RMCAT WG
Internet-Draft
Intended status: Informational
Expires: September 11, 2014
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March 10, 2014

Evaluating Congestion Control for Interactive Real-time Media
draft-ietf-rmcat-eval-criteria-01

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

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Table of Contents

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.ietf-rmcat-cc-requirements]). 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 help prevent a congestion collapse, promote fair capacity usage and optimize the media flow's throughput. 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

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. Metrics that are of interest for interactive multimedia are:

- Throughput.
- Minimizing oscillations in the transmission rate (stability) when the end-to-end capacity varies slowly.
- Delay.
- Reactivity to transient events.
- Packet losses and discards.
- Section 2.1 of [RFC5166] discusses the tradeoff between throughput, delay and loss.

Each experiment is expected to log every incoming and outgoing packet (the RTP logging format is described in Section 3.1). The logging can be done inside the application or at the endpoints using pcap (packet capture, e.g., tcpdump, wireshark). The following are calculated based on the information in the packet logs:

1. Sending rate, Receiver rate, Goodput
2. Packet delay

3. Packet loss

4. If using, retransmission or FEC: residual loss

5. Packets discarded from the playout or de-jitter buffer

[Open issue (1): The "unfairness" test is (measured at 1s intervals):
1. Does not trigger the circuit breaker.
2. Over 3 times or less than 1/3 times the throughput for an RMCAT media stream compared to identical RMCAT streams competing on a bottleneck, for a case when the competing streams have similar RTTs.
3. Over 3 times delay compared to RTT measurements performed before starting the RMCAT flow or for the case when competing with identical RMCAT streams having similar RTTs.
]

[Open issue (2): Possibly using Jain-fairness index.]

Convergence time: the time taken to reach a stable rate at startup, after the available link capacity changes, or when new flows get added to the bottleneck link.

Bandwidth Utilization, defined as ratio of the instantaneous sending rate to the instantaneous bottleneck capacity. This metric is useful when an RMCAT flow is by itself or competing with similar cross-traffic.

From the logs the statistical measures (min, max, mean, standard deviation and variance) for the whole duration or any specific part of the session can be calculated. Also the metrics (sending rate, receiver rate, goodput, latency) can be visualized in graphs as variation over time, the measurements in the plot are at 1 second intervals. Additionally, from the logs it is possible to plot the histogram or CDF of packet delay.

3.1. RTP Log Format

The log file is tab or comma separated containing the following details:

```
        Send or receive timestamp (unix)
        RTP payload type
        SSRC
        RTP sequence no
        RTP timestamp
        marker bit
        payload size
```

If the congestion control implements, retransmissions or FEC, the evaluation should report both packet loss (before applying error-resilience) and residual packet loss (after applying error-resilience).

4. Guidelines

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

4.1. Avoiding Congestion Collapse

The congestion control algorithm is expected to take an action, such as reducing the sending rate, when it detects congestion. Typically, it should intervene before the circuit breaker [I-D.ietf-avtcore-rtp-circuit-breakers] is engaged.

Does the congestion control propose any changes to (or diverge from) the circuit breaker conditions defined in [I-D.ietf-avtcore-rtp-circuit-breakers].

4.2. Stability

The congestion control should be assessed for its stability when the path characteristics do not change over time. Changing the media encoding rate estimate too often or by too much may adversely affect the application layer performance.

4.3. Media Traffic

The congestion control algorithm should be assessed with different types of media behavior, i.e., the media should contain idle and data-limited periods. For example, periods of silence for audio, varying amount of motion for video, or bursty nature of I-frames.

The evaluation may be done in two stages. In the first stage, the endpoint generates traffic at the rate calculated by the congestion controller. In the second stage, real codecs or models of video codecs are used to mimic application-limited data periods and varying video frame sizes.

4.4. Start-up Behaviour

The congestion control algorithm should be assessed with different start-rates. The main reason is to observe the behavior of the congestion control in different test scenarios, such as when competing with varying amount of cross-traffic or how quickly does the congestion control algorithm achieve a stable sending rate.

4.5. Diverse Environments

The congestion control algorithm should be assessed in heterogeneous environments, containing both wired and wireless paths. Examples of wireless access technologies are: 802.11, GPRS, HSPA, or LTE. One of the main challenges of the wireless environments for the congestion control algorithm is to distinguish between congestion induced loss and transmission (bit-error) loss. Congestion control algorithms may incorrectly identify transmission loss as congestion loss and reduce the media encoding rate by too much, which may cause oscillatory behavior and deteriorate the users' quality of experience. Furthermore, packet loss may induce additional delay in networks with wireless paths due to link-layer retransmissions.

