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RFC 8888
RTP Control Protocol (RTCP) Feedback for Congestion Control

Abstract

An effective RTP congestion control algorithm requires more fine-grained feedback on packet loss, timing, and Explicit Congestion Notification (ECN) marks than is provided by the standard RTP Control Protocol (RTCP) Sender Report (SR) and Receiver Report (RR) packets. 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 back to the sender RTCP feedback packets containing the information the sender needs to perform congestion control.

Status of This Memo

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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) [RFC3550] 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) [RFC3550] 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 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 a generic congestion control feedback format.

To help achieve interoperability for unicast RTP congestion control, this memo specifies a common RTCP feedback packet format that can be used by Network-Assisted Dynamic Adaptation (NADA) [RFC8698], Self-Clocked Rate Adaptation for Multimedia (SCReAM) [RFC8298], Google Congestion Control [Google-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**", "**NOT RECOMMENDED**", "**MAY**", and "**OPTIONAL**" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

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

3. RTCP Feedback for Congestion Control

Based on an analysis of NADA [RFC8698], SCReAM [RFC8298], Google Congestion Control [Google-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 feed the sequence numbers of the received RTP packets back 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 feed the arrival time of each RTP packet back 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 feed back 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 Congestion Experienced (ECN-CE). ("ECT" stands for "ECN-Capable Transport".) If the path used by the RTP traffic is ECN capable, the sender can use 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=11  |   PT = 205    |          length               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                 SSRC of RTCP packet sender                    |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                   SSRC of 1st RTP Stream                      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|          begin_seq            |          num_reports          |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|R|ECN|  Arrival time offset    | ...                           .
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
.                                                               .
.                                                               .
.                                                               .
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                   SSRC of nth RTP Stream                      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|          begin_seq            |          num_reports          |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|R|ECN|  Arrival time offset    | ...                           |
.                                                               .
.                                                               .
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                 Report Timestamp (32 bits)                    |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 1: RTCP Congestion Control Feedback Packet Format

The first 8 octets comprise a standard RTCP header, with PT=205 and FMT=11 indicating that this is a congestion control feedback packet, and with the SSRC set to that of the sender of the RTCP packet.

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 that has a 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 0, 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 comprise the R, ECN, and ATO fields as follows:

Received (R, 1 bit): A boolean that indicates whether the packet was received. 0 indicates that the packet was not yet received and the subsequent 15 bits (ECN and ATO) in this 16-bit packet metric block are also set to 0 and **MUST** be ignored. 1 indicates that the packet was received and the subsequent bits in the block need to be parsed.

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

Arrival time offset (ATO, 13 bits): The arrival time of the RTP packet at the receiver, as an offset before the time represented by the Report Timestamp (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 the 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 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. If feedback reports covering overlapping sequence number ranges are sent, information in later feedback reports may update any data sent in previous reports for RTP packets included in both feedback reports.

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. ([RTCP-Multimedia-Feedback] discusses how to choose the reporting interval; specifications for RTP congestion control algorithms can also provide guidance.)

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 for that 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 0 (or the report can simply be omitted). The corresponding Sender Report / Receiver Report (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. [RTCP-Multimedia-Feedback] discusses this trade-off, suggests desirable RTCP feedback rates, and provides guidance on how to configure, for example, the RTCP bandwidth fraction to make appropriate use of the reporting block described in this memo. Specifications for RTP congestion control algorithms can also provide guidance.

It is generally understood that congestion control algorithms work better with more frequent feedback. However, RTCP bandwidth and transmission rules put some upper limits on how frequently the RTCP feedback messages can be sent from an RTP receiver to the RTP sender. In many cases, sending feedback once per frame is an upper bound before the reporting overhead becomes excessive, although this will depend on the media rate and more frequent feedback might be needed with high-rate media flows [RTCP-Multimedia-Feedback]. Analysis [feedback-requirements] has also shown that some candidate congestion control algorithms can operate with less frequent feedback, using a feedback interval range of 50-200 ms. Applications need to negotiate an appropriate congestion control feedback interval at session setup time, based on the choice of congestion control algorithm, the expected media bitrate, and the acceptable feedback overhead.

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.

RTCP packets do not contain a sequence number, so loss of feedback packets has to be inferred based on the time since the last feedback packet. 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, then the media sender **SHOULD** rapidly reduce its sending rate as this likely indicates a path failure. The RTP circuit breaker specification [RFC8083] provides further guidance.

