# Characterization of the Buffers in Real Internet Paths

Luis Sequeira, Julián Fernández-Navajas, Jose Saldana
Communications Technology Group (GTC)-Aragón Inst. of Engeneering Research (I3A)
Dpt. IEC. Ada Byron Building. EINA Univ. Zaragoza
50018 Zaragoza, Spain
Email: {sequeira, navajas, jsaldana}@unizar.es

***Abstract*—The behaviour of the routers' buffer is of primary importance when studying network traffic, since it may modify some of this characteristics, as delay or jitter, and may also drop packets affecting the Quality of Service (QoS) of different services. As a consequence, the characterization of this buffer is interesting, especially when real-time flows are being transmitted.**

**This work presents a preliminary study of how to determine the technical and functional characteristics of buffers (as e.g., behaviour, size, limits, input and output rate) of a network path. Two different methodologies are considered on two test scenarios; real measurements permit the estimation of some parameters of the intermediate buffers as size, input and output rates, in a network path including different devices across the Internet.**

***Index Terms*—Buffer size, queueing, unattended measurements.**

## I. Introduction

Multimedia servicesgenerate a significant amount of network traffic over the Internet, since the number of users grows every day. Moreover, the expectation of future growth for the use of multimedia applications (e.g., videoconferencing and VoIP), indicates that this tendency will increase. On the one hand, the user demands a good experience with multimedia services and on the other hand, the heterogeneous characteristics of the different Internet accesses technologies, makes it necessary to define the Quality of Service (QoS) that they offer, especially when the access networks have to support to real-time applications.

This kind of services may have an important impact on network resources depending on to the nature of the information transmitted and its size. While some services inject traffic with a constant bit rate in order to provide a certain QoS level and a better user's experience, other applications generate bursty traffic, with a different number of frames into each burst. And regarding packet size, while some real-time applications as e.g., VoIP generate small packets (in the order of a few tens of bytes), others use large packets, as e.g., videoconferencing.

At the same time, some network points become critical bottlenecks, mainly in access networks, because these networks' capabilities are lower than the ones available in the backbone; in addition, bottlenecks may also appear at critical points of high-performance networks, being the main cause of packet loss the discarding of packets in router queues. So the design characteristics of router buffers and the implemented scheduling policies, are of primary importance in order to ensure the correct delivery of the traffic of different applications and services.

Buffers are used as a traffic regulation mechanism in network devices. Mid and low-end routers, which do not implement advanced traffic management mechanisms, are usually used in access networks. Thus, buffer size becomes an important design parameter. The buffer can be measured in different ways: maximum number of packets, amount of bytes, or even queueing time limit [1] [2]. Moreover, the buffer has an important role in network planning because it can influence the packet loss on different services and applications and therefore QoS can be affected by the buffer behaviour, size and scheduling policies.

Hence, characterization of the technical and functional parameters of this device becomes critical when trying to provide certain levels of QoS. This knowledge can be useful for applications and services in order to make correct decisions in the way the traffic is generated. As a consequence, if the size of the buffer and its behaviour are known, some techniques can be used so as to improve link utilization, e.g., multiplexing a number of small packets into a big one or fragmentation. However, a problem appears when using these techniques: manufacturers do not include all the implementation details in the technical specifications of the devices, but just part of them, mainly related to the technology used. On the other hand, if a communication has to cross different networks over the Internet, some knowledge about the device's characteristics or the buffer's behaviour will be interesting. For these situations, our group is currently working on the development of a tool able to discover some characteristics of the buffer and its behaviour. The objective is to permit measurements not only when physical access to the "System Under Test" is granted but also in the case of only having remote access.

The paper is organized as follows: section II discusses the related work. The test methodology is presented in section III. The next section covers the experimental results, and the paper ends with the conclusions.

