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Research Article

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IP/MPLS Network analysis of multimedia QoS flows in network

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Abstract The technology used in packet-switched telecommunications networks is constantly evolving .We are in a phase of development and massive use of multimedia services such as Voice over IP, IPTV, VoD and web radio on the Internet. With the convergence of these services on the same network, it is important to control and manage the quality of service. QoS is a crucial factor for the success and use of these multimedia services. With MPLS technology in multimedia service networks, services can be provided thanks to a service differentiation model (DiffServ) for real-time streams while maintaining quality of service requirements. That said, ESMT welcomed us as seekers and allowed us to work in that direction. Therefore, the aim of this work is to first study the above multimedia services and the MPLS network on which they can be deployed, and then to carry out an analysis of the behavior of each service and its influence on the other services in the IP/MPLS network. Network we set up and after this analysis make a proposal to optimize the QoS.

Keywords Multimedia services, IPTV, Voice over IP, QoS, MPLS, optimization, DiffServ, Flux

1. Introduction

In the last decades we have seen exponential growth in the ICT sector. Numerous studies and research are carried out in order to continuously improve the related services and bring innovations in this field. Multimedia therefore benefits from these advances, making it possible to offer a variety of services and keep the customer base happy. However, all this has consequences: to ensure the best quality in reception, and given the growing multimedia traffic over IP, networks must have a large capacity in transmission the quality of transmission in its networks, since these are indicator factors of network performance. This performance is characterized by packet loss, delay, jitter and reception quality perceived by the multimedia user. It also needs to be able to define priority rules in its network so that during congestion, packets are routed in order of priority and with the lowest possible loss rate. From this perspective, we decided to work on this topic to find solutions to improve the quality of service in any type of network.

2. Material and methods

2.1 Examination of services and network protocols for the methods

2.1.1 IPTV

IP television or IPTV (Internet Protocol Television) is the delivery of programs per video data stream encoded as a series of IP packets. TV programs are routed to receivers (TVs, tablets, computers) through a broadband connection band, rather than being delivered via cable distribution or traditional broadcasting. IPTV uses MPEGTS which is a standard for transmission, multiplexing and synchronization audio and video.

Video and program channels are delivered to receivers via a broadband connection. band (2 to 8 Mbps), instead of being delivered by cable and broadcast formats classics. Video streams are encoded into a series of Internet Protocol packets and then routed through the Internet to be received by anyone with a set-top box and a subscription for the service. IPTV is usually bundled or bundled with VoIP and access to

Internet: it can thus be described as a "Triple Play" service. Thus, an IP television service standard is a complete package that allows customers to watch TV, browse through

Internet and make long distance calls using VoIP.

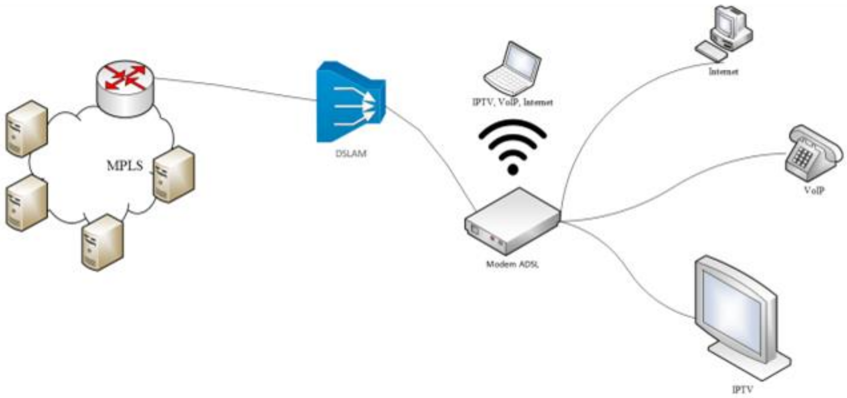


Figure 1: IPTV network topology

IPTV is provided by the service provider using a closed network infrastructure. The flows pass through an ATM or MPLS network before arriving at the DSLAM to be then directed to the client (decoder, IP television or PC). A decoder, commonly called "Set Top Box" or STB connected to a television allows the user to choose the programming to be using interactive menus. STN can be a separate unit, or a computer equipped with appropriate software (multimedia station).

