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Performance Evaluation of Real Time Traffic over MPLS Networks under Different QoS Conditions

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Abstract: One of the most crucial problems in the Internet has been the quality of service (QoS) provisioning. Multiprotocol Label Switching (MPLS) technology guarantees real time and multimedia applications QoS using different resource allocation techniques. Also MPLS contributes high scalability in data network. This paper aims to evaluate MPLS performance based on multimedia service average throughput, total number of packets received, end to end delay, jitter, and packet loss ratio using OPNET simulator. It also compares MPLS network performance to that provided by IP networks. This study shows the scalability of MPLS by simulating small and large networks under different loading conditions. The simulation also shows the performance of different MPLS QoS configurations.

1. Introduction

Internet traffic is exploding as it is doubling every few months and the speed of technology is doubled every two years. Emerging multimedia applications and real time applications will make the explosion faster; moreover multimedia and real time applications require timing and other QoS guarantees, besides bandwidth [1]. Traditional Internet Protocol (IP) networks provide only best effort service which is unacceptable for real time applications. Providing QoS means the ability to provide different guaranteed services based on the applications requirements. QoS provisioning is typically based on end-to-end mechanisms (e.g., connection admission control), edge mechanisms (e.g., shaping and policing), core mechanisms (e.g., buffering, queue management, and scheduling), or any combination of the three. Given that the traffic behavior has gone through remarkable changes the last few years, there is a need for new classification and scheduling techniques that takes into account those changes and offer efficiently QoS [2]. The problem with common generic IP is that it only provides point-to-point connectivity, operates on a first come- first-served basis, and is subject to variable and unpredictable queuing delays as well as congestion losses. Also, IP can’t allocate band-width on a particular link to applications with different performance requirements which is unacceptable for applications such as multimedia and real time. MPLS has emerged as an elegant solution to meet service requirement for next generation IP networks and to offer the required requirement for multimedia and real time traffic. Several studies focused on real time application QoS over different types of networks like ATM, MPLS, and IPv6. QoS of MPLS has been introduced in [3] but without simulation. In [4] the end to end delay is the only QoS parameter discussed, also the number of network nodes is limited as for example in [5].

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The remaining of this paper is organized as follows: Section 2 provides a review of multimedia applications and QoS requirements. Section 3 and 4 give brief information about performance measures and MPLS respectively. The simulation results are introduced in section 5 while section 6 concludes the paper.

2. Multimedia Applications and Quality of Service

QoS policing and management functions control and handle end-to-end traffic across the network. Traffic in a network is made up of flows originated by a variety of applications on end stations. These applications differ in their service and performance requirements. Hence, understanding the application types is a key to understand the different service needs of flows within a network. The network's capability to deliver service needed by specific network applications with some level of control over performance measures—that is, bandwidth, delay/jitter, and loss—is categorized into three service levels: Best-effort service, Differentiated service and guaranteed service. Guaranteed service requires prior network resource reservation over the connection path. Table 1 shows that different user applications have different QoS requirements [6].

Table 1 Applications QoS requirements

<table>
<tr><th>Voice</th><th>Data</th><th>Video</th><th>Interactive Video</th></tr>
<tr><td>Low delay</td><td>High delay</td><td>Higher delay than voice</td><td rowspan="3">Requires low delay and low loss for control but can accept greater delay variation for control</td></tr>
<tr><td>Low delay variation</td><td>High delay variation</td><td rowspan="2">Data loss may have noticeable effects</td></tr>
<tr><td>Tolerant to some data loss</td><td>Tolerant to varying bandwidth</td></tr>
</table>

3. Performance Measures

QoS deployment intends to provide a connection with certain performance bounds from the network. Bandwidth, packet delay and jitter, and packet loss are the common measures used to characterize a connection's performance within a network. Transmission time includes delay due to codec processing as well as propagation delay. ITU-T Recommendation G.114 recommends the following one-way transmission time limits for connections with adequately controlled echo (complying with G.131) [7]:

- 0 to 150 ms: acceptable for most user applications;
- 150 to 400 ms: acceptable for international connections;
- 400 ms: unacceptable for general network planning purposes.

3.1. The E-Model

The E-model defined in the ITU-T Rec G.107 [8] is an analytical model of voice quality used for network planning purposes. A basic result of the E-Model is the calculation of the R-factor which is a simple measure of voice quality ranging from a best case of 100 to a worst case of 50. The R-factor uniquely determines the Mean Opinion Score (MOS), which is the arithmetic average of opinion of voice quality as shown in the next table [9]:

Table 2 MOS value

R-factor	Quality of voice rating	MOS
90<R<100	BEST	4.34 – 4.5
80<R<90	HIGH	4.03 – 4.34
70<R<80	MEDUIM	3.6 – 4.03
60<R<70	LOW	3.1 – 3.6
50<R<60	POOR	2.58 – 3.1

4. MPLS

MPLS flows are connection-oriented and packets are routed along paths pre-configured by service providers called LSP (Label Switched Paths). Ingress routers at the edge of the MPLS network classify each packet potentially using a range of attributes, not just the packet's destination address, to determine which LSP to use. Inside the network, the MPLS routers use only the LSP labels to forward the packet to the egress router. A different label is used for each hop, and it is chosen by the router or switch performing the forwarding operation. MPLS uses two methods for choosing the LSP for an FEC (route selection). They are hop by hop routing and explicit routing. MPLS QoS is an important component of MPLS. In an MPLS network, QoS information is carried in the label header's MPLS CoS field .MPLS uses the same IP QoS functions to provide differentiated QoS for traffic within an MPLS network. Multiple LSPs in parallel can be established through Label Distribution Protocol (LDP) to support traffic with multiple precedence values. Each established LSP is mapped to carry traffic of certain MPLS CoS values [10].

4.1. The QoS Toolset

In practical terms, QoS involves using a range of functions and features (e.g. classification, scheduling, policing, shaping). The mechanisms used for engineering the QoS in a network can be broken down into data plane and control plane mechanisms applied on network devices such as routers. In Data plane, QoS is applied at network nodes and can directly impact the forwarding behavior of packets. This plane includes classification, marking, prioritization and maximum rate assurance. Control plane QoS mechanisms deal with admission control and resource reservation. Control plane QoS functions are implemented as software processes, along with other control plane functions such as routing protocols [11].

4.2. MPLS traffic engineering

MPLS traffic engineering allows constraint-based routing of IP traffic. One of the constraints satisfied by CBR is the availability of required bandwidth over a selected path. Diff-Serv-aware Traffic Engineering extends MPLS traffic engineering to enable performing constraint-based routing of “guaranteed” traffic. Assuming QoS mechanisms are also used on every link to queue guaranteed traffic separately from regular traffic. This is essential for transport of applications that have very high QoS requirements. The MPLS traffic engineering Internet Protocol explicit address exclusion feature provides a means to exclude a link or node from the path for an MPLS traffic engineering label switched path (LSP) [12]. One of the goals of MPLS traffic engineering is to guarantee bandwidth reservations for different service classes. For these goals two functions are defined: Class-type (CT) and bandwidth constraint (BC). For the mapping between BCs and CTs the maximum allocation model (MAM), max allocation with reservation (MAR), and Russian dolls model (RDM) are used [13].

