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Simulation analysis of real-time video QoS over IP and IP/MPLS networks

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Abstract: **video over IP applications such as Video conferencing, peer-to peer video is gaining popularity in the recent years. These applications are described as real-time and require stringent QoS when transporting them over a network. The IP architecture proved to be an efficient, economic and a simple way for transporting real-time video applications, however there are situations where IP may be inappropriate for video transport such as congestion and QoS which is provided per hop. MPLS is a label based forwarding technology that was designed to improve the IP performance by using traffic engineering capabilities it steers the traffic from congested links to less congested ones, additionally MPLS provide an end-to-end QoS by setting up explicit paths between an ingress and egress nodes, with traffic constraints on data following that path and certain QoS. This paper presents a comparative based analysis for QoS metrics (loss, delay, delay variations) of video applications while transported over IP and IP/MPLS networks. Several scenarios are performed to observe the performance of QoS, results shows that MPLS provides better performance than IP networks.**

Keywords: MPLS, QoS, OPNET

I. INTRODUCTION

The internet is a technology that proves to be an efficient and low cost of way of communication between people, which as became an integral part of people life. Video transport over the internet is a service which gained the most usage over the internet due to the advent of low price computers and hand held devices which made the access to these services easy. Cisco index shows that video traffic will consume 90% of the internet traffic [1].

Providing video services over the internet is a challenges task to internet service providers, depending on the type of video application used over the internet, different requirements is needed for each type of application. for example video streaming requires an available bandwidth (500k-2M) and a response time of (2-5 sec) and video-on-demand requires bandwidth range (500k-60 M) and a response time (2-5 sec), other video application such as videoconferencing requires a response time of (<100ms)[2]. The description of video application requirements implies that video traffic may occupy 30-90% of the bandwidth capacity this may cause bandwidth utilization instability and may results a slow response time for other applications. A suitable for quality of service (QoS) policy is needed to control the behavior of video traffic.

The current internet relays on the TCP/IP stack for data delivery. The internet protocol (IP) was designed as a connectionless protocol, meaning there are no guarantees that data will be delivered to destination correctly and in ordered fashion and there is no discrimination between application protocols. The internet network is made of large networks connected together, and in order to manage these networks the internet is divided into smaller areas called Autonomous systems (AS). The IP relies on interior exterior routing gateway protocol (IGRP, EGP) for exchanging routing information between ASs and building routing tables and calculating the best path to the destination. The best path is computed by calculating the shortest path to the destination. This kind of situation may results links being over utilized and causes congestion in the network, which means that the best effort IP delivery may not able to meet video applications requirements.

In order to provide services for different applications over the internet service classes are needed[3]. The Internet Engineering Task Force (IETF) has proposed several service models and mechanisms to meet the demand of QoS namely: the Integrated Service (IntServ)[4].model, Differentiated Service (DS) model, Multi Label Protocol Switch (MPLS)[5], Traffic Engineering (TE). The integrated service model is based on resource reservation protocol (RSVP), where resources are reserved for high priority applications only. However the main problem of this scheme is scalability where the number of resources needs to be increased as the number of flows increases. In DiffServ model the packets are marked differently with create several packet classes, in which each packet in a class received a different service. MPLS is a forwarding mechanism. In which packets are labeled at the ingress of an MPLS capable domain followed by classification and forwarding based on the label values. Traffic engineering is the process of controlling traffic flows in the network.

As mentioned above MPLS is a forwarding mechanism based on labels. the main advantage of using labels for forwarding is to simplify forwarding, where the router use the label value only for forwarding (short prefix match) without inspecting the rest of the packet header. MPLS also simplify the classification process by classifying packets at the ingress LSR. Where the main goal of MPLS was to increase the speed of forwarding but this argument no longer holds because the advent of fast switching routers. These routers contain a dedicated high performance circuits to do the routing lookups.

MPLS is used for providing QoS. The MPLS service model is based on the DiffServ model. In this architecture, LSPs are first configured between each ingress-egress pair. different packets will be send on different LSP based on the packet priority.

