Egyptian Computer Science Journal ,ECS ,Vol. 36 No. 1, January 2012 ISSN-1110-2586

Enhancement of Voice Transmission over the Intranet

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Abstract

Traditional IP networks are based on an open architecture to support best effort applications. Transmission of Multimedia over IP networks has many limitations. These limitations affect service quality of Multimedia applications. Several technologies have been introduced to support Quality of Service (QoS), including Frame Relay (FR), Asynchronous Transfer Mode (ATM) and Multiprotocol Label Switching (MPLS). In this paper, a comparison of the use of these technologies when overlayed on IP infrastructure is introduced. The average delay and its variations (jitter) are the evaluation metrics on the basis of real time applications. Comnet III network simulator and the field measurements are the tools used for evaluating performance in this study. The study reveals that using Multiprotocol Label Switching (MPLS) over IP could be an adequate choice for multimedia transmission.

Keywords: *IPoFR, IPoATM, MPLSoATM, IPoMPLS, l3MPLS-VPN, QoS, Comnet III, MP Intranet.*

1. Introduction

In the past, there were two different networks for voice and data transmission. PSTN is used for voice transmission and Internet Protocol (IP) networks, (and other technologies) are used for data transmission. Nowadays, Internet Protocol (IP) networks (Converged Networks) are used for data and multimedia transmission. However traditional IP networks was mainly support best effort applications and provide little predictability of service, which is unacceptable for voice, video and future real-time applications. In order to support multimedia applications over IP networks, there must be a way that guarantees the Quality of Service (QoS). Multi Protocol Label Switching (MPLS) technology offers the Quality of Service (QoS) that guarantees data communication service as Frame Relay (FR) and Asynchronous Transfer Mode (ATM) do. MPLS is able to speed up the traffic flow by using labels and 3-bit class of service (CoS) field for providing different priorities for different traffic types. These advantages make MPLS play a key role in new converged networks that is used for different services.

The following section presents a survey on recent researches regarding multimedia transmission over different data networks. It addresses the following cases:

1. VoIP compared to VoATM

The authors show the major factors affecting the performance of integrated voice and data networks are QOS parameters and coding schemes. They used simulation tool (Opnet modeler 7) to predict delay and link efficiency. The simulation show that VoIP was found to deliver better performance compared to VoATM for a lower voice call density, while VoATM was found to be more robust at higher call volumes. In terms of the coding schemes to be used for voice encoding, G.729 with silence suppression gave the best results for both ATM and IP networks [12].

2. MPLS over ATM and IP over ATM methods for multimedia applications

By comparing MPLS over ATM model and IP over ATM model for multimedia application traffics with different quality of service (QOS) requirements using a simulation tool (Comnet III). MPLS over ATM method providing ABR, CBR, and VBR QOS support and IP over ATM method providing a primitive UBR QOS support. MPLS over ATM model and IP over ATM model are operated under varying offered loads to measure the average end-to-end delay and delay variation for each value of the load. The authors show that MPLS over ATM model has yielded not only better and lower average end- to- end delay and delay variation results than those of the IP over ATM model but also it has differentiated the multimedia traffics according to their required QOS support. Therefore, voice, video and data traffics experience different end- to- end average delays and delay variations due to fair and efficient use of ATM backbone resources offered with AAL1, AAL2 and AAL3/4 connections. The simulation results illustrate that the IP over ATM approach is insufficient to provides these applications with end-to- end connections with guaranteed bandwidth and required priority, leading to low and unpredictable average end to- end delay and end- to- end delay variations [4].

3. Voice over MPLS compared to Voice over other Packet Transport Technologies

This study shows the advantages of VoMPLS compared to voice transport over other technologies (VoATM, VoIP, and VoFR) from the viewpoints of bandwidth efficiency, and implementation issues in the access and backbone networks. VoMPLSoPPP is more efficient than VoMPLSoATM or VoMPLSoEthernet in the network backbone. VoAAL2oATM has intermediate efficiency, and VoIP is highly inefficient without header compression, but VoMPLSoPPP remains most efficient. By comparison, VoATM has an intermediate level of bandwidth efficiency, and VoIP is highly inefficient unless RTP multiplexing is used. The efficiency defined by the authors reflects the amount of headers related to total length of PDU. The smaller the amount of header the larger the efficiency [5].

4. Improving Triple Play Services using Multi Protocol Label Switching Technology

The final paper shows the effect of MPLS technology on triple play services (voice, data, video) based on the average throughput of the network, total number of packets received at destination nodes and packet loss rates. By comparing the performance of MPLS to that provided by the IP networks [6].

The aim of this study is to enhance voice transmission over Intranet by evaluating average end-to-end delay and end-to-end delay variation results for IPoFR, IPoATM, MPLSoATM, and IPoMPLS models.

The remaining sections in this paper are organized as follows:

Section 2 presents the methodology of the research. Section 3 presents the results of experiments. Section 4 presents the analysis of results and finally section 5 is concerned with the conclusion.

2. Methodology

This research is a quantitative research, where I run two experiments, one on Comnet III simulator (IPoFR, IPoATM, MPLSoATM, and IPoMPLS models) and the other on MP Intranet (IPoFR, and IPoMPLS) to select the best WAN technology for multimedia transmission among remote sites on MP Intranet.

