Evaluating Congestion Control for Interactive Real-time Media.
draft-singh-rmcat-cc-eval-01.txt

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

   This Internet-Draft is submitted in full conformance with the
   provisions of BCP 78 and BCP 79.

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   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 April 25, 2013.

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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.
* Steady-state: 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.]

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. 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 or
varying amount of motion for video.

4.4. 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.

4.5. 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. 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.

[Editor's Note: Different types of queueing mechanisms? Random Early
Detection or only DropTail?].

4.6. 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.

4.7. 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.

4.8. 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].

   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.

4.9. 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. [TCP-eval-suite] contains examples of TCP traffic
   load and scenario settings.

      [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: definition of many and few short TCP flows may depend on the bottleneck link capacity.]

[Editor's Note: clarify if media packets are generated using a traffic generator.]

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.

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   valuable feedback on earlier versions of this draft.

11. References

11.1. Normative References

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11.2. Informative References

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Appendix A. Change Log

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

A.1. Changes in draft-singh-rmcat-cc-eval-01

   o Removed QoE metrics.
   o Changed stability to steady-state.
   o Added measuring impact against few and many flows.
   o Added guideline for idle and data-limited periods.
   o Added reference to TCP evaluation suite in example evaluation
      scenarios.

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