5. Simulations

This section shows the impact of MPLS in small and large networks, light and heavy loaded networks and the effect of applying QoS to MPLS network using OPNET simulation. All networks traffic includes voice, video and data. The simulation time is taken 30 seconds, then, the performance of the voice and video for the IP and MPLS networks are compared.

5.1. Small Network

This network consists of three LANs. Each of them is connected to the core network through edge routers LER 1, 2, and 3 consequently (figure 1). The core network consists of six LSRs (LSRs1to LSR 5) and all links between routers are DS3 with no background traffic while all LANs links are Ethernet 100 base T duplex links. This network contains voice, video, and data traffic (table 3).
LAN 1 consists of seven workstations (V11- V12- V13- Video 11- Video 12 – FTP – FTP 2).
LAN 2 consists of five workstations (V21- V22- V23- Video 21- Video 22) and FTP server.
LAN 3 consists of five workstations (V41- V42- FTP 4)

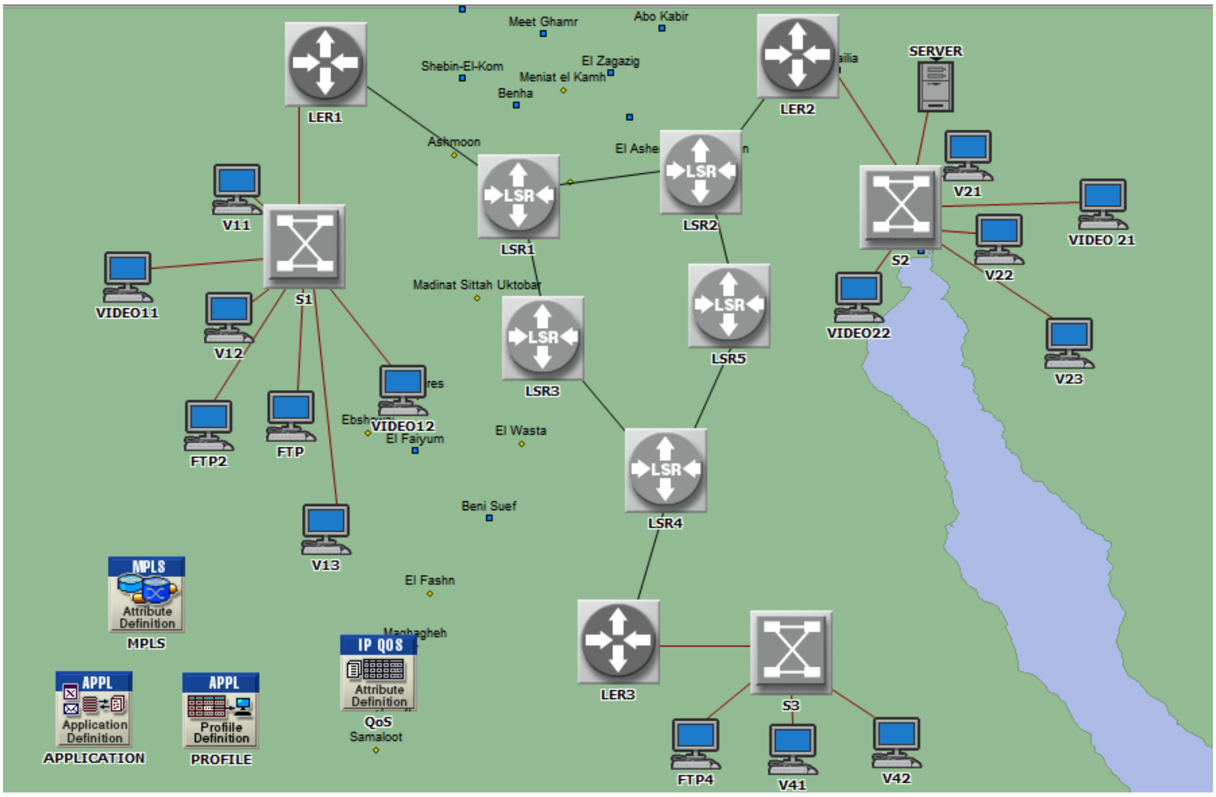


Figure (1) Small network

Table 3 Types of scenario traffic

Traffic type	Workstation name	Description	Profile duration	Repeatability
VoIP	V11-V12-V13-V41-V42-V21-V22-V23	PCM Quality speech	End of profile	Unlimited
VoIP2	V12-V13-V41-V42-V22-V23	PCM Quality speech	Constant (50)	Constant (55)
FTP	All FTP workstations	High Load (50KB file size)	Constant (10)	Exponential (30)
VIDEO	All video workstations	High Resolution (15f\s) 128 x 240 pixel	End of profile	Unlimited

a) Results for Video conference

Figures 2, 3 and 4 show the packet delay variation, end to end delay (in seconds) and the received traffic in bytes for video traffic. (The blue curve represents IP while the red one represents MPLS). For video communication, IP has about 60 ns packet delay variation (neglected variation) and 21 ms end to end delay which is acceptable (less than 150 ms) and almost all packets was received while MPLS has 40 ns packet delay variation (neglected variation) and 22 ms end to end delay almost all packets were received which is acceptable too.

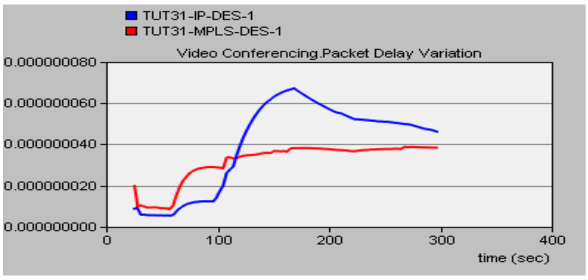


Figure (2) Packet delay variation

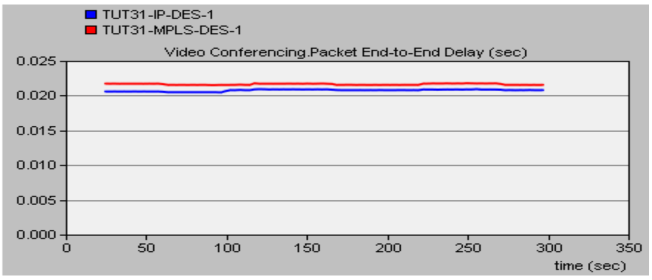


Figure (3) End – to – end delay

Figure (4) Traffic received

b) Results for Voice

Figures 5, 6 and 7 show the packet delay variation, end to end delay (in seconds) and the received traffic in bytes for voice traffic. (The blue curve represents IP while the red one represents MPLS). For voice communication, IP has about 25 µs packet delay variation and 62 ms end to end delay which is acceptable while MPLS has 30 µs packet delay variation and 62 ms end to end delay which is acceptable too.

