Evaluation of QoS in Internet accesses for Multimedia applications (EQoSIM)

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Abstract

This paper presents a Java-based on-line Quality of Service (QoS) estimation system for Internet accesses specially aimed at real-time multimedia applications. The system is capable of estimating access capacity, available bandwidth and delay as the critical QoS parameters for this kind of applications, and to do it from the point of view of the final user. The algorithm selected for QoS estimations is one-way, and is based on the packet train technique. The system has been developed following the client-server model, where a central web server contains, among other things, the Java applet that implements the client side of the system, and has been validated using several commercial Internet accesses with different technologies: analog modem, Asymmetric Digital Subscriber Loop (ADSL), cable modem and Universal Mobile Telecommunications System (UMTS).

1. Introduction

From its beginning, Internet has experienced a huge increase in the number of users, services and data transferred. Different Internet access characteristics lead to very diverse levels of Quality of Service (QoS) [1], which can have a great impact on service performance, particularly on real-time ones.

There are several actors interested in estimating QoS, namely network operators, Internet Service Providers (ISP) and final users. Network operators are concerned about network planning, ISPs want to ensure a certain degree of service for the final user, who finally wants to assess the QoS obtained. Thus, these three agents can greatly benefit from a system designed to estimate QoS, making it possible for them to compare what is theoretically offered with what is actually obtained from an Internet access. This is especially important for final users, given the fact that over the last years, problems derived from poor Internet accesses are in the top positions in the number of complaints to consumer associations. Moreover, QoS can also be considered from different points of view: security, performance, speed, reliability, overall user impression, etc. As a result, the complex set of elements that influences QoS makes its measurement a difficult task.

Several QoS-related network parameters estimation tools have been designed through the last years, being bandwidth one of the most widely measured parameters. Other parameters such as delay or packet loss rate are also frequently used, but to a lesser extent. Reference [2] presents a review of some of the most popular bottleneck link bandwidth estimation techniques that tools as Nettimer [3] or Pathchar [4] use. However, other estimation tools use a more direct approach to bandwidth estimation, especially the common on-line bandwidth speed tests [5]-[8]. The majority of these systems measure the time required to transfer one or several fixed-size files to different servers in order to calculate bandwidth. Nevertheless, this method has a major drawback: only the bandwidth for Transmission Control Protocol (TCP) file transfers is estimated. Real-time applications, on the other hand, are usually transmitted using the Real-time Transport Protocol (RTP) [9], that in turn uses the User Datagram Protocol (UDP), so existing bandwidth speed tests are not well suited to this type of applications.

In this context, this paper presents a Java-based on-line QoS estimation system for Internet accesses specially aimed at real-time multimedia applications called EQoSIM (Evaluation of QoS in Internet accesses for Multimedia applications). It is capable of estimating access capacity, available bandwidth and delay as QoS parameters using UDP packet trains. It has been developed using Java technology, so it can be widely, easily and quickly accessible for the final user.

The rest of the article is structured as follows: section 2 presents the materials and methods used for EQoSIM, as well as the motivations behind its selection. Section 3 presents an overview of the system architecture. A description of the tests carried out to evaluate EQoSIM performance is included in section 4. Finally, section 5 presents the evaluation results obtained and the conclusions are summarized in section 6.

2. Materials and methods

As it has been stated before, the majority of the publicly available on-line bandwidth speed tests [5]-[8] use TCP file transfers as the basis to estimate bandwidth. This approach, however, has several drawbacks:

- Only the bandwidth for TCP file transfers can be estimated. UDP-based applications, mainly real-time ones, are not considered.
- The bandwidth estimation process is highly intrusive, and it is frequently required that the user does not send any other network traffic while the bandwidth speed test is being carried out. This is not a realistic situation since typical Internet users generate different traffics at the same time and the access capacity is a value not as useful as the available bandwidth [10].
- Usually, delay and packet loss rate are not considered. There are specific tools that take them into account [11]-[13], but they are not intended for the non-expert Internet user.

The following subsections explain the main characteristics of EQoSIM in more detail.

