Packet Probing Study to Assess Sustainability in Available Bandwidth Measurements: Case of High-Speed Cellular Networks

Al-Hussein Hameed Jasim[1], Niclas Ögren[2], Dimitar Minovski[1,2], and Karl Andersson[1*]
[1]*Department of Computer Science, Electrical and Space Engineering*
Luleå University of Technology, SE-931 87, Skellefteå, Sweden
jasalh-7@student.ltu.se, {dimitar.minovski, karl.andersson}@ltu.se
[2]*InfoVista Sweden AB, SE-931 62, Skellefteå, Sweden*
{niclas.ogren, dimitar.minovski}@infovista.com

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Abstract

Facts and figures indicate that the latest generations of cellular networks are likely to become the dominant medium of the global data exchange. This may pose a challenge to service providers trying to improve the Quality of Service (QoS) provided that is usually specified in Service Level Agreement (SLA), which, in turn, has to be practically verified. It requires methods and tools to measure the adopted QoS metrics such as bandwidth and Round-Trip Time (RTT). For this research, the InfoVista's proprietary tool (Blixt™) was used. The research discusses how the probing packet parameters play a vital role in determining the accuracy of the measurements, the level of intrusiveness in a shared-resources network, and its implications for sustainability for this application. The experiments were carried out in a live commercial network and the performance was also compared with other cutting-edge available bandwidth measurement tools in a multi-carrier scenario.

Keywords: network performance, active measurement, available bandwidth, 4G/LTE

1 Introduction

Over the past ten years, the cellular networks technologies have witnessed a phenomenal development from High-Speed Packet Access (HSPA) to the 4G Long Term Evolution-Advanced (LTE-A) technology [1] and launching the fifth-generation (5G) networks has become closer than ever according to Ericsson mobility report [2].

At the end of 2017, LTE became the most widely used mobile access technology in the world and its subscriptions continue to increase markedly as it is predicted to form about 60 percent of the total number of subscriptions to reach 5.5 billion in three years [2]. Ericsson's report also foresees that video applications are expected to account for 73 percent of the total mobile data exchange in 2023 which is due to users' preference for using rich in content and cost-effective networks over traditional telephony services, namely SMS and voice calling. The necessity of having higher bandwidth and low latency in accessing different services such as data, voice, and video wirelessly and ubiquitously, is the main incentive associated with that shift [3]. The promising progress of cellular technologies makes the Internet of Things (IoT) more accessible to all with the use of real-time applications to control machines via mobile

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*Corresponding author: Department of Computer Science, Electrical and Space Engineering, Luleå University of Technology, SE-931 87, Skellefteå, Sweden, Tel: +46-(0)910-585364

devices, especially through 4G networks and possibly over the upcoming 5G networks in the near future [3]. The number of cellular IoT connections continues to grow and it is predicted to reach 3.5 billion by 2023 following an annual growth rate of 30% [2]. The exceptional evolution of data transmission rates over the LTE-Advanced and the forthcoming 5G networks will enhance the use of real-time applications such as video streaming.

LTE networks are heterogeneous cells that have various Quality of Service (QoS) metrics, for instance, signal strength, bandwidth, coverage range, access time, etc. One of the pivotal QoS metrics is the end-to-end available bandwidth which represents the average of the unused (spare) capacity of a link or a path during the given period of time [4]. In other words, the maximum throughput that the data is delivered with over a path without affecting the service quality or leading to considerable degradation of other ongoing data flows [5]. The link with the minimum available bandwidth is called the tight link of the end-to-end path, whereas the link with the minimum capacity is called the narrow link [6]. To put the available bandwidth into a mathematical model of a link l [7, 8], U_l is the average usage of link l which varies from 0 to 1, and C_l represents the capacity of link l. Then, A_l is the average available bandwidth of the same link l is specified by the unused portion of the capacity:

$$A_l = (1 - U_l)\,C_l \tag{1}$$

For N links a path, the bandwidth definition can be expanded as follows:

$$A = \min_{l=1,\ldots,N} A_l \tag{2}$$

The available bandwidth of the path that has the lowest available bandwidth of all N links is referred to as A.

The growth of bandwidth demand and users' Quality of Experience (QoE) in the context of mobile applications have become indispensable [9]. The available bandwidth has to be measured as an essential metric to keep track of network performance and to ensure the QoS [9]. Attaining a precise measurement of available bandwidth on network hops or end-to-end paths is critical for different network functions [10]. The available bandwidth plays a major role in network management and capacity planning. Generally, bandwidth estimation approaches are used for analyzing network performance, improve end-to-end transfer performance, service level agreement (SLA) verification, load balancing, admission control, capacity planning support, congestion control, etc. [11]. Furthermore, the knowledge attained from bandwidth estimation techniques has been used to enhance the QoS of video streaming applications multimedia services since the available bandwidth directly impacts the application performance [12]. Bandwidth estimation plays a vital role in the case of multimedia communications since it notoriously degrades the performance based on the variance in mobile networks throughput. In cellular networks, if the transmitting rate of the video call surpasses the available bandwidth could lead to needless delays in video frames which affect the interactivity of the video call [13].

The high-speed networks have to be taken into consideration as they are boosted by the explosion of multimedia services and the latest communication technology. The next generation of high-speed networks is about to emerge, propelling bandwidth measurement methods to extend their compatibility to ultra high-speeds [14]. The existing tools for measuring the available bandwidth are primarily drafted and developed for controlled links' paths [11]. Wireless networks have different characteristics than those in wired networks like rapid fluctuations and unsteadiness. Thus, a number of tools have been evaluated to determine their adequacy to be used for cellular networks. The issue that comes with the majority of these tools is the underestimation of the available bandwidth [15] in consequence of the probing process and the post-analysis for estimation delivery that requires some time. Additionally, the lack of featuring real-time measurements appears to be the main constraint of such tools e.g. pathload [16] over mobile networks. In most cases, there are instantaneous variations in radio conditions. Hence, if the tool takes considerable time to estimate, it increases the probability of not reflecting the present bandwidth condition.

The intrusiveness created by the probing traffic is a serious concern that active measurement tools have to take into account. It can be defined as the property of a tool related to the amount of traffic it injects into the network [7]. The traffic depends on the tool used and particularly on the measured parameter. Prasad et al. [4] studied the intrusiveness concept in bandwidth estimation tools and stated that *"an active measurement tool is intrusive when its average probing traffic during the measurement process is significant compared to the available bandwidth in the path."*. This intrusive (or invasive) characteristic can modify the properties which are meant to be measured. First, it can result in increasing the load on the network; secondly, it can lead to estimation errors or bias.