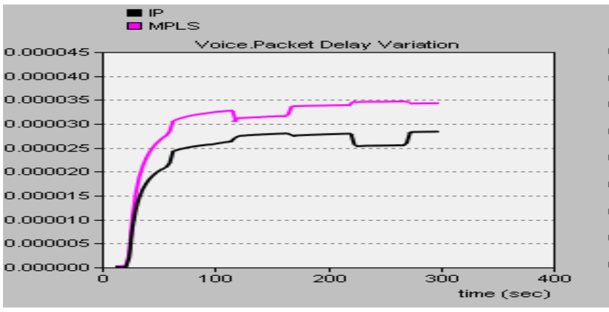


Figure (5) Packet delay variation

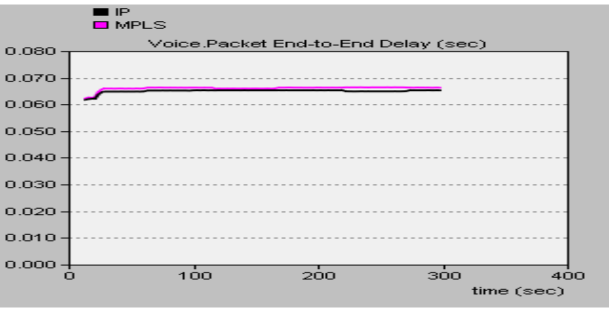


Figure (6) Packet end-to- end delay

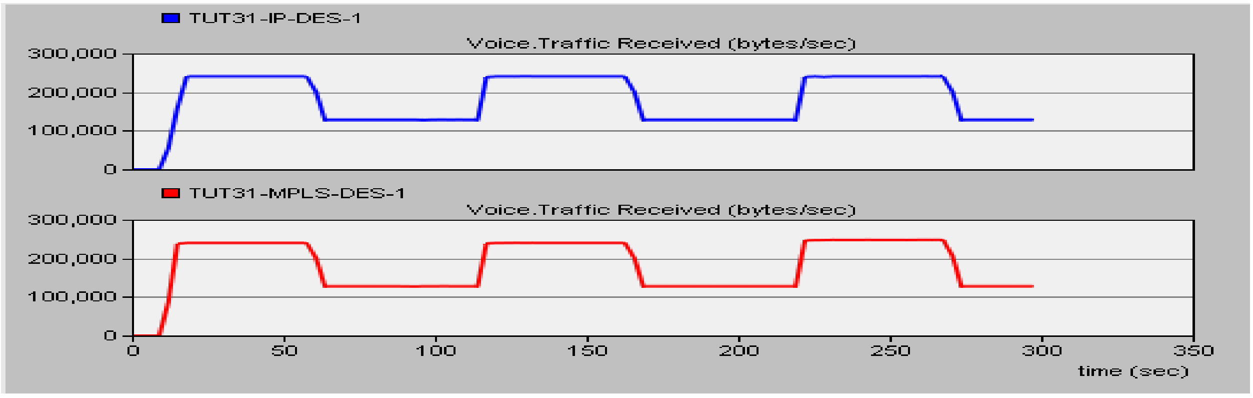


Figure (7) Traffic received

5.2. Large network

Here, the network size is larger than the previous one (figure 8).All links are OC-3(155Mbps) capacity. There are three different scenarios: light load, heavy load, and heavy load with QoS. The purpose of these scenarios is to show the improvement obtained by using MPLS rather than IP. It also shows the advantages of using QoS in MPLS network.

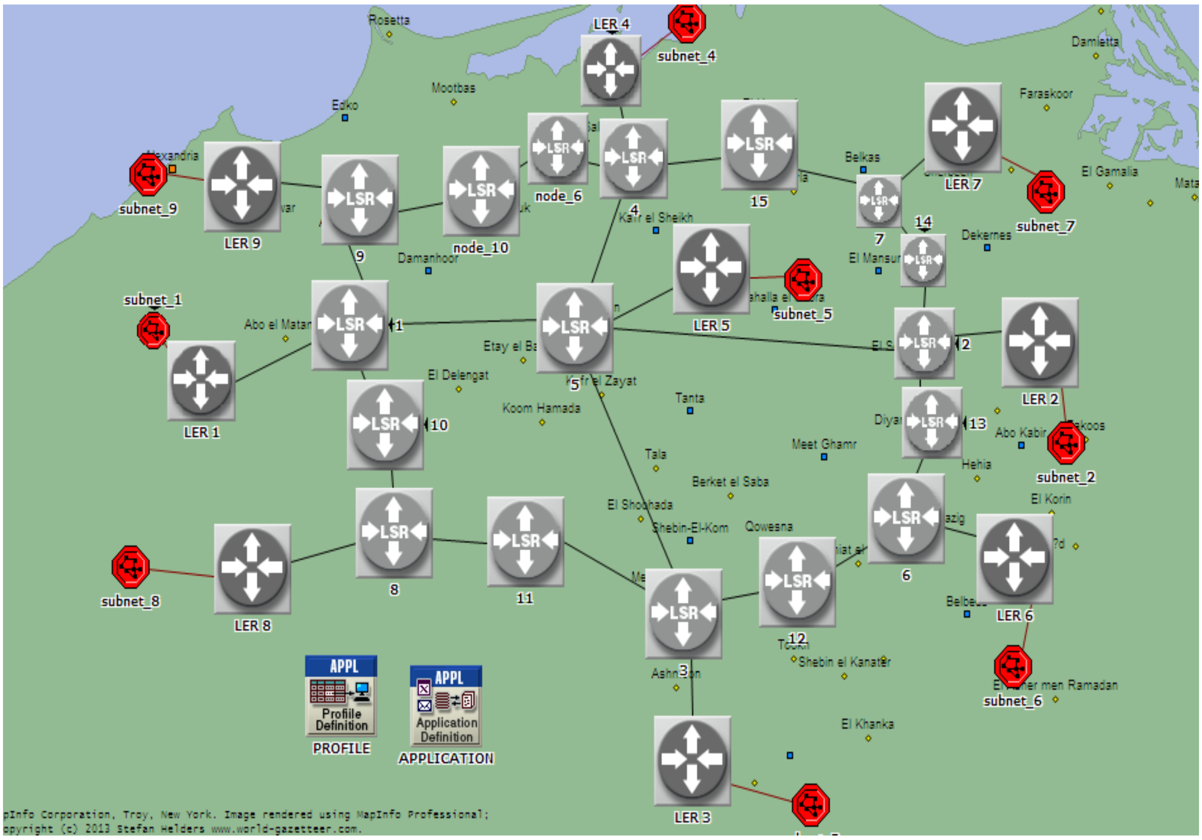


Figure (8) large network

5.2.1 Light load

This scenario has 9 Subnets. Subnets 3, 4, and 5 are server subnets (Print – Email – FTP).Each subnet has two servers. Each of the rest of the subnets has voice LAN, video workstation, E-mail LAN, FTP LAN and Print LAN. Each LAN consists of 10 users. The voice and video communications are between subnets 1 & 2, subnets 6 & 9, and subnets 7 & 9 consequently. The load of the network is as shown in Table 4.

Table 4 Network load

Traffic type	Description	Profile duration	Repeatability
FTP	-10 MB file size -Inter request time = const(50)	End of profile	Unlimited
VoIP	PCM Quality speech	End of profile	Unlimited
VIDEO	High Resolution	End of profile	Unlimited
PRINT	-30KB:9MB file size -Print interval time = exp(30)	End of profile	Unlimited
EMAIL	200KB size	End of profile	Unlimited

a) Results for Video

In this scenario the IP network can't offer the required QoS to achieve continuous high resolution video communication, so there isn't any video packet received (but it can achieve discrete low resolution video communication).

The MPLS network can offer the QoS requirement for video communication. Figures 9, 10 and 11 show the packet delay variation, end to end delay (in seconds) and the received traffic in bytes for video traffic in MPLS network. MPLS has 1.6 µs packet delay variation and 8 ms end to end delay almost all packets were received which is acceptable.

