Performance Evaluation of Multimedia over IP/MPLS Networks

Azeddien M. Sllame, *Member, IACSIT*, and Mohamed Aljafari

***Abstract*—This paper describes performance evaluation of multimedia data streaming over IP and MPLS networks using OPNET simulation tools. The experimental study is carried out by employing VOIP and video streaming applications in both networks with same parameters; but with different routing mechanisms. Therefore, same network model scenario is built with MPLS and TCP/IP networks by replacing MPLS routers by normal IP routers running OSPF routing and disabling MPLS functions. Furthermore, the evaluation process is done using three different queuing mechanisms; namely (FIFO, PQ, and WPQ) with the following performance parameters: the delay jitter (sec), packet delay variation, packet end-to-end delay (sec), and number of packets sent/received which indicates the traffic load, bandwidth and throughput for both networks. Results obtained are clear evidence that the MPLS networks are much appropriate for multimedia applications than conventional IP networks.**

***Index Terms*—MPLS, OPNET, evaluation, multimedia, networks.**

I. Introduction

Networks have become the most important part of today information systems. They form the backbone for information sharing in enterprises, governmental sites, and scientific clusters. That information can take several forms. It can be documents, data folders, data to be shared and processed by another individuals, and multimedia file streams. Multimedia data requires higher bandwidth than any other data types over the internet. Also, the number of Internet users growing rapidly and their usage to the multimedia applications are increasing quickly. Moreover, user are asking more advanced features of the multimedia applications which guided by mobile devices growth from the industry side. Therefore, this results in consuming more and more bandwidth. New technologies such as *dense wave division multiplexing* (DWDM) are evolving to meet these high bandwidth requirements being placed on the Internet. Besides the usage of DWDM networking techniques, some backbone networks based on *multiprotocol label switching* (MPLS) protocol are implemented in many countries to speed up the available Internet networks between different parts of the world.

Multimedia applications, such as internet telephony (i.e. *voice over IP* (VOIP)), video streaming and videoconferencing systems are very sensitive to variable delays and can tolerate some amount of packet loss during their transmission cycle in the Internet. This imposes the usage of the *quality of service* (QOS) concept to guarantee a specific QOS-level for real-time multimedia applications on the Internet. A QOS can be defined as a set of parameters that describe the quality (for example, bandwidth, buffering, priority, and CPU usage) of a specific stream of data. One idea behind the development of MPLS is to support the guarantee of QOS in existing IP and *asynchronous transfer mode* (ATM) networks. It was based on the observation that there exists a sequence of correlated packets for multimedia streams. Such streams are wanted to be processed in the same routing path by a uniform way and we did not want to repetitively examine all the headers of those packets. The observation showed that the headers in those related packets are the same or similar because those related packets in a stream desire consistent and similar processing actions. Hence, MPLS uses new technique to make short-term connection in a path for a sequence of correlated IP packets. MPLS now is considered as the guiding vehicle in the continued effort of developing so called multilayer switching. Generalized MPLS (GMPLS) extends MPLS to encompass time-division (for example, SONET/SDH), wavelength (e.g. DWDM), and spatial switching (for example, incoming port or fiber to outgoing port or fiber). The focus of GMPLS is on the control plane of these various layers to dynamically provision resources and to provide network with persist capability using protection and restoration techniques [1].

II. Multimedia over Internet

The multimedia traffic can be classified into:

- *Data traffic:* This is much more varied. It can be smooth or bursty, benign or greedy, or drop and delay-insensitive, and involves transmission control protocol (TCP) for send/receive acknowledgment and retransmit ion. Traffic patterns vary by application, and data classes must support several different priorities or application categories.
- *Voice traffic*: This is smooth, drop-sensitive, and delay-sensitive, and is typically UDP-based. Bandwidth per call depends on the particular codes adopted, sampling rate, and Layer 2 media employed. Voice quality is directly affected by all three QOS quality factors (loss, delay, and delay variation).
- *Video traffic:* This is bursty, bandwidth-greedy, drop-sensitive, and delay sensitive. IP-based videoconferencing has some of the same sensitivities as voice traffic.

Due to the fast increase use of multimedia applications over the Internet; different protocols that are supporting

Manuscript received March 29, 2014; revised June 24, 2014.

Azeddien M. Sllame is with the Faculty of Information Technology, University of Tripoli, Libya (e-mail: aziz239@yahoo.com).

DOI: 10.7763/IJCTE.2015.V7.972

multimedia in different aspects such as session initiation, session control, multimedia real-time transporting have been specified in the international standards. Among those protocols are: *session initiation protocol* (SIP), *real-time transport protocol* (RTP), *real-time transport control protocol* (RTCP), *real-time streaming protocol* (RTSP), and *resource reservation protocol* (RSVP) are commonly used to support the real time multimedia applications over the internet [2].

A. SIP Protocol

SIP is a client-server application layer protocol designed to address session control in distributed call control architecture, i.e. it initiates, maintains, and discusses session issues like bandwidth, hardware requirements and terminate sessions. SIP is designed specifically as a signaling protocol for Internet conferencing and VOIP; also it is used for event notification and instant messaging. Furthermore SIP supports user's mobility by using proxy and redirecting requests to the user's current location [1].

B. RTP Protocol

RTP is designed to run on top of a connectionless transport protocols such as UDP. UDP provides the multiplexing and checksum services to RTP packets. RTP designed to provide end-to-end delivery services for data that has real-time properties such as VOIP. Other services such as payload type identification, sequence numbering, time-stamping and delivery monitoring are also made available by RTP protocol [2], [3]. Such services are used by receivers to reconstruct the sender's packet sequence and to determine proper location of packets while playing the multimedia streams at the destination side. However, RTP does not guarantee delivery nor prevent out-of-order delivery. Another feature of RTP is that the encapsulation of multimedia streams by RTP is only seen at the end systems and it is not seen by intermediate routers. Hence, allowing routers at least provide the best-effort service if they are not employing other types of QOS services such as reservations made by RSVP protocol [4].