The estimations produced by the currently available tools show that they are clearly impacted by conditions in wireless networks. For instance, link speeds variations, the adaptation rate, the complexity of network functionality and traffic contention [17], a varying convergence time, as well as a considerable level of intrusiveness. The challenges mentioned make those tools impractical, as it is required to have accurate and fast measurements. Additionally, time precision is another issue that emerges in high-speed links where the estimations become more sensitive to errors due to the reduction in time intervals between packets.

This research follows a Quantitative research method since it was carried out with an experimental-based approach. This method is generally used when the theoretical approach is unattainable or insufficient. The experimental approach was followed since there is no theoretical basis to build upon. This research addresses the available bandwidth measurement of the downlink of LTE networks, given that the downlink is the preponderant in this type of networks as the data traffic size passing through this path can be ten times higher (it utilizes carrier aggregation technique) as that of the uplink due to the asymmetry of the system.

The article is organized as follows: In the second section, the traditional models and methods used to measure the available bandwidth are discussed and mainly focused on the active measurement methods. Blixt™'s implementation and technologies used are illustrated in the third section. In section four, the experiments, findings, and the measurement accuracy evaluation are presented followed by the challenges in the fifth section. In section six, a conclusion is drawn.

2 Related work

There is a set of tools that estimate the network's end-to-end available bandwidth and capacity. In general, the bandwidth of a link or a path can either be measured by passively monitoring the amount of traffic flowing through it, or by actively probing the link to estimate its bandwidth.

The passive measurements are based on observing the data traffic passes through the network without interference, where the available bandwidth is estimated based on the delay, packet loss, etc. [12]. The passive monitoring mechanism requires access to intermediate network elements or administrative resources [18]. Active measurements, on the other hand, do not require to access all nodes in the network. The active measurements are based on probing the network by transmitting packets between two points to observe the network's behavior and to estimate the available bandwidth accordingly [19]. Active probing is usually considered due to estimates' efficiency and reliability.

The active measurements also are divided into two models, the Probe Rate Model (PRM) and the Probe Gap Model (PGM). The PRM is also known as the Iterative Probing Self-induced congestion-based technique [12]. There are two cases to consider in PRM, if probe packets are sent at lower rates than the available bandwidth, the arrival rate of the packets at the receiver is anticipated to match the transmission rate set at the sender [20]. On the contrary, if the probe-packet rate is higher than the available bandwidth, congestion occurs in the bottleneck hop and the probe packets face delays; the estimation method is then to determine the rate at which the transition between these two behaviors happens. The available bandwidth, in this case, is equal to the maximum rate at which the sending rate matches the receiving rate. This model has been applied in several tools, including pathload [16], pathChirp [21], BART [22] and PTR. On the other hand, PGM utilizes the information in the time interval gap between the arrival of two consecutive probes in the receiver [23], Spruce tool [24] is an example of where this model is used. The concept of PGM is to capture the correlation between the observed dispersion of a packet-pair at the receiver and the rate of cross-traffic at the bottleneck node of a path. The PGM assumes that both the tight link and the narrow link are the same and represent the bottleneck link [25]. According to the assumption that the end-to-end capacity C is known (or measured) and by knowing the cross-traffic rate at the bottleneck CT, it is possible to calculate the available bandwidth of the path, $A = C - CT$. The available bandwidth is calculated by subtracting the cross-traffic rate at the bottleneck from the end-to-end capacity.

There are several techniques used for estimating the available bandwidth, and the most common ones are Variable Packet Size (VPS) probing, Packet Pair/Train Dispersion (PPTD), Self-Loading Periodic Streams (SLoPS), Trains of Packet Pairs (ToPP), and packet tailgating technique [7]. VPS is used for measuring the link capacity, whilst PPTD is used in estimating the path capacity. SLoPS and TOPP are used for measuring the end-to-end available bandwidth. The accuracy of available bandwidth-related measurements remains a challenge. Plenty of researchers' efforts have been put into using filters to improve the estimation accuracy, for example, Exponential Weight Moving Average Filter (EWMA), Vertical Horizontal Filter (VHF), Kalman Filter and Fuzzy Filter [26, 27]. These filters have been applied to various measurement tools.

Based on research in this area [18], it has been found that one of the main factors in active measurements of the available bandwidth is how the probing packets are sequenced and structured. For instance, there are some simple tools and protocols which use single packets such as ping, whereas Spruce, and IGI use packet pair technique. Moreover, TOPP, BART, and pathload use packet trains with equally-spaced gaps between the trains. PathChirp also uses packet trains but the difference between these tools lies in the number of packets and the distance between them (pathChirp uses exponentially-spaced gaps).

The tools mentioned are mainly developed to be compatible with wired networks, E. Bergfeldt et al. [11] assessed the performance of pathChirp, BART and Pathload in HSDPA live commercial network, and they have found that pathChirp and BART deliver more accurate estimations in radio environment due to their real-time properties. Most of the tools remain slightly insufficient since the packet probing and the analysis require some time to provide an estimate.

2.1 Packet Probe

The structure of the probing packet directly impacts the available bandwidth estimation which resulted in using different forms of probing packets. V. Mohan et al. [7] define the probe packet as an additional packet that can be in any format depending on the information required to be observed. A small UDP packet that has a little payload and a timestamp could serve as an example of a probe packet. Probe packets and their properties have to be properly chosen to simulate the actual network traffic [28]. For instance, to measure the network delay, it is inadvisable to use ICMP packets since routers don't prioritize them, thus they are not treated as normal traffic. In the interest of getting a clear view of the network delay, UDP packets are required to be used instead.

2.2 The State-of-The-Art Tools

In the following sub-sections, the latest methods to measure the bandwidth of cellular networks are presented.

2.2.1 NEXT

A. K. Paul et al. [29] proposed NEXT (New Enhanced Available Bandwidth Measurement Technique) for estimating the available bandwidth based on pathChirp's excursion detection algorithm. The structure of their proposed probing method has a region in which the packet rates are sampled more often than in other areas. Another algorithm was also proposed to adjust the packets' rate in each cycle in a way that the predicted available bandwidth would fit into the packet compact areas. The rate adjustment algorithm has two spread factors to decrease the number of packets sent over the network. One of the spread factors is the *"ratio of successive packet inter-spacing times within a chirp"*. The transmitter node of NEXT systematically sends probing streams of length P packets, each packet has a different sending rate through controlling the inter-packet gaps. On the other hand, the NEXT receiver captures the arrival time of packets individually and it calculates the queuing delay based on the information obtained from the sender's time stamps. In case that the packet's rate is lower than the available bandwidth, no queuing delay occurs. Contrarily, the queuing delay happens when the packet's rate is higher than the available bandwidth. The receiver finds the turning point in which the queuing delay starts to constantly increase and use it for available bandwidth estimation.

