

Influence of quality service in IP/MPLS network load with IPTV and VoD services.

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Abstract - The purpose of this research was to obtain results that clearly show the influence of quality of service in network IP/MPLS load with IPTV and VoD services.

Estimating the traffic and the quality of transmission in these networks is a major indication of network performance and thus ensures optimal QoS. Measure quality of service in terms of bandwidth, latency, and jitter and loss rate. The objective of this study is to first set up an IP/MPLS network with installed multimedia services, then study the network behavior in case of congestion and finally ensure the quality of service both in transmission and do the measurement of QoS. We used the DiffServ architecture to deploy QoS and then to measure again de QoS (latency, jitter, loss rate and bandwidth).

Keywords – Quality of Service (QoS), Streams, IP/MPLS, Latency, Loss rate, Jitter, DiffServ, IPTV, VoD, IP network, EDGE Router.

I. Introduction

With the simultaneous evolution of computers, the emergence of smartphones and new devices at low costs, such as microphones and digital cameras, applications are changing. They are now part of a society where media dominate. Whether in the field of video-on-demand entertainment, in the professional field for tele teaching, telemedicine and co-operative work environments or simply in the home with videoconferencing, today's applications are widespread use of media such as audio and video. The generic name of multimedia applications is given to these applications. Thus, the latter will invade the world (IP). Their main characteristics are high data rates, continuous flows that have strong time constraints, and synchronization needs to preserve the coherence of their presentation.

In addition, high-speed Internet access technologies are widely deployed and becoming more and more accessible. As a result, most multimedia content providers, such as new televisions, radios and entertainment industries (games), are setting up their own streaming infrastructures to deliver their content, either live or on demand.

However, the Internet Protocol (IP) as it was designed is not perfectly adapted to meet these constraints related to multimedia services. Today there is a network technology called Multi-Protocol Label Switching (MPLS) that offers new possibilities for this situation. For the purpose of this thesis, it is a question

of measuring and analyzing the different parameters of the quality of services, namely the bandwidth, the latency, the jitter and the loss rate of the different multimedia streams circulating in an IP/MPLS network.

II. Context, research methodology and multimedia network

1. Context and problem

The Internet protocol (IP) network was originally designed as a simple data network. Originally, the operation of the network in best effort (could do its best) could be suitable for a single type of traffic, non-real-time data traffic.

With the growth of multimedia traffic, social networks and Over The Top (OTT) services, the IP network has had to evolve to meet the needs of multimedia services that have time and bandwidth constraints. The perceived quality of its services decreases when transmission conditions deteriorate [4].

Estimating traffic and transmission quality in these networks is a major indication of network performance.

The parameters characterizing this performance are:

- The loss of packages;
- The transmission time;
- The variation of delay (jitter);
- The reception quality perceived by the multimedia user.

The purpose of this configuration is to allow a smooth reading of the decoded content regardless of the variation in the arrival time of the packages. The problem of packet loss in a network must be addressed by lost packet prediction algorithms or by using redundancy.

With the exponential increase in the number of real-time applications deployed in IP networks, fast recovery during outages is increasingly desired to ensure the continuity of communications services. Different fault-tolerance techniques have been developed to avoid and / or reduce the cutoff time of multimedia services. The purpose of these techniques is to determine backup paths capable of receiving and re-routing the multimedia communications traffic affected by a failure.

The TCP protocol guarantees the reception of packets by retransmission of unacknowledged packets. However, interactive services like videoconferencing cannot tolerate too much retransmission time. The interruptions of services offered by multimedia networks or deterioration in the quality of

transmission of streams is less and less acceptable and constitute the new requirements of quality of service.

Multimedia networks need regular monitoring of the network to implement traffic engineering techniques. The main purpose of these techniques is to avoid the traffic congestion phenomena and resulting degradation of the service. Such an approach to multimedia network engineering must then be based on a new form of traffic forecasting, bandwidth and transmission quality.

