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Z. Sarker
I. Johansson
Ericsson AB
X. Zhu
J. Fu
W. Tan
M. Ramalho
Cisco Systems

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Evaluation Test Cases for Interactive Real-Time Media over Wireless Networks
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Abstract

The Real-time Transport Protocol (RTP) is used for interactive multimedia communication applications. A congestion control algorithm is typically required by these applications. To ensure seamless and robust user experience, a well-designed RTP-based congestion control algorithm should work well across all access network types. This document describes test cases for evaluating performances of such congestion control algorithms over LTE and Wi-Fi networks.

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1. Introduction

Wireless networks (both cellular and Wi-Fi [IEEE802.11]) are an integral part of the Internet. Mobile devices connected to the wireless networks account for an increasingly more significant portion of the media traffic over the Internet. Application scenarios range from video conferencing calls in a bus or train to media consumption by someone sitting on a living room couch. It is well known that the characteristics and technical challenges for supporting multimedia services over wireless are very different from those of providing the same service over a wired network. Even though basic test cases for evaluating RTP-based congestion control schemes as defined in [I-D.ietf-rmcat-eval-test] have covered many effects of the impairments common to both wired and wireless networks, there remain characteristics and dynamics unique to a given wireless environment. For example, in LTE networks, the base station maintains individual queues per radio bearer per user hence it leads to a different nature of interaction between traffic flows of different users. This contrasts with wired network, where traffic from all users share the same queue. Furthermore, user mobility patterns in a cellular network differs from those in a Wi-Fi network. Therefore, it is important to evaluate the performance of proposed candidate RTP-based congestion control solutions over cellular mobile networks and over Wi-Fi networks respectively.

RMCAT evaluation criteria [I-D.ietf-rmcat-eval-criteria] document provides the guideline for evaluating candidate algorithms and recognizes the importance of testing over wireless access networks. However, it does not describe any specific test cases for performance evaluation of candidate algorithms. This document describes test cases specifically targeting cellular networks such as LTE networks and Wi-Fi networks.

2. Terminologies

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

3. Cellular Network Specific Test Cases

A cellular environment is more complicated than a wireline ditto since it seeks to provide services in the context of variable available bandwidth, location dependencies and user mobilities at different speeds. In a cellular network the user may reach the cell edge which may lead to a significant amount of retransmissions to

deliver the data from the base station to the destination and vice versa. These network links or radio links will often act as a bottleneck for the rest of the network which will eventually lead to excessive delays or packet drops. An efficient retransmission or link adaptation mechanism can reduce the packet loss probability but there will still be some packet losses and delay variations. Moreover, with increased cell load or handover to a congested cell, congestion in transport network will become even worse. Besides, there are certain characteristics which make the cellular network different from and more challenging than other types of access networks such as Wi-Fi and wired network. In a cellular network -

- The bottleneck is often a shared link with relatively few users.
  - The cost per bit over the shared link varies over time and is different for different users.
  - Left over/ unused resource can be grabbed by other greedy users.
- Queues are always per radio bearer hence each user can have many of such queues.
- Users can experience both Inter and Intra Radio Access Technology (RAT) handovers ("handover" definition in [HO-def-3GPP]).
- Handover between cells, or change of serving cells (see in [HO-LTE-3GPP] and [HO-UMTS-3GPP]) might cause user plane interruptions which can lead to bursts of packet losses, delay and/or jitter. The exact behavior depends on the type of radio bearer. Typically, the default best effort bearers do not generate packet loss, instead packets are queued up and transmitted once the handover is completed.
- The network part decides how much the user can transmit.
- The cellular network has variable link capacity per user
  - Can vary as fast as a period of milliseconds.
  - Depends on lots of facts (such as distance, speed, interference, different flows).
  - Uses complex and smart link adaptation which makes the link behavior ever more dynamic.
  - The scheduling priority depends on the estimated throughput.

- o Both Quality of Service (QoS) and non-QoS radio bearers can be used.

Hence, a real-time communication application operating in such a cellular network need to cope with shared bottleneck link and variable link capacity, event likes handover, non-congestion related loss, abrupt change in bandwidth (both short term and long term) due to handover, network load and bad radio coverage. Even though 3GPP define QoS bearers [QoS-3GPP] to ensure high quality user experience, adaptive real-time applications are desired.

