Characterization of Real Internet Paths by Means of Packet Loss Analysis

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Abstract—**The behaviour of the routers' buffer may affect the Quality of Service (QoS) of network services under certain conditions, since it may modify some traffic characteristics, as delay or jitter, and may also drop packets. As a consequence, the characterization of the buffer is interesting, especially when multimedia flows are transmitted and even more if they transport information with real-time requirements. This work presents a packet loss analysis with the aim of determining the technical and functional characteristics of the real buffers (as, e.g., behaviour, size, limits, input and output rate) of a network path. An improved methodology is considered in which two different buffers are concatenated. It permits the estimation of some parameters of the intermediate buffers (size, input and output rate) in a network path including different devices across the Internet. The method presented in this paper permits the characterization of commercial router buffer by means of the analysis of the dropped packets in the buffer.**

Keywords-Buffer size; queueing; unattended measurements.

I. Introduction

The number of users of multimedia services (e.g., videoconferencing and Voice over IP, VoIP) grows every day and they generate an increasingly significant amount of traffic over the Internet. At the same time, users demand a good experience with these services. In this context, Internet Service Providers (ISP's) have to grant a high performance network with a certain degree of Quality of Service (QoS), especially when the access networks have to support to real-time applications.

Traditionally, the available bandwidth between two end-to-end devices has been used as a parameter that can give a rough idea of the expected quality. But nowadays, we know that QoS is also affected by the behaviour of the intermediate buffers, which is mainly determined by their size and their management policies. As it was observed in [1], the policies implemented by the router buffer may cause different packet loss behaviour, and may also modify the quality of the service (VoIP in that case, measured in terms of R-factor, Transmission Rating Factor). The influence of the router buffer on another real-time service (i.e., an online game) was studied in [2], showing the mutual relationship between the size and policies of the buffer, and the obtained subjective quality, which mainly depends on delay and jitter in this case. 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 big packets, typical of services requiring bulk data transfers [3], such as e-mail, web browsing or File Transfer Protocol (FTP). However, other applications (e.g., P2P (Peer to Peer) video streaming, online games, etc.) generate high rates 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 [4]. Finally, in P2P-TV services, the generation of high rates of small packets [5] may penalize the video packets and consequently the peer's behaviour within a P2P structure may not be as expected.

As a consequence of the increase of the amount of small packets generated by emerging services, certain network points may become critical bottlenecks, mainly in access networks. In addition, bottlenecks may also appear at critical points of high-performance networks, being the discarding in router queues the main cause of packet loss. 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 commonly used in access networks. Thus, the 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 [6] [7]. Moreover, the buffer must play an important role when planning a network because it can influence the packet loss of different services and applications. Therefore, the QoS of the services can be affected by the size of the buffer and its scheduling policies.

Hence, the 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 and the behaviour of the buffer 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, fragmentation, etc. However, a problem appears when using these techniques: device manufacturers do not include all the implementation details in the technical specifications of the devices, but just part of them, mainly those related to the technology used. Thus, if a communication has to cross different networks over the Internet, some knowledge about the device's characteristics

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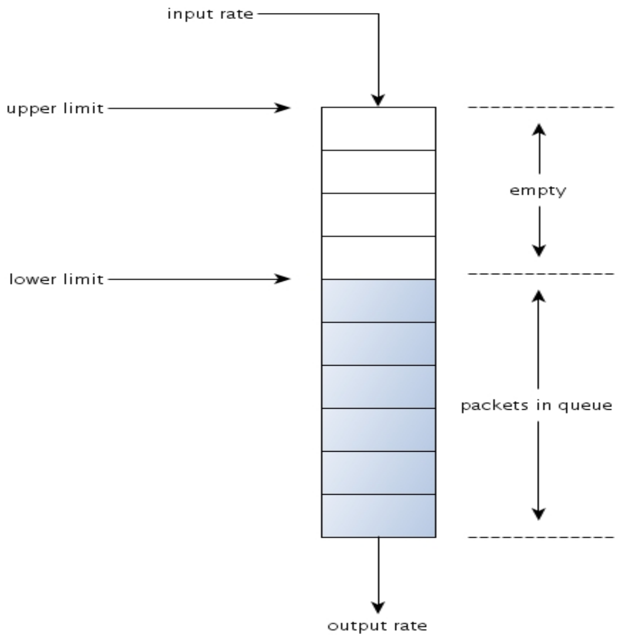

Figure 1. A particular buffer behaviour.