2. Objectives

The objective of this work is first of all to make an assessment of the performance indicators of the empty network in the first place and then evaluate these network performance indicators in the presence of new UDP flows. That is, perform measurements on the IP / MPLS network with a maximum of load flows to see the impact of these on the different services deployed while meeting their requirements in terms of latency, jitter and bandwidth and packet loss rate to satisfy users.

Then, we will focus on the ambiguity of the estimation and circulation of multimedia traffic flows in an IP / MPLS network between different nodes connecting the server and the client. This will allow us to take into account random length paths in order to respond more likely to the problem of traffic intermittency. And finally, under certain conditions, we will determine simple rules for estimating and sizing traffic.

3. Research methodology

The research methodology consists in setting up flows in the ESMT laboratory IP/MPLS network which includes four [4] routers capable of supporting the various multimedia services while respecting their requirements in terms of latency, jitter, bandwidth and the packet loss rate in order to satisfy users and to study the main elements of the existing IP / MPLS architecture as well as its extensions known as GMPLS and the mechanisms and protocols of tag distributions.

This aims at proposing techniques of optimization of the traffic and quality of transmission of the multimedia services (streaming audio and video, ToIP) in an IP/MPLS, so that the quality of transmission is optimal on the real network, afterwards to observe the impact of network load on services.

More precisely, we are interested in the problems of estimate and the circulation of the fluxes of multimedia traffic in a network IP / MPLS between different knots linking up waiters and customers. This allows us to take into account paths of arbitrary length in order to respond more effectively to the problem of sporadic traffic. We set, under certain conditions, simple rules for estimating and sizing traffic [5].

4. MPLS and multimedia flow

Our network will be implemented on the real equipment that constitutes the MPLS (HUAWEI) model of our switching laboratory of the ESMT. This model consists of four (4) routers of which we have two (2) PE routers and two (2) routers CE.

The CE1 router is linked to a switch on which we connected an Asterisk server, and the multimedia server (IPTV, VoD, Streaming Audio, etc.). The network is shown in Figure 1.

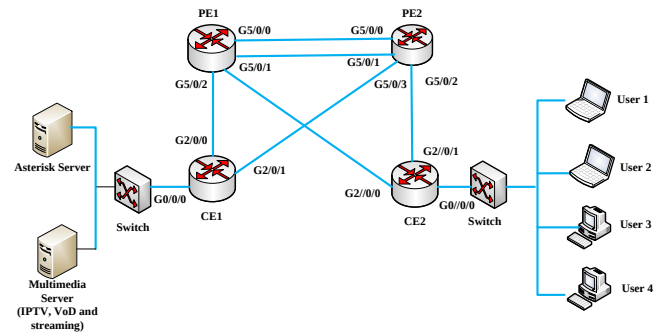


Figure 1: IP/MPLS Network and services

The address of the servers is given in Table 1 [1].

Router	Interface	@IP
PE1	G5/0/0	8.8.8.1 / 30
	G5/0/1	8.8.8.5 / 30
	G5/0/2	7.7.7.13 / 30
	G5/0/3	7.7.7.5 / 30
	Loopback0	10.1.1.1 / 32
PE2	G5/0/0	8.8.8.2 / 30
	G5/0/1	8.8.8.6 / 30
	G5/0/2	7.7.7.1 / 30
	G5/0/3	7.7.7.9 / 30
	Loopback0	10.2.2.2 / 32
CE1	G2/0/0	7.7.7.14 / 30
	G2/0/1	7.7.7.10 / 30
	G0/0/0	192.168.1.1
	Loopback0	10.3.3.3 / 32
CE2	G0/0/0	7.7.7.2 / 30
	G0/0/1	7.7.7.6 / 30
	G0/0/0	192.168.2.1
	Loopback0	10.4.4.4 / 32
Asterisk Server	-	192.168.1.5 / 24
IPTV server	-	192.168.1.2 / 24
VoD server	-	192.168.1.3 / 24

Table 1 addressing interfaces and servers

III. ANALYSIS AND MEASUREMENT OF QOS PARAMETERS

1. Measurement procedure

In the network set up, it is important to know and change the indicators acting in the QoS to know the impact that is presented in relation to these indicators. The Quality of Service study of this network consists of measuring the effect of each service operating in the network. That is to say, depending on the service, we will have to determine the bandwidth occupied the

variation of the jitter, the latency and the loss rate. This lets you know what effect each service has on the network.

