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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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Table of Contents

1. Introduction	3
2. Terminology	3
3. Metrics	3
3.1. RTP Log Format	5
4. Guidelines	5
4.1. Avoiding Congestion Collapse	5
4.2. Stability	5
4.3. Media Traffic	5
4.4. Start-up Behaviour	6
4.5. Diverse Environments	6
4.6. Varying Path Characteristics	6
4.7. Reacting to Transient Events or Interruptions	7
4.8. Fairness With Similar Cross-Traffic	7
4.9. Impact on Cross-Traffic	7
4.10. Extensions to RTP/RTCP	7
5. Minimum Requirements for Evaluation	7
6. Evaluation Parameters	7
6.1. Bottleneck Traffic Flows	8
6.2. Access Links	8
6.3. Bottleneck Link Parameters	9
6.4. Router Queue Parameters	10
6.5. Media Flow Parameters	10
6.6. Cross-traffic Parameters	11
7. Status of Proposals	11
8. Security Considerations	11
9. IANA Considerations	11
10. Contributors	12
11. Acknowledgements	12
12. References	12
12.1. Normative References	12
12.2. Informative References	13
Appendix A. Proposal to evaluate Self-fairness of RMCAT congestion control algorithm	13
A.1. Evaluation Parameters	15
A.1.1. Media Traffic Generator	15
A.1.2. Bottleneck Link Bandwidth	15
A.1.3. Bottleneck Link Queue Type and Length	15
A.1.4. RMCAT flows and delay legs	16
A.1.5. Impairment Generator	16
A.2. Proposed Passing Criteria for Experiment	16
A.3. Extensability	17
Appendix B. Change Log	17
B.1. Changes in draft-singh-rmcat-cc-eval-03	17
B.2. Changes in draft-singh-rmcat-cc-eval-02	17
B.3. Changes in draft-singh-rmcat-cc-eval-01	18
Authors' Addresses	18

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-rmcat-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, 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:

- o Throughput
- o Minimizing oscillations in encoding rate (stability)
- o Delay
- o Reactivity to transient events
- o Packet losses and discards.
- o Users' quality of experience

Each experiment logs every incoming and outgoing packet. The logging can be done inside the application or at the endpoints using pcap (packet capture, e.g., tcpdump, wireshark). The following are measured based on the information in the logs:

1. Sending rate, Receiver rate, Goodput
2. Packet latency
3. Packet loss
4. Post-repair loss: residual loss after retransmission or FEC.
5. Packets discarded from the playout or de-jitter buffer

[Open issue (1): Instead of defining fairness, there has been discussion on defining "unfairness". The criteria are:

1. Do not trigger the circuit breaker.
2. Over 3 times or less than 1/3 times the throughput for a stream compared to similar competing streams on a bottleneck. The competing streams have similar RTTs.
3. Over 3 times delay compared to RTT measurements performed before starting the flow.

Here, rather than discussing the number, do people agree with the criteria?]

[Open issue (2): Convergence time was discussed briefly in the meetings as well. It is defined as: time it takes the congestion control to reach a stable rate (at startup or after new RMCAT flows are added). What is stable rate?]

[Open issue (3): previous versions of the document had Bandwidth Utilization, defined as ratio of sending rate to the available bottleneck capacity. This makes sense when the RMCAT flow is by itself or competing with similar flows (where the assumption would be that all flows get an equal share). rate fairness may not always be the case, so remove this?]

From the logs the statistical measures (min, max, mean, standard deviation and variance) for the whole duration of the call can be calculated for each metric.

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 should be possible to plot the histogram or CDF of delay.

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

[Open issue (4): Application trade-off is yet to be defined. see RMCAT requirements [I-D.jesup-rmcat-reqs] document.]

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

[Open issue (5): Should the retransmissions and FEC for post-repair loss metric be logged in a separate file? the repair streams have different payload type and/or SSRC.]

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. However, the evaluation can be done in two stages. In the first stage, media stream can generate traffic at the rate calculated by the congestion controller. In the second stage, real codecs or models of video codecs should be used to mimic real-world cases.

