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Test Cases for Evaluating RMCAT Proposals
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Abstract

The Real-time Transport Protocol (RTP) is used to transmit media in multimedia telephony applications. These applications are typically required to implement congestion control. This document describes the test cases to be used in the performance evaluation of such congestion control algorithms in a controlled environment.

Status of This Memo

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1. Introduction

This memo describes a set of test cases for evaluating congestion control algorithm proposals in controlled environments for real-time interactive media. It is based on the guidelines enumerated in [I-D.ietf-rmcat-eval-criteria] and the requirements discussed in

[I-D.ietf-rmcat-cc-requirements]. The test cases cover basic usage scenarios and are described using a common structure, which allows for additional test cases to be added to those described herein to accommodate other topologies and/or the modelling of different path characteristics. The described test cases in this memo should be

used to evaluate any proposed congestion control algorithm for real-time interactive media.

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. Structure of Test cases

All the test cases in this document follow a basic structure allowing implementers to describe a new test scenario without repeatedly explaining common attributes. The structure includes a general description section that describes the test case and its motivation. Additionally the test case defines a set of attributes that characterize the testbed, for example, the network path between communicating peers and the diverse traffic sources.

- Define the test case:

  * General description: describes the motivation and the goals of the test case.

  * Expected behavior: describes the desired rate adaptation behavior.

  * Define a list of metrics to evaluate the desired behavior: this indicates the minimum set of metrics (e.g., link utilization, media sending rate) that a proposed algorithm needs to measure to validate the expected rate adaptation behavior. It should also indicate the time granularity (e.g., averaged over 10ms, 100ms, or 1s) for measuring certain metrics. Typical measurement interval is 200ms.

o Define testbed topology: every test case needs to define an evaluation testbed topology. Figure 1 shows such an evaluation topology. In this evaluation topology, S1..Sn are traffic sources. These sources generate media traffic and use the congestion control algorithm(s) under investigation. R1..Rn are the corresponding receivers. A test case can have one or more such traffic sources (S) and their corresponding receivers (R). The path from the source to destination is denoted as "forward" and the path from a destination to a source is denoted as "backward". The following basic structure of the test case has been described from the perspective of media generating endpoints

attached on the left-hand side of Figure 1. In this setup, the media flows are transported in forward direction and corresponding feedback/control messages are transported in the backward direction. However, it is also possible to set up the test with media in both forward and backward directions. In that case, unless otherwise specified by the test case, it is expected that the backward path does not introduce any congestion related impairments and has enough capacity to accommodate both media and feedback/control messages. It should be noted that depending on the test cases it is possible to have different path characteristics in either of the directions.

```
   +---+                                                           +---+
   |S1 |====== \                 Forward -->              / =======|R1 |
   +---+       \\                                        //        +---+
                \\                                      //
   +---+       +-----+                               +-----+       +---+
   |S2 |=======|  A  |------------------------------>|  B  |=======|R2 |
   +---+       |     |<------------------------------|     |       +---+
               +-----+                               +-----+
   (...)         //                                     \\         (...)
                //          <-- Backward                 \\
   +---+       //                                         \\       +---+
   |Sn |====== /                                           \ ======|Rn |
   +---+                                                           +---+
```

Figure 1: Example of A Testbed Topology

In a testbed environment with real equipments, there may exist a significant amount of unwanted traffic on the portions of the network path between the endpoints. Some of this traffic may be generated by other processes on the endpoints themselves (e.g., discovery protocols) or by other endpoints not presently under test. Such unwanted traffic should be removed or avoided to the greatest extent possible.

- o Define testbed attributes:

  * Duration: defines the duration of the test in seconds.

  * Path characteristics: defines the end-to-end transport level path characteristics of the testbed for a particular test case. Two sets of attributes describe the path characteristics, one for the forward path and the other for the backward path. The path characteristics for a particular path direction is applicable to all the Sources "S" sending traffic on that path. If only one attribute is specified, it is used for both path

directions, however, unless specified the reverse path has no capacity restrictions and no path loss.

    + Path direction: forward or backward.

    + Minimum bottleneck-link capacity: defines minimum capacity of the end-to-end path

    + Reference bottleneck capacity: defines a reference value for the bottleneck capacity for test cases with time-varying bottleneck capacities. All bottleneck capacities will be specified as a ratio with respect to the reference capacity value.

    + One-way propagation delay: describes the end-to-end latency along the path when network queues are empty, i.e., the time it takes for a packet to go from the sender to the receiver without encountering any queuing delay.

    + Maximum end-to-end jitter: defines the maximum jitter that can be observed along the path.

