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RTCP message for Receiver Estimated Maximum Bitrate
draft-alvestrand-rmcat-remb-03

Abstract

This document proposes an RTCP message for use in experimentally-deployed congestion control algorithms for RTP-based media flows.

It also describes an absolute-value timestamp option for use in bandwidth estimatoin.

Requirements Language

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 RFC 2119 [RFC2119].

Status of this Memo

Copyright Notice

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1. Introduction

This document proposes an RTCP feedback message signalling the estimated total available bandwidth for a session.

If this function is available, it is possible to implement the algorithm in [I-D.alvestrand-rtcweb-congestion], or other algorithms with the same kind of feedback messaging need, in a fashion that covers multiple RTP streams at once.

2. Receiver Estimated Max Bitrate (REMB)

2.1. Semantics

This feedback message is used to notify a sender of multiple media streams over the same RTP session of the total estimated available bit rate on the path to the receiving side of this RTP session.

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 notification. The "SSRC of media source" is not used and SHALL be set to 0. This usage of the value zero is also done in other RFCs.

The reception of a REMB message by a media sender conforming to this specification SHALL result in the total bit rate sent on the RTP session this message applies to being equal to or lower than the bit rate in this message. The new bit rate constraint should be applied as fast as reasonable. The sender is free to apply additional bandwidth restrictions based on its own restrictions and estimates.

2.2. Message format

This document describes a message using the application specific payload type. This is suitable for experimentation; upon standardization, a specific type can be assigned for the purpose.

The message is an RTCP message with payload type 206. RFC 3550 [RFC3550] defines the range, RFC 4585 defines the specific PT value 206 and the FMT value 15.


```
 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=15  |   PT=206      |             length            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                  SSRC of packet sender                        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                  SSRC of media source                         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|  Unique identifier 'R' 'E' 'M' 'B'                            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|  Num SSRC     | BR Exp    |  BR Mantissa                      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|   SSRC feedback                                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|  ...                                                          |
```

The fields V, P, SSRC, and length are defined in the RTP specification [2], the respective meaning being summarized below:

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

padding (P) (1 bit): If set, the padding bit indicates that the packet contains additional padding octets at the end that are not part of the control information but are included in the length field. Always 0.

Feedback message type (FMT) (5 bits): This field identifies the type of the FB message and is interpreted relative to the type (transport layer, payload- specific, or application layer feedback). Always 15, application layer feedback message. RFC 4585 section 6.4.

Payload type (PT) (8 bits): This is the RTCP packet type that identifies the packet as being an RTCP FB message. Always PSFB (206), Payload-specific FB message. RFC 4585 section 6.4.

Length (16 bits): 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 [3]. RFC 4585 section 6.4.

SSRC of packet sender (32 bits): The synchronization source identifier for the originator of this packet. RFC 4585 section 6.4.

SSRC of media source (32 bits): Always 0; this is the same convention as in [RFC5104] section 4.2.2.2 (TMMBN).

Unique identifier (32 bits): Always 'R' 'E' 'M' 'B' (4 ASCII characters).

Num SSRC (8 bits): Number of SSRCs in this message.

BR Exp (6 bits): The exponential scaling of the mantissa for the maximum total media bit rate value, ignoring all packet overhead. The value is an unsigned integer [0..63], as in RFC 5104 section 4.2.2.1.

BR Mantissa (18 bits): The mantissa of the maximum total media bit rate (ignoring all packet overhead) that the sender of the REMB estimates. The BR is the estimate of the traveled path for the SSRCs reported in this message. The value is an unsigned integer in number of bits per second.

SSRC feedback (32 bits) Consists of one or more SSRC entries which this feedback message applies to.

2.3. Signaling of use of this extension

We negotiate use of the message in SDP using a header extension according to RFC 4585 section 4.2, with the value "goog-remb":

a=rtcp-fb:<payload type> goog-remb

3. Absolute Send Time

Google has found that there are issues with relative send time offset when packets are relayed at nodes that are not the source of the RTP clock; it is hard to generate accurate offsets when you have to regenerate the base clock from the incoming packets before you can figure out what time to match; also, using signals from multiple flows becomes impossible unless the timestamps come from a common clock.

The Absolute Send Time extension is used to stamp RTP packets with a timestamp showing the departure time from the system that put this packet on the wire (or as close to this as we can manage).

- o Name: "Absolute Sender Time" ; "RTP Header Extension for Absolute Sender Time".
- o Formal name: "http://www.webrtc.org/experiments/rtp-hdrext/abs-send-time".
- o Wire format: 1-byte extension, 3 bytes of data. total 4 bytes extra per packet (plus shared 4 bytes for all extensions present: 2 byte magic word 0xBEDE, 2 byte # of extensions).
- o Encoding: Timestamp is in seconds, 24 bit 6.18 fixed point, yielding 64s wraparound and 3.8us resolution (one increment for each 477 bytes going out on a 1Gbps interface).
- o Relation to NTP timestamps: abs_send_time_24 = (ntp_timestamp_64

>> 14) & 0x00ffffff ; NTP timestamp is the number of seconds since the epoch, in 32.32 bit fixed point format.

o Notes: Packets are time stamped when going out, preferably close to metal. Intermediate RTP relays (RTP entities possibly altering the relative timing of packets in the stream) should remove the extension or overwrite its value with its own timestamp.

When signalled in SDP, the standard mechanism for RTCP extensions [RFC5285] is used:

a=extmap:3 http://www.webrtc.org/experiments/rtp-hdrext/abs-send-time

4. IANA considerations

Upon publication of this document as an RFC (if it is decided to publish it), IANA is requested to register the string "goog-remb" in its registry of "rtcp-fb" values in the SDP attribute registry group.

5. Security Considerations

If the RTCP packet is not protected, it is possible to inject fake RTCP packets that can increase or decrease bandwidth. This is not different from security considerations for any other RTCP message.

6. Acknowledgements

This proposal has emerged from discussions between, among others, Justin Uberti, Magnus Flodman, Patrik Westin, Stefan Holmer and Henrik Lundin.

7. References

7.1. Normative References

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.

[RFC3550] Schulzrinne, H., Casner, S., Frederick, R., and V.

Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, RFC 3550, July 2003.

[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, July 2006.

[RFC5104] Wenger, S., Chandra, U., Westerlund, M., and B. Burman, "Codec Control Messages in the RTP Audio-Visual Profile with Feedback (AVPF)", RFC 5104, February 2008.

[RFC5285] Singer, D. and H. Desineni, "A General Mechanism for RTP Header Extensions", RFC 5285, July 2008.

7.2. Informative References

[I-D.alvestrand-rtcweb-congestion] Holmer, S. and H. Alvestrand, "A Google Congestion Control Algorithm for Real-Time Communication on the World Wide Web", draft-alvestrand-rtcweb-congestion-03 (work in progress), October 2012.

[RFC5450] Singer, D. and H. Desineni, "Transmission Time Offsets in RTP Streams", RFC 5450, March 2009.

Appendix A. Change log

A.1. From appendix of -congestion-01 to -00

The timestamp option was removed. Discussion concluded that the RFC 5450 [RFC5450] "transmission time offset" header likely gives accurate enough send-time information for our purposes.

A.2. From -00 to -02

No changes. These are "keepalive" publications.

A.3. From -02 to -03

Added information on the absolute-timestamp extension and on SDP negotiation of REMB.

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