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RTP over QUIC
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Abstract

QUIC is a UDP-based protocol for congestion controlled reliable data transfer, while RTP serves carrying (conversational) real-time media over UDP. This draft discusses design aspects and issues of carrying RTP over QUIC.

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[1.](#) Introduction

The Real-time Transport Protocol (RTP) [[RFC3550](#)] is a protocol for carrying media with real-time properties. It is usually mapped to UDP, possibly with DTLS [[RFC5763](#)] [[RFC5764](#)] in-between, as UDP allows RTP full control over packet transmission timing and congestion control. A number of media-specific and media-independent error control mechanisms have been developed in the AVTCORE and AVTEXT WGs to cope with the unreliability of UDP (e.g., [[RFC4588](#)]), and several congestion control mechanisms are presently being explored in the RMCAT WG (e.g., [[I-D.ietf-rmcat-scream-cc](#)] [[I-D.ietf-rmcat-gcc](#)] [[I-D.ietf-rmcat-nada](#)] [[I-D.ietf-rmcat-coupled-cc](#)] [[I-D.singh-rmcat-adaptive-fec](#)]), in addition to the basic circuit breaker mechanism [[RFC8083](#)]). RTP could also run over TCP or DCCP, but experiments have shown that the operational range in terms of underlying network conditions is fairly limited [[Delay-TCP](#)].

How to use of RTP is usually agreed upon between two endpoints using a signaling channel (e.g., SIP [[RFC3261](#)]) or WebRTC [[I-D.ietf-rtcweb-overview](#)] [[I-D.ietf-rtcweb-rtp-usage](#)], both with the offer/answer exchange [[RFC3264](#)] using the Session Description Protocol (SDP) [[RFC4566](#)]. RTP can run on top connectionless as well

as connection-oriented transport protocols. The signaling channel is also exploited to support NAT traversal RTP using ICE [[RFC5245](#)].

The QUIC transport protocol [[I-D.ietf-quic-transport](#)] [[I-D.ietf-quic-tls](#)] [[I-D.ietf-quic-recovery](#)] [[I-D.ietf-quic-manageability](#)] [[I-D.ietf-quic-applicability](#)] [[I-D.ietf-quic-http](#)] is being developed as a secure, reliable, congestion controlled UDP-based transport protocol with web applications as the primary target. In particular, QUIC allows for low latency establishment of secure connections and supports extensive multiplexing of many independent streams within a single connection (over a single UDP port), making it attractive for bundling of multiple media streams currently specified in SDP using [[I-D.ietf-mmusic-sdp-bundle-negotiation](#)]

The document discusses the possible use of RTP over QUIC with three main purposes:

- o Understanding and defining a sensible mapping of RTP sessions onto one (or more) QUIC connections ([section 3](#));
- o Deriving a wishlist for QUIC functionality to be fed into the QUIC WG ([section 4](#)); and
- o Defining a profile of the QUIC protocol with the necessary signaling extensions to enable RTP over QUIC ([section 5](#)).

Editor's note: [Section 4](#) is intended to document requirements for now and may disappear later if those are met or formally folded into a separate document. Also sections [3](#) and [5](#) may ultimately become separate Internet drafts for considerations by different working groups (e.g., AVTCORE and MMUSIC).

2. RTP-to-Transport Interface

The Real-time Transport Protocol defines the notion of RTP sessions to describe an elementary communication relationship between two or more parties. An RTP session comprises a uni-, bi-, or multidirectional flow of RTP packets carrying media as well as flows of RTCP packets providing feed forward from RTP senders to receivers and feedback from RTP receivers to senders.

Each media source is identified by a 32-bit Synchronization Source (SSRC) identifier, unique within an RTP session. An RTP session comprise the set of media sources that have the same view of the SSRC space. A single endpoint may use multiple SSRC identifiers (e.g., one for audio and one for video). Multiple media streams of a single endpoint are tied together by means of a common Canonical Name (CNAME) carried as part of the RTCP Source Description (SDES)

packets. This allows receivers to, e.g., determine which media streams to synchronize.

Originally, in an RTP session the RTP and RTCP streams each used different port numbers, so that a single RTP session would use two port numbers (historically, when used with multicast conferencing, these were adjacent port numbers, RTP on the even and RTCP on the next higher odd port number). However, the use of unicast RTP has, (not just) due to the presence of NATs, motivated the multiplexing of both RTP and RTCP on a single port number [[RFC5761](#)]. The payload structure and number spaces used for RTP and RTCP packets were designed to support this easily.