+ Bottleneck queue type: for example, "tail drop" [RFC7567], Flow Queue -CoDel (FQ-CoDel)[RFC8290], or Proportional Integral controller Enhanced (PIE)[RFC8033].

+ Bottleneck queue size: defines the size of queue in terms of queuing time when the queue is full (in milliseconds).

+ Path loss ratio: characterizes the non-congested, additive, losses to be generated on the end-to-end path. This must describe the loss pattern or loss model used to generate the losses.

* Application-related: defines the traffic source behavior for implementing the test case

  + Media traffic Source: defines the characteristics of the media sources. When using more than one media source, the different attributes are enumerated separately for each different media source.

    - Media type: Video/Voice

    - Media flow direction: forward, backward or both.

    - Number of media sources: defines the total number of media sources

    - Media codec: Constant Bit Rate (CBR) or Variable Bit Rate (VBR)

    - Media source behavior: describes the media encoder behavior. It defines the main parameters that affect the adaptation behavior. This may include but is not limited to:

      o Adaptability: describes the adaptation options. For example, in the case of video it defines the following ranges of adaptation: bit rate, frame rate, video resolution. Similarly, in the case of voice, it defines the range of bit rate adaptation, the sampling rate variation, and the variation in packetization interval.

o Output variation : for a VBR encoder it defines the encoder output variation from the average target rate over a particular measurement interval. For example, on average the encoder output may vary between 5% to 15% above or below the average target bit rate when measured over a 100 ms time window. The time interval over which the variation is specified must be provided.

o Responsiveness to a new bit rate request: the lag in time between a new bit rate request from the congestion control algorithm and actual rate changes in encoder output. Depending on the encoder, this value may be specified in absolute time (e.g. 10ms to 1000ms) or other appropriate metric (e.g. next frame interval time).

More detailed discussions on expected media source behavior, including those from synthetic video traffic sources, is at [I-D.ietf-rmcat-video-traffic-model].

- Media content: describes the chosen video scenario. For example, video test sequences are available at: [xiph-seq] and [HEVC-seq]. Different video scenarios give different distribution of video frames produced by the video encoder. Hence, it is important to specify the media content used in a particular test. If a synthetic video traffic souce [I-D.ietf-rmcat-video-traffic-model] is used, then the synthetic video traffic source needs to configure according to the characteristics of the media content specified.

- Media timeline: describes the point when the media source is introduced and removed from the testbed. For example, the media source may start transmitting immediately when the test case begins, or after a few seconds.

- Startup behavior: the media starts at a defined bit rate, which may be the minimum, maximum bit rate, or a value in between (in Kbps).

+ Competing traffic source: describes the characteristics of the competing traffic source, the different types of competing flows are enumerated in [I-D.ietf-rmcat-eval-criteria].

  - Traffic direction: forward, backward or both.

  - Type of sources: defines the types of competing traffic sources. Types of competing traffic flows are listed in [I-D.ietf-rmcat-eval-criteria]. For example, the number of TCP flows connected to a web browser, the mean size and distribution of the content downloaded.

  - Number of sources: defines the total number of competing sources of each media type per traffic direction.

  - Congestion control: enumerates the congestion control used by each type of competing traffic.

  - Traffic timeline: describes when the competing traffic starts and ends in the test case.

* Additional attributes: describes attributes essential for implementing a test case which are not included in the above structure. These attributes must be well defined, so that the other implementers of that particular test case are able to implement it easily.

Any attribute can have a set of values (enclosed within "[]"). Each member value of such a set must be treated as different value for the same attribute. It is desired to run separate tests for each such attribute value.

The test cases described in this document follow the above structure.

4. Recommended Evaluation Settings

This section describes recommended test case settings and could be overwritten by the respective test cases.

4.1. Evaluation metrics

To evaluate the performance of the candidate algorithms the implementers must log enough information to visualize the following metrics at a fine enough time granularity:

1. Flow level:

   A. End-to-end delay for the congestion controlled media flow(s). For example - end-to-end delay observed on IP packet level, video frame level.

   B. Variation in sending bit rate and throughput. Mainly observing the frequency and magnitude of oscillations.

   C. Packet losses observed at the receiving endpoint.

   D. Feedback message overhead.

   E. Convergence time - time to reach steady state for the congestion controlled media flow(s). Each occurrence of convergence during the test period need to be presented.

2. Transport level:

   A. Bandwidth utilization.

   B. Queue length (milliseconds at specified path capacity).

4.2. Path characteristics

Each path between a sender and receiver as described in Figure 1 have the following characteristics unless otherwise specified in the test case.

- Path direction: forward and backward.

- Reference bottleneck capacity: 1Mbps.

