Performance Evaluation of Multimedia Streaming Applications in MPLS Networks Using OPNET

Reema A. Saad, Mariam Abojella Msaad
Faculty of Information Technology, University of Tripoli,
Tripoli, Libya
Ranem_reem@yahoo.com, Meemee_02@yahoo.com

Azeddien M. Sllame
Faculty of Information Technology, University of Tripoli,
Tripoli, Libya
Aziz239@yahoo.com

Abstract— **This paper describes the place of MPLS in current state-of-the-art networking as a quality of service means through performing performance analysis of VoIP and video conferencing applications by comparing the effect of different protocols (OSPF, IS-IS, EIRGP) and the effect of various queuing techniques (PQ, WFQ, MWRR) in order to find the good combination of MPLS, routing, and queuing type that provides efficient suitable quality of service levels. The obtained results illustrate a competent combination of MPLS with queuing discipline, and routing could be achieved for each application, such as MPLS and EIGRP with WFQ queuing is an efficient arrangement for video conferencing application.**

Keywords: MPLS; QoS; performance evaluation; routing; queuing; modeling, simulation.

I. Introduction

Multiprotocol label switching (MPLS) protocol is an efficient developed technology that enables resourceful packet forwarding, provides assured and reliable supply of the Internet services across heterogeneous wide area networks, owns high transmission speeds; exhibits lower network transfer delays, and provide scalable networking architecture. The crucial mark of MPLS is its Traffic Engineering (TE). TE is employed to manage the resources along the paths of the network efficiently. As a result, all of that offered quality of service (QoS) through networks that are using MPLS with multiple type of services (ToS) or distinct class-of-services (CoS) [1][2][3]. However, because of reduced network transfer delay, faster forwarding mechanism, scalability, and expectable services' performance of MPLS–based networks; it resulted in considering MPLS as a key technique for satisfying the requirements of the real-time applications such as VoIP and video conferencing and streaming. In addition, the MPLS mechanism establishes a connection-like network of virtual circuits over the ordinary IP datagram that are known as connectionless networks. Thus, MPLS allows the network management to enable accessing an added rich features and options to supervisor different flow streams within TCP/IP networks by approaching new MPLS networking technique which integrates the intelligence of layer 3 routing with efficient of layer 2 switching by label tagging approach. However, MPLS protocol uses standard IP routing protocols such as border gateway protocol (BGP), open shortest path first (OSPF), enhanced interior gateway routing protocol (EIGRP), and intermediate system-to-intermediate system (IS-IS) to understand the structure of the network's topology so as to create different routing and label forwarding tables employed in routers with MPLS capabilities. After the routing tables have been distributed, the label distribution protocol (LDP); of MPLS protocol suite; will dynamically links labels with every IP route found inside the routing table, and disseminates all formed bindings of labels and paths to all router's peers. On another hand, multimedia streaming applications and broadcasting sites are exploding around the world due to the fast growth and distribution of mobile operators, reduction of multimedia conferencing, multimedia streaming and VoIP prices, and a wide-range of computing devices offerings, which require higher bandwidth and certain QoS levels that are higher than other information running on Internet. Therefore, these requirements can be satisfied by MPLS-based networks [4][5][6].

This paper is organized as follows: Previous work is described in section 2. Definitions related to paper context are summarized in section 3. QoS models are outlined in section 4. MPLS technique is described in section 5. Section 6 illustrates queuing mechanisms applied in this paper. Section 7 demonstrates the experimental results of the case study. Section 8 dictates the conclusions.

II. Previous Work

There are many related work papers reported in many international conferences and scientific journals. Abdul-Bary et al. in [14] illustrated a comparison study between IP and MPLS networks with TE. KeerthiPramukh Jannu and Radhakrishna in [15] reported a performance comparison analysis study for VoIP in two analogous scenarios of IP and MPLS networks using OPNET tool. Sllame and Aljafari in [16] illustrated a performance experimental study of VoIP over IP/MPLS networks using OPNET tool with different routing protocols. Vesna et al. in [17] presented a simulation study between DiffServ model and DiffServ with MPLS TE. The results illustrated that the case study of MPLS TE with DiffServ has improved the network performances more than DiffServ alone scenario.

