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QUIC Multiplexing
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Abstract

If QUIC is to be used in a peer-to-peer manner, with NAT traversal, then it is necessary to be able to demultiplex QUIC and other protocols used in WebRTC on a single UDP port. This memo discusses options for demultiplexing.

Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on April 23, 2019.

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1. Introduction

QUIC [I-D.ietf-quic-transport] is a new network transport protocol. While it is initially intended as a replacement for TCP in order to better support HTTP/2 [RFC7540] it should eventually be useful as a general purpose transport. HTTP is an asymmetric client-server protocol, but other uses of QUIC might operate in a peer-to-peer manner and so will need effective NAT traversal using ICE [RFC5245], which which makes use of STUN [RFC5389] and TURN [RFC5766] to discover NAT bindings. Therefore for QUIC to be utilized for peer-to-peer data transport, QUIC and STUN must be able to multiplex on the same port.

In a WebRTC scenario where RTP is used to transport audio and video and QUIC is used for data exchange, SRTP [RFC3711] is keyed using DTLS-SRTP [RFC5764] and therefore SRTP/SRTCP [RFC3550], STUN, TURN, DTLS [RFC6347] and QUIC will need to be multiplexed on the same port.

Within the W3C, a Javascript API for the use of QUIC for peer-to-peer data exchange [WEBRTC-QUIC] is under development within the ORTC

Community Group.

As noted in [RFC7983] Figure 3, protocol demultiplexing currently relies upon differentiation based on the first octet, as follows:

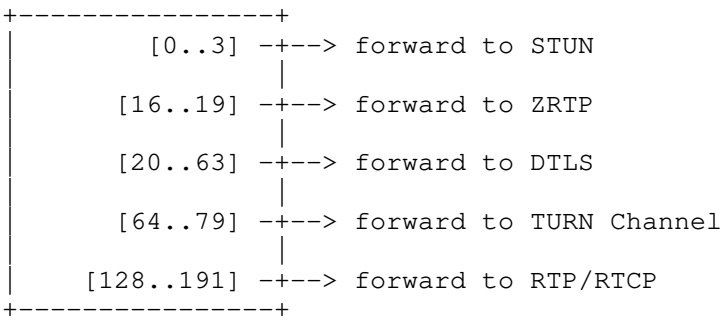


Figure 1: RFC 7983 packet demultiplexing algorithm.

As noted by Colin Perkins and Lars Eggert in [QUIC-Issue] this created a potential conflict with the design of the QUIC headers described in versions of [I-D.ietf-quic-transport] prior to -08.

1.1. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

2. Solution

At IETF 100, Colin Perkins presented a demultiplexing proposal [QUIC-MULTI]. The proposal which was subsequently proposed as a Pull Request to the QUIC Transport specification and merged in draft-ietf-quic-transport-08, involved renumbering of the QUIC long header packet type field as well as inverting the sense of the "C" bit in the short header packet.

The demultiplexing algorithm resulting from the changes appears as follows:

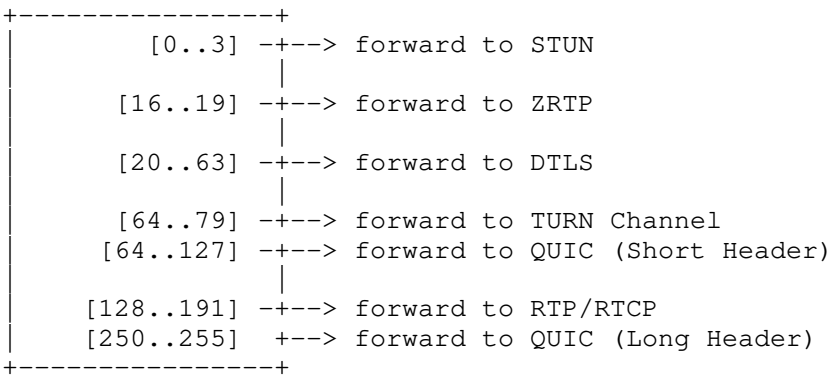
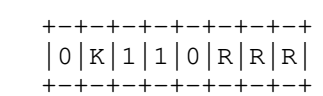


Figure 2: Revised packet demultiplexing algorithm.

Note that while the above diagram has a potential conflict between packets sent in TURN Channels and the QUIC short header, this conflict is not considered serious for WebRTC where TURN Channels are rarely used.

2.1. Subsequent changes

Since then, additional changes have been made to the QUIC transport headers. While the QUIC Long Header packet type field retains its original allocations between 0x7C and 0x7F, as of draft -15, the first octet of the Short Header now appears as follows:



Where:

K = indicates the key phase.
R = reserved bits, set randomly by endpoints not actively using them.

This potentially produces values of the first octet in the ranges 48-55 which potentially conflicts with DTLS, and 80-87 which conflicts with TURN channels (not an issue).

3. Security Considerations

The solutions discussed in this document could potentially introduce some additional security considerations beyond those detailed in

[RFC7983].

Due to the additional logic required, if mis-implemented, heuristics have the potential to mis-classify packets.

When QUIC is used for only for data exchange, the TLS-within-QUIC exchange [I-D.ietf-quic-tls] derives keys used solely to protect the QUIC data packets. If properly implemented, this should not affect the transport of SRTP nor the derivation of SRTP keys via DTLS-SRTP, but if badly implemented, both transport and key derivation could be adversely impacted.

4. IANA Considerations

This document does not require actions by IANA.

5. References

5.1. Informative References

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Acknowledgments

We would like to thank Martin Thomson, Roni Even and other participants in the IETF QUIC and AVTCORE working groups for their discussion of the QUIC multiplexing issue, and their input relating to potential solutions.

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RTP Control Protocol (RTCP) Feedback for Congestion Control
draft-ietf-avtcore-cc-feedback-message-02

Abstract

This document describes an RTCP feedback message intended to enable congestion control for interactive real-time traffic using RTP. The feedback message is designed for use with a sender-based congestion control algorithm, in which the receiver of an RTP flow sends RTCP feedback packets to the sender containing the information the sender needs to perform congestion control.

Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

Internet-Drafts are working documents of the Internet Engineering Task Force (IETF). Note that other groups may also distribute working documents as Internet-Drafts. The list of current Internet-Drafts is at http://datatracker.ietf.org/drafts/current/.

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1. Introduction

For interactive real-time traffic, such as video conferencing flows, the typical protocol choice is the Real-time Transport Protocol (RTP) running over the User Datagram Protocol (UDP). RTP does not provide any guarantee of Quality of Service (QoS), reliability, or timely delivery, and expects the underlying transport protocol to do so. UDP alone certainly does not meet that expectation. However, the RTP Control Protocol (RTCP) provides a mechanism by which the receiver of an RTP flow can periodically send transport and media quality metrics to the sender of that RTP flow. This information can be used by the sender to perform congestion control. In the absence of standardized messages for this purpose, designers of congestion control algorithms have developed proprietary RTCP messages that convey only those parameters needed for their respective designs. As a direct result, the different congestion control (i.e., rate adaptation) designs are not interoperable. To enable algorithm evolution as well as interoperability across designs (e.g., different rate adaptation algorithms), it is highly desirable to have generic congestion control feedback format.

To help achieve interoperability for unicast RTP congestion control, this memo proposes a common RTCP feedback packet format that can be used by NADA [I-D.ietf-rmcat-nada], SCReAM [RFC8298], Google

Congestion Control [I-D.ietf-rmcat-gcc] and Shared Bottleneck Detection [RFC8382], and hopefully also by future RTP congestion control algorithms.

2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

In addition the terminology defined in [RFC3550], [RFC3551], [RFC3611], [RFC4585], and [RFC5506] applies.

3. RTCP Feedback for Congestion Control

Based on an analysis of NADA [I-D.ietf-rmcat-nada], SCReAM [RFC8298], Google Congestion Control [I-D.ietf-rmcat-gcc] and Shared Bottleneck Detection [RFC8382], the following per-RTP packet congestion control feedback information has been determined to be necessary:

- RTP sequence number: The receiver of an RTP flow needs to feedback the sequence numbers of the received RTP packets to the sender, so the sender can determine which packets were received and which were lost. Packet loss is used as an indication of congestion by many congestion control algorithms.

- Packet Arrival Time: The receiver of an RTP flow needs to feedback the arrival time of each RTP packet to the sender. Packet delay and/or delay variation (jitter) is used as a congestion signal by some congestion control algorithms.

- Packet Explicit Congestion Notification (ECN) Marking: If ECN [RFC3168], [RFC6679] is used, it is necessary to feedback the 2-bit ECN mark in received RTP packets, indicating for each RTP packet whether it is marked not-ECT, ECT(0), ECT(1), or ECN-CE. If the path used by the RTP traffic is ECN capable the sender can use Congestion Experienced (ECN-CE) marking information as a congestion control signal.

Every RTP flow is identified by its Synchronization Source (SSRC) identifier. Accordingly, the RTCP feedback format needs to group its reports by SSRC, sending one report block per received SSRC.

As a practical matter, we note that host operating system (OS) process interruptions can occur at inopportune times. Accordingly, recording RTP packet send times at the sender, and the corresponding RTP packet arrival times at the receiver, needs to be done with deliberate care. This is because the time duration of host OS

interruptions can be significant relative to the precision desired in the one-way delay estimates. Specifically, the send time needs to be recorded at the last opportunity prior to transmitting the RTP packet at the sender, and the arrival time at the receiver needs to be recorded at the earliest available opportunity.

3.1. RTCP Congestion Control Feedback Report

Congestion control feedback can be sent as part of a regular scheduled RTCP report, or in an RTP/AVPF early feedback packet. If sent as early feedback, congestion control feedback MAY be sent in a non-compound RTCP packet [RFC5506] if the RTP/AVPF profile [RFC4585] or the RTP/SAVPF profile [RFC5124] is used.

Irrespective of how it is transported, the congestion control feedback is sent as a Transport Layer Feedback Message (RTCP packet type 205). The format of this RTCP packet is shown in Figure 1:

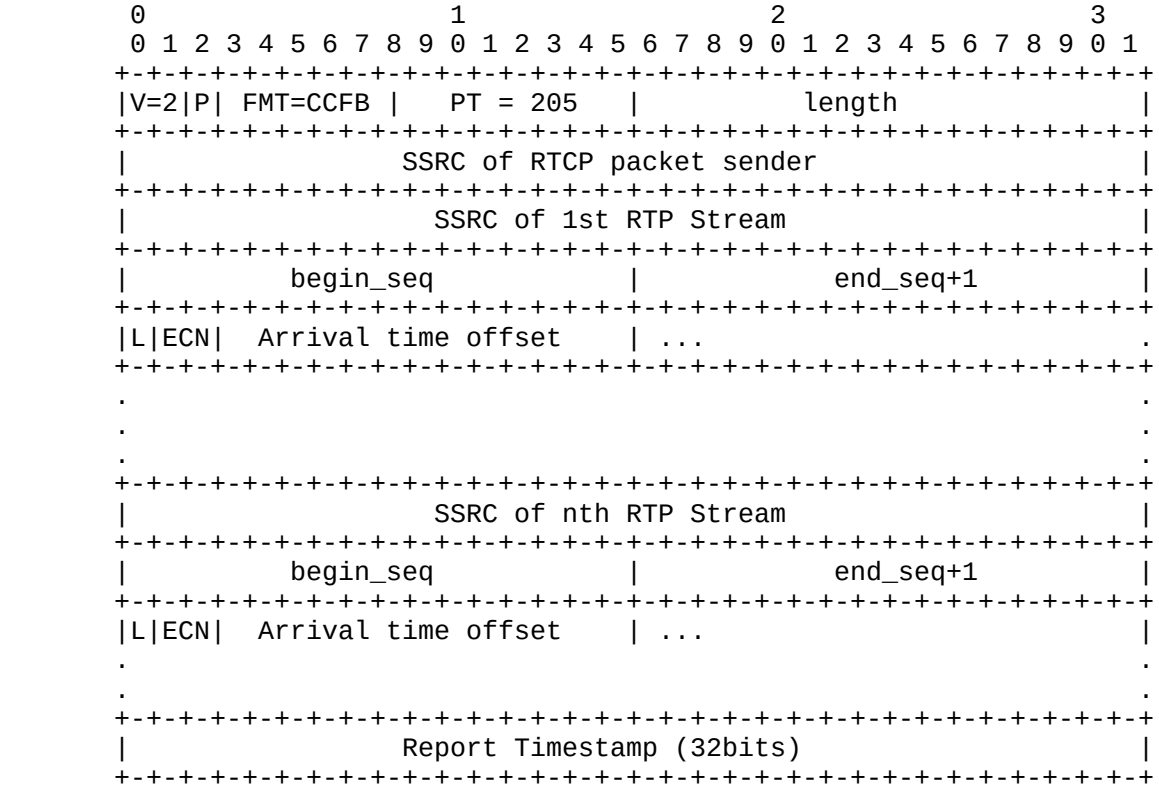

Figure 1: RTCP Congestion Control Feedback Packet Format

The first eight octets comprise a standard RTCP header, with PT=205 and FMT=CCFB indicating that this is a congestion control feedback packet, and with the SSRC set to that of the sender of the RTCP packet. (NOTE TO RFC EDITOR: please replace CCFB here and in the above diagram with the IANA assigned RTCP feedback packet type, and remove this note)

Section 6.1 of [RFC4585] requires the RTCP header to be followed by the SSRC of the RTP flow being reported upon. Accordingly, the RTCP header is followed by a report block for each SSRC from which RTP packets have been received, followed by a Report Timestamp.

Each report block begins with the SSRC of the received RTP Stream on which it is reporting. Following this, the report block contains a 16-bit packet metric block for each RTP packet with sequence number, seq, in the range begin_seq <= seq < end_seq+1 (calculated using arithmetic modulo 65535 to account for possible sequence number wrap-around). If the number of 16-bit packet metric blocks included in the report block is not a multiple of two, then 16 bits of zero padding MUST be added after the last packet metric block, to align the end of the packet metric blocks with the next 32 bit boundary. Each report block MUST NOT include more than 16384 packet metric blocks (i.e., it MUST NOT report on more than one quarter of the sequence number space in a single report).

The contents of each 16-bit packet metric block comprises the L, ECN, and ATO fields are as follows:

- L (1 bit): is a boolean to indicate if the packet was received. 0 represents that the packet was not yet received and all the subsequent bits (ECN and ATO) are also set to 0. 1 represent the packet was received and the subsequent bits in the block need to be parsed.

- ECN (2 bits): is the echoed ECN mark of the packet. These are set to 00 if not received, or if ECN is not used.

- Arrival time offset (ATO, 13 bits): is the arrival time of the RTP packet at the receiver. It is measured as an offset from the time at which the RTCP congestion control feedback report packet is sent. The arrival time offset is calculated by subtracting the reception time of the RTP packet denoted by this 16 bit packet metric block from the Report Timestamp (RTS) field of the RTCP congestion control feedback report packet in which the packet metric report block is contained. The arrival time offset is measured in units of 1/1024 seconds (this unit is chosen to give exact offsets from the RTS field). If the measured value is greater than 8189/1024 seconds (the value that would be coded as

0x1FFD), the value 0x1FFE MUST be reported to indicate an over-range positive measurement. If the measurement is unavailable, the value 0x1FFF MUST be reported.

The RTCP congestion control feedback report packet concludes with the Report Timestamp field (RTS, 32 bits). This represents the time instant when the report packet was generated. The value of RTS field is derived from the same wallclock used to generate the NTP timestamp field in RTCP Sender Report (SR) packets. It is formatted as the middle 32 bits of an NTP format timestamp, as described in Section 4 of [RFC3550].

RTCP congestion control feedback packets SHOULD include a report block for each SSRC that is being congestion controlled. The sequence number ranges reported on in consecutive reports for an SSRC SHOULD be consecutive and SHOULD NOT overlap (i.e., begin_seq for a report is expected to be one greater, modulo 65535, than end_seq of the previous report for that SSRC). If overlapping reports are sent, the information in the later report updates that in any previous reports for packets included in both reports (although note that such updated information will likely arrive too late to affect congestion control decisions at the sender). Reports that cover RTP sequence number ranges that are more than 16384 (i.e., one quarter of the sequence number space) ahead of the last end_seq received from an SSRC, or behind the last begin_seq received from an SSRC, modulo 65535 to account for wrap-around, SHOULD be ignored. An exception to this occurs if sender has sent RTP packets using more than one quarter of the sequence number space since it last received an RTCP congestion control feedback packet, then a report on recently sent RTP packets ought to be accepted, to allow recovery from report packet loss.

If no packets are received from an SSRC in a reporting interval, a report block MAY be sent with begin_seq and end_seq+1 both set to the highest sequence number previously received from that SSRC (or, the report can simply to omitted). The corresponding SR/RR packet will have a non-increased extended highest sequence number received field that will inform the sender that no packets have been received, but it can ease processing to have that information available in the congestion control feedback reports too.

4. Feedback Frequency and Overhead

There is a trade-off between speed and accuracy of reporting, and the overhead of the reports. [I-D.ietf-rmcat-rtp-cc-feedback] discusses this trade-off, suggests desirable RTCP feedback rates, and provides guidance on how to configure the RTCP bandwidth fraction, etc., to make appropriate use of the reporting block described in this memo.

Specifications for RTP congestion control algorithms can also provide guidance.

It is a general understanding that the congestion control algorithms will work better with more frequent feedback - per packet feedback. However, RTCP bandwidth and transmission rules put some upper limits on how frequently the RTCP feedback messages can be send from the RTP receiver to the RTP sender. It has been shown [I-D.ietf-rmcat-rtp-cc-feedback] that in most cases a per frame feedback is a reasonable assumption on how frequent the RTCP feedback messages can be transmitted. It has also been noted that even if a higher frequency of feedback is desired it is not viable if the feedback messages starts to compete against the RTP traffic on the feedback path during congestion period. Analyzing the feedback interval requirement [feedback-requirements] it can be seen that the candidate algorithms can perform with a feedback interval range of 50-200ms. A value within this range need to be negotiated at session setup.

5. SDP Signalling

A new "ack" feedback parameter, "ccfb", is defined to indicate the use of the RTP Congestion Control feedback packet format defined in Section 3. The ABNF definition of this SDP parameter extension is:

```
rtcp-fb-ack-param = <See Section 4.2 of [RFC4584]>
rtcp-fb-ack-param =/ ccfb-par
ccfb-par          = SP "ccfb"
```

The offer/answer rules for these SDP feedback parameters are specified in the RTP/AVPF profile [RFC4585].

6. Design Rationale

The primary function of RTCP SR/RR packets is to report statistics on the reception of RTP packets. The reception report blocks sent in these packets contain information about observed jitter, fractional packet loss, and cumulative packet loss. It was intended that this information could be used to support congestion control algorithms, but experience has shown that it is not sufficient for that purpose. An efficient congestion control algorithm requires more fine grained information on per packet reception quality than is provided by SR/RR packets to react effectively.

The Codec Control Messages for the RTP/AVPF profile [RFC5104] include a Temporary Maximum Media Bit Rate (TMMBR) message. This is used to convey a temporary maximum bit rate limitation from a receiver of RTP packets to their sender. Even though it was not designed to replace

congestion control, TMMBR has been used as a means to do receiver based congestion control where the session bandwidth is high enough to send frequent TMMBR messages, especially when used with non-compound RTCP packets [RFC5506]. This approach requires the receiver of the RTP packets to monitor their reception, determine the level of congestion, and recommend a maximum bit rate suitable for current available bandwidth on the path; it also assumes that the RTP sender can/will respect that bit rate. This is the opposite of the sender based congestion control approach suggested in this memo, so TMMBR cannot be used to convey the information needed for a sender based congestion control. TMMBR could, however, be viewed a complementary mechanism that can inform the sender of the receiver's current view of acceptable maximum bit rate.

A number of RTCP eXtended Report (XR) blocks have previously been defined to report details of packet loss, arrival times [RFC3611], delay [RFC6843], and ECN marking [RFC6679]. It is possible to combine several such XR blocks to report the detailed loss, arrival time, and ECN marking marking information needed for effective sender-based congestion control. However, the result has high overhead both in terms of bandwidth and complexity, due to the need to stack multiple reports.

Considering these issues, we believe it appropriate to design a new RTCP feedback mechanism to convey information for sender based congestion control algorithms. The new congestion control feedback RTCP packet described in Section 3 provides such a mechanism.

7. Acknowledgements

This document is an outcome of RMCAT design team discussion. We would like to thank all participants specially Sergio Mena, Xiaoquing Zhu, Stefan Holmer, David, Ingemar Johansson, Randell Jesup, Ingemar Johansson, and Magnus Westerlund for their valuable contribution to the discussions and to the document.

8. IANA Considerations

The IANA is requested to register one new RTP/AVPF Transport-Layer Feedback Message in the table for FMT values for RTPFB Payload Types [RFC4585] as defined in Section 3.1:

```
Name:      CCFB
Long name: RTP Congestion Control Feedback
Value:     (to be assigned by IANA)
Reference: (RFC number of this document, when published)
```

The IANA is also requested to register one new SDP "rtcp-fb" attribute "ack" parameter, "ccfb", in the SDP ("ack" and "nack" Attribute Values) registry:

```
Value name:  ccfb
Long name:   Congestion Control Feedback
Usable with: ack
Reference:   (RFC number of this document, when published)
```

9. Security Considerations

The security considerations of the RTP specification [RFC3550], the applicable RTP profile (e.g., [RFC3551], [RFC3711], or [RFC4585]), and the RTP congestion control algorithm that is in use (e.g., [I-D.ietf-rmcat-nada], [RFC8298], [I-D.ietf-rmcat-gcc], or [RFC8382]) apply.

A receiver that intentionally generates inaccurate RTCP congestion control feedback reports might be able trick the sender into sending at a greater rate than the path can support, thereby congesting the path. This will negatively impact the quality of experience of that receiver. Since RTP is an unreliable transport, a sender can intentionally leave a gap in the RTP sequence number space without causing harm, to check that the receiver is correctly reporting losses.

An on-path attacker that can modify RTCP congestion control feedback packets can change the reports to trick the sender into sending at either an excessively high or excessively low rate, leading to denial of service. The secure RTCP profile [RFC3711] can be used to authenticate RTCP packets to protect against this attack.