- One-Way propagation delay: 50ms. Implementers are encouraged to run the experiment with additional propagation delays mentioned in [I-D.ietf-rmcat-eval-criteria]

- o Maximum end-to-end jitter: 30ms. Jitter models are described in [I-D.ietf-rmcat-eval-criteria]

- o Bottleneck queue type: "tail drop". Implementers are encouraged to run the experiment with other AQM schemes, such as FQ-CoDel and PIE.

- o Bottleneck queue size: 300ms.

- o Path loss ratio: 0%.

Examples of additional network parameters are discussed in [I-D.ietf-rmcat-eval-criteria].

For test cases involving time-varying bottleneck capacity, all capacity values are specified as a ratio with respect to a reference capacity value, so as to allow flexible scaling of capacity values along with media source rate range. There exist two different mechanisms for inducing path capacity variation: a) by explicitly modifying the value of physical link capacity; or b) by introducing background non-adaptive UDP traffic with time-varying traffic rate. Implementers are encouraged to run the experiments with both mechanisms for test cases specified in Section 5.1, Section 5.2, and Section 5.3.

4.3. Media source

Unless otherwise specified, each test case will include one or more media sources as described below.

- o Media type: Video

    - * Media codec: VBR

    - * Media source behavior:

        - + Adaptability:

            - - Bit rate range: 150 Kbps - 1.5 Mbps. In real-life applications the bit rate range can vary a lot depending on the provided service, for example, the maximum bit rate can be up to 4Mbps. However, for running tests to evaluate the congestion control algorithms it is more important to have a look at how they are reacting to certain amount of bandwidth change. Also it is possible that the media traffic generator used in a particular simulator or testbed is not capable of generating higher bit rate. Hence we have selected a suitable bit rate

range typical of consumer-grade video conferencing applications in designing the test case. If a different bit rate range is used in the test cases, then the end-to-end path capacity values will also need to be scaled accordingly.

- Frame resolution: 144p - 720p (or 1080p). This resolution range is selected based on the bit rate range. If a different bit rate range is used in the test cases then the frame resolution range also need to be selected suitably.

- Frame rate: 10fps - 30fps. This frame rate range is selected based on the bit rate range. If a different bit rate range is used in the test cases then the frame rate range also need to be adjusted suitably.

+ Variation from target bit rate: +/-5%. Unless otherwise specified in the test case(s), bit rate variation should be calculated over one (1) second period of time.

+ Responsiveness to new bit rate request: 100ms

* Media content: The media content should represent a typical video conversational scenario with head and shoulder movement. We recommend to use Foreman video sequence[xiph-seq].

* Media startup behavior: 150Kbps. It should be noted that applications can use smart ways to select an optimal startup bit rate value for a certain network condition. In such cases the candidate proposals may show the effectiveness of such smart approach as an additional information for the evaluation process.

o Media type: Audio

* Media codec: CBR

* Media bit rate: 20Kbps

5. Basic Test Cases

5.1. Variable Available Capacity with a Single Flow

In this test case the minimum bottleneck-link capacity between the two endpoints varies over time. This test is designed to measure the responsiveness of the candidate algorithm. This test tries to address the requirements in [I-D.ietf-rmcat-cc-requirements], which

requires the algorithm to adapt the flow(s) and provide lower end-to-end latency when there exists:

- o an intermediate bottleneck
- o change in available capacity (e.g., due to interface change, routing change, abrupt arrival/departure of background non-adaptive traffic).
- o maximum media bit rate is greater than link capacity. In this case, when the application tries to ramp up to its maximum bit rate, since the link capacity is limited to a value lower, the congestion control scheme is expected to stabilize the sending bit rate close to the available bottleneck capacity.

It should be noted that the exact variation in available capacity due to any of the above depends on the underlying technologies. Hence, we describe a set of known factors, which may be extended to devise a more specific test case targeting certain behaviors in a certain network environment.

Expected behavior: the candidate algorithm is expected to detect the path capacity constraint, converge to the bottleneck link's capacity and adapt the flow to avoid unwanted media rate oscillation when the sending bit rate is approaching the bottleneck link's capacity. Such oscillations might occur when the media flow(s) attempts to reach its maximum bit rate but overshoots the usage of the available bottleneck capacity then to rectify, it reduces the bit rate and starts to ramp up again.

Evaluation metrics : as described in Section 4.1.