III. Definitions Related to Paper Context

The following terms are needed to understand the paper context [1][2][5][6] [13][16]:

- **Packet**: it is the smallest piece of data transfer in packet switching networks that has a predefined size which holds part of user data, with its header contains: addressing, sequencing, and flow and error control information.
- **Flow**: it is an individual unidirectional stream of packets that requires specific QoS level; the flow is generated by the same user's application's instance toward an intended receiver (s). The flow can be defined by 5-tuple (transport

protocol, source address, source port number, destination address, destination port number).

- **Bandwidth**: capability of any network to provide specified different QoS levels to distinct applications agreeing to their different requirements contained by the real capacity available on the network.
- **Packet loss**: it is the percentage of packets correctly received by the destination to amount of packets sent by the source per second over a link or a network. Packet loss may occur due to traffic over capacity, devices errors, link congestions, and improper use of network resources.
- **Throughput:** the sum of error-free transmitted and received packets between a sender and a receiver; measured as bps.
- **End-to-end delay (latency):** the delay time calculated as the difference between the packet receiving time at the receiver and the sending time from the sender; that includes: transmission delay, nodes processing delays, queuing delays, and propagation delay.
- **Delay jitter**: it is the change of the delay between two successive packets within a single packet flow; queues at routers are used to minimize the delay jitter.
- **Packet delay variation**: the changes among end-to-end delays for VoIP packets received by any node.
- **QoS**: is the capability of a network to minimize delay jitter, end-to-end delay, packet loss and maximizing throughput to efficiently use the available network bandwidth.
- **Real-time application**: it is the application that requires constrained timing requirements on every packet on the flow, in such a way that every single packet have to be reached at the destination by a definite time, which if not received by that time at the sender it is counted useless. However, VoIP is a real-time application that needs short end-to-end delay and firm delay jitter, while video streaming requires short delay jitter with acceptable insignificant packet loss.
- **Response time:** the amount of time between a time of requesting a specific network service and a response time to that requested service.
- **MPLS node:** a node that is able to perform MPLS switching and routing.
- **MPLS domain**: it is a continuous set of nodes composing of a single administrative area that are uses MPLS routing and switching techniques based on labels. The MPLS domain consists of label switch routers (LSRs) as core routers and label edge routers (LERs) as edge routers.
- **Forward equivalence class (FEC):** unique FEC is assigned to any group of packets that have related characteristics such as belong to same stream or protocol or QoS class in order to be manipulated equally by MPLS routers within the MPLS domain such that it forwarded along the same LSP path, inserted with the same MPLS label.
- **Label Switch router (LSR)**: its function is to perform packet's forwarding based on label switching, which is also able to achieve layer 3 routing.
- **Label switch router (LSR)**: capable to perform layer 3 routing and forwards packets using label switching. Ingress LER router inserts labels to packets, assigns packets to FEC. Egress LER router removes labels from packets leaving MPLS domain.

IV. QoS Models

A. Integrated service model

Integrated service (IntServ) model is also referred to as end-to-end or hard QoS. IntServ QoS needs an application to indicate that a specific level of service is required. The admission control protocol reacts to this request by assigning or allocating end-to-end resources for the application. If the resources cannot be assigned to a specific request, they will be declined. The Resource Reservation Protocol (RSVP) is used with the IntServ QoS resource reservation model. However, each end-to-end device must support the IntServ QoS protocol. IntServ QoS is not a scalable solution because: (i) there is only a limited amount of bandwidth available to reserve; (ii) IntServ QoS protocols adds additional overheads to end-to-end devices, as each traffic flow state must be fully maintained.

B. Differentiated Service Model

Differentiated Service (DiffServ) has been designed to be a scalable QoS solution. Traffic types are arranged into various classes and then labeled to identify their classification. Policies are then established on a per-hop basis to offer a specific level of service, based on traffic classification. DiffServ QoS is common for its scalability and flexibility in enterprise environments. Though, DiffServ QoS is considered soft QoS because it does not guarantee services like the IntServ QoS model [7].