10. References

10.1. Normative References

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Sending Multiple Types of Media in a Single RTP Session
draft-ietf-avtcore-multi-media-rtp-session-13

Abstract

This document specifies how an RTP session can contain RTP Streams with media from multiple media types such as audio, video, and text. This has been restricted by the RTP Specification, and thus this document updates RFC 3550 and RFC 3551 to enable this behaviour for applications that satisfy the applicability for using multiple media types in a single RTP session.

Status of This Memo

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1. Introduction

The Real-time Transport Protocol [RFC3550] was designed to use separate RTP sessions to transport different types of media. This implies that different transport layer flows are used for different RTP streams. For example, a video conferencing application might send audio and video traffic RTP flows on separate UDP ports. With increased use of network address/port translation, firewalls, and other middleboxes it is, however, becoming difficult to establish multiple transport layer flows between endpoints. Hence, there is pressure to reduce the number of concurrent transport flows used by RTP applications.

This memo updates [RFC3550] and [RFC3551] to allow multiple media types to be sent in a single RTP session in certain cases, thereby reducing the number of transport layer flows that are needed. It makes no changes to RTP behaviour when using multiple RTP streams containing media of the same type (e.g., multiple audio streams or multiple video streams) in a single RTP session. However

[I-D.ietf-avtcore-rtp-multi-stream] provides important clarifications to RTP behaviour in that case.

This memo is structured as follows. Section 2 defines terminology. Section 3 further describes the background to, and motivation for, this memo and Section 4 describes the scenarios where this memo is applicable. Section 5 discusses issues arising from the base RTP and RTCP specification when using multiple types of media in a single RTP session, while Section 6 considers the impact of RTP extensions. We discuss signalling in Section 7. Finally, security considerations are discussed in Section 8.

2. Terminology

The terms Encoded Stream, Endpoint, Media Source, RTP Session, and RTP Stream are used as defined in [RFC7656]. We also define the following terms:

Media Type: The general type of media data used by a real-time application. The media type corresponds to the value used in the <media> field of an SDP m= line. The media types defined at the time of this writing are "audio", "video", "text", "image", "application", and "message". [RFC4566] [RFC6466]

Quality of Service (QoS): Network mechanisms that are intended to ensure that the packets within a flow or with a specific marking are transported with certain properties.

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

3. Background and Motivation

RTP was designed to support multimedia sessions, containing multiple types of media sent simultaneously, by using multiple transport layer flows. The existence of network address translators, firewalls, and other middleboxes complicates this, however, since a mechanism is needed to ensure that all the transport layer flows needed by the application can be established. This has three consequences:

1. increased delay to establish a complete session, since each of the transport layer flows needs to be negotiated and established;

2. increased state and resource consumption in the middleboxes that can lead to unexpected behaviour when middlebox resource limits are reached; and

3. increased risk that a subset of the transport layer flows will fail to be established, thus preventing the application from communicating.

Using fewer transport layer flows can hence be seen to reduce the risk of communication failure, and can lead to improved reliability and performance.

One of the benefits of using multiple transport layer flows is that it makes it easy to use network layer quality of service (QoS) mechanisms to give differentiated performance for different flows. However, we note that many RTP-using application don't use network QoS features, and don't expect or desire any separation in network treatment of their media packets, independent of whether they are audio, video or text. When an application has no such desire, it doesn't need to provide a transport flow structure that simplifies flow based QoS.

Given the above issues, it might seem appropriate for RTP-based applications to send all their RTP streams bundled into one RTP session, running over a single transport layer flow. However, this is prohibited by the RTP specification, because the design of RTP makes certain assumptions that can be incompatible with sending multiple media types in a single RTP session. Specifically, the RTP control protocol (RTCP) timing rules assume that all RTP media flows in a single RTP session have broadly similar RTCP reporting and feedback requirements, which can be problematic when different types of media are multiplexed together. Various RTP extensions also make assumptions about SSRC use and RTCP reporting that are incompatible with sending different media types in a single RTP session.

This memo updates [RFC3550] and [RFC3551] to allow RTP sessions to contain more than one media type in certain circumstances, and gives guidance on when it is safe to send multiple media types in a single RTP session.

4. Applicability

This specification has limited applicability, and anyone intending to use it needs to ensure that their application and use case meets the following criteria:

Equal treatment of media: The use of a single RTP session normally results in similar network treatment for all types of media used within the session. Applications that require significantly different network quality of service (QoS) or RTCP configuration for different RTP streams are better suited by sending those RTP streams in separate RTP session, using separate transport layer

flows for each, since that gives greater flexibility. Further guidance on how to provide differential treatment for some media is given in [I-D.ietf-avtcore-multiplex-guidelines] and [RFC7657].

Compatible RTCP Behaviour: The RTCP timing rules enforce a single RTCP reporting interval for all participants in an RTP session. Flows with very different media sending rate or RTCP feedback requirements cannot be multiplexed together, since this leads to either excessive or insufficient RTCP for some flows, depending on how the RTCP session bandwidth, and hence reporting interval, is configured. For example, it is likely infeasible to find a single RTCP configuration that simultaneously suits both a low-rate audio flow with no feedback, and a high-quality video flow with sophisticated RTCP-based feedback. Thus, combining these into a single RTP session is difficult and/or inadvisable.

Signalled Support: The extensions defined in this memo are not compatible with unmodified [RFC3550]-compatible endpoints. Their use requires signalling and mutual agreement by all participants within an RTP session. This requirement can be a problem for signalling solutions that can’t negotiate with all participants. For declarative signalling solutions, mandating that the session is using multiple media types in one RTP session can be a way of attempting to ensure that all participants in the RTP session follow the requirement. However, for signalling solutions that lack methods for enforcing that a receiver supports a specific feature, this can still cause issues.

Consistent support for multiparty RTP sessions: If it is desired to send multiple types of media in a multiparty RTP session, then all participants in that session need to support sending multiple type of media in a single RTP session. It is not possible, in the general case, to implement a gateway that can interconnect an endpoint using multiple types of media sent using separate RTP sessions, with one or more endpoints that send multiple types of media in a single RTP session.

One reason for this is that the same SSRC value can safely be used for different streams in multiple RTP sessions, but when collapsed to a single RTP session there is an SSRC collision. This would not be an issue, since SSRC collision detection will resolve the conflict, except that some RTP payload formats and extensions use matching SSRCs to identify related flows, and break when a single RTP session is used.

A middlebox that remaps SSRC values when combining multiple RTP sessions into one also needs to be aware of all possible RTCP packet types that might be used, so that it can remap the SSRC

values in those packets. This is impossible to do without restricting the set of RTCP packet types that can be used to those that are known by the middlebox. Such a middlebox might also have difficulty due to differences in configured RTCP bandwidth and other parameters between the RTP sessions.

Finally, the use of a middlebox that translates SSRC values can negatively impact the possibility for loop detection, as SSRC/CSRC can't be used to detect the loops; instead some other RTP stream or media source identity name space that is common across all interconnect parts is needed.

Ability to operate with limited payload type space: An RTP session has only a single 7-bit payload type space for all its payload type numbers. Some applications might find this space limiting when using different media types and RTP payload formats within a single RTP session.

Avoids incompatible Extensions: Some RTP and RTCP extensions rely on the existence of multiple RTP sessions and relate RTP streams between sessions. Others report on particular media types, and cannot be used with other media types. Applications that send multiple types of media into a single RTP session need to avoid such extensions.

5. Using Multiple Media Types in a Single RTP Session

This section defines what needs to be done or avoided to make an RTP session with multiple media types function without issues.

5.1. Allowing Multiple Media Types in an RTP Session

Section 5.2 of "RTP: A Transport Protocol for Real-Time Applications" [RFC3550] states:

> For example, in a teleconference composed of audio and video media encoded separately, each medium SHOULD be carried in a separate RTP session with its own destination transport address.
>
> Separate audio and video streams SHOULD NOT be carried in a single RTP session and demultiplexed based on the payload type or SSRC fields.

This specification changes both of these sentences. The first sentence is changed to:

> For example, in a teleconference composed of audio and video media encoded separately, each medium SHOULD be carried in a separate

> RTP session with its own destination transport address, unless specification [RFCXXXX] is followed and the application meets the applicability constraints.

The second sentence is changed to:

> Separate audio and video media sources SHOULD NOT be carried in a single RTP session, unless the guidelines specified in [RFCXXXX] are followed.

Second paragraph of Section 6 in RTP Profile for Audio and Video Conferences with Minimal Control [RFC3551] says:

> The payload types currently defined in this profile are assigned to exactly one of three categories or media types: audio only, video only and those combining audio and video. The media types are marked in Tables 4 and 5 as "A", "V" and "AV", respectively. Payload types of different media types SHALL NOT be interleaved or multiplexed within a single RTP session, but multiple RTP sessions MAY be used in parallel to send multiple media types. An RTP source MAY change payload types within the same media type during a session. See the section "Multiplexing RTP Sessions" of RFC 3550 for additional explanation.

This specification’s purpose is to override that existing SHALL NOT under certain conditions. Thus this sentence also has to be changed to allow for multiple media type’s payload types in the same session. The sentence containing "SHALL NOT" in the above paragraph is changed to:

> Payload types of different media types SHALL NOT be interleaved or multiplexed within a single RTP session unless [RFCXXXX] is used, and the application conforms to the applicability constraints. Multiple RTP sessions MAY be used in parallel to send multiple media types.

RFC-Editor Note: Please replace RFCXXXX with the RFC number of this specification when assigned.

5.2. Demultiplexing media types within an RTP session

When receiving packets from a transport layer flow, an endpoint will first separate the RTP and RTCP packets from the non-RTP packets, and pass them to the RTP/RTCP protocol handler. The RTP and RTCP packets are then demultiplexed based on their SSRC into the different RTP streams. For each RTP stream, incoming RTCP packets are processed, and the RTP payload type is used to select the appropriate media

decoder. This process remains the same irrespective of whether multiple media types are sent in a single RTP session or not.

As explained below, it is important to note that the RTP payload type is never used to distinguish RTP streams. The RTP packets are demultiplexed into RTP streams based on their SSRC, then the RTP payload type is used to select the correct media decoding pathway for each RTP stream.

5.3. Per-SSRC Media Type Restrictions

An SSRC in an RTP session can change between media formats of the same type, subject to certain restrictions [RFC7160], but MUST NOT change media type during its lifetime. For example, an SSRC can change between different audio formats, but cannot start sending audio then change to sending video. The lifetime of an SSRC ends when an RTCP BYE packet for that SSRC is sent, or when it ceases transmission for long enough that it times out for the other participants in the session.

The main motivation is that a given SSRC has its own RTP timestamp and sequence number spaces. The same way that you can’t send two encoded streams of audio with the same SSRC, you can’t send one encoded audio and one encoded video stream with the same SSRC. Each encoded stream when made into an RTP stream needs to have the sole control over the sequence number and timestamp space. If not, one would not be able to detect packet loss for that particular encoded stream. Nor can one easily determine which clock rate a particular SSRCs timestamp will increase with. For additional arguments why RTP payload type based multiplexing of multiple media sources doesn’t work, see [I-D.ietf-avtcore-multiplex-guidelines].

Within an RTP session where multiple media types have been configured for use, an SSRC can only send one type of media during its lifetime (i.e., it can switch between different audio codecs, since those are both the same type of media, but cannot switch between audio and video). Different SSRCs MUST be used for the different media sources, the same way multiple media sources of the same media type already have to do. The payload type will inform a receiver which media type the SSRC is being used for. Thus the payload type MUST be unique across all of the payload configurations independent of media type that is used in the RTP session.

5.4. RTCP Considerations

When sending multiple types of media that have different rates in a single RTP session, endpoints MUST follow the guidelines for handling RTCP described in Section 7 of [I-D.ietf-avtcore-rtp-multi-stream].

6. Extension Considerations

This section outlines known issues and incompatibilities with RTP and RTCP extensions when multiple media types are used in a single RTP sessions. Future extensions to RTP and RTCP need to consider, and document, any potential incompatibility.

6.1. RTP Retransmission Payload Format

The RTP Retransmission Payload Format [RFC4588] can operate in either SSRC-multiplexed mode or session-multiplex mode.

In SSRC-multiplexed mode, retransmitted RTP packets are sent in the same RTP session as the original packets, but use a different SSRC with the same RTCP SDES CNAME. If each endpoint sends only a single original RTP stream and a single retransmission RTP stream in the session, this is sufficient. If an endpoint sends multiple original and retransmission RTP streams, as would occur when sending multiple media types in a single RTP session, then each original RTP stream and the retransmission RTP stream have to be associated using heuristics. By having retransmission requests outstanding for only one SSRC not yet mapped, a receiver can determine the binding between original and retransmission RTP stream. Another alternative is the use of different RTP payload types, allowing the signalled "apt" (associated payload type) parameter of the RTP retransmission payload format to be used to associate retransmitted and original packets.

Session-multiplexed mode sends the retransmission RTP stream in a separate RTP session to the original RTP stream, but using the same SSRC for each, with association being done by matching SSRCs between the two sessions. This is unaffected by the use of multiple media types in a single RTP session, since each media type will be sent using a different SSRC in the original RTP session, and the same SSRCs can be used in the retransmission session, allowing the streams to be associated. This can be signalled using SDP with the BUNDLE [I-D.ietf-mmusic-sdp-bundle-negotiation] and FID grouping [RFC5888] extensions. These SDP extensions require each "m=" line to only be included in a single FID group, but the RTP retransmission payload format uses FID groups to indicate the m= lines that form an original and retransmission pair. Accordingly, when using the BUNDLE extension to allow multiple media types to be sent in a single RTP session, each original media source (m= line) that is retransmitted needs a corresponding m= line in the retransmission RTP session. In case there are multiple media lines for retransmission, these media lines will form an independent BUNDLE group from the BUNDLE group with the source streams.

An example SDP fragment showing the grouping structures is provided in Figure 1. This example is not legal SDP and only the most important attributes have been left in place. Note that this SDP is not an initial BUNDLE offer. As can be seen there are two bundle groups, one for the source RTP session and one for the retransmissions. Then each of the media sources are grouped with its retransmission flow using FID, resulting in three more groupings.

```
a=group:BUNDLE foo bar fiz
a=group:BUNDLE zoo kelp glo
a=group:FID foo zoo
a=group:FID bar kelp
a=group:FID fiz glo
m=audio 10000 RTP/AVP 0
a=mid:foo
a=rtpmap:0 PCMU/8000
m=video 10000 RTP/AVP 31
a=mid:bar
a=rtpmap:31 H261/90000
m=video 10000 RTP/AVP 31
a=mid:fiz
a=rtpmap:31 H261/90000
m=audio 40000 RTP/AVPF 99
a=rtpmap:99 rtx/90000
a=fmtp:99 apt=0;rtx-time=3000
a=mid:zoo
m=video 40000 RTP/AVPF 100
a=rtpmap:100 rtx/90000
a=fmtp:199 apt=31;rtx-time=3000
a=mid:kelp
m=video 40000 RTP/AVPF 100
a=rtpmap:100 rtx/90000
a=fmtp:199 apt=31;rtx-time=3000
a=mid:glo
```

Figure 1: SDP example of Session Multiplexed RTP Retransmission

6.2. RTP Payload Format for Generic FEC

The RTP Payload Format for Generic Forward Error Correction (FEC) [RFC5109] (and its predecessor [RFC2733]) can either send the FEC stream as a separate RTP stream, or it can send the FEC combined with the original RTP stream as a redundant encoding [RFC2198].

When sending FEC as a separate stream, the RTP Payload Format for generic FEC requires that FEC stream to be sent in a separate RTP session to the original stream, using the same SSRC, with the FEC stream being associated by matching the SSRC between sessions. The

RTP session used for the original streams can include multiple RTP streams, and those RTP streams can use multiple media types. The repair session only needs one RTP Payload type to indicate FEC data, irrespective of the number of FEC streams sent, since the SSRC is used to associate the FEC streams with the original streams. Hence, it is RECOMMENDED that the FEC stream use the "application/ulpfec" media type for [RFC5109], and the "application/parityfec" media type for [RFC2733]. It is legal, but NOT RECOMMENDED, to send FEC streams using media specific payload format names (e.g., using both the "audio/ulpfec" and "video/ulpfec" payload formats for a single RTP session containing both audio and video flows), since this unnecessarily uses up RTP payload type values, and adds no value for demultiplexing since there might be multiple streams of the same media type).

The combination of an original RTP session using multiple media types with an associated generic FEC session can be signalled using SDP with the BUNDLE extension [I-D.ietf-mmusic-sdp-bundle-negotiation]. In this case, the RTP session carrying the FEC streams will be its own BUNDLE group. The m= line for each original stream and the m= line for the corresponding FEC stream are grouped using the SDP grouping framework using either the FEC-FR [RFC5956] grouping or, for backwards compatibility, the FEC [RFC4756] grouping. This is similar to the situation that arises for RTP retransmission with session multiplexing discussed in Section 6.1.

The Source-Specific Media Attributes [RFC5576] specification defines an SDP extension (the "FEC" semantic of the "ssrc-group" attribute) to signal FEC relationships between multiple RTP streams within a single RTP session. This cannot be used with generic FEC, since the FEC repair packets need to have the same SSRC value as the source packets being protected. There was work on an Unequal Layer Protection (ULP) extension to allow it be use FEC RTP streams within the same RTP Session as the source stream [I-D.lennox-payload-ulp-ssrc-mux].

When the FEC is sent as a redundant encoding, the considerations in Section 6.3 apply.

6.3. RTP Payload Format for Redundant Audio

The RTP Payload Format for Redundant Audio [RFC2198] can be used to protect audio streams. It can also be used along with the generic FEC payload format to send original and repair data in the same RTP packets. Both are compatible with RTP sessions containing multiple media types.

This payload format requires each different redundant encoding use a different RTP payload type number. When used with generic FEC in sessions that contain multiple media types, this requires each media type to use a different payload type for the FEC stream. For example, if audio and text are sent in a single RTP session with generic ULP FEC sent as a redundant encoding for each, then payload types need to be assigned for FEC using the audio/ulpfec and text/ulpfec payload formats. If multiple original payload types are used in the session, different redundant payload types need to be allocated for each one. This has potential to rapidly exhaust the available RTP payload type numbers.

7. Signalling

Establishing a single RTP session using multiple media types requires signalling. This signalling has to:

1. ensure that any participant in the RTP session is aware that this is an RTP session with multiple media types;

2. ensure that the payload types in use in the RTP session are using unique values, with no overlap between the media types;

3. ensure RTP session level parameters, for example the RTCP RR and RS bandwidth modifiers, the RTP/AVPF trr-int parameter, transport protocol, RTCP extensions in use, and any security parameters, are consistent across the session; and

4. ensure that RTP and RTCP functions that can be bound to a particular media type are reused where possible, rather than configuring multiple code-points for the same thing.

When using SDP signalling, the BUNDLE extension [I-D.ietf-mmusic-sdp-bundle-negotiation] is used to signal RTP sessions containing multiple media types.

8. Security Considerations

RTP provides a range of strong security mechanisms that can be used to secure sessions [RFC7201], [RFC7202]. The majority of these are independent of the type of media sent in the RTP session; however it is important to check that the security mechanism chosen is compatible with all types of media sent within the session.

Sending multiple media types in a single RTP session will generally require that all use the same security mechanism, whereas media sent using different RTP sessions can be secured in different ways. When different media types have different security requirements, it might

be necessary to send them using separate RTP sessions to meet those different requirements. This can have significant costs in terms of resource usage, session set-up time, etc.

9. IANA Considerations

This memo makes no request of IANA.

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Guidelines for using the Multiplexing Features of RTP to Support Multiple Media Streams
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Abstract

The Real-time Transport Protocol (RTP) is a flexible protocol that can be used in a wide range of applications, networks, and system topologies. That flexibility makes for wide applicability, but can complicate the application design process. One particular design question that has received much attention is how to support multiple media streams in RTP. This memo discusses the available options and design trade-offs, and provides guidelines on how to use the multiplexing features of RTP to support multiple media streams.

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1. Introduction

The Real-time Transport Protocol (RTP) [RFC3550] is a commonly used protocol for real-time media transport. It is a protocol that provides great flexibility and can support a large set of different applications. RTP was from the beginning designed for multiple participants in a communication session. It supports many topology paradigms and usages, as defined in [RFC7667]. RTP has several multiplexing points designed for different purposes. These enable support of multiple RTP streams and switching between different encoding or packetization of the media. By using multiple RTP sessions, sets of RTP streams can be structured for efficient processing or identification. Thus, the question for any RTP application designer is how to best use the RTP session, the RTP stream identifier (SSRC), and the RTP payload type to meet the application's needs.

There have been increased interest in more advanced usage of RTP. For example, multiple RTP streams can be used when a single endpoint has multiple media sources (like multiple cameras or microphones) that need to be sent simultaneously. Consequently, questions are raised regarding the most appropriate RTP usage. The limitations in some implementations, RTP/RTCP extensions, and signalling has also been exposed. The authors also hope that clarification on the usefulness of some functionalities in RTP will result in more complete implementations in the future.

The purpose of this document is to provide clear information about the possibilities of RTP when it comes to multiplexing. The RTP application designer needs to understand the implications that come from a particular usage of the RTP multiplexing points. The document will recommend against some usages as being unsuitable, in general or for particular purposes.

The document starts with some definitions and then goes into the existing RTP functionalities around multiplexing. Both the desired behaviour and the implications of a particular behaviour depend on which topologies are used, which requires some consideration. This is followed by a discussion of some choices in multiplexing behaviour and their impacts. Some designs of RTP usage are discussed. Finally, some guidelines and examples are provided.

2. Definitions

2.1. Terminology

The definitions in Section 3 of [RFC3550] are referenced normatively.

The taxonomy defined in [RFC7656] is referenced normatively.

The following terms and abbreviations are used in this document:

Multiparty: A communication situation including multiple endpoints. In this document, it will be used to refer to situations where more than two endpoints communicate.

RTP Source: The originator or source of a particular RTP stream sent from an endpoint. Identified using an SSRC in a particular RTP session. An RTP source is the source of a single RTP stream, and is associated with a single endpoint and a single media source. An RTP Source is just called a Source in RFC 3550. An endpoint can have multiple RTP sources.

RTP Sink: An endpoint that receives RTP streams. The RTP Sink is identified using one or more SSRCs. The SSRCs used by an RTP sink can be both RTP source ones, as well as used soley to represent the RTP sink. There can be more than one RTP Sink for one RTP source.