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 evaluation scenarios, such as when competing with varying amount of cross-traffic or how quickly does the congestion control algorithm achieve a stable sending rate.

[Editor's note: requires a robust definition for unfriendliness and convergence time.]

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 corruption) 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 bottle-neck link and a router's queue length. 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.

[Open issue (6): Different types of queueing mechanisms? Random Early Detection or only DropTail?].

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.

[Editor's note: If we define Unfriendliness then that criteria should be applied here.]

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]

6. Evaluation Parameters

An evaluation scenario is created from a list of network, link and flow characteristics.

6.1. Bottleneck Traffic Flows

The network scenario describes the types of flows sharing the common bottleneck, they are:

1. Single RMCAT flow.
2. Competing with similar RMCAT flows. These competing flows may use the same algorithm or another candidate RMCAT algorithm.
3. Compete with long-lived TCP.
4. Compete with bursty TCP.
5. Compete with LEDBAT flows.
6. Compete with unresponsive interactive media flows (i.e., not only CBR).

Figure 1 shows an example evaluation topology, where S1..Sn are traffic sources, these sources are either RMCAT or a mixture of traffic flows listed above. R1..Rn are the corresponding receivers. A and B are routers that can be configured to introduce impairments. Access links are in between the sender/receiver and the router, while the bottleneck link is between the Routers A and B.

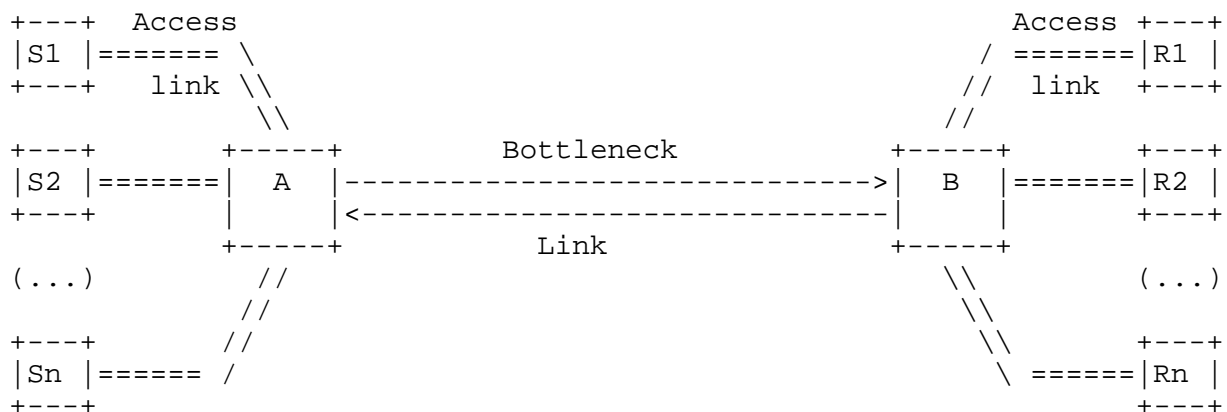


Figure 1: Simple Topology

[Open Issue (7): Discuss more complex topologies]

6.2. Access Links

The media senders and receivers are typically connected to the bottleneck link over high bandwidth, low latency links. They are:

1. Ethernet (LAN)
2. Wireless LAN (WLAN)
3. 3G/LTE

[Open issue (8): need to describe parameters or traces to model WLAN and 3G/LTE.]

A real-world network typically consists of a mixture of links, the most important aspect in this case is to identify the location of the bottleneck link. Depending on amount of cross-traffic or due to the varying link capacity will cause the bottleneck link to move from one node to another. The design of the experiments should take this into account and in the simplest case the access link may not be the bottleneck link.

6.3. Bottleneck Link Parameters

The bottleneck link carries multiple flows, these flows may be other RMCAT flows or other types of cross-traffic. The performance of the congestion control is measured in terms of unfairness (occupancy of the bottleneck link capacity), variation in end-to-end delay due to queuing, etc. The experiments should dimension the bottleneck link based on the number of flows and the expected behavior. For example, if 5 media flows are expected to share the bottleneck link equally, the bottleneck link is set to 5 times the media rate.