Testbed topology: One media source S1 is connected to the corresponding R1. The media traffic is transported over the forward path and corresponding feedback/control traffic is transported over the backward path.

```
                             Forward -->
+----+        +------+                               +------+       +----+
|S1  |=======|  A   |------------------------------->|  B   |=======|R1  |
+----+        |      |<-------------------------------|      |       +----+
              +------+                               +------+
                          <-- Backward
```

Figure 2: Testbed Topology for Limited Link Capacity

Testbed attributes:

- o Test duration: 100s
- o Path characteristics: as described in Section 4.2
- o Application-related:
  - * Media Traffic:
    - + Media type: Video
      - - Media direction: forward.
      - - Number of media sources: one (1)
      - - Media timeline:
        - o Start time: 0s.
        - o End time: 99s.
    - + Media type: Audio
      - - Media direction: forward.
      - - Number of media sources: one (1)

- Media timeline:

  o Start time: 0s.

  o End time: 99s.

* Competing traffic:

  + Number of sources : zero (0)

o Test Specific Information:

  * One-way propagation delay: [50 ms, 100 ms]. on the forward path direction

  * This test uses bottleneck path capacity variation as listed in Table 1

  * When using background non-adaptive UDP traffic to induce time-varying bottleneck , the physical path capacity remains at 4Mbps and the UDP traffic source rate changes over time as (4 -

(Y x r)), where r is the Reference bottleneck capacity in Mbps and Y is the path capacity ratio specified in Table 1

Variation pattern index	Path direction	Start time	Path capacity ratio
One	Forward	0s	1.0
Two	Forward	40s	2.5
Three	Forward	60s	0.6
Four	Forward	80s	1.0

Table 1: Path capacity variation pattern for forward direction

5.2. Variable Available Capacity with Multiple Flows

This test case is similar to Section 5.1. However in addition this test will also consider persistent network load due to competing traffic.

Expected behavior: the candidate algorithm is expected to detect the variation in available capacity and adapt the media stream(s) accordingly. The flows stabilize around their maximum bit rate as the maximum link capacity is large enough to accommodate the flows. When the available capacity drops, the flows adapt by decreasing their sending bit rate, and when congestion disappears, the flows are again expected to ramp up.

Evaluation metrics : as described in Section 4.1.

Testbed Topology: Two (2) media sources S1 and S2 are connected to their corresponding destinations R1 and R2. The media traffic is transported over the forward path and corresponding feedback/control traffic is transported over the backward path.


```
   +----+                                                        +----+
   |S1 |===== \                                         / =======|R1 |
   +----+     \\             Forward -->               //        +----+
               \\                                     //
               +------+                              +------+
               |  A   |----------------------------->|  B   |
               |      |<-----------------------------|      |
               +------+                              +------+
                 //                                    \\
                //          <-- Backward                \\
   +----+      //                                        \\       +----+
```

```
|S2 |====== /                                               \ ======|R2 |
+----+                                                              +----+
```

Figure 3: Testbed Topology for Variable Available Capacity

Testbed attributes:

Testbed attributes are similar as described in Section 5.1 except the test specific capacity variation setup.

Test Specific Information: This test uses path capacity variation as listed in Table 2 with a corresponding end time of 125 seconds. The reference bottleneck capacity is 2Mbps. When using background non-adaptive UDP traffic to induce time-varying bottleneck for congestion controlled media flows, the physical path capacity is 4Mbps and the UDP traffic source rate changes over time as (4 - (Y x r)), where r is the Reference bottleneck capacity in Mbps and Y is the path capacity ratio specified in Table 2.

Variation pattern index	Path direction	Start time	Path capacity ratio
One	Forward	0s	2.0
Two	Forward	25s	1.0
Three	Forward	50s	1.75
Four	Forward	75s	0.5
Five	Forward	100s	1.0

Table 2: Path capacity variation pattern for forward direction

5.3. Congested Feedback Link with Bi-directional Media Flows

Real-time interactive media uses RTP hence it is assumed that RTCP, RTP header extension or such would be used by the congestion control algorithm in the backchannel. Due to the asymmetric nature of the

link between communicating peers it is possible for a participating peer to not receive such feedback information due to an impaired or congested backchannel (even when the forward channel might not be impaired). This test case is designed to observe the candidate congestion control behavior in such an event.

Expected behavior: It is expected that the candidate algorithms are able to cope with the lack of feedback information and adapt to minimize the performance degradation of media flows in the forward channel.

It should be noted that for this test case: logs are compared with the reference case, i.e, when the backward channel has no impairments.

Evaluation metrics : as described in Section 4.1.