2.1.2 VoD

VoD is a technology for transmitting digital video content. The VOD service is offered or sold over wired networks such as the Internet or wireless networks such as 3G or 4G telephony. Video-on-demand has been developing since the early 2000s following the operation of broadband access offered to individuals. VoD uses the principleStreaming, a principle based on unicast broadcasting, is a logical progression from broadcast technologies such as pay-per-view (pay-per-view). More flexible for customers who are not dependent on airtime. Using a digital decoder or computer, the user can order movies or TV shows stored on servers. The user has a predetermined rental time (usually 24 hours) for the movie or show they have ordered. This service is made possible by the development of broadband.

2.1.3 Webradio

A web radio is a radio station specifically designed for web broadcasting. It is a computer installation that, thanks to streaming technology, allows radio broadcasts on the Internet. The idea of web radio is to combine the specific characteristics of radio with others offered on the web to have a new one. The web radio works in a client-server model, it allows you to connect to the server and continuously broadcast data. Once the server is started and the internet radio is connected, clients can connect to the server and access the stream.

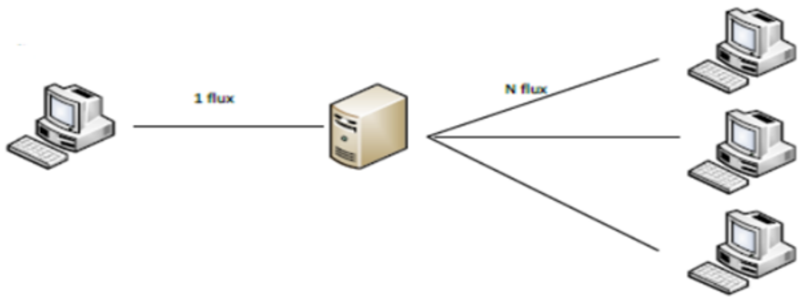


Figure 2: Webradio network topology

2.1.4 Voice over IP

Voice over IP is an alternative to the dial-up telephone system. It enables the routing of digital voice signals in IP networks. The technology enables voice, data and video to converge on a single network that is now more reliable and accessible.

In general, an IP telephony network mostly includes terminals, a router, a gateway to other networks, a gatekeeper and server communication. terminals: These can be hardware types (hardphone) or analog phones equipped with an ATA box, or even PCs equipped with a softphone. The gateway (gateway) / the gatekeeper: The gateway, in Telephony IP, is a computer that provides an interface where the convergence between the switched telephone network (PSTN) and networks occurs based on the switching of TCP/IP packets. It is an integral part of the telephony network architecture IP. The gatekeeper is the element that provides information to the gateway. Depending on the architecture, we can separate the hardware and software parts of a pedestrian bridge.

The gatekeeper thus corresponds to the software component and the gateway to the hardware component.

The IPBX can be defined as a private telephone exchange that uses Internet Protocol (IP) and manages calls between its customers who.

