                   Test Cases for Evaluating RMCAT Proposals
                      draft-folks-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 WebRTC. This document
   describes the test cases needed to evaluate the performance of those
   candidate algorithms.

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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 any implementer to provide new test cases to
   fit their test scenario and link characteristics. Each test cases
   incorporates the metrics, evaluation guidelines and parameters
   described in [I-D.ietf-rmcat-eval-criteria].

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, it enables
   implementers to describe new test scenarios without explaining common
   attributes repeatedly. The structure includes a general description
   section that describe 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 that a proposed algorithm
         needs to measure to validate the expected rate adaptation
         behaviour.

   o Define testbed attributes:

      * Duration: defines the duration of the test case.

      * Path characteristics: defines the transport level
         characteristics of a test case. The characteristics describes
         two sets of characteristics, one each for the upstream and the
         downstream direction. If only one is specified, it is used for
         both directions.

         + Path direction: upstream or downstream.

         + Number of bottlenecks and the link capacity for each
            bottleneck link.

         + One-way propagation delay: describes the end-to-end latency
            along the path.

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

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

+ Bottleneck queue size: (in miliseconds).

+ Link loss ratio: chatacterize the non-congested losses
   observed on a specific link, for e.g., at the access link or
   a bottleneck link. Also 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.

   - Media flow direction: upstream, downstream or both.

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

   - Media source configuration: describes the media encoder
      behavior. This may include but not limited to

      o Bit rate generation: Constant Bit Rate (CBR) if the
         chosen media stream is produced at the exact rate as
         the target bit rate. Variable Bit Rate (VBR) if there
         exits some variation at the media output bit rate than
         the target bit rate for RMCAT media streams.

      o If the media stream is sending VBR, the test case MUST
         define the maximum and minimum encoding rate, frame
         resolution, and frame rate.

      o Variation from target bit rate: the encoder produces a
         bit rate close to the tartget rate. For example it
         may vary between 5% to 15% above or below the 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.

- Number and Types of sources: defines the total number of competing sources of each type. 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.

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

- Traffic timeline: describes when the competing 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. This occurs when defining different attributes for the same type of traffic source.

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 end-to-end path 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 capcity 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 targering 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 candidate algorithm tracks the available capacity
   as closely as possible, i.e., if there is sufficient capacity the
   flow(s) reach their respective maximum bit rate. When the available
   capacity drops, the flow(s) adapts by decreasing its bit rate, and
   when congestion disappears, the flow(s) are again expected to rampup.

   To evaluate the performance of the candidate algorithms it is
   expected to log enough information to visualize the following
   metrics:

   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 attributes:

   o Test duration: 60s

   o Path characteristics:

      * Path direction: Upstream and downstream.

      * Number of bottlenecks : One (1).

      * Bottleneck link capacity : 2Mbps.

      * One-Way propagation delay: 100ms.

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

      * Bottleneck queue type: Drop tail.

      * Bottleneck queue size: 300ms.

      * Link loss ratio: 0%.

   o Application-related:

      * Media Source:

         + Media direction: Upstream.

         + Number of media sources: Two (2).

         + 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 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,1500Kbps].

* Competing traffic

+ Number of sources : Zero (0)

o Test specific setup

* Number of bandwidth variation: Three (3)

* Bottleneck link vairation pattern:

+ Sequence number: 1

+ Path direction: Upstream

+ Bottleneck Capacity: 1Mbps.

+ Start time: 30s

+ End behaviour: Bandwith is restored to the 80% of bottleneck link speed

* Bottleneck link vairation pattern:

+ Sequence number: 2

+ Path direction: Upstream

+ Bottleneck Capacity: 0.5Mbps

+ Start time: 45s

* Bottleneck link vairation pattern:

+ Sequence number: 3

+ Path direction: Upstream

+ Bottleneck Capacity: 2Mbps

           + Start time: 50s

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 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 eventhough
   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 canidate algorithm is expected to detect the
   limitation in available capacity and avoid future bit rate
   oscillations as it approaches the bottleneck link capacity. The
   oscillations occur when the media flow(s) attempts to reach its
   maximum bit rate, overshoots the available bottleneck capacity
   causing overuse, to rectify it reduces the bit rate and starts to
   probe again.

   To evaluate the performance of the candidate algorithms it is
   expected to log enough information to visualize the following
   metrics:

   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 attributes:

o Test duration: 60s

o Path characteristics:

   * Path direction: Upstream and downstream.