Research methodology design: In this study, the practical work divided into two parts:

First, Modeling and Simulation of IPoFR, IPoATM, MPLSoATM and IPoMPLS scenarios in Comnet III 2.5.2 simulator. Providing Model, Network Configuration and Parameters and Results and Analysis from the viewpoints of average end-to-end delay and average end-to-end delay variation for each case.

Second, using Military Production (MP) Intranet as a practical network to compare average end-to-end data delay and average end-to-end data delay variation for two different cases: IPoFR and IPoMPLS.

Finally Comparison of Results of average end-to-end delay and delay variation for all Comnet III simulation Models and comparing the Resulting end-to-end data delay and data delay variation that produced from Military Production (MP) Intranet.

3. Results of Experiments

3.1 Simulation Environments

This part is concerned with experimentation of multimedia transmission over Intranet. The experiments are carried out using Comnet III network simulator, in addition to the use of actual network serving Ministry of Military Production.

1. Case 1: The Results of IPoFR model
2. Case 2: The Results of IPoATM model
3. Case 3: The Results of MPLSoATM model
4. Case 4: The Results of IPoMPLS model

The results of field tests address the following cases:

1. The Results of IPoFR network
2. The Results of IPoMPLS network

The objective of experimentations is the choice of the adequate infrastructure that allows the minimum delay and delay variation in multimedia (MM) traffic.

The simulation tool used in this research was based on Comnet III ver. 2.5.2 network simulator. In Internet Protocol over Frame Relay (IPoFR), Internet Protocol over Asynchronous Transfer Mode (IPoATM), Multiprotocol Label Switching over Asynchronous Transfer Mode (MPLSoATM), and Internet Protocol over

Multiprotocol Label Switching (IPoMPLS) models the Voice1, Video1 and Data1 message sources at Node1 introduce 5000 - 50,000 byte of data with an Exponential (1) seconds interarrival time, destined to Node2 across Frame Relay (FR) or ATM or MPLS network, and vice versa. Each data traffic is carried over ATM and MPLS cloud with a different quality of service required (CBR for voice, VBR for video, ABR for data). In MPLSoATM model, CrossComm XL5 Multiprotocol Routers and Xylan Omni-9Wx ATM switches are employed. The Routers and ATM switches in MPLS cloud are linked with OC-3(150 mbps) link. The links between the Routers and the nodes are IEEE 802.3Z Gigabit Ethernet with CSMA/CD.

Figure (1) shows the configuration used in simulation. The different technologies are defined as follows:

Table (1): Configuration of different technologies

Case	Router	Cloud
IPoFR	IP Router	Frame Relay
IPoATM	IP Router	ATM
MPLSoATM	MPLS Router	ATM
IPoMPLS	IP Router	MPLS

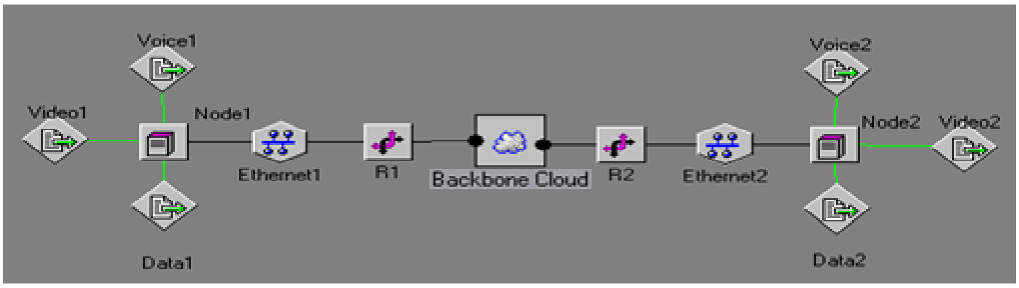

Figure 1: The simulation model

3.2 Simulation Results

1. The Results of IPoFR model

The IP over FR (IPoFR) simulation model given in fig. 1 show that Data1, Voice 1 and Video1 message sources at Node1 introduce 5000 – 50,000 byte of data with an Exp (1) seconds interarrival time, destined to the Node2 across the Frame Relay (FR) network, and vice versa. Data transferred within FR network through dual permanent virtual circuit (PVC), one used from Node1 to Node2 and the other one used from Node2 to Node1. Average end-to-end delay and average end-to-end delay variation results obtained from figures (2),(3) respectively show that, average data delay and delay variation has greater values than in case of voice and video average delay and delay variation. Average end-to-end delay results for voice and video traffics are 400 ms for offered loads less than 37,000 byte. This is acceptable value for voice quality which is tolerable to (50 – 400 ms), but above 37,000 byte load the results is not acceptable.