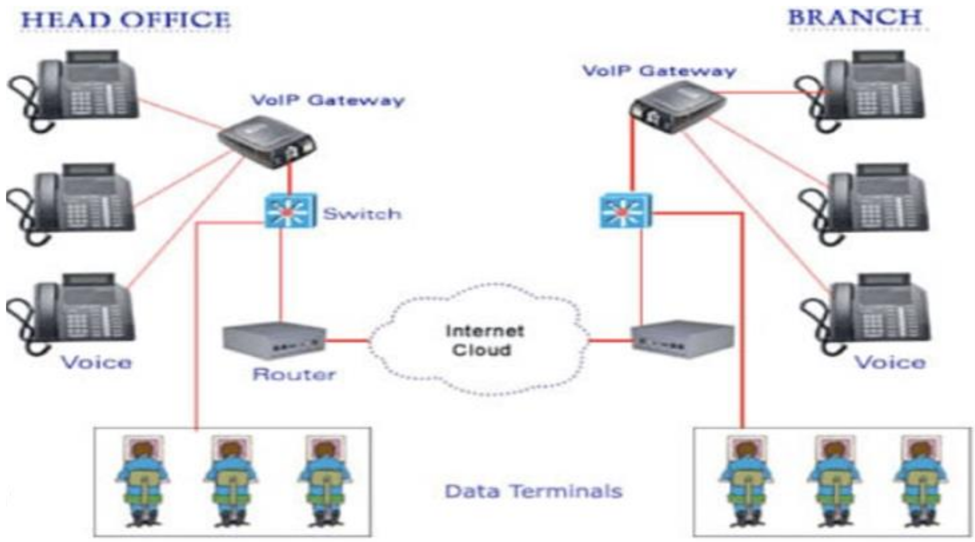


Figure 3: VoIP network topology

2.1.5 MPLS concept

MPLS for MultiProtocol Label Switching was born from the founding of a working group at the IETF in spring 97. In fact, MPLS is a protocol standard proposed by the IETF whose main purpose is to combine the concepts of level 3 IP routing and level 2 switching mechanisms as implemented in ATM or Frame Relay; MPLS is therefore level 2.5. One of the original goals was to increase the speed of datagram processing in all intermediate devices. With the introduction of giga routers, this will has now receded into the background.

The architecture of an MPLS network is composed of two points: the control plane and the data plane.

The control plane: controls level 3 routing information thanks to protocols such as (OSPF, IS-IS or BGP) and labels using protocols as (LDP, MP BGP or RSVP) exchanged between adjacent devices.

The data plane: is independent of the routing and exchange algorithms of label Use of a base called Label Forwarding Information Base (LFIB) to forward packets with the correct labels. This base is filled by the label exchange protocols.

In MPLS domain we have following terminologies:

LSR (Label Switching Router): router or switch capable of forwarding labeled packets and implementing label distribution;

LER (Label Edge Router): resides at the edge of the MPLS network, inserts or removes the LSPs that are included in each packet that occurs at the edge of the MPLS network;

LSP (Label Switching Path): is a series of labels starting from the source and going to the destination, and is unidirectional. LSPs are established prior to the transmission of data or the detection of a flow wishing to traverse the network.

FEC (Forwarding Equivalence Class): This is a set of IP addresses with the same address prefix and/or class of service. All packages from one class are treated equally. They have: the same label, the same exit interface, the same NextHop, the same queue (QoS) at the entrance of the network.

Label: MPLS makes it possible to associate an FEC with a packet for its entire traversal of the network. This association is determined at the LSR ingress based on the domain management policy information contained in the packet. For each link, this association is identified by a short and fixed value: the label.

LDP (Label Distribution Protocol): defines a set of procedures and messages used by LSRs to inform each other about the mapping between labels and the flow. LDP is bi-directional and allows dynamic discovery of neighboring nodes.

2.1.6 Diffserv: QoS

The principle of DiffServ is to differentiate flow aggregations grouped by classes of services (real-time, transactional or "best effort" for example). DiffServ does not use resource reservation mechanisms.

In the differentiated service model, the application does not need to send its request fornetwork resource reservation before sending packets. Instead, the appinforms network nodes of its quality of service requirements by defining the quality of service parameters in the IP header. So the routers along the way get the request by parsing the packet header. DiffServ relies on the DSCP (DiffServ Code Point) field of the IP header.

The DSCP field is defined by edge routers. This field is used to indicate to LSRs the behavior they should adopt according to its value. This behavior is called the Per Hop Behavior.

A code point is used to select a PHB service class composed of 2 parts:

Class selector: class definition;

Drop precedence: priority with which packets are dropped. 3 types of PHBs are defined:

AF (Assured Forwarding): for flows requiring bandwidth limited. Excess traffic is discarded progressively according to a mechanism of priority based on 4 classes of 3 rejection priorities each;

EF (Expedited Forwarding): (DSCP = 101110). For flows requiring a guaranteed bandwidth, with low jitter, latency and loss;

CS (Class Selector): for compatibility with the IP Precedence field (CS0/CS1/CS2/CS3/CS4/CS5/CS6/CS7).

2.1.7 Diffserv: Implementation

To implement Diff-Serv, the access router classifies the packets and marks the class of service (CoS) in the IP packet header. The downstream routers then identify the CoS and forward the packets based on CoS. Diff-Serv is therefore a solution based on QoS classes.