Multiplexing: The operation of taking multiple entities as input, aggregating them onto some common resource while keeping the individual entities addressable such that they can later be fully and unambiguously separated (de-multiplexed) again.

RTP Session Group: One or more RTP sessions that are used together

to perform some function. Examples are multiple RTP sessions used to carry different layers of a layered encoding. In an RTP Session Group, CNAMEs are assumed to be valid across all RTP sessions, and designate synchronisation contexts that can cross RTP sessions; i.e. SSRCs that map to a common CNAME can be assumed to have RTCP SR timing information derived from a common clock such that they can be synchronised for playout.

Signalling: The process of configuring endpoints to participate in one or more RTP sessions.

2.2. Subjects Out of Scope

This document is focused on issues that affect RTP. Thus, issues that involve signalling protocols, such as whether SIP, Jingle or some other protocol is in use for session configuration, the particular syntaxes used to define RTP session properties, or the constraints imposed by particular choices in the signalling protocols, are mentioned only as examples in order to describe the RTP issues more precisely.

This document assumes the applications will use RTCP. While there are applications that don’t send RTCP, they do not conform to the RTP specification, and thus can be regarded as reusing the RTP packet format but not implementing the RTP protocol.

3. RTP Multiplexing Overview

3.1. Reasons for Multiplexing and Grouping RTP Streams

There are several reasons why an endpoint might choose to send multiple media streams. In the below discussion, please keep in mind that the reasons for having multiple RTP streams vary and include but are not limited to the following:

- Multiple media sources
- Multiple RTP streams might be needed to represent one media source (for instance when using layered encodings)
- A retransmission stream might repeat some parts of the content of another RTP stream
- An FEC stream might provide material that can be used to repair another RTP stream
- Alternative encodings, for instance using different codecs for the same audio stream

- Alternative formats, for instance multiple resolutions of the same video stream

For each of these reasons, it is necessary to decide if each additional RTP stream is sent within the same RTP session as the other RTP streams, or if it is necessary to use additional RTP sessions to group the RTP streams. The choice suitable for one reason, might not be the choice suitable for another reason. The clearest understanding is associated with multiplexing multiple media sources of the same media type. However, all reasons warrant discussion and clarification on how to deal with them. As the discussion below will show, in reality we cannot choose a single one of SSRC or RTP session multiplexing solutions. To utilise RTP well and as efficiently as possible, both are needed. The real issue is finding the right guidance on when to create additional RTP sessions and when additional RTP streams in the same RTP session is the right choice.

3.2. RTP Multiplexing Points

This section describes the multiplexing points present in the RTP protocol that can be used to distinguish RTP streams and groups of RTP streams. Figure 1 outlines the process of demultiplexing incoming RTP streams:

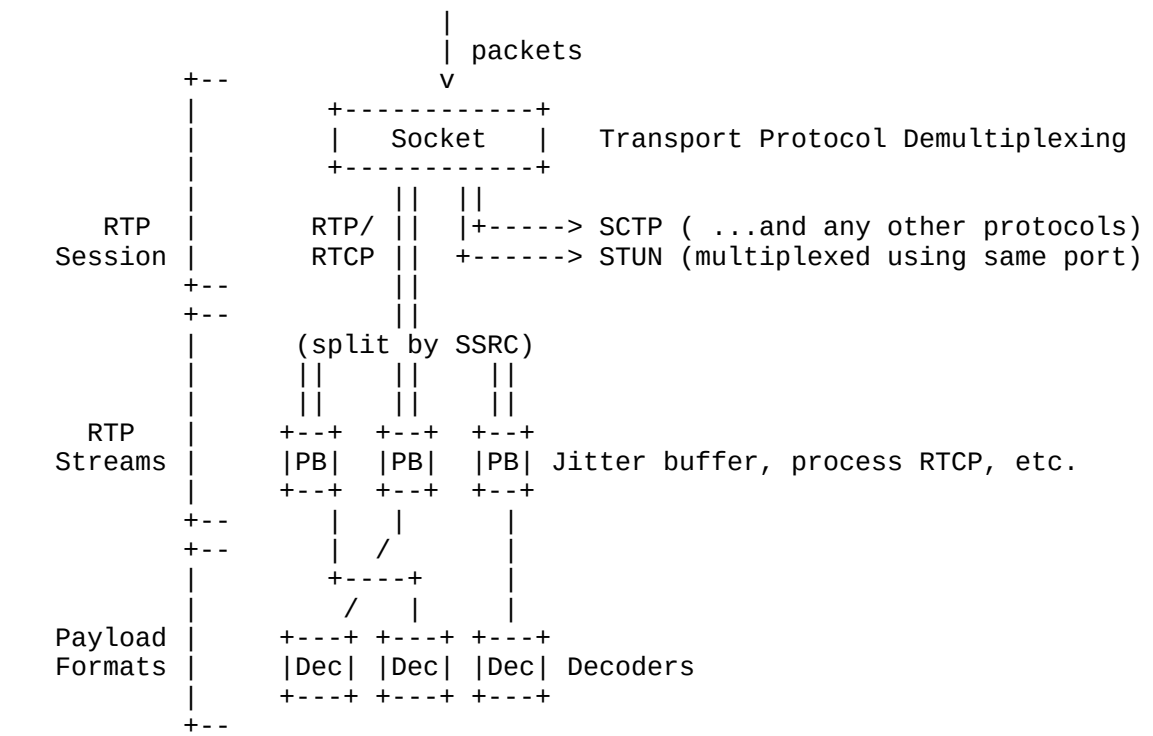

Figure 1: RTP Demultiplexing Process

3.2.1. RTP Session

An RTP Session is the highest semantic layer in the RTP protocol, and represents an association between a group of communicating endpoints. RTP does not contain a session identifier, yet RTP sessions must be possible to separate both across different endpoints and within a single endpoint.

For RTP session separation across endpoints, the set of participants that form an RTP session is defined as those that share a single synchronisation source space [RFC3550]. That is, if a group of participants are each aware of the synchronisation source identifiers belonging to the other participants, then those participants are in a single RTP session. A participant can become aware of a synchronisation source identifier by receiving an RTP packet containing it in the SSRC field or CSRC list, by receiving an RTCP packet mentioning it in an SSRC field, or through signalling (e.g., the Session Description Protocol (SDP) [RFC4566] "a=ssrc:" attribute [RFC5576]). Thus, the scope of an RTP session is determined by the participants' network interconnection topology, in combination with

RTP and RTCP forwarding strategies deployed by the endpoints and any middleboxes, and by the signalling.

For RTP session separation within a single endpoint, RTP relies on the underlying transport layer, and on the signalling to identify RTP sessions in a manner that is meaningful to the application. A single endpoint can have one or more transport flows for the same RTP session. The signalling layer might give RTP sessions an explicit identifier, or the identification might be implicit based on the addresses and ports used. Accordingly, a single RTP session can have multiple associated identifiers, explicit and implicit, belonging to different contexts. For example, when running RTP on top of UDP/IP, an RTP endpoint can identify and delimit an RTP session from other RTP sessions by receiving the multiple UDP flows used as identified based on their UDP source and destination IP addresses and UDP port numbers. Another example is SDP media descriptions (the "m=" line and the following associated lines) signals the transport flow and RTP session configuration for the endpoints part of the RTP session. SDP grouping framework [RFC5888] allows labeling of the media descriptions, for example used so that RTP Session Groups can be created. With Negotiating Media Multiplexing Using the Session Description Protocol (SDP)[I-D.ietf-mmusic-sdp-bundle-negotiation], multiple media descriptions where each represents the RTP streams sent or received for a media source are part of a common RTP session.

The RTP protocol makes no normative statements about the relationship between different RTP sessions, however the applications that use more than one RTP session will have some higher layer understanding of the relationship between the sessions they create.

3.2.2. Synchronisation Source (SSRC)

A synchronisation source (SSRC) identifies an RTP source or an RTP sink. Every endpoint has at least one SSRC identifier, even if it does not send RTP packets. RTP endpoints that are only RTP sinks still send RTCP and use their SSRC identifiers in the RTCP packets they send. An endpoint can have multiple SSRC identifiers if it contains multiple RTP sources (i.e., if it sends multiple RTP streams). Endpoints that are both RTP sources and RTP sinks use the same SSRC in both roles. At any given time, an RTP source has one and only one SSRC - although that can change over the lifetime of the RTP source or sink.

The SSRC is a 32-bit identifier. It is present in every RTP and RTCP packet header, and in the payload of some RTCP packet types. It can also be present in SDP signalling. Unless pre-signalled, e.g. using the SDP "a=ssrc:" attribute [RFC5576], the SSRC is chosen at random. It is not dependent on the network address of the endpoint, and is

intended to be unique within an RTP session. SSRC collisions can occur, and are handled as specified in [RFC3550] and [RFC5576], resulting in the SSRC of the colliding RTP sources and/or sinks changing. An RTP source that changes its network transport address during a session have to choose a new SSRC identifier to avoid being interpreted as looped source, unless the transport layer mechansism, e.g ICE [RFC5245], handle such changes

SSRC identifiers that belong to the same synchronisation context (i.e., that represent RTP streams that can be synchronised using information in RTCP SR packets) use identical CNAME chunks in corresponding RTCP SDES packets. SDP signalling can also be used to provide explicit SSRC grouping [RFC5576].

In some cases, the same SSRC identifier value is used to relate streams in two different RTP sessions, such as in RTP retransmission [RFC4588]. This is to be avoided since there is no guarantee that SSRC values are unique across RTP sessions. For the RTP retransmission [RFC4588] case it is recommended to use explicit binding of the source RTP stream and the redundancy stream, e.g. using the RepairedRtpStreamId RTCP SDES item [I-D.ietf-avtext-rid].

Note that RTP sequence number and RTP timestamp are scoped by the SSRC. Each RTP source will have a different SSRC, and the corresponding RTP stream will have a separate RTP sequence number and timestamp space.

An SSRC identifier is used by different type of sources as well as sinks:

Real Media Source: Connected to a "physical" media source, for example a camera or microphone.

Conceptual Media Source: A source with some attributed property generated by some network node, for example a filtering function in an RTP mixer that provides the most active speaker based on some criteria, or a mix representing a set of other sources.

RTP Sink: A source that does not generate any RTP stream in itself (e.g. an endpoint or middlebox only receiving in an RTP session). It still needs an SSRC for use as source in RTCP reports.

Note that an endpoint that generates more than one media type, e.g. a conference participant sending both audio and video, need not (and should not) use the same SSRC value across RTP sessions. RTCP Compound packets containing the CNAME SDES item is the designated method to bind an SSRC to a CNAME, effectively cross-correlating

SSRCs within and between RTP Sessions as coming from the same endpoint. The main property attributed to SSRCs associated with the same CNAME is that they are from a particular synchronisation context and can be synchronised at playback.

An RTP receiver receiving a previously unseen SSRC value will interpret it as a new source. It might in fact be a previously existing source that had to change SSRC number due to an SSRC conflict. However, the originator of the previous SSRC ought to have ended the conflicting source by sending an RTCP BYE for it prior to starting to send with the new SSRC, so the new SSRC is anyway effectively a new source.

3.2.3. Contributing Source (CSRC)

The Contributing Source (CSRC) is not a separate identifier. Rather an SSRC identifier is listed as a CSRC in the RTP header of a packet generated by an RTP mixer, if the corresponding SSRC was in the header of one of the packets that contributed to the mix.

It is not possible, in general, to extract media represented by an individual CSRC since it is typically the result of a media mixing (merge) operation by an RTP mixer on the individual media streams corresponding to the CSRC identifiers. The exception is the case when only a single CSRC is indicated as this represent forwarding of an RTP stream, possibly modified. The RTP header extension for Mixer-to-Client Audio Level Indication [RFC6465] expands on the receiver’s information about a packet with a CSRC list. Due to these restrictions, CSRC will not be considered a fully qualified multiplexing point and will be disregarded in the rest of this document.

3.2.4. RTP Payload Type

Each RTP stream utilises one or more RTP payload formats. An RTP payload format describes how the output of a particular media codec is framed and encoded into RTP packets. The payload format used is identified by the payload type (PT) field in the RTP packet header. The combination of SSRC and PT therefore identifies a specific RTP stream encoding format. The format definition can be taken from [RFC3551] for statically allocated payload types, but ought to be explicitly defined in signalling, such as SDP, both for static and dynamic payload types. The term "format" here includes whatever can be described by out-of-band signalling means. In SDP, the term "format" includes media type, RTP timestamp sampling rate, codec, codec configuration, payload format configurations, and various robustness mechanisms such as redundant encodings [RFC2198].

The RTP payload type is scoped by the sending endpoint within an RTP session. PT has the same meaning across all RTP streams in an RTP session. All SSRCs sent from a single endpoint share the same payload type definitions. The RTP payload type is designed such that only a single payload type is valid at any time instant in the RTP source's RTP timestamp time line, effectively time-multiplexing different payload types if any change occurs. The payload type used can change on a per-packet basis for an SSRC, for example a speech codec making use of generic comfort noise [RFC3389]. If there is a true need to send multiple payload types for the same SSRC that are valid for the same instant, then redundant encodings [RFC2198] can be used. Several additional constraints than the ones mentioned above need to be met to enable this use, one of which is that the combined payload sizes of the different payload types ought not exceed the transport MTU. If it is acceptable to send multiple formats of the same media source as separate RTP streams (with separate SSRC), simulcast [I-D.ietf-mmusic-sdp-simulcast] can be used.

Other aspects of RTP payload format use are described in How to Write an RTP Payload Format [RFC8088].

The payload type is not a multiplexing point at the RTP layer (see Appendix A for a detailed discussion of why using the payload type as an RTP multiplexing point does not work). The RTP payload type is, however, used to determine how to consume and decode an RTP stream. The RTP payload type number is sometimes used to associate an RTP stream with the signalling; this is not recommended since a specific payload type value can be used in multiple bundled "m=" sections [I-D.ietf-mmusic-sdp-bundle-negotiation]. This association is only possible if unique RTP payload type numbers are used in each context.

3.3. Issues Related to RTP Topologies

The impact of how RTP multiplexing is performed will in general vary with how the RTP session participants are interconnected, described by RTP Topology [RFC7667].

Even the most basic use case, denoted Topo-Point-to-Point in [RFC7667], raises a number of considerations that are discussed in detail in following sections. They range over such aspects as:

- Does my communication peer support RTP as defined with multiple SSRCs per RTP session?
- Do I need network differentiation in form of QoS?
- Can the application more easily process and handle the media streams if they are in different RTP sessions?

- Do I need to use additional RTP streams for RTP retransmission or FEC?
- etc.

For some point to multi-point topologies (e.g. Topo-ASM and Topo-SSM in [RFC7667]), multicast is used to interconnect the session participants. Special considerations (documented in Section 4.2.3) are then needed as multicast is a one-to-many distribution system.

Sometimes an RTP communication can end up in a situation when the communicating peers are not compatible for various reasons:

- No common media codec for a media type thus requiring transcoding.
- Different support for multiple RTP sources and RTP sessions.
- Usage of different media transport protocols, i.e RTP or other.
- Usage of different transport protocols, e.g. UDP, DCCP, TCP.
- Different security solutions, e.g. IPsec, TLS, DTLS, SRTP with different keying mechanisms.

In many situations this is resolved by the inclusion of a translator between the two peers, as described by Topo-PtP-Translator in [RFC7667]. The translator's main purpose is to make the peers look compatible to each other. There can also be other reasons than compatibility to insert a translator in the form of a middlebox or gateway, for example a need to monitor the RTP streams. If the stream transport characteristics are changed by the translator, appropriate media handling can require thorough understanding of the application logic, specifically any congestion control or media adaptation.

The point to point topology can contain one to many RTP sessions with one to many media sources per session, each having one or more RTP sources per media source.

3.4. Issues Related to RTP and RTCP Protocol

Using multiple RTP streams is a well-supported feature of RTP. However, for most implementers or people writing RTP/RTCP applications or extensions attempting to apply multiple streams, it can be unclear when it is most appropriate to add an additional RTP stream in an existing RTP session and when it is better to use multiple RTP sessions. This section discusses the various considerations needed.

3.4.1. The RTP Specification

RFC 3550 contains some recommendations and a bullet list with 5 arguments for different aspects of RTP multiplexing. Let's review Section 5.2 of [RFC3550], reproduced below:

"For efficient protocol processing, the number of multiplexing points should be minimised, as described in the integrated layer processing design principle [ALF]. In RTP, multiplexing is provided by the destination transport address (network address and port number) which is different for each RTP session. For example, in a teleconference composed of audio and video media encoded separately, each medium SHOULD be carried in a separate RTP session with its own destination transport address.

Separate audio and video streams SHOULD NOT be carried in a single RTP session and demultiplexed based on the payload type or SSRC fields. Interleaving packets with different RTP media types but using the same SSRC would introduce several problems:

1. If, say, two audio streams shared the same RTP session and the same SSRC value, and one were to change encodings and thus acquire a different RTP payload type, there would be no general way of identifying which stream had changed encodings.

2. An SSRC is defined to identify a single timing and sequence number space. Interleaving multiple payload types would require different timing spaces if the media clock rates differ and would require different sequence number spaces to tell which payload type suffered packet loss.

3. The RTCP sender and receiver reports (see Section 6.4) can only describe one timing and sequence number space per SSRC and do not carry a payload type field.

4. An RTP mixer would not be able to combine interleaved streams of incompatible media into one stream.

5. Carrying multiple media in one RTP session precludes: the use of different network paths or network resource allocations if appropriate; reception of a subset of the media if desired, for example just audio if video would exceed the available bandwidth; and receiver implementations that use separate processes for the different media, whereas using separate RTP sessions permits either single- or multiple-process implementations.

Using a different SSRC for each medium but sending them in the same RTP session would avoid the first three problems but not the last two.

On the other hand, multiplexing multiple related sources of the same medium in one RTP session using different SSRC values is the norm for multicast sessions. The problems listed above don't apply: an RTP mixer can combine multiple audio sources, for example, and the same treatment is applicable for all of them. It might also be appropriate to multiplex streams of the same medium using different SSRC values in other scenarios where the last two problems do not apply."

Let's consider one argument at a time. The first argument is for using different SSRC for each individual RTP stream, which is very basic.

The second argument is advocating against demultiplexing RTP streams within a session based on their RTP payload type numbers, which still stands as can been seen by the extensive list of issues found in Appendix A.

The third argument is yet another argument against payload type multiplexing.

The fourth argument is against multiplexing RTP packets that require different handling into the same session. As we saw in the discussion of RTP mixers, the RTP mixer must embed application logic to handle streams anyway; the separation of streams according to stream type is just another piece of application logic, which might or might not be appropriate for a particular application. One type of application that can mix different media sources "blindly" is the audio-only "telephone" bridge; most other types of applications need application-specific logic to perform the mix correctly.

The fifth argument discusses network aspects that we will discuss more below in Section 4.2. It also goes into aspects of implementation, like Split Component Terminal (see Section 3.10 of [RFC7667]) endpoints where different processes or inter-connected devices handle different aspects of the whole multi-media session.

A summary of RFC 3550's view on multiplexing is to use unique SSRCs for anything that is its own media/packet stream, and to use different RTP sessions for media streams that don't share a media type. This document supports the first point; it is very valid. The latter needs further discussion, as imposing a single solution on all usages of RTP is inappropriate. Multiple Media Types in an RTP Session specification [I-D.ietf-avtcore-multi-media-rtp-session]

provides a detailed analysis of the potential issues in having multiple media types in the same RTP session. This document provides a wider scope for an RTP session and considers multiple media types in one RTP session as a possible choice for the RTP application designer.

3.4.2. Multiple SSRCs in a Session

Using multiple SSRCs at one endpoint in an RTP session requires resolving some unclear aspects of the RTP specification. These could potentially lead to some interoperability issues as well as some potential significant inefficiencies, as further discussed in "RTP Considerations for Endpoints Sending Multiple Media Streams" [RFC8108]. An RTP application designer should consider these issues and the possible applicaiton impact from lack of appropriate RTP handling or optimization in the peer endpoints.

Using multiple RTP sessions can potentially mitigate application issues caused by multiple SSRCs in an RTP session.

3.4.3. Binding Related Sources

A common problem in a number of various RTP extensions has been how to bind related RTP sources and their RTP streams together. This issue is common to both using additional SSRCs and multiple RTP sessions.

The solutions can be divided into a few groups:

- RTP/RTCP based
- Signalling based (SDP)
- grouping related RTP sessions
- grouping SSRCs within an RTP session

Most solutions are explicit, but some implicit methods have also been applied to the problem.

The SDP-based signalling solutions are:

SDP Media Description Grouping: The SDP Grouping Framework [RFC5888] uses various semantics to group any number of media descriptions. These has previously been considered primarily as grouping RTP sessions, [I-D.ietf-mmusic-sdp-bundle-negotiation] groups multiple media descriptions as a single RTP session.

SDP SSRC grouping: Source-Specific Media Attributes in SDP [RFC5576] includes a solution for grouping SSRCs the same way as the Grouping framework groups Media Descriptions.

SDP MSID grouping: Media Stream Identifiers [I-D.ietf-mmusic-msid] specifies a Session Description Protocol (SDP) Grouping mechanism for RTP streams that can be used to specify relations between RTP streams. This mechanism is used to signal the association between the SDP concept of "media description" and the WebRTC concept of "MediaStream" / "MediaStreamTrack" (Corresponds to the [RFC7656] term "Source Stream") using SDP signalling.

This supports a lot of use cases. All these solutions have shortcomings in cases where the session's dynamic properties are such that it is difficult or resource consuming to keep the list of related SSRCs up to date.

An RTP/RTCP-based solution is to use the RTCP SDES CNAME to bind the RTP streams to an endpoint or synchronization context. For applications with a single RTP stream per type (Media, Source or Redundancy) this is sufficient independent if one or more RTP sessions are used. However, some applications choose not to use it because of perceived complexity or a desire not to implement RTCP and instead use the same SSRC value to bind related RTP streams across multiple RTP sessions. RTP Retransmission [RFC4588] in multiple RTP session mode and Generic FEC [RFC5109] both use this method. This method may work but might have some downsides in RTP sessions with many participating SSRCs. When an SSRC collision occurs, this will force one to change SSRC in all RTP sessions and thus resynchronize all of them instead of only the single media stream having the collision. Therefore, it is not recommended to use identical SSRC values to relate RTP streams.