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 final objective is to permit these measurements not only when physical access to the System Under Test (SUT) 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 issues

Buffers are used to reduce packet loss by absorbing transient bursts of traffic when routers cannot forward them at that moment. They are instrumental in keeping output links fully utilized during congestion times.

The so-called *rule of thumb* has been used to obtain the amount of buffering needed at a router's output interface [8] but in [9], a *small buffer* model was proposed, in which buffer size is obtained by the capacity, C, round-trip time, RTT and the number of flows, N, so, $B = C \times RTT/\sqrt{N}$. In [10], it was suggested the use of even smaller buffers, called *tiny buffers*, considering a size of some tens of packets.

Traditional First In First Out (FIFO) queues accept a new packet when there is enough space. However, this is not the only buffer behaviour in commercial devices. In [11], a particular buffer behaviour was observed and characterized: once the buffer gets completely full, no more packets are accepted until a certain amount of memory is available. Thus, an *upper limit* and a *lower limit* can be defined (see Fig. 1): when the *upper limit* is reached, no more packets are accepted until the size of the buffer corresponds to the *lower limit*.

Figure 2. A typical topology in end-to-end communication.

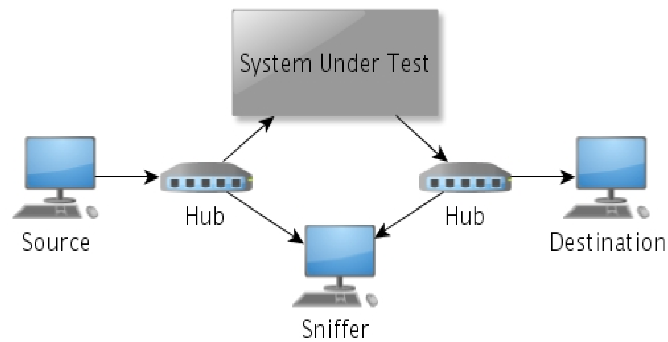

Figure 3. Topology used for tests.

III. Improved methodology for the characterization of Internet paths

When traffic is crossing a network in an end-to-end communication (Fig. 2), all the packets traverse a high number of network devices which may drop packets in some cases. The network path is uncertain for most applications and services which usually only measure the available bandwidth, in order to limit the generated traffic and rarely to smooth it. So, we proposed a method to discover and describe network characteristics which may be useful to correctly modify the traffic by applications. One of the premise of the present work is that most of the network characteristics can be explained by buffers characteristics.

In a previous work [12], we described a methodology to determine the technical and functional characteristics of buffers (as e.g., behaviour, size, limits, input and output rate) of a network path even when more that one buffer is in the path, finding interesting results but with some inaccuracies when obtaining input rate estimations and buffer size. An improvement of this methodology is presented in this article.

A. Test procedure

The scheme of the tests is shown in Fig. 3. There is a System Under Test (SUT), which may be either a single device or an entire network. The test is based on the sending of a burst of UDP (User Datagram Protocol) packets from the Source to the Destination machine, so as to produce a buffer overflow in the SUT.