If the experiment carries only media in one direction, then the upstream bottleneck link carries media packets while the downstream bottleneck carries the feedback packets. The bottleneck link parameters discussed in this section apply only to a single direction, hence bottleneck link in the reverse direction can choose to have the same or different parameters.

The link latency corresponds to the propagation delay, the experiments are expected to verify that the congestion control are able to work in challenging situations, for example over trans-continental and/or satellite links. The experiment should pick one of the following:

1. Very low delay: 1ms
2. Low delay: 50ms
3. High delay: 150ms
4. Extreme delay: 300ms

Similarly, to model lossy links, the experiments can choose one of the following loss rates, the fractional loss is ratio of packets lost and packets sent.

1. no loss: 0%
2. 1%
3. 5%
4. 10%
5. 20%

[Open issue (10): how is the loss generated? using traces, Gilbert-Elliot model, randomly (uncorrelated) loss.]

6.4. Router Queue Parameters

The router queue length is measured as the time taken to drain the FIFO queue, they are:

1. QoS-aware (or short): 70ms
2. Nominal: 500ms
3. Buffer-bloated: 2000ms

However, the size of the queue is typically measured in bytes or packets and to convert the queue length measured in seconds to 1) queue length in bytes:

$\text{QueueSize (in bytes)} = \text{QueueSize (in sec)} \times \text{Throughput (in bps)} / 8$

and 2) queue length in packets:

$\text{QueueSize (in pkts)} = \text{QueueSize (in bytes)} / \text{MTU}, \text{MTU}=1500$

[Open issue (11): define parameters for other types of queues?]

6.5. Media Flow Parameters

The media sources can be modeled in two ways. In the first, the sources always have data to send, i.e., have no data limited intervals and are able to generate the media rate requested by the RMCAT congestion control algorithm. In the second, the traffic generator models the behavior of a media codec, mainly the burstiness.

At the beginning of the session, the media sources are configured to start at a given start rate, use one of the following start rates:

1. 200 kbps
2. 800 kbps
3. 1300 kbps
4. 4000 kbps

The scenario described in Appendix A takes into account these network and flow considerations.

6.6. Cross-traffic Parameters

[Open issue(12): TCP cross-traffic parameters are still TBD, mainly the bursty TCP. Long-lived TCP flows will download data throughout the session and are expected to have infinite amount of data to send or receive.]

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. 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 the scenario discussed in Appendix A.

11. Acknowledgements

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

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12. References

12.1. 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-rmcat-reqs]

Jesup, R., "Congestion Control Requirements For RMCAT", draft-jesup-rmcat-reqs-01 (work in progress), February 2013.

[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-01 (work in progress), October 2012.

12.2. 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.

[SA4-EVAL]

R1-081955, 3GPP., "LTE Link Level Throughput Data for SA4 Evaluation Framework", 3GPP R1-081955, 5 2008.

[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. Proposal to evaluate Self-fairness of RMCAT congestion control algorithm

The goal of the experiment discussed in this section is to initially take out as many unknowns from the experiments. Later experiments can define more complex environments, topologies and media behavior. This experiment evaluates the performance of the RMCAT sender competing with other similar RMCAT flows (running the same algorithm or other RMCAT proposals) on the bottleneck link. There are up to 20 RMCAT flows competing for capacity on the bottleneck link, but media only flows in one direction, from senders (S1..S20) to receivers (R1..R20) and the feedback packets flow in the reverse direction.

Figure 2 shows the experiment setup and compared to the simple topology in Figure 1 it has a maximum of 20 flows. Groups of 10

receivers are connected to the bottleneck link through two different routers (Router C and D). The rationale for adding these additional routers is to create two delay legs, i.e., two groups of endpoints with different network latencies and measure the performance of the RMCAT congestion control algorithm. If fewer than 10 sources are initialized, all traffic flows experience the same delay because they share the same delay leg.