   * Number of bottlenecks : One (1)

   * Bottleneck link speed : 1Mbps

   * One-Way propagation delay: 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

   * Link loss ratio: 0%

o Application-related:

   * Media Source:

      + Media direction: Upstream.

      + Number of media sources: One (1)

      + 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 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 use 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 desitination 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 taffic
   sources in the bottleneck link and that there is sufficient capacity
   to accomodate all the flows. While this appears to be a variant of
   the previous test case, however it tests the capacity sharing
   distribution of the candidate algorithm. Whereas, the previous test
   case measures the stability of the candidate algorithm. This test
   case particularly addresses the requirements 2,3 and 10 in
   [I-D.ietf-rmcat-cc-requirements].

   It is expected that the competing flows will converge to an optimum
   bit rate to accomodate all the flows with minumum 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 accomodate 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:

   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 attributes:

   o Test duration: 60s

   o Path characteristics

      * Path direction: Upstream, Downstream

      * Number of bottlenecks: One (1)

      * Bottleneck link capacity: 3.5Mbps

      * One-Way propagation delay: 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 direction: Upstream

         + Number of media sources: Three (3)

         + 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: 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 ime: 10s

          + End time: 59s

       * Media flow appearance:

          + Flow no: Three (3)

          + Start ime: 25s

          + 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 requirements 8 in
   [I-D.ietf-rmcat-cc-requirements].

   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:

   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 attributes:

o Test duration: 120s

o Path characteristics

   * Path direction: Upstream, Downstream

   * Number of bottlenecks: one

   * Bottleneck link capacity: 2Mbps

   * One-Way propagation delay: [50ms, 200ms]

   * 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]

   * Link loss ratio: 0.0%

o Application-related:

   * Media Source:

      + Media direction: Upstream and Downstream

      + Number of media sources: One (1)

      + 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 video

      + Media timeline:

         - Start time: 5s.

                  - End time: 59s.

              + 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: 59s.

   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 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:

   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 attributes:

   o Test duration: 300s

   o Path characteristics:

     * Path direction: Upstream, Downstream

     * Number of bottlenecks: One (1)

     * Bottleneck link capacity: 2.0Mbps

     * One-Way propagation delay: [50ms, 200ms]

     * 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

     * Link loss ratio: 0.0%

   o Application-related:

     * Media Source:

+ Media direction: Upstream and Downstream

+ Number of media sources: One (1)

+ 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 video

+ Media timeline:

- Start time: 0s.

- End time: 59s.

+ 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.

o Test specific setup:

* Short-TCP traffic model

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

           + Idle period: 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 extention or such would be used as
   signalling means for the adaptation algorithm in the backchannel.
   Due to assymetry 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 will the
   lack of backchannel information and adapt to minimize the performace
   of media flows in the forward channel.

   To evaluate the performance of the candidate algorithms it is
   expected to log enough information to visualize the following
   metrics:

   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

   It should be noted that for this test case log is needed for the
   reference case where the downstream channel have no impairments.

   Example Testbed attributes:

   o Test duration: 60s

o Path characteristics: Same as test case 4.1

o Application-related:

   * Media Source:

      + Media direction: Upstream, Downstream

      + Number of media sources: two (2).

      + 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 bitrate: +/-5%

         - Responsiveness to new bit rate request: 60ms

      + 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)

   * Link vairation pattern:

      + Sequence number: 1

      + Path direction: Upstream

+ Amount of change: 50% of bottleneck link speed

+ Duration: 10s

+ Start time: 10s

+ End behaviour: bandwith is restored to the 80% of bottleneck
  link speed

* Link vairation pattern:

+ Sequence number: 2

+ Path direction: Downstream

+ Amount of change: 50% of bottleneck link speed

+ Duration: 5s

+ Starttime: 15s

+ End behaviour: bandwith is restored to the 100% of
  bottleneck link speed

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 taffic sources in the bottleneck link and that
there is sufficient capacity to accomodate 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 accomodate all the flows with minumum 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:

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 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: 25ms, 50ms, 100ms,
       150ms, 200ms.

    * 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 video

       + 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
   fairshare 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

osillation 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 metics:

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 attributes: The gerenal description of the test bed
parameters are same as test case 4.3 with only chages 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: One (1)

+ 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

5.2. Wi-Fi Network Specific Test Cases

TBD

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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