C. MPLS

MPLS is a protocol that uses labels to route packets instead of IP addresses. With MPLS, just the first device does a routing lookup, it finds the ultimate destination along with a route to that destination. The path of the MPLS packet is called a label-switched path (LSP). In addition, MPLS adds one or more labels to a packet so that it would follow the LSP to the destination. Each switch will pop off its label and send the packet to the next switch label in the sequence [8].

D. Traffic Engineering (TE)

TE can be considered as a mechanism that controls network traffic flows and provides performance by making optimal use of network resources [6]. Reservation of resources, fault-tolerance, and optimum utilization of resources can be considered as key features of TE [9]. To control network traffic, information obtained through the measurement, modeling and characterization of the network can be used to make effective use of the network. The main objective of TE is to improve traffic performance through efficient and reliable network operation with optimum use of network resources [10].

V. MPLS Technique

MPLS is an advanced technology for providing reliable services by speeding up network traffic and service quality through the use of DiffServ and MPLS TE to provide reliable transmission of real-time multimedia applications. However, traditional IP forwarding networks are not suitable for the provision of guaranteed services such as VoIP in relation to their mechanism technique, where the routing lookup is performed on the basis of L3 addresses at each routing node (router) in the network (IP address). Packet delay would also suffer from high and responsive applications such as VoIP and real-time applications. Thus, MPLS is a more efficient solution for providing stable end-to-end communication that can establish multiple network topology routes, but MPLS is unable to monitor traffic classifying per flow, manipulating traffic with the same actions, so using DiffServ with MPLS is efficient way to help real-time services that are vulnerable to delay, jitter, and packet loss, with high reliable connectivity [11]. The control-driven model is used by MPLS to initiate the assignment and distribution of label bindings for LSPs formation. By concatenating one or more label-switched hops, an LSP is created, allowing a packet to be forwarded across the MPLS domain from one label-switching router (LSR) to another LSR. The MPLS network architecture consists of the heart of the network's LSR and the edge of the label-edge routers (LER). In order to find the corresponding forwarding equivalence class (FEC) and LSP, the LER routers have the task of analyzing the IP header of each arrived packet, which facilitates the label swapping feature in LSR nodes. Packet forwarding, classification and QoS are determined by the labels and the service type (CoS) fields within the MPLS domain.

VI. Queuing

Different queuing mechanisms can be used by the router to manage requests using a queuing discipline. It is possible to design a queuing discipline to give priority to one traffic group and not to another. One of the fundamental factors that define the consistency of the service rendered is the management of the queuing system, as it provides a fundamental distinction between the service levels of the various traffic classes. The following are the queuing methods used with case study presented in this paper [12].

(1) *First in First Out (FIFO) queuing:* This is the queuing discipline which is most widely practiced. This method of queuing is implemented by very large numbers of network devices. The order in which the packet arrives is retained here and no preference is given for any packet. This means that every packet is not given any preferential treatment. This queuing method is also known as queuing first come first served (FCFS).

(2) *Priority Queuing (PQ):* After FIFO, this was the first queue to become popular. Priority queuing is based on the packet being listed into a service class. It is put into the required queue based on the classification. The scheduling of the queue is very simple; until there are no packets in higher priority queues, a lower priority queue will not be serviced.

(3) *Weighted Fair Queuing (WFQ):* Weighted Fair Queuing attempts to equally schedule packets based on the weights assigned to each traffic stream. So if small packet sizes are used by packets such as telnet and voice, voice can be given greater weight so that priority is given over the telnet packets. The flows are individually known as FIFO. This is very similar to queuing based on classes. If the bandwidth is not completely used, then depending on the weights, it is distributed among other flows.

(4) *Round robin (RR)* queuing: The new arrived packets are categorized and put in different queues in this queuing technique, all queues are set with the same weight, and all queues are polled in a cyclic order. However, if a non-empty queue is encountered, a single packet is transmitted from it, allowing the RR queuing technique to have maximum strength to deal equally with all queues [12].