Different mobile operators deploy their own cellular network with their own set of network functionalities and policies. Usually, a mobile operator network includes 2G, EDGE, 3G and 4G radio access technologies. Looking at the specifications of such radio technologies it is evident that only 3G and 4G radio technologies can support the high bandwidth requirements from real-time interactive video applications. The future real-time interactive application will impose even greater demand on cellular network performance which makes 4G (and beyond radio technologies) more suitable access technology for such genre of application.

The key factors to define test cases for cellular networks are

- o Shared and varying link capacity

- o Mobility

- o Handover

However, for cellular network it is very hard to separate such events from one another as these events are heavily related. Hence instead of devising separate test cases for all those important events we have divided the test case in two categories. It should be noted that in the following test cases the goal is to evaluate the performance of candidate algorithms over radio interface of the cellular network. Hence it is assumed that the radio interface is the bottleneck link between the communicating peers and that the core network does not add any extra congestion in the path. Also the combination of multiple access technologies such as one user has LTE connection and another has Wi-Fi connection is kept out of the scope of this document. However, later those additional scenarios can also be added in this list of test cases. While defining the test cases we assumed a typical real-time telephony scenario over cellular networks where one real-time session consists of one voice stream and one video stream. We recommend that an LTE network simulator is used for the test cases defined in this document, for example-NS-3 LTE simulator [LTE-simulator].

3.1. Varying Network Load

The goal of this test is to evaluate the performance of the candidate congestion control algorithm under varying network load. The network load variation is created by adding and removing network users a.k.a. User Equipments (UEs) during the simulation. In this test case, each of the user/UE in the media session is an RMCAT compliant endpoint. The arrival of users follows a Poisson distribution, which is proportional to the length of the call, so that the number of users per cell is kept fairly constant during the evaluation period. At the beginning of the simulation there should be enough time to warm-up the network. This is to avoid running the evaluation in an empty network where network nodes are having empty buffers, low interference at the beginning of the simulation. This network initialization period is therefore excluded from the evaluation period.

This test case also includes user mobility and competing traffic. The competing traffic includes both same kind of flows (with same adaptation algorithms) and different kind of flows (with different service and congestion control). The investigated congestion control algorithms should show maximum possible network utilization and stability in terms of rate variations, lowest possible end to end frame latency, network latency and Packet Loss Rate (PLR) at different cell load level.

3.1.1. Network Connection

Each mobile user is connected to a fixed user. The connection between the mobile user and fixed user consists of a LTE radio access, an Evolved Packet Core (EPC) and an Internet connection. The mobile user is connected to the EPC using LTE radio access technology which is further connected to the Internet. The fixed user is connected to the Internet via wired connection with sufficiently high bandwidth, for instance, 10 Gbps, so that the system is resource-limited on the wireless interface. The Internet and wired connection in this setup does not introduce any network impairments to the test; it only adds 10ms of one-way propagation delay.

The path from the fixed user to mobile user is defines as "Downlink" and the path from mobile user to the fixed user is defined as "Uplink". We assume that only uplink or downlink is congested for the mobile users. Hence, we recommend that the uplink and downlink simulations are run separately.

```
                          uplink
         ++)))        +-------------------------->
         ++-+      ((o))
         |  |       / \     +--------+     +-------+    +---+
         +--+      /   \----+       +------+      +-----+   |
                  /     \   +--------+     +-------+    +---+
          UE         BS         EPC        Internet    fixed
                      <--------------------------+
                               downlink
```

Figure 1: Simulation Topology

3.1.2. Simulation Setup

The values enclosed within " [ ] " for the following simulation attributes follow the notion set in [I-D.ietf-rmcat-eval-test]. The desired simulation setup as follows-