NEXT, in like manner to pathChirp, it also estimates the end-to-end available bandwidth by sending a number of packet chirps P (numbered as $P = 1, 2, \ldots, n$) from sender to receiver. Based on rates and the spreading factor, each train of packets has a specific number of packets. The traffic structure and spacing between the packets represent the differences between pathChirp and NEXT despite that both tools use a train of packets of increasing delays. The tool was implemented and evaluated in a simulated testbed and it was also compared with other Available Bandwidth Measurements (ABM) tools, such as Spruce, pathChirp, and IGI. The implementation of each of these tools was done in the ns-2 network simulator. According to the estimations, NEXT showed better estimations than in pathChirp in terms of accuracy and intrusiveness.

A. K. Paul et al. [29] introduced NEXT-V2 [20] which is an updated version of NEXT with the aim of achieving more accurate estimations. In NEXT-V2, the algorithms of packet loss recovery and excursion detection have been modified. NEXT-V2 estimates the available bandwidth by finding the turning point based on the observation of a one-way queuing delay.

NEXT-V2 is not the latest version of NEXT, A. K. Paul et al. extended their tool and presented NEXT-FIT [30] which targets estimating the available bandwidth over LTE networks. NEXT-FIT uses long probe-packet trains and a "parameter-independent curve-fitting" technique was used instead of pathChirp's excursion detection algorithm to estimate the available bandwidth from a one-way queuing delay (as in the case of NEXT-V2). The tool was evaluated in a commercial 4G/LTE mobile communication network of a Japanese mobile operator.

2.2.2 PathQuick

PathQuick [31] follows PRM and it is designed to understand and reflect on the complex behavior of queuing delays over operational networks. Generally, the idea of PathQuick can be briefly explained as having a sender that sends a UDP probing packet train to a receiver. At the sender's node, each packet is timestamped and used by the receiver coupled with its own timestamps to deliver an estimation of the available bandwidth and reports it back to the sender through a UDP packet. PathQuick3 [32] is based on the concept of the *"curve fitting along with the nonlinear least-squares method"*.

The authors of PathQuick3 stated that it represents the successful version of their previous releases, as both PathQuick and PathQuick2 were primarily intended to be used for wired networks. PathQuick3 was evaluated practically against pathChirp over an LTE network in the downlink path and the experiments showed that pathQuick3 estimations are more accurate.

2.2.3 PathML

In 2017, N. Sato et al. [33] proposed PathML which is an approach that uses machine learning algorithms with a large amount of data in a data-driven model to estimate the available bandwidth. Supervised learning was applied as the task is to use queuing delays (input) to predict the available bandwidth (output) from. Four machine learning algorithms were selected, which are: kernel ridge regression, convolutional neural network, random forests, and support vector regression. The machine learning libraries, TensorFlow [34] and Scikit-learn [35], were used to implement the system with the previously mentioned algorithms. The experiments were carried out over an operational LTE network (Japan's primary mobile operator) and the observation was that the convolutional neural network algorithm seemed to deliver the most accurate estimates. This approach was also evaluated against PathQuick3.

The learning procedure can be briefly explained as follows: initially, the training dataset was prepared which is a considerable amount of queuing delays pairs captured at a receiver and corresponding available bandwidth values. The next step was generating a predictor that predicts the values of the available bandwidth based on the observed queuing delays. The predictor uses patterns according to the relationship between available bandwidth and the queuing delays, for instance, in the case of a small number of packets, the transition point refers to low available bandwidth and in contrary to a large number of packets. By adding a new set of queuing delays into the predictor, it becomes feasible to obtain the available bandwidth estimated value.

3 Blixt™: Case Study

Blixt™ ABM tool was selected to study the aspects of sustainability in available bandwidth active measurement for high-speed cellular networks.

3.1 TWAMP

Blixt™ relies on a time-stamping protocol which is an improved version of the Two-Way Active Measurement Protocol (TWAMP). TWAMP is considered as the first standardized protocol of its kind which enables an effective performance monitoring process by collecting two-way metric data in one measurement as it is defined in RFC 5357 [36]. The TWAMP protocol is an expanded version of the One-Way Active Measurement Protocol (OWAMP) specification defined in RFC 4656 with the addition of the performance measurement of round-trip and two-way metrics for IP based networks. TWAMP provides a flexible way to accurately measure unidirectional performance and routing between two endpoints supported by TWAMP. TWAMP, in turn, consists of two interrelated protocols, TWAMP-Control and TWAMP-Test. TWAMP-Control protocol initiates, starts, and stops test sessions as well as fetching the results. Practically, the Control-Client and Session-Sender roles are implemented as ”Controller” on one host while the Session-Reflector and the Server roles are implemented as ”Responder” on the other host. This model supports Full-TWAMP protocol architecture. C. Kocak and K. Zaim [37] highly recommended using TWAMP to measure LTE's QoS metrics due to their importance and sensitivity to changes.

3.2 Blixt™ Algorithm and Measurement Procedure

The concept of design is based primarily on a client that sends packets to an ABM server, the server, in turn, reflects the packets back when received to the sender along with timestamps and other information contained in the packets. The client-side can be easily configured to test different points of the network by connecting to different servers. The packet trains used for probing the network are structured to fully utilize the maximum available bandwidth, without constraining the throughput by the slow start mechanisms or low-load scheduling. The data is sent in short bursts (”chirps”) with much longer gaps between them. The aim of using short and intense bursts is to meet the requirements of a sufficiently high provisional resolution (with minimal interference), which implies that at least once in a while it can be expected that the optimal radio conditions of the radio may prevail throughout the burst data (As long as the device location and the network configuration allow this to start with). As there are multiple packets are being sent back-to-back and scheduled in successive TTIs, the network’s full-capacity is ensured to be allotted to the ABM tool at least at some TTIs in the middle of the burst. The reason why multiple contiguous packets are being sent is that one TTI is presumably not to be filled or scheduled via TTI if only one packet is being sent which means that the available bandwidth will not be fully utilized in each TTI. The peak load is fairly high to reach the network’s theoretical maximum, whereas the average load remains low. This scheme enables the possibility of examining the available bandwidth while keeping the use of network resources as minimum as possible. The algorithm is suited for network configuration parameters which implies the amount of data sent is adapted following the maximum throughput of the network while keeping the intrusiveness level to a minimum at any point.

Data is sent in bursts at a rate of one-second intervals (by default). Bursts always form a small fraction of a second and nothing is sent between these bursts. Each burst of data consists of a number of packets transmitted respectively and jointly referred to as a packet train. According to TWAMP specifications, the packets get four timestamps, the first one t_1 when the session-sender sends the packets to the session-reflector, while t_2 represents the timestamp at the server (session-reflector) once it receives the packets. In a similar manner, the reflector timestamps the packets as soon as it reflects them back to the client t_3 and the last one t_4 happens at the time that the client receives the reflected packets from the server.