C. RTCP Protocol

RTCP is the control part of RTP. RTCP is used to gather end-to-end statistics about the flow and the quality of the session to each participant, i.e., the recipients is sending feedback to the source(s) to adjust (increase) the QOS, by limiting flow or using a different codec. This in turn, allows the RTP to concentrate on data-only communications between senders and receivers. Receiver/sender session status transmitted via RTCP contains the following information: last sequence number of a last packet received from various senders, observed loss rates from various senders, observed jitter information from various senders, member information (canonical name, e-mail, etc.), and control algorithm (limits RTCP transmission rate) [3].

D. RTSP Protocol

RTSP is a client-server application layer protocol used to control multimedia streaming sessions, i.e. *rewind*, *fast forward*, *pause*, *resume*, *repositioning*, etc. Server maintains session labels to look after the multimedia streaming flow with different clients. It establishes and controls either a single or several time-synchronized streams of continuous media such as audio and video. It does not typically deliver the continuous streams itself and it doesn't restrict how streamed media is transported (UDP or TCP possible) [3].

E. RSVP Protocol

RSVP is a network control protocol of type receiver-oriented reservation that allows data receiver to request a special end-to-end QOS for its data flows [2]. It is used to set up reservations for network resources and also in charge for maintaining router and host states to deliver the claimed service. Receivers employing RSVP protocol are responsible for choosing their own levels of QOS, initiating the reservation and keeping it active as long as it required by the application. Whereas, senders divide traffic in several flows, each is a separate RSVP flow with different level of QOS. Finally, RSVP provides multicast as a "first class" service [2], [3].

III. MPLS

MPLS is an emerging technology that guarantees reliable distribution of the Internet services with high transmission speeds and lower delays. The key feature of MPLS is its Traffic Engineering (TE), which is used for efficiently managing the networks for effective deployment of network resources. Due to lower network delay, efficient forwarding mechanism, scalability and predictable performance of the services provided by MPLS technology makes the most appropriate tool for fulfilling real-time applications requirements such as voice and video. Hence, MPLS has been evolved as a well-designed technique to deal with the bandwidth-management and service demands for next-generation IP–based crucial networks. That is because MPLS introduces a connection-oriented structure over connectionless IP networks with integrating of layer 2 switching with layer 3 routing. Moreover, MPLS can be present over existing asynchronous transfer mode (ATM) and frame-relay networks [4]. An MPLS-ready IP router forwards packets based on a label that is very similar in functionality to the VPI/VCI value carried in the header of an ATM cell. The label is a numerical value agreed upon two MPLS nodes to confirm a connection along label switching path (LSP). Moreover, the MPLS-ready router, known as a *label switched router* (LSR), maintains a table of labels. Then LSR forwards a packet based on the value of a label encapsulated in the packet. The most important protocol in MPLS technology is the Label Switching Protocol (LDP). The LDP protocol defined for distributing labels. LDP associates a *Forwarding Equivalence Class* (FEC) with each LSP it creates. The FEC associated with an LSP specifies which packets are mapped to that LSP. LSPs are extended through a network as each LSR maps incoming labels for an FEC to the outgoing label to the next hop for the given FEC. Therefore, the FEC is a set of packets that are treated identically by a router, i.e., forwarded out by the same interface with the same next hop and label, and assigned the same class of service. When a packet enters the MPLS domain at the ingress node, it is mapped into an appropriate

FEC. The mapping can be done according to a number of factors, i.e., the address prefix, source/destination address pair, or ingress interface. A group of IP packets that are forwarded over the same path and treated in the same manner and can be mapped to a single label by a LSR [4], [5].

IV. Quality of Service

A network that provides QOS is a network that offers certain assurance value for the delivery of packets. In a packet switched network, the quality may include packet transfer delay, delay variation, and packet loss ratio. In today's service delivery environment, all service providers are expected to offer personalized media-rich application services. In order to reduce operational costs and to enhance user experience, providers are migrating toward offering all killer application over a single IP/MPLS core infrastructure. QOS features enable network to handle traffic for efficient multi service delivery. The basic architecture of QOS introduces the three fundamental sections for QOS architecture implementation: (a) QOS within a single network element (for example, queuing, scheduling, and traffic shaping tools), (b) QOS signaling techniques for coordinating QOS from end to end between network elements, and (c) QOS policy, management, and accounting functions to control and run end-to-end traffic across a network [6]-[8].

A. Service Levels

Service levels refer to the actual end-to-end QOS capabilities, meaning the ability of a network to deliver service needed by specific network traffic from end to end. The services differ in their level of QOS requirements which describes how tightly the service can be guaranteed by specific bandwidth, delay, jitter, and loss characteristics. There are three basic levels of end-to-end QOS can be provided across a heterogeneous network:

1) *Best-effort service*; also known as lack of QOS. It is the original internet service. Makes best effort to transfer packets, but provides no guarantees. Best-effort service does not employ any prioritization scheme, hence, in case of congestion, any packet may be dropped.
2) *Differentiated service* (DiffServ); also called soft QOS; different priorities are assigned to different applications. Hence, some traffic is treated better than the rest (faster handling, more bandwidth on average, and lower loss rate on average). This is a statistical preference, not a hard and fast guarantee.
3) *Integrated service* (IntServ); also called hard QOS; an absolute reservation of network resources for specific traffic. In this class, the devices on the network through signaling can negotiate, request and adjust priority levels for different types of traffic based on the previously agreed values. However, RSVP protocol is deployed in the IntServ framework to implement per flow resource reservation and admission control [9].

