                   Test Cases for Evaluating RMCAT Proposals
                       draft-sarker-rmcat-eval-test-00

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. The RMCAT working group is
   currently working on candidate algorithms for such interactive real-
   time multimedia applications. This document describes the test cases
   to be used in the evaluation of the performance of those candidate
   algorithms.

Status of This Memo

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

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1. Introduction

   This memo describes a set of test cases for evaluating candidate
   RMCAT congestion control algorithm proposals, it is based on the
   guidelines enumerated in [I-D.ietf-rmcat-eval-criteria] and
   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
   modeling of different link characteristics. Each test case
   incorporates the metrics, evaluation guidelines and parameters
   described in [I-D.ietf-rmcat-eval-criteria]. It is the intention of
   this work to capture the consensus of the RMCAT working group
   participants regarding the test cases upon which the performance of
   the candidate RMCAT proposals should be evaluated.

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], Support for Reduced-Size RTCP [RFC5506], and
   RTP Circuit Breaker algorithm [I-D.ietf-avtcore-rtp-circuit-breakers]
   apply.

3. Basic Structure of Test cases

   All test cases in this document follow a basic structure allowing
   implementers to describe a new test scenarios without repeatedly
   explaining common attributes. The structure includes a general
   description section that describes the test case and motivations,
   additionally it defines a set of attributes that characterize the
   testbed, i.e., the network path between communicating peers and the
   diverse traffic sources.

   o Define the test case:

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

      * Additionally, describe the desired rate adaptation behaviour.

      * Define a checklist to evaluate the desired behaviour: this
         indicates the minimum set of metrics (e.g., link utiilization,
         media sending rate) that a proposed algorithm needs to measure
         to validate the expected rate adaptation behaviour. 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 either generate media traffic and use an
      RMCAT candidate congestion control algorithm or use a different
      congestion control algorithm than that one used in the media
      source. R1..Rn are the corresponding receivers. A test case can
      have one or more such traffic source (S) and corresponding
      receiver (R). The path from the source to destination is denoted
      as upstream and the path from a destination to a source is denoted
      as downstream. It is possible for a testbed to have different
      path characteristics in either directions. The following basic
      structure of test case has been described from the perspective of
      endpoints attached on the left-hand side of Figure 1.

o

```
+---+                                                        +---+
|S1 |====== \               UPSTREAM -->          / ======|R1 |
+---+        \\                                   //         +---+
              \\                                 //
+---+        +-----+                         +-----+        +---+
|S2 |======|  A  |------------------------------>|  B  |======|R2 |
+---+      |     |<------------------------------|     |      +---+
           +-----+                               +-----+
(...)        //                                    \\          (...)
            //            <-- DOWNSTREAM             \\
+---+      //                                          \\     +---+
|Sn |====== /                                    \ ======|Rn |
+---+                                                        +---+
```
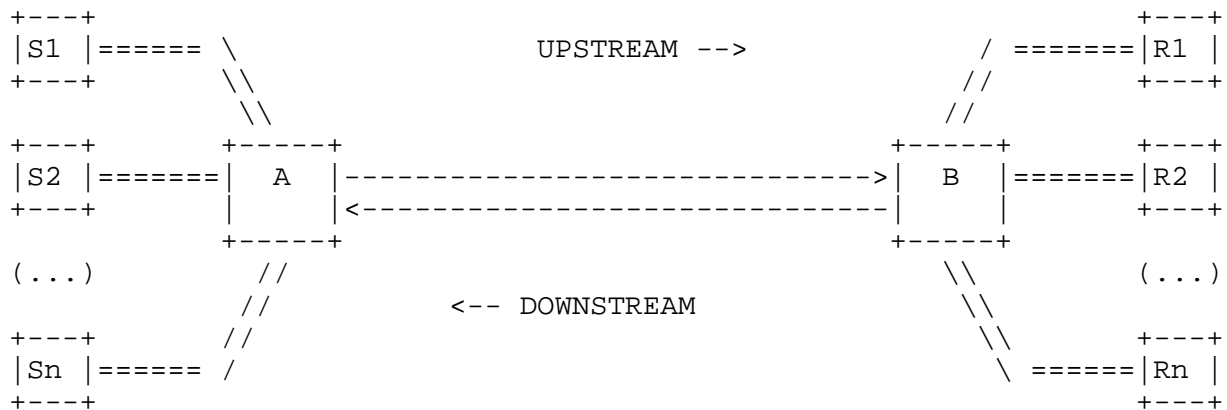
                    Figure 1: Example of A Testbed Topology

o In a laboratory testbed environment there exists large amount of
   traffic on the network path between the endpoints, which may be
   contributed by other devices connected to such a testbed. It is
   recommended to not route non-test traffic through the testbed.

o Define testbed attributes:

   * Duration: defines the duration of the test.