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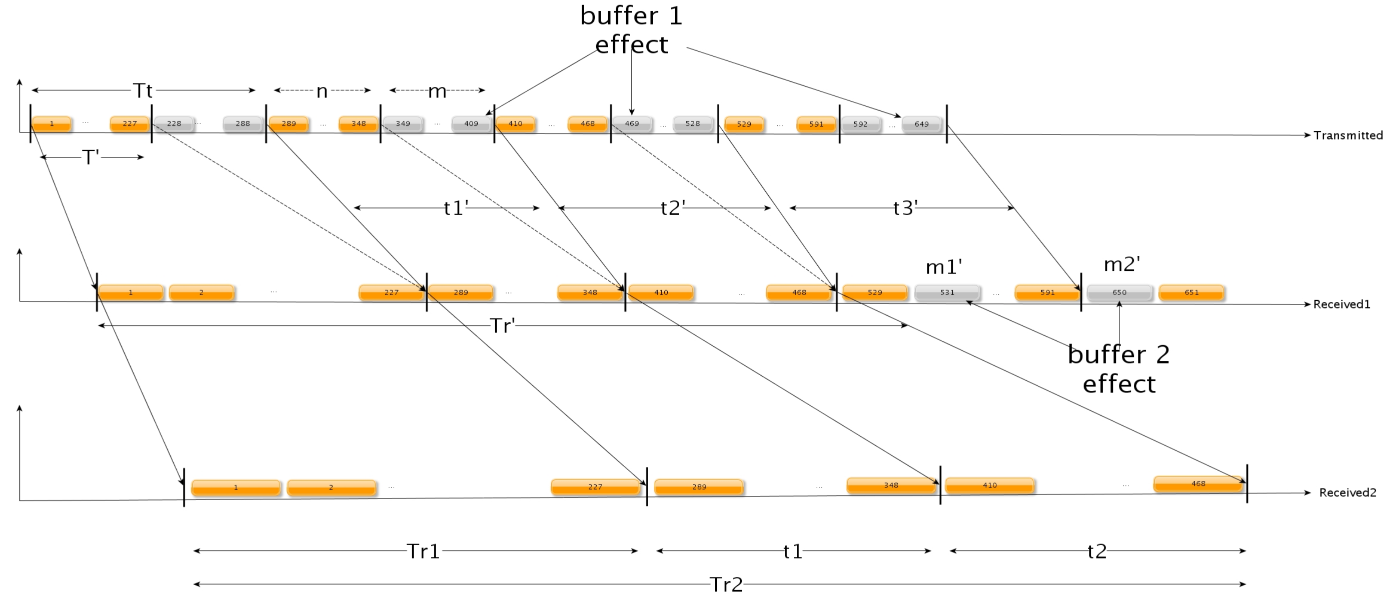

Figure 4. Estimating packets in concatenated queues with remote access.

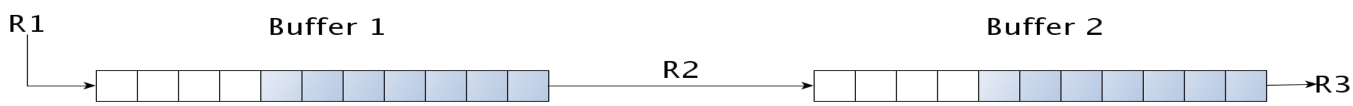

Figure 5. Estimating packets in queue in remote access (particular case).

B. New methodology

The methodology is based on the premise that the output rate can be obtained from traffic capture at the destination device. This output rate depends on the technology used in each case (Ehternet, WiFi). The output rate can be determined because the remote capture includes the n received packets in t seconds, and packet length is known. For calculating the input rate, we know 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, the output and input rates can be estimated only from the data contained in the *destination*.

In Fig. 4, the *Transmitted* trace corresponds to 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 available trace in order to determinate all the link characteristics. Dropped packets are the grey ones.

Fig. 5 represents an example in which two buffers are concatenated, *Buffer* 1 has an *upper limit* and a *lower limit*, as described in [11]; *Buffer* 2 uses the traditional FIFO policy. These two buffers will fill when $R_1 > R_2 > R_3$. 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. *Buffer* 2 has a different behaviour in congestion time because if a packet gets out, another one can get into the buffer, so packets will not be discarded in bursts. This figure clearly shown two different packet loss patterns which corresponds to each buffer.

TABLE I. EQUATIONS FOR ESTIMATING BUFFER PARAMETERS OF FIG. 4.