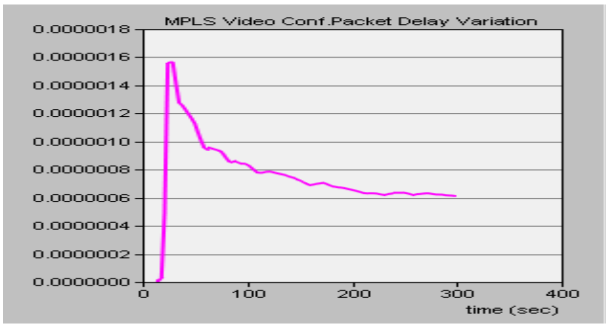


Figure (9) Packet delay variation

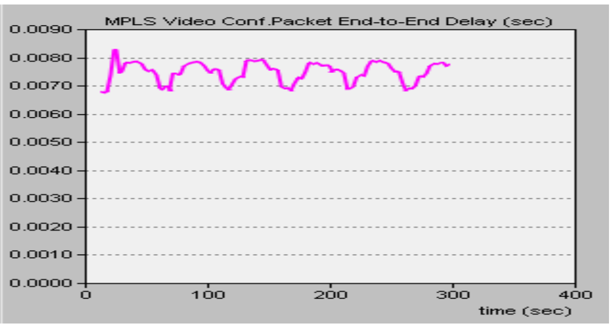


Figure (10) Packet end-to- end delay

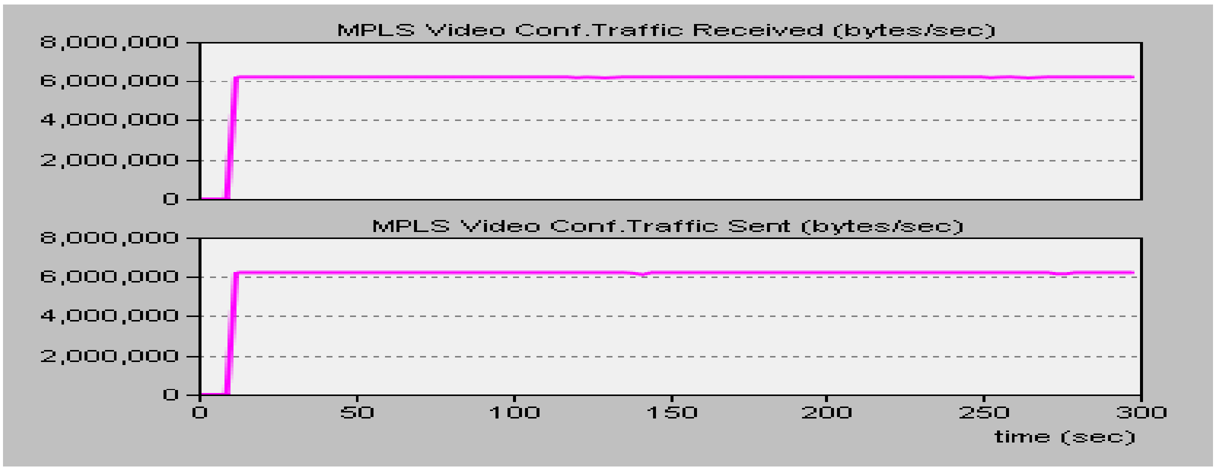


Figure (11) Traffic received

b) Results for Voice

Figures 12, 13 and 14 show the MOS value, packet delay variation and end to end delay (in seconds) while table 5 shows transmitted, received packets and total packet loss for voice traffic. For voice traffic, the MOS value is equal for both networks (3.7). IP has about 0.1 µs packet delay variation and 62 ms end to end delay which is acceptable while MPLS has 0.5 µs packet delay variation and 62 ms end to end delay which is acceptable too.

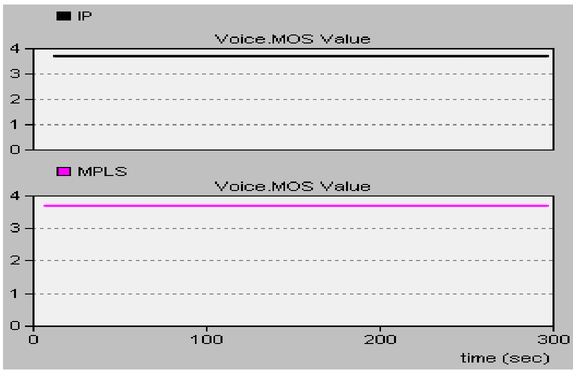


Figure (12) MOS value

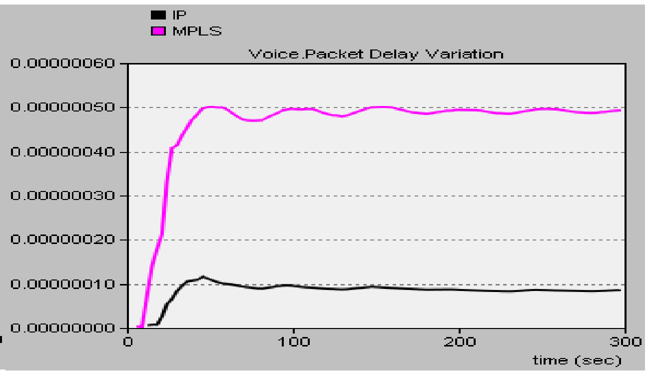


Figure (13) Packet delay variation (0.5 µsec for MPLS- 0.1µsec for IP)

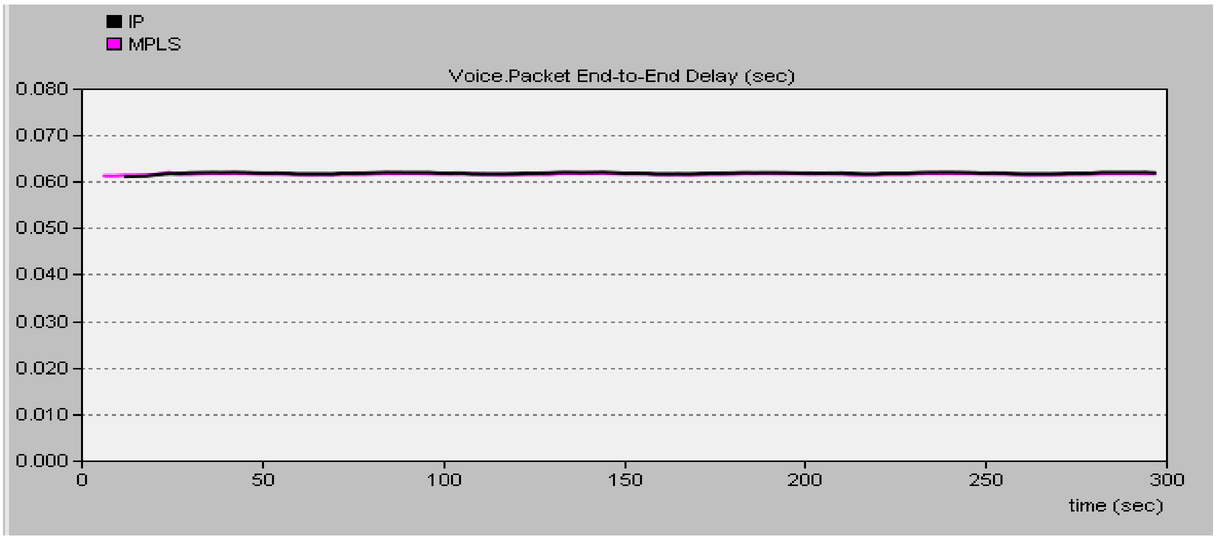


Figure (14) Packet end to end delay (IP and MPLS have the same delay 62 msec approximately)

Table 5 Total Packet Loss

	IP	MPLS
Total transmitted packets	641,912	644,422
Total received packets	614,845	642,752
Total packet loss	27,067	1,669
Packet loss ratio %	4.2 %	2.6%

This scenario shows that IP can't offer the required QoS to achieve continuous high resolution video communication, but it has the same performance as MPLS in voice communication while MPLS has an accepted performance for video communication (1.6 µs packet delay variation and 8 ms end to end delay) and better throughput and less packet loss than IP for voice. This is because MPLS TE can find a path in the network that meets a series of constraints (bandwidth, delay…) by using Constraints Shortest Path First (CSPF) at ingress node.