Testbed topology: One (1) media source S1 is connected to corresponding R1, but both endpoints are additionally receiving and sending data, respectively. The media traffic (S1->R1) is transported over the forward path and corresponding feedback/control traffic is transported over the backward path. Likewise media traffic (S2->R2) is transported over the backward path and corresponding feedback/control traffic is transported over the forward path.

```
      +---+                                                         +---+
      |S1 |===== \                Forward -->              / =======|R1 |
      +---+      \\                                       //        +---+
                  \\                                     //
               +-----+                               +-----+
               |  A  |------------------------------>|  B  |
               |     |<------------------------------|     |
               +-----+                               +-----+
                  //                                     \\
                 //            <-- Backward               \\
     +---+      //                                         \\       +---+
     |R2 |===== /                                           \ ======|S2 |
     +---+                                                          +---+
```

Figure 4: Testbed Topology for Congested Feedback Link

Testbed attributes:

o Test duration: 100s

o Path characteristics:

      * Reference bottleneck capacity: 1Mbps.

   o Application-related:

      * Media Source:

         + Media type: Video

            - Media direction: forward and backward

            - Number of media sources: two (2)

            - Media timeline:

               o Start time: 0s.

               o End time: 99s.

         + Media type: Audio

            - Media direction: forward and backward

            - Number of media sources: two (2)

            - Media timeline:

               o Start time: 0s.

               o End time: 99s.

      * Competing traffic:

         + Number of sources : zero (0)

   o Test Specific Information: this test uses path capacity variations to create congested feedback link. Table 3 lists the variation patterns applied to the forward path and Table 4 lists the variation patterns applied to the backward path. When using background non-adaptive UDP traffic to induce time-varying bottleneck for congestion controlled media flows, the physical path capacity is 4Mbps for both directions and the UDP traffic source rate changes over time as (4-x)Mbps in each direction, where x is the bottleneck capacity specified in Table 4.

Variation pattern index	Path direction	Start time	Path capacity ratio
One	Forward	0s	2.0
Two	Forward	20s	1.0
Three	Forward	40s	0.5
Four	Forward	60s	2.0

Table 3: Path capacity variation pattern for forward direction

Variation pattern index	Path direction	Start time	Path capacity ratio
One	Backward	0s	2.0
Two	Backward	35s	0.8
Three	Backward	70s	2.0

Table 4: Path capacity variation pattern for backward direction

5.4. Competing Media Flows with same Congestion Control Algorithm

In this test case, more than one media flow share the bottleneck link and each of them uses the same congestion control algorithm. This is a typical scenario where a real-time interactive application sends more than one media flow to the same destination and these flows are multiplexed over the same port. In such a scenario it is likely that the flows will be routed via the same path and need to share the available bandwidth amongst themselves. For the sake of simplicity it is assumed that there are no other competing traffic sources in the bottleneck link and that there is sufficient capacity to accommodate all the flows individually. While this appears to be a variant of the test case defined in Section 5.2, it focuses on the capacity sharing aspect of the candidate algorithm. The previous test case, on the other hand, measures adaptability, stability, and responsiveness of the candidate algorithm.

Expected behavior: It is expected that the competing flows will converge to an optimum bit rate to accommodate all the flows with minimum possible latency and loss. Specifically, the test introduces three media flows at different time instances, when the second flow appears there should still be room to accommodate another flow on the bottleneck link. Lastly, when the third flow appears the bottleneck link should be saturated.

Evaluation metrics : as described in Section 4.1.

Testbed topology: Three media sources S1, S2, S3 are connected to R1, R2, R3 respectively. The media traffic is transported over the forward path and corresponding feedback/control traffic is transported over the backward path.

```
   +----+                                                        +----+
   |S1 |===== \                Forward -->              / =======|R1 |
   +----+     \\                                       //        +----+
               \\                                     //
   +----+      +------+                              +------+      +----+
   |S2 |=======|  A   |----------------------------->|  B   |=======|R2 |
   +----+      |      |<-----------------------------|      |      +----+
               +------+                              +------+
               //          <-- Backward               \\
   +----+     //                                       \\       +----+
   |S3 |===== /                                         \ ======|R3 |
   +----+                                                       +----+
```

Figure 5: Testbed Topology for Multiple congestion controlled media Flows

Testbed attributes:

o Test duration: 120s

o Path characteristics:

   * Reference bottleneck capacity: 3.5Mbps

   * Path capacity ratio: 1.0

o Application-related:

  * Media Source:

    + Media type: Video

      - Media direction: forward.

      - Number of media sources: three (3)

      - Media timeline: new media flows are added sequentially, at short time intervals. See test specific setup below.

    + Media type: Audio

      - Media direction: forward.

      - Number of media sources: three (3)

      - Media timeline: new media flows are added sequentially, at short time intervals. See test specific setup below.

  * Competing traffic:

    + Number of sources : zero (0)

o Test Specific Information: Table 5 defines the media timeline for both media type.