Router A has a single forward direction bottleneck link (i.e., the bottleneck capacity and delay constraints applies only to the media packets going from the sender to the receiver, the feedback packets are unaffected). Hence, the Round-Trip Time (RTT) is primarily composed of the bottleneck queue delay and any forward path (propagation) latency. The main reason for not applying any constraints on the return path is to provide the best-case performance scenario for the congestion control algorithm. In later experiments, it is possible to add similar capacity and delay constraints on the return path.

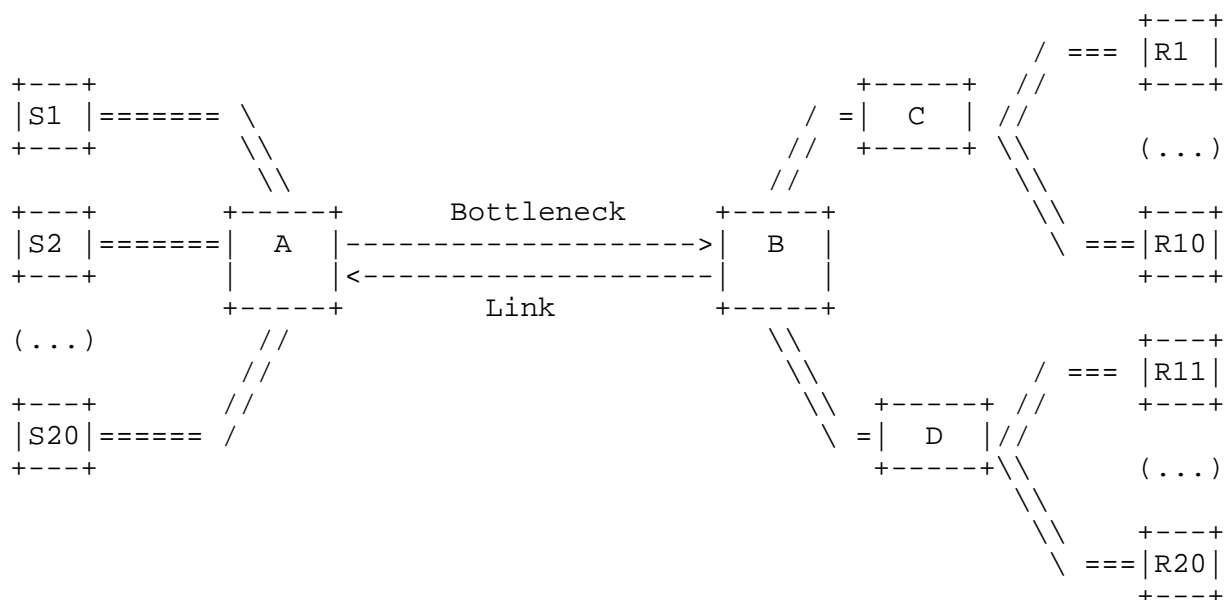


Figure 2: Self-fairness Evaluation Setup

Loss impairments are applied at Router C and Router D, but only to the feedback flows. If the losses are set to 0%, it represents a case where the return path is over-provisioned for all traffic. In later experiments the loss impairments can be added to the media path as well.

The media sources are configured to send infinite amount of data, i.e., the sources always have data to send and have no data limited intervals. Additionally, the media sources are always successful in generating the media rate requested by the RMCAT congestion control algorithm. In this experiment, we avoid the potentially complicated scenario of using media traffic generators that try to model the behavior of media codecs (mainly the burstiness).

A.1. Evaluation Parameters

A.1.1. Media Traffic Generator

The media source always generates at the rate requested by the congestion control. Furthermore, the media packet generator is subject to the following constraints:

1. It MUST emit a packet at least once per 100 ms time interval.
2. For low media rate source: when generating data at a rate less than a maximum length MTU every 100 ms would allow (e.g., 120 kbps = 1500 bytes/packet * 10 packets/sec * 8 bits/byte for 1500 byte packets), the RMCAT source must modulate the packet size (RTP payload size) of RTP packets that are sent every 100 ms to attain the desired rate.
3. For high media rate sources: when generating data at a rate greater than a maximum length MTU every 100 ms would allow, the source must do so by sending (approximately) maximum MTU sized packets and adjusting the inter-departure interval to be approximately equal. The intent of this to ensure the data is sent relatively smoothly independent of the bitrate, subject to the first constraint.

