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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 implement congestion control. This document describes the test cases required to evaluate the performance of congestion control for interactive point-to-point real-time media.

Status of This Memo

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1. Introduction

This memo describes a set of test cases for evaluating candidate RMCAT proposals, it based on the guidelines enumerated in [I-D.ietf-rmcat-eval-criteria] and requirements discussed in [I-D.ietf-rmcat-cc-requirements]. This document discusses an initial set of scenarios that cover common use-cases. Additionally, the document provides a basic structure to escribe test cases, which can be customized by an implementer to fit their test scenario.

2. Terminology

The terminology defined in RTP [RFC3550], RTP Profile for Audio and Video Conferences with Minimal Control [RFC3551], RTCP Extended Report (XR) [RFC3611], Extended RTP Profile for RTCP-based Feedback (RTP/AVPF) [RFC4585] and Support for Reduced-Size RTCP [RFC5506] apply.

3. Basic Structure of Test cases

Defining a common structure enables implementers to describe new scenarios.

- 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 mimimum set of metrics that a proposed algorithm needs to achieve to pass the test case.

- o Define testbed parameters:

    - * 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.

        - + 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 bottleneck link. Also describe the loss pattern or model to implement the losses.

* Application-related: defines different applications behaviour included in the test case

  + Media Source: defines the characteristics of the media sources present in the test case. When using more than one media source, the following set of attibutes are defined for each such media source.

    - 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 Maximum Bit rate.

      o Minimum Bit rate.

      o Bit rate generation: Constant bit rate if the chosen media stream is not congestion controlled (for example, IPTV). Variable bit rate for RMCAT media streams.

      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, the desired frame rate, resolution, etc. Test sequences are available at: [xiph-seq] [HEVC-seq].

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

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

4. Basic Test Cases

4.1. Variable Available Capacity

In this testcase the end-to-end path capacity between the two endpoint varies over time. This test is designed to measure the responsiveness of the candidate algorithm. The 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 intermediated 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 impairments depends on the under-lying technology. Hence, we describe a set of known factors, which may be extended to devise a more specific test case.

Expected behavior: the candidate algorithm should detect the variation and adapt the media stream accordingly. The candidate algorithm track 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 RMCAT flow(s) adapts by decreasing its bit rate, and when congestion disappears, the flow(s) are expected to reach their maximum bit rate.

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. Losses observed at the receiving endpoint

2. Transport level

    a. Bandwidth utilization

    b. Queue length (ms)

        + average over the length of the session

        + 5 ad 95 percentile

Example Testbed parameters:

- Test duration: 60s

- Transport

    * Path direction: Upstream and downstream.

    * Number of bottlenecks : One.

    * 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:

            - Maximum Bit rate: 2Mpbs.

            - Minimum Bit rate: 200Kbps.

            - Bit rate generation: Variable bit rate.

            - Variation from target bitrate: 5%

            - Responsiveness to new bit rate request: 60ms

         + Media content: Foreman media sequence

         + Media timeline: all sources appear at start up, end 1s
            before shutdown.

      * Media startup behaviour: starts at minimum bit rate (200kbps).

      * Competing traffic

         + Number of sources : Zero (0)

   o Test specific setup

      * Number of bandwidth variation: Two (2)

      * Link vairation pattern:

         + Sequence number: 1

         + Amount of change: 50% of bottleneck link speed

         + Duration: 5s

        + start time: 10s

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

    * Link vairation pattern:

        + Sequence number: 2

        + Amount of change: 50% of bottleneck link speed

        + Duration: 10s

        + starttime: 30s

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

4.2. Maximum Media Bit Rate is Greater than Link Capacity

In this case, the application will attempt to reach 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 can occur when the endpoints are connected via thin Internet pipes 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].

In this test case the canidate algorithm is expected to detect the limitation in available capacity and avoid 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 causing overuse, then 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 metics:

1. Flow level

    a. End-to-end delay.

    b. RTP packet losses observed at the receiving endpoint.

    c. Oscillation in sending bit rate.

    d. Convergence period.

2. Transport level

   a. Bandwidth utilization.

   b. Queue length (ms)

      + average over the length of the session

      + 5 an 95 percentile

Testbed parameters:

- o Test duration: 120s

- o Transport

  * Number of bottlenecks : One

  * Bottleneck link speed : 1Mbps

  * One-Way propagation delay: 100 ms

  * Bottleneck queue type: one of Drop tail

  * Bottleneck size: 300ms

  * Link loss ratio: 0%

- o Application-related:

  * Media Source:

    + Number of media sources: one (1)

    + Encoder configuration: Variable bit rate.

    + Variation from target bit rate: 5%

    + Media content: Foreman video sequence

    + Media timeline: at startup, shutdown 1s before the end.

  * Media startup behaviour: 1) start at minimum bit rate, 2) start at maximum bit rate.

  * Competing traffic

    + Number of sources : Zero (0)

o Test specific setup : none

4.3. Competing with similar RMCAT flows

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 testcase measures the stability of the candidate algorithm. This testcase 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 metics:

1. Flow level

    a. End-to-end delay.

    b. RTP packet losses observed at the receiving endpoint.

    c. Oscillation in sending bit rate.

    d. Convergence period.

2. Transport level

    a. Bandwidth utilization.

    b. Queue length (ms)

        + average over the length of the session

        + 5 an 95 percentile

Testbed parameters:

- o Test duration: 40s
- o Transport
  - * Path direction: Upstream, Downstream
  - * Number of bottlenecks: one
  - * Bottleneck link capacity: 2.5Mbps
  - * One-Way propagation delay: 50ms
  - * Maximum end to end jitter: 30ms
  - * Bottleneck queue type: droptail
  - * Bottleneck queue size: 250ms
  - * Link loss ratio: 0.0%
- o Application-related:
  - * Media Source:
    - + Media direction: Upstream
    - + Number of media sources: Three (3)
    - + Encoder configuration:
      - - Maximum Bit rate:2Mbps
      - - Minimum Bit rate:200Kbps
      - - Bit rate generation: VBR
      - - Variation from target bit rate: 5%
      - - Responsiveness to new bit rate request: 60ms
    - + Media content: Foreman video video
    - + Media timeline: sequentially, at short time intervals. See test specific setup below.

        + Media startup behaviour: starts at minimum bit rate (200kbps).

      * Competing traffic

        + Number of sources : Zero (0)

   o Other test specific setup:

      * Media flow timeline:

        + Flow no: one

        + Start ime: 0s

      * Media flow appearance:

        + Flow no: two

        + Start ime: 10s

      * Media flow appearance:

        + Flow no: one

        + Start ime: 25s

4.4. RMCAT Flow competing with a Long TCP Flow

TBD

4.5. RMCAT Flow competing with short TCP Flows

TBD

4.6. Media Pause and Resume

TBD

4.7. Congested Feedback Link

TBD

4.8. Round Trip Time Fairness

TBD

4.9. Startup Behavior

TBD

4.10. Response to Explicit Congestion Notification

TBD

5. Cellular Network Specific Test Cases

TBD

6. Wi-Fi Network Specific Test Cases

TBD

7. Security Considerations

Security issues have not been discussed in this memo.

8. IANA Considerations

There are no IANA impacts in this memo.

9. 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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