Bandwidth measurements and capacity exploitation in Gigabit Passive Optical Networks2014 Fotonica AEIT Italian Conference on Photonics Technologies

Original
Bandwidth measurements and capacity exploitation in Gigabit Passive Optical Networks2014 Fotonica AEIT Italian Conference on Photonics Technologies / Arianna Rufini;Edion Tego;Francesco Matera;Marco Mellia. - STAMPA. - (2014), pp. 1-4. ((Intervento presentato al convegno Fotonica 2014 tenutosi a Napoli nel 12 May 2014.

Availability:
This version is available at: 11583/2565556 since:

Publisher:
IEEE - INST ELECTRICAL ELECTRONICS ENGINEERS INC

Published
DOI:10.1109/Fotonica.2014.6843833

Bandwidth Measurements and Capacity Exploitation in Gigabit Passive Optical Networks

Arianna Rufini, Edion Tego, Francesco Matera
Fondazione Ugo Bordoni
Viale del Policlinico 147
00161 Rome, Italy
mat@fub.it

Marco Mellia
Politecnico di Torino
Corso Duca degli Abruzzi 24
10129 Turin, Italy

***Abstract*—We report an experimental investigation on the measurement of the available bandwidth for users in Gigabit Passive Optical Networks (GPON) and the limitations caused by the Internet protocols. We point out that the huge capacity offered by the GPON highlights the enormous differences that can be showed among the available and actually exploitable bandwidth in the case of TCP. In this ultrabroadband environment we also investigated on use of the UDP and of the multisession TCP. A correlation in terms of QoE is also reported.**

Keywords—QoS; QoE; throughput; access capacity, GPON

I. INTRODUCTION

The ubiquity of Internet access, and the wide variety of Internet-enabled devices, has made the Internet a principal pillar of the Information Society. As the importance of the Internet to everyday life grows, reliability of the characteristics of Internet service (availability, throughput, delay, etc.) grows important as well. Service Level Agreements (SLAs) between providers and customers of Internet services regulate the minimum level of service provided in terms of one or more measurable parameters.

Currently SLAs are tested in terms of some network performance parameters as "bandwidth" (generally expressed in terms of raw throughput). However, with the evolution of the applications, SLAs will regard aspects more and more related to user perception and verification towards applications. Therefore we also need to define new metrics that take into account Quality of Experience (QoE). This also implies that metrics and measurements have to be applied at the application layer; therefore suitable SLA parameters must be defined for specific Web services such as YouTube. These Application SLAs (ASLAs) will necessarily depend on lower-layer SLAs: clearly, for high-quality HD video delivery it is necessary to have a very good large broadband physical connection that has to be guaranteed by a suitable SLA between ISP and user. This is a necessary but not sufficient condition for high quality of a service at application layer.

In this scenario, each user may have SLAs with different providers (e.g. ISP, VoIP, IPTV); we therefore need to consider correlation among different SLAs.

It is well known that the amount of useful bits for seconds (defined as either *goodput* if considered at application layer, or *throughput* if at transport layer) can differ very much with respect to the capacity offered at the physical layer, or line capacity [1-5]; for instance, in the case of the Transfer Control Protocol (TCP) the measured throughput can be much lower than the line capacity. This results particularly evident in the case of network conditions with high values of bandwidth delay product [3], which are especially common in high capacity accesses with line rate higher than 20 Mb/s [6]. This difference between line capacity and throughput is a cause of several problems, especially in terms of SLA verification between users and ISPs. Therefore tools are needed to simultaneously evaluate throughput (or goodput) and line capacity. The most reliable method to estimate the line capacity is to locate a specific measurement device at the gateway of the user (modem, CpE,…). This method is clearly not scalable and it is very expensive, since it requires installing monitoring boxes very close to the customers' access links. Conversely, cheap methods to evaluate the user connection quality are based on “speed test” tools which measure the time to download a specific data file from a dedicated server. Therefore such speed tests measure the application layer goodput (or the transport throughput) and they are not reliable to evaluate the access capacity. In this paper we investigate on network performance of the ultra broadband accesses in the contest of Fiber to Building/Home networks based on Gigabit Passive Optical Networks (GPON), with all the relative implications at service and application level.