$Rate$	$Buffer\ size$
$R_3 = \frac{n_{rx}}{t_{r2}} \times packet_{size}$	$L_{Buffer1} = \frac{T'_r}{\frac{1}{R_1-R_2}+\frac{1}{R_2}}$
$R_2 = \frac{n_{rx}+m'_{tx}}{t_n} \times packet_{size}$	$L_{Buffer2} = \frac{T_{r1}}{\frac{1}{R_2-R_3}+\frac{1}{R_3}}$
$R_1 = \frac{n_{rx}+m_{tx}+m'_{rx}}{t_{r2}} \times packet_{size}$	

Analysing the Fig. 4, we can deduce the rates R and the buffer length L_{Buffer} from the remote capture, we can obtain all the parameters using the expressions shown in Table I, which corresponds to the same variables. In theses equations, the most important parameters are the m values because they correspond to the packet loss pattern of each buffer and they give information about the buffer size. The m values are determined by observing the packet loss patterns and it is given by: $m = number\ of\ patterns$.

We have described the methodology using an example, in which a buffer that drops one packet at a time, is concatenated with other which drops packets in bursts, so the m values or patterns can be easily determined. With the aim of test our methodology, the next section presents a more complex scenario with two concatenated buffers whose packet loss is in burst.

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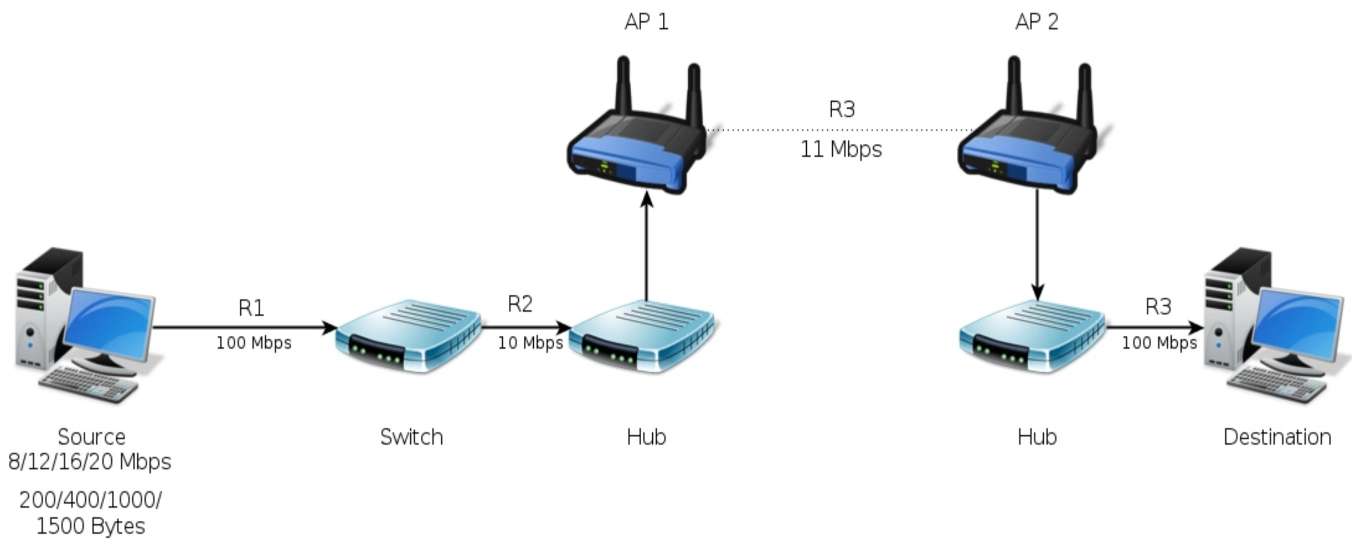

Figure 6. Topology used for estimating the buffer size in wired and wireless network.

IV. Experimental results

Real tests have been deployed in a testbed and results are analysed according to the procedures cited above. Tests have been performed for two different conditions: a single buffer overflow and two different concatenated buffers. The tests are repeated using different values for the packet size and the bandwidth.

A. Laboratory environment

We have implemented a controlled network environment in order to study two different devices: a switch (3COM) and an access point (Linksys WAP54G). The topology is shown in Fig. 6 in which different bandwidth limits were set in the hubs and the access points in order to create a bottleneck to be measured. With the proposed methodology, we will characterize the output buffers of the switch and the access point. Real machines have been used (Linux kernel $2.6.38-7$, $Intel^{\circledR}\ Core^{TM}\ i3\ CPU\ 2.4\ GHz$) for sending and receiving the test packets, in order to identify the buffer behaviour of the devices across the network path.