5.2.2 Heavy Load Scenario

In this scenario network load will be increased by increasing the number of subnets to 17 subnets. Each additional subnet will have three voice LANs and one FTP LAN. The additional subnets will have interactive voice communication as follows: subnet 10 with 14, subnet 11 with 15, subnet 12 with 16, subnet 13 with 17.So each subnet has 30 voice subscribers and 10 FTP subscribers. Also voice LANs will be three LANs for subnet 1 and subnet 2 (figure 15).

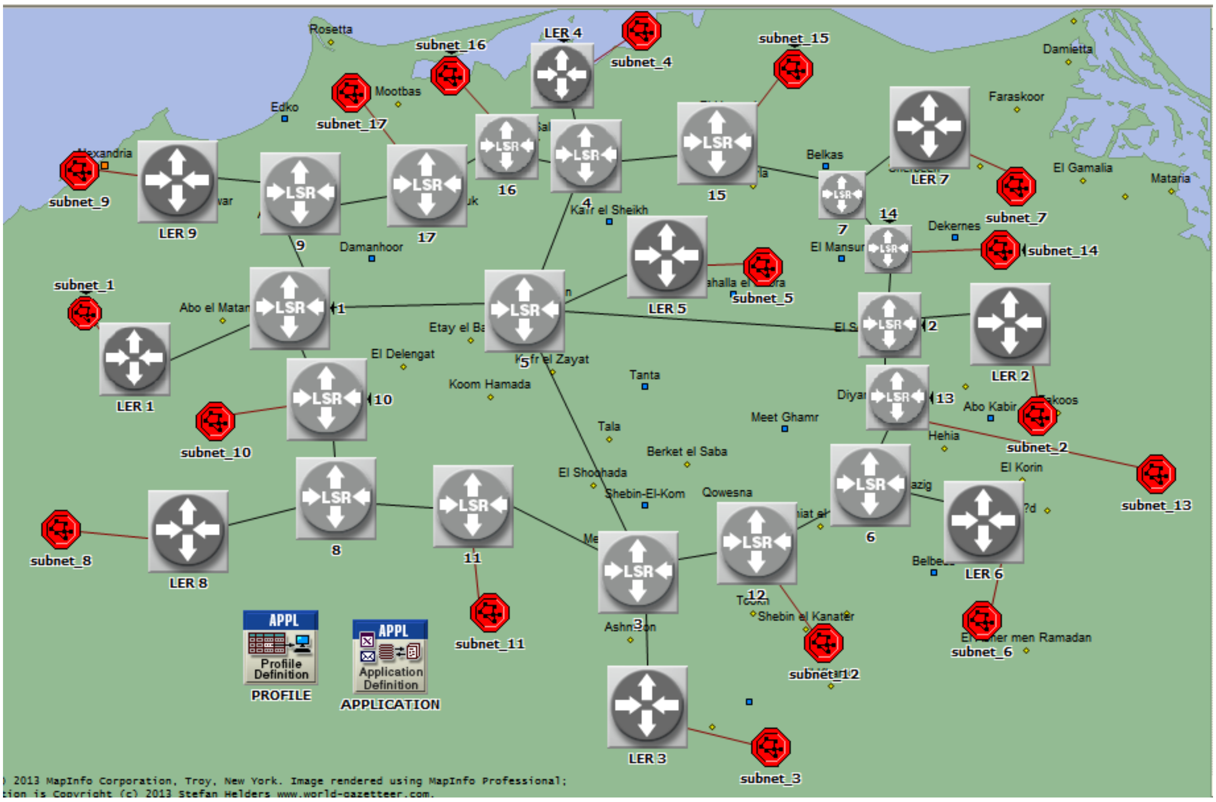


Figure (15) heavy load network

a) Results for Video

In this scenario the IP network can't offer the required QoS to achieve continuous high resolution video communication, so there isn't any video packet received.

The MPLS network can offer the QoS requirement for video communication. Figure 16, 17 and 18 show the packet delay variation, end to end delay (in seconds) and the received traffic in bytes for video traffic in MPLS network. MPLS has 1.2 µs packet delay variation and 9 ms end to end delay and almost all packets were received.