Deciding which type of service is appropriate to deploy in the network depends on several factors:

1) The application or problem the customer is trying to solve. Each of the three types of service is appropriate for certain applications. This does not implies that a customer must migrate to differentiated and then to guaranteed service. It is depending on the customer application requirements.
2) The rate at which customers can realistically upgrade their infrastructures. This means the availability of the technology that enables employing guaranteed services instead of differentiated services.
3) The cost of applying and setting up guaranteed service is expected to be more than that for a differentiated service.

B. Fundamental QOS Features

To implement a QOS model, many QOS features are required. To achieve network QOS in general, and mostly for DiffServ QOS; the following features are vital: traffic classification, queuing and buffering, scheduling, rate limiting, and filtering [6]-[8].

- *Traffic classification*: in DiffServ as the traffic arrives at the access ingress, packets are classified into different forwarding classes, and within a forwarding class into high or low queuing priority. Traffic classification can be based on multiple header fields, i.e. packets may be marked with standard *type of service* (ToS) field markings in IPv4 header. At the subsequent nodes, the traffic is classified according to the standard marking present in the ToS field.
- *Queuing and buffering*: packets flowing through a node may wait before being serviced by a scheduler toward their corresponding destinations. The wait is expected, if the arrival rate of packets destined for a particular egress port is greater than the rate at which they leave. However, waiting in networking referred as queuing delay or latency. Moreover packets belonging to different classes of service are queued in distinguished queues. The packet belonging to a high priority traffic class is assured of buffering space. On the order hand, overflow may occur in the queues assigned to low priority traffic classes. There are four main types of queues that are used [8]:

1) *First in first out (FIFO):* it is one of the simplest techniques it consist of buffering and forwarding of packets in the same order of their arrival. FIFO queue type hasn't any priority or traffic classification schemas. However, in FIFO queue all packets are treated equally in the same way. When FIFO is used, some transmitting sources that are not well optimized can absorb all the available bandwidth. Moreover, bursty sources can cause delays in real-time traffic or important flows; hence this can cause dropping to real-time traffic or data since the less important traffic occupies the queue.
2) *Priority Queuing (PQ)*: with PQ, packets are classified to a certain priority class, then those belonging to higher priority class of traffic are sent before all lower priority traffic to guarantee their delivery in timing and prevents packets loss as much as possible. PQ is considered as a method of traffic differentiation, but it is priority classification schema is not optimal, since it affects handling of low priority queue's packets. Moreover, in the worst case, the lower priority queue may be prevented from sending its packets under limited bandwidth concerns, which will make starvation's

situation is possible [6]-[8]. This type is easy to realize but it isn't a max-min fair method, so it must be used with some other mechanism to control traffic into queues.

3) *Fair Queuing (FQ)*: using FQ the packets are classified into several groups, and each one has its own queue. This overcomes some of the FIFO and PQ limitations. However, in FQ method the regulation alternates service between the active queues (those queues whose have packets). Hence, active queues share the link equally.
4) *Weighted Fair Queuing (WFQ)*: in WFQ, the service is set according to the queue weight, i.e. each queue is given a slice from the link proportional to its prearranged weight. WFQ employs sorting and interleaving of individual packets by flow and then queue each flow based on the volume of traffic in this flow. However, by using this technique, larger flows are prevented from consuming network's bandwidth. [6]-[8]. However, WFQ is max-min fair technique and it provides some QOS control, and it is used in some industrial routers, but it is relatively complex to realize and it involves heavy computational overhead per packet in the flow.
5) *Round Robin (RR)*: in this approach the new arrived packets are classified and placed into different queues. However, all the queues are given the same weight. The queues are polled in a cyclic order, once a non-empty queue encountered one single packet from it is transmitted. RR technique gives a maximum effort to handle all queues equally.
6) *Weighted Round Robin (WRR)*: In this method the queue is treated proportional to its weight. In a cycle, some queues may be polled more frequently than others. Therefore, some queues, when polled, may be able to transmit more than one packet. The number of packets transmitted determined by the queue's weight.

- *Scheduling*: this function is done within a node where it decides the order in which queues allocated to different forwarding classes are serviced. Typically, queues that are assigned to high priority forwarding classes are serviced before queues belonging to low priority forwarding classes.
- *Rate limiting*: this is applied on streams in order to ensure customer traffic is conforming to a negotiated *service level agreement* (SLA), service providers may rate limit incoming traffic and drop nonconforming packets.
- *Traffic filtering*: this process is a network security measure. However, filtering is not obligatory function for a network service, but, traffic filtering, based on certain criteria, prevents some packets from flowing through the node. Hence, traffic filtering impacts the overall QOS that are provided in the networks.