The main technologies used to implement DiffServ are:

Traffic classification: allows objects to be identified according to specific rules. It's here prerequisite of DiffServ which is used to identify packets according to defined rules.

Traffic policing: controls traffic rate. The rate of traffic entering the network is monitored and traffic exceeding the predefined limit is blocked. This allows to optimize the use of network resources and protects the interests of service providers.

Congestion management: manages resource allocation during network congestion. It first stores the packets in the queue, and then uses a dispatch algorithm to determine the packet transmission sequence.

Congestion avoidance: it monitors network resource usage, and automatically drops packets case of heavy congestion. This solves the problem of network overload.

Traffic shaping: its automatically adjusts outbound traffic volume based on network resources which can be allocated by the downstream router to prevent packet loss and congestion.

3. Results & Discussion

The result of our work has been tested in this network topology

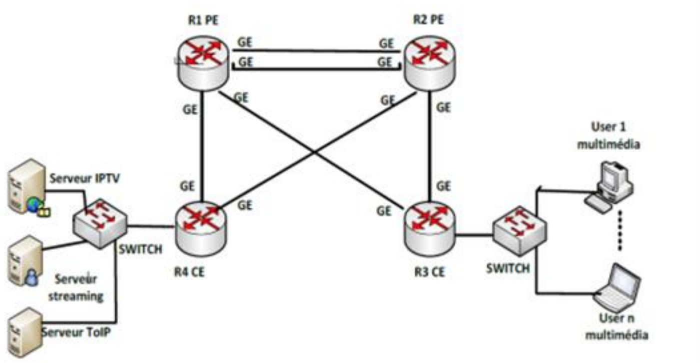


Figure 4: Target network topology

The Pes in this figure are AR2220 type routers, and the CEs of AR1220 are also type of routers. We therefore obtain the following topology on tool.

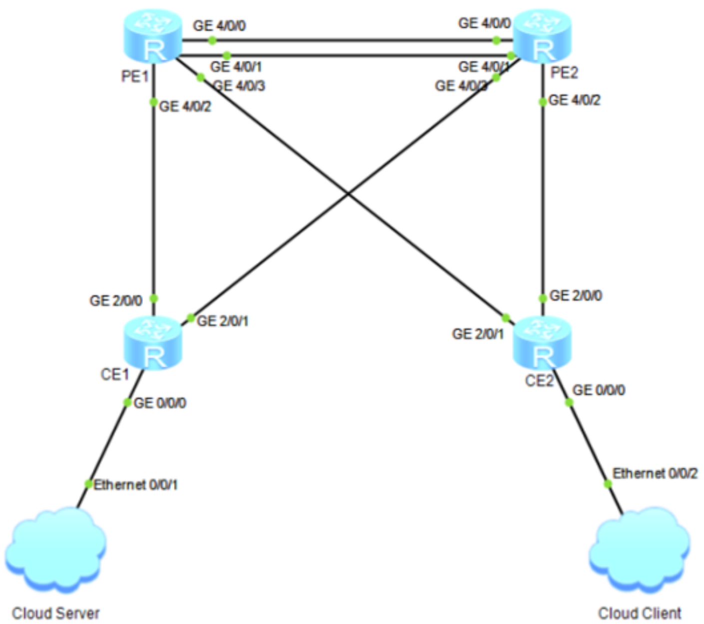


Figure 5: Summary cloud target network topology

Table 1: IP address plan of network

Router	Interface	@IP
PE1	G4/0/0	8.8.8.1/30
	G4/0/1	8.8.8.5/30
	G4/0/2	7.7.7.1/30
	G4/0/3	7.7.7.5/30
	Loopback	1.1.1.1/32
PE2	G4/0/0	8.8.8.6/30
	G4/0/1	8.8.8.1/30
	G4/0/2	7.7.7.9/30

