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RTP Control Protocol (RTCP) Feedback for Congestion Control
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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.

Table of Contents

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:

```
  0                   1                   2                   3
  0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |V=2|P| FMT=CCFB |   PT = 205    |          length               |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |                 SSRC of RTCP packet sender                    |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |                   SSRC of 1st RTP Stream                      |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |          begin_seq            |          num_reports          |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |L|ECN|  Arrival time offset    | ...                           .
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 .                                                               .
 .                                                               .
 .                                                               .
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |                   SSRC of nth RTP Stream                      |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |          begin_seq            |          num_reports          |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |L|ECN|  Arrival time offset    | ...                           |
 .                                                               .
 .                                                               .
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |                 Report Timestamp (32bits)                     |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

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 in the range begin_seq to begin_seq+num_reports inclusive (calculated using arithmetic modulo 65536 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. The value of num_reports MAY be zero, indicating that there are no packet metric blocks included for that SSRC. 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, as an offset before the time represented by the RTS field of this RTCP congestion control feedback report. The ATO field is in units of 1/1024 seconds (this unit is chosen to give exact offsets from the RTS field) so, for example, an ATO value of 512 indicates that the corresponding RTP packet arrived exactly half a second before the time instant represented by 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 measurement. If the measurement is unavailable, or if the arrival time of the RTP packet is after the time represented by the RTS field, then an ATO value of 0x1FFF MUST be reported for the packet.

The RTCP congestion control feedback report packet concludes with the Report Timestamp field (RTS, 32 bits). This denotes the time instant on which this packet is reporting, and is the instant from which the arrival time offset values are calculated. The value of RTS field is derived from the same clock 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 every active SSRC. The sequence number ranges reported on in consecutive reports for a given SSRC will generally be contiguous, but overlapping reports MAY be sent (and need to be sent in cases

where RTP packet reordering occurs across the boundary between consecutive reports). If reports covering overlapping sequence number ranges are sent, information in later reports updates that in sent previous reports for RTP packets included in both reports. If an RTP packet was reported as received in one report, that packet MUST also be reported as received in any overlapping reports sent later that cover its sequence number range.

RTCP congestion control feedback packets can be large if they are sent infrequently relative to the number of RTP data packets. If an RTCP congestion control feedback packet is too large to fit within the path MTU, its sender SHOULD split it into multiple feedback packets. The RTCP reporting interval SHOULD be chosen such that feedback packets are sent often enough that they are small enough to fit within the path MTU ([I-D.ietf-rmcat-rtp-cc-feedback] provides guidance on how to choose the reporting interval).

If duplicate copies of a particular RTP packet are received, then the arrival time of the first copy to arrive MUST be reported. If any of the copies of the duplicated packet are ECN-CE marked, then an ECN-CE mark MUST be reported that for packet; otherwise the ECN mark of the first copy to arrive is reported.

If no packets are received from an SSRC in a reporting interval, a report block MAY be sent with begin_seq set to the highest sequence number previously received from that SSRC and num_reports set to zero (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.

A report block indicating that certain RTP packets were lost is not to be interpreted as a request to retransmit the lost packets. The receiver of such a report might choose to retransmit such packets, provided a retransmission payload format has been negotiated, but there is no requirement that it do so.

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. Response to Loss of Feedback Packets

Like all RTCP packets, RTCP congestion control feedback packets might be lost. All RTP congestion control algorithms MUST specify how they respond to the loss of feedback packets.

If only a single congestion control feedback packet is lost, an appropriate response is to assume that the level of congestion has remained roughly the same as the previous report. However, if multiple consecutive congestion control feedback packets are lost, the sender SHOULD rapidly reduce its sending rate towards zero, as this likely indicates a path failure. The RTP circuit breaker [RFC8083] provides further guidance.

6. SDP Signalling

A new "ack" feedback parameter, "ccfb", is defined for use with the "a=rtcp-fb:" SDP extension 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 [RFC4585]>
rtcp-fb-ack-param =/ ccfb-par
ccfb-par          = SP "ccfb"
```

When used with "ccfb" feedback, the wildcard payload type ("*") MUST be used. This implies that the congestion control feedback is sent for all payload types in use in the session, including any FEC and retransmission payload types. An example of the resulting SDP attribute is:

```
a=rtcp-fb:* ack ccfb
```

The offer/answer rules for these SDP feedback parameters are specified in Section 4.2 of the RTP/AVPF profile [RFC4585].

An SDP offer might indicate support for both the congestion control feedback mechanism specified in this memo and one or more alternative congestion control feedback mechanisms that offer substantially the same semantics. In this case, the answering party SHOULD include only one of the offered congestion control feedback mechanisms in its answer. If a re-invite offering the same set of congestion control feedback mechanisms is received, the generated answer SHOULD choose the same congestion control feedback mechanism as in the original answer where possible.

When the SDP BUNDLE extension [I-D.ietf-mmusic-sdp-bundle-negotiation] is used for multiplexing, the "a=rtcp-fb:" attribute has multiplexing category IDENTICAL-PER-PT [I-D.ietf-mmusic-sdp-mux-attributes].

7. Relation to RFC 6679

Use of Explicit Congestion Notification (ECN) with RTP is described in [RFC6679]. That specifies how to negotiate the use of ECN with RTP, and defines an RTCP ECN Feedback Packet to carry ECN feedback reports. It uses an SDP "a=ecn-capaable-rtp:" attribute to negotiate use of ECN, and the "a=rtcp-fb:" attributes with the "nack" parameter "ecn" to negotiate the use of RTCP ECN Feedback Packets.

The RTCP ECN Feedback Packet is not useful when ECN is used with the RTP Congestion Control Feedback Packet defined in this memo since it provides duplicate information. Accordingly, when congestion control feedback is to be used with RTP and ECN, the SDP offer generated MUST include an "a=ecn-capable-rtp:" attribute to negotiate ECN support, along with an "a=rtcp-fb:" attribute with the "ack" parameter "ccfb" to indicate that the RTP Congestion Control Feedback Packet is to be used for feedback. The "a=rtcp-fb:" attribute MUST NOT include the "nack" parameter "ecn", so the RTCP ECN Feedback Packet will not be used.

8. 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.

9. Acknowledgements

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10. 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)
```

11. 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.

12. References

12.1. Normative References

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