## II. Related work

### A. Buffer size

The fact of having different rates at the input and output link of a router may produce bottlenecks in the network, so packet loss may occur. Buffers are used to reduce packet loss

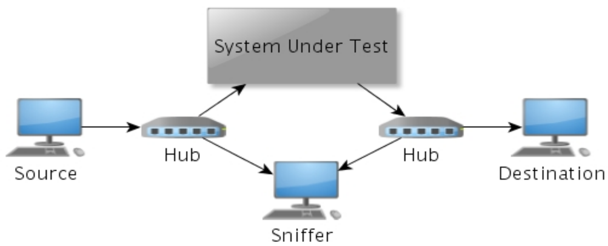


Fig. 1. Topology used for test.

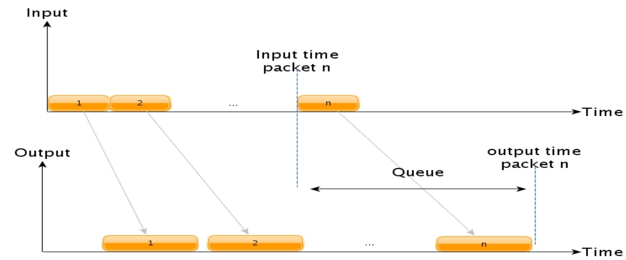


Fig. 2. Estimating packets in queue.

by absorbing transient bursts of traffic when routers cannot forward them at that moment. They are instrumental in keeping output links fully utilised during congestion times.

For many years, researchers accepted the so-called *rule of thumb* to obtain the amount of buffering needed at a router's output interface. This rule was proposed in 1994 [3] and it is given by $B = C \times RTT$, where $B$ is the buffer size, $RTT$ is the average round-trip time and $C$ the capacity of the router's network interface. This *rule of thumb* is also called the Bandwidth Delay Product (BDP). In [4] it was proposed a reduced buffer size by dividing BDP by the square root of the number of TCP flows $B = C \times RTT/\sqrt{N}$. This model was called *small buffer*. In [5] it was suggested the use of even smaller buffers, called *tiny buffers*, considering a size of some tens of packets. However, the use of this model presents a trade-off: reducing buffers to only a few dozen of $KBytes$ can produce a $10\% - 20\%$ drop probability.

It has also been observed in the literature that the buffer size can be measured in different ways: e.g., in [6] the routers of two manufacturers are compared, and one gives the information in packets, whereas the other one measures it in milliseconds.

However, this design characteristic is important when planning a network. The reason for this is that there is a relationship between router buffer size and link utilization, since an excessive amount of memory would generate a significant latency increment when the buffer is full. On the other hand, a very small amount of memory in the buffer will increase packet loss in congestion time. As a consequence, the knowledge of the buffer behaviour is an interesting parameter which can be considered when trying to improve link utilization.

### *B. Impact of the buffer in multimedia services*

Many scientific publications related to the study of the influence of the buffer on different services and applications show how QoS is affected by the buffer behaviour, which is mainly determined by its size and management policies. The influence of the buffer on VoIP was studied in [7], where three different router buffer policies were tested, also using two multiplexing schemes. It was observed that the policies implemented by the router buffer may cause different packet loss behaviour, and also modify voice quality, measured by means of R-factor. In the same paper two multiplexing methods for VoIP flows were studied, were bandwidth with the counterpart of increasing packet size, which has an influence on packet loss, depending on the implementation and buffer size of the router. In this case the VoIP native traffic showed a good behaviour when using a small buffer measured in bytes, as small packets have less probability of being discarded than big ones.

In [8] the authors presented a simulation study of the influence of a multiplexing method on the parameters that define the subjective quality of online games, mainly delay, jitter and packet loss. The results show that small buffers present better characteristics for maintaining delay and jitter in adequate levels, at the cost of increasing packet loss. In addition, buffers whose size is measured in packets also increase packets loss.

Many access network devices are designed for bulk data transfers [9], such as e-mail, web or FTP services. However, other applications (e.g., P2P video streaming, online games, etc.) generate a high rate of small packets, so the routers may experience problems to manage this traffic, since, their processing capacity can become a bottleneck if they have to manage too many packets per second [10]. The generation of hight rates of small packets [11] may penalize the video packets and consequently peer's behaviour within a P2P structure may not be as expected.