B. Test procedure

The test is based on the sending of a burst of UDP packets from the source to the destination, so as to produce a buffer overflow on the different devices. The test is intended to find out the bottlenecks that appear, and whether they have the same or different behaviour, according to the size of the sent packets. This test is repeated using different amounts of bandwidth for R_1 $(8, 12, 16, 20\ Mbps)$. The wireless link is set to $11\ Mbps$. Packets of different sizes $(200, 400, 1000, 1500\ bytes)$ are used so as to determine if the buffer is measured in number of packets or in bytes. This information will also be useful to determine the packet size that generates the best results. This allows us to discover when the effect of bottlenecks appears and it will also be useful to determine if the size of the packets may modify the output rate after the bottleneck.

Once the communication is set, we capture the traffic on the end device. This procedure generates a remote capture in the destination host which is analyzed in order to estimate the bandwidth, packet loss and buffer size. The accuracy of these estimations is compared with the one obtained in previous works [11] to validate the method proposed in the present work.

C. Packet loss patron analysis

1) Finding the number of patterns: When the traffic traverses the SUT, packets are lost according to different patterns. If we are able to identify them, this can give us useful information about the number of the buffers in the path. With the aim of finding a number of patterns, we made a study of the packet loss using different amounts of input bandwidth $(8, 12, 16$ and $20\ Mbps)$ which may correspond at the same number of bottlenecks. All the transmitted packets have a length of $1500\ bytes$.

Fig. 7 shows the results for this test. The charts show different packet loss patrons which can be determined by observing the groups of packets around the same packet loss value (see coloured eclipses). The $8Mbps$ graphic, when the input rate is under $10Mbps$ (switch maximum output rate) we can observe that packet loss remains below under 50 packets. There are only a buffer effect, corresponding to the access point WIFI buffer. The $12, 16$ and $20Mbps$ graphic shows that appear three group of packet loss patron. First of them is similar than to the $8Mbps$ graphic, corresponding to the access point WIFI buffer because the output rate of the switch is set to $10\ Mbps$, so the input rate of the access point has no variations. A second group have packet loss close to $125, 160$ and 205 corresponding to the switch buffer. As expected, packet loss increases when input rate grows. But, also, a third group of packet loss is observed and it corresponds to the sum of the packet loss produced by switch and the access point, so this effect is present when both devices drop packets at the same time.

