

IETF RMCAT Working Group  
Internet-Draft  
Intended status: Standards Track  
Expires: May 3, 2018

Z. Sarker  
Ericsson AB  
C. Perkins  
University of Glasgow  
V. Singh  
callstats.io  
M. Ramalho  
Cisco Systems  
October 30, 2017

**RTP Control Protocol (RTCP) Feedback for Congestion Control**  
**draft-ietf-avtcore-cc-feedback-message-00**

**Abstract**

This document describes a feedback message intended to enable congestion control for interactive real-time traffic. The RTP Media Congestion Avoidance Techniques (RMCAT) Working Group formed a design team to analyze feedback requirements from various congestion control algorithms and to design a generic feedback message to help ensure interoperability across those algorithms. The feedback message is designed for a sender-based congestion control, which means the receiver of the media will send necessary feedback to the sender of the media to perform the congestion control at the sender.

**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 <https://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 May 3, 2018.

**Copyright Notice**

Copyright (c) 2017 IETF Trust and the persons identified as the document authors. All rights reserved.

This document is subject to [BCP 78](#) and the IETF Trust's Legal Provisions Relating to IETF Documents (<https://trustee.ietf.org/license-info>) in effect on the date of publication of this document. Please review these documents carefully, as they describe your rights and restrictions with respect to this document. Code Components extracted from this document must include Simplified BSD License text as described in Section 4.e of the Trust Legal Provisions and are provided without warranty as described in the Simplified BSD License.

## Table of Contents

<a href="#">1.</a>	Introduction . . . . .	<a href="#">2</a>
<a href="#">2.</a>	Terminology . . . . .	<a href="#">3</a>
<a href="#">3.</a>	Feedback Message . . . . .	<a href="#">3</a>
<a href="#">3.1.</a>	RTCP Congestion Control Feedback Report . . . . .	<a href="#">4</a>
<a href="#">4.</a>	Feedback Frequency and Overhead . . . . .	<a href="#">6</a>
<a href="#">5.</a>	Design Rationale . . . . .	<a href="#">7</a>
<a href="#">6.</a>	Acknowledgements . . . . .	<a href="#">7</a>
<a href="#">7.</a>	IANA Considerations . . . . .	<a href="#">8</a>
<a href="#">8.</a>	Security Considerations . . . . .	<a href="#">8</a>
<a href="#">9.</a>	References . . . . .	<a href="#">8</a>
<a href="#">9.1.</a>	Normative References . . . . .	<a href="#">8</a>
<a href="#">9.2.</a>	Informative References . . . . .	<a href="#">9</a>
	Authors' Addresses . . . . .	<a href="#">10</a>

## [1.](#) Introduction

For interactive real-time traffic the typical protocol choice is Realtime Transport Protocol (RTP) over User Datagram Protocol (UDP). RTP does not provide any guarantee of Quality of Service (QoS), reliable or timely delivery and expects the underlying transport protocol to do so. UDP alone certainly does not meet that expectation. However, RTP Control Protocol (RTCP) provides a mechanism to periodically send transport and media metrics to the media sender which can be utilized and extended for the purposes of RMCAT congestion control. For a congestion control algorithm which operates at the media sender, RTCP messages can be transmitted from the media receiver back to the media sender to enable congestion control. In the absence of standardized messages for this purpose, the congestion control algorithm designers have designed proprietary RTCP messages that convey only those parameters required for their respective designs. As a direct result, the different congestion control (a.k.a. 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 format that can be used by NADA [[I-D.ietf-rmcat-nada](#)], SCReAM [[I-D.ietf-rmcat-scream-cc](#)], Google Congestion Control [[I-D.ietf-rmcat-gcc](#)] and Shared Bottleneck Detection [[I-D.ietf-rmcat-sbd](#)], and hopefully future RTP congestion control algorithms as well.

## 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. Feedback Message

The design team analyzed the feedback requirements from the different proposed candidate in RMCAT WG. The analysis showed some commonalities between the proposed solution candidate and some can be derived from other information. The design team has agreed to have following packet information block in the feedback message to satisfy different requirement analyzed.