The bundle framework [[I-D.ietf-mmusic-sdp-bundle-negotiation](#)] allows multiplexing of multiple RTP streams on a single address:port combination. All the RTP streams in a bundled group are part of a single RTP session sharing a single SSRC number space [[RFC3550](#)].

These two efforts also reduce the number of ICE candidates to be validated as part of a multimedia call or conference setup procedure. They are particularly required in conjunction with WebRTC to reduce the signaling and resource requirements, which would affect NATs as well as STUN and TURN servers. We note, however, that ICE is not currently usable with QUIC, since QUIC and STUN packets are not readily distinguished on a single UDP port, due to poor choice of packet formats.

WebRTC deserves particular consideration because its potential close relationship to QUIC: WebRTC uses HTTP/1.1 (possibly using WebSockets), or HTTP/2 to connect to web servers, and thus will likely use QUIC in the future as a signaling transport. Moreover, WebRTC supports peer-to-peer data channels, which currently target using SCTP over UDP over DTLS: SCTP for stream multiplexing within a connection and UDP for better NAT traversal properties. Since QUIC would seem to support these two functions, it could be a natural choice to be used for the data channel as well - although this would require changes to the QUIC packet formats to allow demultiplexing with STUN for NAT traversal.

For the actual media transmission, RTP use codec-specific payload formats that define how a piece of encoded media is broken down into data units that can fit into an MTU-sized packet for transmission. One important goal of RTP payload format design is allowing decoding packets as much as possible independent of each other as some may be lost due to the best-effort nature of the underlying UDP [[RFC2736](#)]. This implies, on the one hand, that RTP senders have to perform codec-level fragmentation in a semantically meaningful manner and, on the other hand, that are in control of packet boundaries and

transmission scheduling and timing as well as retransmission decisions.

On the receiving side, RTP expects a detailed understanding of packet reception timing, possible reordering, and losses, as this information is used for the feedback statistics.

3. RTP-to-QUIC Mapping

This section address the necessary considerations to realize one possible way of carrying RTP-over-QUIC.

Editor's note: At this point, this section is intended to explore the design space and briefly describe a number of different options without making specific recommendations about which option(s) to choose. Future revisions of this document move towards taking concrete decisions.

3.1. Mapping Semantic Units

RTP payload formats define a mapping of media data units (e.g., video or audio frames, audio samples, etc.) to packets. Assuming that we will preserve the structure of RTP header, optional header extension, and payload, there are two obvious options:

Preserve the previous RTP assumptions about semantic fragmentation at MTU size boundaries; i.e., use the same packetization mechanism as before, just then drop the resulting RTP packet into a QUIC payload. Note that the MTU size may be smaller since QUIC packet headers are larger than plain UDP headers.

Operate solely on semantic units such as video frames, and map each semantic unit to a QUIC payload. This approach leaves the final packetization decision to QUIC. In this case, our "MTU size" would not be defined by the IP layer but by QUIC. It is possible in this case for video frame composed of multiple RTP packets to use one RTP header for the whole video frame; no need to break the video frame to multiple RTP packet, put all payload as one RTP packet whose size may be bigger than MTU and send it as QUIC payload.

If we assume that semantic units are to be received and processed atomically for best performance results, then option 2) would be preferred. If we consider that subunits are meaningful (e.g., slices in case of video), then option 1) may be preferred. In any case, however, it would be up to the payload definition to determine what a semantic unit.

3.2. Encapsulating Media Units

QUIC streams do not preserve packet boundaries but rather offer the same abstraction as TCP does. Therefore, if multiple identifiable media units are to be transmitted on the the same stream, the encapsulation mechanisms **MUST** provide boundaries for media data units, e.g., similar to the approach chosen for carrying RTP in TCP.

The exception would be if only a single frame is ever transmitted across a single stream (see option 3 in [section 3.3](#)) so that stream termination signifies the end of the respective packet.

3.3. Mapping Media to Streams

There are three basic distinct options for mapping media to streams:

Map an RTP session to a QUIC stream. In this case, all media packets of the RTP session would be carried within a single QUIC stream.

Map an RTP stream to a QUIC stream. In case, as presently discussed in the QUIC WG, the QUIC stream would be unidirectional and we will have one QUIC stream per transmission direction.

Note that both options would map, e.g., FEC or retransmission sessions to different QUIC streams. Note also that both 1. and 2. implicitly create the problem of head-of-line blocking since QUIC streams are reliable and order preserving. This would thus not serve the real-time nature of RTP packets well.

Map each independently decodable groups of frames, video frame, or even packet, depending on the encapsulation chosen to an individual QUIC stream. This is independent of whether streams, would be uni- or bi-directional.