2.1. Bandwidth estimation algorithm

In the communication path there is usually a link that sets QoS parameters, and it is commonly called the *bottleneck link* [2],[10],[14]. Different estimation tools focused on discovering bandwidth in the bottleneck link (also called the *bottleneck bandwidth*) use measurement methods that can be classified into passive [13],[15], and active [3],[4]. Active measurement methods can be further divided into those that measure Round Trip Time (RTT) [12] and those that only measure one traffic direction (One-Way) [3]. The most used protocols in these measurement systems are UDP, TCP and Internet Control Message Protocol (ICMP).

Once the different measurement acquisition methods have been presented, it is very important to identify the most relevant QoS parameters for real-time applications. Two common and ubiquitous parameters used to measure QoS levels are bandwidth and delay [2],[10]-[11],[16]-[17]. These two parameters have been selected for EQoSIM because they make it possible for the client to check the performance of his Internet access, especially when it is used for real-time communications. As this kind of communications mainly uses RTP, which in turn uses UDP, this is the protocol selected for the estimations.

The bandwidth estimation algorithm selected for EQoSIM is One-way based on the transmission, in both directions of communication, of bursts of k UDP packets with constant packet size (S) (packet trains).

Given a path between two network end points that includes n links L_1, L_2, … L_n with bandwidths BW_1, BW_2, … BW_n, the bottleneck bandwidth (BBW) can be defined as [14]:

$$BBW = \min (BW_1, BW_2, \dots , BW_n) \qquad (1)$$

Next, given a link L_i with bandwidth BW_i and traffic load TL_i, the available bandwidth (ABW) in the link is defined as [14]:

$$ABW_i = BW_i - TL_i \qquad (2)$$

The procedure to calculate the available bottleneck bandwidth (ABBW) consists of sending packets in the burst at a rate equal to the already estimated BBW [14]. Moreover, the estimation of the percentage of packet loss (PL_{Rate}) can be calculated as the percentage of lost packets in the burst. Finally, delay can be measured if the packet train sender and the receiver are properly synchronized.

The parameters that characterize this algorithm (packet length, number of packets per burst, packet spacing in a burst and time between bursts) are fully configurable to select those better suited for each particular scenario.

It is important to note that this estimation method is much less intrusive than traditional bandwidth speed tests and produces acceptable results with a minimum bandwidth waste. It is also capable of estimating bandwidth under realistic circumstances, i.e. when the user is generating other network traffics, which makes the value of ABBW a crucial parameter in order to decide whether a particular real-time application can be used in conjunction with other traffics.

3. System architecture

EQoSIM has been is developed according to the scenario presented in Fig. 1. As can be seen, there is a central node where users come to get their QoS measurements. This central node contains a web server that hosts Hypertext Mark-up Language (HTML) pages and the Java applet that implements the client-side application to be displayed in a Java-compatible browser. In addition, a Network Time Protocol (NTP) server and an UDP bursts server (the server that receives the UDP packet bursts and replies to them) that takes the appropriate measurements are installed.

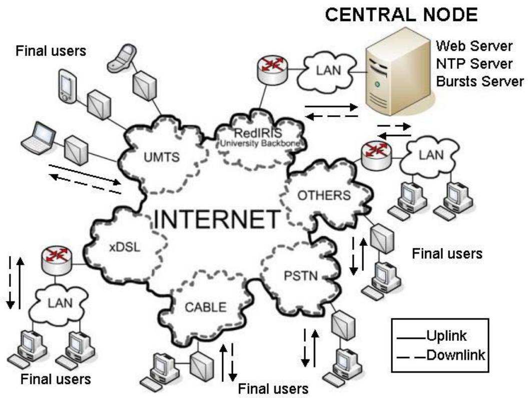

Figure 1: General network scenario

The data flow diagram of Fig. 2 shows the process of making a QoS estimation with EQoSIM. When a user loads the main web page (Fig. 3), an applet is downloaded showing three different versions: simple, advanced (for advanced users) and monitoring (designed to do scheduled QoS estimations). Then, the applet exchanges NTP messages with the server in order to be synchronized. As soon as the time offset between the server and the client is corrected, TCP communication is used to establish the burst parameters (number of bursts, frames per burst, frame length, etc.). When the server processes those parameters, the UDP bursts client is accepted or refused through the TCP connection. If the answer is affirmative, several UDP bursts are sent in the uplink.

To notice the end of burst, a TCP message is sent. Once the TCP message reaches the server, this one sends other UDP bursts to the client in order to evaluate the downlink. Finally, client and server exchange their measurements using TCP. This way, the results can be shown by the applet in the browser and stored by the server for further processing.