Analyses were performed by the following tools:

- Iperf: is network performance measurement software, available on many platforms (Linux, BSD, Mac, Windows ...). It comes in the form of a command line to run on two machines (client / server) arranged at the ends of the network. Iperf measures bandwidth, latency, jitter, and packet loss rate [10]
- Wireshark: is an open source network protocol analyzer. Its goal is to capture frames revealing security flaws, see performance on the network.

After installing Iperf on the server machine and client machine, we measure bandwidth, latency, jitter, and loss rate by running the following commands:

- On the server: **iperf -s -u**: option -s to specify the server, option -u for UDP protocol;
- On the client: **iperf -c @IP -u -b 2m -t 60** : option -c to specify the client, @IP address of the server, option -u for UDP protocol, -b for bandwidth.

2. IPTV Service

Table 2 shows the network measurement results for the IPTV service when the IP/MPLS network is not loaded by the services. Table 3 is the measurements when the network is loaded into services (IPTV, VoD and ToIP).

IPTV streams (Mbits/s)	Bandwidth (Mbits/s)	Latency (ms)	Jitter (ms)	Loss Rate (%)
2 M	2	1	0	0
4 M	4	1	0	0
8 M	8	1	0.352	0
16 M	16	1	0.554	0
32 M	32	1	0.309	0
64 M	64.3	2	0.799	0
128 M	95.8	15	0.245	0.032

Table 2: Measuring the QoS of the network without streams

IPTV streams (Mbits/s)	Bandwidth (Mbits/s)	Latency (ms)	Jitter (ms)	Loss Rate (%)
2 M	2	2	0.352	0
4 M	3.96	2	0.530	0
8 M	7.99	2	0.513	0
16 M	15.9	2	0.591	0
32 M	30.3	2	0.3	0
64 M	59.2	3	0.194	0
128 M	83.5	8	0.129	0.56

Table 3: Measuring the QoS of the network streams

2.1 Latency

Network latency increases slightly, which is normal, but when bandwidth drops from 64Mbps to 128Mbps latency is doubled.

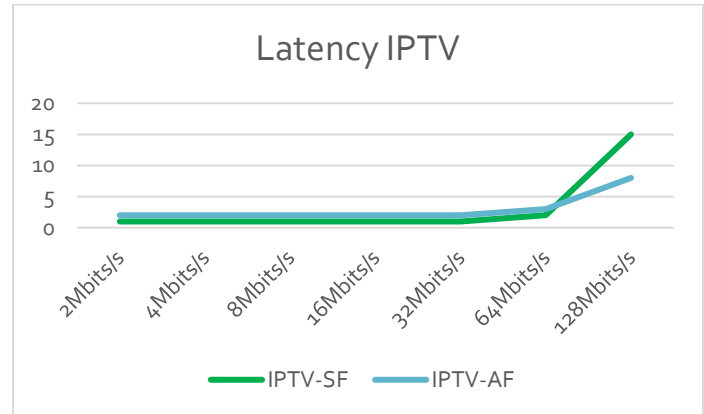


Figure 3: Latency variation curve
IPTV-SF measures without flows in the network.
IPTV-AF measurement with Network Flow

2.2 Jitter

The jitter, which represent variation of packets latency in the network. Network jitter increases when the network is loaded and decreases as the network load increases to 128Mbps.