1. Radio environment

    A. Deployment and propagation model : 3GPP case 1[Deployment]

    B. Antenna: Multiple-Input and Multiple-Output (MIMO), [2D, 3D]

    C. Mobility: [3km/h, 30km/h]

    D. Transmission bandwidth: 10Mhz

    E. Number of cells: multi cell deployment (3 Cells per Base Station (BS) * 7 BS) = 21 cells

    F. Cell radius: 166.666 Meters

    G. Scheduler: Proportional fair with no priority

    H. Bearer: Default bearer for all traffic.

    I. Active Queue Management (AQM) settings: AQM [on,off]

2. End to end Round Trip Time (RTT): [40, 150]

3. User arrival model: Poisson arrival model

4. User intensity:

    * Downlink user intensity: {0.7, 1.4, 2.1, 2.8, 3.5, 4.2, 4.9, 5.6, 6.3, 7.0, 7.7, 8.4, 9,1, 9.8, 10.5}

   * Uplink user intensity : {0.7, 1.4, 2.1, 2.8, 3.5, 4.2, 4.9, 5.6, 6.3, 7.0}

5. Simulation duration: 91s

6. Evaluation period : 30s-60s

7. Media traffic

   1. Media type: Video

      a. Media direction: [Uplink, Downlink]

      b. Number of Media source per user: One (1)

      c. Media duration per user: 30s

      d. Media source: same as define in section 4.3 of [I-D.ietf-rmcat-eval-test]

   2. Media Type : Audio

      a. Media direction: Uplink and Downlink

      b. Number of Media source per user: One (1)

      c. Media duration per user: 30s

      d. Media codec: Constant BitRate (CBR)

      e. Media bitrate : 20 Kbps

      f. Adaptation: off

8. Other traffic model:

   * Downlink simulation: Maximum of 4Mbps/cell (web browsing or FTP traffic following default TCP congestion control [RFC5681])

   * Unlink simulation: Maximum of 2Mbps/cell (web browsing or FTP traffic following default TCP congestion control [RFC5681])

3.2. Bad Radio Coverage

The goal of this test is to evaluate the performance of candidate congestion control algorithm when users visit part of the network with bad radio coverage. The scenario is created by using larger

cell radius than previous test case. In this test case each of the user/UE in the media session is an RMCAT compliant endpoint. The arrival of users follows a Poisson distribution, which is proportional to the length of the call, so that the number of users per cell is kept fairly constant during the evaluation period. At the beginning of the simulation there should be enough amount of time to warm-up the network. This is to avoid running the evaluation in an empty network where network nodes are having empty buffers, low interference at the beginning of the simulation. This network initialization period is therefore excluded from the evaluation period.

This test case also includes user mobility and competing traffic. The competing traffic includes same kind of flows (with same adaptation algorithms) . The investigated congestion control algorithms should show maximum possible network utilization and stability in terms of rate variations, lowest possible end to end frame latency, network latency and Packet Loss Rate (PLR) at different cell load level.

3.2.1. Network connection

Same as defined in Section 3.1.1

3.2.2. Simulation Setup

The desired simulation setup is same as Varying Network Load test case defined in Section 3.1 except following changes:

1. Radio environment: Same as defined in Section 3.1.2 except the following:

    A. Deployment and propagation model : 3GPP case 3 [Deployment]

    B. Cell radius: 577.3333 Meters

    C. Mobility: 3km/h

2. User intensity = {0.7, 1.4, 2.1, 2.8, 3.5, 4.2, 4.9, 5.6, 6.3, 7.0}

3. Media traffic model: Same as defined in Section 3.1.2

4. Other traffic model:

    * Downlink simulation: Maximum of 2Mbps/cell (web browsing or FTP traffic following default TCP congestion control [RFC5681])

   * Unlink simulation: Maximum of 1Mbps/cell (web browsing or FTP traffic following default TCP congestion control [RFC5681])

3.3. Desired Evaluation Metrics for cellular test cases

RMCAT evaluation criteria document [I-D.ietf-rmcat-eval-criteria] defines metrics to be used to evaluate candidate algorithms. However, looking at the nature and distinction of cellular networks we recommend at minimum following metrics to be used to evaluate the performance of the candidate algorithms for the test cases defined in this document.

The desired metrics are-

o Average cell throughput (for all cells), shows cell utilizations.

o Application sending and receiving bitrate, goodput.

o Packet Loss Rate (PLR).

o End to end Media frame delay. For video, this means the delay from capture to display.

o Transport delay.

o Algorithm stability in terms of rate variation.