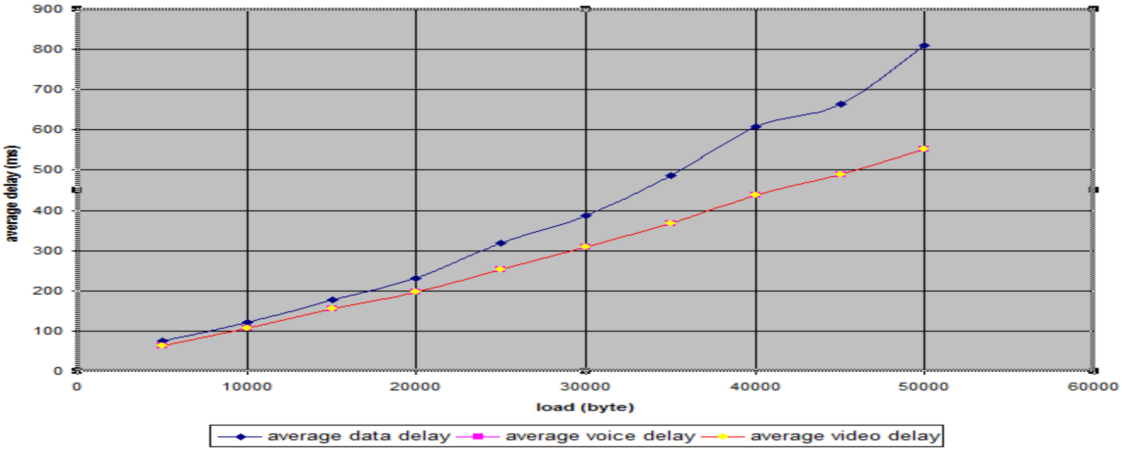

Figure 2: IPoFR average end-to-end message delay (ms)

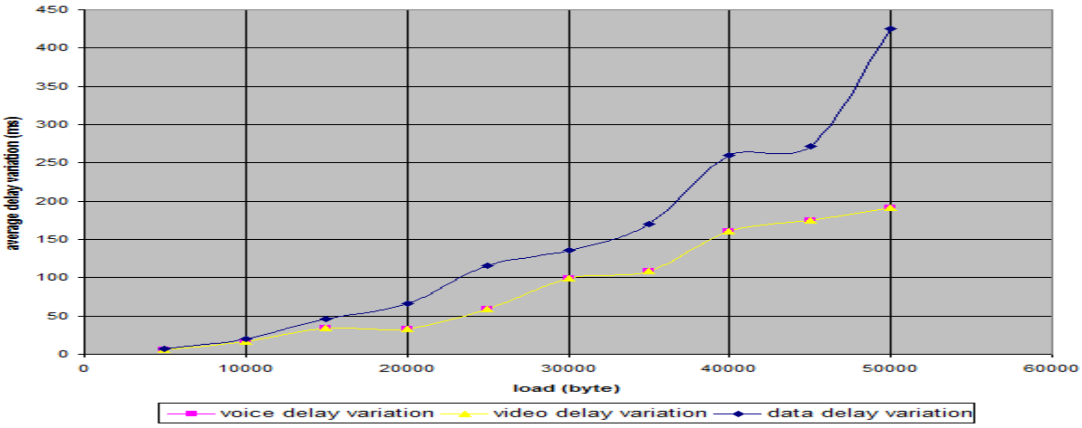

Figure 3: IPoFR average end-to-end message delay variation (ms)

2. The Results of IPoATM model

The quality of service (QoS) parameters for traffic sources are ABR with data source, CBR with voice source and VBR with video source. Average end-to-end delay and average end-to-end delay variation results obtained from figures (4),(5) respectively show that, average data delay and delay variation has greater values than in case of voice and video average delay and delay variation. Average delay and average delay variation increased with increasing load. Average end-to-end delay results for voice and video traffics are 325.29 ms for offered load equal 50,000 byte. This is acceptable value for voice quality which is tolerable to (50 – 400 ms) end to end delay.

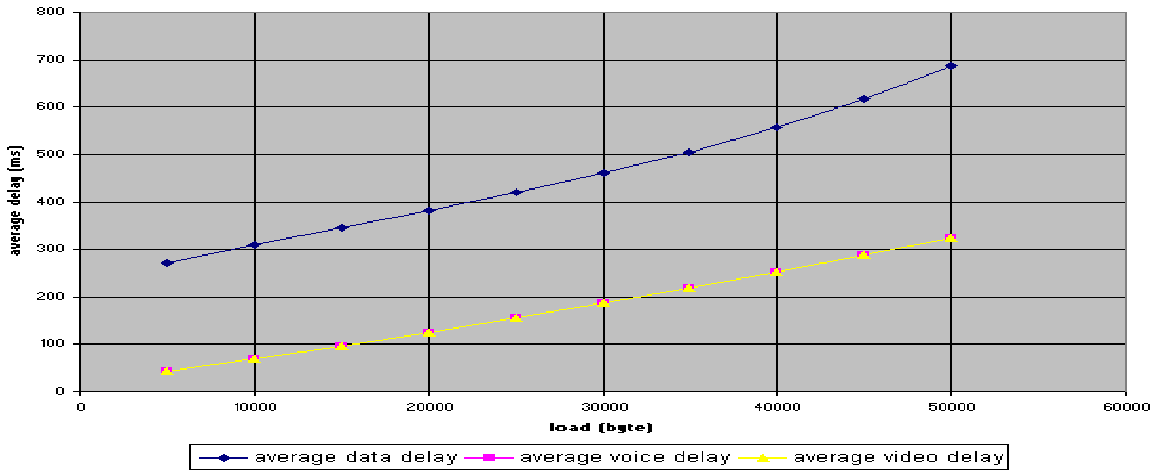

Figure 4: IPoATM average end-to-end message delay (ms)