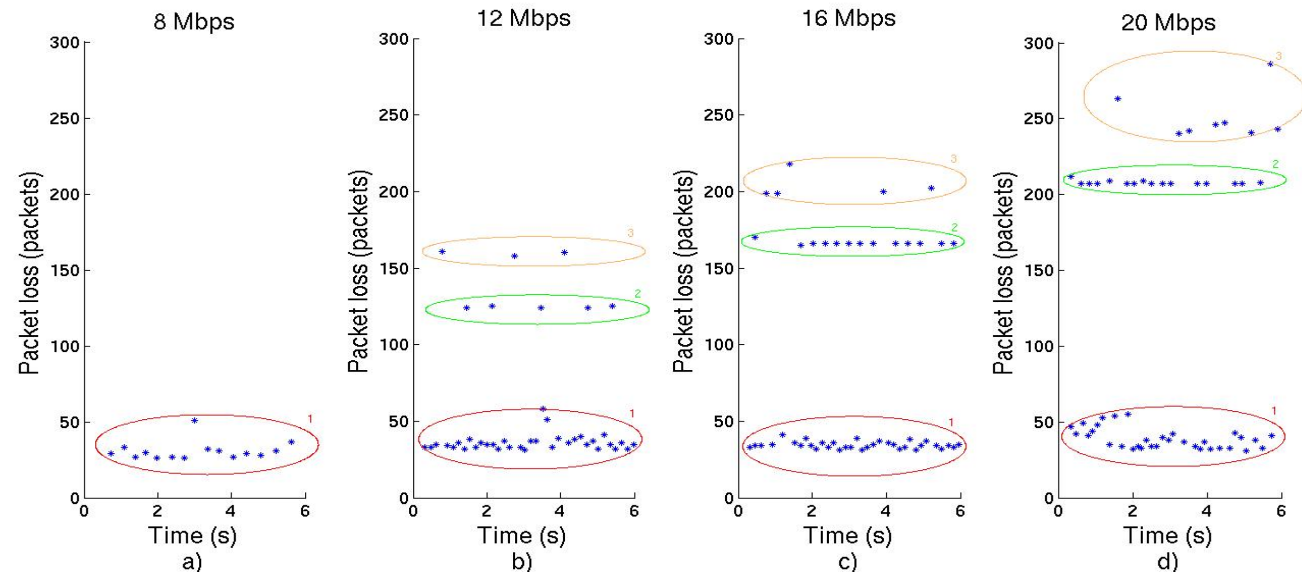

Figure 7. Packet loss patron in the switch and the access point for different bandwidth amounts when packet size is 1500 $bytes$.

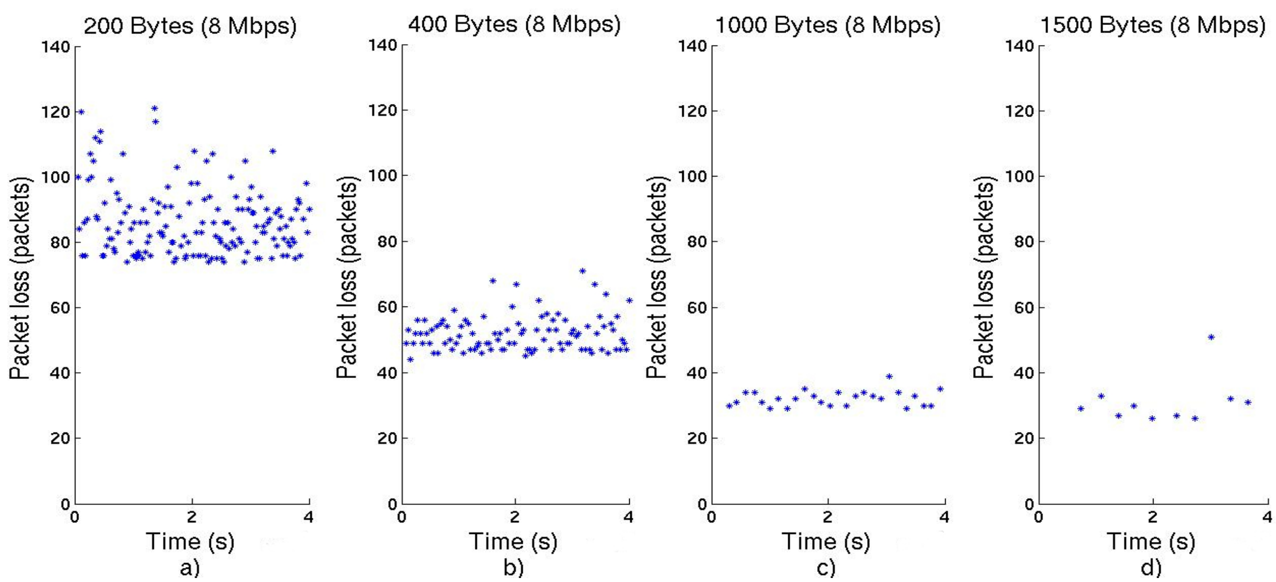

Figure 8. Packet loss patron in the access point for different packet sizes when the bandwidth is 8 $Mbps$.

2) Analyzing the effect of the packet length: In this case we developed two different tests. In the first test, we have selected a bandwidth of 8 $Mbps$ with the aim of producing an overflow on the second buffer ($AP1$) but not on the first one ($Switch$). The test is repeated using packets of $200, 400, 1000$ and 1500 $bytes$. The results are shown in Fig. 8, where different packet loss patterns can be observed.

The amount of lost packets is roughly the same for every test (Fig. 8) and they correspond to the ones obtained in Fig. 7, due to the relationship between the buffer size and the input and output rate of the access point buffer. In the WIFI technology, the output rate increases when packet size increases, so packet loss value is affected by packet length, and it decreases for bigger packets. In addition, packet loss has less dispersion when packet size is bigger.