With the use of the information related to the timestamping, the packet size and the number of packets sent to the server, it becomes feasible to estimate the available bandwidth for the downlink D_l by using the following general formula:

$$D_l(bps) = \frac{packet\ size(bytes) \times 8 \times no.\ of\ packets\ per\ train}{t_{4\ of\ the\ last\ packet} - t_{4\ of\ the\ first\ packet}} \tag{3}$$

Where the packet size is in bytes.

3.3 Blixt™ Implementation

The implementation of Blixt™ considers a two-host scenario which includes a session-sender and a session-reflector. The ABM tool is developed for mobile devices powered by Android to serve as a session sender by generating probing packets, whereas the TWAMP server or the session-reflector was implemented in a Linux machine.

Following the PRM technique, Blixt™ session-sender initiates the TWAMP session by sending UDP packets as probe traffic to the session-reflector. As an input, the IP address and port number associated with the session are delivered by the tool to the TWAMP session-reflector. Moreover, it is also possible to specify the size of the packet, the number of packets per train to be sent within the interval time. In order to compute the available bandwidth, the application receives the reflected train of packets from the TWAMP server along with the timestamps. The measurement’s values are displayed in a graph. It displays the values in Mbps for downlink available bandwidth. The device was used for the measurements is Sony XZ which is powered by Qualcomm® Snapdragon™ 820, 64-bit Quad-core processor and it supports LTE (4G) Cat 9 (device specifications according to SONY mobile).

Blixt™ session-reflector is a Linux-based server that communicates through a single port as specified in TWAMP standards for exchanging TWAMP test packets, on which the server takes the inbound UDP packets coming from different clients. The server receives a train of packets from the client which are timestamped on reception and buffered until the last packet in the train is received. Upon reception of the last packet in the train, it reflects back the entire train of packets with an interval as specified in the received packet information. Thus, the server solely serves as a reflector that does not carry out any computations on the received packets in contrast to the OWAMP.

InfoVista has tested out the performance of Blixt™ and compared its accuracy to the FTP throughput. The figure below (Figure 1) illustrates how well the Downlink (DL) ABM correlates with the FTP

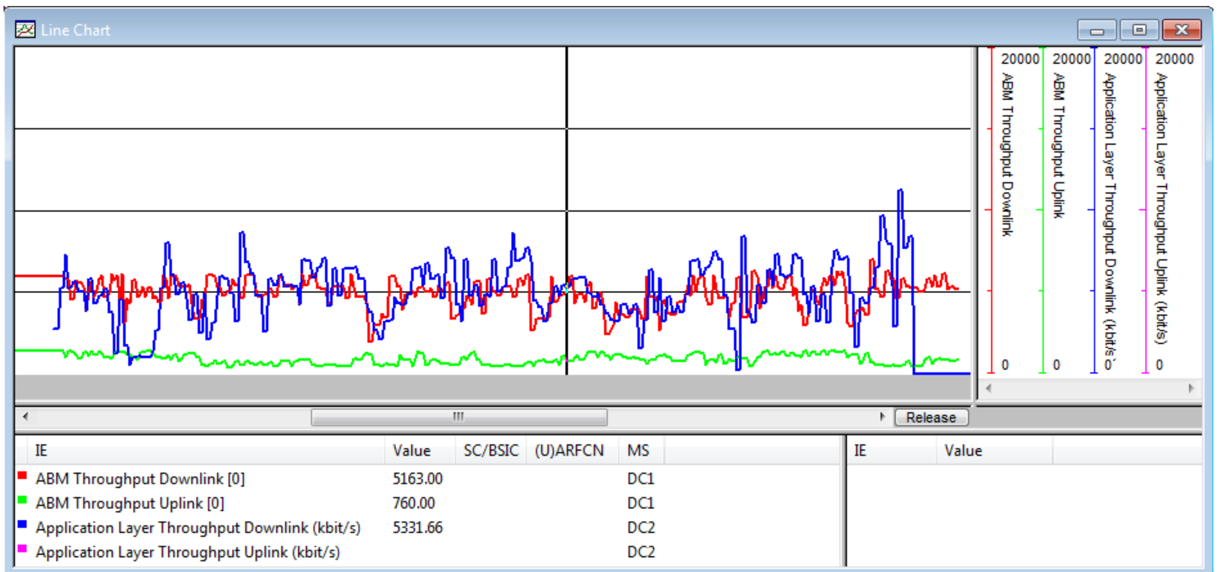


Figure 1: Blixt™'s ABM vs the FTP throughput in single carrier scenario

throughput in the single carrier scenario. The application layer throughput represents the FTP throughput, and Blixt's application layer throughput is represented by the ABM throughput for the DL.

3.4 Probing Packet Parameters

In order to find the correlation between the variance in parameters' values and accuracy of the measurements, Blixt™ has been modified to manually tune these values. The user interface illustrated in Figure 2 shows the probing packet parameters fields as inputs.

3.4.1 Probing Packet Size

The probing packets size is a crucial parameter and its value should be chosen appropriately. If the size of the packet is chosen to be relatively large, this can lead to an increase in time interval and results in interfering with the data exchanged in the network. On the other hand, choosing a small packet size may not be an optimal solution either. The packet dispersion recedes as the packet length dwindles and makes the measurements prone to the case in which the first packet encounters a longer delay than the rest. The experiments also present that sending small probing packets increases the susceptibility to interference which makes it sound to send reasonably large probing packets.

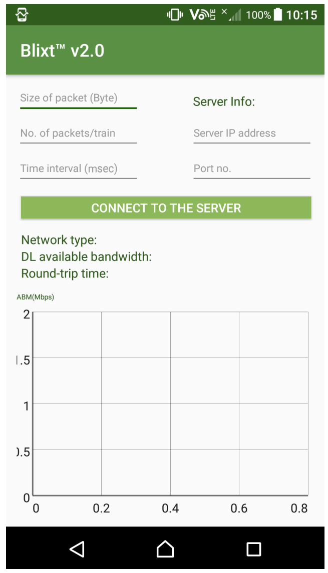


Figure 2: Blixt™'s client UI

3.4.2 Number of Probing Packets

The data traffic over the internet consists of data burst which implies a short glance is not sufficient to test the average traffic load which is the reason for sending a somewhat fairly large number of packet trains to probe the network [38]. It should also be borne in mind that excessively sending packets can lead to larger queues and cause to increase the network load as well as packet loss.

3.4.3 Transmission Time Interval (TTI)

The concept of drive testing is to collect RSL measurements from different regions, which in turn vary in the distances adopted in the test. The procedure for shaping a coverage area is known as binning. Mobile operators ordinarily measure the coverage area based on bin sizes which are 50x50 and 25x25 meters of resolution. In the case of a car driving at speed of 90 km/h, the available bandwidth has to be measured every 1s, $T_t = (25 * 3{,}600) / (90{,}000) = 1$s, where T_t is the transmission time interval.