	G4/0/3	7.7.7.13/30
	Loopbac0	2.2.2.2/30
	G2/0/0	7.7.7.2/30
CE1	G2/0/1	7.7.7.14/30
	G0/0/0	192.168.1.1
	Loopback	3.3.3.3/32
	G2/0/0	7.7.7.10/30
CE2	G2/0/1	7.7.7.6/30
	G0/0/0	192.168.2.1
	Loopback0	4.4.4.4/32
Serveur Asterisk		192.168.1.5/24
Serveur IPTV and VoD		192.168.1.4/24
Serveur Webradio		192.168.1.6/24

3.1 Analysis of parameters and Optimization of the QoS

When setting up the network, it is important to know and change the indicators that act in the QoS in order to know the impact that occurs in relation to these indicators. Studying the quality of service of this network consists in measuring the impact of each service operating on the network. That is, depending on the service, we need to determine the occupied bandwidth, fluctuations in jitter, latency and loss rate. This lets us know how each service is impacting the network and allows us to prioritize specific services in the event of network congestion.

Analyze were performed on a server PC and a client PC with Wireshark tools 2.2.7 (Windows) and Iperf 3.1 (Linux).

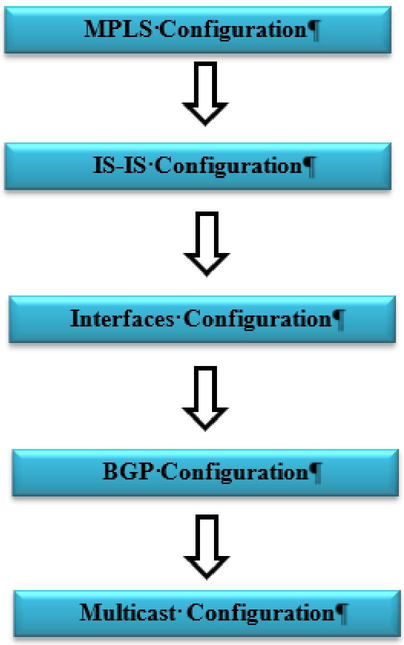
Iperf: Is a network performance measurement software available on many platforms (Linux, Windows) developed by the National Laboratory for Applied Network Research. It comes in the form of a command line that runs on two computers (client/server) located at the ends of the network. Iperf measures bandwidth, latency, jitter and packet loss. The first machine starts in Iperf inserver mode (with the -s option), the second in client mode (the -c option).

Iperf is installed on Linux with the apt-get install iperf command.

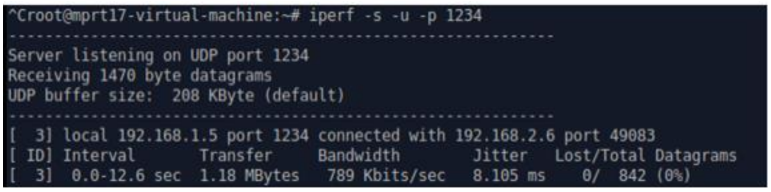
Wireshark: Is an open source network protocol analyzer. Its goal is to capture frames that reveal security vulnerabilities, see performance on the network.

The tool can be used on multiple platforms.

Network configuration is done using the following configuration steps:



3.2 Behavior of services without QoS



```
^Croot@mprt17-virtual-machine:~# iperf -s -u -p 1234
------------------------------------------------------------
Server listening on UDP port 1234
Receiving 1470 byte datagrams
UDP buffer size:  208 KByte (default)
------------------------------------------------------------
[  3] local 192.168.1.5 port 1234 connected with 192.168.2.6 port 49083
[ ID] Interval       Transfer     Bandwidth        Jitter   Lost/Total Datagrams
[  3]  0.0-12.6 sec  1.18 MBytes   789 Kbits/sec   8.105 ms    0/  842 (0%)
```

Screenshot 1: Analysis of an IPTV stream with Iperf

The results of the analysis allowed us to have the following curve after put the settings in tool:

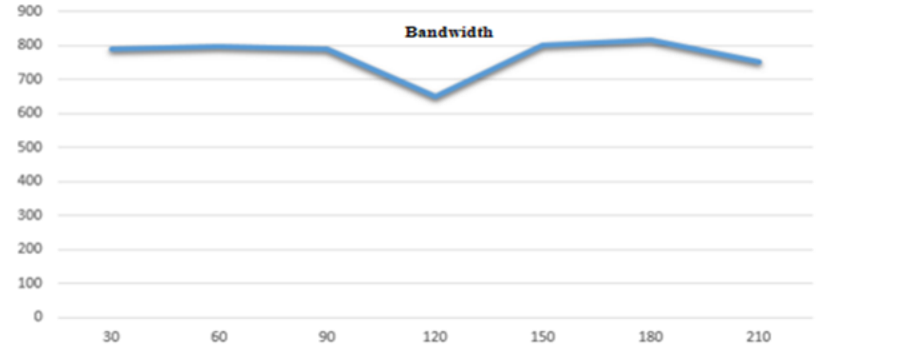


Figure 6: IPTV bandwidth

Reading of result N°01

We see a small variation in bandwidth over time. After a few minutes, the maximum bandwidth observed on this channel is **801 Kbps**.

If the bandwidth has average values, then this is beneficial for other network services. However, it results in poor image quality during transmission.

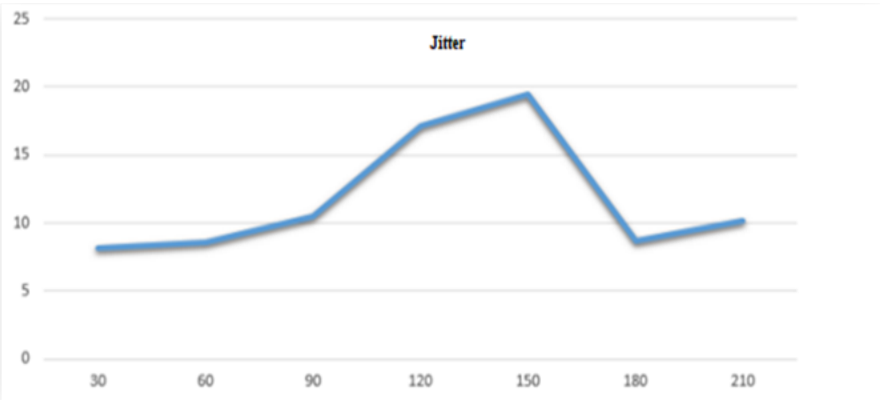


Figure 7: IPTV Jitter

Reading of result N°02

We see strong jitter values. Jitter is a very important parameter for IPTV.

Jitter values that are too high lead to image degradation during diffusion. If packages do not arrive on time, they are considered lost. Lost packets are predicted and replaced. But often these predicted packets are wrong. This is noticeable in the images as pixels are missing or out of place.

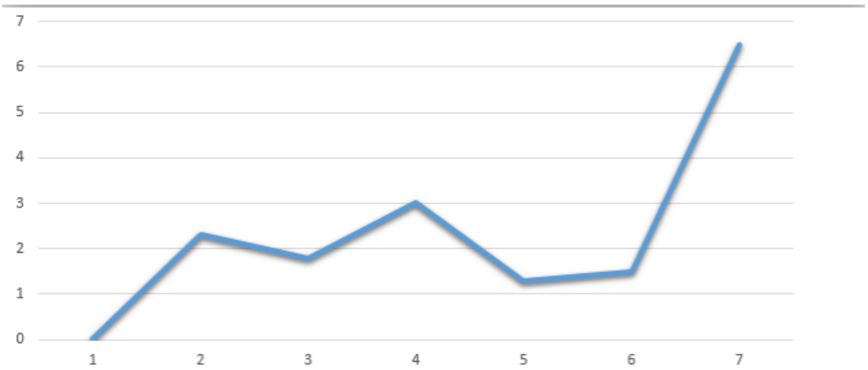


Figure 8: IPTV packet loss

Reading of result N°03

This graph shows us a loss rate of less than 10% and with frequent fluctuations.
This can be due to a high processor load or too much waiting time.
These losses therefore lead to images with missing pixels or jerky transmission of images.
By analyzing the network with Wireshark, we understood that:
Video streams can occupy a maximum bandwidth of 4 Mbit/s to almost 9 Mbit/s.
Achieving these values results in sharper images with the pixels in their place.
However, these values are quite high considering the other services present in the network and therefore affect the other services.