(5) *Weighted round robin (WRR) queuing:* The queue is manipulated in a cyclic manner in proportion to its assigned weight, so some queues can be polled more frequently than others. However, this allows those queues to transmit more than one packet when scanned, as the number of packets transmitted is determined by the weight of each queue. OPNET tool implements modified *weighted round robin (MWRR) queuing* which employ class-based approach by assigning specific weight to each ToS of IPv6 with MPLS or IP4 classification according to DSCP [5] [6].

VII. Experimetial Study

In this performance evaluation study OPNET simulation tool [18] is employed to model and simulate the MPLS network with three different queuing techniques namely are (PQ, WFQ, MWRR) using three different routing protocols are (OSPF, EIGRP, ISIS); as shown in the design model shown in Fig. 1. The model contains wide area networks interconnected with a core MPLS-based network represented by 6 MPLS routers in the center of the model. However, in this experimental study there are nine different scenarios are evaluated with VoIP and multimedia streaming applications. The VoIP traffic is created between LANs of the model by using "create traffic flow" key-option of OPNET tool with the following input parameters: (call rate: 17 calls per hour for every, average call duration: 180s (3 min), voice flow duration: 3600s (60 min), with G.711 voice encoder, with interactive voice as ToS.

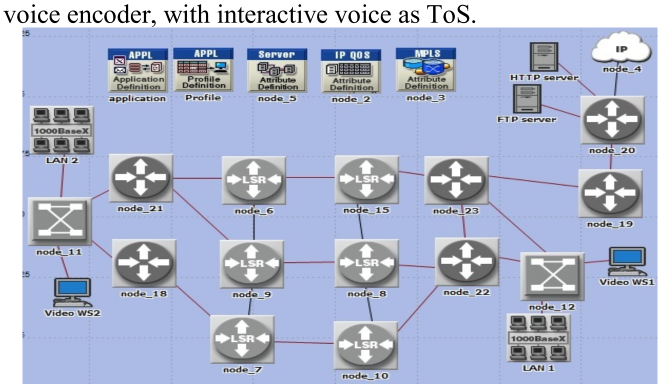

Fig. 1. Case study of MPLS-based network topology with Internet cloud

Video conferencing as a multimedia streaming application is generated between two video stations in the model over a wide area MPLS based network. For the sake of making the simulation more realistic HTTP and FTP applications are generated between the LANs of the given topology. However, the ToS summary is as follows: for video conference: (ToS=156 with 85.495 GB in 17 flows); for VoIP application: (ToS=220 with 287.616 MB in 17 flows); for FTP application: (ToS=92 with85.495 GB in 17 flows); for HTTP application: (ToS = Best Effort (0) with85.495 GB in 17 flows), as a result a total of 256.767 GB in 68 flows are injected inside the model which is a very huge data.

Therefore, the implemented scenarios are 9 scenarios:

(1) **3** scenarios for: MPLS with OSPF routing protocol, each with 3 queuing technique; all with VoIP, video conferencing, HTTP, and FTP applications.

(2) **3** scenarios for: MPLS with EIGRP routing protocol, each with 3 queuing technique; all with VoIP, video conferencing, HTTP, and FTP applications;

(3) **3** scenarios for: MPLS with IS-IS routing protocol, each with 3 queuing technique; all with VoIP, video conferencing, HTTP, and FTP applications;

Simulation results that show the effect of routing and queuing on QoS of multimedia applications illustrated:

VoIP application: impact of routing protocols (OSPF, EIGRP, ISIS) on QoS of MPLS network with PQ, WFQ and MWRR queuing techniques:

Fig. 2 illustrates the packet end-to-end delay performance parameter of VoIP application with PQ queuing for three routing scenarios. The results show that the packet end-to-end delay of ISIS routing protocol scenario is the best since it recorded the lowest end-to-end delay value nearly of 0.0600 second (red line), whereas the OSPF scenario reported the worst case with a value of upper that 0.0608 seconds (green line). But, all of them within the standard specified limits.