## III. Test methodology

### *A. Test procedure*

The scheme of the tests is shown in Fig. 1. There is a "System Under Test" (SUT from now), which may be either a device or a network. Traffic is sent from a source, and arrives to the destination traversing the SUT. Two hubs and a sniffer are used in order to capture the traffic at the input and at the output of the SUT. The test is based on the sending of a burst of UDP packets from the source to the destination machine, so as to produce a buffer overflow in the SUT. This test is repeated using different amounts of bandwidth. Packets of different sizes are used so as to determine if the buffer is measured in number of packets or in bytes.

### *B. Test methodology for a single buffer*

We will use two methodologies to estimate the characteristics of the buffers of the network traversed: of size, limits and output rate, using the methodology detailed in [12]. The methods are based on the premise that output rate can be obtained from *destination* capture. Output rate depends on the technology used in each case (Ehternet, WiFi).

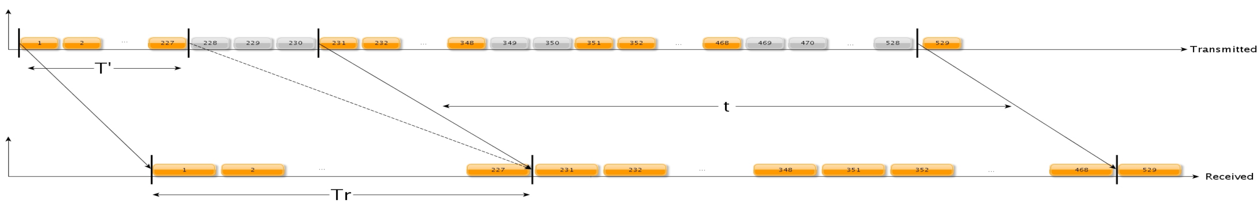


Fig. 3. Estimating packets in queue in remote access (general case).

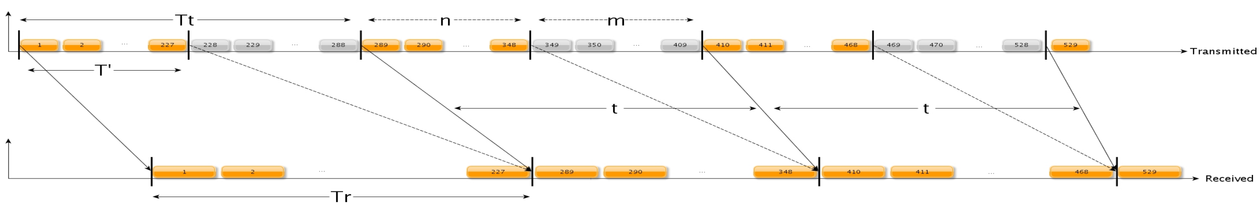


Fig. 4. Estimating packets in queue in remote access (particular case).

- **Method 1:** Counting the number of packets in the queue in the moment that a packet arrives at the buffer.
- **Method 2:** If the delay of a packet in the buffer can be determined, then the variations of this delay can give us useful information for estimating buffer size.

The first option brings a more accurate estimation, but it requires physical access to the SUT. The second option can also be used when there is not direct access to the system.

*1) Finding buffer size with physical access:* The method used in this case is shown in Fig. 2. All the transmitted packets are identified by a sequence number included in the payload, so the size of the buffer is estimated by the number of packets in the queue between the arrival and departure time.

When physical access to the SUT is guaranteed, a *sniffer* captures traffic at the ends of the device. Two captures are obtained an stored in files, and are processed with a *shell script* to calculate packet delay, packet loss, interarrival packet time, input and output buffer rate and filling buffer rate, and buffer size according to [12].