C. QOS Parameters

QOS is quantitatively defined in terms of guarantees or bounds on certain network performance parameters. The most common performance parameters are the bandwidth, packet delay and jitter, and packet loss. Moreover, QOS makes a sense only if the network is up and running all the time, i.e. it is applied on reliable networks. On another hand, network throughput is the effective number of data units transported per unit time (e.g., bits/second). This parameter is usually specified as a "bandwidth guarantee". The bandwidth guarantee involves allocation of the link capacity as well as processing capacity of the intermediate nodes. A bandwidth bottleneck can put at risk the bandwidth guarantee for the entire path. The following are the details of the above mentioned parameters [6]-[8]:

1) *The packet delay*: is defined as the difference in the time at which the packet enters the network and the time at which it leaves the network; from sender to destination. Delay is also commonly referred to as latency. Each element through which a packet flows in a traffic path will increase the delay experienced by the packet. Moreover, it will impose a processing delay to the traffic flowing through them. From SLA perspective, the delay is the average fixed delay that an application's traffic will experience within the service provider's network. The packet delay composed of: i) Propagation delay**:** the time to travel across the network from end to end. It's based on the speed of light and the distance the signal must travel. ii) Transport delay**:** the time to get through the network devices along the path. iii) Packetization delay: the time for the codec to digitize the analog signal and build frames and undo it at the other end. iv) Jitter buffer delay**:** is introduced for a compensation of a jitter.
2) *Packet Delay Variation (Jitter)*: Jitter represents the variation in packet latency, and is sometimes called packet delay variation. Jitter is the variation in the network delay experienced by packets. More specifically, it is measured as the delay variation between two consecutive packets belonging to the same traffic stream. Although queuing is the main cause of traffic jitter, lengthy reroute propagation delays and additional processing delays can also affect traffic jitter. The jitter may be caused by i) variations in queue length; ii) variations in the processing time needed to reorder packets that are out of order arrived as a result of rerouting; iii) variation in processing time due to reassembling of segmented-packets.
3) *Packet loss*: it is the number of packets dropped in the path of a one way traffic flow between the sender and the receiver that may occur in a service network. However, packet-switched network does not provide mechanisms for reserving resources within the network on behalf of a particular packet "flow". Hence, packet loss is unavoidable under conditions of heavy and bursty loads with different streams using network resources in different ways. Even though network nodes equipped with buffer space to temporarily queue packets, the packet loss can't be eliminated. The following factors affect the packet loss: congestion, traffic rate limiting, physical layer errors, network element failures, and loss period and loss distance in the sequence transmission.
4) *Bandwidth*: it is the capability of the network to provide a better service to selected network traffic within TCP/IP networks. Therefore, bandwidth management provides proper priority to identified network traffic including dedicated bandwidth, controlled jitter and latency that is required by real time applications while improving

quality by reducing packet loss. This parameter's relationship with the QOS is becoming critical issue to enhance network performance in delivering real-time multimedia such as VOIP, News broadcasting over internet, and daily SMS updates of many software packages such antiviruses and multimedia applications. Of course there should be a trade-off between increasing bandwidth (storage) in network devices and developing efficient routing algorithms. Bear in mind that, robust and effective QOS deployment does allow maximum use of available bandwidth.

V. OPNET Simulation Tool

Communication systems are very complex structures. Due to that complexity and the cost concerns of building such systems; modeling and simulation is extensively used for the development/validation/enhancement of new or working communication architectures, and network protocols. Modeling is the process of producing a model; a model is a demonstration of the structure and working procedure of a system. The model should be a close approximation to the real system and includes most of its important features [9].One purpose of a model is to assist the analyst to predict the effect of different features of the system under concern. The model should not be so complex or difficult to understand or difficult to use experimentally. Simulation complements the theory and experimental studies and play progressively central roles in education and training fields. Simulation contribute to our understanding of how things function and are essential to the effective and efficient design exploration, evaluation, and increase understanding of the operation of new systems. Simulation results provide important information for developing and improving systems under design especially in very complex fields such as communication.

OPNET is a software product that can be applied in modeling and simulation of computer network. It allows researchers examining the application behavior and the background traffic of the designed networks [10]. However, applications such as VOIP and multimedia real time applications behaviors in Internet can be analyzed efficiently.

VI. Results

1) *VOIP application*: voice over IP is not only a way of voice communication. It is a full range of procedures that control call sessions i.e.; initiate, maintain and disconnect the data flows in different applications. In network technical words it is the technique of transmitting and routing the voice passing through the packet-switched networks. VoIP is transmitted by using the combination of RTP/UDP/IP protocols, while SIP or H.323 is used for session control. Moreover, RTCP protocol is used to allow monitoring of the data delivery and to control the flow and quality of data handled by RTP protocol. Even though TCP/IP is a reliable network protocol suite, it is not used in real-time communications because of its retransmissions with unbounded delays, it has no provision for time stamping, TCP congestion control has slow-start, and TCP does not support multicast [1]-[3]. However, normally, multimedia applications run RTP on top of UDP to benefit from UDP's multiplexing and check-sum services. In addition, there are many factors that affect the quality of voice e.g., the choice of codec, packet loss, packet delay variation (jitter), and packet delay, etc. For VoIP applications it is required that end-to-end packet delay shouldn't exceed 150*ms* to make sure that the quality of the established VOIP call is acceptable [3].
2) Multimedia streaming: data contains audio and video content ("continuous media"), can be of three classes; *streaming*, *unidirectional real time*, and *interactive real time* applications. Each class might be broadcast (multicast) or may be simply a unicast. While the networks use UDP to avoid TCP congestion control (delays) for time-sensitive streams; the client-side adaptive playout delay to compensate for delay and server side matches stream bandwidth to offered client-to-server path bandwidth by using any available techniques such as choosing different stream rate. Streaming of stored audio/video can tolerate higher delays considerably using initial buffering before playing back at the receiver end.
3) Fig. 1 shows the MPLS network topology illustrating node icons in OPNET which consists of the following network elements:
   a) Six router LERs (R1, R2, R3, R4, R5 and R6)
   b) Four routers LSRs (MPLS_R1, MPLS_R3, MPLS_R3 and MPLS_R4)
   c) Two VOIP stations (VOIP_West and VOIP_East)
   d) DS1 links (1.544Mbps) and 10Base T Links (10Mbps) are used for connecting all the routers with workstations.

