by means of in-band RTCP packets using the bitmap-based RTP Retransmission payload format specified in RFC 4588 [59].

- RIST’s range-based NACK retransmission mechanism [7] (as described in clause 4.2.4) may additionally or alternatively be used in this context.

In this respect, QRT and RTP over QUIC offer a similar feature set to RIST Main Profile, as described in clause 4.2.4.

The following optional RTP-related features are additionally available in comparison with current RIST specifications:

**- Congestion Control (CC)** through the use of RTCP-signalled feedback mechanisms, such as that described in RFC 8888 [60], alongside congestion control algorithms such as Network-Assisted Dynamic Adaptation (NADA) [61], Self-Clocked Rate Adaptation for Multimedia (SCReAM) [62], Google Congestion Control (Google-CC) [63] or Shared Bottleneck Detection [64].

By using QUIC, which has a predominant use in underpinning HTTP/3, QRT also inherits the following features:

- It is well understood by many application firewalls and proxies.

- Because connections are identified by a pair of abstract connection identifiers (rather than by a traditional 5‑tuple) active QUIC connections can be migrated between network endpoints without performing the security connection handshake again and without interrupting application-level flows.

- By probing additional network links before performing a connection migration, minimal delay and/or interruption is incurred.

A draft multipath extension [65] recently adopted by the IETF QUIC Working Group allows a pair of QUIC endpoints to use multiple network paths simultaneously (i.e. "link bonding"), either to increase the aggregate capacity of a connection, or to improve the robustness/resilience of the connection, or a combination of these.

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#### 6.3.4.3 Solution Example B: Fine-grained separation with separated media

In this example, a finer-grained separation of media is used:

- Within Group 1, the audio elementary stream has a higher priority than the video elementary stream.

- Talkback (Group 2) audio has a lower priority than Group 1 traffic.

- In Group 3, tally light control has a higher priority than general camera control.

As result, the individual media flows should be separated into separate application flows, e.g. UDP/IP flows or TCP/IP flows.

In order to enable the 5G System to prioritise the audio elementary stream higher than the video elementary stream in Group 1, the elementary streams need to be carried as individual UDP/IP media flows.

- RIST Simple profile (see clause 4.2.4) allows usage of separated RTP sessions for different elementary streams, when a native RTP payload format (like RFC 7798 [47] for HEVC or RFC 6416 [49] for AAC) is used.

- RIST Main profile (see clause 4.2.4) uses GRE tunnelling to multiplex all media flows in order to simplify NAT/firewall traversal.

- QRT [54] and RTP over QUIC [55] (see clause 4.6.2) can leverage the multiplexing capabilities of QUIC. It can achieve the same multiplexing effect as RIST Main profile.

NOTE: The usage of a tunnelling protocol prevents the 5G System from differentiating individual media flows, and thus inhibits its ability to apply different network QoS to the flows multiplexed inside a tunnel.

The talkback audio flow needs to be separated from the main output using dedicated TCP/IP or UDP/IP transmission resources.

If tally light control requires a higher priority than other camera control messages, the event messages should be carried using uniquely identifiable network resources. When MQTT is used for carrying control event messages, the camera needs to set up two MQTT/TCP connections, which can then be clearly prioritized by the 5G System. When WebSockets are used for carrying the event message, the camera should set up two WebSocket/TCP connections to enable separate message prioritization.

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## 6.6 Key Issue #5: Dynamic bit rate adaptation

### 6.6.1 General

Dynamic bit rate adaptation describes the capability to adjust the encoding bit rate of a compressed stream during operation in order to handle short term network variations, by varying the quality of the encoded media stream. Those network variations can be caused e.g. by high load, interference or mobility events. There can be different triggers for rate adaptation, e.g. a control signal from the network or continuous monitoring the network performance (e.g. by estimating the available bandwidth). Such a capability may not be required for Tier 1 AV productions, since Tier 1 AV productions are typically well planned from a capacity and coverage perspective. Dynamic bit rate adaptation could, however, become an important tool for Tier 2 or Tier 3 production scenarios to improve the overall robustness of the system, e.g. to increase the usage flexibilty and simplify SLA negotiations and fulfillment.

This type of adaptive bit rate is not widely available for professional applications so adoption by the media production industry is needed.

- Solutions can describe different realizations (e.g. using the Temporary Maximum Media Bit Rate (TMMBR) RTCP transport layer feedback message defined in RFC 5104 [41] and section 6.2 of RFC 4585 [42], the RTCP-based congestion control feedback message defined in RFC 8888 [60], etc)

- Support can be an optional feature of a media protocol.

- Trade-off between packet loss, quality, etc (different parameters to fit into the bit rate budget) should be studied.

Editor’s Note: More input needed on acceptable performance, potential SLA requirements, bit rate boundaries, such as accaptable minimal bit rate, etc needed from media producer side.

NOTE: Dynamic bit rate adaptation is typically applied to video signals, but can also be applied to audio.

### 6.6.2 Dynamic bit rate adaptation solutions leveraging RTCP-based congestion control signalling

RTP-based media transport protocols such as RIST (see clause 4.2.4) as well as QRT and RTP over QUIC (see clause 4.6.2) can make use of RTCP-based congestion control mechanisms to apply rate control to RTP media flows [44] and to influence the bit rate of a media encoder in real time so that it operates within a capacity envelope dicatated by prevailing network conditions.

A congestion control feedback packet is defined in RFC 8888 [60] for this purpose and it may be combined with a number of different congestion control algorithms, including (but not limited to) Network-Assisted Dynamic Adaptation (NADA), defined in RFC 8698 [61], Self-Clocked Rate Adaptation for Multimedia (SCReAM), defined in RFC  [62], Google Congestion Control [63] and Shared Bottleneck Detection, defined in RFC 8382 [64].

END OF CHANGES