We have adopted the novel method proposed in the framework of the FP7 MPLANE project (D.5.1, sect. III) that allows us to simultaneously measure Round Trip Time, jitter, packet loss, throughput and access capacity. We also test the correlation between QoS and QoE for some web TV services adopting a QoE procedures defined in [7-8].

After this introduction in Section II we describe our methodology to measure QoS and QoE and in particular the software agent realized to measure QoS parameters at two OSI Layers: L2 and L4, which allows us to well typify the bandwidth characteristics from the user and ISP point of view. Section III reports the experimental setup for the ultrabroadband environment (GPON) and the results in terms of throughput versus capacity with the relative impact on the QoE. Finally conclusions are drawn in Section IV.

II. Method to Measure QoS Parameters at Different Layer (Channel Capacity and Throughput) and MOS Evaluation

Currently, the Internet services and applications are based on IP protocol and almost of them rely on TCP, most notably HTTP. This implies that download throughput is regulated by TCP at the transport layer. Theory and experimental tests [1-6] showed that TCP flow and congestion control mechanisms suddenly become the major bottleneck when exploiting high capacity paths with large Round Trip Time (RTT).

The difference between throughput and line capacity could be deeply limited by avoiding mechanisms of flow control (flow control algorithms) as for an instance using UDP, even though such a protocol is not able to control and manage packet loss processes and does not perform congestion control mechanism. These simple considerations point out that the QoS parameters should be defined for each Layer of OSI stack, and as a consequence we could define corresponding SLAs related to different OSI Layers, from the physical Layer to the application one.

From the user point of view the most important metric is the Quality of Experience (QoE) that is just related to the user perception and depends on several human factors [7-9]. The QoE measures the fact if the user likes a service, or not. Therefore QoE should consider all degrading effects such as blur, jerkiness, blocking, freezes and so on [9]. These effects depend on the network performance but the relationship with network parameters is usually rather complex. QoE is measured in terms of Mean Opinion Score (MOS) [10-13] and it can be evaluated with either subjective or objective tests. MOS based on subjective tests is generally evaluated by a pool of reviewers that look at the video services and manifest a score either following the quality evolution in time or at the end of the service [14-15]. Conversely QoE based on objective tests uses software and algorithms tools to quantify the service degradation, for an instance also by using a reference service [9]. In the case of HTTP video streaming applications like YouTube, previous studies [7-8] have shown that both the number of stalling events and their duration are the most important features influencing the QoE undergone by the end-user. A stalling event corresponds to the interruption of the video playback due to the depletion of the playback buffer at the user's terminal. It occurs when the available bandwidth is lower than the required video encoding bitrate. A deep investigation on the relationship between MOS and stalling events was reported in [7] and such results were used for our QoE evaluation.

The multiservice level agreement verification method defined in MPLANE permits to measure:

- the line capacity to verify ISP SLA;
- the throughput (as available bandwidth for the user);
- the goodput (as available bandwidth for the user at application layer).

Therefore for our investigation we realized a novel software agent based on the following steps:

- First, it performs throughput measurement (B_t) by adopting a TCP file download to evaluate Bt.
- It measures other QoS parameters as delay t_d (RTT measured with a PING), congestion window and threshold value.
- After the RTT and B_t measurements, an estimation of the line capacity, B_c, can be obtained by means of the following equation:

$$B_t \leq \frac{\min(CWND, RWND)}{RTT} \leq \min\left(B_c, \frac{RWND}{RTT}\right)$$

  where CWND is the congestion window, B_t the throughput and B_c the line capacity.
- To measure the line capacity it performs an UDP test to obtaining other QoS parameters such as jitter and packet loss. A UDP stream is sent from the server with a line capacity of B_c', measuring the lost datagrams with respect to the transmitted datagrams. In such a way we obtain the useful transmitted bits, BT_U, and from the total transmitted bits BT_T, we can achieve the estimated line capacity as $B_c = (BT_U/BT_T) * B_c'$. In the second step the effective line capacity is measured by means of an UDP test based on a downloading of a file and forcing the transmission rate $B_c^* = B_c - \epsilon$ (i.e. $\epsilon=0.001 * B_c$), verifying that a packet loss is lower than 0.1%. The value of ϵ depends on the required reliability of the measurement and in particular on the SLA between ISP and user. If the required packet loss is verified B_c^* is the line capacity.