4. Evaluation tests

4.1. Test scenarios

EQoSIM has been validated in commercial Internet accesses. This paper presents several evaluation results obtained with the following commercial accesses:

- Analog modem over PSTN (Modem): V.90 standard modem with or without compression. The bandwidth of this access is low but it remains constant and 100% of its capacity is offered to the user. Frame size and flow control are configurable by the user. Depending of the modem standard, uplink and downlink can be symmetrical or asymmetrical.

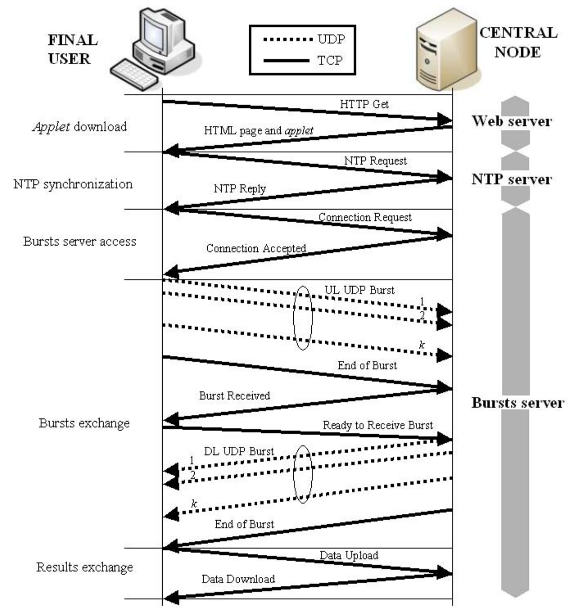

Figure 2: Data flow diagram

Figure 3: Main web page of EQoSIM

- Asymmetric Digital Subscriber Loop (ADSL): 128 kbps in the uplink and 256 kbps in the downlink, with 10% guaranteed in the contract. The capacity is greater than that of analog modem but only a percentage of it is available. Information is transmitted using fixed size Asynchronous Transfer Mode (ATM) cells [13]. As its name indicates, both capacity and available bandwidth are asymmetrical.
- Cable modem (Cable): 128 kbps with 20% guaranteed and a 1:4 concurrence ratio. As opposed to ADSL, the access is shared between several users. Bandwidth is similar to that of ADSL and the access is asymmetric too. In the uplink, Medium Access Control (MAC) is based on Time Division Multiple Access (TDMA) [18], whereas bandwidth assignation in the downlink is controlled by the Cable Modem Termination System (CMTS).
- Universal Mobile Telecommunication System (UMTS): 64 kbps in the uplink and in the downlink. The access is shared between some users. However, available bandwidth should remain almost constant,

but its value can vary depending on the radio link conditions. Delay is greater than in the other accesses due to channel coding and interleaving.

It is important to remark that a typical user of EQoSIM only knows the access parameters given by his ISP (access capacity and guaranteed bandwidth), but this is not enough in order to characterize the behaviour of the access in a working situation. The bandwidth available to a particular user may vary through the time in a particular access, since ISPs only guarantee a given percentage of it. As a result, real tests with commercial accesses can produce more significant results. A bandwidth monitoring process would be of special interest, and for that reason EQoSIM has the monitoring option (see Fig. 3).

4.2. Test parameters

The results presented in the next section correspond to several tests that consisted of:

- Number of bursts sent: 49 bursts in both uplink and downlink.
- Time between two consecutive bursts: 15 min.
- Variable frame size (*S*): 90, 120, 200, 500 bytes of UDP data without overhead.
- Packets per burst: *k*=5 and *k*=10 have been chosen.
- Test conditions: No competing traffic, in order to measure BBW instead of ABBW.
- Different tests over the same link have been interleaved to concur at the same hour of the day.

5. Results and discussion

The results obtained in the evaluation tests are presented in Table 1. It shows the mean (μ) and the standard deviation (σ) of the BBW for each of the tests presented in the previous section.