   * Path characteristics: defines the end-to-end transport level
      path characteristics of the testbed in a particular test case.
      Two sets of attributes describes the path characteristics, one
      for the upstream path and the other for the downstream 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.

      + Path direction: upstream or downstream.

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

      + 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,
         it does not include any queuing delay.

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

    + Bottleneck queue type: for example, Droptail, FQ-CoDel, or
       PIE.

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

    + Path loss ratio: characterize the non-congested losses
       observed on the end-to-end path. MUST describe the loss
       pattern or loss model.

  * Application-related: defines the media-related behaviour for
     implementing the test case

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

       - Media type: Video/Voice/Application/Text

       - Media flow direction: upstream, downstream 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 behaviour: describes the media encoder
          behavior. It MUST define the main parameters than affect
          the adaptation behaviour. This may include but not
          limited to:

          o Adaptability: describes the adaptation options. For
             example, in case of video it defines the range of
             bitrate adaptation, frame rate adaption, video
             resolution adaptation. In case of voice, it defines
             the range of bitrate adaptation, sampling rate, frame
             size.

          o Output variation : for VBR encoder, it defines the
             encoder output variation from the average target rate.
             For example on average the encoder output may vary
             between 5% to 15% above or below the average target
             bit rate.

          o Encoder's responsiveness to a new bit rate request:
             value typically between 10ms to 1000ms.

- Media content: describes the chosen media sequences; For example, test sequences are available at: [xiph-seq] [HEVC-seq].

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

- Startup behaviour: 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: Upstream, downstream 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 type.

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

- Traffic timeline: describes when the competing traffic is added and removed from 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 other implementers are able to implement it.

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. Basic Test Cases

4.1. Variable Available Capacity

   In this test case the 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 requirement 1(a) 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 due to interface change and/or
      routing change.

   o persistent network load due to competing traffic

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

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

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

   1. Flow level:

       a. End-to-end delay

       b. Losses observed at the receiving endpoint

       c. Feedback message overhead

   2. Transport level:

       a. Bandwidth utilization

b. Queue length (ms):

+   average over the length of the session

+   5 and 95 percentile

+   median, maximum, minimum

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

```
+---+                                              +---+
|S1 |===== \              UPSTREAM -->      / ======|R1 |
+---+      \\                              //       +---+
            \\                            //
            +-----+                  +-----+
            |  A  |------------------------------>|  B  |
            |     |<------------------------------|     |
            +-----+                  +-----+
             //                                \\
             //            <-- DOWNSTREAM        \\
+---+       //                                    \\       +---+
|S2 |===== /                                       \ ======|R2 |
+---+                                                       +---+
```
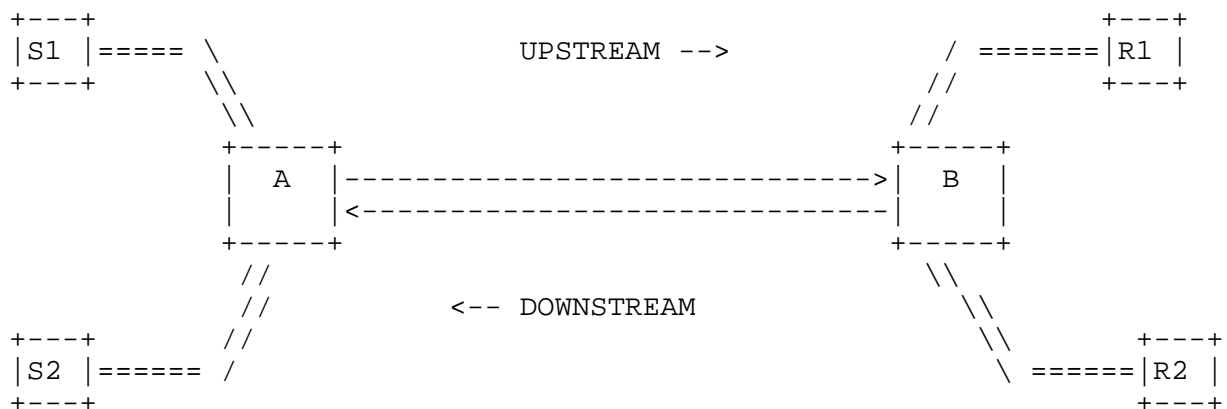
        Figure 2: Testbed Topology for Variable Available Capacity

Testbed attributes:

o Test duration: 120s

o Path characteristics:

    * Path direction: Upstream and downstream.