Such a method was implemented by adopting IPERF tool [16].

Subjective MOS measurements are generally based on evaluation carried out by a group of reviewers for some video lab test [7-9]. However such a method is difficult to be implemented, especially in field environment. On the other hand our interest for MOS evaluation was mainly to analyze the network conditions that could induce some form of degradation on video services. Therefore our approach was based on the detection of events that induced also a minimum MOS degradation (i.e. from 5 to 4) and we considered less relevant a different interpretation of an event in terms of MOS between 2 and 3. Therefore, according to the MOS investigation reported in [7] we adopted a MOS evaluation based on an only reviewer that had to decide a score according to the following guidelines:

- 5: no stalling;
- 4: only one stalling with a duration shorter than 0.5s;
- 3: only one stalling event with a total duration between 0.5 and 3s;
- 2: either for 2 stalling events with a duration lower than 4s or for one stalling with a duration between 3 and 8s;
- 1: for all the worse cases.

The evaluation was carried out on different video services according to the available throughput under test, and in particular from YouTube standard (360p) to HD (1080p), up to H.264 video (1920 x 1080 pixel requiring a video encoding bitrate equal to 10Mb/s), and MPEG-2, (1280 x 720 pixel, 18Mb/s).

III. GPON EXPERIMENTAL PLATFORM AND RESULTS

The adopted test bed is shown in fig.1 [15]: the core part is composed of four Juniper M10 routers interconnected using 1Gb/s long haul optical links in the Rome area. Three Cisco 3845 edge routers are deployed at the access part of the network by means of Gigabit Ethernet optical links. Finally, the test bed is completed with GPON access networks composed of an OLT (Optical Line Termination) and up to eight Optical Network Terminals (ONT), offering a shared bandwidth equal to 1.244Gbit/s. A desktop PC is connected to the user ONT. A high-end server is then connected using Gigabit Ethernet links. To emulate different RTT a delay generator is added on the path between Server and Client to introduce an additional delay *d*.

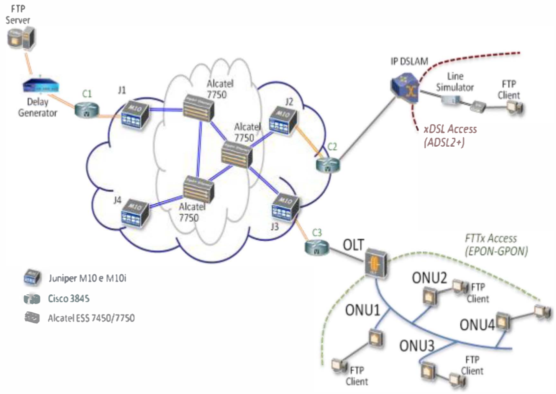

Fig. 1. Test bed setup

To guarantee an End-to-End minimum bandwidth in the backbone path, we use the technique described in [15] that allows us to assign a guaranteed bandwidth between two end-points of the network by means of the Virtual LAN and Virtual Private LAN Service (VPLS) technique. So this technique enforces strict bandwidth requirements so that background traffic that may be present on the testbed does not interfere with our tests. Three different Operating Systems (OSes) are considered as client, namely Windows XP SP3, Windows 7 and Linux Ubuntu 9.10. The maximum CWND at the server was set to 512kB, which is larger than the actual maximum RWND imposed by default for all considered OSes.

To outline the differences that can be measured between throughput and line capacity, we carried out QoS parameters measurements in the GPON access profiling different line capacities at the ONU output ranging between 20 and 100 Mb/s and considering different Operating Systems (Oses).

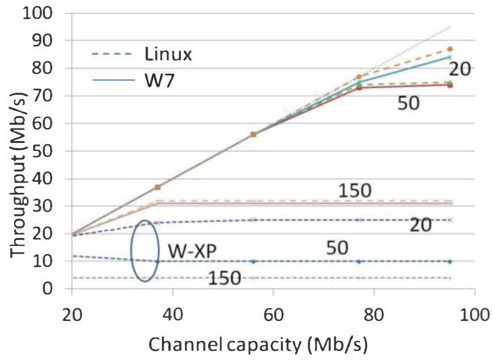

Fig. 2. Throughput vs line capacity for Linux, W7, Windows XP for different RTT (20, 50, 150 ms)

In fig. 2 we report throughput versus line capacity (channel) for different RTT. The difference is relevant for high RTT, and overall by considering Windows XP (W-XP).