- o Packet Identifier : RTP sequence number. The RTP packet header includes an incremental packet sequence number that the sender needs to correlate packets sent at the sender with packets received at the receiver.
- o Packet Arrival Time : Arrival time stamp at the receiver of the media. The sender requires the arrival time stamp of the respective packet to determine delay and jitter the packet had experienced during transmission. In a sender based congestion control solution the sender requires to keep track of the sent packets - usually packet sequence number, packet size and packet send time. With the packet arrival time the sender can detect the delay and jitter information. Along with packet loss and delay information the sender can estimate the available bandwidth and thus adapt to the situation.
- o Packet Explicit Congestion Notification (ECN) Marking : If ECN [[RFC3168](#)] is used, it is necessary to report on the 2-bit ECN mark in received packets, indicating for each packet whether it is marked not-ECT, ECT(0), ECT(1), or ECN-CE. If the path on which the media traffic traversing is ECN capable then the sender can use the Congestion Experienced (ECN-CE) marking information for congestion control. It is important that the receiver sends the



ECN-CE marking information of the packet back to the sender to take the advantages of ECN marking. Note that how the receiver gets the ECN marking information at application layer is out of the scope of this design team. Additional information for ECN use with RTP can be found at [[RFC6679](#)].

The feedback messages can have one or more of the above information blocks. For RTCP based feedback message the packet information block will be grouped by Synchronization Source (SSRC) identifier.

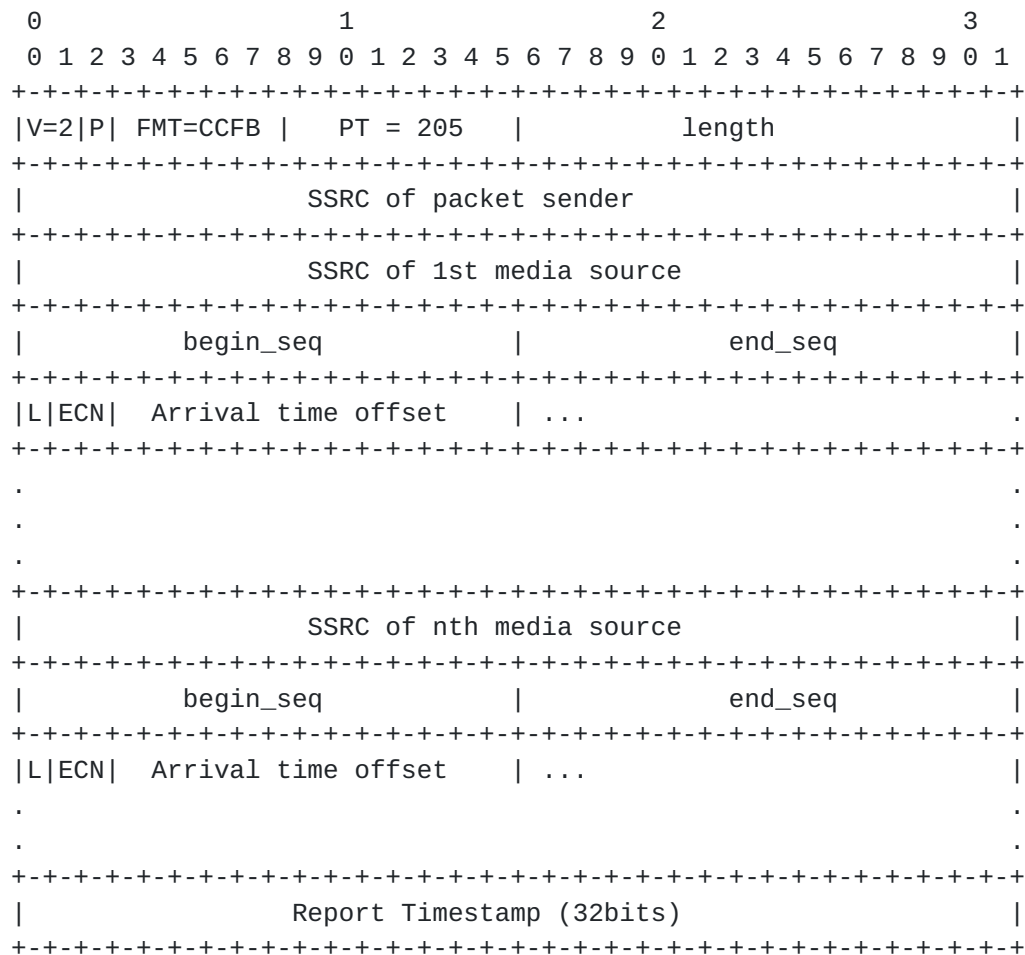
As a practical matter, we note that host Operating System (OS) process interruptions can occur at inopportune times. Thus, the recording of the sent times at the sender and arrival times at the receiver should be made with deliberate care. This is because the time duration of host OS interruptions can be significant relative to the precision desired in the one-way delay estimates. Specifically, the send time should be recorded at the latest opportunity prior to outputting the media packet at the sender (e.g., socket or RTP API) and the arrival time at the receiver (e.g., socket or RTP API) should be recorded at the earliest opportunity available to the receiver.