A.1.2. Bottleneck Link Bandwidth

The bottleneck link bandwidth is dimensioned such that each RMCAT flow in an ideal situation (with perfectly equal capacity sharing for all the flows on the bottleneck) obtains the following throughputs: 200 kbps, 800 kbps, 1.3 Mbps and 4 Mbps.

For example, experiments with five RMCAT flows with an 800 kbps/flow target rate should set the bottleneck link capacity to 4 Mbps.

A.1.3. Bottleneck Link Queue Type and Length

The bottleneck link queue (Router A) is a simple FIFO queue having a buffer length corresponding to 70 ms, 500 ms or 2000 ms of delay at the bottleneck link rate (i.e., actual buffer lengths in bytes are dependent on bottleneck link bandwidth).

A.1.4. RMCAT flows and delay legs

Experiments run with 1, 3, 5, 10 and 20 RMCAT sources, they are outlined as follows:

1. Experiments with 1, 3, and 5 RMCAT flows, all RMCAT flows commence simultaneously. A single delay leg is used and the link latency is set to one of the following : 0 ms, 50 ms and 150 ms.
2. For 10 and 20 source experiments where all RMCAT flows begin simultaneously the sources are split evenly into two different bulk delay legs. One leg is set to 0 ms bulk delay leg and the other is set to 150 ms.
3. For 10 and 20 source experiments where the first set will use 0 ms of bulk delay and the second set will use 150 ms bulk delay.
 1. Random starts within interval [0 ms, 500 ms].
 2. One "early-coming" flow (i.e., the 1st flow starting and achieving steady-state before the next N-1 simultaneously begin).
 3. One "late-coming" flow (i.e., the Nth flow starting after steady-state has occurred for the existing N-1 flows).

These cases assess if there are any early or late-comer advantages or disadvantages for a particular algorithm and to see if any unfairness is reproducible or unpredictable.

[Open issue (A.1): which group does the early and late flow belong to?]

[Open issue (A.2): Start rate for the media flows]

A.1.5. Impairment Generator

Packet loss is created in the reverse path (affects only feedback packets). Cases of 0%, 1%, 5% and 10% are studied for the 1, 3, and 5 RMCAT flow experiments, losses are not applied to flows with 10 or 20 RMCAT flows.

A.2. Proposed Passing Criteria for Experiment

[Editor's note: there has been little or no discussion on the below criteria, however, they are listed for the sake of completeness]

No unfairness is observed, i.e., at steady state each flow attains a throughput between $[B/(3*N), (3*B)/N]$, where B is the link bandwidth and N is the number of flows.

No flow experiences packet loss when queue length is set to 500 ms or greater

All individual sources must be in their steady state within twenty LRTTs (where LRTT is defined as the RTT associated with the flow with the Largest RTT in the experiment).]

A.3. Extensability

The above scenario describes only RMCAT sources competing for capacity on the bottleneck link, however, future experiments can use different types of cross-traffic (as described in Section 6.1).

Currently, the forward path (carrying media packets) is characterized to add delay and a fixed bottleneck link capacity, in the future packet losses and capacity changes can be applied to mimic a wireless link layer (for e.g., WiFi, 3G, LTE). Additionally, only losses are applied to the reverse path (carrying feedback packets), later experiments can apply the same forward path (carrying media packets) impairments to the reverse path.

Appendix B. Change Log

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

B.1. Changes in draft-singh-rmcat-cc-eval-03

- o Incorporate the discussion within the design team.
- o Added a section on evaluation parameters, it describes the flow and network characteristics.
- o Added Appendix with self-fairness experiment.

B.2. Changes in draft-singh-rmcat-cc-eval-02

- o Added scenario descriptions.

B.3. 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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