*2) Finding buffer size with remote access:* A good estimation can also be obtained even if there is no physical access. Fig. 3 shows the relationship between sent and received packet times. $T_r$ is the sum of two times: the delay for completely filling the buffer until a packet loss occurs; plus the time the last accepted packet needs for traversing the buffer. As a consequence, this time will be noticed at the out-capture when the first packet is missing. So,

$$T_r = T_{fill} + T_{empty} \tag{1}$$

Let $R_{in}$ and $R_{out}$ be the input and output rates of the buffer respectivelly. We define $R_{fill}$ as the rate in which the buffer fills when $R_{in}$ is bigger than $R_{out}$ ($R_{fill} = R_{in} - R_{out}$). $L_{buffer}$ is the size of the buffer in bytes. A packet spends $L_{buffer}/rate$ to cross the full buffer, so we can obtain $T_r$ as,

$$T_r = \frac{L_{buffer}}{R_{fill}} + \frac{L_{buffer}}{R_{out}} \tag{2}$$

therefore,

$$L_{buffer} = \frac{T_r}{\frac{1}{R_{in}-R_{out}} + \frac{1}{R_{out}}} \tag{3}$$

The output rate can be easily determined, because the remote capture includes the $n$ received packet in $t$ seconds and packet length is known. For calculating the input rate, we know that the amount of transmitted packets $n+m$ (received and dropped packets respectively) in $t$ seconds. Where $m$ can be known since all the packets have a unique identifier. With this information, output and input rates can be estimated only from the data contained in the *destination* capture, using the following expressions:

$$R_{out} = \frac{n_{tx}}{t} \times packet_{size} \tag{4}$$

$$R_{in} = \frac{n_{tx} + m_{tx}}{t} \times packet_{size} \tag{5}$$

As we have commented above, to determinate buffer size it is necessary to obtain an accurate value of $t$. The way we select the value of this variable has to be determined by the buffer type. Traditional FIFO queues allow entering a packet when there is enough space, so $t$ must have a value that permits to determinate an accurate input rate, allowing an amount of packets as high as possible.

In [12] a particular buffer behaviour was observed and characterized: when the buffer is completely full, no more packets are accepted until a certain amount of memory is available. Thus an *upper limit* and *lower limit* can be defined. For these cases, $t$ is the time that the buffer size requires for changing from the *upper limit* to *lower limit*. As is shown in Fig. 4, $t$ can be exactly measured and it will have the same value in both extremes. In this case, when *destination* receives $n$ consecutive packets, the *source* has sent $n+m$ packets (where $m$ is the number of dropped packets).

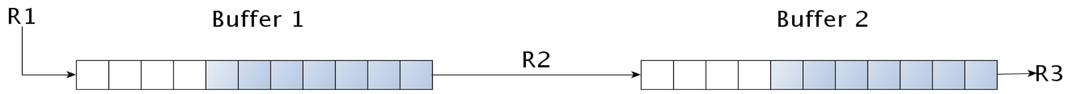


Fig. 5. Estimating packets in queue in remote access (particular case).

