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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 . . . . .	6
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 . . . . .	15
A.1.5. Impairment Generator . . . . .	16
A.2. Proposed Passing Criteria . . . . .	16
A.3. Extensability of the Experiment . . . . .	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 . . . . .	17
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 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 important to interactive multimedia are:

- o Throughput.
- o Minimizing oscillations in the transmission rate (stability) when the end-to-end capacity varies slowly.
- o Delay.
- o Reactivity to transient events.
- o Packet losses and discards.

Each experiment logs 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. Packets discarded from the playout or de-jitter buffer

[Editor's note: How to handle packet re-transmissions? loss before retransmission, after retransmission?]

[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 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.

Here, rather than discussing the number '3'? Does the criteria capture Unfairness adequately?]

[Open issue (2): Convergence time was discussed briefly in the design meetings. It is defined as: the time it takes the congestion control to reach a stable rate (at startup or after new RMCAT flows are added). What is a 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 is useful when the RMCAT flow is by itself or competing with similar flows (where the assumption would be that all flows get an equal share). Remove this?]

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.

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. Perhaps each experiment should define the application's expectation or trade-off.]

### 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 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 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 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. The parameters discussed in the following subsections are meant to aid in creating evaluation scenarios and do not describe an evaluation scenario. The scenario discussed in Appendix A takes into account all these parameters.

6.1. Bottleneck Traffic Flows

The network scenario describes the types of flows sharing the common bottleneck with a single RMCAT flow, they are:

1. A single RMCAT flow by itself.
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 LEBBAT 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.

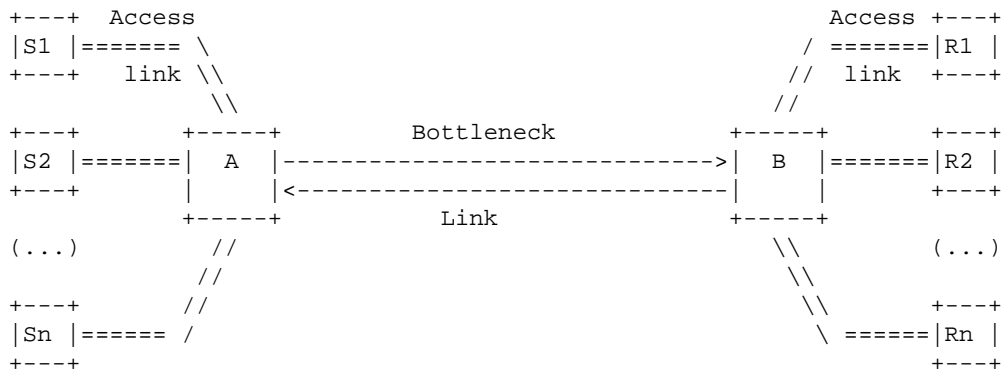


Figure 1: Simple Topology

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

6.2. Access Links

The media senders and receivers are typically connected to the bottleneck link, common access links 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 is to identify the location of the bottleneck link. The bottleneck link can move from one node to another depending on the amount of cross-traffic or due to the varying link capacity. The design of the experiments should take this into account. In the simplest case the access link may not be the bottleneck link but an intermediate node.

### 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 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 desired transmission rate.

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

The link latency corresponds to the propagation delay of the link, i.e., the time it takes for a packet to traverse the bottleneck link, it does not include queuing delay. In an experiment with several links the experiment should describe if the links add latency or not. It is possible for experiments to have multiple hops with different link latencies. Experiments are expected to verify that the congestion control is able to work in challenging situations, for example over trans-continental and/or satellite links. The experiment should pick link latency values from the following:

1. Very low latency: 0-1ms
2. Low latency: 50ms

3. High latency: 150ms
4. Extreme latency: 300ms

[Editor's note: currently describes the latency for a single link, instead of end-to-end delay. Which is preferred? or both?]

Similarly, to model lossy links, the experiments can choose one of the following loss rates, the fractional loss is the 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-Elliott 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 queue length in bytes:

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

[Open issue (11): Confirm the above values, do we need to 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 (time-varying data produced by a video GOP).

At the beginning of the session, the media sources are configured to start at a given start rate, they are:

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

#### 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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## 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 scenario. 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, but the media only flows in one direction, from senders (S1..S20) to receivers (R1..R20) and the feedback packets flow in the reverse direction.



## A.1. Evaluation Parameters

### A.1.1. Media Traffic Generator

The media source always generates at the rate requested by the congestion control and has infinite data to send. 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), 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 is to ensure the data is sent relatively smoothly independent of the bit rate, subject to the first constraint.

### A.1.2. Bottleneck Link Bandwidth

The bottleneck link capacity 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 (defined in Section 6.4) 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

[Editor's note: there has been little or no discussion on the below criteria, however, they are listed here 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 of the Experiment

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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