To evaluate the MPLS-based networks performance it is needed to compare it with conventional TCP/IP same network topology. Hence, a simulation scenario is built also with TCP/IP network by replacing MPLS (LSR) routers by normal routers and disabling MPLS functions in LER routers in Fig. 1 and enabling open shortest path first (OSPF) routing. Moreover, the evaluation process is done using three different queuing mechanisms; namely (FIFO, PQ, and WPQ). The following performance parameters are used to perform the evaluation process: (The Delay (sec), Control traffic sent and received (packet/sec), Traffic Dropped (packet/sec), Jitter (sec), Packet delay variation, and Packet End-to-End delay (sec)) for each network with different queue techniques.

The model of both topologies in OPNET needs to define application profiles; namely VOIP application and video application profiles. In such profiles the designer should present some parameters for each application and associate the profile with specific nodes. Therefore, for VOIP workstations that are intended to make /receive calls need to enable VOIP application on them. However, the call volume was defined for the simulation as 1000 call with 300ms/call using G.711encoder with voice flow duration as 90000 seconds and the analysis includes delays overhead bytes of TCP/UDP/IP and creation of full mesh between all the topology nodes; hence, the OPNET created 240 voice traffic using traffic center to start the simulation; however, this indicates a huge data is used for the simulation run in the case study to investigate the network topology running the VOIP application comprehensively.

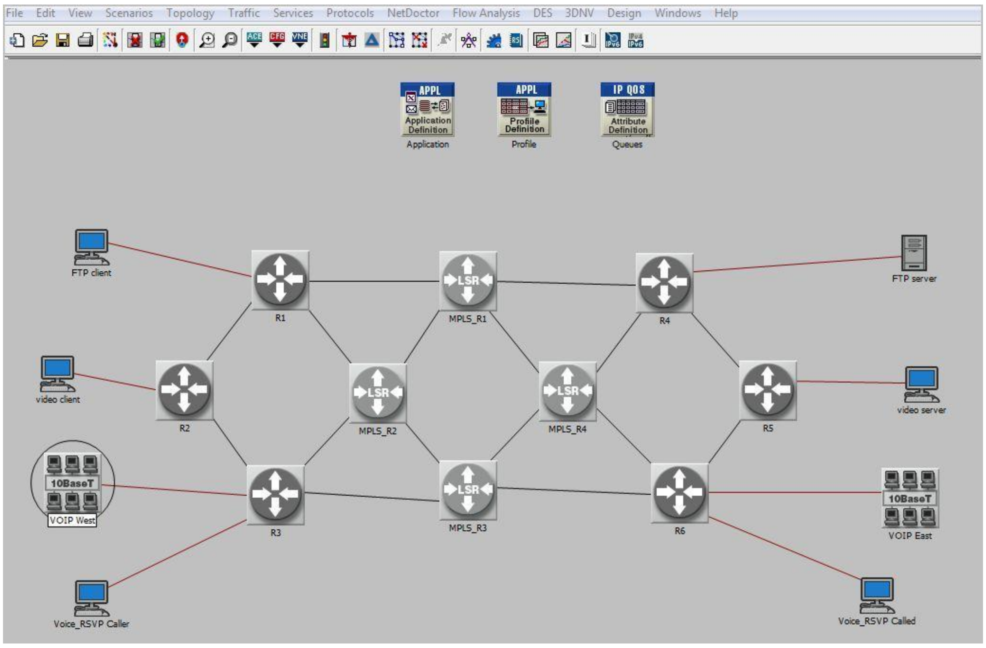

Fig. 1. MPLS simulation model.

In this section, some points are read-out from the given result figures is presented. However, in the exploration study of this paper, more than 100 figures were produced by OPNET simulator. Even though a large set of produced figures is presented in the paper, only some representative figures is described here in this section; because the conclusion of the result can be obtained directly from the figures; which describes the efficiency of MPLS-based networks over the IP-based networks in carrying different multimedia applications. The list of Fig. 2 to Fig. 15 is highlighted:

- Fig. 2 the analysis was for end-to-end delay for both MPLS and IP networks with PQ queuing technique. The IP network has much higher end-to-end packet delay than MPLS network. In IP network it goes higher than 0.35 values and approaching 0.4 in some points of the curve, while it is slightly above 0.20 for MPLS and its curve is with less oscillations.

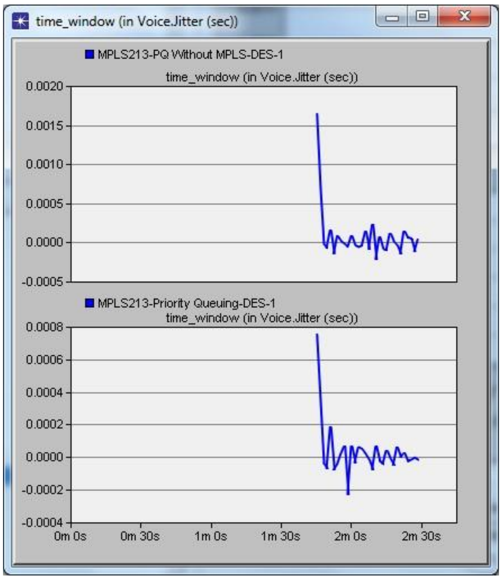

Fig. 3. Voice: Jitter (sec) for TCP/IP (UP) and MPLS (DOWN) with PQ mechanism.