### **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 as follows:





The first 8 octets are the RTCP header, with PT=205 and FMT=CCFB specifying the remainder is a congestion control feedback packet, and including the SSRC of the packet sender. (NOTE TO RFC EDITOR: please replace CCFB here and in the above diagram with the IANA assigned RTCP feedback packet type)

[Section 6.1 of \[RFC4585\]](#) requires the RTCP header to be followed by the SSRC of the media source being reported upon. Accordingly, the RTCP header is followed by a report for each SSRC received, followed by the Report Timestamp.

The report for each SSRC received starts with the SSRC of that media source. Then, each sequence number between the begin\_seq and end\_seq (both inclusive) is represented by a packet metric block of 16-bits that contains the L, ECN, and ATO fields. If an odd number of reports are included, i.e., end\_seq - begin\_seq is odd then 16 bits of zero padding MUST be added after the last report, to align the RTCP packet to a four (4) bytes boundary. 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 relative arrival time of the RTP packets at the receiver before this feedback report was generated measured in milliseconds. It is calculated by subtracting the reception timestamp of the RTP packet denoted by this 16bit block and the timestamp (RTS) of this report. If the measured value is greater than 8.189 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.

Report Timestamp (RTS, 32 bits): represents the timestamp when this report was generated. The sender of the feedback message decides on the wall-clock. Usually, it should be derived from the same wall-clock that is used for timestamping RTP packets arrival. Consistency in the unit and resolution (10th of millisecond should be good enough) is important here. In addition, the media sender can ask for a specific resolution it wants.

#### **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, and the possible rates of feedback.

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 media receiver to the media 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. The design team also have noted that even if a higher frequency of feedback is desired it is not viable if the feedback messages starts to compete against the media 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. Design Rationale

The primary function of RTCP Sender Report (SR) / Receiver Report (RR) is to report the reception quality of media. The regular SR / RR reports contain information about observed jitter, fractional packet loss and cumulative packet loss. The original intent of this information was to assist flow and congestion control mechanisms. Even though it is possible to do congestion control based on information provided in the SR/RR reports it is not sufficient to design an efficient congestion control algorithm for interactive real-time communication. An efficient congestion control algorithm requires more fine grain information on per packet (see [Section 3](#)) to react to the congestion or to avoid further congestion on the path.

Codec Control Message for AVPF [[RFC5104](#)] defines Temporary Maximum Media Bit Rate (TMMBR) message which conveys a temporary maximum bitrate limitation from the receiver of the media to the sender of the media. Even though it is 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 with reduced sized reports [[RFC5506](#)]. This requires the receiver of the media to analyze the data reception, detect congestion level and recommend a maximum bitrate suitable for current available bandwidth on the path with an assumption that the sender of the media always honors the TMMBR message. This requirement is completely opposite of the sender based congestion control approach. Hence, TMMBR cannot be as a signaling means for a sender based congestion control mechanism. However, TMMBR should be viewed a complimentary signaling mechanism to establish receiver's current view of acceptable maximum bitrate which a sender based congestion control should honor.

There are number of RTCP eXtended Report (XR) blocks have been defined for reporting of delay, loss and ECN marking. It is possible to combine several XR blocks to report the loss and ECN marking at the cost of overhead and complexity. However, there is no existing RTCP XR block to report packet arrival time.

Considering the issues discussed here it is rational to design a new congestion control feedback signaling mechanism for sender based congestion control algorithm.

## 6. Acknowledgements

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



## **7. IANA Considerations**

IANA is requested to assign a new value in the "FMT Values for RTPFB Payload Types" registry for the CCFB transport layer feedback packet described in [Section 3.1](#).

## **8. Security Considerations**

There is a risk of causing congestion if an on-path attacker modifies the feedback messages in such a manner to make available bandwidth greater than it is in reality. [More on security consideration TBD.]

## **9. References**

### **9.1. Normative References**

- [I-D.ietf-rmcat-rtp-cc-feedback]  
Perkins, C., "RTP Control Protocol (RTCP) Feedback for Congestion Control in Interactive Multimedia Conferences", [draft-ietf-rmcat-rtp-cc-feedback-03](#) (work in progress), November 2016.
- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", [BCP 14](#), [RFC 2119](#), DOI 10.17487/RFC2119, March 1997, <<https://www.rfc-editor.org/info/rfc2119>>.
- [RFC3168] Ramakrishnan, K., Floyd, S., and D. Black, "The Addition of Explicit Congestion Notification (ECN) to IP", [RFC 3168](#), DOI 10.17487/RFC3168, September 2001, <<https://www.rfc-editor.org/info/rfc3168>>.
- [RFC3550] Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, [RFC 3550](#), DOI 10.17487/RFC3550, July 2003, <<https://www.rfc-editor.org/info/rfc3550>>.
- [RFC3551] Schulzrinne, H. and S. Casner, "RTP Profile for Audio and Video Conferences with Minimal Control", STD 65, [RFC 3551](#), DOI 10.17487/RFC3551, July 2003, <<https://www.rfc-editor.org/info/rfc3551>>.
- [RFC3611] Friedman, T., Ed., Caceres, R., Ed., and A. Clark, Ed., "RTP Control Protocol Extended Reports (RTCP XR)", [RFC 3611](#), DOI 10.17487/RFC3611, November 2003, <<https://www.rfc-editor.org/info/rfc3611>>.