Another solution to bind SSRCs is an implicit method used by RTP Retransmission [RFC4588] when doing retransmissions in the same RTP session as the source RTP stream. The receiver missing a packet issues an RTP retransmission request, and then awaits a new SSRC carrying the RTP retransmission payload and where that SSRC is from the same CNAME. This limits a requestor to having only one outstanding request on any new source SSRCs per endpoint.

RTP Payload Format Restrictions [I-D.ietf-mmusic-rid] provides an RTP/RTCP based mechanism to unambiguously identify the RTP streams within an RTP session and restrict the streams' payload format parameters in a codec-agnostic way beyond what is provided with the regular Payload Types. The mapping is done by specifying an "a=rid" value in the SDP offer/answer signalling and having the corresponding "rtp-stream-id" value as an SDES item and an RTP header extension.

The RID solution also includes a solution for binding redundancy RTP streams to their original source RTP streams, given that those use RID identifiers.

It can be noted that Section 8.3 of the RTP Specification [RFC3550] recommends using a single SSRC space across all RTP sessions for layered coding. Based on the experience so far however, we recommend to use a solution doing explicit binding between the RTP streams so what the used SSRC values are do not matter. That way solutions using multiple RTP streams in a single RTP session and multiple RTP sessions uses the same solution.

3.4.4. Forward Error Correction

There exist a number of Forward Error Correction (FEC) based schemes for how to reduce the packet loss of the original streams. Most of the FEC schemes will protect a single source flow. The protection is achieved by transmitting a certain amount of redundant information that is encoded such that it can repair one or more packet losses over the set of packets the redundant information protects. This sequence of redundant information also needs to be transmitted as its own media stream, or in some cases, instead of the original media stream. Thus, many of these schemes create a need for binding related flows as discussed above. Looking at the history of these schemes, there are schemes using multiple SSRCs and schemes using multiple RTP sessions, and some schemes that support both modes of operation.

Using multiple RTP sessions supports the case where some set of receivers might not be able to utilise the FEC information. By placing it in a separate RTP session and if separating RTP sessions on transport level, FEC can easily be ignored already on transport level.

In usages involving multicast, having the FEC information on its own multicast group allows for similar flexibility. This is especially useful when receivers see very heterogeneous packet loss rates. Those receivers that are not seeing packet loss don’t need to join the multicast group with the FEC data, and so avoid the overhead of receiving unnecessary FEC packets, for example.

4. Considerations for RTP Multiplexing

4.1. Interworking Considerations

There are several different kinds of interworking, and this section discusses two; interworking between different applications including the implications of potentially different RTP multiplexing point

choices and limitations that have to be considered when working with some legacy applications.

4.1.1. Application Interworking

It is not uncommon that applications or services of similar but not identical usage, especially the ones intended for interactive communication, encounter a situation where one want to interconnect two or more of these applications.

In these cases, one ends up in a situation where one might use a gateway to interconnect applications. This gateway must then either change the multiplexing structure or adhere to the respective limitations in each application.

There are two fundamental approaches to gatewaying: RTP Translator interworking (RTP bridging), where the gateway acts as an RTP Translator, with the two applications being members of the same RTP session, and Gateway Interworking (with RTP termination), where there are independent RTP sessions running from each interconnected application to the gateway.

4.1.2. RTP Translator Interworking

From an RTP perspective the RTP Translator approach could work if all the applications are using the same codecs with the same payload types, have made the same multiplexing choices, and have the same capabilities in number of simultaneous RTP streams combined with the same set of RTP/RTCP extensions being supported. Unfortunately, this might not always be true.

When one is gatewaying via an RTP Translator, an important consideration is if the two applications being interconnected need to use the same approach to multiplexing. If one side is using RTP session multiplexing and the other is using SSRC multiplexing with bundle, the mapping of SDP "m=" lines between both sides requires that the order in bundled and not bundled sides will be the same to allow routing without mapping, it is possible for the RTP translator to map the RTP streams between both sides. There are also challenges with SSRC collision handling since there may be a collision on the SSRC multiplexing side but the RTP session multiplexing side will not be aware of any collision unless SSRC translation is applied on the RTP translator. Furthermore, if one of the applications is capable of working in several modes (such as being able to use additional RTP streams in one RTP session or multiple RTP sessions at will), and the other one is not, successful interconnection depends on locking the more flexible application into the operating mode where interconnection can be successful, even if no participants are using

the less flexible application when the RTP sessions are being created.

4.1.3. Gateway Interworking

When one terminates RTP sessions at the gateway, there are certain tasks that the gateway has to carry out:

- Generating appropriate RTCP reports for all RTP streams (possibly based on incoming RTCP reports), originating from SSRCs controlled by the gateway.
- Handling SSRC collision resolution in each application's RTP sessions.
- Signalling, choosing and policing appropriate bit-rates for each session.

For applications that uses any security mechanism, e.g. in the form of SRTP, the gateway needs to be able to decrypt incoming packets and re-encrypt them in the other application's security context. This is necessary even if all that's needed is a simple remapping of SSRC numbers. If this is done, the gateway also needs to be a member of the security contexts of both sides, of course.

Other tasks a gateway might need to apply include transcoding (for incompatible codec types), media-level adaptations that cannot be solved through media negotiation (such as rescaling for incompatible video size requirements), suppression of content that is known not to be handled in the destination application, or the addition or removal of redundancy coding or scalability layers to fit the needs of the destination domain.

From the above, we can see that the gateway needs to have an intimate knowledge of the application requirements; a gateway is by its nature application specific, not a commodity product.

This fact reveals the potential for these gateways to block application evolution by blocking RTP and RTCP extensions that the applications have been extended with but that are unknown to the gateway.

If one uses security functions, like SRTP, and as can be seen from above, they incur both additional risk due to the requirement to have the gateway in the security association between the endpoints (unless the gateway is on the transport level), and additional complexities in form of the decrypt-encrypt cycles needed for each forwarded packet. SRTP, due to its keying structure, also requires that each

RTP session needs different master keys, as use of the same key in two RTP sessions can for some ciphers result in two-time pads that completely breaks the confidentiality of the packets.

4.1.4. Multiple SSRC Legacy Considerations

Historically, the most common RTP use cases have been point to point Voice over IP (VoIP) or streaming applications, commonly with no more than one media source per endpoint and media type (typically audio and video). Even in conferencing applications, especially voice-only, the conference focus or bridge has provided a single stream with a mix of the other participants to each participant. It is also common to have individual RTP sessions between each endpoint and the RTP mixer, meaning that the mixer functions as an RTP-terminating gateway.

When establishing RTP sessions that can contain endpoints that aren't updated to handle multiple streams following these recommendations, a particular application can have issues with multiple SSRCs within a single session. These issues include:

1. Need to handle more than one stream simultaneously rather than replacing an already existing stream with a new one.

2. Be capable of decoding multiple streams simultaneously.

3. Be capable of rendering multiple streams simultaneously.

This indicates that gateways attempting to interconnect to this class of devices has to make sure that only one RTP stream of each type gets delivered to the endpoint if it's expecting only one, and that the multiplexing format is what the device expects. It is highly unlikely that RTP translator-based interworking can be made to function successfully in such a context.

4.2. Network Considerations

The RTP multiplexing choice has impact on network level mechanisms that need to be considered by the implementer.

4.2.1. Quality of Service

When it comes to Quality of Service mechanisms, they are either flow based or packet marking based. RSVP [RFC2205] is an example of a flow based mechanism, while Diff-Serv [RFC2474] is an example of a packet marking based one. For a packet marking based scheme, the method of multiplexing will not affect the possibility to use QoS.

However, for a flow based scheme there is a clear difference between the multiplexing methods. Additional SSRC will result in all RTP streams being part of the same 5-tuple (protocol, source address, destination address, source port, destination port) which is the most common selector for flow based QoS.

It must also be noted that packet marking based QoS mechanisms can have limitations. A general observation is that different Differentiated Services Code Points (DSCP) can be assigned to different packets within a flow as well as within an RTP stream. However, care must be taken when considering which forwarding behaviours that are applied on path due to these DSCPs. In some cases the forwarding behaviour can result in packet reordering. For more discussion of this see [RFC7657].

The method for assigning marking to packets can impact what number of RTP sessions to choose. If this marking is done using a network ingress function, it can have issues discriminating the different RTP streams. The network API on the endpoint also needs to be capable of setting the marking on a per-packet basis to reach the full functionality.

4.2.2. NAT and Firewall Traversal

In today’s network there exist a large number of middleboxes. The ones that normally have most impact on RTP are Network Address Translators (NAT) and Firewalls (FW).

Below we analyse and comment on the impact of requiring more underlying transport flows in the presence of NATs and Firewalls:

End-Point Port Consumption: A given IP address only has 65536 available local ports per transport protocol for all consumers of ports that exist on the machine. This is normally never an issue for an end-user machine. It can become an issue for servers that handle large number of simultaneous streams. However, if the application uses ICE to authenticate STUN requests, a server can serve multiple endpoints from the same local port, and use the whole 5-tuple (source and destination address, source and destination port, protocol) as identifier of flows after having securely bound them to the remote endpoint address using the STUN request. In theory the minimum number of media server ports needed are the maximum number of simultaneous RTP Sessions a single endpoint can use. In practice, implementation will probably benefit from using more server ports to simplify implementation or avoid performance bottlenecks.

NAT State: If an endpoint sits behind a NAT, each flow it generates

to an external address will result in a state that has to be kept in the NAT. That state is a limited resource. In home or Small Office/Home Office (SOHO) NATs, memory or processing are usually the most limited resources. For large scale NATs serving many internal endpoints, available external ports are likely the scarce resource. Port limitations is primarily a problem for larger centralised NATs where endpoint independent mapping requires each flow to use one port for the external IP address. This affects the maximum number of internal users per external IP address. However, it is worth pointing out that a real-time video conference session with audio and video is likely using less than 10 UDP flows, compared to certain web applications that can use 100+ TCP flows to various servers from a single browser instance.

NAT Traversal Extra Delay: Performing the NAT/FW traversal takes a certain amount of time for each flow. It also takes time in a phase of communication between accepting to communicate and the media path being established which is fairly critical. The best case scenario for how much extra time it takes after finding the first valid candidate pair following the specified ICE procedures are: 1.5*RTT + Ta*(Additional_Flows-1), where Ta is the pacing timer. That assumes a message in one direction, and then an immediate triggered check back. The reason it isn't more, is that ICE first finds one candidate pair that works prior to attempting to establish multiple flows. Thus, there is no extra time until one has found a working candidate pair. Based on that working pair the needed extra time is to in parallel establish the, in most cases 2-3, additional flows. However, packet loss causes extra delays, at least 100 ms, which is the minimal retransmission timer for ICE.

NAT Traversal Failure Rate: Due to the need to establish more than a single flow through the NAT, there is some risk that establishing the first flow succeeds but that one or more of the additional flows fail. The risk that this happens is hard to quantify, but ought to be fairly low as one flow from the same interfaces has just been successfully established. Thus only rare events such as NAT resource overload, or selecting particular port numbers that are filtered etc., ought to be reasons for failure.

Deep Packet Inspection and Multiple Streams: Firewalls differ in how deeply they inspect packets. There exist some potential that deeply inspecting firewalls will have similar legacy issues with multiple SSRCs as some stack implementations.

Using additional RTP streams in the same RTP session and transport flow does not introduce any additional NAT traversal complexities per RTP stream. This can be compared with normally one or two additional

transport flows per RTP session when using multiple RTP sessions. Additional lower layer transport flows will be needed, unless an explicit de-multiplexing layer is added between RTP and the transport protocol. At time of writing no such mechanism was defined.

4.2.3. Multicast

Multicast groups provides a powerful tool for a number of real-time applications, especially the ones that desire broadcast-like behaviours with one endpoint transmitting to a large number of receivers, like in IPTV. There are also the RTP/RTCP extension to better support Source Specific Multicast (SSM) [RFC5760]. Another application is the Many to Many communication, which RTP [RFC3550] was originally built to support. But the multicast semantics do result in a certain number of limitations.

One limitation is that for any group, sender side adaptation to the actual receiver properties causes degradation for all participants to what is supported by the receiver with the worst conditions among the group participants. For broadcast type of applications this is not acceptable. Instead, various receiver-based solutions are employed to ensure that the receivers achieve best possible performance. By using scalable encoding and placing each scalability layer in a different multicast group, the receiver can control the amount of traffic it receives. To have each scalability layer on a different multicast group, one RTP session per multicast group is used.

In addition, the transport flow considerations in multicast are a bit different from unicast; NATs with port translation are not useful in the multicast environment, meaning that the entire port range of each multicast address is available for distinguishing between RTP sessions.

Thus, when using broadcast applications it appears easiest and most straightforward to use multiple RTP sessions for sending different media flows used for adapting to network conditions. It is also common that streams that improve transport robustness are sent in their own multicast group to allow for interworking with legacy or to support different levels of protection.

For many to many applications there is different needs. Here it will depend on how the actual application is realized what is the most appropriate choice. With sender side congestion control there might not exist any benefit with using multiple RTP session.

The properties of a broadcast application using RTP multicast:

1. Uses a group of RTP sessions, not one. Each endpoint will need to be a member of a number of RTP sessions in order to perform well.

2. Within each RTP session, the number of RTP sinks is likely to be much larger than the number of RTP sources.

3. The applications need signalling functions to identify the relationships between RTP sessions.

4. The applications need signalling or RTP/RTCP functions to identify the relationships between SSRCs in different RTP sessions when needs beyond CNAME exists.

Both broadcast and many to many multicast applications do share a signalling requirement; all of the participants will need to have the same RTP and payload type configuration. Otherwise, A could for example be using payload type 97 as the video codec H.264 while B thinks it is MPEG-2. It is to be noted that SDP offer/answer [RFC3264] is not appropriate for ensuring this property in broadcast/ multicast context. The signalling aspects of broadcast/multicast are not explored further in this memo.

Security solutions for this type of group communications are also challenging. First, the key-management and the security protocol needs to support group communication. Second, source authentication requires special solutions. For more discussion on this please review Options for Securing RTP Sessions [RFC7201].

4.3. Security and Key Management Considerations

When dealing with point-to-point, 2-member RTP sessions only, there are few security issues that are relevant to the choice of having one RTP session or multiple RTP sessions. However, there are a few aspects of multiparty sessions that might warrant consideration. For general information of possible methods of securing RTP, please review RTP Security Options [RFC7201].

4.3.1. Security Context Scope

When using SRTP [RFC3711] the security context scope is important and can be a necessary differentiation in some applications. As SRTP’s crypto suites are (so far) built around symmetric keys, the receiver will need to have the same key as the sender. This results in that no one in a multi-party session can be certain that a received packet really was sent by the claimed sender and not by another party having access to the key. At least unless TESLA source authentication [RFC4383], which adds delay to achieve source authentication. In

most cases symmetric ciphers provide sufficient security properties, but there are a few cases where this does create issues.

The first case is when someone leaves a multi-party session and one wants to ensure that the party that left can no longer access the RTP streams. This requires that everyone re-keys without disclosing the keys to the excluded party.

A second case is when using security as an enforcing mechanism for differentiation. Take for example a scalable layer or a high quality simulcast version that only premium users are allowed to access. The mechanism preventing a receiver from getting the high quality stream can be based on the stream being encrypted with a key that user can't access without paying premium, having the key-management limit access to the key.

SRTP [RFC3711] has no special functions for dealing with different sets of master keys for different SSRCs. The key-management functions have different capabilities to establish different sets of keys, normally on a per endpoint basis. For example, DTLS-SRTP [RFC5764] and Security Descriptions [RFC4568] establish different keys for outgoing and incoming traffic from an endpoint. This key usage has to be written into the cryptographic context, possibly associated with different SSRCs.

4.3.2. Key Management for Multi-party sessions

Performing key-management for multi-party session can be a challenge. This section considers some of the issues.

Multi-party sessions, such as transport translator based sessions and multicast sessions, cannot use Security Description [RFC4568] nor DTLS-SRTP [RFC5764] without an extension as each endpoint provides its set of keys. In centralised conferences, the signalling counterpart is a conference server and the media plane unicast counterpart (to which DTLS messages would be sent) is the transport translator. Thus, an extension like Encrypted Key Transport [I-D.ietf-perc-srtp-ekt-diet] or a MIKEY [RFC3830] based solution that allows for keying all session participants with the same master key is needed.

4.3.3. Complexity Implications

The usage of security functions can surface complexity implications from the choice of multiplexing and topology. This becomes especially evident in RTP topologies having any type of middlebox that processes or modifies RTP/RTCP packets. Where there is very small overhead for an RTP translator or mixer to rewrite an SSRC

value in the RTP packet of an unencrypted session, the cost is higher when using cryptographic security functions. For example, if using SRTP [RFC3711], the actual security context and exact crypto key are determined by the SSRC field value. If one changes SSRC, the encryption and authentication must use another key. Thus, changing the SSRC value implies a decryption using the old SSRC and its security context, followed by an encryption using the new one.

5. RTP Multiplexing Design Choices

This section discusses how some RTP multiplexing design choices can be used in applications to achieve certain goals, and a summary of the implications of such choices. For each design there is discussion of benefits and downsides.

5.1. Single SSRC per Endpoint

In this design each endpoint in a point-to-point session has only a single SSRC, thus the RTP session contains only two SSRCs, one local and one remote. This session can be used both unidirectional, i.e. only a single RTP stream or bi-directional, i.e. both endpoints have one RTP stream each. If the application needs additional media flows between the endpoints, they will have to establish additional RTP sessions.

The Pros:

1. This design has great legacy interoperability potential as it will not tax any RTP stack implementations.

2. The signalling has good possibilities to negotiate and describe the exact formats and bit-rates for each RTP stream, especially using today’s tools in SDP.

3. It is possible to control security association per RTP stream with current key-management, since each RTP stream is directly related to an RTP session, and the most used keying mechanisms operates on a per-session basis.

The Cons:

a. The number of RTP sessions grows directly in proportion with the number of RTP streams, which has the implications:

   * Linear growth of the amount of NAT/FW state with number of RTP streams.

* Increased delay and resource consumption from NAT/FW traversal.

* Likely larger signalling message and signalling processing requirement due to the amount of session related information.

* Higher potential for a single RTP stream to fail during transport between the endpoints.

b. When the number of RTP sessions grows, the amount of explicit state for relating RTP stream also grows, linearly, depending on how the application needs to relate RTP streams.

c. The port consumption might become a problem for centralised services, where the central node’s port consumption grows rapidly with the number of sessions.

d. For applications where the RTP stream usage is highly dynamic, i.e. entering and leaving, the amount of signalling can grow high. Issues can also arise from the timely establishment of additional RTP sessions.

e. If, against the recommendation, the same SSRC value is reused in multiple RTP sessions rather than being randomly chosen, interworking with applications that use a different multiplexing structure will require SSRC translation.

RTP applications that need to interwork with legacy RTP applications can potentially benefit from this structure. However, a large number of media descriptions in SDP can also run into issues with existing implementations. For any application needing a larger number of media flows, the overhead can become very significant. This structure is also not suitable for multi-party sessions, as any given RTP stream from each participant, although having same usage in the application, needs its own RTP session. In addition, the dynamic behaviour that can arise in multi-party applications can tax the signalling system and make timely media establishment more difficult.

5.2. Multiple SSRCs of the Same Media Type

In this design, each RTP session serves only a single media type. The RTP session can contain multiple RTP streams, either from a single endpoint or from multiple endpoints. This commonly creates a low number of RTP sessions, typically only one for audio and one for video, with a corresponding need for two listening ports when using RTP/RTCP multiplexing.

The Pros:

1. Low number of RTP sessions needed compared to Single SSRC per Endpoint case. This implies:

   * Reduced NAT/FW state

   * Lower NAT/FW Traversal Cost in both processing and delay.

2. Works well with Split Component Terminal (see Section 3.10 of [RFC7667]) where the split is per media type.

3. Enables Flow-based QoS with different prioritisation between media types.

4. For applications with dynamic usage of RTP streams, i.e. frequently added and removed, having much of the state associated with the RTP session rather than per individual SSRC can avoid the need for in-session signalling of meta-information about each SSRC.

5. Low overhead for security association establishment.

The Cons:

a. Some potential for concern with legacy implementations that don’t support the RTP specification fully when it comes to handling multiple SSRC per endpoint.

b. Not possible to control security association for sets of RTP streams within the same media type with today’s key- management mechanisms, unless these are split into different RTP sessions.

For RTP applications where all RTP streams of the same media type share same usage, this structure provides efficiency gains in amount of network state used and provides more fate sharing with other media flows of the same type. At the same time, it is still maintaining almost all functionalities when it comes to negotiation in the signalling of the properties for the individual media type, and also enables flow based QoS prioritisation between media types. It handles multi-party session well, independently of multicast or centralised transport distribution, as additional sources can dynamically enter and leave the session.

5.3. Multiple Sessions for one Media type

This design goes one step further than above (Section 5.2) by using multiple RTP sessions also for a single media type. The main reason for going in this direction is that the RTP application needs separation of the RTP streams due to their usage. Some typical

reasons for going to this design are scalability over multicast, simulcast, need for extended QoS prioritisation of RTP streams due to their usage in the application, or the need for fine- grained signalling using today's tools.

The Pros:

1. More suitable for multicast usage where receivers can individually select which RTP sessions they want to participate in, assuming each RTP session has its own multicast group.

2. The application can indicate its usage of the RTP streams on RTP session level, in case multiple different usages exist.

3. Less need for SSRC specific explicit signalling for each media stream and thus reduced need for explicit and timely signalling.

4. Enables detailed QoS prioritisation for flow-based mechanisms.

5. Works well with Split Component Terminal (see Section 3.10 of [RFC7667]).

6. The scope for who is included in a security association can be structured around the different RTP sessions, thus enabling such functionality with existing key-management.

The Cons:

a. Increases the amount of RTP sessions compared to Multiple SSRCs of the Same Media Type.

b. Increased amount of session configuration state.

c. For RTP streams that are part of scalability, simulcast or transport robustness, a method to bind sources across multiple RTP sessions is needed.

d. Some potential for concern with legacy implementations that does not support the RTP specification fully when it comes to handling multiple SSRC per endpoint.

e. Higher overhead for security association establishment due to the increased number of RTP sessions.

f. If the applications need finer control than on RTP session level over which participants that are included in different sets of security associations, most of today's key-management will have difficulties establishing such a session.