In this work a simulation based analysis will be presented in order to evaluate the performance of video applications over. Several scenarios has been prototyped in order to compare the performance between IP and MPLS networks. Analysis of application performance over MPLS has been studied before. In [6] authors presented an analysis of MPLS QoS, by separating UDP and TCP traffic on different links and different traffic trunks. In [7] authors presented a simulation study of MPLS-DiffServ improvement over IP network. In[8] authors presented a study on impact of Diffserv-MPLS on network parameters such as jitter ,throughput and delay for real-time applications. In this paper a simulation study is presented to evaluate the performance of video applications in the presence of background traffic.

II. THEORETICAL BACKGROUND

MPLS Operation

Once a packet enters an MPLS network, a label is appended to the packet header (label pushing) by the ingress edge router. The decision of assigning label values to the incoming packets is made by the ingress label switch router (LS a router with MPLS capability). The decision is made on the packets forwarding equivalence class (FEC). An FEC is a set of packets the shares a common attributes such as packets destination address, DiffServ values, source application port, and other factors. all packets belongs to an FEC will share the same path in an MPLS network. a path that labeled packets follows is called label switch path (LSP), which is a virtual connection between two points , starts with an ingress LSR and ends with egress LSR. The intermediate LSRs forward packets based on the label's value contained in the packet's label. When an LSR receives a packet, it looks up the label entry in the Label Information Base (LIB) and determines the outgoing interface and the corresponding label value (label swapping). At the egress LSR the label is removed (with pop operation) and forward it with normal IP operation[9].

In the context of MPLS we have traffic trunks. A traffic trunk is an aggregation of traffic flows which are placed inside an LSP.a trunk are routable objects, they can be moved from a path to another. Traffic trunks capture traffic characteristics such as peak rate, average rate, and average burst size[5].

III. Simulation

A. Simulation Aim and Environment

The aim of the simulation is to observe the video application performance when transported on traditional IP network and IP/MPLS network.In terms of results analysis, we will focus on the Received Packet /Packet loss, End-to-End delay (ETE), Packet Delay Variations (PDV) collected at the destination nodes.We have prototyped number of scenarios in order to compare different results under different situations.

B. Simulation Setup and Details

The network topology (shown in figure-1)used for the simulation consists of edge networks on the source and destination sides and a core network connecting source to destination. The edge networks consist of source and destination workstations and edge routers connected to the core network. The edge network devicesareconnected with a 100BaseT Ethernet line. The core network is formed by fiverouters (R1-R5) connected with each other using PPP-DS3 link type. All routers in the core network are MPLS capable. They have been configured such that their label mapping and switching algorithms are enabled only when LSPs are defined in the network. When no LSPs are defined, these routers will use the routes advertised by the dynamic routing protocol running on their interfaces (default route R1-R2-R5) to forward packets to their destination.

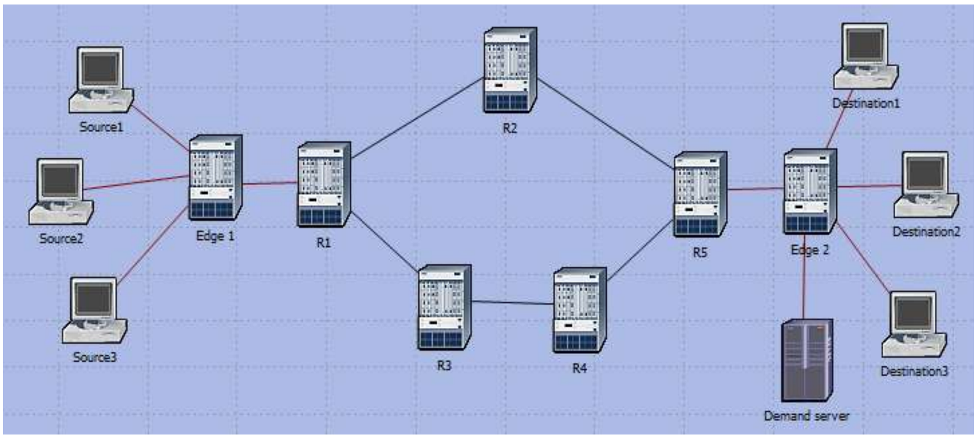


Figure 1: Simulated Network

C. Traffic parameters

- Video conferencing: video application is deployed at source and corresponding destination station. three types of video conferencing are configured , the configurations is shown in the table-1 .