This is due to enhanced TCP algorithm implementation, related to adaptive parameters in the algorithm (i.e., Auto-Tuning of the RWND which can grow much larger than the 64kB offered by Windows XP). As already shown in [15] the TCP limitation could strongly degrade video services based HD streaming. As an example in fig. 3 we report the MOS versus throughput in case W-XP of fig. 2, for H264 video and for MPEG2 video. The line capacity was 50 Mb/s and the throughput varies changing RTT. We see as RTT has a fundamental role in terms of QoE; we further observe a weak better immunity of MPEG2 with respect to H264 in terms of throughput threshold, even though it requires a higher video encoding bitrate.

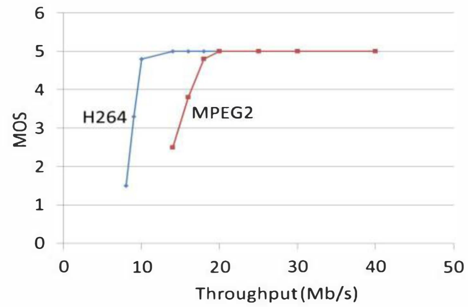

Fig. 3. MOS vs Throughput for W-XP in case of 50 Mb/s line capacity for two different RTT; 50 ms for H264 and 20 ms for MPEG2

Fig. 3 also shows that W-XP can show strong throughput reduction for high RTT, which could induce degradation also for YouTube videos. For an example for RTT=150 ms the W-XP throughput was equal to 4.7 Mb/s and by considering the YouTube “Avatar trailer” (having a duration of about 4 min) we verified a MOS reduction from 5 to 4. The time behavior of throughput for an ONU line capacity of 100 Mb/s is reported in fig. 4, adopting for the user a PC with Windows 7. The results show the bandwidth at disposal of the user is much lower than the line capacity (100 Mb/s).

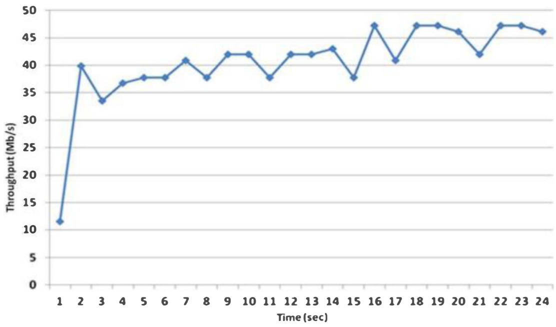

Fig. 4. Time evolution of the TCP Throughput for a channel capacity of 100 Mb/s and RTT=100 ms

Conversely in fig. 5 we show how UDP can permit to fully exploit the GPON capacity.

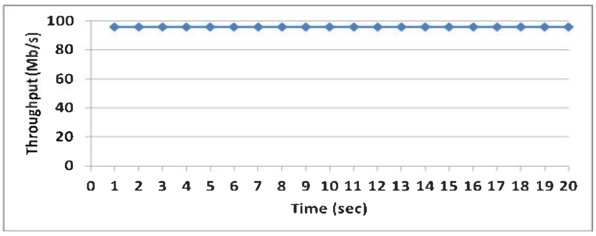

Fig. 5. Time evolution of the UDP Throughput for a line capacity of 100 Mb/s and RTT=100 ms

We also show how it is possible to better exploit the physical layer capacity by adopting multiple TCP connections to avoid the bottleneck of a single connection and in fig.6 an example is reported.

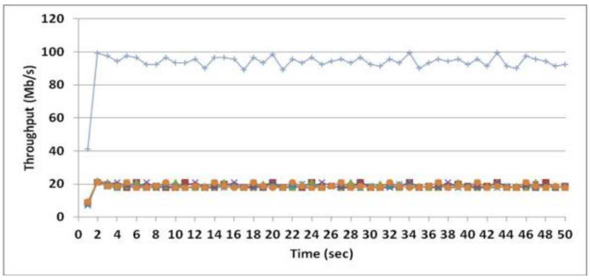

Fig. 6. Evolution over time of the TCP Throughput for a Multisession with 5 flows. The highest curve is the sum of the five flows. GPON line capacity of 100 Mb/s, RTT=100 ms

IV. CONCLUSIONS

In this work we report a method to simultaneously measure the throughput and the line capacity in order to evaluate the effective bandwidth at disposal of the user and the one offered by the access technology provided by the ISP. We also correlate these measurements with MOS tests to verify the suitability of some video web services in the case of ultrabroadband based on GPON, but that it could be strongly reduced by TCP behavior.