    * Bottleneck-link capacity: 4Mbps.

    * One-Way propagation delay: [10ms, 100ms].

    * Maximum end-to-end jitter: 30ms.

    * Bottleneck queue type: Drop tail.

    * Bottleneck queue size: 300ms.

    * path loss ratio: 0%.

o Application-related:

   * Media Source:

      + Media type: Video

      + Media direction: Upstream.

      + Number of media sources: Two (2).

      + Media codec: VBR

      + Media source behaviour:

         - Adaptability:

            o Bit rate range: 150 Kbps - 1.5 Mbps

            o Frame Resolution: 144p - 720p (or 1080p)

            o Frame rate: 10fps - 30fps

         - Variation from target bitrate: +/-5%

         - Responsiveness to new bit rate request: 100ms

      + Media content: Foreman media sequence.

      + Media timeline:

         - Start time: 0s.

         - End time: 119s.

   * Media startup behaviour: [200Kbps, 1500Kbps].

   * Competing traffic:

      + Number of sources : Zero (0)

o Test specific setup:

   * Number of bandwidth variations: Three (4)

   * Variation pattern:

      + Sequence number: 1

        + Path direction: Upstream

        + Bottleneck Capacity: 2Mbps.

        + Start time: 30s

     * Variation pattern:

        + Sequence number: 2

        + Path direction: Upstream

        + Bottleneck Capacity: 3.5Mbps

        + Start time: 50s

     * Variation pattern:

        + Sequence number: 3

        + Path direction: Upstream

        + Bottleneck Capacity: 1Mbps

        + Start time: 70s

     * Variation pattern:

        + Sequence number: 4

        + Path direction: Upstream

        + Bottleneck Capacity: 2Mbps

        + Start time: 100s

4.2. Maximum Media Bit Rate is Greater than Link Capacity

   In this case, the application will attempt 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. This situation can
   occur when the endpoints are connected via thin long networks even
   though the advertised capacity of the access network may be higher.
   The test case addresses the requirement 1 and 10 of the
   [I-D.ietf-rmcat-cc-requirements].

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

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

```
                              UPSTREAM -->
+---+         +-----+                                 +-----+         +---+
|S1 |=======|  A  |-------------------------------->|  B  |=======|R1 |
+---+         |     |<--------------------------------|     |         +---+
              +-----+                                 +-----+
                              <-- DOWNSTREAM
```
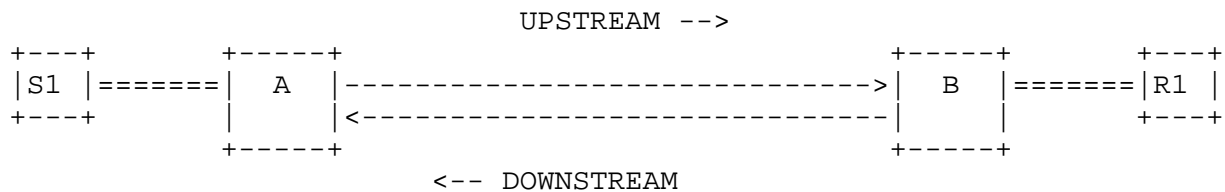
              Figure 3: Testbed Topology for Limited Link Capacity

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

1. Flow level:

    a. End-to-end delay.

    b. RTP packet losses observed at the receiving endpoint.

    c. Variation in sending bit rate and goodput. Mainly observing
        the frequency and magnitude of oscillations.

    d. Convergence time.

    e. Feedback message overhead.

2. Transport level:

    a. Bandwidth utilization.

    b. Queue length (ms):

        + average over the length of the session

        + 5 and 95 percentile

         + median

   Testbed attributes:

   o Test duration: 60s

   o Path characteristics:

      * Path direction: Upstream and downstream.