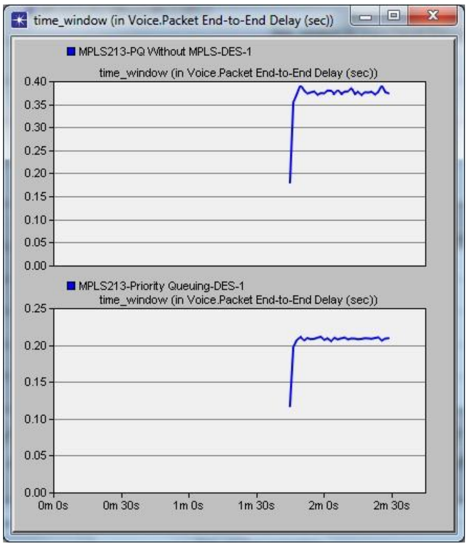

Fig. 2. Voice: Packet end-to-end delay (sec) for TCP/IP (UP) and MPLS (DOWN) with PQ mechanism.

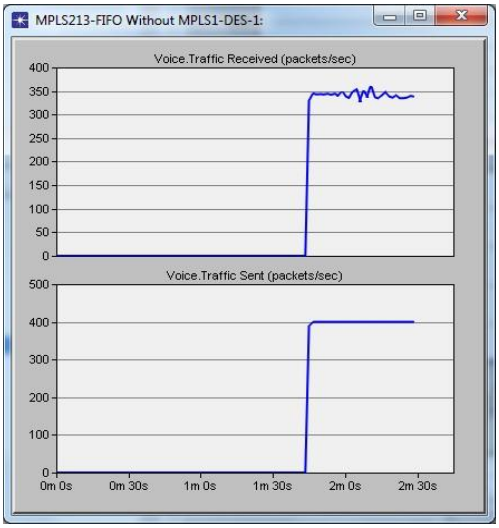

Fig. 4. Voice: Traffic sent and received (packets/sec) for MPLS with FIFO mechanism.

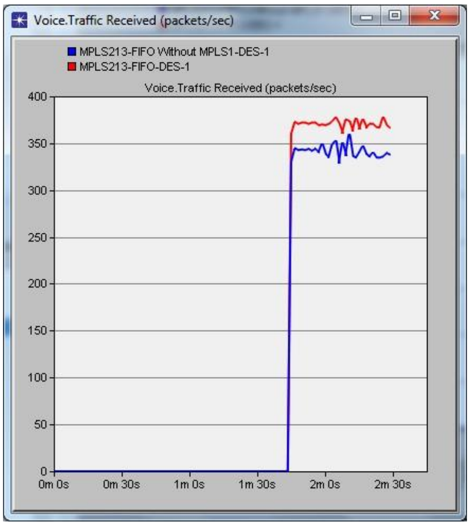

Fig. 5. Voice: traffic received (packets/sec) for TCP/IP (Blue) and MPLS (Red) with FIFO queuing mechanism.

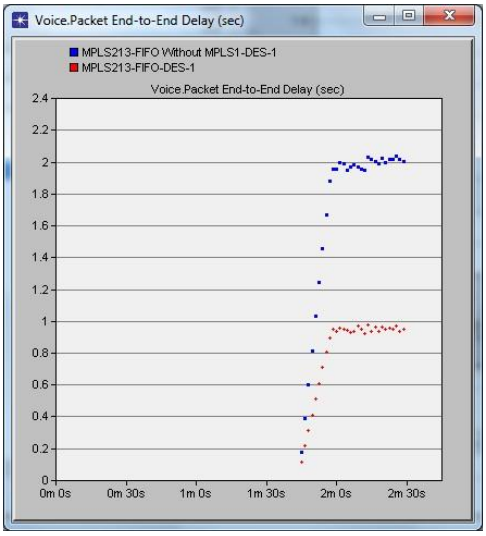

Fig. 6. Voice: Delay jitter (sec) for TCP/IP (Blue) and MPLS (Red) with FIFO mechanism.

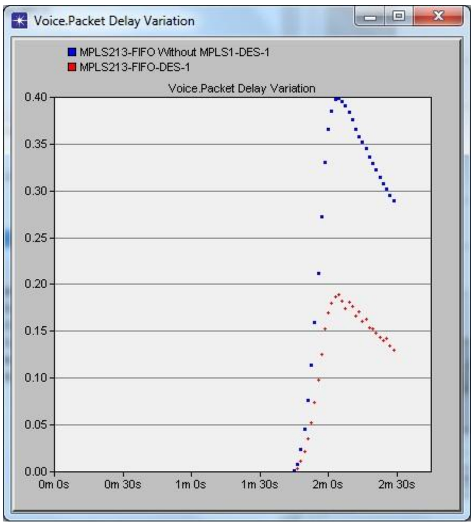

Fig. 7. Voice: Packet delay variation (sec) for TCP/IP (Blue) and MPLS (Red) with FIFO mechanism.

- Fig. 6 illustrates delay jitter (sec) for VOIP application in IP (Blue) and MPLS (Red) with FIFO mechanism. The figure is direct evidence that MPLS network has much less delay jitter than conventional IP network.
- Fig. 10 describes video conferencing application simulation results of packet end-to-end delay (sec) for TCP/IP (UP) and MPLS (DOWN) with WFQ queuing mechanism. The result clearly shows that MPLS has less end-to-end packet delay than IP network.

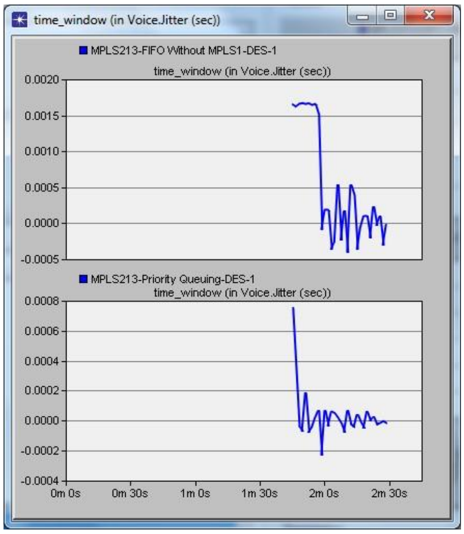

Fig. 8. Voice: Voice jitter (sec) for MPLS with (FIFO: UP) (PQ: DOWN).