Fig. 3 depicts the jitter delay as a performance measure of VoIP application with PQ queuing for three routing protocols over MPLS network. The comparison results illustrate that the VoIP jitter with ISIS routing protocol scenario is reported better values which is very near to zero than EIGRP and OSPF protocols scenarios.

Fig. 4 demonstrates the packet end-to-end delay performance factor of VoIP application with WFQ queuing for three routing cases. The results describe that the packet end-to-end delay of ISIS routing protocol scenario with WFQ is the best where it registered the lowest packet end-to-end delay value which is slightly upper than 0.0600 second (red line), whereas the EIGRP scenario recorded the worst case with a value of upper that 0.0608 and reaching 0.0612 seconds (Blue line). But, all of them still committed to the standard specified limits.

Fig. 5 describes the jitter delay as a performance quantity of VoIP application with WFQ queuing for three routing protocols over MPLS network. The evaluation results show that the VoIP jitter with ISIS routing protocol case is informed efficient queuing delay results almost near zero value, whereas both EIGRP and OSPF cases reported negative delay; but still within the standard limits.

Fig. 6 shows the packet end-to-end delay performance parameter of VoIP application with MWRR queuing for three routing protocols. The results demonstrates that the packet end-to-end delay of ISIS routing protocol scenario with MWRR queuing discipline is the best where it reported the lowest packet end-to-end delay value which is almost equal to 0.0600 second (red line), however the EIGRP scenario recorded the worst case with a value of upper that 0.0608 seconds (Blue line). But, all of them still committed to the standard specified limits.

Fig. 7 illustrates the jitter delay as a performance factor of VoIP application with MWRR queuing for three routing protocols over MPLS network. Assessment results describe that the VoIP jitter with MWRR queuing discipline is almost the same as the WFQ queuing type where ISIS routing protocol case is recorded the results which almost near zero value, whereas both EIGRP and OSPF cases reported negative delay; but still within the standard limits.

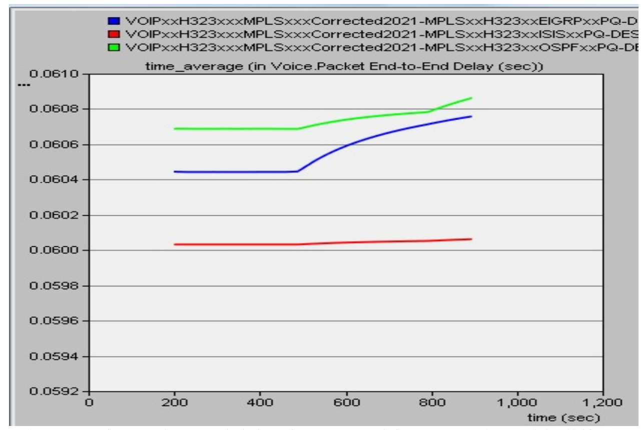

Fig. 2. Packet end-to-end delay for VoIP with PQ queuing with different routing protocols.