4 Experiments and Results

The experiments were carried out over a live commercial 4G/LTE network, where the radio conditions are likely to vary over time. Two mobile operators in Sweden, A, and B, were chosen for the experiments (the results provided are obtained from the experiments performed over the network of one operator). To have higher bandwidth, the experiments were mainly conducted in the late evenings. A sports venue was chosen to perform the measurements at, based on the information gathered from the Bredbandskartan.se website which provides information about the broadband connection in Sweden. Figure 3 shows the measurements setup for Blixt™.

The maximum achievable FTP throughput was used as ground truth of the actual available bandwidth [18] as it is infeasible to access the intermediate network nodes of the mobile operator to get the available bandwidth. It has also been used by other researchers along with the TCP throughput [30]. For the FTP download test, TEMS™ pocket's test example was used which is an automated testing service provided by a previously written script with the possibility of starting and stopping the test with menu

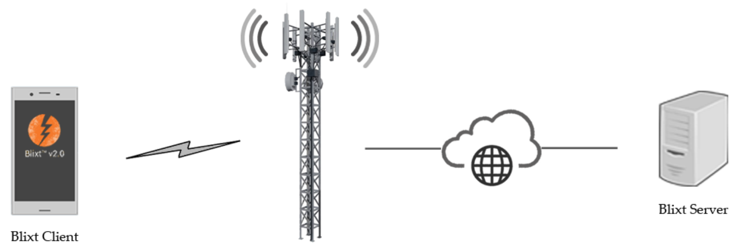


Figure 3: Blixt™ measurements testbed

commands. The file used for the FTP test is a Linux distribution (Arch Linux) for computers based on x86-64 architectures with a size of approximately 550MB. In the beginning, the FTP test was performed to measure the throughput which was followed by measuring the available bandwidth of the downlink using Blixt™. The procedure was then reversed to observe if any changes occur.

With the objective to assure the consistency of the results, and due to the infeasibility of designing a controlled testbed for measuring the available bandwidth in mobile networks, the alternative was to carry out multiple experiments in comparable circumstances (Same device, location and time). The tests were alternated between Blixt™ and the FTP test to align the assessment results. In the first set of experiments, the values of the three main probing packet parameters which are: the size of a packet, the number of packets per train, and the Transmission Time Interval, were tuned (one at a time) to observe how the available bandwidth varies accordingly.

The second part of the experiments lay emphasis evaluating Blixt™ against the FTP test, iPerf 3, and nPerf in terms of the accuracy and the intrusiveness.

4.1 Probing Packet Parameters Study

In the first set of experiments, the probing packet parameters were studied.

4.1.1 Packet Size

In an effort to examine how changing the packet size impacts the available bandwidth, the number of packets and the TTI were kept constant, and equal to 190 packets, 1,000 ms respectively [39]. As illustrated in Figure 4, the larger the packet size is, the higher the available bandwidth estimations are.

The underlying rationale for these observations is attributable to the active probing method's behavior, i.e. the larger the packet probes is, the closer to the actual throughput achieved by downloading a large file it gets. With FTP throughput as a reference, it was observed that the packet size of 1,400 bytes makes ameliorated measurements than those of smaller packet sizes.

4.1.2 Number of Packets Per Train

The number of packets is a pivotal parameter that highly impacts the measurement accuracy and the intrusiveness level which represent the main concerns regarding the available bandwidth measurement in high-speed mobile networks. The number of packets ranged between 20 up to 4,000, whilst the packet size and the TTI remained as constants and equal to 1,400 bytes and 1,000 ms respectively. The measurements obtained were also compared to the FTP test's throughput values.

The observation is that the available bandwidth value is an increasing function of the number of packets and its average comes closer to the FTP throughput. The acceptable estimation occurs at the range of 130 to 500 packets and to keep the level of intrusiveness as minimum as possible, it is recommended

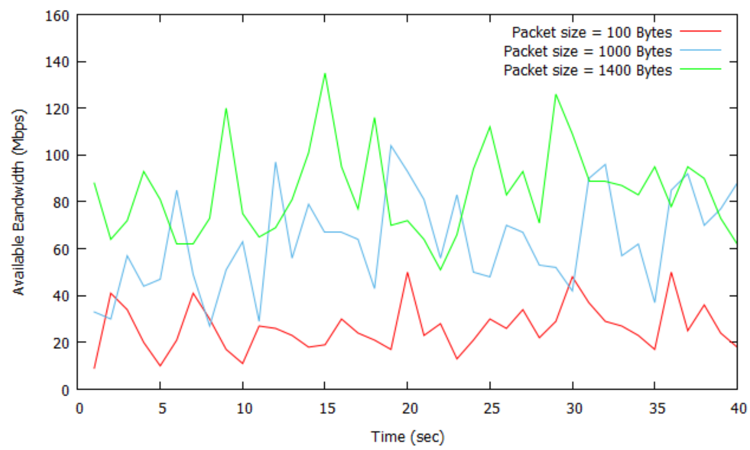
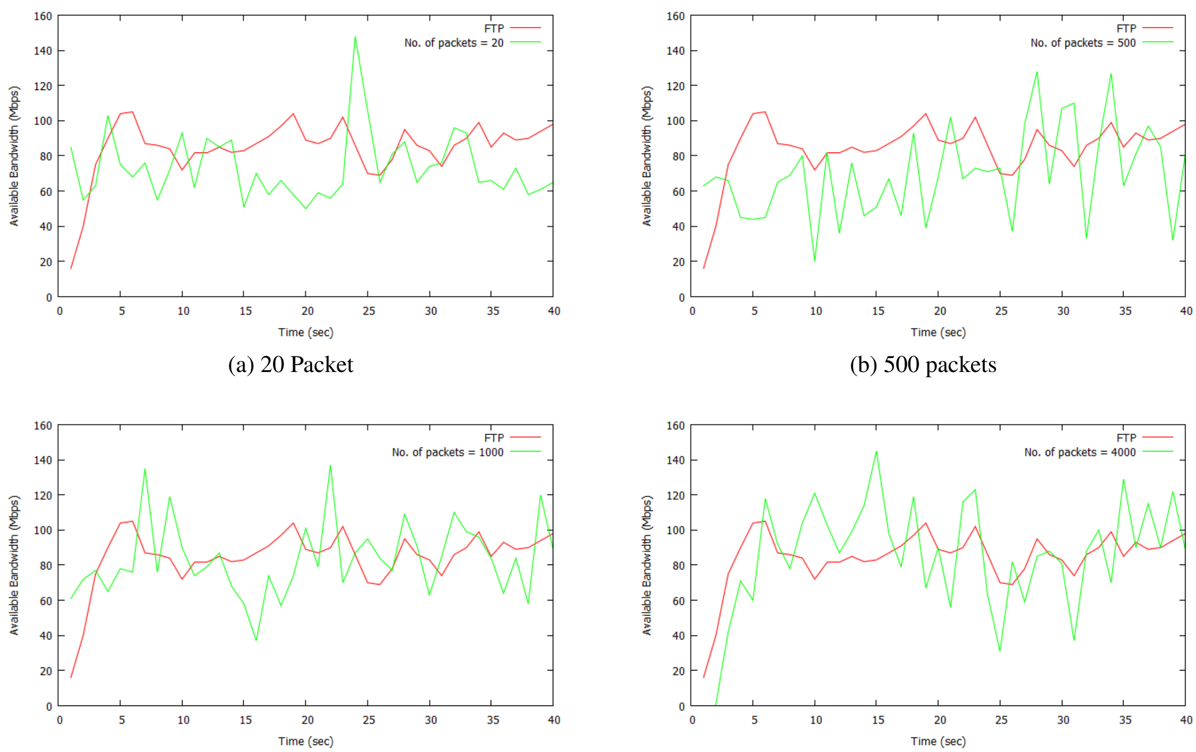