4. Wi-Fi Networks Specific Test Cases

Given the prevalence of Internet access links over Wi-Fi, it is important to evaluate candidate RMCAT congestion control solutions over test cases that include Wi-Fi access lines. Such evaluations should also highlight the inherent different characteristics of Wi-Fi networks in contrast to wired networks:

o The wireless radio channel is subject to interference from nearby transmitters, multipath fading, and shadowing, causing fluctuations in link throughput and sometimes an error-prone communication environment

o Available network bandwidth is not only shared over the air between cocurrent users, but also between uplink and downlink traffic due to the half duplex nature of wireless transmission medium.

o Packet transmissions over Wi-Fi are susceptible to contentions and collisions over the air. Consequently, traffic load beyond a certain utilization level over a Wi-Fi network can introduce

frequent collisions over the air and significant network overhead, as well as packet drops due to buffer overflow at the transmitters. This, in turn, leads to excessive delay, retransmissions, packet losses and lower effective bandwidth for applications. Note, however, that the consequent delay and loss patterns caused by collisions are qualitatively different from those induced by congestion over a wired connection.

- The IEEE 802.11 standard (i.e., Wi-Fi) supports multi-rate transmission capabilities by dynamically choosing the most appropriate modulation scheme for a given received signal strength. A different choice of physical-layer rate leads to different application-layer throughput.

- Presence of legancy 802.11b networks can significantly slow down the the rest of a modern Wi-Fi Network. As discussed in [Heusse2003]since it takes longer to transmit the same packet over a slower link than over a faster link.

- Handover from one Wi-Fi Access Point (AP) to another may lead to packet delay and losses during the process.

- IEEE 802.11e defined EDCA/WMM (Enhanced DCF Channel Access/Wi-Fi Multi-Media) to give voice and video streams higher priority over pure data applications (e.g., file transfers).

In summary, presence of Wi-Fi access links in different network topologies can exert different impact on the network performance in terms of application-layer effective throughput, packet loss rate, and packet delivery delay. These, in turn, influence the behavior of end-to-end real-time multimedia congestion control.

Unless otherwise mentioned, test cases in this section are described using the underlying PHY- and MAC-layer parameters based on the IEEE 802.11n Standard. Statistics collected from enterprise Wi-Fi networks show that the two dominant physical modes are 802.11n and 802.11ac, accounting for 41% and 58% of connected devices. As Wi-Fi standards evolve over time, for instance, with the introduction of the emerging Wi-Fi 6 (802.11ax) products, the PHY- and MAC-layer test case specifications need to be updated accordingly to reflect such changes.

Typically, a Wi-Fi access network connects to a wired infrastructure. Either the wired or the Wi-Fi segment of the network could be the bottleneck. In the following sections, we describe basic test cases for both scenarios separately. The same set of performance metrics as in [I-D.ietf-rmcat-eval-test]) should be collected for each test case.

All test cases described below can be carried out using simulations, e.g. based on [ns-2] or [ns-3]. When feasible, it is also encouraged to perform testbed-based evaluations using Wi-Fi access points and endpoints running up-to-date IEEE 802.11 protocols, such as 802.11ac and the emerging Wi-Fi 6, to verify the viability of the candidate schemes.

4.1. Bottleneck in Wired Network

The test scenarios below are intended to mimic the setup of video conferencing over Wi-Fi connections from the home. Typically, the Wi-Fi home network is not congested and the bottleneck is present over the wired home access link. Although it is expected that test evaluation results from this section are similar to those from test cases defined for wired networks (see [I-D.ietf-rmcat-eval-test]), it is worthwhile to run through these tests as sanity checks.