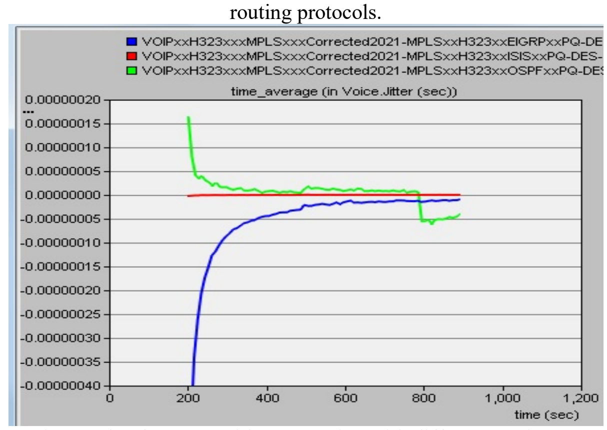

Fig. 3. Jitter Delay for VoIP with PQ queuing with different routing protocols

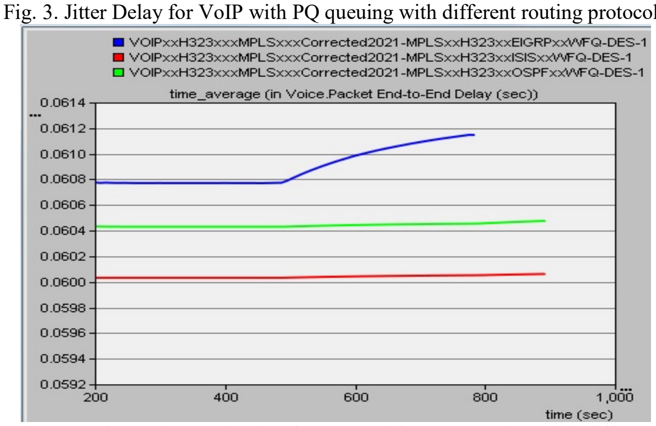

Fig. 4. Packet end-to-end Delay for VoIP with WFQ queuing with different routing protocols.

Video conferencing application: impact of routing protocols (OSPF, EIGRP, ISIS) on QoS of MPLS network with PQ, WFQ and MWRR queuing techniques:

Fig. 8 illustrates the packet end-to-end delay performance parameter of video conferencing application with different queuing disciplines (PQ, WFQ, MWRR) for three routing scenarios. The results describe that the packet end-to-end delay of EIGRP routing protocol with MWRR queuing scenario is the best since it recorded the lowest packet end-to-end delay value nearly of 0.0015 second (blue line). However EIGRP protocol with PQ and WFQ also reported better results with values which under 0.0020 seconds. Whereas the ISIS with MWRR (blue sky line) case reported the worst value that about 0.0033 value which is twice as that of EIGRP scenarios. Also, OSPF with MWRR queuing and OSPF with PQ queuing reported the second worst values which is around the 0.0027 seconds of packet end-to-end delay values.

Fig. 9 depicts the packet delay variation performance measure of video conferencing application with different queuing types (PQ, WFQ, MWRR) for three routing scenarios (OSPF, ISIS, EIGRP). The results clarifies that the packet delay variation of EIGRP routing protocol with WFQ and MWRR queuing scenarios reported the best values which are below 0.000002 seconds. Whereas, OSPF with (PQ, WFQ, MWRR) and ISIS with (PQ, WFQ, MWRR) recorded delay values around 0.000008 seconds, which 4 times higher that of EIGRP with WFQ scenario.

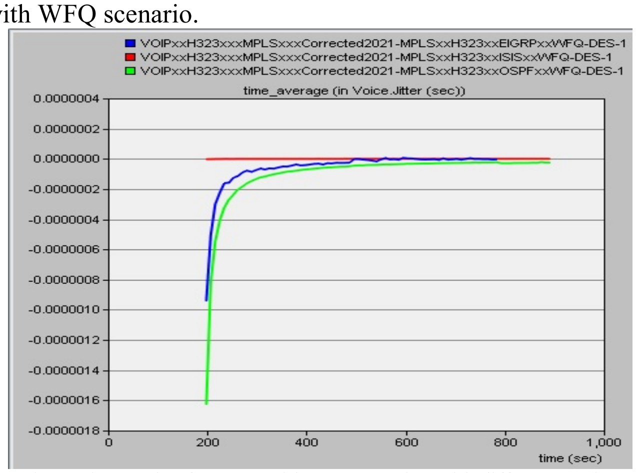

Fig. 5. Jitter Delay for VoIP with WFQ queuing with different routing protocols.

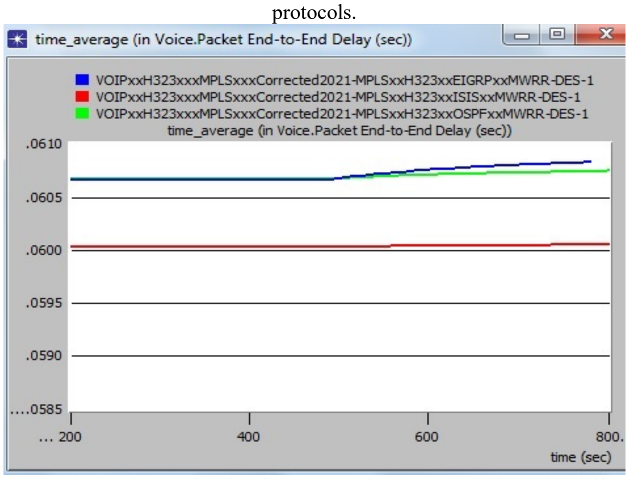

Fig. 6. Packet end-to-end Delay for VoIP with MWRR queuing with different routing protocols.