Figure 4: Blixt™'s measurements for different values of packet's size



(a) 20 Packet

(b) 500 packets

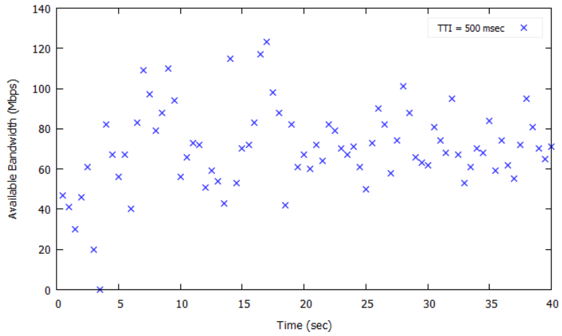
(c) 1,000 Packet

(d) 4,000 packets

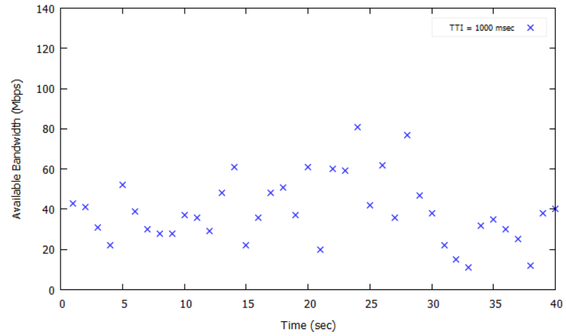
Figure 5: Blixt™ measurements versus the FTP throughput for different numbers of probing packets

to use the lowest number of probing packets that gives accurate estimations. Figure 5 depicts how the available bandwidth varies in accordance with the number of packets versus the FTP throughput.

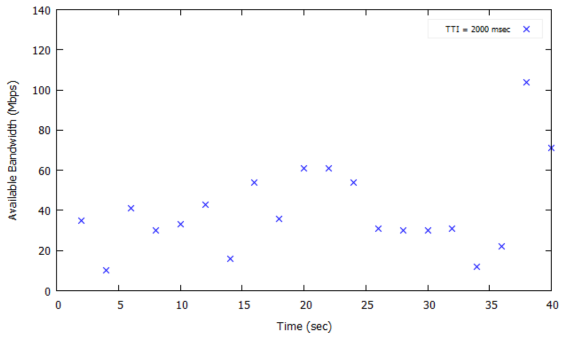
The cons of sending long packet trains are the overhead, the delay, and the increased risk of packet loss, in addition to traffic shaping in network nodes, especially in or close to the base station.



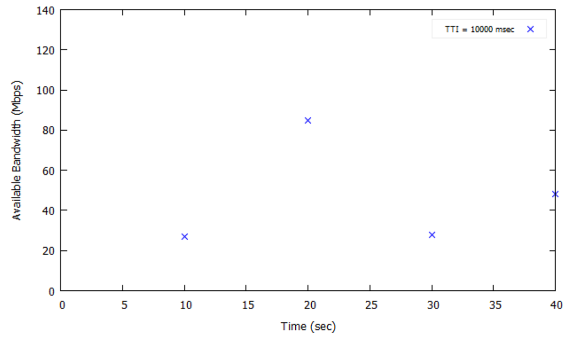
(a) 500 ms



(b) 1,000 ms



(c) 2,000 ms



(d) 10,000 ms

Figure 6: Blixt™ measurement for different values of TTI

4.1.3 Transmission Time Interval

TTI was the third studied parameter to catch a glimpse of how it affects the measurements' performance. Various values were chosen, ranging from 500 ms up to 10,000 ms, the packet size was 1,400 bytes and the number of packets was 190 packets throughout the entire experiment. Figure 6 delineates how the measurements vary as a result of modifying the TTI value.

The observation is that as the TTI increases, the points of measurements lessen, and the available bandwidth values dwindle as well due to the reduced probing traffic over the same period of time. Table 1 lists the values used for the experiments.

Table 1: The probing packet parameters' values used for the experiments

Packet size (Bytes)	Number of Packets	TTI (ms)
100	20	500
1,000	190	1,000
1,200	500	2,000
1,400	1,000	5,000
2,000	4,000	10,000

4.2 Performance Evaluation

The second set of experiments targets the performance evaluation of Blixt™ against the FTP test, iPerf 3, and nPerf. The benchmarking was limited to the widely used and commonly known tools since the details of how ABM tools are implemented and how the measurements are carried out are not broadly available. The most favorable values of the probing packet parameters that deliver better measurements were selected based on the first stage of the measurements which were later used in the evaluation experiments. The values selected were as follows: The packet size is 1,400 bytes; the number of packets is 190, and TTI is 1,000 ms.

IPerf [1] is an open-source tool and often used to monitor the network path performance and it can be considered as an unofficial standard in networking research groups. IPerf initiates TCP/UDP data packets to be sent between client and server hops with the aim of measuring the network's throughput. IPerf usability and performance are investigated in several research efforts; S. Abolfazli, Z. Sanaei et al. [40] have devised an approach to measure the bandwidth in 4G networks by using iPerf and OOKLA speed test. Another benchmarking tool was used for the comparison, which is nPerf.

nPerf [2] is a software (also works on Android) developed by nPerf SAS, a French-based company, and it aims to test the internet connection performance. In order to examine the performance of iPerf 3, magic iPerf was used which is an Android application that serves as an iPerf 3 client. IPerf 3 server was installed and ran over an Amazon EC2 instance, which is located in (eu-central-1, Frankfurt) region to be accessible over the internet with a public IP. On the other hand, the nPerf server is located in Stockholm. Figure 7 show's iPerf's experimental setup:

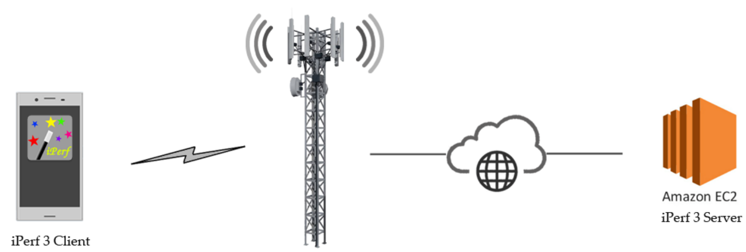


Figure 7: iPerf 3 measurements testbed

It was observed that Blixt™ delivers the most precise measurements compared to iPerf 3 and nPerf, and it has the lowest level of intrusiveness (about 10 times less than that of the FTP test) among the tools used for the experiments.