4.1.1. Network topology

Figure 2 shows topology of the network for Wi-Fi test cases. The test contains multiple mobile nodes (MNs) connected to a common Wi-Fi access point (AP) and their corresponding wired clients on fixed nodes (FNs). Each connection carries either RMCAT or TCP traffic flow. Directions of the flows can be uplink, downlink, or bi-directional.

```
                                   uplink
                             +----------------->+
            +------+                                      +------+
            | MN_1 |))))                             /=====| FN_1 |
            +------+    ))                          //     +------+
                .        ))                        //          .
                .         ))                      //           .
                .          ))                    //            .
            +------+         +----+         +-----+        +------+
            | MN_N | ))))))) |    |         |     |========| FN_N |
            +------+         |    |         |     |        +------+
                             | AP |=========| FN0 |
           +----------+      |    |         |     |      +----------+
           | MN_tcp_1 | )))) |    |         |     |======| MN_tcp_1 |
           +----------+      +----+         +-----+      +----------+
                 .          ))                 \\              .
                 .         ))                   \\             .
                 .        ))                     \\            .
           +----------+  ))                       \\     +----------+
           | MN_tcp_M |)))                         \=====| MN_tcp_M |
           +----------+                                  +----------+
                            +<-----------------+
                                    downlink
```

Figure 2: Network topology for Wi-Fi test cases

4.1.2. Test setup

- o Test duration: 120s

- o Wi-Fi network characteristics:

  * Radio propagation model: Log-distance path loss propagation model [NS3WiFi]

  * PHY- and MAC-layer configuration: IEEE 802.11n

  * MCS Index at 11: 16-QAM 1/2, Raw Data Rate@52Mbps

- o Wired path characteristics:

  * Path capacity: 1Mbps

  * One-Way propagation delay: 50ms.

  * Maximum end-to-end jitter: 30ms

  * Bottleneck queue type: Drop tail.

      * Bottleneck queue size: 300ms.

      * Path loss ratio: 0%.

- Application characteristics:

  * Media Traffic:

    + Media type: Video

    + Media direction: See Section 4.1.3

    + Number of media sources (N): See Section 4.1.3

    + Media timeline:

      - Start time: 0s.

      - End time: 119s.

  * Competing traffic:

    + Type of sources: long-lived TCP or CBR over UDP

    + Traffic direction: See Section 4.1.3

    + Number of sources (M): See Section 4.1.3

    + Congestion control: Default TCP congestion control [RFC5681] or constant-bit-rate (CBR) traffic over UDP.

    + Traffic timeline: See Section 4.1.3

4.1.3. Typical test scenarios

- Single uplink RMCAT flow: N=1 with uplink direction and M=0.

- One pair of bi-directional RMCAT flows: N=2 (with one uplink flow and one downlink flow); M=0.

- One pair of bi-directional RMCAT flows, one on-off CBR over UDP flow on uplink: N=2 (with one uplink flow and one downlink flow); M=1 (uplink). CBR flow ON time at 0s-60s, OFF time at 60s-119s.

- One pair of bi-directional RMCAT flows, one off-on CBR over UDP flow on uplink: N=2 (with one uplink flow and one downlink flow); M=1 (uplink). OFF time for UDP flow: 0s-60s; ON time: 60s-119s.

- One RMCAT flow competing against one long-live TCP flow over uplink: N=1 (uplink) and M = 1(uplink), TCP start time at 0s and end time at 119s.

4.1.4. Expected behavior

- Single uplink RMCAT flow: the candidate algorithm is expected to detect the path capacity constraint, to converge to bottleneck link capacity and to adapt the flow to avoid unwanted oscillation when the sending bit rate is approaching the bottleneck link capacity. No excessive rate oscillations should be present.

- Bi-directional RMCAT flows: It is expected that the candidate algorithm is able to converge to the bottleneck capacity of the wired path on both directions despite presence of measurment noise over the Wi-Fi connection. In the presence of background TCP or CBR over UDP traffic, the rate of RMCAT flows should adapt in a timely manner to changes in the available bottleneck bandwidth.

- One RMCAT flow competing with long-live TCP flow over uplink: the candidate algorithm should be able to avoid congestion collapse, and to stablize at a fair share of the bottleneck link capacity.

4.2. Bottleneck in Wi-Fi Network

These test cases assume that the wired portion along the media path is well-provisioned whereas the bottleneck exists over the Wi-Fi access network. This is to mimic the application scenarios typically encountered by users in an enterprise environment or at a coffee house.

4.2.1. Network topology

Same as defined in Section 4.1.1

4.2.2. Test setup

- Test duration: 120s

- Wi-Fi network characteristics:

  * Radio propagation model: Log-distance path loss propagation model [NS3WiFi]

  * PHY- and MAC-layer configuration: IEEE 802.11n

  * MCS Index at 11: 16-QAM 1/2, Raw Data Rate at 52Mbps

o Wired path characteristics:

  * Path capacity: 100Mbps

  * One-Way propagation delay: 50ms.