6. SDP Signaling

A new "ack" feedback parameter, "ccfb", is defined for use with the "a=rtcp-fb:" Session Description Protocol (SDP) extension to indicate the use of the RTP Congestion Control Feedback Packet format defined in Section 3. The ABNF definition [RFC5234] 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"
```

The payload type used with "ccfb" feedback **MUST** be the wildcard type ("*"). This implies that the congestion control feedback is sent for all payload types in use in the session, including any Forward Error Correction (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 subsequent offer containing 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 [RFC8843] is used for multiplexing, the "a=rtcp-fb:" attribute has multiplexing category IDENTICAL-PER-PT [RFC8859].

7. Relationship to RFC 6679

The use of Explicit Congestion Notification (ECN) with RTP is described in [RFC6679], which 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-capable-rtp:" attribute to negotiate the use of ECN, and the "a=rtcp-fb:" attribute 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. 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 can be used. The "a=rtcp-fb:" attribute **MAY** also include the "nack" parameter "ecn" to indicate that the RTCP ECN Feedback Packet is also supported. If an SDP offer signals support for both RTP Congestion Control Feedback Packets and the RTCP ECN Feedback Packet, the answering party **SHOULD** signal support for one, but not both, formats in its SDP answer to avoid sending duplicate feedback.

When using ECN with RTP, the guidelines in Section 7.2 of [RFC6679] **MUST** be followed to initiate the use of ECN in an RTP session. The guidelines in Section 7.3 of [RFC6679] regarding the ongoing use of ECN within an RTP session **MUST** also be followed, with the exception that feedback is sent using the RTCP Congestion Control Feedback Packets described in this memo rather than using RTP ECN Feedback Packets. Similarly, the guidance in Section 7.4 of [RFC6679] related to detecting failures **MUST** be followed, with the exception that the necessary information is retrieved from the RTCP Congestion Control Feedback Packets rather than from RTP ECN Feedback Packets.

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 feedback format defined in this memo provides such fine-grained feedback.

Several other RTCP extensions also provide more detailed feedback than SR/RR packets:

TMMBR:

The codec control messages for the RTP/AVPF profile [RFC5104] include a Temporary Maximum Media Stream Bit Rate Request (TMMBR) message. This is used to convey a temporary maximum bitrate 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 bitrate suitable for current available bandwidth on the path; it also assumes that the RTP sender can/will respect that bitrate. 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 sender-based congestion control. TMMBR could, however, be viewed as a complementary mechanism that can inform the sender of the receiver's current view of an acceptable maximum bitrate. Mechanisms that convey the receiver's estimate of the maximum available bitrate provide similar feedback.

RTCP Extended Reports (XRs): Numerous RTCP XR blocks have 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 into a compound RTCP packet, to report the detailed loss, arrival time, and ECN marking information needed for effective sender-based congestion control. However, the result has high overhead in terms of both bandwidth and complexity, due to the need to stack multiple reports.

Transport-wide Congestion Control: The format defined in this memo provides individual feedback on each SSRC. An alternative is to add a header extension to each RTP packet, containing a single, transport-wide, packet sequence number, then have the receiver send RTCP reports giving feedback on these additional sequence numbers [RTP-Ext-for-CC]. Such an approach increases the size of each RTP packet by 8 octets, due to the header extension, but reduces the size of the RTCP feedback packets, and can simplify the rate calculation at the sender if it maintains a single rate limit that applies to all RTP packets sent, irrespective of their SSRC. Equally, the use of transport-wide feedback makes it more difficult to adapt the sending rate, or respond to lost packets, based on the reception and/or loss patterns observed on a per-SSRC basis (for example, to perform differential rate control and repair for audio and video flows, based on knowledge of what packets from each flow were lost). Transport-wide feedback is also a less natural fit with the wider RTP framework, which makes extensive use of per-SSRC sequence numbers and feedback.

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. IANA Considerations

The IANA has registered one new RTP/AVPF Transport-Layer Feedback Message in the "FMT Values for RTPFB Payload Types" table [RFC4585] as defined in Section 3.1:

Name: CCFB
Long name: RTP Congestion Control Feedback
Value: 11
Reference: RFC 8888

The IANA has also registered 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
Mux: IDENTICAL-PER-PT
Reference: RFC 8888

10. 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 being used (e.g., [RFC8698], [RFC8298], [Google-GCC], or [RFC8382]) apply.

A receiver that intentionally generates inaccurate RTCP congestion control feedback reports might be able to trick the sender into sending at a greater rate than the path can support, thereby causing congestion on the path. This scenario will negatively impact the quality of experience of that receiver, potentially causing both denial of service to other traffic sharing the path and excessively increased resource usage at the media sender. Since RTP is an unreliable transport, a sender can intentionally drop a packet, leaving a gap in the RTP sequence number space without causing serious harm, to check that the receiver is correctly reporting losses. (This needs to be done with care and some awareness of the media data being sent, to limit impact on the user experience.)

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.

An off-path attacker that can spoof RTCP Congestion Control Feedback Packets can similarly trick a sender into sending at an incorrect rate, leading to denial of service. This attack is difficult, since the attacker needs to guess the SSRC and sequence number in addition to the destination transport address. As with on-path attacks, the secure RTCP profile [RFC3711] can be used to authenticate RTCP packets to protect against this attack.

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