      * Bottleneck-link capacity: 1Mbps

      * One-Way propagation delay: [10ms,100 ms]

      * Maximum end-to-end jitter: 30ms.

      * Bottleneck queue type: Droptail. Additional tests with other
         AQM schemes are recommended: FQ-CoDel, PIE

      * Bottleneck size: 300ms

      * Path loss ratio: 0%

   o Application-related:

      * Media Source:

         + Media type: Video

         + Media direction: Upstream.

         + Number of media sources: One (1)

         + Media codec: VBR

         + Media source behaviour:

            - Adaptability:

               o Bit rate range: 150 Kbps - 1.5 Mbps

               o Frame Resolution: 144p - 720p (or 1080p)

               o Frame rate: 10fps - 30fps

            - Variation from target bitrate: +/-5%

            - Responsiveness to new bit rate request: 100ms

             + Media content: Foreman video sequence

             + Media timeline:

                - Start time: 0s.

                - End time: 59s.

          * Media startup behaviour: [200Kbps, 1500Kbps].

          * Competing traffic:

             + Number of sources : Zero (0)

     o Test specific setup: None

4.3. Competing Flows with same RMCAT Algorithm

     In this test case, more than one RMCAT media flow shares the
     bottleneck link and each of them uses the same congestion control
     algorithm. This is a typical scenario wherein a real-time
     interactive application sends more than one media flows 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 4.1 , however it tests the capacity sharing
     distribution of the candidate algorithm. Whereas, the previous test
     case measures the stability and responsiveness of the candidate
     algorithm. This test case particularly addresses the requirements
     2,3 and 10 in [I-D.ietf-rmcat-cc-requirements].

     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.

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

     1. Flow level:

a. End-to-end delay.

b. RTP packet losses observed at the receiving endpoint.

c. Variation in sending bit rate and goodput. Mainly observing
    the frequency and magnitude of oscillations.

d. Convergence time.

e. Feedback message overhead.

2. Transport level:

a. Bandwidth utilization.

b. Queue length (ms):

    + average over the length of the session

    + 5 and 95 percentile

    + median

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

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

        Figure 4: Testbed Topology for Multiple RMCAT Flows

Testbed attributes:

o Test duration: 60s

   o Path characteristics:

      * Path direction: Upstream, Downstream

      * Bottleneck-link capacity: 3.5Mbps

      * One-Way propagation delay: [10ms, 50ms]

      * Maximum end to end jitter: 30ms

      * Bottleneck queue type: Droptail

      * Bottleneck queue size: 300ms

      * Link loss ratio: 0.0%

   o Application-related:

      * Media Source:

         + Media type: Video

         + Media direction: Upstream

         + Number of media sources: Three (3)

         + Media codec: VBR

         + Media source behaviour:

            - Adaptability:

               o Bit rate range: 150 Kbps - 1.5 Mbps

               o Frame Resolution: 144p - 720p (or 1080p)

               o Frame rate: 10fps - 30fps

            - Variation from target bitrate: +/-5%

            - Responsiveness to new bit rate request: 100ms

         + Media content: Foreman video sequence

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

         + Media startup behaviour: 200Kbps.

       * Competing traffic:

          + Number of sources : Zero (0)

   o Test specific setup:

      * Media flow timeline:

         + Flow no: One (1)

         + Start time: 0s

         + End time: 59s

      * Media flow appearance:

         + Flow no: Two (2)

         + Start time: 20s

         + End time: 59s

      * Media flow appearance:

         + Flow no: Three (3)

         + Start time: 40s

         + End time: 59s

4.4. RMCAT Flow competing with a long TCP Flow

   In this test case, one or more RMCAT media flow shares the bottleneck
   link with at least one long lived TCP flows. 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
   wherein 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 8 in
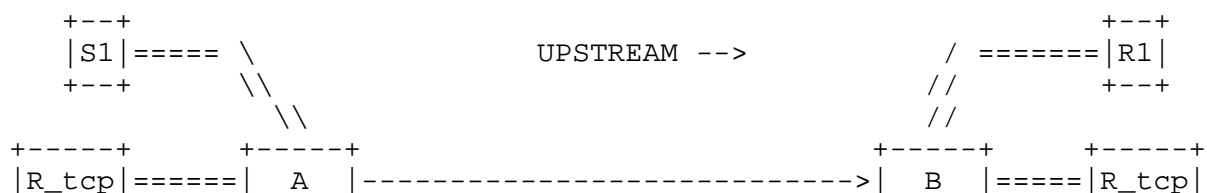   [I-D.ietf-rmcat-cc-requirements].