Flow ID	Media type	Start time	End time
1	Video	0s	119s
2	Video	20s	119s
3	Video	40s	119s
4	Audio	0s	119s
5	Audio	20s	119s
6	Audio	40s	119s

Table 5: Media Timeline for Video and Audio media sources

5.5. Round Trip Time Fairness

In this test case, multiple media flows share the bottleneck link, but the one-way propagation delay for each flow is different. For the sake of simplicity it is assumed that there are no other competing traffic sources in the bottleneck link and that there is sufficient capacity to accommodate all the flows. While this appears to be a variant of test case 5.2, it focuses on the capacity sharing aspect of the candidate algorithm under different RTTs.

Expected behavior: It is expected that the competing flows will converge to bit rates to accommodate all the flows with minimum possible latency and loss. The effectiveness of the algorithm depends on how fast and fairly the competing flows converge to their steady states irrespective of the RTT observed.

Evaluation metrics : as described in Section 4.1.

Testbed Topology: Five (5) media sources S1,S2,..,S5 are connected to their corresponding media sinks R1,R2,..,R5. The media traffic is transported over the forward path and corresponding feedback/control

traffic is transported over the backward path. The topology is the same as in Section 5.4.

Testbed attributes:

- o Test duration: 300s

- o Path characteristics:

  - * Reference bottleneck capacity: 4Mbps

  - * Path capacity ratio: 1.0

  - * One-Way propagation delay for each flow: 10ms for S1-R1, 25ms for S2-R2, 50ms for S3-R3, 100ms for S4-R4, and 150ms S5-R5.

- o Application-related:

  - * Media Source:

+ Media type: Video

  - Media direction: forward

  - Number of media sources: five (5)

  - Media timeline: new media flows are added sequentially, at short time intervals. See test specific setup below.

+ Media type: Audio

  - Media direction: forward.

  - Number of media sources: five (5)

  - Media timeline: new media flows are added sequentially, at short time intervals. See test specific setup below.

* Competing traffic:

  + Number of sources : zero (0)

o Test Specific Information: Table 6 defines the media timeline for both media type.

Flow IF	Media type	Start time	End time
1	Video	0s	299s
2	Video	10s	299s
3	Video	20s	299s
4	Video	30s	299s
5	Video	40s	299s
6	Audio	0	299s
7	Audio	10s	299s
8	Audio	20s	299s
9	Audio	30s	299s

```
| 10        | Audio       | 40s         | 299s      |
+----------+------------+------------+----------+
```

Table 6: Media Timeline for Video and Audio media sources

5.6. Media Flow Competing with a Long TCP Flow

In this test case, one or more media flows share the bottleneck link with at least one long lived TCP flow. Long lived TCP flows download data throughout the session and are expected to have infinite amount of data to send and receive. This is a scenario where a multimedia application co-exists with a large file download. The test case measures the adaptivity of the candidate algorithm to competing traffic. It addresses the requirement 3 in [I-D.ietf-rmcat-cc-requirements].

Expected behavior: depending on the convergence observed in test case 5.1 and 5.2, the candidate algorithm may be able to avoid congestion collapse. In the worst case, the media stream will fall to the minimum media bit rate.

Evaluation metrics : following metrics in addition to as described in Section 4.1.

1. Flow level:

    A. TCP throughput.

    B. Loss for the TCP flow

Testbed topology: One (1) media source S1 is connected to the corresponding media sink, R1. In addition, there is a long-live TCP flow sharing the same bottleneck link. The media traffic is transported over the forward path and corresponding feedback/control traffic is transported over the backward path. The TCP traffic goes

over the forward path from, S_tcp with acknowledgment packets go over the backward path from, R_tcp.

```
+---+                                                   +---+
|S1|===== \                Forward -->              / =====|R1|
```

```
      +---+      \\                                    //      +---+
                  \\                                  //
                  +-----+                            +-----+
                  |  A  |--------------------------->|  B  |
                  |     |<---------------------------|     |
                  +-----+                            +-----+
                  //        <-- Backward              \\
     +-----+     //                                    \\    +-----+
     |S_tcp|=== /                                       \ ===|R_tcp|
     +-----+                                                 +-----+
```

Figure 6: Testbed Topology for TCP vs congestion controlled media Flows

Testbed attributes:

- o Test duration: 120s
- o Path characteristics:
  - * Reference bottleneck capacity: 2Mbps
  - * Path capacity ratio: 1.0
  - * Bottleneck queue size: [300ms, 1000ms]
- o Application-related:
  - * Media Source:
    - + Media type: Video
      - - Media direction: forward
      - - Number of media sources: one (1)
      - - Media timeline:
        - o Start time: 5s.
        - o End time: 119s.
    - + Media type: Audio

- Media direction: forward

- Number of media sources: one (1)

- Media timeline:

  o Start time: 5s.

  o End time: 119s.

