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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 such interactive real-time multimedia applications. This document describes the test cases needed to evaluate the performance of those candidate algorithms.

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Table of Contents

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 observed along the path.

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

+ Bottleneck queue size: (in milliseconds).

+ Link loss ratio: characterize 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 target 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:

- an intermediate bottleneck

- change in available capacity due to interface change and/or routing change.

- 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 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 ramp up.

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 variation pattern:

  + Sequence number: 1

  + Path direction: Upstream

  + Bottleneck Capacity: 1Mbps.

  + Start time: 30s

* Bottleneck link variation pattern:

  + Sequence number: 2

  + Path direction: Upstream

  + Bottleneck Capacity: 0.5Mbps

  + Start time: 45s

* Bottleneck link variation 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 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 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 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. 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 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:

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 time: 10s
  + End time: 59s

* Media flow appearance:

  + Flow no: Three (3)
  + Start time: 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:

- Test duration: 120s

- 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%

- 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 sequence

    + 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:

- Test duration: 300s

- 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%

- 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 sequence

         + 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 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 will the lack of backchannel information and adapt to minimize the performance 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 variation pattern:

      + Sequence number: 1

      + Path direction: Upstream

    + Amount of change: 50% of bottleneck link speed

    + Duration: 10s

    + Start time: 10s

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

  * Link variation pattern:

    + Sequence number: 2

    + Path direction: Downstream

    + Amount of change: 50% of bottleneck link speed

    + Duration: 5s

    + Start time: 15s

    + End behaviour: Bandwidth 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 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:

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:

- Test duration: 60s

- 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%

- 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:

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 general description of the test bed parameters are same as test case 4.3 with changes in the test specific setup as below-

- 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.

9. References

9.1. Normative References

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