Option 3 eliminates the head of line blocking problem of options 1. and 2. because QUIC does not provide any ordering across different streams. Using larger semantic units (e.g., GOPs) for stream mapping, would provide for more efficient stream number usage. However, all stream frames are still transmitted reliably. This implies that QUIC will perform retransmissions even for packets that would be too late already.

Mapping each video frame or packet to a different stream would raise an issue with stream numbering unless all RTP sessions are multiplexed on a single UDP socket anyway and then all RTP packets would simply be mapped to different streams.

An open question here would be how to deal with additional data channels that don't use RTP. Ideally, it should be possible that those be within the same QUIC connection (if QUIC is used as transport) to avoid consuming again more port numbers. Since, on the one hand, data channels can be set up and torn down at any time and, on the other hand, media packets are transmitted continuously, a need arises to set aside streams for data channels. One option would be "reserving" those streams in some form. But then, how many to reserve? Moreover, this would be incompatible with the slides stream number window being used by QUIC. Alternatively, one would need to synchronize the use of QUIC streams in real-time between the signaling and application channels and the media packet transmission. This may be hard to achieve and also suffers from the problem of the stream id window moving fast with frame transmissions. A third option would be adding another demultiplexing structure (e.g., to different RTP headers from data packets) and use a similar scheme of one application data unit (ADU) per stream for other applications. While feasible, this appears somewhat cumbersome in the mapping.

We finally need to consider inter RTP stream synchronisation and how/ if this would be affected by use of multiple QUIC streams.

None of the above schemes appear truly satisfactory from a system design perspective. This may call for some refined design considerations for QUIC, which we will begin discussing in [section 4](#).

3.4. Mapping RTCP packets

RTCP is a bi-directional stream unlike RTP streams which are unidirectional. There can be for example a video stream receiver that only receives video content but will send and receive RTCP messages.

The current discussion on uni-directional streams direction will allow both uni- and bi- dirctional QUIC streams in the same QUIC connection. Such a solution will allow multiplexing of RTP and RTCP streams in the same QUIC connection.

An issue to consider is the encryption of RTCP messages. The RTP secure profiles SAVP [[RFC3711](#)] and SAVPF [[RFC5124](#)] allow NULL cipher for RTCP with message integrity. Using a NULL cipher allow RTP middleboxes to monitor the RTP delivery quality.

Whether to use a single stream for forward RTCP and another for reverse could be a function of the streams being uni- or bidirectional in the end. Another question to answer is if there should be one stream per SSRC per direction for RTCP. Finally, RTCP

packets may also be lost and they contain timing information. Avoiding HoL blocking may thus also be important.

3.5. Mapping of RTP header extensions

QUIC provides a reliable protocol which addresses the requirement in [[I-D.ietf-avtcore-rfc5285-bis](#)] to transmit the RTP header extension in a couple of RTP packets to provide better reliability. Still if we use mapping option 3 we will still need to transmit the RTP header extensions more than once. Using QUIC as a transport for RTP will have all RTP header extensions encrypted allowing only entities that terminate a QUIC connection to decode them. RTP header extension as defined in [[I-D.ietf-avtcore-rfc5285-bis](#)] can be sent in the clear and provide information to RTP middleboxes enabling them to route encrypted RTP packets. Currently the following header extensions are used for routing of encrypted RTP streams. Client to mixer audio level [[RFC6464](#)]. Frame marking [[I-D.ietf-avtext-framemarking](#)] and splicing interval [[I-D.ietf-avtext-splicing-notification](#)].

4. Design considerations for QUIC

This section will address design implications for QUIC and the interaction with QUIC of both RTP and RTCP. We expect to discuss the following aspects in the future:

- Reliability (or retransmission) control for stream frames
- Congestion control adaptation
- RTCP mapping
- Priming QUIC 0-RTT
- API
- Multiparty operation

5. SDP Extensions for Negotiating RTP-over-QUIC

TBD

6. Security Considerations

RTP is used as a plain payload for QUIC, exploiting its multiplexing capabilities. To this end, the RTP packets are protected (confidentiality) by the QUIC security mechanisms. Hence, the security considerations pertinent to QUIC apply.

QUIC is by its very nature a transport layer security mechanisms. RTP traffic will thus be protected on a single transport hop only. As soon RTP topologies more complex than a point-to-point connection are used (e.g., [[RFC7667](#)]), RTP traffic will lose its end-to-end

protection as transport connections are terminated at the intermediary, even if this acts just as a relay.

7. IANA Considerations

There are no IANA considerations at this point.

8. Acknowledgments

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