   Expected behavior: depending on the convergence observed in test case
   4.1 and 4.2, 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.

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

1. Flow level:

    a. End-to-end delay for the RMCAT flow.

    b. RTP packet losses observed at the receiving endpoint.

    c. Variation in sending bit rate and goodput. Mainly observing
        the frequency and magnitude of oscillations.

    d. Variation in the sending rate of the TCP flow

    e. TCP throughput.

    f. Convergence time.

    g. Feedback message overhead.

2. Transport level:

    a. Bandwidth utilization.

    b. Queue length (ms):

        + average over the length of the session

        + 5 and 95 percentile

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->S2) is
transported over the upstream path and corresponding feedback/control
traffic is transported over the downstream path. Likewise media
traffic (S2->S1) is transported over the downstram path and
corresponding feedback/control traffic is transported over the
upstream path. The TCP uploads traffic over upstream path and
downloads over downstream path. Hence, TCP traffic exits in both
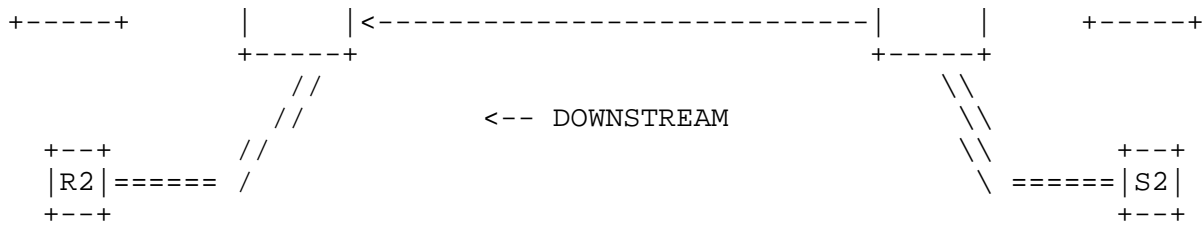path directions.

```
    +--+                                              +--+
    |S1|===== \              UPSTREAM -->      / ======|R1|
    +--+       \\                             //        +--+
                \\                           //
+-----+      +-----+                     +-----+      +-----+
|R_tcp|======|  A  |------------------------->|  B  |=====|R_tcp|
```

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

               Figure 5: Testbed Topology for TCP vs RMCAT Flows

    Testbed attributes:

    o Test duration: 120s

    o Path characteristics:

       * Path direction: Upstream, Downstream

       * Bottleneck-link capacity: 2Mbps

       * One-Way propagation delay: [10ms, 150ms]

       * Maximum end to end jitter: 30ms

       * Bottleneck queue type: Droptail, but would benefit from running
          the same test with different AQM schemes: FQ-Codel, or PIE.

       * Bottleneck queue size: [20ms, 250ms, 1000ms]

       * Path loss ratio: 0.0%

    o Application-related:

       * Media Source:

          + Media type: Video

          + Media direction: Upstream and Downstream

          + Number of media sources: One (1)

          + Media codec: VBR

          + Media source behaviour:

             - Adaptability:

                o Bit rate range: 150 Kbps - 1.5 Mbps

    o Frame Resolution: 144p - 720p (or 1080p)

    o Frame rate: 10fps - 30fps

   - Variation from target bitrate: +/-5%

   - Responsiveness to new bit rate request: 100ms

  + Media content: Foreman video sequence

  + Media timeline:

   - Start time: 5s.

   - End time: 119s.

  + Media startup behaviour: [200Kbps, 1500Kbps].

 * Competing traffic:

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

  + Traffic direction : Downstream

  + Congestion control: Default TCP congestion control.

  + Traffic timeline:

   - Start time: 0s.

   - End time: 119s.

 o Test specific setup: None

4.5. RMCAT Flow competing with short TCP Flows

In this test case, one or more RMCAT media flow shares the bottleneck
link with at multiple short-lived TCP flows. Short-lived TCP flows
resemble the on/off pattern observed in the web traffic, wherein
clients (browsers) connect to a server and download a resource
(typically a webpage, few images, text files, etc.) using several
TCP connections (up to 4). This scenario shows the performance of
the 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 2 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 behaviour of the candidate algorithm.