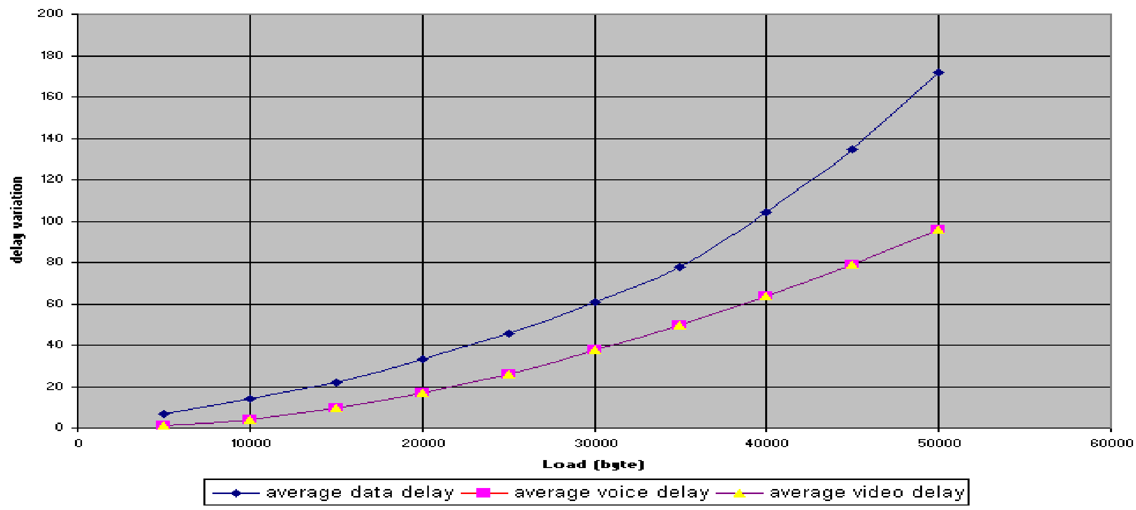

Figure 5: IPoATM average end-to-end message delay variation (ms)

3. The Results of MPLSoATM model

Each data traffic is carried over the ATM cloud with a different quality of service required (with ABR for data, CBR for voice, and VBR for video). LSR1 (LSR2) ingress edge router adds 4 byte MPLS labels to all data1, voice1 and video1 packets before they enter MPLS network. Then these packets are transferred over ATM cloud supporting the required level of QOS. MPLS labels are removed at the LSR4 (LSR3) egress router, which completes the Node1 to Node2 multimedia traffic transfer over MPLS network. Average end-to-end delay and average end-to-end delay variation results obtained from figures (6), (7) respectively show that, Average delay and average delay variation increased with increasing load. Average data delay and delay variation have higher values than voice and video delay and delay variation. Average end-to-end delay results for voice and video traffics are 310.307 ms for offered load equal 50,000 byte. This is acceptable value for voice quality which is tolerable to (50-400 ms) end to end delay.

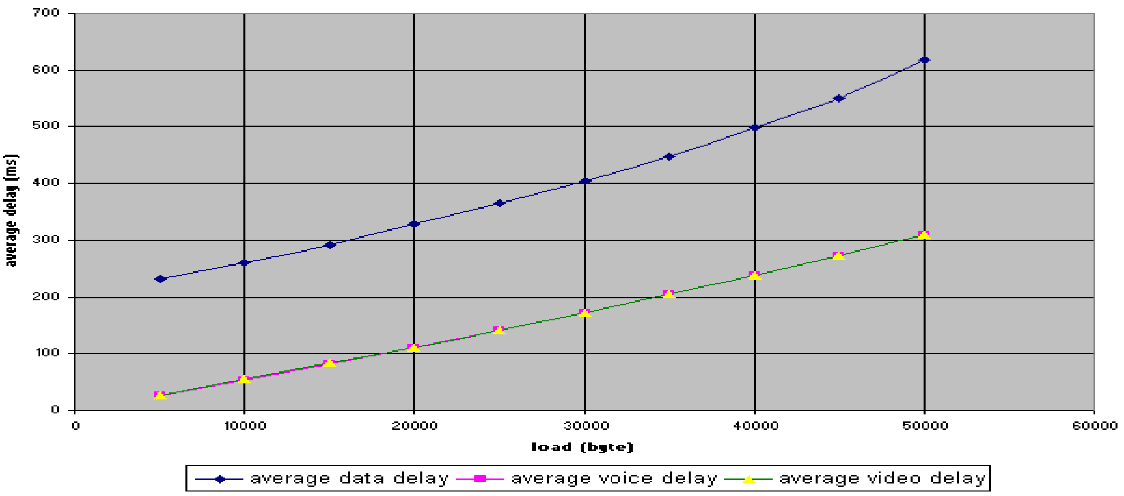

Figure 6: MPLSoATM average end-to-end message delay (ms)