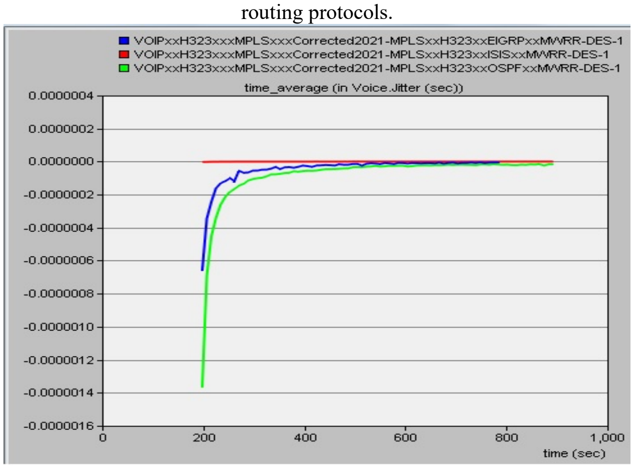

Fig. 7. Jitter Delay for VoIP with MWRR queuing with different routing protocols.

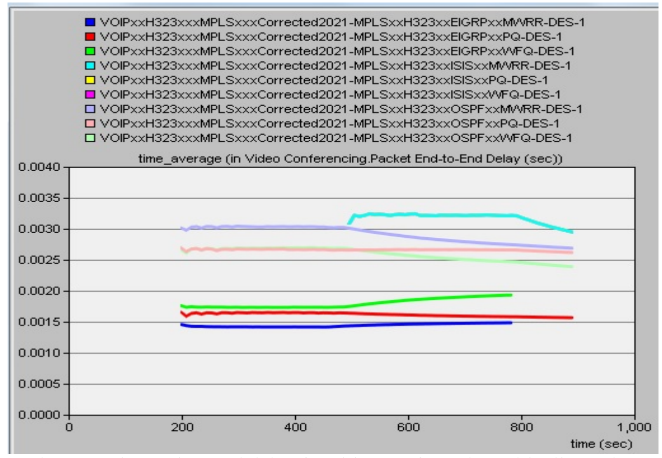

Fig. 8. Packet end-to-end delay for video conferencing with all queuing techniques and with different routing protocols.

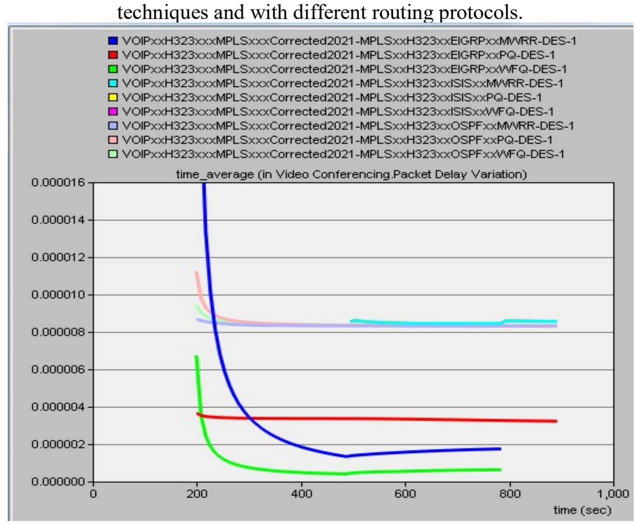

Fig. 9. Packet delay variation for video conferencing with all queuing techniques and with different routing protocols.