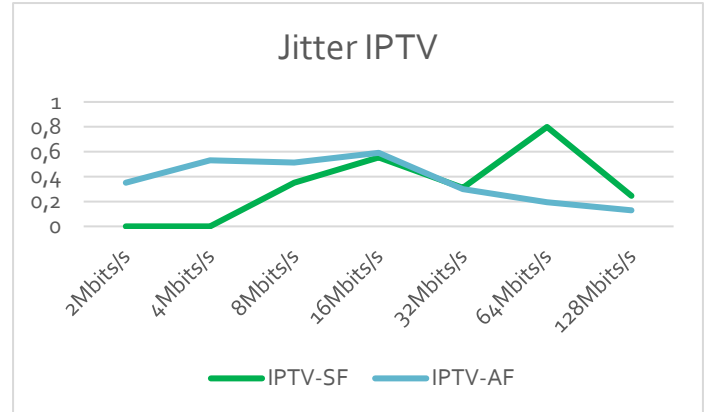


Figure 4: Jitter variation

2.3 Loss Rate

The loss rate is the number of packets that do not arrive correctly to their destinations.

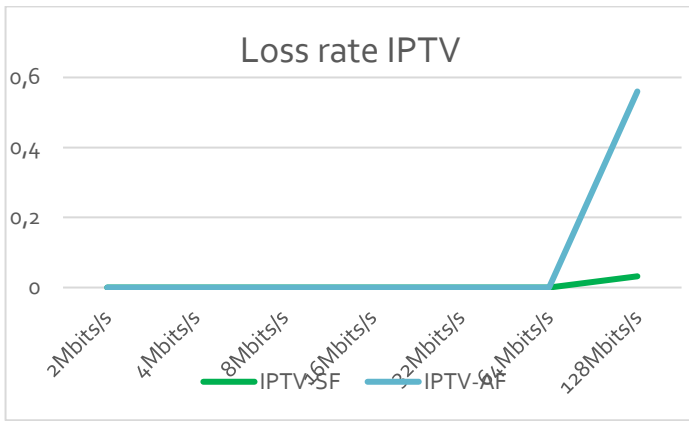


Figure 5: Loss Rate variation

3. VoD Service

Table 4 shows the network measurement results for the Video on Demand (VoD) service when the IP / MPLS network is not loaded by the services. Table 5 is the measurements when the network is loaded into services (IPTV, VoD and ToIP).

VoD streams (Mbits/s)	Bandwidth (Mbits/s)	Latency (ms)	Jitter (ms)	Loss Rate (%)
2 M	2	1	0.272	0
4 M	4	1	0.357	0
8 M	8	1	0.904	0
16 M	16	1	0.623	0
32 M	32	1	0.282	0
64 M	64.3	2	0.276	0
128 M	95.8	15	0.314	0.006

Table 4 : Measure of QoS in the network without streams

VOD streams (Mbits/s)	Bandwidth (Mbits/s)	Latency (ms)	Jitter (ms)	Loss Rate (%)
2 M	2	2	0.451	0
4 M	4	2	0.638	0
8 M	8	2	0.608	0
16 M	15.7	2	0.237	0
32 M	29.8	2	0.316	0
64 M	61.9	3	0.135	0.046
128 M	85.5	9	0.267	0.16