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

1. Flow level:

    a. End-to-end delay for the RMCAT flow.

    b. RTP packet losses observed at the receiving endpoint.

    c. Variation in sending bit rate and goodput. Mainly observing
        the frequency and magnitude of oscillations.

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

    e. TCP throughput.

    f. Convergence time.

    g. Feedback message overhead.

2. Transport level:

    a. Bandwidth utilization.

    b. Queue length (ms):

        + average over the length of the session

        + 5 and 95 percentile

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->S2) is
transported over the upstream path and corresponding feedback/control
traffic is transported over the downstream path. Likewise media
traffic (S2->S1) is transported over the downstram path and
corresponding feedback/control traffic is transported over the
upstream path. The TCP uploads traffic over upstream path and
downloads over downstream path. Hence, TCP traffic exits in both
path directions. The topology described here is similar to the one
described in Figure 5.

Testbed attributes:

o Test duration: 300s

o Path characteristics:

   * Path direction: Upstream, Downstream

   * Bottleneck-link capacity: 2.0Mbps

   * One-Way propagation delay: [10ms, 150ms]

   * Maximum end to end jitter: 30ms

   * Bottleneck queue type: Droptail, but would benefit from running
      the same test with different AQM schemes: FQ-Codel, or PIE.

   * Bottleneck queue size: 300ms

   * Path loss ratio: 0.0%

o Application-related:

   * Media Source:

      + Media type: Video

      + Media direction: Upstream and Downstream

      + Number of media sources: One (1)

      + Media codec: VBR

      + Media source behaviour:

         - Adaptability:

            o Bit rate range: 150 Kbps - 1.5 Mbps

            o Frame Resolution: 144p - 720p (or 1080p)

            o Frame rate: 10fps - 30fps

         - Variation from target bitrate: +/-5%

         - Responsiveness to new bit rate request: 100ms

      + Media content: Foreman video sequence

   + Media timeline:

     - Start time: 0s.

     - End time: 299s.

   + Media startup behaviour: [200Kbps, 1500Kbps].

 * Competing traffic:

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

   + Traffic direction : Downstream

   + Congestion algorithm: Default TCP Congestion control.

   + Traffic timeline: Each short TCP flow is modeled as a
     sequence of file downloads interleaved with idle periods.
     See test specific setup. Not all short TCPs start at the
     same time, 2 start in the ON state while 8 start in an OFF
     stats. The model for the idle times for the OFF state is
     discussed in the Short-TCP model.

 o Test specific setup:

   * Short-TCP traffic model:

     + File sizes: uniform distribution between 100KB to 1MB

     + Idle period: the duration of the OFF state is derived from
       an exponential distribution with the mean value of 10
       seconds.

4.6. Congested Feedback Link

   RMCAT WG has been chartered to define algorithms for RTP hence it is
   assumed that RTCP, RTP header extension or such would be used as
   signalling means for the adaptation algorithm in the backchannel.
   Due to asymmetry nature of the link between communicating peers it is
   possible to observer lack such backchannel information due to
   impaired backchannel link (even when forward channel might be
   unimpaired). This test case is designed to observer candidate
   congestion control behaviour in such an event. This test case
   addresses requirement number 5 and in particular, requirement number
   7.

It is expected that the candidate algorithms should cope with the lack of backchannel information and adapt to minimize the performance degradation of media flows in the forward channel.
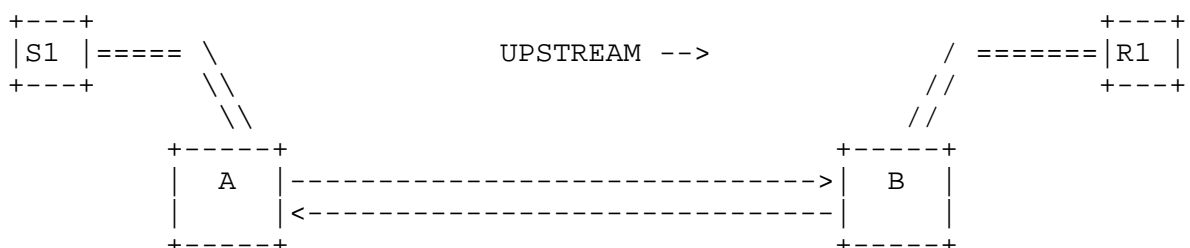
It SHOULD be noted that for this test case log is needed for the reference case where the downstream channel has no impairments

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

1. Flow level:

    a. End-to-end delay

    b. Losses observed at the receiving endpoint

    c. Feedback message overhead

2. Transport level:

    a. Bandwidth utilization

    b. Queue length (ms):

        + average over the length of the session

        + 5 and 95 percentile

.