Table-1 video application parameters

Application name	Application Attributes	Rate (bit/sec)	Deployment	Simulation Time(sec)
Video1	Frame size=50KB , frame rate =10 frames/sec	4 Mbps	Source1->Destination1	200
Video2	Packet size=50KB , frame rate =15 frames/sec	6Mbps	Source2-> Destination2	200
Video3	Packet size=50KB , frame rate =30frames/sec	12Mbps	Source3-> Destination3	200
Background	2Mbit/sec	2Mbps	Source 1,2,3->Background Server	200
Background	4Mbit/sec	4Mbps	Source 1,2,3->Background Server	300
Background	8Mbit/sec	8Mbps	Source 1,2,3->Background Server	400

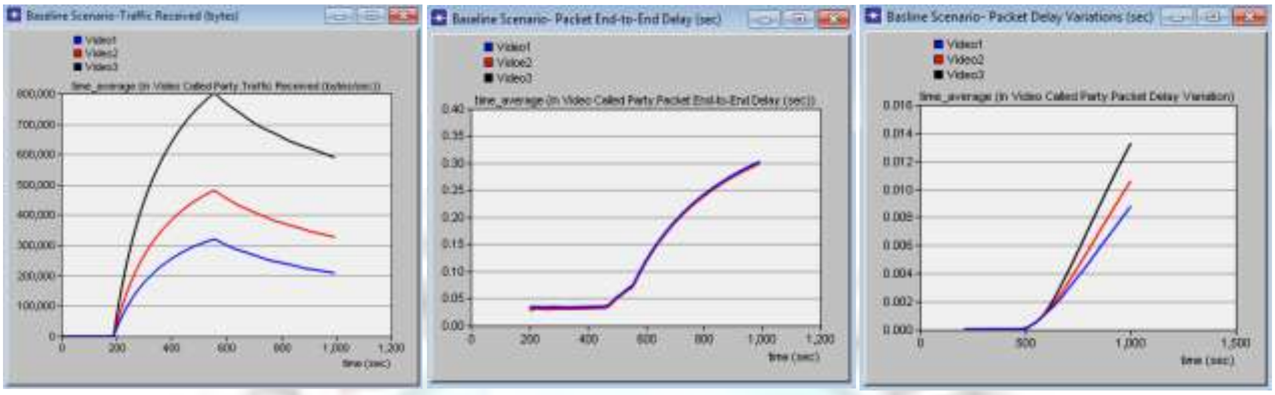
- Background traffic: demands are used for background traffic generation. Demands are configured from each source station to the Demand server. The amount of background traffic is varied through simulation (Table-1) in order to observe the performance under different conditions. The primary purpose of the background traffic is to model the effect of general traffic in the network on selected traffic of interest. This effect primarily takes the form of delay, utilization and throughput of the link which enables modeling the traffic flow as in real network.

IV. Simulation Results and Analysis

A. Baseline Scenario

The purpose of this scenario is to observe the performance of the three network metrics for the best effort IP delivery. Routers are configured to run OSPF as a dynamic routing protocol at the core network, so any packet comes across the core network will take the shortest path advertised by the dynamic routing protocol which in this case (R1-R2-R6). All routers in this scenario are configured with First-Come-First-Served (FCFS) service.

The performance metrics are shown in figure 2. The figures shows a good performance at the beginning of simulation, however when the amount of background increases, the performance parameters degrades. Figure-2a shows the packets received for all video flows. The figure shows for the period (200-430 sec) of simulation time the packets received is increasing , however, when the background traffic starts to increase the average packets received starts to decrease for all video flows at time (560 sec and above), the value continue to decrease until the end of simulation. Figure-2-b shows the end-to-end delay for all video flows. The figure shows tolerable values for the ETE for the period (200-430 sec) of simulation time, however, when the background traffic starts to increase the values of End-to-End delay also increases for period (430 sec and above), the value continue to increase until the end of simulation.Figure2-c shows the Packet Delay Variations for all video flows, at time of simulation (200-470 sec) the average values of PDV is about 22µs which an acceptable value for a real-time video application. For simulation time above (470 sec) the PDV will keep increasing until the end of simulation, the average value for the PDV is about 15ms. from the figure it is also observed that PDV changed from 20 µs to 30ms with the increase in the background traffic.