Table 5: Measure of QoS in the network with streams

3.1 Latency

For VoD server, latency drops as bandwidth increases to 128Mbps

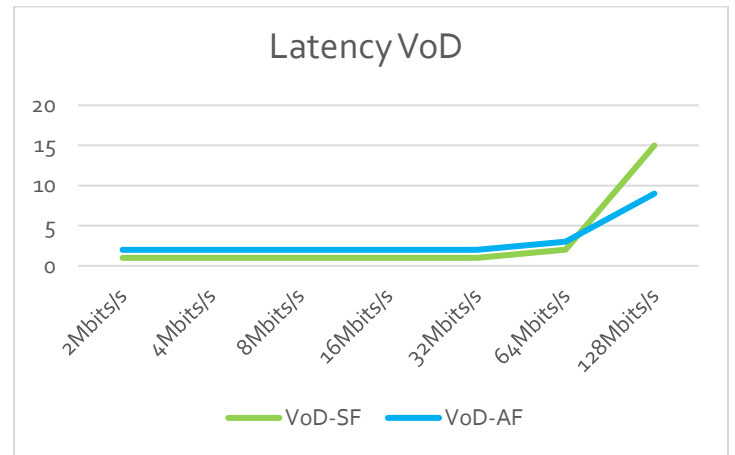


Figure 6: Latency variation

3.2 The Jitter

The jitter of the network with the flows decreases considerably

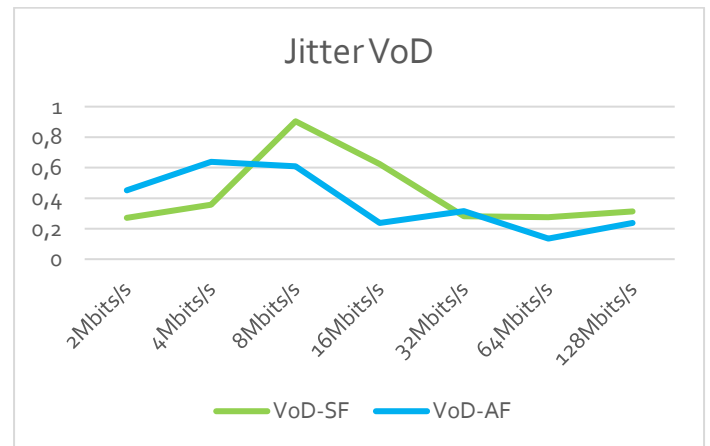


Figure 7 : Variation de la gigue

3.3 Loss Rate

The loss rate for the VoD service is zero up to 64Mbps. When the network is loaded the VoD loss rate is constant at first increases from 64Mbps.

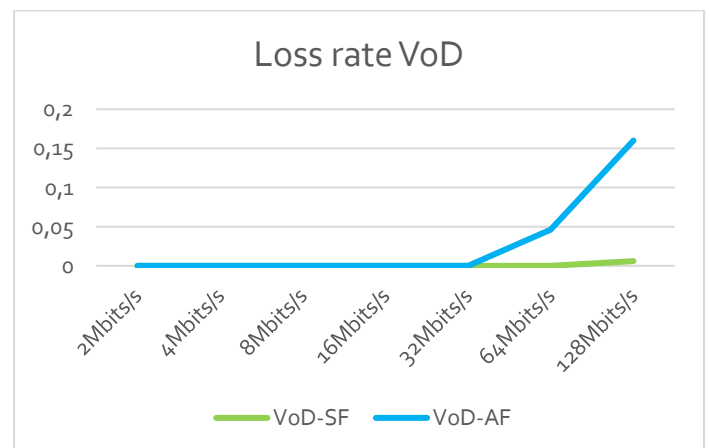


Figure 8: Loss Rate Variation

IV. OPTIMIZATION OF QOS PARAMETERS

1. Implementation of the QoS

From previous results we can deduce that a network optimization is required. To optimize the network we will implement the quality of service in our network through the diffServ protocol. Then identify the flows and determine in which classes of services they will be assigned, then perform the configurations of network equipment.

In our example, we implement three [3] service classes based on EF, AF and Best Effort behaviors. Here is the table to identify the classes we will create for our network. Priority will be given to all services to see the behavior of our multimedia services in the IP/MPLS network and we will take example for a single priority in the configurations.

We will give the highest priority to the IPTV service. The table below gives the different classes of services.

Type of traffic	CIR	PIR	Priority
IPTV	30000	50000	46
VOIP	2000	50000	38
VoD	30000	50000	36
Data	--	--	--

Table 6: classification of services by priority

For the implementation of the QoS, the configurations are done only on the router connected to the servers (CE1). The stages of QoS implementation are as follows [B5]:

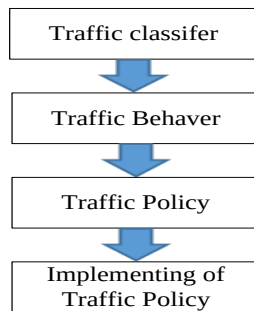


Figure 9: Implementation of QoS on services

2. Analysis of QoS parameters

The objective is to study the performance of our various services in the network by giving priority to each of our three services with the quality of service. To measure our services implemented in the network we will load all the streams of the different services on our client machines (5 IPTV streams, 5 VoD streams and calls), then use the Iperf tool to analyze the flows in our network. After installing the Iperf tool on the server and client, we execute the following command :

```
#iperf -s -u
#iperf -c 192.168.1.2 -u -b 4m
```

Measurements of bandwidth jitter and latency with UDP, generating all the streams on our different client bandwidth

machines of 2Mbps, 4Mbps, 8Mbps/s, 16Mbps/s, 32Mbps/s, 64Mbps/s, 128Mbps/s, 512Mbps/s et 1024Mbps/s.

3. QoS measurements with the IPTV service in priority

3.1 Jitter

The following table gives the measurement of the jitter when the IPTV service has priority.

	2M	4M	8M	16M	32M	64M	128M
IPTV (ms)	0.35 2	0.53 3	0.51 3	0.59 1	0.3 4	0.19 4	0.12 9
IPTV -QoS (ms)	0.70 7	0.69 1	0.53 8	0.59 9	0.35 8	0.27 7	0.33 6

Table 7: Measure of jitter

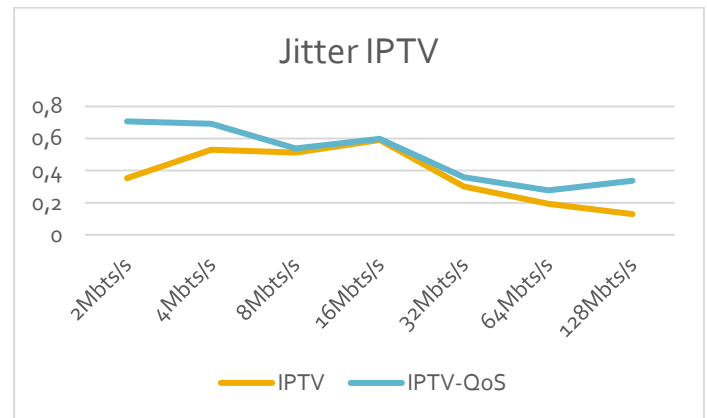


Figure 10: variation of the IPTV service jitter

We see a variation in jitter when IPTV packets are prioritized and when they are not prioritized. The highest jitter value for packets that are not IPTV types when IPTV packets are not prioritized is 16Mbps which corresponds to a jitter value of 0.591ms. And the highest jitter value of packets that are not IPTV types when IPTV packets are prioritized is 2Mbps which corresponds to a jitter value of 0.707ms. Jitter is a very important parameter for IPTV. Too high jitter values cause image degradation during streaming. In our case we do not have a degradation of the images which implies that the different values of the jitter remain nevertheless of very low value

The measurement of the jitter of the service VoD is on the table below:

	2M	4M	8M	16M	32M	64M	128M
VoD (ms)	0.451	0.638	0.608	0.2 37	0.3 16	0.1 35	0.2 37
VoD-QoS (ms)	0.622	0.764	0.532	0.5	0.3 42	0.3 12	0.2 71

Tableau 8: measure the jitter of the VoD service

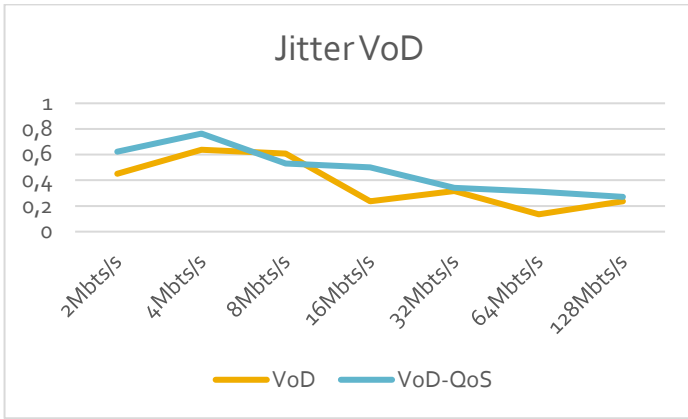


Figure 11: variation de la gigue du service VoD

We find that VoD jitter with QoS increases when IPTV service is prioritized.