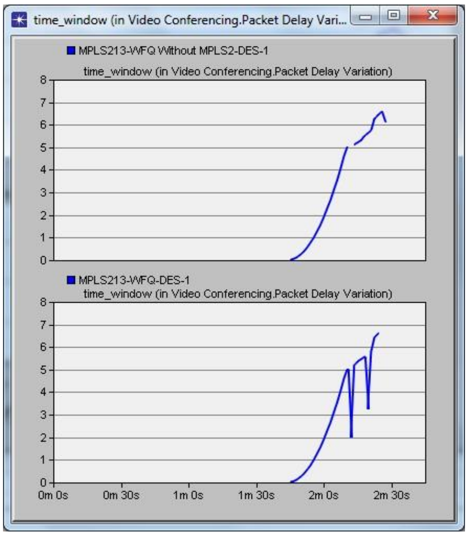

Fig. 9. Video conferencing: Packet delay variation for TCP/IP (UP) and MPLS (DOWN) with WFQ mechanism.

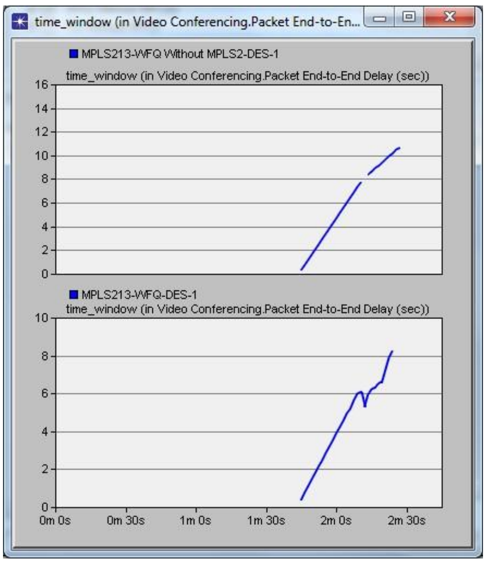

Fig. 10. Video conferencing: Packet end-to-end delay (sec) for TCP/IP (UP) and MPLS (DOWN) with WFQ mechanism.

- Fig. 11 shows video conferencing traffic received for IP (UP) and MPLS (DOWN) with WFQ queuing mechanism. The figure clearly states that MPLS network received higher data than the IP normal network.

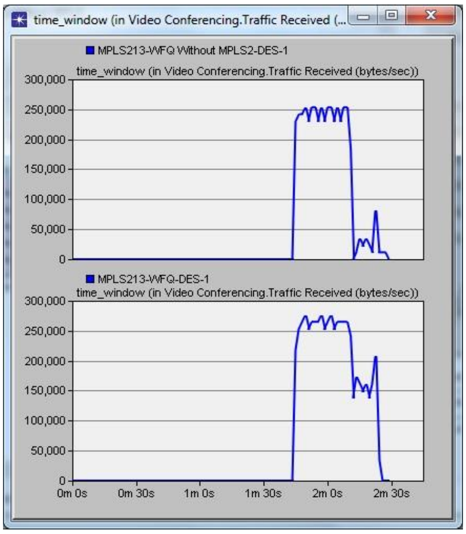

Fig. 11. Video conferencing: Traffic received for TCP/IP (UP) and MPLS (DOWN) with WFQ mechanism.

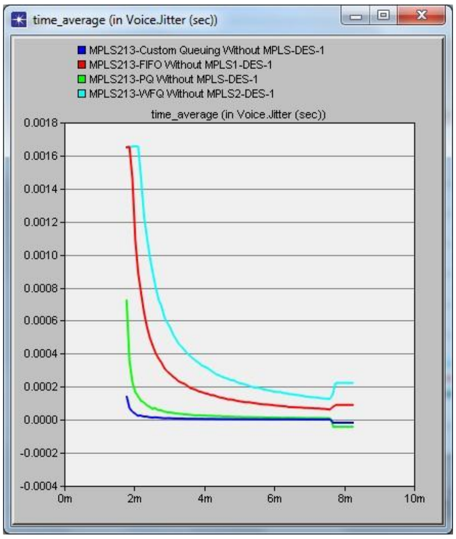

Fig. 12. Voice jitter (sec) for VOIP on the TCP/IP with (FIFO, PQ, WFQ, custom) queuing mechanisms.

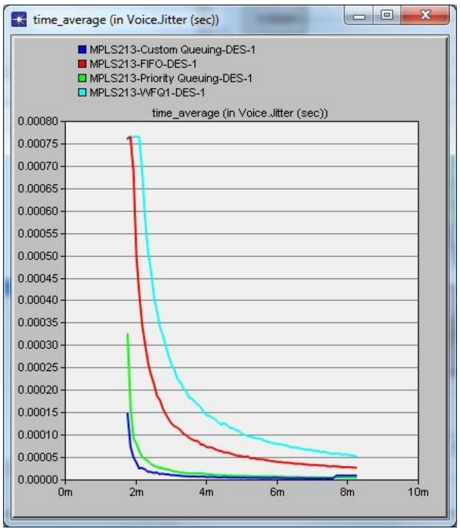

Fig. 13. Voice jitter (sec) for VOIP on the MPLS with (FIFO, PQ, WFQ, custom) queuing.