* Additionally, implementers are encouraged to run the experiment with multiple media sources.

* Competing traffic:

  + Number and Types of sources : one (1) and long-lived TCP

  + Traffic direction : forward

  + Congestion control: default TCP congestion control[RFC5681]. Implementers are also encouraged to run the experiment with alternative TCP congestion control algorithm.

  + Traffic timeline:

    - Start time: 0s.

    - End time: 119s.

o Test Specific Information: none

5.7. Media Flow Competing with Short TCP Flows

In this test case, one or more congestion controlled media flow shares the bottleneck link with multiple short-lived TCP flows. Short-lived TCP flows resemble the on/off pattern observed in the web traffic, wherein clients (for example, browsers) connect to a server and download a resource (typically a web page, few images, text files, etc.) using several TCP connections. This scenario shows the performance of a multimedia application when several browser windows are active. The test case measures the adaptivity of the candidate algorithm to competing web traffic, it addresses the requirements 1.E in [I-D.ietf-rmcat-cc-requirements].

Depending on the number of short TCP flows, the cross-traffic either appears as a short burst flow or resembles a long TCP flow. The intention of this test is to observe the impact of short-term burst on the behavior of the candidate algorithm.

Expected behavior: The candidate algorithm is expected to avoid flow starvation during the presence of short and bursty competing TCP flows, streaming at least at the minimum media bit rate. After competing TCP flows terminate, the media streams are expected to be robust enough to eventually recover to previous steady state behavior, and at the very least, avoid persistent starvation.

Evaluation metrics : following metrics in addition to as described in Section 4.1.

1. Flow level:

    A. Variation in the sending rate of the TCP flow.

    B. TCP throughput.

Testbed topology: The topology described here is same as the one described in Figure 6.

Testbed attributes:

- o Test duration: 300s
- o Path characteristics:
  - * Reference bottleneck capacity: 2.0Mbps
  - * Path capacity ratio: 1.0
- o Application-related:
  - * Media source:
    - + Media type: Video
      - - Media direction: forward
      - - Number of media sources: two (2)
      - - Media timeline:
        - o Start time: 5s.

o End time: 299s.

+ Media type: Audio

- Media direction: forward

- Number of media sources: two (2)

- Media timeline:

o Start time: 5s.

o End time: 299s.

* Competing traffic:

+ Number and Types of sources : ten (10), short-lived TCP flows.

+ Traffic direction : forward

+ Congestion algorithm: default TCP Congestion control [RFC5681]. Implementers are also encouraged to run the experiment with alternative TCP congestion control algorithm.

+ Traffic timeline: each short TCP flow is modeled as a sequence of file downloads interleaved with idle periods. Not all short TCP flows start at the same time, 2 of them start in the ON state while rest of the 8 flows start in an OFF state. For description of short TCP flow model see test specific information below.

o Test Specific Information:

* Short-TCP traffic model: The short TCP model to be used in this test is described in [I-D.ietf-rmcat-eval-criteria].

5.8. Media Pause and Resume

In this test case, more than one real-time interactive media flows share the link bandwidth and all flows reach to a steady state by utilizing the link capacity in an optimum way. At this stage one of the media flows is paused for a moment. This event will result in more available bandwidth for the rest of the flows as they are on a shared link. When the paused media flow resumes it would no longer have the same bandwidth share on the link. It has to make its way through the other existing flows in the link to achieve a fair share of the link capacity. This test case is important specially for real-time interactive media which consists of more than one media flows and can pause/resume media flows at any point of time during the session. This test case directly addresses the requirement number 5 in [I-D.ietf-rmcat-cc-requirements]. One can think it as a variation of test case defined in Section 5.4. However, it is

different as the candidate algorithms can use different strategies to increase its efficiency, for example in terms of fairness, convergence time, reduce oscillation etc, by capitalizing the fact that they have previous information of the link.

Expected behavior: During the period where the third stream is paused, the two remaining flows are expected to increase their rates and reach the maximum media bit rate. When the third stream resumes, all three flows are expected to converge to the same original fair share of rates prior to the media pause/resume event.

Evaluation metrics : following metrics in addition to as described in Section 4.1.

1. Flow level:

   A. Variation in sending bit rate and throughput. Mainly observing the frequency and magnitude of oscillations.

Testbed Topology: Same as test case defined in Section 5.4

Testbed attributes: The general description of the testbed parameters are same as Section 5.4 with changes in the test specific setup as below-

- Other test specific setup:

  * Media flow timeline:

+ Flow ID: one (1)

+ Start time: 0s

+ Flow duration: 119s

+ Pause time: not required

+ Resume time: not required

* Media flow timeline:

  + Flow ID: two (2)

  + Start time: 0s

  + Flow duration: 119s

  + Pause time: at 40s

  + Resume time: at 60s

* Media flow timeline:

  + Flow ID: three (3)

  + Start time: 0s

  + Flow duration:119s

  + Pause time: not required

  + Resume time: not required

6. Other potential test cases

It has been noticed that there are other interesting test cases besides the basic test cases listed above. In many aspects, these additional test cases can help further evaluation of the candidate algorithm. They are listed as below.