3.2 Latency

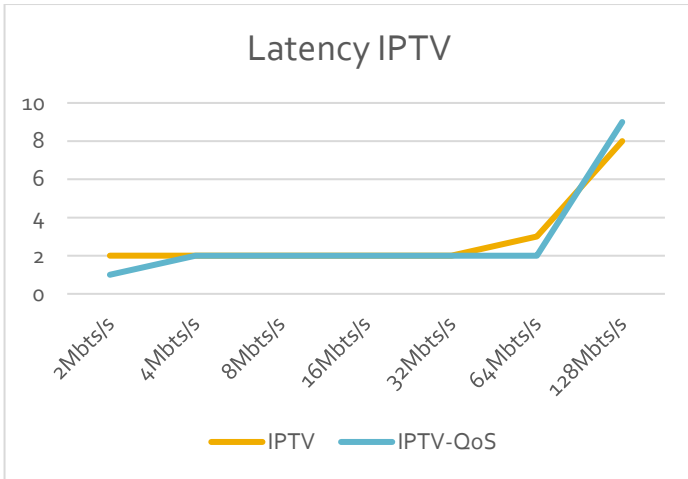


Figure 12: variation latency with priority IPTV

Latency decreases when IPTV service is prioritized. It becomes slightly larger between bandwidths of 32 to 64Mbps and decreases as bandwidth goes up to 128Mbps.

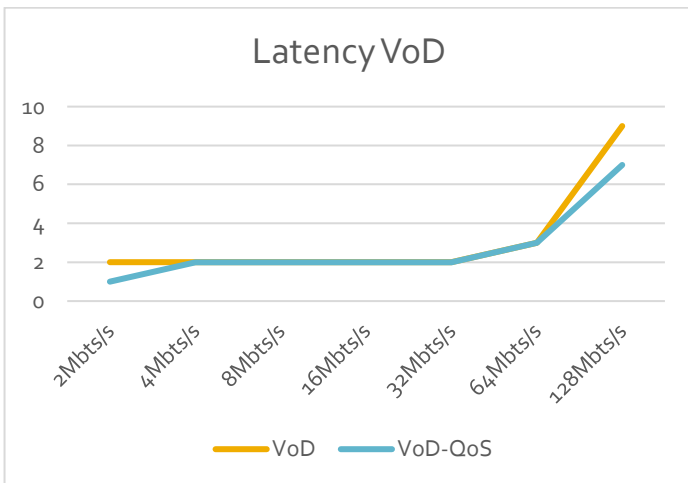


Figure 13: variation of VoD jitter with IPTV

VoD latency decreases when IPTV is a priority. We find that packages that are not IPTV types when IPTV packages are not prioritized will have a constant time display between 2MbtsVs to 32MbtsVs and there is an increase to 64MbtsVs. And those packages that are not IPTV type when IPTV packages are prioritized have a constant time display between 4MbtsVs to 32MbtsVs and there is an increase to 64MbtsVs.