Figure 8 shows that Blixt™ closely follows the FTP throughput compared to iPerf 3 and even nPerf.

Figure 9 depicts the level of intrusiveness introduced by each tool compared to that of the FTP test.

It has to be noted that the intrusiveness level is relative to the intrusiveness of the FTP test which is presumed to be 100% as it capitalizes on the entire bandwidth. Moreover, these measurements scrutinize the carrier aggregation case, Blixt™ makes less than 5% of intrusiveness according to the tests conducted prior to the design by InfoVista and which was validated in a single carrier scenario.

5 Discussion and Challenges

The end-to-end measurement of the available bandwidth in 4G/LTE networks is a challenging task due to fluctuations in radio-channel conditions, modulation techniques, scheduling, pre-configured QoS pa-

[1] https://iperf.fr/
[2] https://nperf.com/

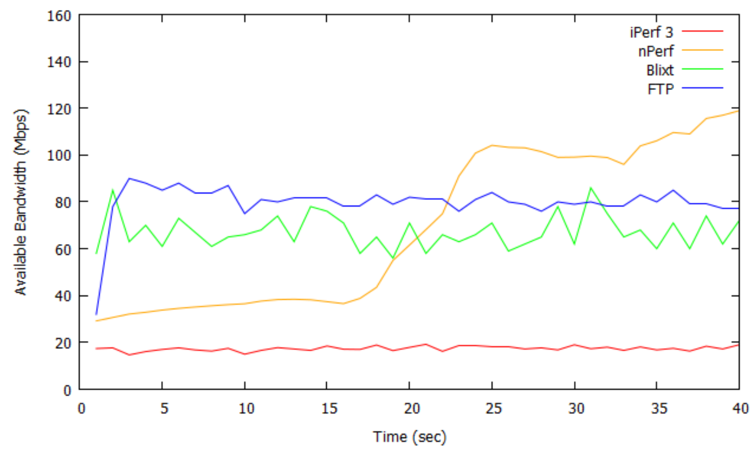


Figure 8: ABM of Blixt™ vs FTP, iPerf 3, and nPerf

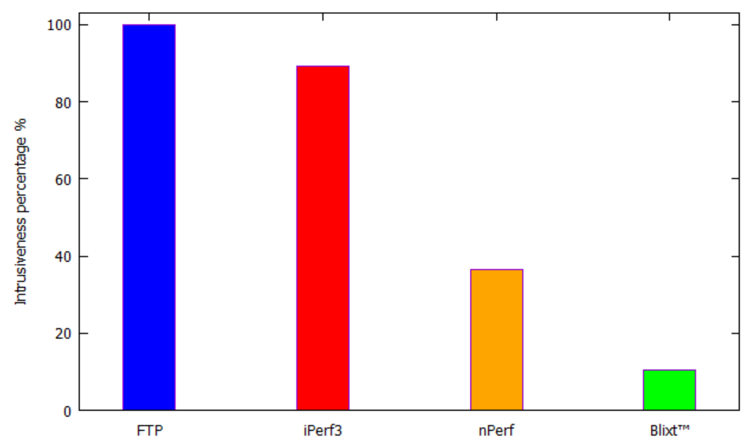


Figure 9: Intrusiveness level compared to that of the FTP test

rameters, hardware, and operating systems. Moreover, the challenge is how to precisely time-stamps the packet at the receiver node and handling the noise along the path, in addition to the difficulty in sending an accurate measurement of the inter-packet gaps. Another challenge to address is to lessen the level of intrusiveness and to restrain the measurements from impacting the QoE of subscribers since the measurements are conducted in a shared, live commercial networks. In the case of the data inserted by the tool into the network is rather high if compared to the available bandwidth at the time of the measurement, the tool or technique can be categorized as an intrusive tool. The probing traffic thus has to be restricted to avoid network disruption and measurement errors.

In cellular networks, ISPs modify the scheduler's configuration to control the transmission time and defer it until the conditions of the carrier channel become adequate. Additionally, the bandwidth in mobile networks is constrained, which is the major incentive for ISPs to use traffic-shaping mechanisms for bandwidth-hungry applications like video streaming to enhance the utilizable bandwidth for end users. In order to act quickly on network changes, it becomes a necessity to estimate available bandwidth expeditiously and at the earliest possible moment. Additionally, the measurement duration depends on the number of samples issued per iteration. The fewer samples are sent, the faster the measurement can be completed.

As opposed to wired networks, the throughput of mobile networks is hardly equal to the data rate due to various reasons which were previously mentioned. Wireless networks have several properties, namely inconstant packet error rate, transmission retries, contention, and wireless link rate adaptation. Furthermore, the wireless medium is also a shared channel which implies that the available bandwidth varies depending on the number of users vying for the resources offered by the channel which makes it more challenging to estimate the bandwidth for mobile networks. To exacerbate the issue, ISPs use buffer configurations and different admission control policies following their business model or operational strategy. It is common that the operational LTE networks use a proportional fair scheduler which considers the variations of radio quality and number of UEs, nevertheless, it stays as one of the main grounds for the measurements' precision deterioration as the behavior of the packet scheduler at the LTE link-layer varies dynamically which may result in severe disruption for the queuing delays at a receiver.

It is infeasible to compute the cross-traffic effect on the available bandwidth since there is no simple formula to do so, as it largely relies on the fluctuating wireless channel conditions, packet size, scheduling and modulation techniques, pre-configured QoS metrics, etc. The challenges also involve the interference along with the resource constraints (data and power) on UEs. Real-time nature aggravates the issue which makes it crucial to balance the precision of the measurements, the amount of traffic used to probe the network, and the time it takes to deliver an estimation.

In general, the ABM tools face three key challenges in the case of high-speed networks, which are finding a subtle inter-packet gap at the sender, precisely timestamping packet arrival at the receiver, and noise handling [39]. In practice, most of ABM tools have been showing inferior performance in the case of real environment assessments in contrast to their performance in simulation testbeds. The fine-scale buffering between the sender and receiver in a shared resource system creates noise in the gap separating the packets, which is one of the factors that can influence the analysis of one-way delays or inter-packet gaps at the receiver for such bandwidth estimation. It has a greater impact on high-speed networks as gaps and delays are much finer in scale. Moreover, the available bandwidth estimation techniques need to handle the noisy data that comes from measurements which, for example, is attributable to imprecise time-stamping.