  * Maximum end-to-end jitter: 30ms

  * Bottleneck queue type: Drop tail.

  * Bottleneck queue size: 300ms.

  * Path loss ratio: 0%.

o Application characteristics:

  * Media Traffic:

    + Media type: Video

    + Media direction: See Section 4.2.3

    + Number of media sources (N): See Section 4.2.3

    + Media timeline:

      - Start time: 0s.

      - End time: 119s.

  * Competing traffic:

    + Type of sources: long-lived TCP or CBR over UDP

    + Number of sources (M): See Section 4.2.3

    + Traffic direction: See Section 4.2.3

    + Congestion control: Default TCP congestion control [RFC5681] or constant-bit-rate (CBR) traffic over UDP

    + Traffic timeline: See Section 4.2.3

4.2.3. Typical test scenarios

This section describes a few test scenarios that are deemed as important for understanding the behavior of a RMCAT candidate solution over a Wi-Fi network.

- Multiple RMCAT Flows Sharing the Wireless Downlink: N=16 (all downlink); M = 0. This test case is for studying the impact of contention on competing RMCAT flows. For an 802.11n network, given the MCS Index of 11 and the corresponding raw data rate of 52Mbps, the total application-layer throughput (assuming reasonable distance, low interference and infrequent contentions caused by competing streams) is around 20Mbps. Consequently, a total of N=16 RMCAT flows are needed to saturate the wireless interface in this experiment. Evaluation of a given candidate solution should focus on whether downlink RMCAT flows can stablize at a fair share of total application-layer throughput.

- Multiple RMCAT Flows Sharing the Wireless Uplink: N = 16 (all downlink); M = 0. When multiple clients attempt to transmit video packets uplink over the wireless interface, they introduce more frequent contentions and potential collisions. Per-flow throughput is expected to be lower than that in the previous downlink-only scenario. Evaluation of a given candidate solution should focus on whether uplink flows can stablize at a fair share of application-layer throughput.

- Multiple Bi-directional RMCAT Flows: N = 16 (8 uplink and 8 downlink); M = 0. The goal of this test is to evaluate performance of the candidate solution in terms of bandwidth fairness between uplink and downlink flows.

- Multiple Bi-directional RMCAT Flows with on-off CBR traffic: N = 16 (8 uplink and 8 downlink); M = 5(uplink). The goal of this test is to evaluate adaptation behavior of the candidate solution when its available bandwidth changes due to departure of background traffic. The background traffic consists of several (e.g., M=5) CBR flows transported over UDP, which are ON at times t=0-60s and are OFF at times t=61-120s.

- Multiple Bi-directional RMCAT Flows with off-on CBR traffic: N = 16 (8 uplink and 8 downlink); M = 5(uplink). The goal of this test is to evaluate adaptation behavior of the candidate solution when its available bandwidth changes due to arrival of background traffic. The background traffic consists of several (e.g., M=5) parallel CBR flows transported over UDP, which are OFF at times t=0-60s and are ON at times t=61-120s.

- Multiple Bi-directional RMCAT flows in the presence of background TCP traffic: N=16 (8 uplink and 8 downlink); M = 5 (uplink). The goal of this test is to evaluate how RMCAT flows compete against TCP over a congested Wi-Fi network for a given candidate solution. TCP start time: 40s, end time: 80s.

- o Varying number of RMCAT flows. A series of tests can be carried out for the above test cases with different values of N, e.g., N = [4, 8, 12, 16, 20]. The goal of this test is to evaluate how a candidate RMCAT solution responds to varying traffic load/demand over a congested Wi-Fi network. The start time of these RMCAT flows is randomly distributed within a window of t=0-10s, whereas their end times are randomly distributed within a window of t=110-120s.

4.2.4. Expected behavior

- o Multiple downlink RMCAT flows: each RMCAT flow should get its fair share of the total bottleneck link bandwidth. Overall bandwidth usage should not be significantly lower than that experienced by the same number of concurrent downlink TCP flows. In other words, the performance of multiple concurrent TCP flows will be used as a performance benchmark for this test scenario. The end-to-end delay and packet loss ratio experienced by each flow should be within acceptable range for real-time multimedia applications.