(a) (b) (c)

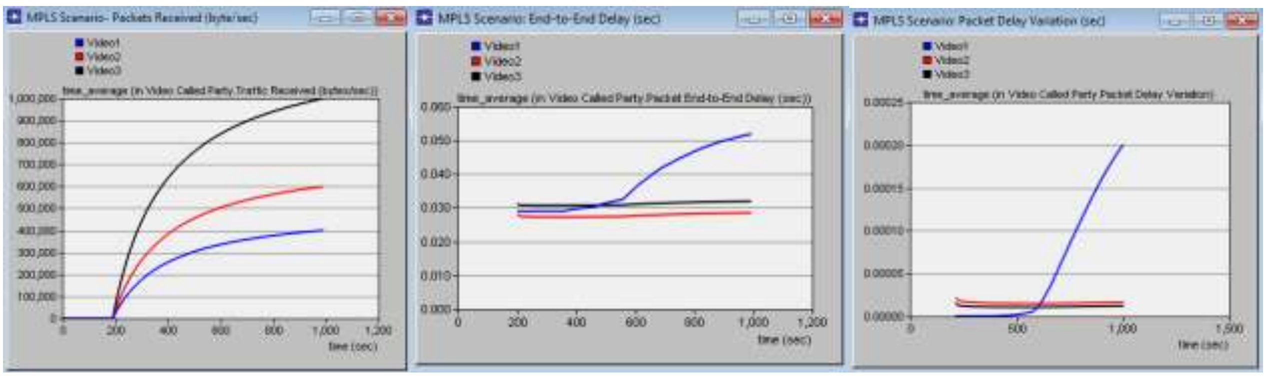
Figure 2: performance metrics for the baseline scenario (a) Packets received (b) Packet End-to-End Delay (c) Packet Delay Variation

B.

In this scenario MPLS is enabled at the core network. Two LSPs are established to carry traffic along the upper and the lower path at the core network. FEC are established to classify packets based on their source-destination address. Four traffic trunks are configured, three for each video traffic flow and one for background traffic. Traffic Mapping configurations are set such that the traffic flows coming from source-1, and Source-2 are configured to take the upper LSP, while source-3 are configured to take the lower LSP. The trafficreceived at the destination stations are shown in figure-3-a.As shown there is no packet loss even when the background traffic is increased the received packet is not affected, since all links at the core stations are utilized, the core network is able to satisfy all traffic demands.

Thepacket ETE delay is shown in figure-3b. For Video-2 and Video-3 traffic the average delay is about 31 ms for video3 packet flow and 27 ms for video3 packet flowfor all simulation time, source-1flow is sent into the lower LSP, not only it takes a longer path to destination but the path is also shared with the background traffic as a result Video3 traffic is suffering more delays and the delay varying as the background traffic increases.The average value for the packet ETE is about 30 ms for time (200-470 sec), for (470 sec) and above the ETE will increase to an average of 40 ms.

Figure-3c shows the packet delay variations for each video traffic flow. as shown in the figure the PDV for Video-1 flow is 56 µs, for Video-2 15 µs for Video-3 is 15 µs. it also noted that the PDV value for Video-1 is increasing to the end of the simulation.