Randomness could also be considered as one of the issues. A practical solution to cope with this issue is to repeat the measurements multiple times and average the results, which was followed in pathChirp and Spruce, or to decide according to the most repetitive, as it was done in Pathload. This method was also used in BART by performing repetitive measurements to observe changes in the available bandwidth. Besides the three (and the randomness) challenges, it also faces two major challenges in cellular networks which are:

1. Handover procedures and admission control policies: are governed by the ISPs and responsible for load balancing and bandwidth reservation. It highly depends on the instantaneous number of users.

2. Mobility: the user's mobility causes tremendous signal fluctuation, which, in turn, results in bandwidth degradation, while it is comparatively consistent if the mobile device does not move or moves at regular speed.

In summary, the bandwidth in cellular networks is observably variable if compared with Wi-Fi and wired networks. This is the reason why the majority of bandwidth estimation tools flunk in mobile networks, thereby an accurate bandwidth estimation is commonly achieved at the cost of high data charges and delays which may result in becoming invalid quickly.

According to the results, Blixt™ measurements follow a similar paradigm to that of FTP throughput in most of the values [39]. In very few points, it was spotted slightly lower values considering that the experiments were not conducted concurrently as using two devices would not help since it measures the available bandwidth, which means the residual bandwidth. Also, the eNodeB propends to select a lower bit rate when the mobile experiences poor radio conditions to preclude a large number of bit errors.

A lightweight packet train is used to obtain quick and precise estimation. At a train length of 130-190 packets, the estimations become stable and accurate enough to be of value to the application, the longer the packet train length is, the more reliable the measurements are.

The active measurement techniques require strong time-related constraints, which are:

1. Precise timing in sending the probing packets which have to match the inter-departure constraints.

2. Arrived probing packets have to be time-stamped precisely.

3. Accurate synchronization of the sender and the receiver in order to compare the clock between the two endpoints.

Studying the feasibility of sustainability is one of the most important aspects to consider. To address the environmental impact, the carbon emissions can be reduced by measuring three metrics (Available Bandwidth for both, the uplink and the downlink, in addition to the Round-Trip Time RTT) simultaneously in one drive-test using Blixt™. In addition, it reduces energy consumption in both cells and networks by reducing the process of data compared to other tools that adopt other protocols such as File Transfer Protocol (FTP).

On the other hand, it reduces the cost required for benchmarking competitors. Additionally, minimizing the probing data sent to the network which utilizes the shared resources network, i.e. the cellular network, contributes to improving the QoE for the subscribers as it has less impact in terms of the intrusiveness compared to other techniques.

6 Conclusion

In cellular networks, the available bandwidth has an impact on both, the QoS and the QoE, in most of the mobile services as it actively involves improving QoS in the network in addition to a variety of network applications and technologies. As is known, signal strength is not the only aspect to consider, as there are more aspects (for example, admission control policies and signal fading) affect, in one way or another, the available bandwidth, which makes it a necessity to have a bandwidth measurement application on mobile phones to present the measured bandwidth instantaneously.

The precision of the ABM technique is determined by the packet size, the number of packets per train, the gap that separates two packet trains, as well as by the real-time data rate, that is determined to correspond with the maximum bandwidth following present system information parameters, UE capabilities, and additional settings. Measurements precision correlates with the number of probe samples which have been sent into the data channel per iteration. If smaller packages are used in each train or smaller packet size, this reduces the time the tool takes for estimation, but at the expense of less accuracy. The more samples are inserted into the network, the greater the accuracy of the measurement is obtained from that particular technique, nevertheless, injecting probe packets into the network could also lead to quality of service degradation or cause network congestion. For these reasons, it is crucial to find optimal values of the number of probe samples per iteration that will be large enough for providing an adequate bandwidth measurement accuracy and small enough to not deplete network resources. Blixt™ contributes to solving the issue by conducting the required testing standard (obtaining enough data) while keeping measurement intrusiveness to an absolute minimum. Blixt™ generates short traffic bursts which last for a quite short time, with comparatively considerable empty gaps between the packet trains. Thus, the average probing packets rate represents a tiny part of the total available bandwidth.

The major challenge in active measurement approaches is the amount of probing data to be sent over the network should be kept at a minimum and still attaining precise estimations while keeping the level of intrusiveness low. In fact, it cannot be said that the tool is intrusive if it only sends a small amount of UDP bursts to measure RTT in case the bandwidth available is relatively large. For instance, the tool can be considered intrusive if it consumes more than 10% of the available bandwidth in a shared resources environment.

For future work, the use of machine learning algorithms to forecast the available bandwidth can be considered based on the data collected from previous measurements, thus minimizing the interference with the network. This approach requires training the algorithm with a large set of data to attain better accuracy, and the data should be good enough and representative.

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Author Biography

Al-Hussein Hameed Jasim received the B.Sc. degree in electronics and communications engineering from the University of Baghdad, Iraq. He received three M.Sc. degrees in complex systems engineering, technology, computer science and engineering from the University of Lorraine, France, Lappeenranta University of Technology, Finland, and Luleå University of Technology, Sweden respectively. He is currently working for Tetra Pak as a DevOps engineer. His research interests include Network Monitoring, Cloud Computing, and Green ICT.

Niclas Ögren has a B.Sc. in Electrical Engineering emphasized on Digital Signal Processing from Blekinge Institute of Technology, Karlskrona, Sweden. He has since his degree in 1995 worked with measurement solutions for radio access of mobile telecommunication systems from 2G to 5G, and today works in the role of specialist of Network and Protocols. His research interests include radio access technology, radio measurement, QoS, and QoE.

Dimitar Minovski has a joint M.Sc. degree in Pervasive Computing and Communication from University of Lorraine, Lappeenranta University of Technology and Luleå University of Technology. Currently, he is a PhD candidate at Luleå University of Technology, with research interests in areas such as Quality of Experience, Internet of Things, and Mobile Networks.

Karl Andersson has a M.Sc. degree in Computer Science and Technology from Royal Institute of Technology, Stockholm, Sweden and a Ph.D. degree in Mobile Systems from Luleå University of Technology, Sweden. After pursuing postdoctoral research at the Internet Real-time Laboratory at Columbia University, New York, USA and National Institute of Information and Communications Technology, Tokyo, Japan, he is now Associate Professor of Pervasive and Mobile Computing at Luleå University of Technology, Sweden. His research interests include Green and Mobile Computing, the Internet of Things, Cloud Technologies, and Information Security.