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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.

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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

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 important to interactive multimedia are:

* Delay
* Throughput
* Minimizing oscillations in encoding rate (stability)
* Reactivity to transient events
* Packet loss and discard rate
* Users' quality of experience

[Editor's Note: measurement interval and statistical measures (min, max, mean, median) are yet to be specified.]

Section 2.1 of [RFC5166] discusses the tradeoff between throughput, delay and loss.

(i) Bandwidth Utilization: is the ratio of the encoding rate to the (available) end-to-end path capacity.

* Under-utilization: is the period of time when the endpoint’s encoding rate is lower than the end-to-end capacity, i.e., the bandwidth utilization is less than 1.
* Overuse: is the period of time when the endpoint’s encoding rate is higher than the end-to-end capacity, i.e., the bandwidth utilization is greater than 1.
* Stability: is the period of time when the endpoint’s encoding rate is relatively stable, i.e., the bandwidth utilization is constant.

(ii) Packet Loss and Discard Rate.

(iii) Fair Share.
[Editor’s Note: This metric should match the one defined in the RMCAT requirements [I-D.jesup-rtp-congestion-reqs] document.]

(iv) Quality: There are many different types of quality metrics for audio and video. Audio quality is often expressed by a MOS ("Mean Opinion Score") and can be calculated using an objective algorithm [I-D.ietf-xrblock-rtcp-xr-qoe]. Section 4.7 of [RFC3611] can also be used for VoIP metrics. Similarly, there exist several metrics to measure video quality, for example Peak Signal to Noise Ratio (PSNR).
[Editor’s Note: Should the algorithm compare average PSNR of test video sequences or what other video quality metric can be used?]

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

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 too often or by too much may adversely affect the users' quality of experience.

4.3. Diverse Environments

The congestion control algorithm should be assessed in heterogeneous environments, containing both wired and wireless paths. Examples of wireless access technologies are: 802.11x, 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. Congestion control algorithms may incorrectly identify transmission loss as congestion loss and reduce the media encoding rate 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.

4.4. 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 determine under which circumstances will the proposed congestion control algorithm break down and also determine the operational range of the algorithm.

4.5. 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.

4.6. 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.

4.7. 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-lived TCP 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].

4.8. 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. Minimum Requirements for Evaluation

[Editor’s Note: If needed, a minimum evaluation criteria can be based on the above guidelines]

6. Example Evaluation Scenarios

In the scenarios listed below, all RTP flows are bi-directional and point-to-point.

[S1] 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.

[S2] RTP flow on a variable capacity link: This scenario evaluates the reactivity of the proposed congestion control algorithm to transient network events due to interference and handovers in mobile environments. Sample 3G bandwidth traces are available at [3GPP.R1.081955].

[S3] Fairness to RTP flows running the same congestion control algorithm: This scenario shows if the proposed algorithm can share the bottleneck link equitably, irrespective of number of flows.

[S4] Competing with long-lived TCP flows: In this scenario the proposed algorithm is expected to be TCP-friendly, i.e., it should neither starve out the competing TCP flows (causing a congestion collapse) nor should it be starved out by TCP.

[S5] Competing with short TCP flows: Depending on the level of statistical multiplexing on the bottleneck link, the proposed algorithm may behave differently. If there are a few short TCP flows then the proposed algorithm may observe these flows as transient events and let them complete quickly. Alternatively, if there are many short flows then the proposed algorithm may have to compete with the flows as if they were long lived TCP flows. [Editor's Note: many and few short TCP flows may depend on the bottleneck link capacity.]

7. 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).]

8. Security Considerations

Security issues have not been discussed in this memo.

9. IANA Considerations

There are no IANA impacts in this memo.

10. Acknowledgements

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

11. References

11.1. Normative References

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11.2. Informative References

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