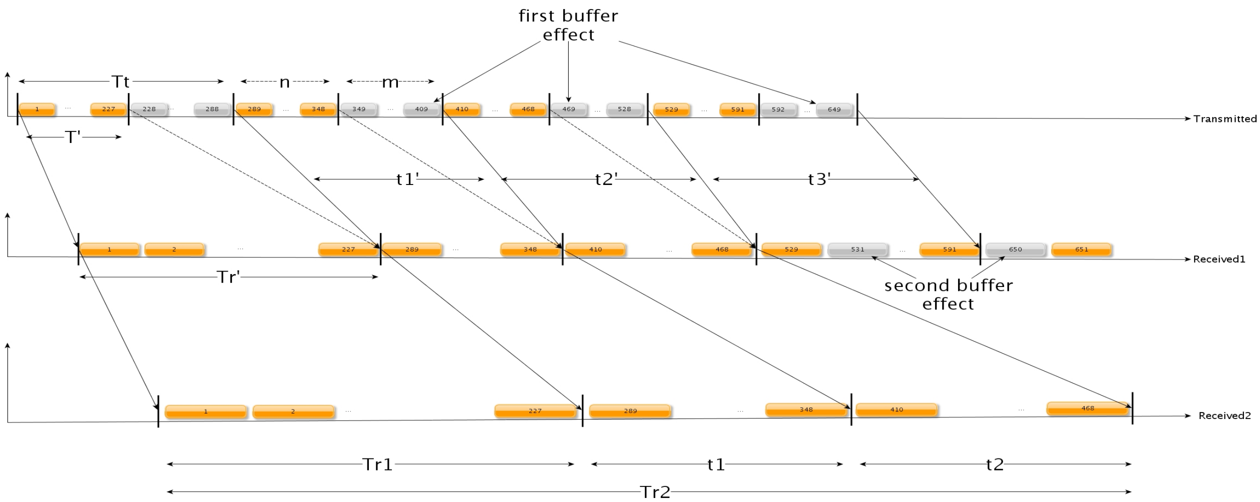


Fig. 6. Estimating packets in concatenate queues with remote access.

### C. Test methodology for concatenated buffers

The methodology is based on the premise that $R_{in} > R_{out}$, so $R_{fill} > 0$. Fig. 5 shows two concatenated buffers, which will fill when $R_1 > R_2 > R_3$. In the case that one buffer has $R_{in}$ lower than $R_{out}$ all packets can be transmitted without storing them and this situation leaves without effect the buffer behaviour in the network path.

Fig. 6 is used in order to explain the methodology when two buffers are in the same path. In this figure, the *Transmitted* trace is at the input of *Buffer* 1, *Received* 1 is the output trace of *Buffer* 1, *Received* 2 is the output trace of *Buffer* 2, and it is the only trace we have available in order to determinate all the link characteristics. *Buffer* 1 is a device with an *upper limit* and a *lower limit*, as described in [12]; *Buffer* 2 using the traditional FIFO policy. Gray packets are the ones dropped.

If $R_1 > R_2 > R_3$, both buffers are filling and they will drop packets in certain moments. When *Buffer* 1 gets into overflow, it drops packets until a certain amount of memory is available, so it will discard a burst of packets. The *Buffer* 2 has a different behaviour on congestion time because if a packet gets out, another can get into the buffer thus packets will not be discarded in bursts. With the remote capture we can obtain $R_3$ as follows:

$$R_3 = \frac{n_{rx}}{t_{r2}} \times packet_{size} \tag{6}$$

We also know which packets are lost and we can determinate a proper time $t$, as we commented above, so $R_2$ is

$$R_2 = \frac{n_{rx} + m_{tx}}{t_n} \times packet_{size} \tag{7}$$

finally, following the same logic we find $R_1$

$$R_1 = \frac{n_{rx} + m_{tx}}{t'_n} \times packet_{size} \tag{8}$$

and so, buffer size:

$$L_{Buffer1} = \frac{T'_r}{\frac{1}{R_1 - R_2} + \frac{1}{R_2}} \tag{9}$$

$$L_{Buffer2} = \frac{T_{r1}}{\frac{1}{R_2 - R_3} + \frac{1}{R_3}} \tag{10}$$

## IV. Preliminary experimental results

In order to illustrate the proposed methodology, real tests have been deployed in a testbed and results are analysed according to the procedures cited above. Real machines have been used (Linux kernel $2.6.38-7$, Atheros $AR9287$ wireless network adapter, $Intel^{®}$ $Core^{TM}$ $i3$ $CPU$ $2.4$ $GHz$), in order to identify the buffer behaviour of different devices. In addition, simulations have been used to analyze FIFO scenarios.

### A. Real scenarios

We have studied two different scenarios, in both cases a host sends traffic to a destination through a network or a device. A sniffer which do not degrade monitoring performance is included at the best location for making captures [13]. Different bandwidths limits were set in the hubs in order to create a bottleneck which has to be measured.

