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        RTP Control Protocol (RTCP) Feedback for Congestion Control
              draft-ietf-avtcore-cc-feedback-message-06

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.

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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:

o 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.

o 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.

o 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:

o 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.

o 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.

o 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 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 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 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.
It has been shown [I-D.ietf-rmcat-rtp-cc-feedback] that in many cases
sending feedback one 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. Analysis [feedback-requirements] has also shown that some
candidate congestion control algorithms can operate with less
frequent feedback, using a feedback interval range of 50-200ms.
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 bit rate, 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.

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 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 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. The Received Estimated
   Maximum Bit-rate (REMB) mechanism [I-D.alvestrand-rmcat-remb]
   provides similar feedback.

RTCP Extended Reports (XR): Numerous RTCP extended report (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
   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.

   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 [I-D.holmer-rmcat-transport-wide-cc-extensions]. Such an
   approach adds the per-packet overhead of the header extension (8
   octets per packet in the referenced format), but reduces the size
   of the 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. Acknowledgements

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   the RTP Media Congestion Avoidance Techniques (RMCAT) working group.
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   feedback.

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.

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