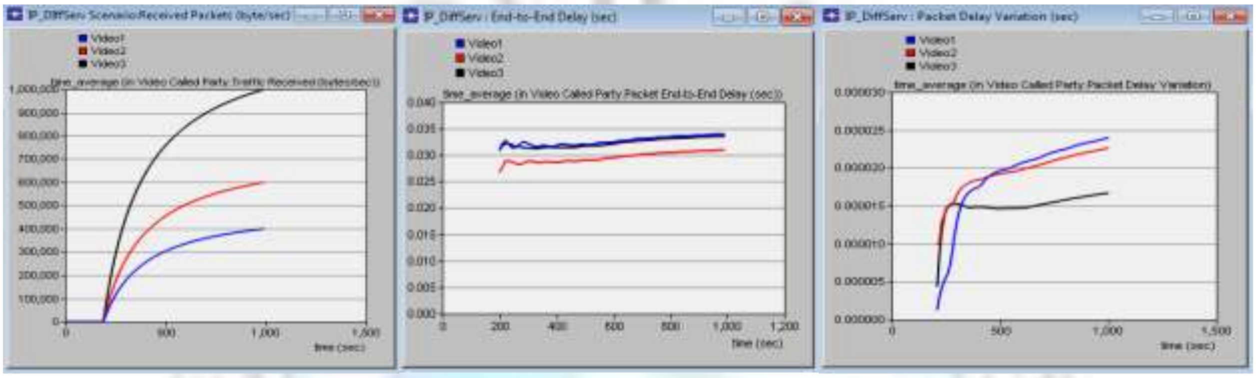
(a) (b) (c)

Figure 3: performance metrics for the MPLS scenario (a) Packets received (b) Packet End-to-End Delay (c) Packet Delay Variation

C. IP with QoS

in this scenario QoS attribute is set at the core routers with priority queuing as a scheduling scheme. Packets are marked with DiffServ points at the source stations. Priority queuing is enabled on the core routers interfaces and the priority is based on DiffServ points. Packets are marked at the source stations with DiffServpoints. The Video traffic flows assigned to multimedia traffic class which is marked with AF4x values, for video-1 flow packets are marked with AF41 and for video-2 packets are marked with AF42, video-3 packets are marked with AF43. The background packets are assigned to background traffic class with priority given AF11.

Figure4a shows the received traffic at destination stations, clearly it tis the same as the case with MPLS scenario almost there is no loss for all video traffic flows, this is because the video traffic gets higher priority than the background traffic for forwarding. The End-to –End delay is shown in figure-4b.the average value of the end to end delay is about 32 ms for Video1 and Video3 , and for Video3 is 28 ms, it is also can be noted there is a little increase in ETE value when the background traffic increases.figure-4c shows the PDV.For video1 the values of the PDV starts at 1.38 µs and increases to 23 µs. for Video-2 starts with PDV values of 10 µs and keeps increasing through the simulation time until it reaches values of 22 µs. Video-3 flow suffers less than the other flows which starts with PDV values of 4 µs and increases to a value of 16 µs.

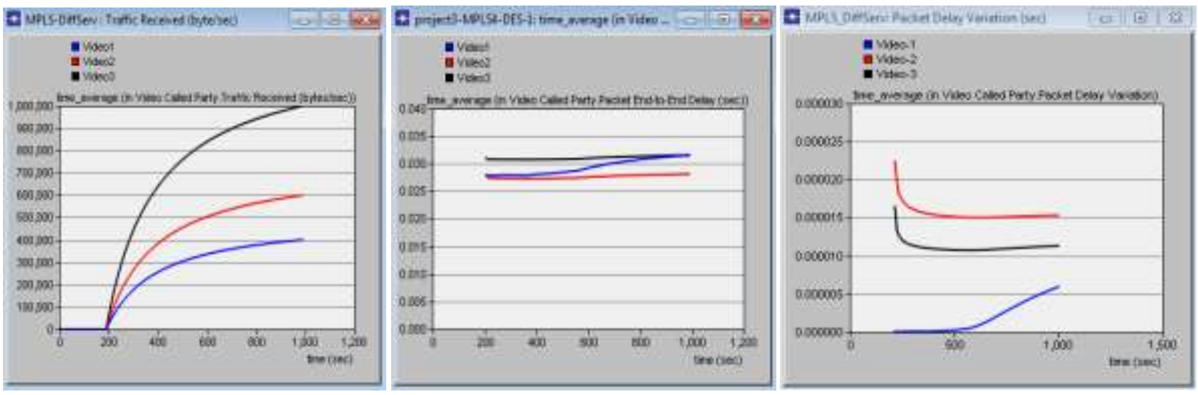


(a) (b) (c)

Figure 4performance metrics for the MPLS scenario (a) Packets received (b) Packet End-to-End Delay (c) Packet Delay Variation