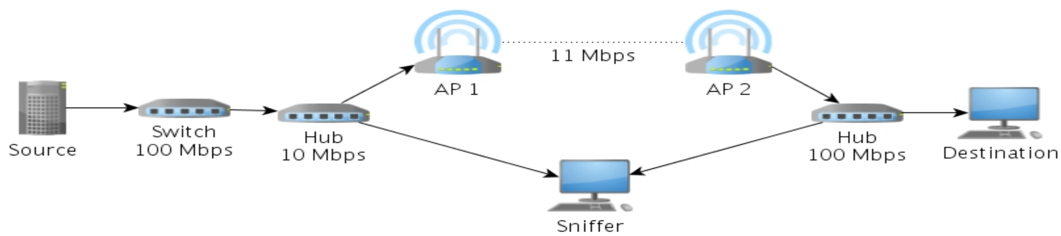


Fig. 7. First scenario: Estimating buffer size in a wireless network.

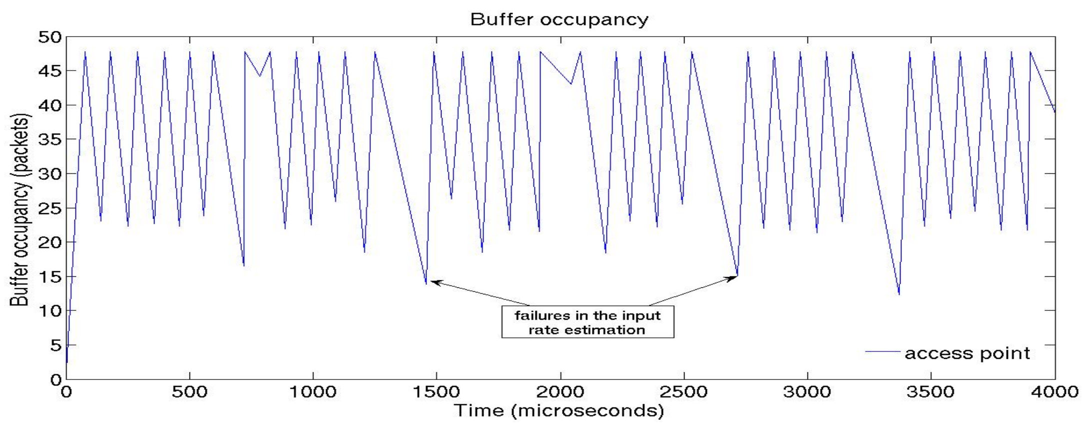


Fig. 8. Estimating the buffer size of an access point .

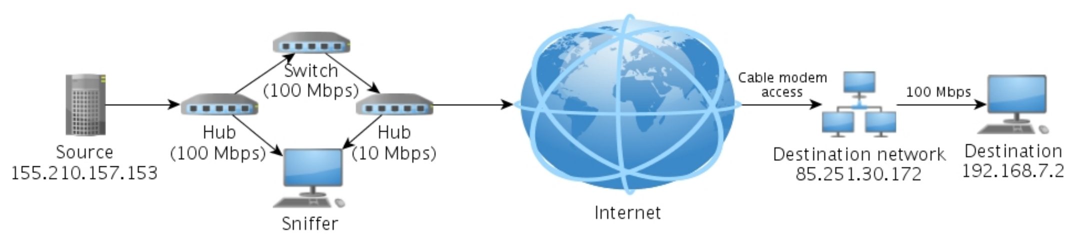


Fig. 9. Second scenario: Estimating buffer size in a wired network.

*1) Laboratory environment:* In the first case, a network path across one swhitch (3COM) and two access points (Linksys WAP54G) has been analized, trying to estimate the buffer size. Fig. 7 shows the topology used: both hosts are connected to the hubs using 100 $Mbps$ and 10 $Mbps$ links. UDP flows are sent with the aim of obtaining the buffer size of the $Switch$ and $AP1$.