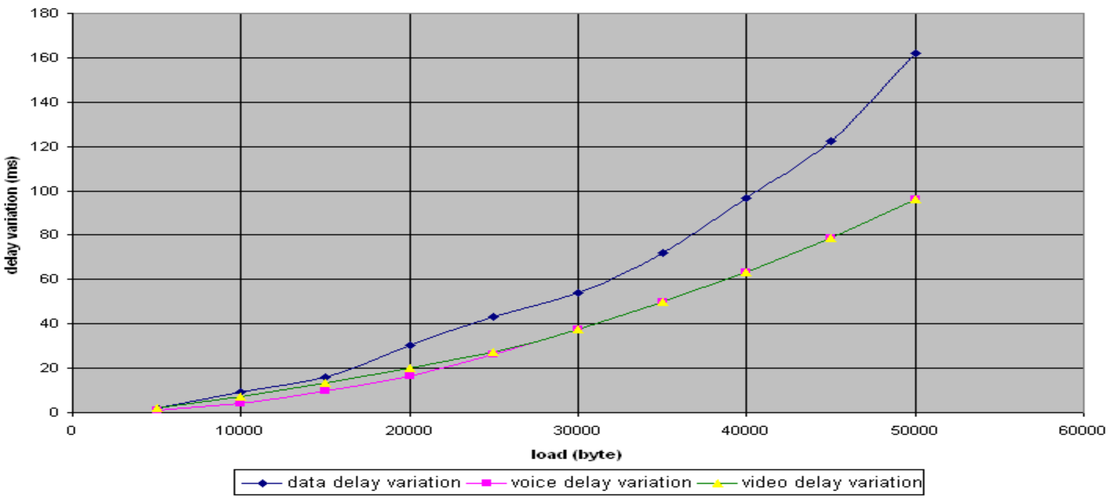

Figure 7: MPLSoATM average end-to-end message delay variation (ms)

4. The Results of IPoMPLS model

Each data traffic is carried over the MPLS cloud with a different quality of service required. Data 1&2 or Voice 1&2 or Video 1&2 are used to transfer traffic from node1 to node 2 and vice versa. Starting with the Data1 message source, which generate data stream and give it to transmission control protocol (TCP) that run on Data 1 message source. TCP puts a sequence number, error check and application port number on the segment (segment consists of data plus TCP header), passes this segment to internet protocol (IP) that runs on Data 1 message source. IP adds the source and destination IP addresses to form an IP packet, which is put in an 802.3 Ethernet LAN frame with Mac address of R1. R1 extracts the IP packet and compares destination IP address with its routing table then find the best path to destination (Node2) then reencapsulate packet in a frame and forward it to MPLS LSR1 ingress router. LER1 examines the IP address on the packet and decides what forwarding Equivalence Class (FEC) the packet belongs to, and implements its decision by labeling the packet with a 20-bit label identifying the FEC. It then does a table lookup to determine what is the router packets with this label are forwarded to, and transmits the labeled packet in a frame on the appropriate label switching path (LSP). When packet reach LSR3 in MPLS core, LSR3 extract the packet then looks at the label and performs a table lookup to determine where to forward it and what priority to give it. When labeled packet is reached LSR4 egress router, it removes the label from the packet and uses IP routing to send the packet in an Ethernet frame to R2. The R2 router decapsulate the frame and looks in its address resolution protocol (ARP) table to find out what Mac address is currently assigned to the destination IP address. When frame reach Ethernet network its switches uses Mac address to forwarding frame to its destination (Node2). The same process takes place when using voice or video source but using user datagram protocol (UDP) as a transport protocol instead of TCP in case of data transmission. Average end-to-end delay and average end-to-end delay variation results obtained from figures (8), (9) respectively show that, Average delay and average delay variation increased with increasing load. Average data delay and delay variation have higher values than voice and video delay and delay variation. . Average end-to-end delay results for voice and video traffics are 310.12 ms for offered load equal 50,000 byte. This is acceptable value for voice quality which is tolerable to (50-400 ms) end to end delay.

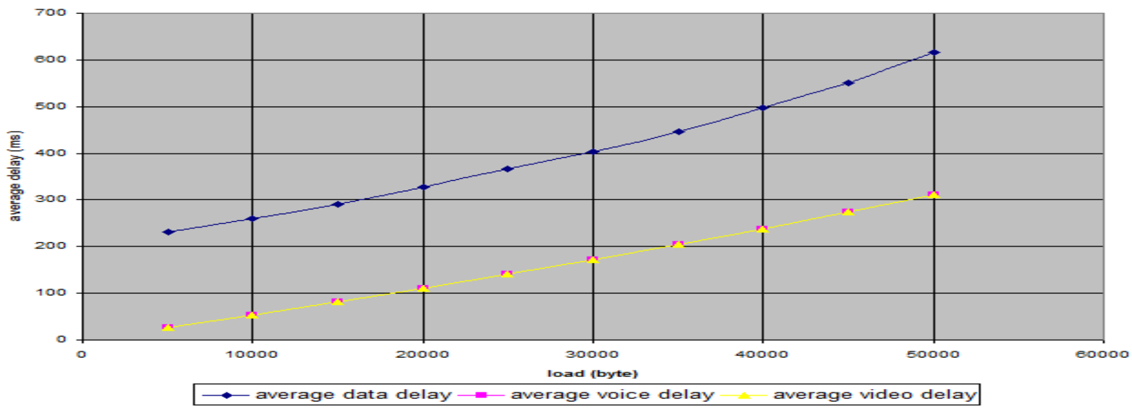

Figure 8: IPoMPLS average end-to-end message delay (ms)