The following points discuss the relevant aspects of the test results for each technology in more detail:

- Modem: Tests were done with *S*=90, 200 and 500 bytes. As can be observed, *S* decreases using *k*=10 packets, and the link is asymmetrical (with a higher BBW in the downlink). The highest value of BBW was obtained when the smallest frames (*S*=90 bytes) were used.
- ADSL: Tests were done with *S*=90, 120 and 500 bytes. Depending on the value of *S* used, the BBW obtained at the IP layer (ADSL-I) varies. (ADSL-II) have been obtained by taking into account ATM headers. Finally, the percentage of the contract bandwidth that the ISP is really providing can be calculated. For *S*=500 bytes and *k*=10, the downlink reaches almost 100% of the contract, but the uplink is only at 20% of its nominal capacity.

Table 1: Bottleneck bandwidth

Access	*S* (bytes)	*k* (frames)	UPLINK		DOWNLINK	
			μ (kbps)	σ (kbps)	μ (kbps)	σ (kbps)
Modem	90	5	34.112	3.2461	47.801	10.976
		10	30.456	1.4617	46.729	6.4270
	200	5	23.810	1.3796	38.748	3.3194
		10	24.576	0.6859	40.704	1.1099
	500	5	25.602	0.9846	36.892	1.7431
		10	25.601	0.2089	38.426	1.3326
ADSL-I	90	5	23.440	1.6963	231.45	98.196
		10	23.565	0.9479	188.57	34.531
	120	5	22.153	1.1216	190.08	45.576
		10	22.248	0.6016	177.37	9.5506
	500	5	26.358	1.2772	202.78	36.384
		10	26.562	0.0833	203.01	26.252
ADSL-II	90	5	31.584	2.2856	311.86	132.31
		10	31.752	1.2772	254.09	46.529
	120	5	31.732	1.6066	272.27	65.284
		10	31.868	0.8617	254.06	13.679
	500	5	31.749	1.5384	244.25	43.826
		10	31.995	0.1003	244.53	31.621
Cable	90	5	233.36	10.299	184.57	60.485
		10	230.56	17.015	359.58	141.45
	120	5	298.39	29.214	193.78	86.228
		10	293.11	14.564	176.87	76.134
	500	5	119.95	1.7057	129.21	12.189
		10	78.421	0.3654	154.64	15.087
UMTS	90	5	57.433	13.514	61.930	7.0163
		10	59.672	10.676	61.061	2.5664
	200	5	59.433	12.973	63.083	4.685
		10	61.142	7.185	62.592	2.016
	500	5	61.387	12.441	64.537	2.9010
		10	63.246	4.9292	62.834	1.4475

- Cable: Tests were done with *S*=90, 120 and 500 bytes. As downlink and uplink have different methods for bandwidth assignation, their results must be treated independently. Thus, a detailed research would be necessary with different values of *k* and *S*.
- UMTS: Tests were done with *S*=90, 200 and 500 bytes. In this access all measurements give similar results. Furthermore, delay is much greater than in the previously mentioned wired accesses and there are more errors due to the mobile channel, but this results are not shown here.

6. Conclusions

A Java-based on-line QoS estimation system specially aimed at real-time multimedia applications has been developed to evaluate Internet accesses. This system is especially useful for final users who want to estimate the quality of their Internet access and check its performance regarding the use of real-time multimedia applications. The usefulness of this system has been evidenced in the evaluation of representative commercial Internet accesses, since ISPs offer wide QoS ranges that can vary through the time.

Evaluation results show that the number of packets per burst and the packet size have a big influence on the estimated QoS, so it is very important to study the particular technologies in depth, obtaining a suitable characterization of each access. The results obtained for analog modem, ADSL and UMTS accesses are conclusive, but further research is required for cable in order to obtain a more complete characterization of this kind of access.

The use of Java technology has many compatibility-related advantages, making it possible for the user to carry out the tests in different computers, with different operating systems and different web browsers, but time accuracy and security restrictions have been problematic. We are currently working on an in-depth study of estimation errors produced by the time precision provided by Java in order to obtain the optimum parameters k and S of the bandwidth estimation algorithm for a given access.

EQoSIM only provides QoS estimations between the end user and a central server, but real-time applications are frequently used in a peer-to-peer basis, so a modification of the application in order to take measurements not only between the user and the central node, but also directly between two users is being considered. Acquired data organization and its automation for further processing is also being considered as a future research line.

Finally, security and server load-balancing considerations must be taken into account if the application is made available to the general public. Regarding server load-balancing, the web server, the NTP server and the UDP bursts servers can be placed in three physically distinct servers using an appropriate network design with an incoming firewall that would also add a greater level of security.

7. Acknowledgment

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