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Evaluating Congestion Control for Interactive Real-time Media.

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Abstract

The Real-time Transport Protocol (RTP) is used to transmit media in telephony and video conferencing applications. This document describes the guidelines to evaluate new congestion control algorithms for interactive point-to-point real-time media.

Status of this Memo

Copyright Notice

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1. Introduction

This memo describes the guidelines to help with evaluating new congestion control algorithms for interactive point-to-point real time media. The requirements for the congestion control algorithm are outlined in [I-D.jesup-rtp-congestion-reqs]). This document builds upon previous work at the IETF: Specifying New Congestion Control Algorithms [RFC5033] and Metrics for the Evaluation of Congestion Control Algorithms [RFC5166].

The guidelines proposed in the document are intended to prevent a congestion collapse, promote fair capacity usage and optimize the media flow’s throughput, delay, loss and quality. Furthermore, the proposed algorithms are expected to operate within the envelope of the circuit breakers defined in [I-D.ietf-avtcore-rtp-circuit-breakers].

This document only provides broad-level criteria for evaluating a new congestion control algorithm and the working group should expect a thorough scientific study to make its decision. The results of the evaluation are not expected to be included within the internet-draft but should be cited in the document.

2. Terminology

[Editor’s Note: Should the document use RFC2119 language.]

The terminology defined in RTP [RFC3550], RTP Profile for Audio and Video Conferences with Minimal Control [RFC3551], RTCP Extended Report (XR) [RFC3611], Extended RTP Profile for RTCP-based Feedback (RTP/AVPF) [RFC4585] and Support for Reduced-Size RTCP [RFC5506] apply.

3. Metrics

[RFC5166] describes the basic metrics for congestion control. Section 2.1 of that document discusses the tradeoff between throughput, delay and loss.

- (i) Bandwidth Utilization
  - under-utilization
  - overshooting
- (ii) Packet Loss Rate
- (iii) Quality

[Editor’s Note: Should the algorithm also compare average PSNR of test video sequences?]

4. Guidelines

A congestion control algorithm should be tested in simulation or a testbed environment, and repeated multiple times to infer statistical significance. The following guidelines may be considered for evaluation:

1. Avoiding Congestion Collapse: Does the congestion control propose any changes to (or diverge from) the circuit breaker conditions [I-D.ietf-avtcore-rtp-circuit-breakers] for avoiding congestion collapse.
2. Stability: changing the media encoding rate too often or by too much may adversely affect the users' quality of experience.
3. Diverse environments: The congestion control algorithm should be assessed in heterogeneous environments, containing both wired and wireless paths. Examples of wireless technologies are: 802.11x, HSPA, WCDMA, or GPRS. One of the main challenges of the wireless environments is the inability to distinguish congestion induced loss from transmission (bit-error) loss. Furthermore, packet loss may induce additional delay in networks with wireless paths due to link-layer retransmissions.
4. Varying path characteristics: The congestion control algorithm should be evaluated for a range of path characteristics such as, different end-to-end capacity and latency, varying amount of cross traffic on a bottle-neck link and a router's queue length. [Editor's Note: Different types of queueing mechanisms? Random Early Detection or only DropTail?]. The main motivation for the previous and current criteria is to figure out under what circumstances will the congestion control algorithm break down and in which environments it may be safe to deploy.
5. Reacting to transient events or interruptions: The congestion control algorithm should be able to handle changes in end-to-end latency due to route updates, link failures, handovers etc. In mobile environment the end-to-end capacity may vary due to the interference, fading, handovers, etc. In wired networks the end-to-end capacity may vary due to changes in resource reservation.
6. Fairness with similar cross-traffic: The congestion control algorithm should be evaluated when competing with other RTP flows using the same congestion control algorithm. The proposal should highlight the bottleneck capacity share of each RTP flow.
7. Impact on cross-traffic: The congestion control algorithm should be evaluated when competing with standard TCP. Short TCP flows may be considered as transient events and the RTP flow may give way to the short TCP flow to complete quickly. However, long TCP flows (also known as elephant flows) may starve out the RTP flow

depending on router queue length. In the latter case the proposed congestion control for RTP should be as aggressive as standard TCP [RFC5681].

8. Extensions to RTP/RTCP: The congestion control algorithm should indicate if any protocol extensions are required to implement it and should carefully describe the impact of the extension.

5. Example Evaluation Scenarios

- o Single RTP flow on a fixed link: This scenario evaluates the ramp-up to the bottleneck capacity and the stability of the proposed congestion control algorithm.
- o Single RTP flow on a variable capacity link: This scenario evaluates the response of the proposed congestion control algorithm to transient network events or due to interference and handovers in mobile environments.
- o Fairness to RTP flows running the same congestion control algorithm: Depending on the levels of
- o Competing with short TCP flows:
- o Competing with long TCP flows:

6. Status of Proposals

Congestion control algorithms are expected to be published as "Experimental" documents until they are shown to be safe to deploy. An algorithm published as a draft should be experimented in simulation, or a controlled environment (testbed) to show its applicability. Every congestion control algorithm should include a note describing the environments in which the algorithm is tested and safe to deploy. It is possible that an algorithm is not recommended for certain environments or perform sub-optimally for the user.

[Editors Note: Should there be a distinction between "Informational" and "Experimental" drafts for congestion control algorithms in RMCAT. [RFC5033] describes Informational proposals as algorithms that are not safe for deployment but are proposals to experiment with in simulation/testbeds. While Experimental algorithms are ones that are deemed safe in some environments but require a more thorough evaluation (from the community).]

7. Security Considerations

Security issues have not been discussed in this memo. However, sending more media data than the available end-to-end capacity may cause a congestion collapse.

8. IANA Considerations

There are no IANA impacts in this memo.

9. References

9.1. Normative References

[RFC3550] Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, RFC 3550, July 2003.

[RFC3551] Schulzrinne, H. and S. Casner, "RTP Profile for Audio and Video Conferences with Minimal Control", STD 65, RFC 3551, July 2003.

[RFC3611] Friedman, T., Caceres, R., and A. Clark, "RTP Control Protocol Extended Reports (RTCP XR)", RFC 3611, November 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.

[RFC5506] Johansson, I. and M. Westerlund, "Support for Reduced-Size Real-Time Transport Control Protocol (RTCP): Opportunities and Consequences", RFC 5506, April 2009.

[I-D.jesup-rtp-congestion-reqs] Jesup, R. and H. Alvestrand, "Congestion Control Requirements For Real Time Media", draft-jesup-rtp-congestion-reqs-00 (work in progress), March 2012.

[I-D.ietf-avtcore-rtp-circuit-breakers] Perkins, C. and V. Singh, "RTP Congestion Control: Circuit Breakers for Unicast Sessions", draft-ietf-avtcore-rtp-circuit-breakers-00 (work in progress), October 2012.

9.2. Informative References

[RFC5033] Floyd, S. and M. Allman, "Specifying New Congestion Control Algorithms", BCP 133, RFC 5033, August 2007.

[RFC5166] Floyd, S., "Metrics for the Evaluation of Congestion

Control Mechanisms", RFC 5166, March 2008.

[RFC5681] Allman, M., Paxson, V., and E. Blanton, "TCP Congestion Control", RFC 5681, September 2009.

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