- o Multiple uplink RMCAT flows: overall bandwidth usage shared by all RMCAT flows should not be significantly lower than that experienced by the same number of concurrent uplink TCP flows. In other words, the performance of multiple concurrent TCP flows will be used as a performance benchmark for this test scenario.

- o Multiple bi-directional RMCAT flows with dynamic background traffic carrying CBR flows over UDP: RMCAT flows should adapt in a timely fashion to the resulting changes in available bandwidth.

- o Multiple bi-directional RMCAT flows with dynamic background traffic over TCP: during the presence of TCP background flows, the overall bandwidth usage shared by all RMCAT flows should not be significantly lower than those achieved by the same number of bi-directional TCP flows. In other words, the performance of multiple concurrent TCP flows will be used as a performance benchmark for this test scenario. All downlink RMCAT flows are expected to obtain similar bandwidth with respect to each other. The throughput of RMCAT flows should decrease upon the arrival of TCP background traffic and increase upon their departure, both reactions should occur in a timely fashion (e.g., within 10s of seconds).

- o Varying number of RMCAT flows: the test results for varying values of N -- while keeping all other parameters constant -- is expected to show steady and stable per-flow throughtput for each value of N. The average throughput of all RMCAT flows is expected to stay constant around the maximum rate when N is small, then gradually

decrease with increasing number of RMCAT flows till it reaches the minimum allowed rate, beyond which the offered load to the Wi-Fi network (with a large value of N) is exceeding its capacity.

4.3. Other Potential Test Cases

4.3.1. EDCA/WMM usage

EDCA/WMM is prioritized QoS with four traffic classes (or Access Categories) with differing priorities. RMCAT flows should achieve better performance (i.e., lower delay, fewer packet losses) with EDCA/WMM enabled when competing against non-interactive background traffic (e.g., file transfers). When most of the traffic over Wi-Fi is dominated by media, however, turning on WMM may actually degrade performance since all media flows now attempt to access the wireless transmission medium more aggressively, thereby causing more frequent collisions and collision-induced losses. This is a topic worthy of further investigation.

4.3.2. Effects of Legacy 802.11b Devices

When there is 802.11b devices connected to modern 802.11 network, it may affect the performance of the whole network. Additional test cases can be added to evaluate the affects of legancy devices on the performance of RMCAT congestion control algorithm.

5. Conclusion

This document defines a collection of test cases that are considered important for cellular and Wi-Fi networks. Moreover, this document also provides a framework for defining additional test cases over wireless cellular/Wi-Fi networks.

6. IANA Considerations

This memo includes no request to IANA.

7. Security Considerations

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

The evaluation of the test cases are intended to be run in a controlled lab environment. Hence, the applications, simulators and network nodes ought to be well-behaved and should not impact the

desired results. It is important to take appropriate caution to avoid leaking non-responsive traffic from unproven congestion avoidance techniques onto the open Internet.

8. Acknowledgments

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Authors' Addresses

Zaheduzzaman Sarker
Ericsson AB
Laboratoriegraend 11
Luleae 97753
Sweden

Phone: +46 107173743
Email: zaheduzzaman.sarker@ericsson.com

Ingemar Johansson
Ericsson AB
Laboratoriegraend 11
Luleae 97753
Sweden

Phone: +46 10 7143042
Email: ingemar.s.johansson@ericsson.com

Xiaoqing Zhu
Cisco Systems
12515 Research Blvd., Building 4
Austin, TX 78759
USA

Email: xiaoqzhu@cisco.com

Jiantao Fu
Cisco Systems
707 Tasman Drive
Milpitas, CA 95035
USA

Email: jianfu@cisco.com

Wei-Tian Tan
Cisco Systems
725 Alder Drive
Milpitas, CA 95035
USA

Email: dtan2@cisco.com

Michael A. Ramalho
Cisco Systems, Inc.
8000 Hawkins Road
Sarasota, FL 34241
USA

Phone: +1 919 476 2038
Email: mramalho@cisco.com