For more complex RTP applications that have several different usages for RTP streams of the same media type and / or uses scalability or simulcast, this solution can enable those functions at the cost of increased overhead associated with the additional sessions. This type of structure is suitable for more advanced applications as well as multicast-based applications requiring differentiation to different participants.

5.4. Multiple Media Types in one Session

This design uses a single RTP session for multiple different media types, like audio and video, and possibly also transport robustness mechanisms like FEC or Retransmission. An endpoint can have zero, one or more media sources per media type. Resulting in a number of RTP streams of various media types and both source and redundancy type.

The Pros:

1. Single RTP session which implies:
    * Minimal NAT/FW state.
    * Minimal NAT/FW Traversal Cost.
    * Fate-sharing for all media flows.
2. Can handle dynamic allocations of RTP streams well on an RTP level. Depends on the application’s needs for explicit indication of the stream usage and how timely that can be signalled.
3. Minimal overhead for security association establishment.

The Cons:

a. Less suitable for interworking with other applications that uses individual RTP sessions per media type or multiple sessions for a single media type, due to the potential need of SSRC translation.

b. Negotiation of bandwidth for the different media types is currently only possible using RID [I-D.ietf-mmusic-rid] in SDP.

c. Not suitable for Split Component Terminal (see Section 3.10 of [RFC7667]).

d. Flow-based QoS cannot provide separate treatment of RTP streams compared to others in the single RTP session.

e. If there is significant asymmetry between the RTP streams' RTCP reporting needs, there are some challenges in configuration and usage to avoid wasting RTCP reporting on the RTP stream that does not need that frequent reporting.

f. Not suitable for applications where some receivers like to receive only a subset of the RTP streams, especially if multicast or transport translator is being used.

g. Additional concern with legacy implementations that do not support the RTP specification fully when it comes to handling multiple SSRC per endpoint, as also multiple simultaneous media types needs to be handled.

h. If the applications need finer control over which session participants that are included in different sets of security associations, most key-management will have difficulties establishing such a session.

5.5. Summary

There are some clear similarities between these designs. Both the "Single SSRC per Endpoint" and the "Multiple Media Types in one Session" are cases that require full explicit signalling of the media stream relations. However, they operate on two different levels where the first primarily enables session level binding, and the second needs SSRC level binding. From another perspective, the two solutions are the two extreme points when it comes to number of RTP sessions needed.

The two other designs "Multiple SSRCs of the Same Media Type" and "Multiple Sessions for one Media Type" are two examples that primarily allows for some implicit mapping of the role or usage of the RTP streams based on which RTP session they appear in. It thus potentially allows for less signalling and in particular reduces the need for real-time signalling in dynamic sessions. They also represent points in between the first two designs when it comes to amount of RTP sessions established, i.e. representing an attempt to balance the amount of RTP sessions with the functionality the communication session provides both on network level and on signalling level.

6. Guidelines

This section contains a number of multi-stream guidelines for implementers or specification writers.

Do not use the same SSRC across RTP sessions: As discussed in Section 3.4.3 there exist drawbacks in using the same SSRC in multiple RTP sessions as a mechanism to bind related RTP streams together. It is instead recommended to use a mechanism to explicitly signal the relation, either in RTP/RTCP or in the signalling mechanism used to establish the RTP session(s).

Use additional RTP streams for additional media sources: In the cases where an RTP endpoint needs to transmit additional RTP streams of the same media type in the application, with the same processing requirements at the network and RTP layers, it is suggested to send them in the same RTP session. For example a telepresence room where there are three cameras, and each camera captures 2 persons sitting at the table, sending each camera as its own RTP stream within a single RTP session is suggested.

Use additional RTP sessions for streams with different requirements:

When RTP streams have different processing requirements from the network or the RTP layer at the endpoints, it is suggested that the different types of streams are put in different RTP sessions. This includes the case where different participants want different subsets of the set of RTP streams.

When using multiple RTP Sessions use grouping: When using Multiple RTP session solutions, it is suggested to explicitly group the involved RTP sessions when needed using a signalling mechanism, for example The Session Description Protocol (SDP) Grouping Framework [RFC5888], using some appropriate grouping semantics.

RTP/RTCP Extensions Support Multiple RTP Streams as well as Multiple RTP sessions:
When defining an RTP or RTCP extension, the creator needs to consider if this extension is applicable to use with additional SSRCs and multiple RTP sessions. Any extension intended to be generic must support both. Extensions that are not as generally applicable will have to consider if interoperability is better served by defining a single solution or providing both options.

Transport Support Extensions: When defining new RTP/RTCP extensions intended for transport support, like the retransmission or FEC mechanisms, they must include support for both multiple RTP streams in the same RTP sessions and multiple RTP sessions, such that application developers can choose freely from the set of mechanisms without concerning themselves with which of the multiplexing choices a particular solution supports.

7. Open Issues

There are currently some issues that needs to be resolved before this document is ready to be published:

1. Does the MSID text need to be updated and clarified based on the evolution of MSID since previous version. Section 3.4.3.

2. Changed definitions needs review and consideration.

8. IANA Considerations

This document makes no request of IANA.

Note to RFC Editor: this section can be removed on publication as an RFC.

9. Security Considerations

The security considerations of the RTP specification [RFC3550] and any applicable RTP profile [RFC3551],[RFC4585],[RFC3711], the extensions for sending multiple media types in a single RTP session [I-D.ietf-avtcore-multi-media-rtp-session], MSID [I-D.ietf-mmusic-msid], RID [I-D.ietf-mmusic-rid], BUNDLE [I-D.ietf-mmusic-sdp-bundle-negotiation], [RFC5760], [RFC5761], apply if selected and thus needs to be considered in the evaluation.

There is discussion of the security implications of choosing multiple SSRC vs multiple RTP sessions in Section 4.3.

10. Contributors

Hui Zheng (Marvin) from Huawei contributed to WG draft versions -04 and -05 of the document.

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Appendix A. Dismissing Payload Type Multiplexing

This section documents a number of reasons why using the payload type as a multiplexing point is unsuitable for most things related to multiple RTP streams. If one attempts to use Payload type multiplexing beyond its defined usage, that has well known negative effects on RTP. To use payload type as the single discriminator for multiple streams implies that all the different RTP streams are being sent with the same SSRC, thus using the same timestamp and sequence number space. This has many effects:

1. Putting restraint on RTP timestamp rate for the multiplexed media. For example, RTP streams that use different RTP timestamp rates cannot be combined, as the timestamp values need to be consistent across all multiplexed media frames. Thus streams are forced to use the same RTP timestamp rate. When this is not possible, payload type multiplexing cannot be used.

2. Many RTP payload formats can fragment a media object over multiple RTP packets, like parts of a video frame. These payload formats need to determine the order of the fragments to correctly decode them. Thus, it is important to ensure that all fragments related to a frame or a similar media object are transmitted in sequence and without interruptions within the object. This can relatively simple be solved on the sender side by ensuring that the fragments of each RTP stream are sent in sequence.

3. Some media formats require uninterrupted sequence number space between media parts. These are media formats where any missing

RTP sequence number will result in decoding failure or invoking a repair mechanism within a single media context. The text/T140 payload format [RFC4103] is an example of such a format. These formats will need a sequence numbering abstraction function between RTP and the individual RTP stream before being used with payload type multiplexing.

4. Sending multiple streams in the same sequence number space makes it impossible to determine which payload type, which stream a packet loss relates to, and thus to which stream to potentially apply packet loss concealment or other stream-specific loss mitigation mechanisms.

5. If RTP Retransmission [RFC4588] is used and there is a loss, it is possible to ask for the missing packet(s) by SSRC and sequence number, not by payload type. If only some of the payload type multiplexed streams are of interest, there is no way of telling which missing packet(s) belong to the interesting stream(s) and all lost packets need be requested, wasting bandwidth.

6. The current RTCP feedback mechanisms are built around providing feedback on RTP streams based on stream ID (SSRC), packet (sequence numbers) and time interval (RTP Timestamps). There is almost never a field to indicate which payload type is reported, so sending feedback for a specific RTP payload type is difficult without extending existing RTCP reporting.

7. The current RTCP media control messages [RFC5104] specification is oriented around controlling particular media flows, i.e. requests are done addressing a particular SSRC. Such mechanisms would need to be redefined to support payload type multiplexing.

8. The number of payload types are inherently limited. Accordingly, using payload type multiplexing limits the number of streams that can be multiplexed and does not scale. This limitation is exacerbated if one uses solutions like RTP and RTCP multiplexing [RFC5761] where a number of payload types are blocked due to the overlap between RTP and RTCP.

9. At times, there is a need to group multiplexed streams and this is currently possible for RTP sessions and for SSRC, but there is no defined way to group payload types.

10. It is currently not possible to signal bandwidth requirements per RTP stream when using payload type multiplexing.

11. Most existing SDP media level attributes cannot be applied on a per payload type level and would require re-definition in that context.

12. A legacy endpoint that does not understand the indication that different RTP payload types are different RTP streams might be slightly confused by the large amount of possibly overlapping or identically defined RTP payload types.

Appendix B. Signalling Considerations

Signalling is not an architectural consideration for RTP itself, so this discussion has been moved to an appendix. However, it is hugely important for anyone building complete applications, so it is deserving of discussion.

The issues raised here need to be addressed in the WGs that deal with signalling; they cannot be addressed by tweaking, extending or profiling RTP.

There exist various signalling solutions for establishing RTP sessions. Many are SDP [RFC4566] based, however SDP functionality is also dependent on the signalling protocols carrying the SDP. RTSP [RFC7826] and SAP [RFC2974] both use SDP in a declarative fashion, while SIP [RFC3261] uses SDP with the additional definition of Offer/Answer [RFC3264]. The impact on signalling and especially SDP needs to be considered as it can greatly affect how to deploy a certain multiplexing point choice.

B.1. Session Oriented Properties

One aspect of the existing signalling is that it is focused around RTP sessions, or at least in the case of SDP the media description. There are a number of things that are signalled on media description level but those are not necessarily strictly bound to an RTP session and could be of interest to signal specifically for a particular RTP stream (SSRC) within the session. The following properties have been identified as being potentially useful to signal not only on RTP session level:

- Bitrate/Bandwidth exist today only at aggregate or as a common "any RTP stream" limit, unless either codec-specific bandwidth limiting or RTCP signalling using TMMBR is used.

- Which SSRC that will use which RTP payload types (this will be visible from the first media packet, but is sometimes useful to know before packet arrival).

Some of these issues are clearly SDP's problem rather than RTP limitations. However, if the aim is to deploy an solution using additional SSRCs that contains several sets of RTP streams with different properties (encoding/packetization parameter, bit-rate, etc.), putting each set in a different RTP session would directly enable negotiation of the parameters for each set. If insisting on additional SSRC only, a number of signalling extensions are needed to clarify that there are multiple sets of RTP streams with different properties and that they need in fact be kept different, since a single set will not satisfy the application's requirements.

For some parameters, such as RTP payload type, resolution and framerate, a SSRC-linked mechanism has been proposed in [I-D.ietf-mmusic-rid]

B.2. SDP Prevents Multiple Media Types

SDP chose to use the m= line both to delineate an RTP session and to specify the top level of the MIME media type; audio, video, text, image, application. This media type is used as the top-level media type for identifying the actual payload format and is bound to a particular payload type using the rtpmap attribute. This binding has to be loosened in order to use SDP to describe RTP sessions containing multiple MIME top level types.

[I-D.ietf-mmusic-sdp-bundle-negotiation] describes how to let multiple SDP media descriptions use a single underlying transport in SDP, which allows to define one RTP session with media types having different MIME top level types.

B.3. Signalling RTP stream Usage

RTP streams being transported in RTP has some particular usage in an RTP application. This usage of the RTP stream is in many applications so far implicitly signalled. For example, an application might choose to take all incoming audio RTP streams, mix them and play them out. However, in more advanced applications that use multiple RTP streams there will be more than a single usage or purpose among the set of RTP streams being sent or received. RTP applications will need to signal this usage somehow. The signalling used will have to identify the RTP streams affected by their RTP-level identifiers, which means that they have to be identified either by their session or by their SSRC + session.

In some applications, the receiver cannot utilise the RTP stream at all before it has received the signalling message describing the RTP stream and its usage. In other applications, there exists a default handling that is appropriate.

If all RTP streams in an RTP session are to be treated in the same way, identifying the session is enough. If SSRCs in a session are to be treated differently, signalling needs to identify both the session and the SSRC.

If this signalling affects how any RTP central node, like an RTP mixer or translator that selects, mixes or processes streams, treats the streams, the node will also need to receive the same signalling to know how to treat RTP streams with different usage in the right fashion.

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Sending Multiple RTP Streams in a Single RTP Session: Grouping RTCP Reception Statistics and Other Feedback draft-ietf-avtcore-rtp-multi-stream-optimisation-12

Abstract

RTP allows multiple RTP streams to be sent in a single session, but requires each Synchronisation Source (SSRC) to send RTCP reception quality reports for every other SSRC visible in the session. This causes the number of RTCP reception reports to grow with the number of SSRCs, rather than the number of endpoints. In many cases most of these RTCP reception reports are unnecessary, since all SSRCs of an endpoint are normally co-located and see the same reception quality. This memo defines a Reporting Group extension to RTCP to reduce the reporting overhead in such scenarios.

Status of This Memo

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1. Introduction

The Real-time Transport Protocol (RTP) [RFC3550] is a protocol for group communication, supporting multiparty multimedia sessions. A single RTP session can support multiple participants sending at once, and can also support participants sending multiple simultaneous RTP streams. Examples of the latter might include a participant with multiple cameras who chooses to send multiple views of a scene, or a participant that sends audio and video flows multiplexed in a single RTP session. Rules for handling RTP sessions containing multiple RTP

streams are described in [RFC3550] with some clarifications in [I-D.ietf-avtcore-rtp-multi-stream].

An RTP endpoint will have one or more synchronisation sources (SSRCs). It will have at least one RTP Stream, and thus SSRC, for each media source it sends, and might use multiple SSRCs per media source when using media scalability features [RFC6190], forward error correction, RTP retransmission [RFC4588], or similar mechanisms. An endpoint that is not sending any RTP stream, will have at least one SSRC to use for reporting and any feedback messages. Each SSRC has to send RTCP sender reports corresponding to the RTP packets it sends, and receiver reports for traffic it receives. That is, every SSRC will send RTCP packets to report on every other SSRC. This rule is simple, but can be quite inefficient for endpoints that send large numbers of RTP streams in a single RTP session. Consider a session comprising ten participants, each sending three media sources, each with their own RTP stream. There will be 30 SSRCs in such an RTP session, and each of those 30 SSRCs will send an RTCP Sender Report/ Receiver Report packet (containing several report blocks) per reporting interval as each SSRC reports on all the others. However, the three SSRCs comprising each participant are commonly co-located such that they see identical reception quality. If there was a way to indicate that several SSRCs are co-located, and see the same reception quality, then two-thirds of those RTCP reports could be suppressed. This would allow the remaining RTCP reports to be sent more often, while keeping within the same RTCP bandwidth fraction.

This memo defines such an RTCP extension, RTCP Reporting Groups. This extension is used to indicate the SSRCs that originate from the same endpoint, and therefore have identical reception quality, hence allowing the endpoints to suppress unnecessary RTCP reception quality reports.

2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

3. RTCP Reporting Groups

An RTCP Reporting Group is a set of synchronization sources (SSRCs) that are co-located at a single endpoint (which could be an end host or a middlebox) in an RTP session. Since they are co-located, every SSRC in the RTCP reporting group will have an identical view of the network conditions, and see the same lost packets, jitter, etc. This allows a single representative to send RTCP reception quality reports

on behalf of the rest of the reporting group, reducing the number of RTCP packets that need to be sent without loss of information.

3.1. Semantics and Behaviour of RTCP Reporting Groups

A group of co-located SSRCs that see identical network conditions can form an RTCP reporting group. If reporting groups are in use, an RTP endpoint with multiple SSRCs MAY put those SSRCs into a reporting group if their view of the network is identical; i.e., if they report on traffic received at the same interface of an RTP endpoint. SSRCs with different views of the network MUST NOT be put into the same reporting group.

An endpoint that has combined its SSRCs into an RTCP reporting group will choose one (or a subset) of those SSRCs to act as "reporting source(s)" for that RTCP reporting group. A reporting source will send RTCP SR/RR reception quality reports on behalf of the other members of the RTCP reporting group. A reporting source MUST suppress the RTCP SR/RR reports that relate to other members of the reporting group, and only report on remote SSRCs. The other members (non reporting sources) of the RTCP reporting group will suppress their RTCP reception quality reports, and instead send an RTCP RGRS packet (see Section 3.2.2) to indicate that they are part of an RTCP reporting group and give the SSRCs of the reporting sources.

If there are large numbers of remote SSRCs in the RTP session, then the reception quality reports generated by the reporting source might grow too large to fit into a single compound RTCP packet, forcing the reporting source to use a round-robin policy to determine what remote SSRCs it includes in each compound RTCP packet, and so reducing the frequency of reports on each SSRC. To avoid this, in sessions with large numbers of remote SSRCs, an RTCP reporting group MAY use more than one reporting source. If several SSRCs are acting as reporting sources for an RTCP reporting group, then each reporting source MUST have non-overlapping sets of remote SSRCs it reports on.

An endpoint MUST NOT create an RTCP reporting group that comprises only a single local SSRC (i.e., an RTCP reporting group where the reporting source is the only member of the group), unless it is anticipated that the group might have additional SSRCs added to it in the future.

If a reporting source leaves the RTP session (i.e., if it sends a RTCP BYE packet, or leaves the session without sending BYE under the rules of [RFC3550] section 6.3.7), the remaining members of the RTCP reporting group MUST either (a) have another reporting source, if one exists, report on the remote SSRCs the leaving SSRC reported on, (b) choose a new reporting source, or (c) disband the RTCP reporting

group and begin sending reception quality reports following [RFC3550] and [I-D.ietf-avtcore-rtp-multi-stream].

The RTCP timing rules assign different bandwidth fractions to senders and receivers. This lets senders transmit RTCP reception quality reports more often than receivers. If a reporting source in an RTCP reporting group is a receiver, but one or more non-reporting SSRCs in the RTCP reporting group are senders, then the endpoint MAY treat the reporting source as a sender for the purpose of RTCP bandwidth allocation, increasing its RTCP bandwidth allocation, provided it also treats one of the senders as if it were a receiver and makes the corresponding reduction in RTCP bandwidth for that SSRC. However, the application needs to consider the impact on the frequency of transmitting of the synchronization information included in RTCP Sender Reports.

3.2. Identifying Members of an RTCP Reporting Group

When RTCP Reporting Groups are in use, the other SSRCs in the RTP session need to be able to identify which SSRCs are members of an RTCP reporting group. Two RTCP extensions are defined to support this: the RTCP RGRP SDES item is used by the reporting source(s) to identify an RTCP reporting group, and the RTCP RGRS packet is used by other members of an RTCP reporting group to identify the reporting source(s).

3.2.1. Definition and Use of the RTCP RGRP SDES Item

This document defines a new RTCP SDES item to identify an RTCP reporting group. The motivation for giving a reporting group an identify is to ensure that the RTCP reporting group and its member SSRCs can be correctly associated when there are multiple reporting sources, and to ensure that a reporting SSRC can be associated with the correct reporting group if an SSRC collision occurs.

This document defines the RTCP Source Description (SDES) RGRP item. The RTCP SDES RGRP item MUST be sent by the reporting sources in a reporting group, and MUST NOT be sent by other members of the reporting group or by SSRCs that are not members of any RTCP reporting group. Specifically, every reporting source in an RTCP reporting group MUST include an RTCP SDES packet containing an RGRP item in every compound RTCP packet in which it sends an RR or SR packet (i.e., in every RTCP packet it sends, unless Reduced-Size RTCP [RFC5506] is in use).

Syntactically, the format of the RTCP SDES RGRP item is identical to that of the RTCP SDES CNAME item [RFC7022], except that the SDES item type field MUST have value RGRP=(TBA) instead of CNAME=1. The value

of the RTCP SDES RGRP item MUST be chosen with the same concerns about global uniqueness and the same privacy considerations as the RTCP SDES CNAME. The value of the RTCP SDES RGRP item MUST be stable throughout the lifetime of the reporting group, even if some or all of the reporting sources change their SSRC due to collisions, or if the set of reporting sources changes.

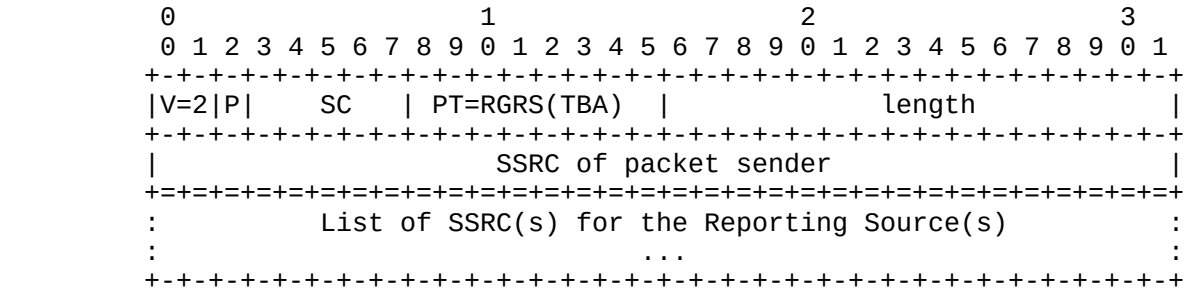
> Note to RFC Editor: please replace (TBA) in the above paragraph with the RTCP SDES item type number assigned to the RGRP item, then delete this note.

An RTP mixer or translator that forwards RTCP SR or RR packets from members of a reporting group MUST forward the corresponding RTCP SDES RGRP items as well, even if it otherwise strips SDES items other than the CNAME item.

3.2.2. Definition and Use of the RTCP RGRS Packet

A new RTCP packet type is defined to allow the members of an RTCP reporting group to identify the reporting sources for that group. This allows participants in an RTP session to distinguish an SSRC that is sending empty RTCP reception reports because it is a member of an RTCP reporting group, from an SSRC that is sending empty RTCP reception reports because it is not receiving any traffic. It also explicitly identifies the reporting sources, allowing other members of the RTP session to know which SSRCs are acting as the reporting sources for an RTCP reporting group, and allowing them to detect if RTCP packets from any of the reporting sources are being lost.