ACKNOWLEDGMENT

Work carried out in the framework of the FP7 MPLANE project.

REFERENCES

[1] Lakshman, T. V., Madhow, U., and Suter, B., “TCP/IP performance with random loss and bidirectional congestion”, IEEE/ACM TRANSACTIONS ON NETWORKING, 2000, 8, (5), pp. 541-555.

[2] T. V. Lakshman, Member, “The Performance of TCP/IP for Networks with High Bandwidth-Delay Products and Random Loss” IEEE, and Upamanyu Madhow IEEE/ACM TRANSACTIONS ON NETWORKING, 1997. 5, (3), pp. 336-350.

[3] Mellia, M., Lo Cigno R., Neri, F.,, “Measuring IP and TCP behavior on edge nodes with Tstat,” Computer Networks, 2005, 47, (1), pp. 1-21.

[4] Bolletta, P., Rea, L., “Monitoring of the User Quality of Service: Network Architecture for Measurements and role of Operating System with consequences for optical accesses” ONDM 2011 – Bologna, Italy February 2011.

[5] Del Grosso, A., Rea, L. Bolletta, P., Valenti, A., Luisi, A. M., “On the Impact of Operative Systems Choice in End-user Bandwidth Evaluation: Testing and Analysis in a Metro-access Network”, IARIA ACCESS 2010 Sept. 2010 Valencia, Spain.

[6] Rea, L., Mammi, E., “Italian QoS Monitoring network: impact on SLA control”, Networks 2012, Rome October 15-18, 2012.

[7] P. Casas, M. Seufert, R. Schatz "YOUQMON: A System for On-line Monitoring of YouTube QoE in Operational 3G Networks" ACM SIGMETRICS Performance Evaluation Review, vol.41, no.2, pp. 44-46, 2013.

[8] P. Casas, R. Schatz, T. Hoßfeld "Monitoring YouTube QoE: Is Your Mobile Network Delivering the Right Experience to your Customers?" in Proceedings of IEEE Wireless Communications and Networking Conference, Shanghai, China, 2013.

[9] R. Stankiewicz, P. Cholda, A. Jajszczyk, “QoX: What is Really?” IEEE Communication Magazine, vol. 49, iss. 4, pp. 148-158, April 2011.

[10] R. K. P. Mok, E. W. W. Chan, R. K. Chang, “Measuring the Quality of Experience of HTTP Video Streaming” 12th IFIP/IEEE IM 2011 Conference, Dublin May 23-27 2011.

[11] K. Seshadrinathan, R. Seshadrinathan, A. Bovik, L. K. Cormack “Study of Subjective and Objective Quality Assesment of Video” IEEE Transactions on Image Processing, vol. 17, 2009, pp. 1-16.

[12] M. Fiendler, T. Hassfeld, P. Tran-Gia, “A Generic Quantitative Relationship between Quality of Experience and Quality of Service” IEEE Networks, march-April 2010, pp. 36-41.

[13] A. Finamore, M. Mellia, M. Munafò, R. Torres, S. G. Rao, “You Tube Everywhere: impact of Device and Infrastructure Synergies on User Experience”, IMC 2011 Berlin (GE) November 2-4 2011.

[14] F. Matera, et al., “Comparison between objective and subjective measurements of quality of service over an Optical Wide Area network”, European Transactions on Telecommunications vol. 19, 2008, pp. 233-245.

[15] Valenti, A., Rufini, A., Pompei, S., Matera, F., Di Bartolo, S., Da Ponte, C., Del Buono, D., Tosi Beleffi, G.M., “QoE and QoS Comparison in an Anycast Digital Television Platform Operating on Passive Optical Network” Networks 2012 - Roma – October 15-18, 2012.

[16] http://code.google.com/p/iperf.