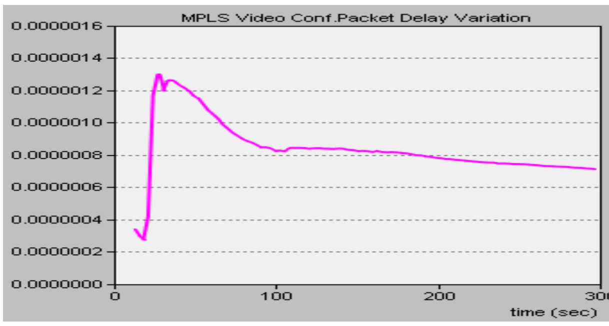


Figure (16) Packet delay variation

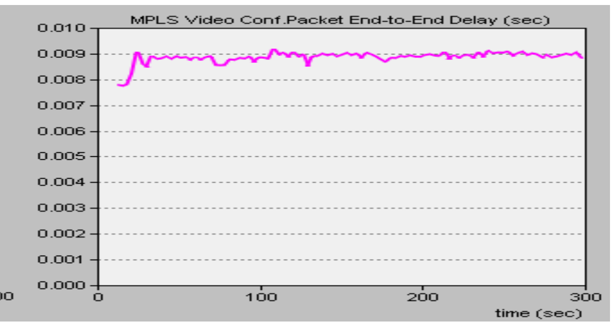


Figure (17) Packet end to end delay

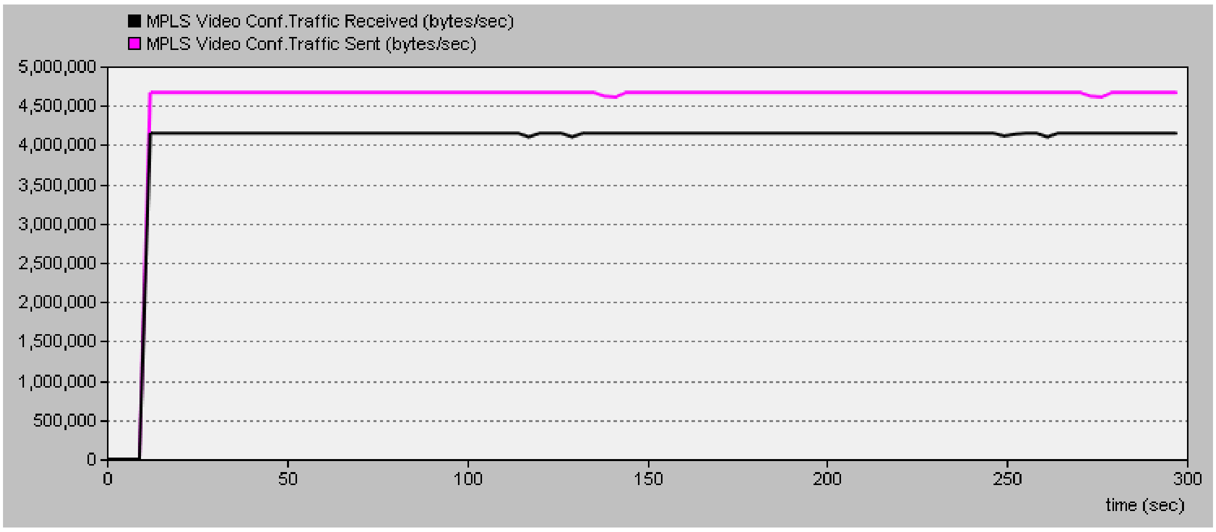
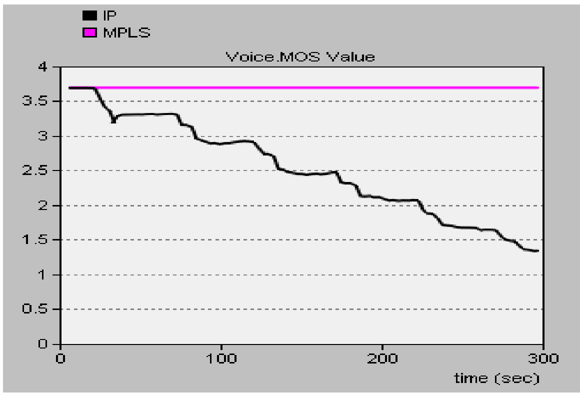


Figure (18) MPLS video traffic sent and received

b) Results for Voice

Figures 19, 20 and 21 show the MOS value, packet delay variation and end to end delay (in seconds) while table 6 shows transmitted, received packets and total packet loss for voice traffic. As shown from the figures IP network performance degrades with time, by the end of simulation time the MOS value reaches 1.5, 10 ms packet delay variation and 450 ms end to end delay which is not accepted while MPLS keeps its good performance as the light load scenario. It has 3.7 MOS value, 0.5 µs packet delay variation and 62 ms end to end delay.



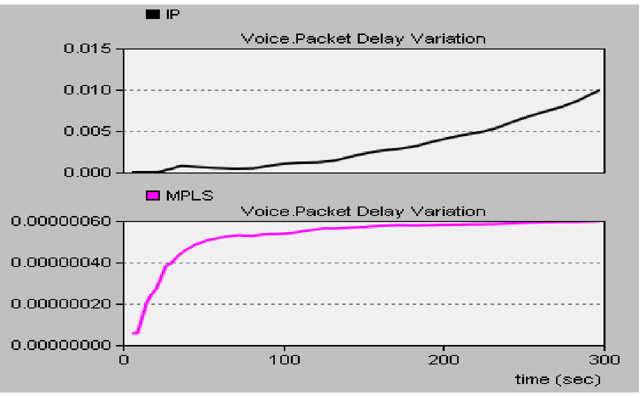


Figure (19) MOS value **Figure (20) Packet delay variation**

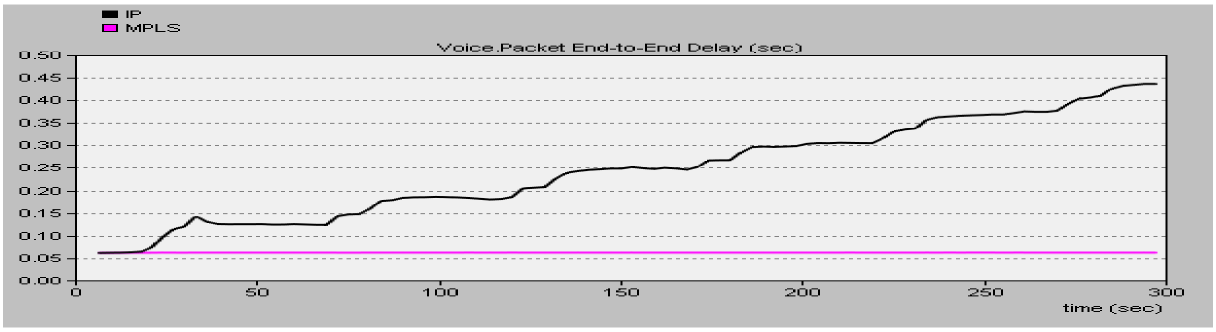


Figure (21) Packet end to end delay

Table 6 Total Packet Loss

	IP	MPLS
Total transmitted packets	3,643,973	3,642,809
Total received packets	3,550,558	3,642,696
Total packet loss	93,414	113
Packet loss ratio %	2.6 %	0.003 %

From this scenario: MPLS has a better performance than IP in voice and video communication. Also MPLS satisfies the QoS requirements for multimedia and real time traffic.

5.2.3. Increasing background traffic

In this scenario we will use the previous network and will see the performance after increasing the background traffic of the network to reach congestion as shown in figure (22):

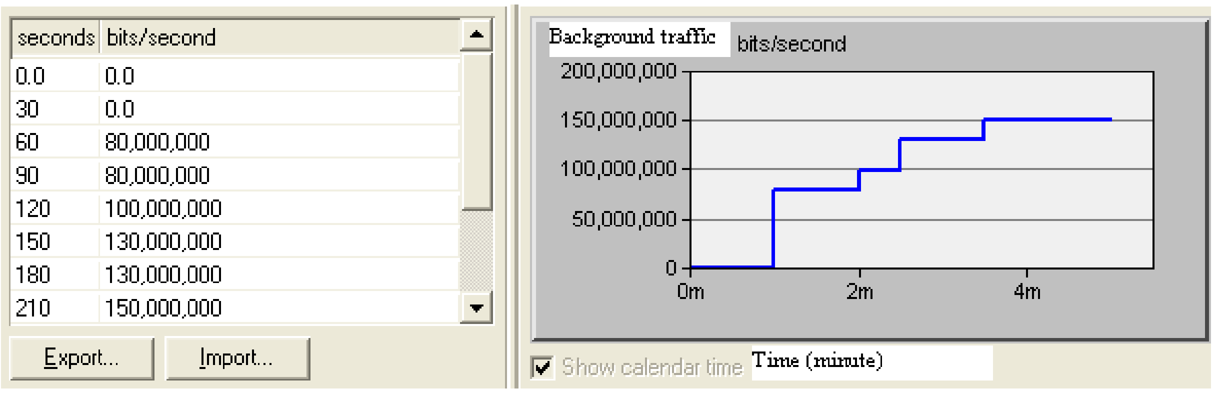
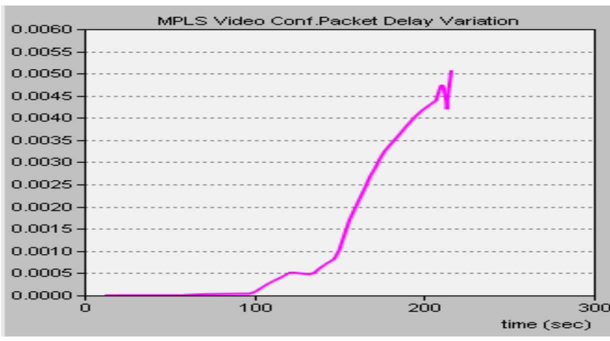


Figure (22) background traffic added to the network verses time

a) Results for Video

Also, in this scenario the IP network can't offer the required QoS to achieve continuous high resolution video communication, so there isn't any video packet received.