The format of the RTCP RGRS packet is defined below. It comprises the fixed RTCP header that indicates the packet type and length, the SSRC of the packet sender, and a list of reporting sources for the RTCP reporting group of which the packet sender is a member.



The fields in the RTCP RGRS packet have the following definition:

version (V): 2 bits unsigned integer. This field identifies the RTP version. The current RTP version is 2.

padding (P): 1 bit. If set, the padding bit indicates that the RTCP packet contains additional padding octets at the end that are not part of the control information but are included in the length field. See [RFC3550].

Source Count (SC): 5 bits unsigned integer. Indicates the number of reporting source SSRCs that are included in this RTCP packet. As the RTCP RGRS packet MUST NOT be not sent by reporting sources, all the SSRCs in the list of reporting sources will be different from the SSRC of the packet sender. Every RTCP RGRS packet MUST contain at least one reporting source SSRC.

Payload type (PT): 8 bits unsigned integer. The RTCP packet type number that identifies the packet as being an RTCP RGRS packet. The RGRS RTCP packet has the value [TBA].

> Note to RFC Editor: please replace [TBA] here, and in the packet format diagram above, with the RTCP packet type that IANA assigns to the RTCP RGRS packet.

Length: 16 bits unsigned integer. The length of this packet in 32-bit words minus one, including the header and any padding. This is in line with the definition of the length field used in RTCP sender and receiver reports [RFC3550]. Since all RTCP RGRS packets include at least one reporting source SSRC, the length will always be 2 or greater.

SSRC of packet sender: 32 bits. The SSRC of the sender of this packet.

List of SSRCs for the Reporting Source(s): A variable length size (as indicated by SC header field) of the 32 bit SSRC values of the reporting sources for the RTCP Reporting Group of which the packet sender is a member.

Every source that belongs to an RTCP reporting group but is not a reporting source MUST include an RTCP RGRS packet in every compound RTCP packet in which it sends an RR or SR packet (i.e., in every RTCP packet it sends, unless Reduced-Size RTCP [RFC5506] is in use). Each RTCP RGRS packet MUST contain the SSRC identifier of at least one reporting source. If there are more reporting sources in an RTCP reporting group than can fit into an RTCP RGRS packet, the members of that reporting group MUST send the SSRCs of the reporting sources in a round-robin fashion in consecutive RTCP RGRS packets, such that all

the SSRCs of the reporting sources are included over the course of several RTCP reporting intervals.

An RTP mixer or translator that forwards RTCP SR or RR packets from members of a reporting group MUST also forward the corresponding RGRS RTCP packets. If the RTP mixer or translator rewrites SSRC values of the packets it forwards, it MUST make the corresponding changes to the RTCP RGRS packets.

3.3. Interactions with the RTP/AVPF Feedback Profile

Use of the RTP/AVPF Feedback Profile [RFC4585] allows SSRCs to send rapid RTCP feedback requests and codec control messages. If use of the RTP/AVPF profile has been negotiated in an RTP session, members of an RTCP reporting group can send rapid RTCP feedback and codec control messages following [RFC4585] and [RFC5104], as updated by Section 5.4 of [I-D.ietf-avtcore-rtp-multi-stream], and by the following considerations.

The members of an RTCP reporting group will all see identical network conditions. Accordingly, one might therefore think that it doesn’t matter which SSRC in the reporting group sends the RTP/AVPF feedback or codec control messages. There might be, however, cases where the sender of the feedback/codec control message has semantic importance, or when only a subset of the members of an RTCP reporting group might want to send RTP/AVPF feedback or a codec control message in response to a particular event. For example, an RTP video sender might choose to treat packet loss feedback received from SSRCs known to be audio receivers with less urgency than feedback that it receives from video receivers when deciding what packets to retransmit, and a multimedia receiver using reporting groups might want to choose the outgoing SSRC for feedback packets to reflect this.

Each member of an RTCP reporting group SHOULD therefore send RTP/AVPF feedback/codec control messages independently of the other members of the reporting group, to respect the semantic meaning of the message sender. The suppression rules of [RFC4585] will ensure that only a single copy of each feedback packet is (typically) generated, even if several members of a reporting group send the same feedback. When an endpoint knows that several members of its RTCP reporting group will be sending identical feedback, and that the sender of the feedback is not semantically important, then that endpoint MAY choose to send all its feedback from the reporting source and deterministically suppress feedback packets generated by the other sources in the reporting group.

It is important to note that the RTP/AVPF timing rules operate on a per-SSRC basis. Using a single reporting source to send all feedback

for a reporting group will hence limit the amount of feedback that can be sent to that which can be sent by one SSRC. If this limit is a problem, then the reporting group can allow each of its members to send its own feedback, using its own SSRC.

If the RTP/AVPF feedback messages or codec control requests are sent as compound RTCP packets, then those compound RTCP packets MUST include either an RTCP RGRS packet or an RTCP SDES RGRP item, depending on whether they are sent by the reporting source or a non-reporting source in the RTCP reporting group respectively. The contents of non-compound RTCP feedback or codec control messages are not affected by the use of RTCP reporting groups.

3.4. Interactions with RTCP Extended Report (XR) Packets

When using RTCP Extended Reports (XR) [RFC3611] with RTCP reporting groups, it is RECOMMENDED that the reporting source is used to send the RTCP XR packets. If multiple reporting sources are in use, the reporting source that sends the SR/RR packets that relate to a particular remote SSRC SHOULD send the RTCP XR reports about that SSRC. This is motivated as one commonly combine the RTCP XR metrics with the regular report block to more fully understand the situation. Receiving these blocks in different compound packets reduces their value as the measuring intervals are not synchronized in those cases.

Some RTCP XR report blocks are specific to particular types of media, and might be relevant to only some members of a reporting group. For example, it would make no sense for an SSRC that is receiving video to send a VoIP metric RTCP XR report block. Such media specific RTCP XR report blocks MUST be sent by the SSRC to which they are relevant, and MUST NOT be included in the common report sent by the reporting source. This might mean that some SSRCs send RTCP XR packets in compound RTCP packets that contain an empty RTCP SR/RR packet, and that the time period covered by the RTCP XR packet is different to that covered by the RTCP SR/RR packet. If it is important that the RTCP XR packet and RTCP SR/RR packet cover the same time period, then that source SHOULD be removed from the RTCP reporting group, and send standard RTCP packets instead.

3.5. Middlebox Considerations

Many different types of middlebox are used with RTP. RTCP reporting groups are potentially relevant to those types of RTP middlebox that have their own SSRCs and generate RTCP reports for the traffic they receive. RTP middleboxes that do not have their own SSRC, and that don’t send RTCP reports on the traffic they receive, cannot use the RTCP reporting groups extension, since they generate no RTCP reports to group.

An RTP middlebox that has several SSRCs of its own can use the RTCP reporting groups extension to group the RTCP reports it generates. This can occur, for example, if a middlebox is acting as an RTP mixer for both audio and video flows that are multiplexed onto a single RTP session, where the middlebox has one SSRC for the audio mixer and one for the video mixer part, and when the middlebox wants to avoid cross reporting between audio and video.

A middlebox cannot use the RTCP reporting groups extension to group RTCP packets from the SSRCs that it is forwarding. It can, however, group the RTCP packets from the SSRCs it is forwarding into compound RTCP packets following the rules in Section 6.1 of [RFC3550] and Section 5.3 of [I-D.ietf-avtcore-rtp-multi-stream]. If the middlebox is using RTCP reporting groups for its own SSRCs, it MAY include RTCP packets from the SSRCs that it is forwarding as part of the compound RTCP packets its reporting source generates.

A middlebox that forwards RTCP SR or RR packets sent by members of a reporting group MUST forward the corresponding RTCP SDES RGRP items, as described in Section 3.2.1. A middlebox that forwards RTCP SR or RR packets sent by member of a reporting group MUST also forward the corresponding RTCP RGRS packets, as described in Section 3.2.2. Failure to forward these packets can cause compatibility problems, as described in Section 4.2.

If a middlebox rewrites SSRC values in the RTP and RTCP packets that it is forwarding, then it MUST make the corresponding changes in RTCP SDES packets containing RGRP items and in RTCP RGRS packets, to allow them to be associated with the rewritten SSRCs.

3.6. SDP Signalling for Reporting Groups

This document defines the "a=rtcp-rgrp" Session Description Protocol (SDP) [RFC4566] attribute to indicate if the session participant is capable of supporting RTCP Reporting Groups for applications that use SDP for configuration of RTP sessions. It is a property attribute, and hence takes no value. The multiplexing category [I-D.ietf-mmusic-sdp-mux-attributes] is IDENTICAL, as the functionality applies on RTP session level. A participant that proposes the use of RTCP Reporting Groups SHALL itself support the reception of RTCP Reporting Groups. The formal definition of this attribute is:

```
Name: rtcp-rgrp
Value:
Usage Level: session, media
Charset Dependent: no
Example:
   a=rtcp-rgrp
```

When using SDP Offer/Answer [RFC3264], the following procedures are to be used:

- Generating the initial SDP offer: If the offerer supports the RTCP reporting group extensions, and is willing to accept RTCP packets containing those extensions, then it MUST include an "a=rtcp-rgrp" attribute in the initial offer. If the offerer does not support RTCP reporting groups extensions, or is not willing to accept RTCP packets containing those extensions, then it MUST NOT include the "a=rtcp-rgrp" attribute in the offer.

- Generating the SDP answer: If the SDP offer contains an "a=rtcp-rgrp" attribute, and if the answerer supports RTCP reporting groups and is willing to receive RTCP packets using the RTCP reporting groups extensions, then the answerer MAY include an "a=rtcp-rgrp" attribute in the answer and MAY send RTCP packets containing the RTCP reporting groups extensions. If the offer does not contain an "a=rtcp-rgrp" attribute, or if the offer does contain such an attribute but the answerer does not wish to accept RTCP packets using the RTCP reporting groups extensions, then the answer MUST NOT include an "a=rtcp-rgrp" attribute.

- Offerer Processing of the SDP Answer: If the SDP answer contains an "a=rtcp-rgrp" attribute, and the corresponding offer also contained an "a=rtcp-rgrp" attribute, then the offerer MUST be prepared to accept and process RTCP packets that contain the reporting groups extension, and MAY send RTCP packets that contain the reporting groups extension. If the SDP answer contains an "a=rtcp-rgrp" attribute, but the corresponding offer did not contain the "a=rtcp-rgrp" attribute, then the offerer MUST reject the call. If the SDP answer does not contain an "a=rtcp-rgrp" attribute, then the offerer MUST NOT send packets containing the RTCP reporting groups extensions, and does not need to process packet containing the RTCP reporting groups extensions.

In declarative usage of SDP, such as the Real Time Streaming Protocol (RTSP) [RFC2326] and the Session Announcement Protocol (SAP) [RFC2974], the presence of the attribute indicates that the session participant MAY use RTCP Reporting Groups in its RTCP transmissions. An implementation that doesn’t explicitly support RTCP Reporting Groups MAY join a RTP session as long as it has been verified that

the implementation doesn't suffer from the problems discussed in Section 4.2.

4. Properties of RTCP Reporting Groups

This section provides additional information on what the resulting properties are with the design specified in Section 3. The content of this section is non-normative.

4.1. Bandwidth Benefits of RTCP Reporting Groups

To understand the benefits of RTCP reporting groups, consider a scenario in which the two endpoints in a session each have a hundred sources, of which eight each are sending within any given reporting interval.

For ease of analysis, we can make the simplifying approximation that the duration of the RTCP reporting interval is equal to the total size of the RTCP packets sent during an RTCP interval, divided by the RTCP bandwidth. (This will be approximately true in scenarios where the bandwidth is not so high that the minimum RTCP interval is reached.) For further simplification, we can assume RTCP senders are following the recommendations regarding Compound RTCP Packets in [I-D.ietf-avtcore-rtp-multi-stream]; thus, the per-packet transport-layer overhead will be small relative to the RTCP data. Thus, only the actual RTCP data itself need be considered.

In a report interval in this scenario, there will, as a baseline, be 200 SDES packets, 184 RR packets, and 16 SR packets. This amounts to approximately 6.5 kB of RTCP per report interval, assuming 16-byte CNAMEs and no other SDES information.

Using the original [RFC3550] everyone-reports-on-every-sender feedback rules, each of the 184 receivers will send 16 report blocks, and each of the 16 senders will send 15. This amounts to approximately 76 kB of report block traffic per interval; 92% of RTCP traffic consists of report blocks.

If reporting groups are used, however, there is only 0.4 kB of reports per interval, with no loss of useful information. Additionally, there will be (assuming 16-byte RGRPs, and a single reporting source per reporting group) an additional 2.4 kB per cycle of RGRP SDES items and RGRS packets. Put another way, the unmodified [RFC3550] reporting interval is approximately 9 times longer than if reporting groups are in use.

4.2. Compatibility of RTCP Reporting Groups

The RTCP traffic generated by receivers using RTCP Reporting Groups might appear, to observers unaware of these semantics, to be generated by receivers who are experiencing a network disconnection, as the non-reporting sources appear not to be receiving a given sender at all.

This could be a potentially critical problem for such a sender using RTCP for congestion control, as such a sender might think that it is sending so much traffic that it is causing complete congestion collapse.

However, such an interpretation of the session statistics would require a fairly sophisticated RTCP analysis. Any receiver of RTCP statistics which is just interested in information about itself needs to be prepared that any given reception report might not contain information about a specific media source, because reception reports in large conferences can be round-robined.

Thus, it is unclear to what extent such backward compatibility issues would actually cause trouble in practice.

5. Security Considerations

The security considerations of [RFC3550] and [I-D.ietf-avtcore-rtp-multi-stream] apply. If the RTP/AVPF profile is in use, then the security considerations of [RFC4585] (and [RFC5104], if used) also apply. If RTCP XR is used, the security consideration of [RFC3611] and any XR report blocks used also apply.

The RTCP SDES RGRP item is vulnerable to malicious modifications unless integrity protected is used. A modification of this item's length field cause the parsing of the RTCP packet in which it is contained to fail. Depending on the implementation, parsing of the full compound RTCP packet can also fail causing the whole packet to be discarded. A modification to the value of this SDES item would make the receiver of the report think that the sender of the report was a member of a different RTCP reporting group. This will potentially create an inconsistency, when the RGRS reports the source as being in the same reporting group as another source with another reporting group identifier. What impact on a receiver implementation such inconsistencies would have are difficult to fully predict. One case is when congestion control or other adaptation mechanisms are used, an inconsistent report can result in a media sender to reduce its bit-rate. However, a direct modification of the receiver report or a feedback message itself would be a more efficient attack, and equally costly to perform.

The new RGRS RTCP Packet type is very simple. The common RTCP packet type header shares the security risks with previous RTCP packet types. Errors or modification of the length field can cause the full compound packet to fail header validation (see Appendix A.2 in [RFC3550]) resulting in the whole compound RTCP packet being discarded. Modification of the SC or P fields would cause inconsistency when processing the RTCP packet, likely resulting it being classified as invalid. A modification of the PT field would cause the packet being interpreted under some other packet type's rules. In such case the result might be more or less predictable but packet type specific. Modification of the SSRC of packet sender would attribute this packet to another sender. Resulting in a receiver believing the reporting group applies also for this SSRC, if it exists. If it doesn't exist, unless also corresponding modifications are done on a SR/RR packet and a SDES packet the RTCP packet SHOULD be discarded. If consistent changes are done, that could be part of a resource exhaustion attack on a receiver implementation. Modification of the "List of SSRCs for the Reporting Source(s)" would change the SSRC the receiver expect to report on behalf of this SSRC. If that SSRC exist, that could potentially change the report group used for this SSRC. A change to another reporting group belonging to another endpoint is likely detectable as there would be a mismatch between the SSRC of the packet sender's endpoint information, transport addresses, SDES CNAME etc and the corresponding information from the reporting group indicated.

In general the reporting group is providing limited impacts attacks. The most significant result from an deliberate attack would be to cause the information to be discarded or be inconsistent, including discard of all RTCP packets that are modified. This causes a lack of information at any receiver entity, possibly disregarding the endpoints participation in the session.

To protect against this type of attacks from external non trusted entities, integrity and source authentication SHOULD be applied. This can be done, for example, by using SRTP [RFC3711] with appropriate key-management, other options exist as discussed in RTP Security Options [RFC7201].

The Report Group Identifier has a potential privacy impacting properties. If this would be generated by an implementation in such a way that is long term stable or predictable, it could be used for tracking a particular end-point. Therefore it is RECOMMENDED that it be generated as a short-term persistent RGRP, following the rules for short-term persistent CNAMEs in [RFC7022]. The rest of the information revealed, i.e. the SSRCs, the size of reporting group and the number of reporting sources in a reporting group is of less sensitive nature, considering that the SSRCs and the communication

would anyway be revealed without this extension. By encrypting the report group extensions the SSRC values would preserved confidential, but can still be revealed if SRTP [RFC3711] is used. The size of the reporting groups and number of reporting sources are likely determinable from analysis of the packet pattern and sizes. However, this information appears to have limited value.

6. IANA Considerations

(Note to the RFC-Editor: in the following, please replace "TBA" with the IANA-assigned value, and "XXXX" with the number of this document, then delete this note)

The IANA is requested to register one new RTCP SDES items in the "RTCP SDES Item Types" registry, as follows:

Value	Abbrev	Name	Reference
TBA	RGRP	Reporting Group Identifier	[RFCXXXX]

The definition of the RTCP SDES RGRP item is given in Section 3.2.1 of this memo.

The IANA is also requested to register one new RTCP packet type in the "RTCP Control Packet Types (PT)" Registry as follows:

Value	Abbrev	Name	Reference
TBA	RGRS	Reporting Group Reporting Sources	[RFCXXXX]

The definition of the RTCP RGRS packet type is given in Section 3.2.2 of this memo.

The IANA is also requested to register one new SDP attribute:

SDP Attribute ("att-field"):

Attribute name:	rtcp-rgrp
Long form:	RTCP Reporting Groups
Type of name:	att-field
Type of attribute:	Media or session level
Subject to charset:	No
Purpose:	Negotiate or configure the use of the RTCP Reporting Group Extension.
Reference:	[RFCXXXX]
Values:	None

The definition of the "a=rtcp-rgrp" SDES attribute is given in Section 3.6 of this memo.

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Frame Marking RTP Header Extension
draft-ietf-avtext-framemarking-08

Abstract

This document describes a Frame Marking RTP header extension used to convey information about video frames that is critical for error recovery and packet forwarding in RTP middleboxes or network nodes. It is most useful when media is encrypted, and essential when the middlebox or node has no access to the media decryption keys. It is also useful for codec-agnostic processing of encrypted or unencrypted media, while it also supports extensions for codec-specific information.

Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on April 26, 2019.

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Table of Contents

1. Introduction

Many widely deployed RTP [RFC3550] topologies [RFC7667] used in modern voice and video conferencing systems include a centralized component that acts as an RTP switch. It receives voice and video streams from each participant, which may be encrypted using SRTP [RFC3711], or extensions that provide participants with private media [I-D.ietf-perc-private-media-framework] via end-to-end encryption where the switch has no access to media decryption keys. The goal is to provide a set of streams back to the participants which enable them to render the right media content. In a simple video configuration, for example, the goal will be that each participant sees and hears just the active speaker. In that case, the goal of the switch is to receive the voice and video streams from each participant, determine the active speaker based on energy in the voice packets, possibly using the client-to-mixer audio level RTP header extension [RFC6464], and select the corresponding video stream for transmission to participants; see Figure 1.

In this document, an "RTP switch" is used as a common short term for the terms "switching RTP mixer", "source projecting middlebox", "source forwarding unit/middlebox" and "video switching MCU" as discussed in [RFC7667].

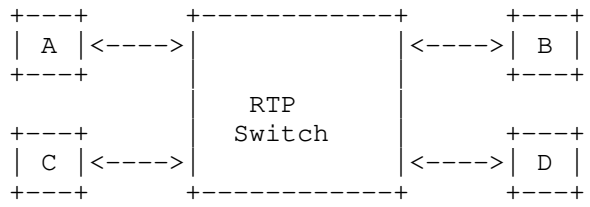


Figure 1: RTP switch

In order to properly support switching of video streams, the RTP switch typically needs some critical information about video frames in order to start and stop forwarding streams.

- Because of inter-frame dependencies, it should ideally switch video streams at a point where the first frame from the new speaker can be decoded by recipients without prior frames, e.g switch on an intra-frame.
- In many cases, the switch may need to drop frames in order to realize congestion control techniques, and needs to know which frames can be dropped with minimal impact to video quality.
- Furthermore, it is highly desirable to do this in a payload format-agnostic way which is not specific to each different video codec. Most modern video codecs share common concepts around frame types and other critical information to make this codec-agnostic handling possible.
- It is also desirable to be able to do this for SRTP without requiring the video switch to decrypt the packets. SRTP will encrypt the RTP payload format contents and consequently this data is not usable for the switching function without decryption, which may not even be possible in the case of end-to-end encryption of private media [I-D.ietf-perc-private-media-framework].

By providing meta-information about the RTP streams outside the encrypted media payload, an RTP switch can do codec-agnostic selective forwarding without decrypting the payload. This document specifies the necessary meta-information in an RTP header extension.

2. Key Words for Normative Requirements

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

3. Frame Marking RTP Header Extension

This specification uses RTP header extensions as defined in [RFC8285]. A subset of meta-information from the video stream is provided as an RTP header extension to allow an RTP switch to do generic selective forwarding of video streams encoded with potentially different video codecs.

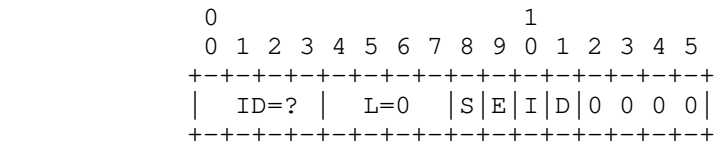
The Frame Marking RTP header extension is encoded using the one-byte header or two-byte header as described in [RFC8285]. The one-byte header format is used for examples in this memo. The two-byte header format is used when other two-byte header extensions are present in the same RTP packet, since mixing one-byte and two-byte extensions is not possible in the same RTP packet.