Different assessments

Fig. 10 describes the TCP segment delay performance parameter of all scenarios with all running applications described above. The results demonstrate that the TCP segment delay ISIS with MWRR queuing case reported the best TCP segment delay values around 0.00004 seconds. While the EIGRP with WFQ queuing recorded 0.000046 seconds; and EIGRP with MWRR recorded (0.00038) and EIGRP with PQ reported the (0.00030), which comes due to the speed and lowest end-to-end delay reported with video conferencing.

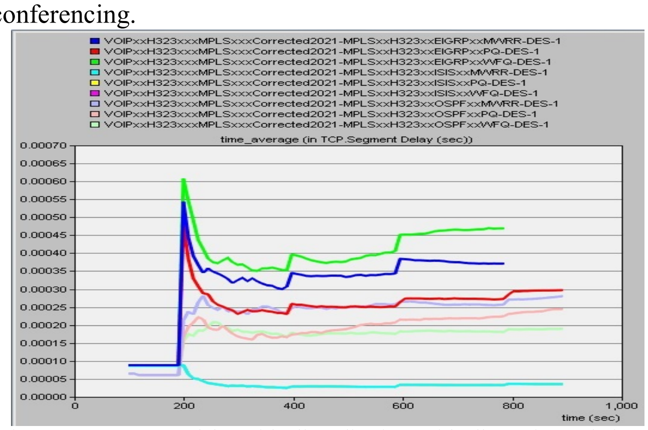

Fig. 10. TCP segment delay with all applications with all queuing techniques and with different routing protocols

Fig. 11 demonstrates the point-to-point queuing delay performance measure of all scenarios with all implemented applications described above. The results illustrate that EIGRP with WFQ reported the best point-to-point queuing delay which is around 0.0000005 seconds; this value is also recorded with ISIS MWRR case. While the OSPF with WFQ and MWRR queuing and ISIS with WFQ recorded the worst values which around 0.0000045 seconds.

Results analysis

From results reported above the IS-IS protocol is producing the least packet end-to-end delay for VoIP application when using PQ, WFQ and MWRR queuing techniques with MPLS networks, since it is registered also the minimum delay jitter for VoIP calls. However, when using video conferencing application with MPLS networks it is recommended to employ EIGRP as a routing protocol since it is registered the minimum packet end-to-end delay and the least packet delay variation with this application.

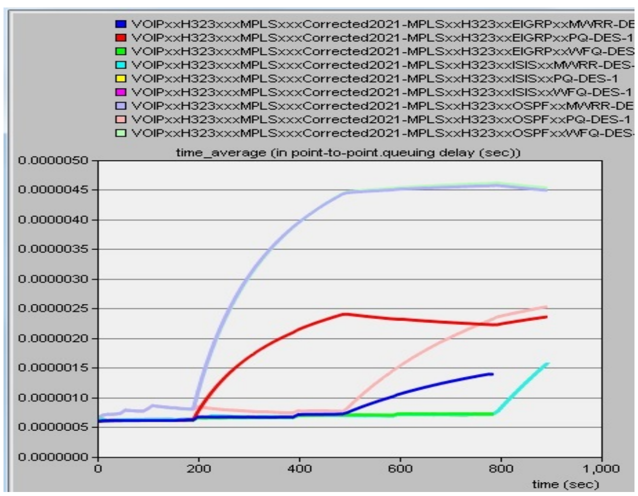

Fig. 11. Point-to-point queuing delay with all applications with all queuing techniques and with different routing protocols.

VIII. CONCLUSION

The simulation results presented that engaging of MPLS into IP networks largely enhanced multimedia streaming applications over IP networks by minimizing delay variation and end-to-end delay of streaming' packets because of the QoS enhanced tools brought by using MPLS technique. Consequently, well-observed that the MPLS carrying out VoIP with better performance when augmented with IS-IS as an essential routing protocol than the EIGRP and OSPF protocols; where delay jitter, delay variation and end-to-end delay of packet is diminished by using IS-IS protocol. In addition, EIGRP as routing protocol providing a good combination with MPLS to handle video conferencing over IP/MPLS networks.

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