For MPLS video communication, the network performance began to degrade at the moment (after 150 second simulation time) the back ground traffic reached 130 Mb/s and there isn't any packet received since the background traffic reached 150 Mb/s (after 210 second simulation time). Figures 23, 24 and 25 show the packet delay variation, end to end delay (in seconds) and the transmitted and received traffic in bytes for video traffic.



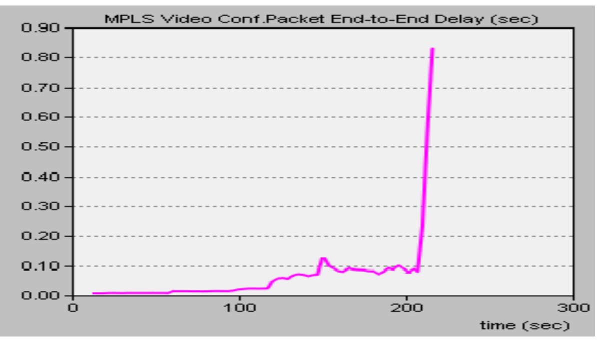


Figure (23) Packet delay variation **Figure (24) Packet end to end delay**

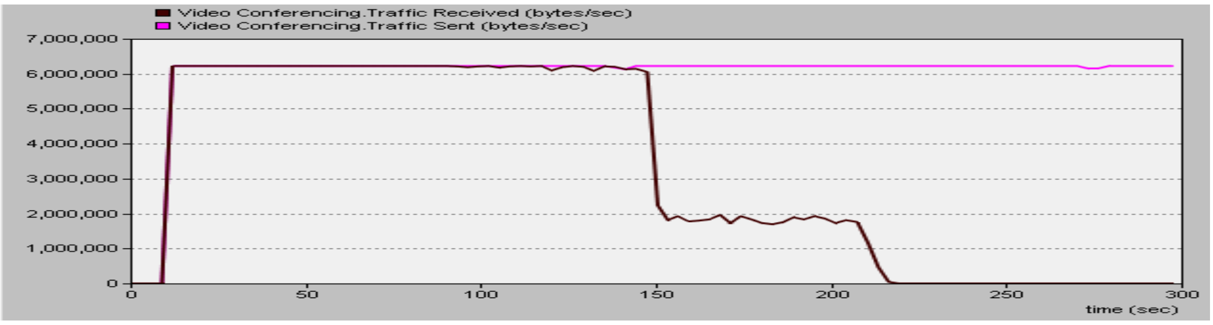


Figure (25) Traffic transmitted (Red) and received (Blue) and packet loss

b) Results for Voice

For both networks, performance began to degrade at the moment (after 150 second simulation time) the back ground traffic reached 130 Mb/s and the degradation rate increased when the background traffic reaches 150 Mb/s. Figure 26, 27 and 28 show the MOS value, packet delay variation and end to end delay (in seconds) while figure 29 andTable 7 shows transmitted, received packets and total packet loss for voice traffic.

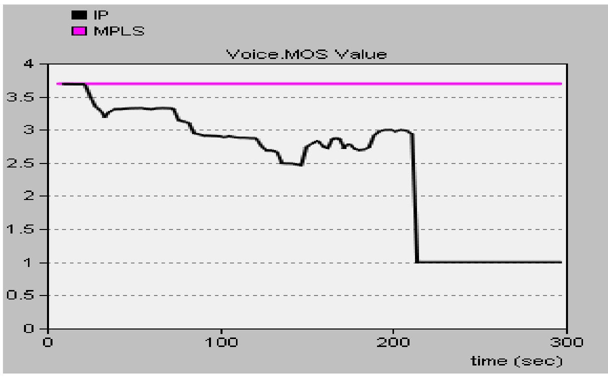


Figure (26) MOS value

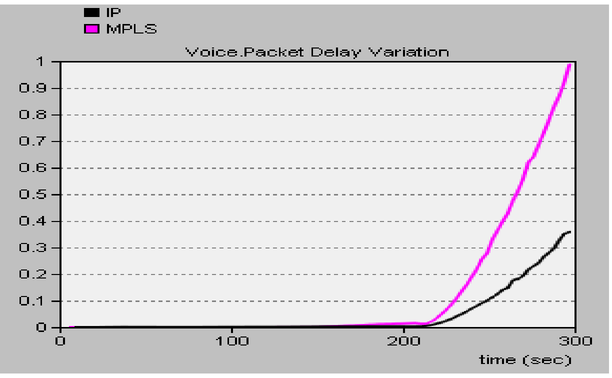


Figure (27) Packet delay variation

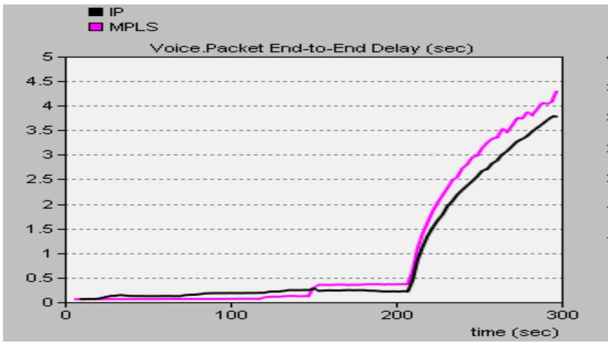


Figure (28) Packet end to end delay

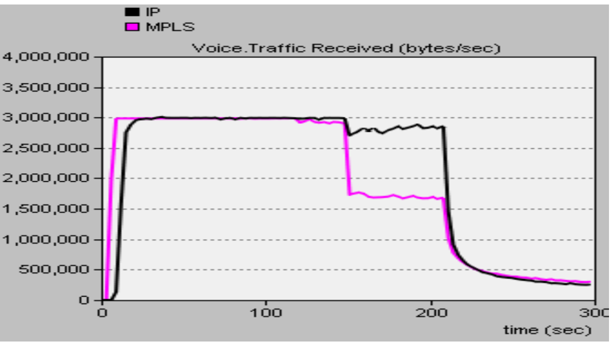


Figure (29) Traffic received and packet loss

Table 7 Total packet loss

	IP	MPLS
Total transmitted packets	3,645,266	3,642,809
Total received packets	2,561,277	2,358,788
Total packet loss	1,083,989	1,284,021
Packet loss ratio %	29.7 %	35 %

At this scenario we applied a background load and that load reached the max capacity of the links beginning from the 150th second from the simulation time (users' traffic is about from 80 to 100 Mbps). So at this time the network discards most of packets.

5.2.4. Enhancing MPLS network by applying QoS

This scenario will use the previous network (with background traffic) and apply QoS by using TOS field and mapping it to MPLS EXP field. Also Weighted Fair Queuing (WFQ) and Weighted Random Early Detect (WRED) will be used.

a) Results for Video

The IP network still can't offer video communication (figure 32).
For MPLS video communication, the application of QoS to MPLS network restores back the good performance for video traffic. Figure 30, 31 and 32 show the packet delay variation, end to end delay (in seconds) and the received traffic in bytes for video traffic. MPLS has 3 µs packet delay variation and 10 ms end to end delay which is acceptable and all packets almost were received.