4.6. 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 bottleneck link and a router's queue length. For the moment, only DropTail queues are used. However, if new Active Queue Management (AQM) schemes become available, the performance of the congestion control algorithm should be again evaluated.

In an experiment, if the media only flows in a single direction, the feedback path should also be tested with varying amounts of impairments.

The main motivation for the previous and current criteria is to identify situations in which the proposed congestion control is less performant.

4.7. Reacting to Transient Events or Interruptions

The congestion control algorithm should be able to handle changes in end-to-end capacity and latency. Latency may change 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.

4.8. Fairness With Similar Cross-Traffic

The congestion control algorithm should be evaluated when competing with other RTP flows using the same or another candidate congestion control algorithm. The proposal should highlight the bottleneck capacity share of each RTP flow.

4.9. 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-lived TCP flows may starve out the RTP flow depending on router queue length.

The proposal should also measure the impact on varied number of cross-traffic sources, i.e., few and many competing flows, or mixing various amounts of TCP and similar cross-traffic.

4.10. 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. Minimum Requirements for Evaluation

[Editor’s Note: If needed, a minimum evaluation criteria can be based on the above guidelines or defined tests/scenarios.]

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.

[Editor’s 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.

8. IANA Considerations

There are no IANA impacts in this memo.

9. Contributors

The content and concepts within this document are a product of the discussion carried out in the Design Team.

Michael Ramalho provided the text for a specific scenario, which is now covered in [I-D.sarker-rmcat-eval-test].

10. Acknowledgements

Much of this document is derived from previous work on congestion control at the IETF.

The authors would like to thank Harald Alvestrand, Luca De Cicco, Wesley Eddy, Lars Eggert, Kevin Gross, Vinayak Hegde, Stefan Holmer, Randell Jesup, Piers O'Hanlon, Colin Perkins, Michael Ramalho, Zaheduzzaman Sarker, Timothy B. Terriberry, Michael Welzl, and Mo Zanaty for providing valuable feedback on earlier versions of this draft. Additionally, also thank the participants of the design team for their comments and discussion related to the evaluation criteria.

11. References

11.1. Normative References

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[RFC3611] Friedman, T., Caceres, R., and A. Clark, "RTP Control Protocol Extended Reports (RTCP XR)", RFC 3611, November 2003.

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11.2. Informative References

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[SA4-LR] S4-050560, 3GPP., "Error Patterns for MBMS Streaming over UTRAN and GERAN", 3GPP S4-050560, 5 2008.

[TCP-eval-suite]
Lachlan, A., Marcondes, C., Floyd, S., Dunn, L., Guillier, R., Gang, W., Eggert, L., Ha, S., and I. Rhee, "Towards a Common TCP Evaluation Suite", Proc. PFLDnet. 2008, August 2008.

Appendix A. Application Trade-off

Application trade-off is yet to be defined. see RMCAT requirements [I-D.ietf-rmcat-cc-requirements] document. Perhaps each experiment should define the application's expectation or trade-off.

A.1. Measuring Quality

No quality metric is defined for performance evaluation, it is currently an open issue. However, there is consensus that congestion control algorithm should be able to show that it is useful for interactive video by performing analysis using a real codec and video sequences.

Appendix B. Change Log

Note to the RFC-Editor: please remove this section prior to publication as an RFC.

B.1. Changes in draft-ietf-rmcat-eval-criteria-01

o Removed Appendix B.

o Removed Section on Evaluation Parameters.

B.2. Changes in draft-ietf-rmcat-eval-criteria-00

o Updated references.

o Resubmitted as WG draft.

B.3. Changes in draft-singh-rmcat-cc-eval-04

o Incorporate feedback from IETF 87, Berlin.

o Clarified metrics: convergence time, bandwidth utilization.

o Changed fairness criteria to fairness test.

o Added measuring pre- and post-repair loss.

o Added open issue of measuring video quality to appendix.

o clarified use of DropTail and AQM.

o Updated text in "Minimum Requirements for Evaluation"

B.4. Changes in draft-singh-rmcat-cc-eval-03

- Incorporate the discussion within the design team.

- Added a section on evaluation parameters, it describes the flow and network characteristics.

- Added Appendix with self-fairness experiment.

- Changed bottleneck parameters from a proposal to an example set.

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B.5. Changes in draft-singh-rmcat-cc-eval-02

- Added scenario descriptions.

B.6. Changes in draft-singh-rmcat-cc-eval-01

- Removed QoE metrics.

- Changed stability to steady-state.

- Added measuring impact against few and many flows.

- Added guideline for idle and data-limited periods.

- Added reference to TCP evaluation suite in example evaluation scenarios.

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