- [RFC4585] Ott, J., Wenger, S., Sato, N., Burmeister, C., and J. Rey, "Extended RTP Profile for Real-time Transport Control Protocol (RTCP)-Based Feedback (RTP/AVPF)", [RFC 4585](#), DOI 10.17487/RFC4585, July 2006, <<https://www.rfc-editor.org/info/rfc4585>>.
- [RFC5124] Ott, J. and E. Carrara, "Extended Secure RTP Profile for Real-time Transport Control Protocol (RTCP)-Based Feedback (RTP/SAVPF)", [RFC 5124](#), DOI 10.17487/RFC5124, February 2008, <<https://www.rfc-editor.org/info/rfc5124>>.
- [RFC5506] Johansson, I. and M. Westerlund, "Support for Reduced-Size Real-Time Transport Control Protocol (RTCP): Opportunities and Consequences", [RFC 5506](#), DOI 10.17487/RFC5506, April 2009, <<https://www.rfc-editor.org/info/rfc5506>>.
- [RFC6679] Westerlund, M., Johansson, I., Perkins, C., O'Hanlon, P., and K. Carlberg, "Explicit Congestion Notification (ECN) for RTP over UDP", [RFC 6679](#), DOI 10.17487/RFC6679, August 2012, <<https://www.rfc-editor.org/info/rfc6679>>.

## **9.2. Informative References**

- [feedback-requirements]  
"RMCAT Feedback Requirements",  
<:/www.ietf.org/proceedings/95/slides/slides-95-rmcat-1.pdf>.
- [I-D.ietf-rmcat-gcc]  
Holmer, S., Lundin, H., Carlucci, G., Cicco, L., and S. Mascolo, "A Google Congestion Control Algorithm for Real-Time Communication", [draft-ietf-rmcat-gcc-02](#) (work in progress), July 2016.
- [I-D.ietf-rmcat-nada]  
Zhu, X., Pan, R., Ramalho, M., Cruz, S., Jones, P., Fu, J., and S. D'Aronco, "NADA: A Unified Congestion Control Scheme for Real-Time Media", [draft-ietf-rmcat-nada-05](#) (work in progress), September 2017.
- [I-D.ietf-rmcat-sbd]  
Hayes, D., Ferlin, S., Welzl, M., and K. Hiorth, "Shared Bottleneck Detection for Coupled Congestion Control for RTP Media.", [draft-ietf-rmcat-sbd-08](#) (work in progress), July 2017.





[I-D.ietf-rmcat-scream-cc]

Johansson, I. and Z. Sarker, "Self-Clocked Rate Adaptation for Multimedia", [draft-ietf-rmcat-scream-cc-13](#) (work in progress), October 2017.

[RFC5104] Wenger, S., Chandra, U., Westerlund, M., and B. Burman, "Codec Control Messages in the RTP Audio-Visual Profile with Feedback (AVPF)", [RFC 5104](#), DOI 10.17487/RFC5104, February 2008, <<https://www.rfc-editor.org/info/rfc5104>>.

#### Authors' Addresses

Zaheduzzaman Sarker  
Ericsson AB  
Luleae  
Sweden

Phone: +46107173743  
Email: [zaheduzzaman.sarker@ericsson.com](mailto:zaheduzzaman.sarker@ericsson.com)

Colin Perkins  
University of Glasgow  
School of Computing Science  
Glasgow G12 8QQ  
United Kingdom

Email: [csp@csperkins.org](mailto:csp@csperkins.org)

Varun Singh  
Nemu Dialogue Systems Oy  
Runeberginkatu 4c A 4  
Helsinki 00100  
Finland

Email: [varun.singh@iki.fi](mailto:varun.singh@iki.fi)  
URI: <http://www.callstats.io/>

Michael A. Ramalho  
Cisco Systems, Inc.  
6310 Watercrest Way Unit 203  
Lakewood Ranch, FL 34202  
USA

Phone: +1 919 476 2038  
Email: [mramalho@cisco.com](mailto:mramalho@cisco.com)