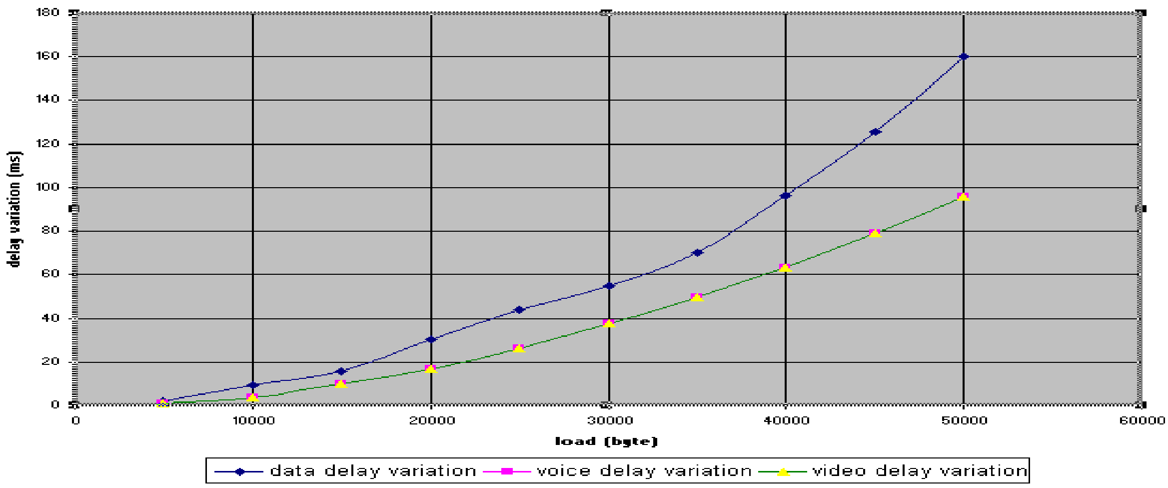

Figure 9: IPoMPLS average end-to-end message delay variation (ms)

3.3 Results of Field Tests

1. The Results of IPoFR network

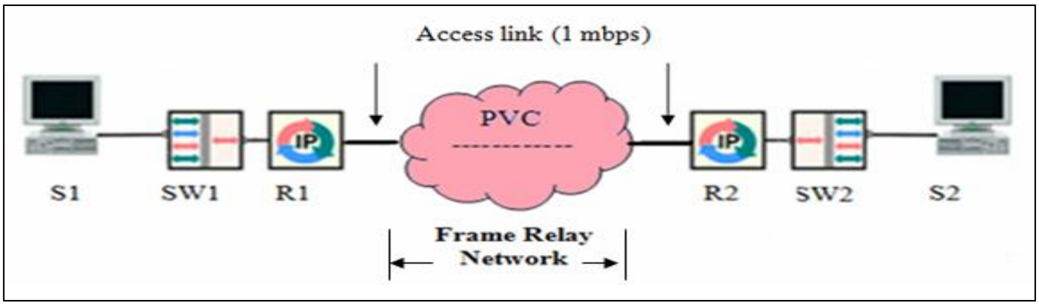

Figure 10: IPoFR network configuration

The IP over FR (IPoFR) Network Scenario given in fig. 10 show that data source S1 generate traffic that designed to S2 data source. Data is encapsulated by Station1 (S1) in an Ethernet frame that contains Mac address of Ethernet interface of router1 (R1) in frame header as the destination Mac address. Data is transferred through unshielded twisted pair (UTP) cable until reach Ethernet switch SW1. SW1 decapsulate Ethernet frame and extract destination Mac address then compare it with its Mac address table to find the switch port that is used to forward Ethernet frame to R1. when frame reached R1, R1 decapsulate packet extract destination IP address from packet header and compare this IP address with its OSPF (open short path first) routing table to find the best path for data transfer and the corresponding exit router interface that can be used to forward packets to its destination. After finding the exit interface for data transfer, R1 decapsulate packet in frame relay frame and forward frame to frame relay switch that is connected to R1 through access link. Encapsulation type on router1 (R1) exit interface (serial 0/0) is frame relay. DLCI number on R1 is 16 and DLCI number on R2 is 17. Two DLCIs are used to identify PVC that is used to carry data from R1 to R2 through frame relay network. Each FR switch in frame relay network using its table to route frames. The table matches an incoming port- DLCI combination with an outgoing port-DLCI combination to select the correct exit port to transfer frame. After frame reach R2, R2 make the same steps of R1 and forward frame out of its Ethernet frame that is connected to SW2. SW2 make the same steps of SW1 and finally switching Ethernet frames to station (S2) and complete the end to end data transmission. Average end-to-end delay and average end-to-end delay variation results obtained from figures (11), (12) respectively show that, average data delay is increased with increasing load. Average end to end delay range from 29 to 100 ms for the corresponding loads from 100 to 1400 byte. Maximum delay value is 100 ms which is acceptable for data transmission over frame relay network without any packet loss. Load traffic generation is controlled on S1 or S2 data sources by using ping command, ping –l size (byte) destination IP address –n iteration number, where size is range from 100 to 1400 byte in this case.