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->S2) is transported over the upstream path and corresponding feedback/control traffic is transported over the downstream path. Likewise media traffic (S2->S1) is transported over the downstram path and corresponding feedback/control traffic is transported over the upstream path.
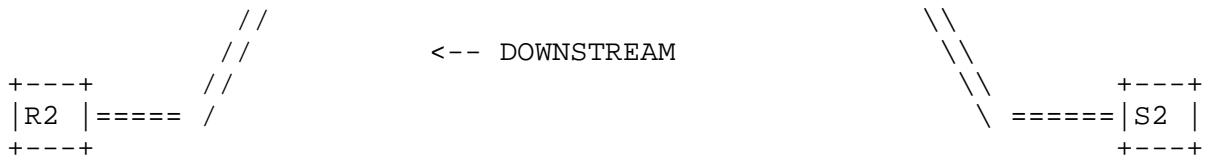
```
  +---+                                                    +---+
  |S1 |===== \              UPSTREAM -->          / ======|R1 |
  +---+       \\                                  //        +---+
               \\                                //
          +-----+                           +-----+
          |  A  |-------------------------->|  B  |
          |     |<--------------------------|     |
          +-----+                           +-----+
```

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

                Figure 6: Testbed Topology for Congested Feedback Link

   Testbed attributes:

   o Test duration: 60s

   o Path characteristics:

      * Path direction: Upstream and downstream.

      * Bottleneck-link capacity: 2Mbps.

      * One-Way propagation delay: [10ms, 100ms].

      * Maximum end-to-end jitter: 30ms.

      * Bottleneck queue type: Drop tail.

      * Bottleneck queue size: 300ms.

      * path loss ratio: 0%.

   o Application-related:

      * Media Source:

         + Media type: Video

         + Media direction: Upstream

         + Number of media sources: One (1).

         + Media codec: VBR

         + Media source behaviour:

            - Adaptability:

               o Bit rate range: 150 Kbps - 1.5 Mbps

               o Frame Resolution: 144p - 720p (or 1080p)

        o Frame rate: 10fps - 30fps

      - Variation from target bitrate: +/-5%

      - Responsiveness to new bit rate request: 100ms

   + Media content: Foreman media sequence

   + Media timeline:

      - Start time: 0s.

      - End time: 59s.

   * Media Source:

   + Media type: Video

   + Media direction: Downstream

   + Number of media sources: One (1).

   + Media codec: VBR

   + Media source behaviour:

      - Adaptability:

        o Bit rate range: 150 Kbps - 1.5 Mbps

        o Frame Resolution: 144p - 720p (or 1080p)

        o Frame rate: 10fps - 30fps

      - Variation from target bitrate: +/-5%

      - Responsiveness to new bit rate request: 100ms

   + Media content: Foreman media sequence

   + Media timeline:

      - Start time: 0s.

      - End time: 59s.

   * Media startup behaviour: 200Kbps.

      * Competing traffic:

         + Number of sources : Zero (0)

   o Test specific setup:

      * Number of bandwidth variation: Two (2)

      * Variation pattern:

         + Sequence number: 1

         + Path direction: Upstream

         + Bottleneck Capacity: 1Mbps

         + Start time: 20s

      * Variation pattern:

         + Sequence number: 2

         + Path direction: Upstream

         + Bottleneck Capacity: 600Kbps

         + Start time: 30s

      * Variation pattern:

         + Sequence number: 3

         + Path direction: Upstream

         + Bottleneck Capacity: 2Mbps

         + Start time: 50s

      * Variation pattern:

         + Sequence number: 1

         + Path direction: Downstream

         + Bottleneck Capacity: 800kbps

         + Start time: 30s

    * Variation pattern:

       + Sequence number: 2

       + Path direction: Downstream

       + Bottleneck Capacity: 2Mbps

       + Start time: 50s

4.7. Round Trip Time Fairness

   In this test case, more than one RMCAT media flow shares the
   bottleneck link, but the end-to-end path latency for each RMCAT 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 the test case 4.2, it tests the
   capacity sharing distribution of the candidate algorithm under
   different RTTs. This test case particularly addresses the
   requirements 2 [I-D.ietf-rmcat-cc-requirements].