3.3 Loss Rate

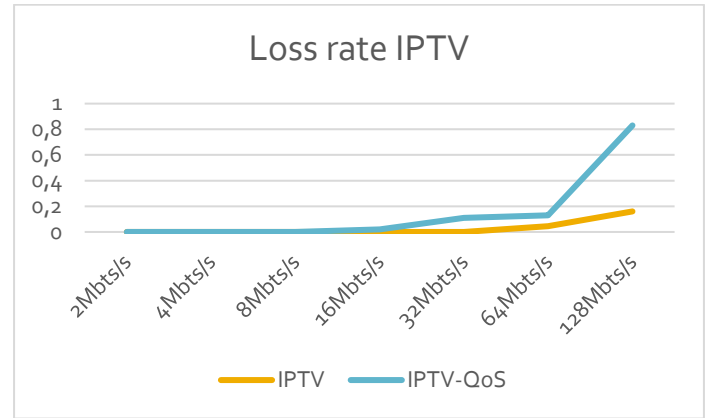


Figure 14: Loss Rate variation

The loss rate of packages that are not IPTV types when IPTV packages are prioritized will have a zero loss rate below 16MbtsVs, and above 16MbtsVs we see a large loss, which means up to 16MbtsVs there is no impact on the q images at reception. And the loss rate of packages that are not IPTV types when IPTV packages are not prioritized, have a slightly increasing loss rate.

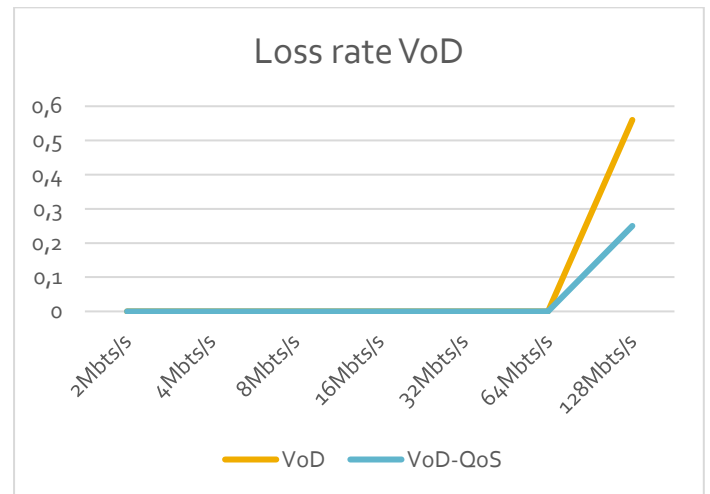


Figure 15: variation in the rate of loss of the VoD service

These curves show us a zero loss rate in the range of 2Mbts / s to 64Mbts / s and we find at 128Mbts / s a high loss rate. These losses thus cause images with missing pixels, or images broadcast in saccade.

V. CONCLUSION

Multimedia technologies are evolving and the IP network has become a multimedia and international network and operators and service providers believe the emergence of new multimedia services that must be provided with quality of service across equipment network. Convergence requires a new network technology that is MPLS to meet the needs of QoS applications namely bandwidth management, jitter, loss rate and latency.

We have implemented IPTV (multicast), VoD and ToIP services. Using analytics tools such as Wireshark and Iperf, we did a study of the different flows from these services. The results of this analysis show that the network being optimal requires the implementation of the QoS because the end user of these services waits for his content provider to offer quality services.

Quality of service estimation has always played an important role in the design and design of network architectures. With the convergence of data, voice and video over IP it is clear that an approach is needed to optimize the use of resources in order to limit investment costs, guarantee quality of service and high availability. MPLS technology by providing its new services such as quality of service management introduces the concept of service class and DiffServ which allows routing of different multimedia services taking into account their priority. Also, through Traffic Engineering, traffic is routed along a tunnel, making transmission even more efficient.

The results obtained clearly show the importance of a guarantee of QoS and its reaction to the receipt of information. However, there is no public offering to benefit from QoS. This may be due to the lack of a business model associated with DiffServ. The fact that DiffServ offers a large number of parameters to adjust in each device and that the policy must be global makes its complex deployment in a network.

Today, the use of multimedia services such as VoD, IPTV, Streaming audio, Voice over IP is growing. The end user of these services waits for his content provider to offer him quality services. It is in this context that we decided to conduct our study. The objectives were achieved. It has been a very rewarding experience and a real opportunity to put into practice a body of knowledge acquired during our training.

Multimedia networks are evolving towards processing and configuration automation through a new architecture that is SDN (Software-Defined Networks) and existing protocols. SDN has been able to bring the support of equipment manufacturers, network administrators and will no doubt profoundly change the business of the network administrator in the years to come.

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