3.3 Disruption of flows

We will now interrupt our IPTV streams by creating another stream using Iperf according to the following syntax:
iperf s u;
iperf c 192.168.1.4 -u -b 1m: 192.168.1.4 corresponds to the IP address of ours Servers,
b 1m to generate a stream of 1 megabits per second.
At the same time, we measure the performance of the network using the following commands:
iperf s u;
iperf c 192.168.1.5 u.
We get the following result:

```
UDP buffer size:  208 KByte (default)
------------------------------------------------------------
[  3] local 192.168.1.5 port 5001 connected with 192.168.2.15 port 60192
[ ID] Interval       Transfer     Bandwidth        Jitter   Lost/Total Datagrams
[  3]  0.0-30.1 sec   932 KBytes   254 Kbits/sec  22.724 ms    6/  655 (0.92%)
[  4] local 192.168.1.5 port 5001 connected with 192.168.2.15 port 40886
[  4]  0.0-31.4 sec  1.83 MBytes   490 Kbits/sec  15.903 ms    0/ 1308 (0%)
```

Screenshot 2: IPTV disruption flows

By varying the flow rate of generated traffic from 256Kbps to 10Mbps, we have the following behavior:

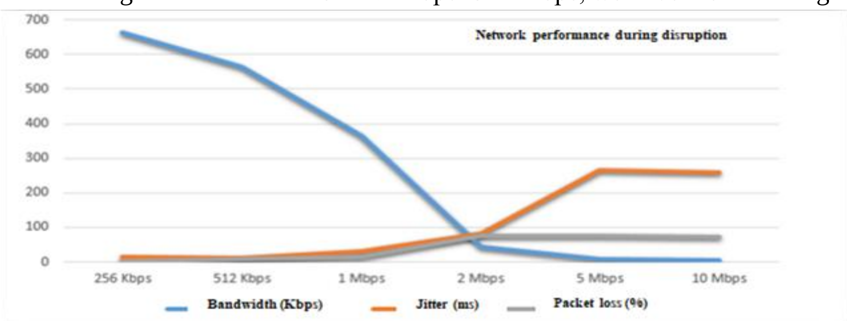


Figure 9: Variation of bandwidth jitter and packet loss rate after disturbance

Reading of result N°04

This curve shows us the impact of the interference generated on video streams.

In addition, more the network bandwidth decreases, the larger the band interference bandwidth.

This therefore means a high loss rate of up to 75%.

The generation of streams with Iperf therefore disrupts the video streams already present in the network and degrades the image quality during streaming

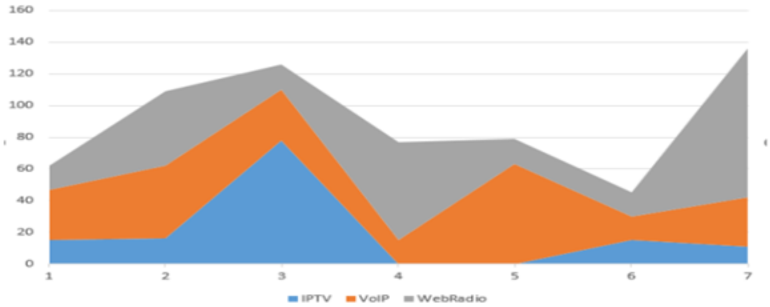


Figure 10: Latency

Reading of result N°05

Internet radio has, on average, higher latency compared to other services.

Nevertheless, the latencies observed with the other services explain the delays observed during their use

5. Key Contributions

Use IPv6 addressing;

Implement the different services on an MPLS network with multicast VPN;

Implementation of monitoring server to progressively see the behavior of the various services in the network, and to be informed in the event of saturation.

6. Conclusion

Today we are faced with a heavy use of multimedia services and a higher demand for quality of these services. To do this, it is important to manage your network. The assessment of the quality of service has always occupied an important place in the conception and design of network architectures. With the convergence of data, voice and video over IP, an approach to optimizing resource utilization is required to achieve a good balance between quality of service and cost.

The MPLS technology, through the contribution of its new services such as the management of the quality of service, introduces the concept of the service class and the DiffServ, which allows the routing of the various multimedia services taking into account their priority.

Acknowledgements

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