Fig. 8 shows the results. In a previous work [12], the buffer size of the studied device was obtained, which fits with the shown results: we can observe a *lower limit* of 25 packets, and an *upper limit* of roughly 50. Thus, we see that the proposed method is able to determine the buffer size of AP1.

However, we also observe some moments in which the lower limit is underestimated. The cause of this is the innacuracy of the estimation of the rate in the ingress of the access point. In order to obtain a better estimation we would need to deploy an exhaustive analysis of the the relationship between packet loss and bandwidth in the received trace.

*2) Real network path across the Internet:* For the second scenario, the topology shown in Fig. 9 has been used. In this case, a typical home network is accessed by other host from a different network across the Internet. Using the remote capture buffer size, the concatenation of different buffers across the internet can be estimated.

In the Fig. 10 can be observed that during the first 2000 $\mu s$, the buffer of the switch gets full, but, the second one is not losing packets yet. So we can observe that the size of the switch's buffer is 120 packets.

However, in $t = 2000$ $\mu s$, the second buffer gets also full, so the effect of the two buffers overlap, so we are not able to estimate their sizes any more, since we cannot obtain an accurate estimation of the input rate of the second buffer.

### B. Simulation scenario

For the simulated tests we have implemented a NS-2 scenario which is shown in Fig. 11. In this case, we confirmed

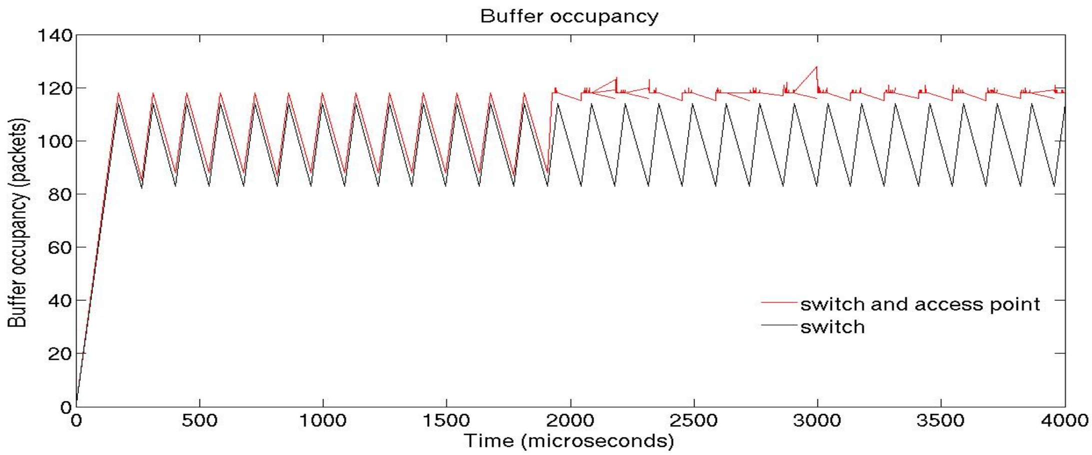


Fig. 10. Buffer size.

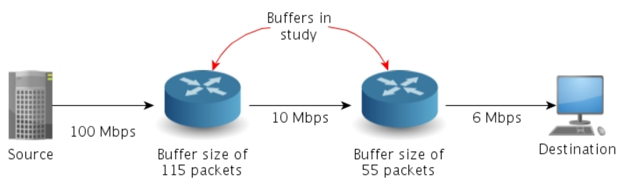


Fig. 11. Simulation scenario: Analysing queue behaviour.

the estimations for FIFO queues.

## V. Conclusion

This article has presented two methods for analyzing the technical and functional characteristics of commercial buffers on a determinate network path. This characterization is important, taking into account that the buffer may modify the traffic characteristics.

Tests using commercial devices have been deployed in two different scenarios, using wired and wireless networks. Accurate results of the buffer size can be obtained when there is physical access to the "System Under Test". In case of having no direct access to the system, an acceptable estimation can also be obtained. As future work, the method has to be improved in order to improve the accuracy, especially when measuring the input rate.

## Acknowledgment

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