This extension is only specified for Source (not Redundancy) RTP Streams [RFC7656] that carry video payloads. It is not specified for audio payloads, nor is it specified for Redundancy RTP Streams. The (separate) specifications for Redundancy RTP Streams often include provisions for recovering any header extensions that were part of the original source packet. Such provisions SHALL be followed to recover the Frame Marking RTP header extension of the original source packet. Source packet frame markings may be useful when generating Redundancy RTP Streams; for example, the I and D bits can be used to generate extra or no redundancy, respectively, and redundancy schemes with source blocks can align source block boundaries with Independent frame boundaries as marked by the I bit.

A frame, in the context of this specification, is the set of RTP packets with the same RTP timestamp from a specific RTP synchronization source (SSRC).

3.1. Short Extension for Non-Scalable Streams

The following RTP header extension is RECOMMENDED for non-scalable streams. It MAY also be used for scalable streams if the sender has limited or no information about stream scalability. The ID is assigned per [RFC8285], and the length is encoded as L=0 which indicates 1 octet of data.

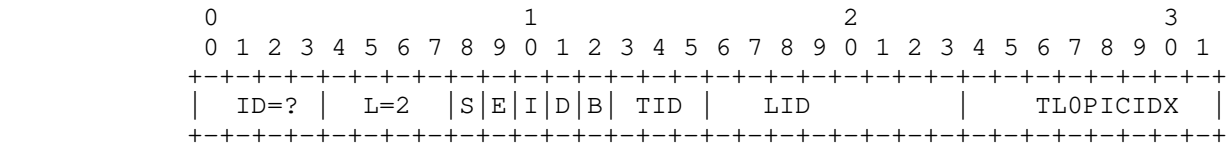


The following information are extracted from the media payload and sent in the Frame Marking RTP header extension.

- S: Start of Frame (1 bit) - MUST be 1 in the first packet in a frame; otherwise MUST be 0.
- E: End of Frame (1 bit) - MUST be 1 in the last packet in a frame; otherwise MUST be 0. Note that this SHOULD match the RTP header marker bit when the latter is reliable.
- I: Independent Frame (1 bit) - MUST be 1 for frames that can be decoded independent of temporally prior frames, e.g. intra-frame, VPX keyframe, H.264 IDR [RFC6184], H.265 IDR/CRA/BLA/RAP [RFC7798]; otherwise MUST be 0.
- D: Discardable Frame (1 bit) - MUST be 1 for frames the sender knows can be discarded, and still provide a decodable media stream; otherwise MUST be 0.
- The remaining (4 bits) - are reserved for future use for non-scalable streams; they MUST be set to 0 upon transmission and ignored upon reception.

3.2. Long Extension for Scalable Streams

The following RTP header extension is RECOMMENDED for scalable streams. It MAY also be used for non-scalable streams, in which case TID, LID and TL0PICIDX MUST be 0. The ID is assigned per [RFC8285], and the length is encoded as L=2 which indicates 3 octets of data.



The following information are extracted from the media payload and sent in the Frame Marking RTP header extension.

- S: Start of Frame (1 bit) - MUST be 1 in the first packet in a frame within a layer; otherwise MUST be 0.

- E: End of Frame (1 bit) - MUST be 1 in the last packet in a frame within a layer; otherwise MUST be 0. Note that the RTP header marker bit MAY be used to infer the last packet of the highest enhancement layer.
- I: Independent Frame (1 bit) - MUST be 1 for frames that can be decoded independent of temporally prior frames, e.g. intra-frame, VPX keyframe, H.264 IDR [RFC6184], H.265 IDR/CRA/BLA/RAP [RFC7798]; otherwise MUST be 0. Note that this bit only signals temporal independence, so it can be 1 in spatial or quality enhancement layers that depend on temporally co-located layers but not temporally prior frames.
- D: Discardable Frame (1 bit) - MUST be 1 for frames the sender knows can be discarded, and still provide a decodable media stream; otherwise MUST be 0.
- B: Base Layer Sync (1 bit) - MUST be 1 if the sender knows this frame only depends on the base temporal layer; otherwise MUST be 0. If no scalability is used, this MUST be 0.
- TID: Temporal ID (3 bits) - The base temporal layer starts with 0, and increases with 1 for each higher temporal layer/sub-layer. If no scalability is used, this MUST be 0.
- LID: Layer ID (8 bits) - Identifies the spatial and quality layer encoded, starting with 0 and increasing with higher fidelity. If no scalability is used, this MUST be 0 or omitted to reduce length. When omitted, TL0PICIDX MUST also be omitted.
- TL0PICIDX: Temporal Layer 0 Picture Index (8 bits) - Running index of base temporal layer 0 frames when TID is 0. When TID is not 0, this indicates a dependency on the given index. If no scalability is used, or the running index is unknown, this MUST be omitted to reduce length. Note that 0 is a valid running index value for TL0PICIDX.

The layer information contained in TID and LID convey useful aspects of the layer structure that can be utilized in selective forwarding. Without further information about the layer structure, these identifiers can only be used for relative priority of layers. They convey a layer hierarchy with TID=0 and LID=0 identifying the base layer. Higher values of TID identify higher temporal layers with higher frame rates. Higher values of LID identify higher spatial and/or quality layers with higher resolutions and/or bitrates.

With further information, for example, possible future RTCP SDES items that convey full layer structure information, it may be possible to map these TIDs and LIDs to specific frame rates, resolutions and bitrates. Such additional layer information may be useful for forwarding decisions in the RTP switch, but is beyond the scope of this memo. The relative layer information is still useful for many selective forwarding decisions even without such additional layer information.

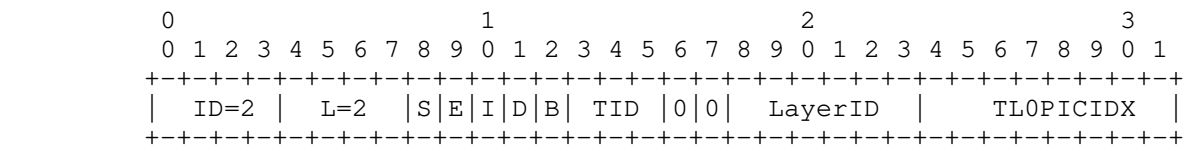
3.2.1. Layer ID Mappings for Scalable Streams

3.2.1.1. H265 LID Mapping

The following shows the H265 [RFC7798] LayerID (6 bits) and TID (3 bits) from the NAL unit header mapped to the generic LID and TID fields.

The I bit MUST be 1 when the NAL unit type is 16-23 (inclusive), otherwise it MUST be 0.

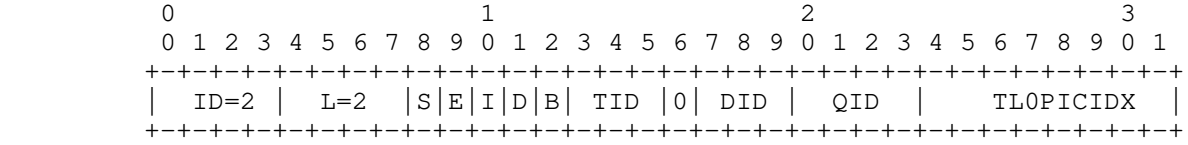
The S and E bits MUST match the corresponding bits in PACI:PHES:TSCI payload structures.



3.2.1.2. H264-SVC LID Mapping

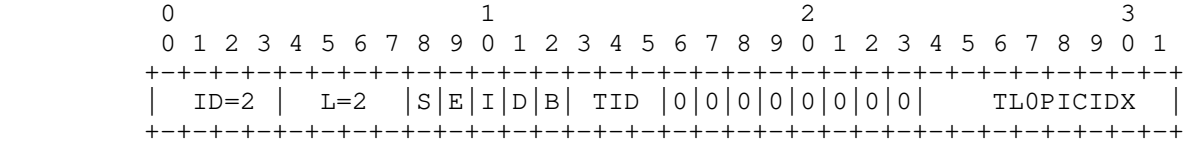
The following shows H264-SVC [RFC6190] Layer encoding information (3 bits for spatial/dependency layer, 4 bits for quality layer and 3 bits for temporal layer) mapped to the generic LID and TID fields.

The S, E, I and D bits MUST match the corresponding bits in PACSI payload structures.



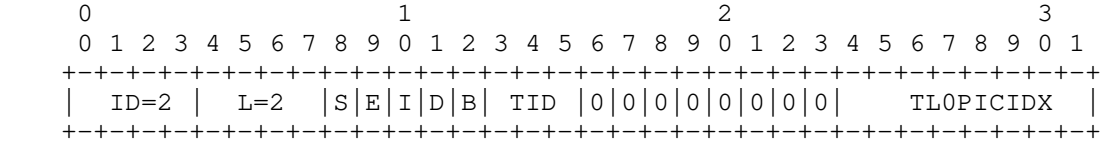
3.2.1.3. H264 (AVC) LID Mapping

The following shows the header extension for H264 (AVC) [RFC6184] that contains only temporal layer information.



3.2.1.4. VP8 LID Mapping

The following shows the header extension for VP8 [RFC7741] that contains only temporal layer information.



3.2.1.5. Future Codec LID Mapping

The RTP payload format specification for future video codecs SHOULD include a section describing the LID mapping and TID mapping for the codec. For example, the LID/TID mapping for the VP9 codec is described in the VP9 RTP Payload Format [I-D.ietf-payload-vp9].

3.3. Signaling Information

The URI for declaring this header extension in an extmap attribute is "urn:ietf:params:rtp-hdrext:framemarking". It does not contain any extension attributes.

An example attribute line in SDP:

```
a=extmap:3 urn:ietf:params:rtp-hdrext:framemarking
```

3.4. Usage Considerations

The header extension values MUST represent what is already in the RTP payload.

When an RTP switch needs to discard a received video frame due to congestion control considerations, it is RECOMMENDED that it preferably drop frames marked with the D (Discardable) bit set, or the highest values of TID and LID, which indicate the highest temporal and spatial/quality enhancement layers, since those typically have fewer dependenices on them than lower layers.

When an RTP switch wants to forward a new video stream to a receiver, it is RECOMMENDED to select the new video stream from the first switching point with the I (Independent) bit set in all spatial layers and forward the same. An RTP switch can request a media source to generate a switching point by sending Full Intra Request (RTCP FIR) as defined in [RFC5104], for example.

3.4.1. Relation to Layer Refresh Request (LRR)

Receivers can use the Layer Refresh Request (LRR) [I-D.ietf-avtext-lrr] RTCP feedback message to upgrade to a higher layer in scalable encodings. The TID/LID values and formats used in LRR messages MUST correspond to the same values and formats specified in Section 3.2.

Because frame marking can only be used with temporally-nested streams, temporal-layer LRR refreshes are unnecessary for frame-marked streams. Other refreshes can be detected based on the I bit being set for the specific spatial layers.

3.4.2. Scalability Structures

The LID and TID information is most useful for fixed scalability structures, such as nested hierarchical temporal layering structures, where each temporal layer only references lower temporal layers or the base temporal layer. The LID and TID information is less useful, or even not useful at all, for complex, irregular scalability structures that do not conform to common, fixed patterns of inter-layer dependencies and referencing structures. Therefore it is RECOMMENDED to use LID and TID information for RTP switch forwarding decisions only in the case of temporally nested scalability structures, and it is NOT RECOMMENDED for other (more complex or irregular) scalability structures.

4. Security Considerations

In the Secure Real-Time Transport Protocol (SRTP) [RFC3711], RTP header extensions are authenticated but usually not encrypted. When header extensions are used some of the payload type information are exposed and visible to middle boxes. The encrypted media data is not exposed, so this is not seen as a high risk exposure.

5. Acknowledgements

Many thanks to Bernard Aboba, Jonathan Lennox, and Stephan Wenger for their inputs.

6. IANA Considerations

This document defines a new extension URI to the RTP Compact HeaderExtensions sub-registry of the Real-Time Transport Protocol (RTP) Parameters registry, according to the following data:

Extension URI: urn:ietf:params:rtp-hdrext:framemarkinginfo
Description: Frame marking information for video streams

Contact: mzanaty@cisco.com
Reference: RFC XXXX

Note to RFC Editor: please replace RFC XXXX with the number of this RFC.

7. References

7.1. Normative References

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[RFC6190] Wenger, S., Wang, Y., Schierl, T., and A. Eleftheriadis, "RTP Payload Format for Scalable Video Coding", RFC 6190, DOI 10.17487/RFC6190, May 2011, <https://www.rfc-editor.org/info/rfc6190>.

[RFC7741] Westin, P., Lundin, H., Glover, M., Uberti, J., and F. Galligan, "RTP Payload Format for VP8 Video", RFC 7741, DOI 10.17487/RFC7741, March 2016, <https://www.rfc-editor.org/info/rfc7741>.

[RFC7798] Wang, Y., Sanchez, Y., Schierl, T., Wenger, S., and M. Hannuksela, "RTP Payload Format for High Efficiency Video Coding (HEVC)", RFC 7798, DOI 10.17487/RFC7798, March 2016, <https://www.rfc-editor.org/info/rfc7798>.

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7.2. Informative References

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Payload Working Group
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June 29, 2017

The Layer Refresh Request (LRR) RTCP Feedback Message
draft-ietf-avtext-lrr-07

Abstract

This memo describes the RTCP Payload-Specific Feedback Message "Layer Refresh Request" (LRR), which can be used to request a state refresh of one or more substreams of a layered media stream. It also defines its use with several RTP payloads for scalable media formats.

Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

Internet-Drafts are working documents of the Internet Engineering Task Force (IETF). Note that other groups may also distribute working documents as Internet-Drafts. The list of current Internet-Drafts is at http://datatracker.ietf.org/drafts/current/.

Internet-Drafts are draft documents valid for a maximum of six months and may be updated, replaced, or obsoleted by other documents at any time. It is inappropriate to use Internet-Drafts as reference material or to cite them other than as "work in progress."

This Internet-Draft will expire on December 31, 2017.

Table of Contents

1. Introduction

This memo describes an RTCP [RFC3550] Payload-Specific Feedback Message [RFC4585] "Layer Refresh Request" (LRR). It is designed to allow a receiver of a layered media stream to request that one or more of its substreams be refreshed, such that it can then be decoded by an endpoint which previously was not receiving those layers, without requiring that the entire stream be refreshed (as it would be if the receiver sent a Full Intra Request (FIR); [RFC5104] see also [RFC8082]).

The feedback message is applicable both to temporally and spatially scaled streams, and to both single-stream and multi-stream scalability modes.

2. Conventions, Definitions and Acronyms

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

2.1. Terminology

A "Layer Refresh Point" is a point in a scalable stream after which a decoder, which previously had been able to decode only some (possibly none) of the available layers of stream, is able to decode a greater number of the layers.

For spatial (or quality) layers, in normal encoding, a subpicture can depend both on earlier pictures of that spatial layer and also on lower-layer pictures of the current picture. A layer refresh, however, typically requires that a spatial layer picture be encoded in a way that references only the lower-layer subpictures of the current picture, not any earlier pictures of that spatial layer. Additionally, the encoder must promise that no earlier pictures of that spatial layer will be used as reference in the future.

However, even in a layer refresh, layers other than the ones being refreshed may still maintain dependency on earlier content of the stream. This is the difference between a layer refresh and a Full Intra Request [RFC5104]. This minimizes the coding overhead of refresh to only those parts of the stream that actually need to be refreshed at any given time.

An illustration of spatial layer refresh of an enhancement layer is shown below. <-- indicates a coding dependency.

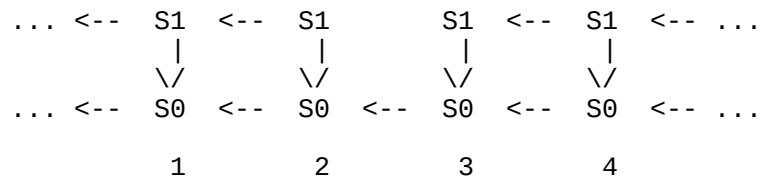

Figure 1

In Figure 1, frame 3 is a layer refresh point for spatial layer S1; a decoder which had previously only been decoding spatial layer S0 would be able to decode layer S1 starting at frame 3.

An illustration of spatial layer refresh of a base layer is shown below. <-- indicates a coding dependency.

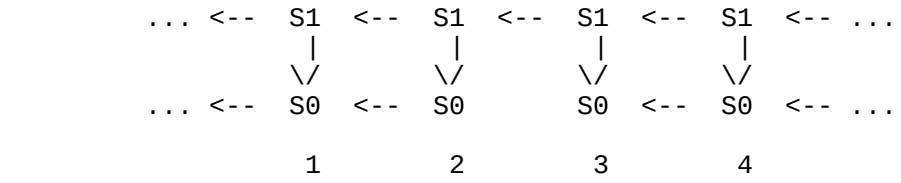

Figure 2

In Figure 2, frame 3 is a layer refresh point for spatial layer S0; a decoder which had previously not been decoding the stream at all could decode layer S0 starting at frame 3.

For temporal layers, while normal encoding allows frames to depend on earlier frames of the same temporal layer, layer refresh requires that the layer be "temporally nested", i.e. use as reference only earlier frames of a lower temporal layer, not any earlier frames of this temporal layer, and also promise that no future frames of this temporal layer will reference frames of this temporal layer before the refresh point. In many cases, the temporal structure of the stream will mean that all frames are temporally nested, in which case decoders will have no need to send LRR messages for the stream.

An illustration of temporal layer refresh is shown below. <-- indicates a coding dependency.

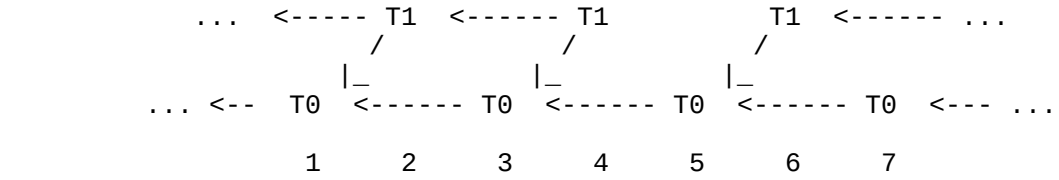

Figure 3

In Figure 3, frame 6 is a layer refresh point for temporal layer T1; a decoder which had previously only been decoding temporal layer T0 would be able to decode layer T1 starting at frame 6.

An illustration of an inherently temporally nested stream is shown below. <-- indicates a coding dependency.

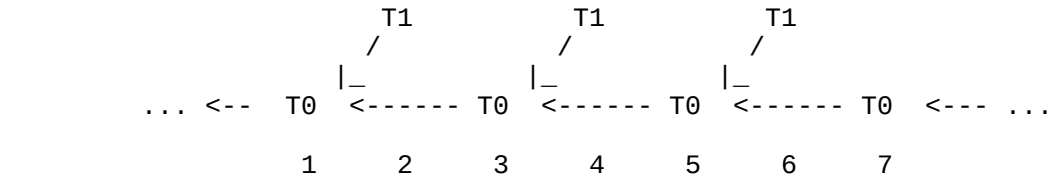

Figure 4

In Figure 4, the stream is temporally nested in its ordinary structure; a decoder receiving layer T0 can begin decoding layer T1 at any point.

A "Layer Index" is a numeric label for a specific spatial and temporal layer of a scalable stream. It consists of the pair of a "temporal ID" identifying the temporal layer, and a "layer ID" identifying the spatial or quality layer. The details of how layers of a scalable stream are labeled are codec-specific. Details for several codecs are defined in Section 4.

3. Layer Refresh Request

A layer refresh frame can be requested by sending a Layer Refresh Request (LRR), which is an RTP Control Protocol (RTCP) [RFC3550] payload-specific feedback message [RFC4585] asking the encoder to encode a frame which makes it possible to upgrade to a higher layer. The LRR contains one or two tuples, indicating the temporal and spatial layer the decoder wants to upgrade to, and (optionally) the currently highest temporal and spatial layer the decoder can decode.

The specific format of the tuples, and the mechanism by which a receiver recognizes a refresh frame, is codec-dependent. Usage for several codecs is discussed in Section 4.

LRR follows the model of the Full Intra Request (FIR) [RFC5104] (Section 3.5.1) for its retransmission, reliability, and use in multipoint conferences.

The LRR message is identified by RTCP packet type value PT=PSFB and FMT=TBD. The FCI field MUST contain one or more LRR entries. Each entry applies to a different media sender, identified by its SSRC.

[NOTE TO RFC Editor: Please replace "TBD" with the IANA-assigned payload-specific feedback number.]

3.1. Message Format

The Feedback Control Information (FCI) for the Layer Refresh Request consists of one or more FCI entries, the content of which is depicted in Figure 5. The length of the LRR feedback message MUST be set to 2+3*N 32-bit words, where N is the number of FCI entries.

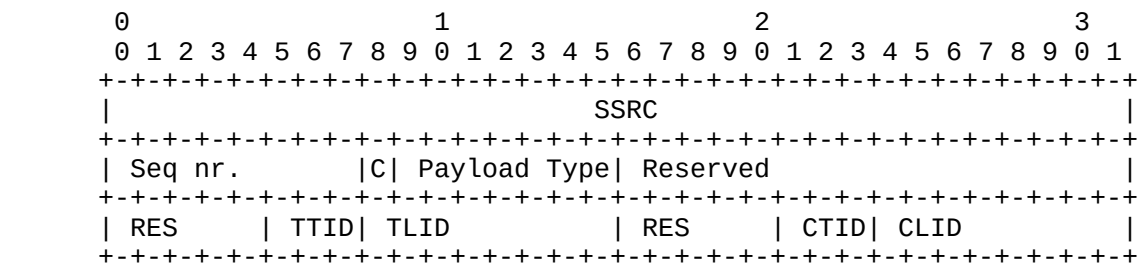

Figure 5

SSRC (32 bits) The SSRC value of the media sender that is requested to send a layer refresh point.

Seq nr. (8 bits) Command sequence number. The sequence number space is unique for each pairing of the SSRC of command source and the SSRC of the command target. The sequence number SHALL be increased by 1 for each new command (modulo 256, so the value after 255 is 0). A repetition SHALL NOT increase the sequence number. The initial value is arbitrary.

C (1 bit) A flag bit indicating whether the "Current Temporal Layer ID (CTID)" and "Current Layer ID (CLID)" fields are present in the FCI. If this bit is 0, the sender of the LRR message is requesting refresh of all layers up to and including the target layer.

Payload Type (7 bits) The RTP payload type for which the LRR is being requested. This gives the context in which the target layer index is to be interpreted.