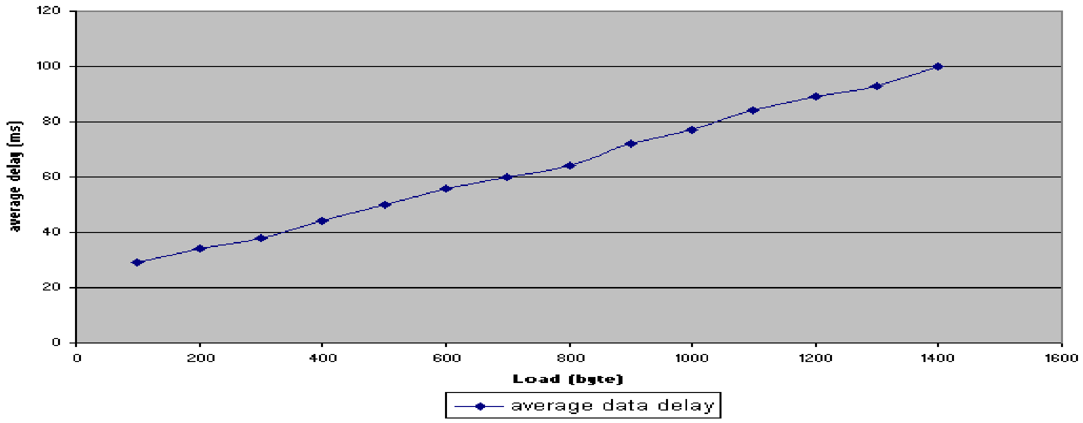

Figure 11: IPoFR average end-to-end message delay (ms)

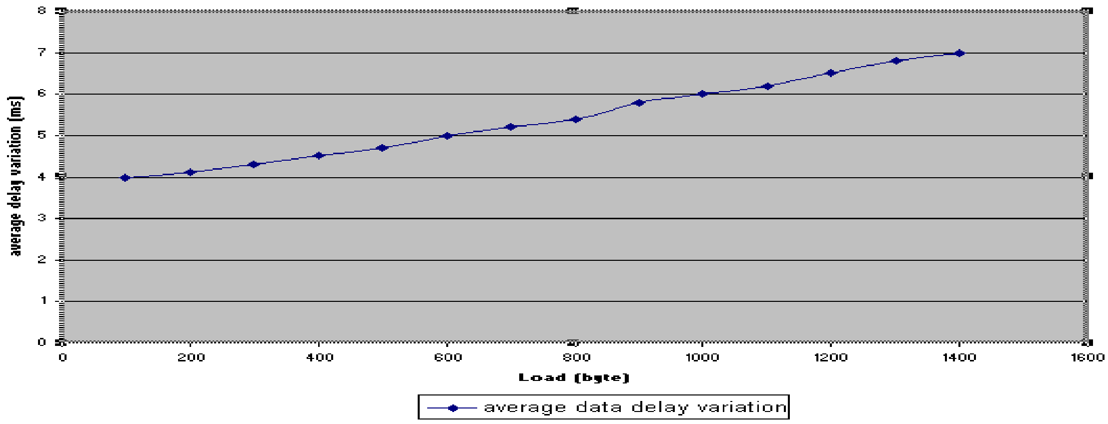

Figure 12: IPoFR average end-to-end message delay variation (ms)