6.1. Media Flows with Priority

In this test case media flows will have different priority levels. This will be an extension of Section 5.4 where the same test will be run with different priority levels imposed on each of the media flows. For example, the first flow (S1) is assigned a priority of 2 whereas the remaining two flows (S2 and S3) are assigned a priority of 1. The candidate algorithm must reflect the relative priorities assigned to each media flow. In this case, the first flow (S1) must arrive at a steady-state rate approximately twice of that of the other two flows (S2 and S3).

The candidate algorithm can use a coupled congestion control mechanism [I-D.ietf-rmcat-coupled-cc] or use a weighted priority scheduler for the bandwidth distribution according to the respective media flow priority or use.

6.2. Explicit Congestion Notification Usage

This test case requires to run all the basic test cases with the availability of Explicit Congestion Notification (ECN) [RFC6679] feature enabled. The goal of this test is to exhibit that the candidate algorithms do not fail when ECN signals are available. With ECN signals enabled the algorithms are expected to perform better than their delay-based variants.

6.3. Multiple Bottlenecks

In this test case one congestion controlled media flow, S1->R1, traverses a path with multiple bottlenecks. As illustrated in Figure 7, the first flow (S1->R1) competes with the second congestion controlled media flow (S2->R2) over the link between A and B which is close to the sender side; again, that flow (S1->R1) competes with the third congestion controlled media flow (S3->R3) over the link between C and D which is close to the receiver side. The goal of this test is to ensure that the candidate algorithms work properly in the presence of multiple bottleneck links on the end to end path.

Expected behavior: The candidate algorithm is expected to achieve full utilization at both bottleneck links without starving any of the three congestion controlled media flows and ensuring fair share of the available bandwidth at each bottlenecks.

```
                               Forward ---->

               +---+          +---+        +---+      +---+
               |S2 |          |R2 |        |S3 |      |R3 |
               +---+          +---+        +---+      +---+
                 |              |            |          |
                 |              |            |          |
  +---+       +-----+       +-----+      +-----+     +-----+       +---+
  |S1 |=======|  A  |------>|  B  |----->|  C  |---->|  D  |=======|R1 |
  +---+       |     |<------|     |<-----|     |<----|     |       +---+
              +-----+       +-----+      +-----+     +-----+

                       1st                       2nd
                Bottleneck (A->B)          Bottleneck (C->D)

                             <------ Backward
```

Figure 7: Testbed Topology for Multiple Bottlenecks

Testbed topology: Three media sources S1, S2, and S3 are connected to respective destinations R1, R2, and R3. For all three flows the media traffic is transported over the forward path and corresponding feedback/control traffic is transported over the backward path.

Testbed attributes:

- o Test duration: 300s

- o Path characteristics:

  * Reference bottleneck capacity: 2Mbps.

  * Path capacity ratio between A and B: 1.0

* Path capacity ratio between B and C: 4.0.

* Path capacity ratio between C and D: 0.75.

* One-Way propagation delay:

  1. Between S1 and R1: 100ms

  2. Between S2 and R2: 40ms

  3. Between S3 and R3: 40ms

o Application-related:

* Media Source:

  + Media type: Video

    - Media direction: Forward

    - Number of media sources: Three (3)

    - Media timeline:

      o Start time: 0s.

      o End time: 299s.

  + Media type: Audio

    - Media direction: Forward

    - Number of media sources: Three (3)

    - Media timeline:

      o Start time: 0s.

      o End time: 299s.

* Competing traffic:

    + Number of sources : Zero (0)

7. Wireless Access Links

Additional wireless network (both cellular network and WiFi network) specific test cases are defined in [I-D.ietf-rmcat-wireless-tests].

8. Security Considerations

The security considerations in [I-D.ietf-rmcat-eval-criteria] and the relevant congestion control algorithms apply. The principles for congestion control are described in [RFC2914], and in particular any new method must implement safeguards to avoid congestion collapse of the Internet.

The evaluation of the test cases are intended to be run in a controlled lab environment. Hence, the applications, simulators and network nodes ought to be well-behaved and should not impact the desired results. Moreover, proper measures must be taken to avoid leaking non-responsive traffic from unproven congestion avoidance techniques onto the open Internet.

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.

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

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