Reserved (RES) (three separate fields, 16 bits / 5 bits / 5 bits) All bits SHALL be set to 0 by the sender and SHALL be ignored on reception.

Target Temporal Layer ID (TTID) (3 bits) The temporal ID of the target layer for which the receiver wishes a refresh point.

Target Layer ID (TLID) (8 bits) The layer ID of the target spatial or quality layer for which the receiver wishes a refresh point. Its format is dependent on the payload type field.

Current Temporal Layer ID (CTID) (3 bits) If C is 1, the ID of the current temporal layer being decoded by the receiver. This message is not requesting refresh of layers at or below this layer. If C is 0, this field SHALL be set to 0 by the sender and SHALL be ignored on reception.

Current Layer ID (CLID) (8 bits) If C is 1, the layer ID of the current spatial or quality layer being decoded by the receiver. This message is not requesting refresh of layers at or below this layer. If C is 0, this field SHALL be set to 0 by the sender and SHALL be ignored on reception.

When C is 1, TTID MUST NOT be less than CTID, and TLID MUST NOT be less than CLID; at least one of TTID or TLID MUST be greater than CTID or CLID respectively. That is to say, the target layer index <TTID, TLID> MUST be a layer upgrade from the current layer index <CTID, CLID>. A sender MAY request an upgrade in both temporal and spatial/quality layers simultaneously.

A receiver receiving an LRR feedback packet which does not satisfy the requirements of the previous paragraph, i.e. one where the C bit is present but TTID is less than CTID or TLID is less than CLID, MUST discard the request.

Note: the syntax of the TTID, TLID, CTID, and CLID fields match, by design, the TID and LID fields in [I-D.ietf-avtext-framemarking].

3.2. Semantics

Within the common packet header for feedback messages (as defined in section 6.1 of [RFC4585]), the "SSRC of packet sender" field indicates the source of the request, and the "SSRC of media source" is not used and SHALL be set to 0. The SSRCs of the media senders to which the LRR command applies are in the corresponding FCI entries. A LRR message MAY contain requests to multiple media senders, using one FCI entry per target media sender.

Upon reception of LRR, the encoder MUST send a decoder refresh point (see Section 2.1) as soon as possible.

The sender MUST respect bandwidth limits provided by the application of congestion control, as described in Section 5 of [RFC5104]. As layer refresh points will often be larger than non-refreshing frames,

this may restrict a sender's ability to send a layer refresh point quickly.

LRR MUST NOT be sent as a reaction to picture losses due to packet loss or corruption -- it is RECOMMENDED to use PLI [RFC4585] instead. LRR SHOULD be used only in situations where there is an explicit change in decoders' behavior, for example when a receiver will start decoding a layer which it previously had been discarding.

4. Usage with specific codecs

In order for LRR to be used with a scalable codec, the format of the temporal and layer ID fields (for both the target and current layer indices) needs to be specified for that codec's RTP packetization. New RTP packetization specifications for scalable codecs SHOULD define how this is done. (The VP9 payload [I-D.ietf-payload-vp9], for instance, has done so.) If the payload also specifies how it is used with the Frame Marking RTP Header Extension [I-D.ietf-avtext-framemarking], the syntax MUST be defined in the same manner as the TID and LID fields in that header.

4.1. H264 SVC

H.264 SVC [RFC6190] defines temporal, dependency (spatial), and quality scalability modes.

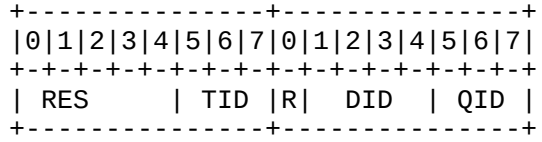

Figure 6

Figure 6 shows the format of the layer index fields for H.264 SVC streams. The "R" and "RES" fields MUST be set to 0 on transmission and ignored on reception. See [RFC6190] Section 1.1.3 for details on the DID, QID, and TID fields.

A dependency or quality layer refresh of a given layer in H.264 SVC can be identified by the "I" bit (idr_flag) in the extended NAL unit header, present in NAL unit types 14 (prefix NAL unit) and 20 (coded scalable slice). Layer refresh of the base layer can also be identified by its NAL unit type of its coded slices, which is "5" rather than "1". A dependency or quality layer refresh is complete once this bit has been seen on all the appropriate layers (in decoding order) above the current layer index (if any, or beginning from the base layer if not) through the target layer index.

Note that as the "I" bit in a PACSI header is set if the corresponding bit is set in any of the aggregated NAL units it describes; thus, it is not sufficient to identify layer refresh when NAL units of multiple dependency or quality layers are aggregated.

In H.264 SVC, temporal layer refresh information can be determined from various Supplemental Encoding Information (SEI) messages in the bitstream.

Whether an H.264 SVC stream is scalably nested can be determined from the Scalability Information SEI message's temporal_id_nesting flag. If this flag is set in a stream's currently applicable Scalability Information SEI, receivers SHOULD NOT send temporal LRR messages for that stream, as every frame is implicitly a temporal layer refresh point. (The Scalability Information SEI message may also be available in the signaling negotiation of H.264 SVC, as the sprop-scalability-info parameter.)

If a stream's temporal_id_nesting flag is not set, the Temporal Level Switching Point SEI message identifies temporal layer switching points. A temporal layer refresh is satisfied when this SEI message is present in a frame with the target layer index, if the message's delta_frame_num refers to a frame with the requested current layer index. (Alternately, temporal layer refresh can also be satisfied by a complete state refresh, such as an IDR.) Senders which support receiving LRR for non-temporally-nested streams MUST insert Temporal Level Switching Point SEI messages as appropriate.

4.2. VP8

The VP8 RTP payload format [RFC7741] defines temporal scalability modes. It does not support spatial scalability.

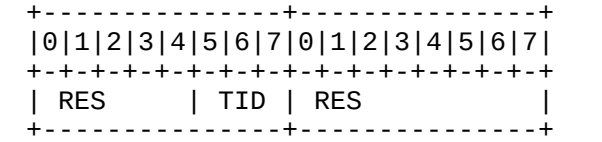

Figure 7

Figure 7 shows the format of the layer index field for VP8 streams. The "RES" fields MUST be set to 0 on transmission and be ignored on reception. See [RFC7741] Section 4.2 for details on the TID field.

A VP8 layer refresh point can be identified by the presence of the "Y" bit in the VP8 payload header. When this bit is set, this and all subsequent frames depend only on the current base temporal layer.

On receipt of an LRR for a VP8 stream, A sender which supports LRR MUST encode the stream so it can set the Y bit in a packet whose temporal layer is at or below the target layer index.

Note that in VP8, not every layer switch point can be identified by the Y bit, since the Y bit implies layer switch of all layers, not just the layer in which it is sent. Thus the use of LRR with VP8 can result in some inefficiency in transmision. However, this is not expected to be a major issue for temporal structures in normal use.

4.3. H265

The initial version of the H.265 payload format [RFC7798] defines temporal scalability, with protocol elements reserved for spatial or other scalability modes (which are expected to be defined in a future version of the specification).

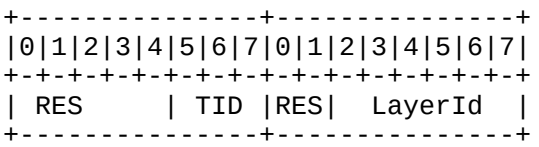

Figure 8

Figure 8 shows the format of the layer index field for H.265 streams. The "RES" fields MUST be set to 0 on transmission and ignored on reception. See [RFC7798] Section 1.1.4 for details on the LayerId and TID fields.

H.265 streams signal whether they are temporally nested, using the vps_temporal_id_nesting_flag in the Video Parameter Set (VPS), and the sps_temporal_id_nesting_flag in the Sequence Parameter Set (SPS). If this flag is set in a stream’s currently applicable VPS or SPS, receivers SHOULD NOT send temporal LRR messages for that stream, as every frame is implicitly a temporal layer refresh point.

If a stream’s sps_temporal_id_nesting_flag is not set, the NAL unit types 2 to 5 inclusively identify temporal layer switching points. A layer refresh to any higher target temporal layer is satisfied when a NAL unit type of 4 or 5 with TID equal to 1 more than current TID is seen. Alternatively, layer refresh to a target temporal layer can be incrementally satisfied with NAL unit type of 2 or 3. In this case, given current TID = TO and target TID = TN, layer refresh to TN is satisfied when NAL unit type of 2 or 3 is seen for TID = T1, then TID = T2, all the way up to TID = TN. During this incremental process, layer refresh to TN can be completely satisfied as soon as a NAL unit type of 2 or 3 is seen.

Of course, temporal layer refresh can also be satisfied whenever any Intra Random Access Point (IRAP) NAL unit type (with values 16-23, inclusively) is seen. An IRAP picture is similar to an IDR picture in H.264 (NAL unit type of 5 in H.264) where decoding of the picture can start without any older pictures.

In the (future) H.265 payloads that support spatial scalability, a spatial layer refresh of a specific layer can be identified by NAL units with the requested layer ID and NAL unit types between 16 and 21 inclusive. A dependency or quality layer refresh is complete once NAL units of this type have been seen on all the appropriate layers (in decoding order) above the current layer index (if any, or beginning from the base layer if not) through the target layer index.

5. Usage with different scalability transmission mechanisms

Several different mechanisms are defined for how scalable streams can be transmitted in RTP. The RTP Taxonomy [RFC7656] Section 3.7 defines three mechanisms: Single RTP Stream on a Single Media Transport (SRST), Multiple RTP Streams on a Single Media Transport (MRST), and Multiple RTP Streams on Multiple Media Transports (MRMT).

The LRR message is applicable to all these mechanisms. For MRST and MRMT mechanisms, the "media source" field of the LRR FCI is set to the SSRC of the RTP stream containing the layer indicated by the Current Layer Index (if "C" is 1), or the stream containing the base encoded stream (if "C" is 0). For MRMT, it is sent on the RTP session on which this stream is sent. On receipt, the sender MUST refresh all the layers requested in the stream, simultaneously in decode order.

6. SDP Definitions

Section 7 of [RFC5104] defines SDP procedures for indicating and negotiating support for codec control messages (CCM) in SDP. This document extends this with a new codec control command, "lrr", which indicates support of the Layer Refresh Request (LRR).

Figure 9 gives a formal Augmented Backus-Naur Form (ABNF) [RFC5234] showing this grammar extension, extending the grammar defined in [RFC5104].

```
rtcp-fb-ccm-param =/ SP "lrr"    ; Layer Refresh Request
```

Figure 9: Syntax of the "lrr" ccm

The Offer-Answer considerations defined in [RFC5104] Section 7.2 apply.

7. Security Considerations

All the security considerations of FIR feedback packets [RFC5104] apply to LRR feedback packets as well. Additionally, media senders receiving LRR feedback packets MUST validate that the payload types and layer indices they are receiving are valid for the stream they are currently sending, and discard the requests if not.

8. IANA Considerations

This document defines a new entry to the "Codec Control Messages" subregistry of the "Session Description Protocol (SDP) Parameters" registry, according to the following data:

Value name: lrr

Long name: Layer Refresh Request Command

Usable with: ccm

Mux: IDENTICAL-PER-PT

Reference: RFC XXXX

This document also defines a new entry to the "FMT Values for PSFB Payload Types" subregistry of the "Real-Time Transport Protocol (RTP) Parameters" registry, according to the following data:

Name: LRR

Long Name: Layer Refresh Request Command

Value: TBD

Reference: RFC XXXX

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RTP Stream Identifier Source Description (SDES)
draft-ietf-avtext-rid-09

Abstract

This document defines and registers two new RTCP Stream Identifier Source Description (SDES) items. One, named RtpStreamId, is used for unique identification of RTP streams. The other, RepairedRtpStreamId, can be used to identify which stream a redundancy RTP stream is to be used to repair.

Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on April 9, 2017.

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1. Introduction

RTP sessions frequently consist of multiple streams, each of which is identified at any given time by its SSRC; however, the SSRC associated with a stream is not guaranteed to be stable over its lifetime. Within a session, these streams can be tagged with a number of identifiers, including CNAMEs and MSIDs [I-D.ietf-mmusic-msid]. Unfortunately, none of these have the proper ordinality to refer to an individual stream; all such identifiers can appear in more than one stream at a time. While approaches that use unique Payload Types (PTs) per stream have been used in some applications, this is a semantic overloading of that field, and one for which its size is inadequate: in moderately complex systems that use PT to uniquely identify every potential combination of codec configuration and unique stream, it is possible to simply run out of values.

To address this situation, we define a new RTCP Stream Identifier Source Description (SDES) identifier, RtpStreamId, that uniquely identifies a single RTP stream. A key motivator for defining this identifier is the ability to differentiate among different encodings of a single Source Stream that are sent simultaneously (i.e., simulcast). This need for unique identification extends to dependent

streams (e.g., where layers used by a layered codec are transmitted on separate streams).

At the same time, when redundancy RTP streams are in use, we also need an identifier that connects such streams to the RTP stream for which they are providing redundancy. For this purpose, we define an additional SDES identifier, RepairedRtpStreamId. This identifier can appear only in packets associated with a redundancy RTP stream. They carry the same value as the RtpStreamId of the RTP stream that the redundant RTP stream is correcting.

2. Terminology

In this document, the terms "source stream", "RTP stream", "source RTP stream", "dependent stream", "received RTP stream", and "redundancy RTP stream" are used as defined in [RFC7656].

The following acronyms are also used:

- CNAME: Canonical End-Point Identifier, defined in [RFC3550]
- MID: Media Identification, defined in [I-D.ietf-mmusic-sdp-bundle-negotiation]
- MSID: Media Stream Identifier, defined in [I-D.ietf-mmusic-msid]
- RTCP: Real-time Transport Control Protocol, defined in [RFC3550]
- RTP: Real-time Transport Protocol, defined in [RFC3550]
- SDES: Source Description, defined in [RFC3550]
- SSRC: Synchronization Source, defined in [RFC3550]

3. Usage of RtpStreamId and RepairedRtpStreamId in RTP and RTCP

The RTP fixed header includes the payload type number and the SSRC values of the RTP stream. RTP defines how you de-multiplex streams within an RTP session; however, in some use cases, applications need further identifiers in order to effectively map the individual RTP Streams to their equivalent payload configurations in the SDP.

This specification defines two new RTCP SDES items [RFC3550]. The first item is ’RtpStreamId’, which is used to carry RTP stream identifiers within RTCP SDES packets. This makes it possible for a receiver to associate received RTP packets (identifying the RTP stream) with a media description having the format constraint specified. The second is ’RepairedRtpStreamId’, which can be used in

redundancy RTP streams to indicate the RTP stream repaired by a redundancy RTP stream.

To be clear: the value carried in a RepairedRtpStreamId will always match the RtpStreamId value from another RTP stream in the same session. For example, if a source RTP stream is identified by RtpStreamId "A", then any redundancy RTP stream that repairs that source RTP stream will contain a RepairedRtpStreamId of "A" (if this mechanism is being used to perform such correlation). These redundant RTP streams may also contain their own unique RtpStreamId.

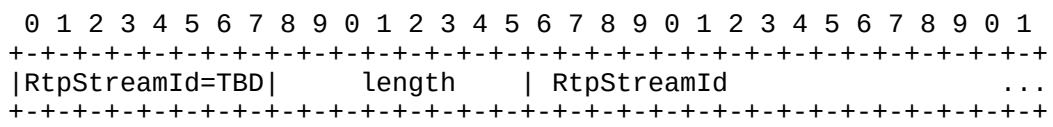
This specification also uses the RTP header extension for RTCP SDES items [I-D.ietf-avtext-sdes-hdr-ext] to allow carrying RtpStreamId and RepairedRtpStreamId values in RTP packets. This allows correlation at stream startup, or after stream changes where the use of RTCP may not be sufficiently responsive. This speed of response is necessary since, in many cases, the stream cannot be properly processed until it can be identified.

RtpStreamId and RepairedRtpStreamId values are scoped by source identifier (e.g., CNAME) and by media session. When the media is multiplexed using the BUNDLE extension [I-D.ietf-mmusic-sdp-bundle-negotiation], these values are further scoped by their associated MID values. For example: an RtpStreamId of "1" may be present in the stream identified with a CNAME of "1234@example.com", and may also be present in a stream with a CNAME of "5678@example.org", and these would refer to different streams. Similarly, an RtpStreamId of "1" may be present with an MID of "A", and again with a MID of "B", and also refer to two different streams.

Note that the RepairedRtpStreamId mechanism is limited to indicating one repaired stream per redundancy stream. If systems require correlation for schemes in which a redundancy stream contains information used to repair more than one stream, they will have to use a more complex mechanism than the one defined in this specification.

As with all SDES items, RtpStreamId and RepairedRtpStreamId are limited to a total of 255 octets in length. RtpStreamId and RepairedStreamId are constrained to contain only alphanumeric characters. For avoidance of doubt, the only allowed byte values for these IDs are decimal 48 through 57, 65 through 90, and 97 through 122.

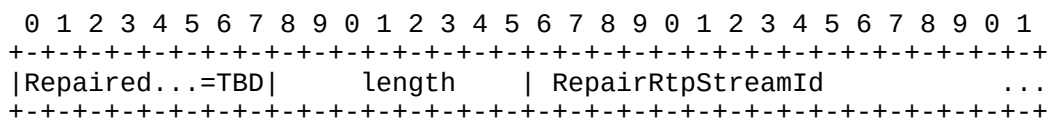
3.1. RTCP 'RtpStreamId' SDES Extension



The RtpStreamId payload is ASCII encoded and is not null-terminated.

RFC EDITOR NOTE: Please replace TBD with the assigned SDES identifier value.

3.2. RTCP 'RepairedRtpStreamId' SDES Extension



The RepairedRtpStreamId payload is ASCII encoded and is not null-terminated.

RFC EDITOR NOTE: Please replace TBD with the assigned SDES identifier value.

3.3. RTP 'RtpStreamId' and 'RepairedRtpStreamId' Header Extensions

Because recipients of RTP packets will typically need to know which streams they correspond to immediately upon receipt, this specification also defines a means of carrying RtpStreamId and RepairedRtpStreamId identifiers in RTP extension headers, using the technique described in [I-D.ietf-avtext-sdes-hdr-ext].

As described in that document, the header extension element can be encoded using either the one-byte or two-byte header, and the identification-tag payload is ASCII-encoded.

As the identifier is included in an RTP header extension, there should be some consideration given to the packet expansion caused by the identifier. To avoid Maximum Transmission Unit (MTU) issues for the RTP packets, the header extension's size needs to be taken into account when encoding media. Note that the set of header extensions included in the packet needs to be padded to the next 32-bit boundary [RFC5285].

In many cases, a one-byte identifier will be sufficient to distinguish streams in a session; implementations are strongly encouraged to use the shortest identifier that fits their purposes. Implementors are warned, in particular, not to include any information in the identifier that is derived from potentially user-identifying information, such as user ID or IP address. To avoid identification of specific implementations based on their pattern of tag generation, implementations are encouraged to use a simple scheme that starts with the ASCII digit "1", and increments by one for each subsequent identifier.

4. IANA Considerations

4.1. New RtpStreamId SDES item

RFC EDITOR NOTE: Please replace RFCXXXX with the RFC number of this document.

RFC EDITOR NOTE: Please replace TBD with the assigned SDES identifier value.

This document adds the RtpStreamId SDES item to the IANA "RTP SDES item types" registry as follows:

Value:	TBD
Abbrev.:	RtpStreamId
Name:	RTP Stream Identifier
Reference:	RFCXXXX

4.2. New RepairRtpStreamId SDES item

RFC EDITOR NOTE: Please replace RFCXXXX with the RFC number of this document.

RFC EDITOR NOTE: Please replace TBD with the assigned SDES identifier value.

This document adds the RepairedRtpStreamId SDES item to the IANA "RTP SDES item types" registry as follows:

Value:	TBD
Abbrev.:	RepairedRtpStreamId
Name:	Repaired RTP Stream Identifier
Reference:	RFCXXXX

4.3. New RtpStreamId Header Extension URI

RFC EDITOR NOTE: Please replace RFCXXXX with the RFC number of this document.

This document defines a new extension URI in the RTP SDES Compact Header Extensions sub-registry of the RTP Compact Header Extensions registry sub-registry, as follows

Extension URI: urn:ietf:params:rtp-hdrext:sdes:rtp-stream-id
Description: RTP Stream Identifier Contact: adam@nostrum.com
Reference: RFCXXXX

4.4. New RepairRtpStreamId Header Extension URI

RFC EDITOR NOTE: Please replace RFCXXXX with the RFC number of this document.

This document defines a new extension URI in the RTP SDES Compact Header Extensions sub-registry of the RTP Compact Header Extensions registry sub-registry, as follows

Extension URI: urn:ietf:params:rtp-hdrext:sdes:repaired-rtp-sream-id
Description: RTP Repaired Stream Identifier Contact: adam@nostrum.com
Reference: RFCXXXX

5. Security Considerations

Although the identifiers defined in this document are limited to be strictly alphanumeric, SDES items have the potential to carry any string. As a consequence, there exists a risk that it might carry privacy-sensitive information. Implementations need to take care when generating identifiers so that they do not contain information that can identify the user or allow for long term tracking of the device. Following the generation recommendations in Section 3.3 will result in non-instance-specific labels, with only minor fingerprinting possibilities in the total number of used RtpStreamIds and RepairedRtpStreamIds.

Even if the SDES items are generated to convey as little information as possible, implementors are strongly encouraged to encrypt SDES items - both in RTCP and RTP header extensions - so as to preserve privacy against third parties.

As the SDES items are used for identification of the RTP streams for different application purposes, it is important that the intended values are received. An attacker, either a third party or malicious RTP middlebox, that removes, or changes the values for these SDES

items, can severely impact the application. The impact can include failure to decode or display the media content of the RTP stream. It can also result in incorrectly attributing media content to identifiers of the media source, such as incorrectly identifying the speaker. To prevent this from occurring due to third party attacks, integrity and source authentication is needed.

Options for Securing RTP Sessions [RFC7201] discusses options for how encryption, integrity and source authentication can be accomplished.

6. Acknowledgements

Many thanks for review and input from Cullen Jennings, Magnus Westerlund, Colin Perkins, Jonathan Lennox, and Paul Kyzivat. Magnus Westerlund provided substantially all of the Security Considerations section.

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