D. MPLS withQoS

In this scenario MPLS is enabled at the core network. Classification of packets is done at the ingress router (R1), where packets are classified based on its source-destination pair and the DSCP values 4 trunk profiles are configured with different priorities are configured at the core network the . Priority scheduling scheme is enabled at the core routers .Detailed configuration is shown in table2. The performance metrics are shown in figure -5. Packets are received with no losses at the destination stations. The packet ETE is shown in figure-5b, the average values for the ETE is 30 ms ,28ms , 31 ms respectively for each flow. Figure-5c shows the PDV for each video flow. the average values for PDV is 5µs 16µs, 11 µs respectively for al video flows.



(a) (b) (c)

Figure 5: performance metrics for the MPLS scenario (a) Packets received (b) Packet End-to-End Delay (c) Packet Delay Variation

Results Statistics

In this section we will discuss the results obtained from the simulated scenarios.

A. Received packets/Packet loss ratio

Table-2 shows the statistics for the overall performance of video applications. The statistics for the baseline scenario shows that the standard deviation is close to the average received packets with min value of 58.4 packet/sec. This means for a period of time the video application experience a packet loss of 40% of packet loss. For other scenarios the standard deviation has small values compared to average. Also the statistics shows there is a small amount of loss.
In the first scenario we have seen that the reason for this performance is when the traffic demands exceeds the link capacity, each video flow suffers from packet loss.
In the second scenario we have seen how MPLS and traffic engineering features could reduce the packet loss, the load is balanced among the two links so the core network is able to satisfy the demands of all traffic and hence the packet loss is almost zero. For IP with QoS scenario each the flows is given a priority higher than background traffic flows , so every packet belongs to a video flow will be forwarded before any other packets. However the cost of priority given to higher packets is unfair for lower priority applications.

Table-2: statistics for overall received packets (packet/sec)

	Average	STDEV	Min	Max
Baseline Scenario	78.165	19.73359982	58.4	100
Network with MPLS	99.9825	0.075933015	99.7	100
Network with QoS	99.9475	0.096751018	99.5	100
Network With MPLS and QoS	99.9825	0.044366141	99.8	100

B. Packet End-to-End Delay

The packet end-to-end delay is a metric that is more susceptible to the change of traffic in the network. We could see for all scenarios when the background traffic increases the packet ETE also increases. The ETE statistic performance is shown in table 3. the same as received packets, the standard deviation is close to the average value. The maximum ETE value is 143.244 ms. the rest of the scenarios the values of the standard deviations is much smaller than the mean value. It is also noted for delay values for the last scenario has smaller values than other scenarios. the table also shows the minimum values for the ETE (when the background traffic is less than capacity of the link capacity). we can also notice MPLS performs better than IP with QoS.

Table-3: ETE statistics for all videoconferencing application(ms)

	Average	STDEV	Min	Max
Baseline Scenario	83.1583724	37.25641628	30.69559	143.2441695
Network with MPLS	31.99013559	2.230927622	29.0227	34.71154319
Network with QoS	32.03647667	0.984868487	29.83074	33.98390764
Network With MPLS and QoS	29.78608028	0.542630192	28.91006	30.5370986

C. Packet Delay Variations

This metric is affected by the background traffic as the same with ETE. The standard deviation is shown in table-4 which is very close the mean value. This is the same with the second scenario. When the QoS is activated in the network it improves the jitter performance.

Table-4: PDV statistics for all videoconferencing application (ms)

	Average	STDEV	Min	Max
Baseline Scenario	13.2698	13.192	0.03026	35.52203
Network with MPLS	0.20622	0.1991	0.01454	0.48631
Network with QoS	0.03043	0.0078	0.02600	0.06931
Network With MPLS and QoS	0.01557	0.0012	0.01453	0.022598

Discussion

In this paper we have prototyped four scenarios in order to evaluate the performance of video applications QoS. The scenarios showed a poor performance when congestion occurs in the network. When enabling QoS in IP networks, the application performance improved but unfairness may be introduced for other applications. Finally by utilizing other network resource through the use of TE engineering and combined it with QoS features of IP we can see better performance for real-time video applications and a fair performance for other lower priority applications in the network.

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