- Fig. 12 illustrates a comparison of voice jitter (sec) for VOIP on the IP network with (FIFO, PQ, WFQ, custom) queuing mechanisms. The figure shows that custom queuing (not standard) is better than other queues, but PQ queuing makes better results than FIFO and WFQ.
- Fig. 13 demonstrates voice jitter (sec) resuls for VOIP on the MPLS network with (FIFO, PQ, WFQ, custom) queuing. The figure confirms that custom queuing (not standard) is better than other queues, but PQ queuing makes better results than FIFO and WFQ.
- However, by comparing Fig. 12 with Fig. 13 we conclude that MPLS network performs much better than IP networks with the four types of queuing.

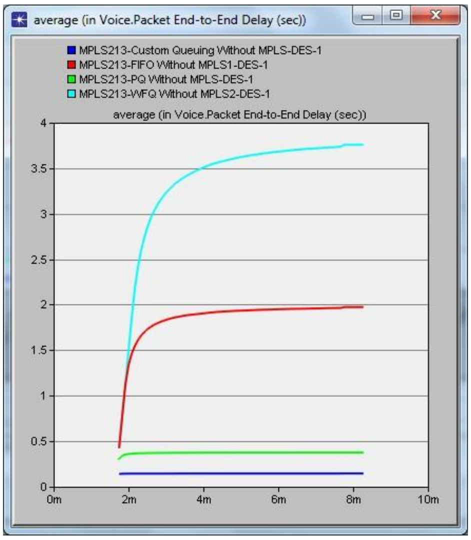

Fig. 14. Voice: packet end-to-end delay (sec) for VOIP on the TCP/IP with (FIFO, PQ, WFQ, custom) queuing.

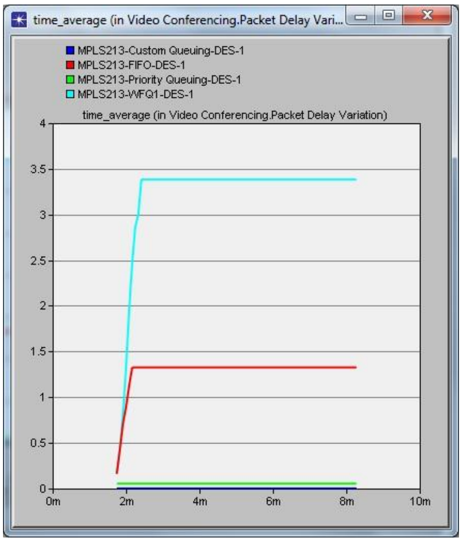

Fig. 15. Video conferencing: Packet delay varation on MPLS with (FIFO, PQ, WFQ, custom) queuing.

- Fig. 15 describes the results of video conferencing packet delay variation on MPLS network with (FIFO, PQ, WFQ, custom) queuing which shows that PQ outperforms FIFO and WFQ techniques.

VII. Conclusion

The main objective of the paper is to look at multimedia over IP/TCP and MPLS networks. During the experimental study we have been utilized OPNET simulation platform as a tool to carry out the analysis study. The VOIP and video conference streaming have been selected as a candidate applications because they are commonly used multimedia applications in the Internet. Moreover, both applications are employing most of the multimedia standard protocols that are mentioned in this paper. The analysis is made by focusing on the commonly used QOS statistics: packet delay variation, packet end-to-end delay, delay jitter, number of packets sent/received (indicates the traffic load, bandwidth and throughput), and effect of different queuing techniques (queuing delay).

The results clearly state that the MPLS based networks is much better in carrying multimedia applications that conventional TCP/IP networks.

REFERENCES

[1] L. Parziale, D. T. Britt, C. Davis, J. Forrester, W. Liu, C. Matthews, and N. Rosselot, "TCP/IP tutorial and technical overview," *IBM RedBook*, USA, 2006.

[2] W. Stallings, *Computer Networking with Internet Protocols and Technology*, Prentice Hall (Pearson Education), USA, 2004.

[3] J. F. Kurose and K. W. Ross, *Computer Networking: A Top-Down Approach Featuring the Internet*, 2012.

[4] H. G. Perros, *Connection-Oriented Networks SONET/SDH, ATM, MPLS and Optical Networks*, John Wiley & Sons Ltd, UK, 2005.

[5] E. Rosen, A. Viswanathan, and R. Callon, "Multiprotocol label switching architecture," *IETF RFC 3031*, p. 61, 2001.

[6] S. N. Bhatti and J. Crowcroft, "QoS-sensitive flows: Issues in IP packet handling," *IEEE Internet Computing*, pp. 48-57, July/August, 2000.

[7] J. Evans and C. Filsfils, *Deploying IP and MPLS QOS for Multiservice Networks: Theory and Practice*, Morgan Kaufmann Publishers, USA, 2007.

[8] R. Balakrishnan, *Advanced QoS for Multi-Service IP/MPLS Networks*, Wiley Publishing, USA, 2008.

[9] D. K. Pace, "Modeling and simulation verification and validation challenges," *Johns Hopkins APL Technical Digest*, vol. 25, no. 2, pp. 163-172, 2004.

[10] *OPNET Product Documentation v.11.0.A*, OPNET Technologies, Inc., Bethesda, MD, 2004.

Azeddien M. Sllame was born in Ben Ghshir, Libya. He received his B.Sc degree in computer engineering in 1990 from Engineering Academy, Tajoura, Libya. He earned his M.Sc degree in computer science and technology from Brno Technical University, Czech Republic in 1997. In 2003 he got his PhD degree in information technology from Brno University of Technology. He published more than 30 scientific papers in many international conferences and journals in the areas of designing of high-performance digital systems, system-on-chip, network-on-chip designs, computer networks and evolvable hardware systems. He is now an assistant professor with Faculty of Information Technology, University of Tripoli, Libya.