   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 five media flows at the
   same time instance.

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

   1. Flow level:

       a. End-to-end delay.

       b. RTP packet losses observed at the receiving endpoint.

       c. Variation in sending bit rate and goodput. Mainly observing
           the frequency and magnitude of oscillations.

       d. Convergence time.

   2. Transport level:

       a. Bandwidth utilization.

       b. Queue length (ms):

        + average over the length of the session

        + 5 an 95 percentile

   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 upstream path and corresponding feedback/control
   traffic is transported over the downstream path. The topology is the
   same as test case 4.3 defined in Section 4.3. The end-to-end path
   delay for S1-R1 is 10ms, S2-R2 is 25ms, S3-R3 is 50ms, S4-R4 is
   100ms, S5-R5 is 150ms.

   Testbed attributes:

   o Test duration: 60s

   o Path characteristics:

      * Path direction: Upstream, Downstream

      * Number of bottlenecks: One (1)

      * Bottleneck link capacity: 3.0Mbps

      * One-Way propagation delay for each path is: 10ms, 25ms, 50ms,
         100ms, 150ms.

      * Maximum end to end jitter: 30ms

      * Bottleneck queue type: Droptail

      * Bottleneck queue size: 300ms

      * Link loss ratio: 0.0%

   o Application-related:

      * Media Source:

         + Media direction: Upstream

         + Number of media sources: Five (5)

         + Encoder configuration:

            - Bit rate generation: VBR

            - Bit rate range: 150 Kbps - 1.5 Mbps

- Frame Resolution: 144p-720p (or 1080p)

- Frame rate: 10fps-30fps

- Variation from target bit rate: +/-5%

- Responsiveness to new bit rate request: 60ms

+ Media content: Foreman video sequence

+ Media timeline:

- Start time: 0s.

- End time: 59s.

+ Media startup behaviour: [200 Kbps, 1500 Kbps].

* Competing traffic:

+ Number of sources : Zero (0)

o Test specific setup: None

4.8. 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 these stage one of
the media flow is paused for a moment. This event will result in
more available bandwidth for the rest of the flows and as they are on
a shared link. When the paused media flow will resume it would no
longer have the same bandwidth share on the link. It has to make it
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 flow at any point of time during the
session. This test case directly addresses the requirement number
1.B in [I-D.ietf-rmcat-cc-requirements]. One can think it as a
variation of test case 4.3. However, it is different as the
candidate algorithms can use different strategies to increase it s
efficiency, for example the fairness, convergence time, reduce
oscillation etc, by capitalizing the fact that they have previous
information of the link.

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

1. Flow level:

    a. End-to-end delay.

    b. RTP packet losses observed at the receiving endpoint.

    c. Variation in sending bit rate and goodput. Mainly observing
        the frequency and magnitude of oscillations.

    d. Convergence time.

    e. Feedback message overhead.

2. Transport level:

    a. Bandwidth utilization.

    b. Queue length (ms):

        + average over the length of the session

        + 5 and 95 percentile

Testbed Topology: Same as test case 4.3 defined in Section 4.3

Testbed attributes: The general description of the testbed parameters
are same as test case 4.3 with changes in the test specific setup as
below-

o Other test specific setup:

   * Media flow timeline:

      + Flow no: One (1)

      + Start time: 0s

      + Flow duration: 59s

      + Pause time: not required

      + Resume time: not required

   * Media flow appearance:

      + Flow no: Two (2)

      + Start time: 0s

             + Flow duration: 59s

             + Pause time: 20s

             + Resume time: 30s

          * Media flow appearance:

             + Flow no: Three (3)

             + Start time: 0s

             + Flow duration:59s

             + Pause time: not required

             + Resume time: not required

4.9. Explicit Congestion Notification Usage

   TBD

5. Wireless Access Links

5.1. Cellular Network Specific Test Cases

   Additional cellular network specific test cases are define in
   [I-D.draft-sarker-rmcat-cellular-eval-test-cases]

5.2. Wi-Fi Network Specific Test Cases

   TBD

   [Editor's Note: We should encourage people to come up with possible
   WiFi Network specific test cases]

6. Security Considerations

   Security issues have not been discussed in this memo.

7. IANA Considerations

   There are no IANA impacts in this memo.

8. 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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