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Evaluating Congestion Control for Interactive Real-time Media
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Abstract

Status of This Memo

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1. Introduction

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. 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.

Figure 1 shows the experiment setup and it has subtle differences compared to the simple topology in Figure 2. 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.

```
                                                               +---+
                                                         / === |R1 |
   +---+                                           +-----+  //     +---+
   |S1 |======= \                               / =|  C  | //
   +---+        \\                             //  +-----+ \\      (...)
             \\                           //            \\
   +---+       +-----+      Bottleneck     +-----+           \\    +---+
   |S2 |=======|  A  |-------------------->|  B  |            \ ===|R10|
   +---+       |     |<--------------------|     |                 +---+
           +-----+        Link         +-----+
   (...)         //                           \\                   +---+
            //                             \\            / === |R11|
   +---+       //                               \\  +-----+ //     +---+
   |S20|====== /                                 \ =|  D  |//
   +---+                                            +-----+\\      (...)
                                                        \\
                                                         \\    +---+
                                                          \ ===|R20|
                                                               +---+
```

Figure 1: 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).

3.1. Evaluation Parameters

3.1.1. 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 to ensure the data is sent relatively smoothly independent of the bit rate, subject to the first constraint.

3.1.2. 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.

3.1.3. 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 4.4) of delay at the bottleneck link rate (i.e., actual buffer lengths in bytes are dependent on bottleneck link bandwidth).

3.1.4. 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]

3.1.5. 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.

3.2. 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).]

3.3. Extensibility 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 4.1).

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.

4. Evaluation Parameters

An evaluation scenario is created from a list of network, link and flow characteristics. The example 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 Section 3 takes into account all these parameters.

4.1. 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 LEDBAT flows.

6. Compete with unresponsive interactive media flows (i.e., not only CBR).

Figure 2 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.

```
+---+  Access                                          Access +---+
|S1 |======= \                                        / =======|R1 |
+---+   link \\                                     //  link  +---+
          \\                                     //
+---+      +-----+         Bottleneck          +-----+      +---+
|S2 |=======|  A  |------------------------------>|  B  |=======|R2 |
+---+      |     |<------------------------------|     |      +---+
        +-----+           Link               +-----+
(...)        //                                    \\        (...)
         //                                     \\
+---+      //                                        \\       +---+
|Sn |====== /                                         \ ======|Rn |
+---+                                                         +---+
```

Figure 2: Simple Topology

[Open Issue: Discuss more complex topologies]

4.2. 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: point to a reference containing 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.

4.3. Example 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

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%

These fractional losses can be generated using traces, Gilbert-Elliot model, randomly (uncorrelated) loss.

4.4. DropTail 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:

QueueSize (in bytes) = QueueSize (in sec) x Throughput (in bps)/8

4.5. 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

4.6. Cross-traffic Parameters

Long-lived TCP flows will download data throughout the session and are expected to have infinite amount of data to send or receive.

[Open issue: short-lived/bursty TCP cross-traffic parameters are still TBD.

5. Security Considerations

Security issues have not been discussed in this memo.

6. IANA Considerations

There are no IANA impacts in this memo.

7. Acknowledgements

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

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

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