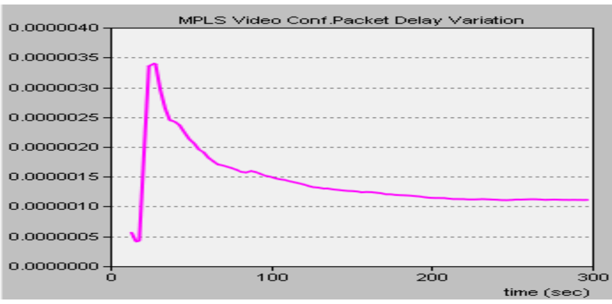


Figure (30) Packet delay variation

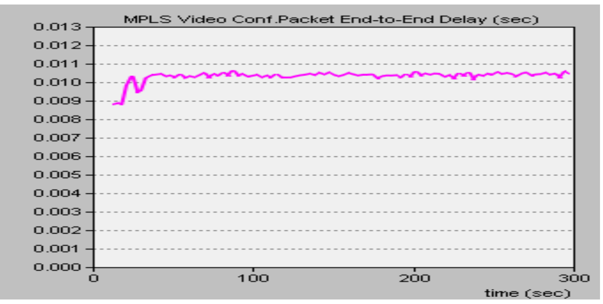
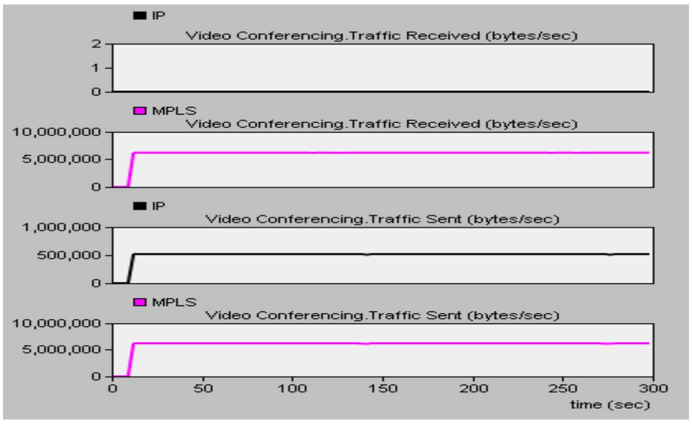


Figure (31) Packet end to end delay



Received IP traffic

Transmitted IP traffic

Received MPLS traffic

Transmitted MPLS traffic

Figure (32) Traffic received and packet loss

Table 8 Total packet loss

	IP	MPLS
Total transmitted packets	1,439	17,274
Total received packets	0	17272
Total packet loss	1,439	2
Packet loss ratio %	100 %	0.01 %

b) Results for Voice

Figures 33, 34 and 35 show the MOS value, packet delay variation and end to end delay (in seconds) while Table 9 shows transmitted, received packets and total packet loss for voice traffic. As shown from the figures IP network performance degrades with time and by the end of simulation time the MOS value reaches 1.5, 10 ms packet delay variation and 450 ms end to end delay which is not accepted while MPLS restore back its good performance for voice communication. It has 3.7 MOS value, 2 µs packet delay variation and 62 ms end to end delay.

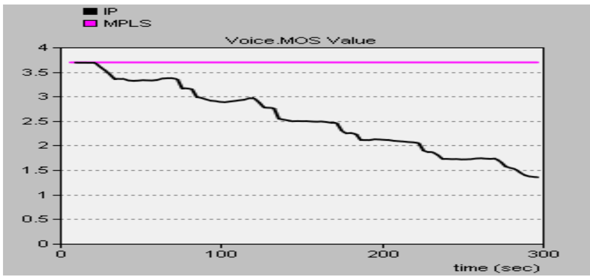


Figure (33) MOS value

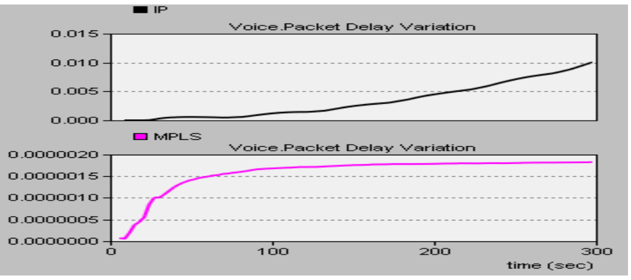


Figure (34) Packet delay variation

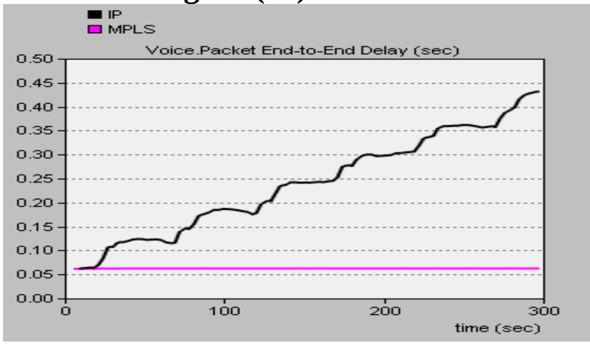


Figure (35) Packet end to end delay

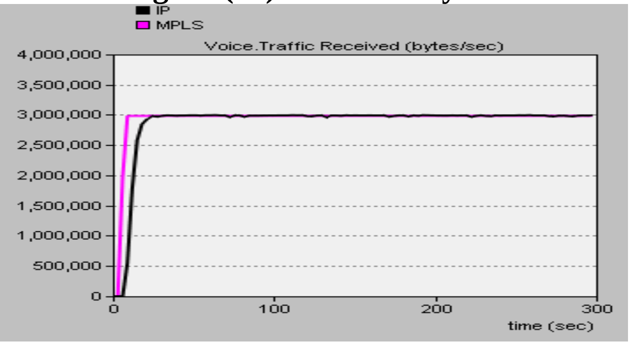


Figure (36) Voice traffic received

Table 9 Total packet loss

	IP	MPLS
Total transmitted packets	3,645,683	3,642,809.333
Total received packets	3,570,612	3,642,696
Total packet loss	75,071	113.333
Packet loss ratio %	2 %	0.003 %

At this scenario, the network has the same background traffic as the previous scenario, but with QoS applied by using ToS field to differentiate between traffic. Also WFQ was used to give priority to real time traffic with a fair chance for best effort traffic to communicate and WRED was used at each node to enhance multimedia and real time packet drop at congestion period. Here MPLS network restore back the same required performance for multimedia and real time traffic.

6. Conclusion

In this paper, MPLS technology is used in small and large network with light and heavy loads. Finally QoS is applied to improve MPLS performance. From the simulation, it is clear that there is no need to apply MPLS to small network with sufficient resources to its traffic, but for large network with heavy load we notice that MPLS has better scalability than IP and improve the performance for multimedia and real time traffic, this is because MPLS TE can find a path in the network that meets a series of constraints (BW – delay). But TE can't distinguish between two types (class) of traffic and can't enforce allocations at a per class granularity. This is illustrated at heavy load scenario. After applying QoS to MPLS, the MPLS network restores the perfect performance back for multimedia and real time application.

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