In the second test (Fig. 9), we used a fixed bandwidth of 20 $Mbps$ in order to flood both buffers. This test allow again us to characterize the relationship between packet size and packet loss pattern. But in this case, the situation is more complex since there are two effects concatenated corresponding to the two buffers. The explanation is similar to the Fig. 7 and 8. There are three groups of packet loss patterns and the different values of the results occurs because the variability of the buffers output rate, which depends on the packets sizes.

3) Estimating the buffer size: We obtained the buffer size for the switch and the access point using to the proposed methodology but analyzing one buffer at a time according to each pattern, and eliminating the effect of the other buffer. In both cases, the appearance of a buffer with *upper limit* and *lower limit* is observed, which maximum size is roughly 120

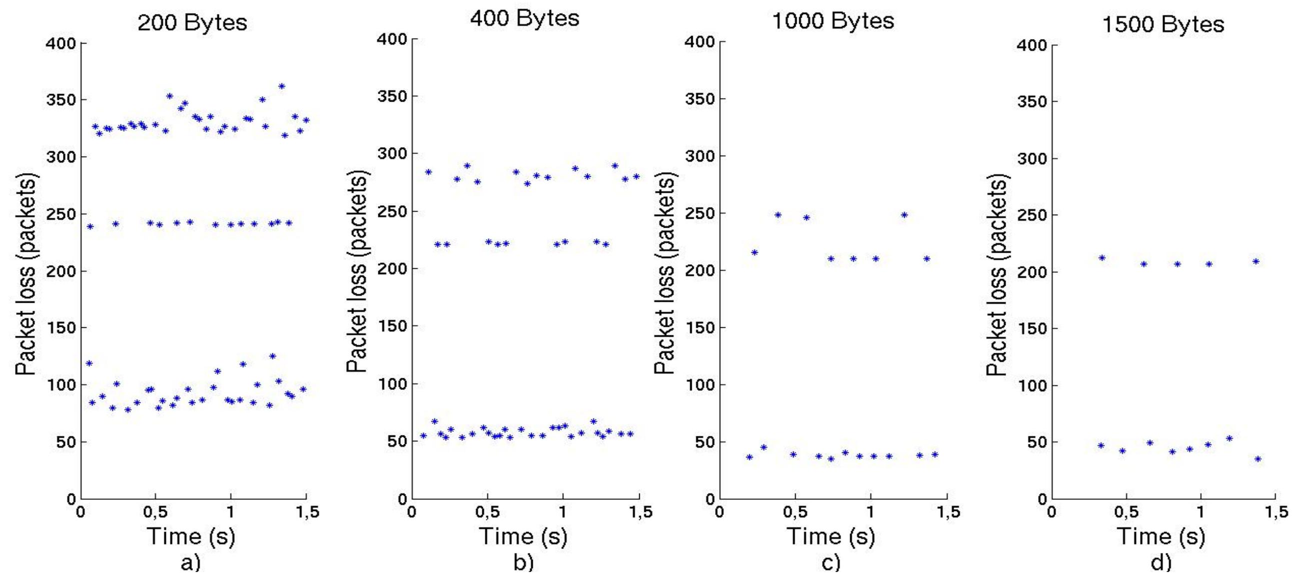

Figure 9. Packet loss patron in the switch and the access point for different packet sizes when the bandwidth is 20 $Mbps$.

packets for the switch and 50 packets for the access point. The more noteworthy is that results are similar that the estimates that we done, in previous works, with isolated buffers.

V. Conclusion

This article has presented a packet loss analysis, which is useful in order to describe the technical and functional characteristics of commercial buffers on a 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 a controlled laboratory scenario, including wired and wireless devices. Accurate results of the buffer size and other parameters have been 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 progress in order to improve the accuracy, especially when measuring the input rate when a wireless link is in the network path. Moreover, it would be interesting to discuss how to use the measures to infer on the buffering strategies, which could then lead to transport protocol adjustments that would consider these strategies to maximize QoS.

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