2. The Results of IPoMPLS network

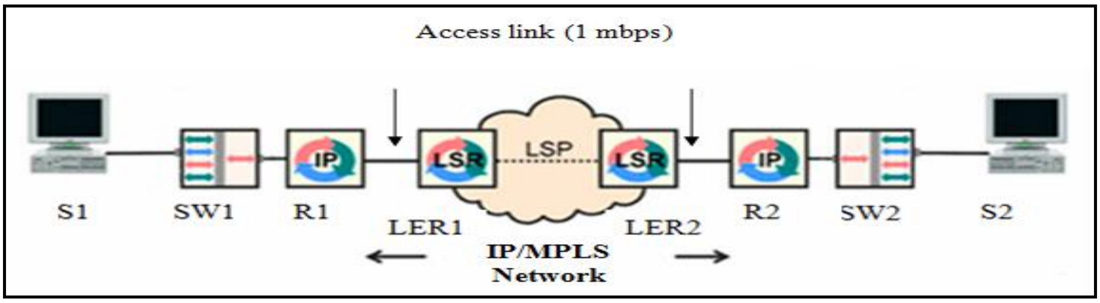

Figure 13: IPoMPLS network configuration

The IP over MPLS (IPoMPLS) Network Scenario given in fig. 13 show that data source S1 generate traffic that designed to S2 data source. Data is encapsulated by Station1 (S1) in an Ethernet frame that contains Mac address of Ethernet interface of router1 (R1) in frame header as the destination Mac address. Data is transferred through unshielded twisted pair (UTP) cable until reach Ethernet switch SW1. SW1 decapsulate Ethernet frame and extract destination Mac address then compare it with its Mac address table to find the switch port that is used to forward Ethernet frame to R1. L3 MPLS-VPN network run using packet forwarding (Data Plane). In this data plane, when data reaches router1 (R1) it originates a data packet with the source address of 192.168.1.2 and destination of 192.168.81.3. when packets reaches label edge router1 (LER1) it receives the data packet and appends the VPN label V1 and LDP label L2 and forwards the packet to the first LSR within MPLS network that is direct connected to LER1. The first LSR within MPLS network after LER1 receives the data packet destined to 192.168.81.3 and swaps LDP label L2 with label L1. The second LSR within MPLS network that is connected to the first LSR receives the data packet destined to 192.168.81.3 and pops the top label (L1). The resulting labeled packet with VPN label V1 is forwarded to LER2 (assume that there are only two LSRs inside MPLS network core). LER2 pops the VPN label and forwards the data packet to R2 where the 192.168.81.0 network is located, so data can reach S2 through SW2. Average end-to-end delay and average end-to-end delay variation results obtained from figures (14), (15) respectively show that, average data delay is increased with increasing load. Average end to end delay range from 8.5 to 41.5 ms for the corresponding loads from 100 to 1400 byte. Maximum delay value is 41.5 ms which is excellent value for data transmission over MPLS network without any packet loss. Load traffic generation is controlled on S1 or S2 data sources by using ping command, ping –l size (byte) destination IP address –n iteration number, where size ranges from 100 to 1400 byte in this case.

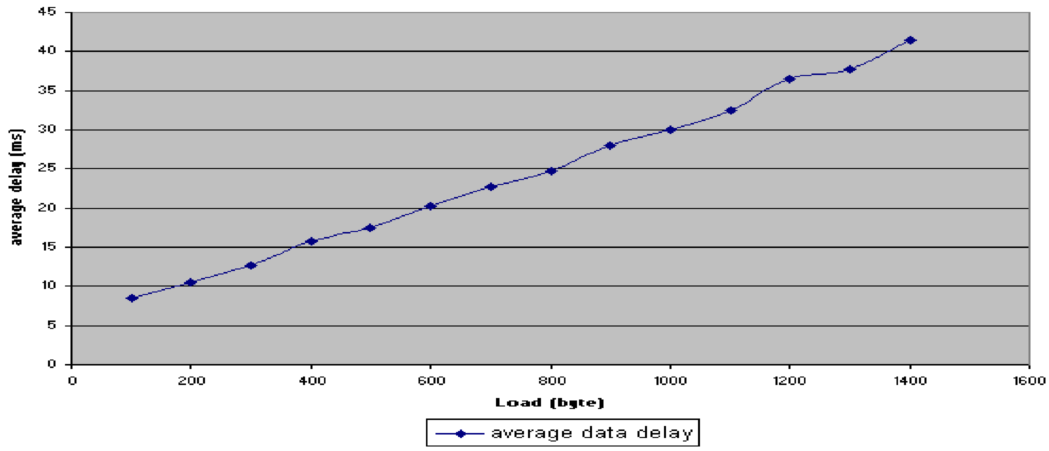

Figure 14: IPoMPLS average end-to-end message delay (ms)

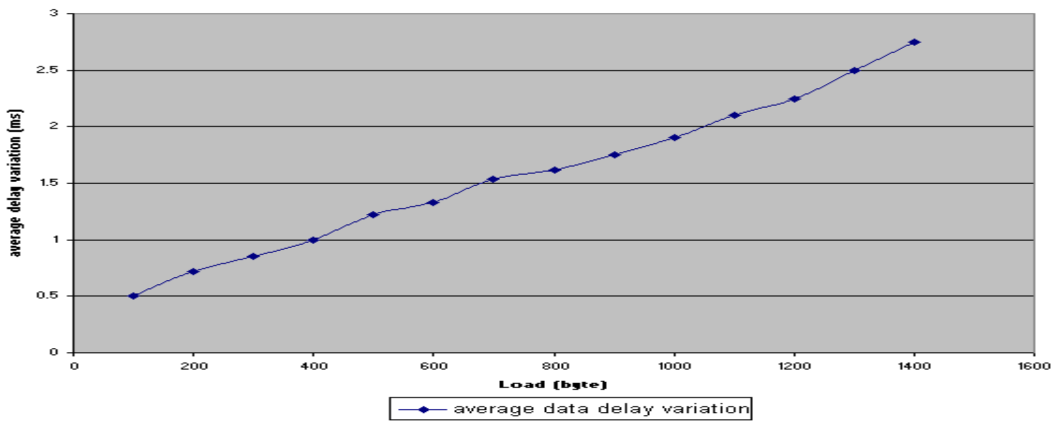

Figure 15: IPoMPLS average end-to-end message delay variation (ms)

4. Discussion

In this section an analysis is conducted to evaluate both simulation and field experiments results.

Comnet III simulation

Results of IPoFR model showed that, Results of voice and video traffics are better than data traffic because of using user datagram protocol (UDP) for voice and video transmission which is fast connection less protocol, less overhead than TCP (UDP OH = 20 byte, TCP OH = 40 byte), not using flow control to control traffic flow and do not send any acknowledgement message from destination to source to ensure correct data received. Beside transport protocol factor, QOS parameters are another important factor that gives better results for voice and video traffic transmission. Voice and video delay and delay variation results are similar because both of them using UDP as transport protocol. Frame Relay network are affected by CIR, B_c, and B_e values of Frame Relay network.

Voice and video traffic results in IPoATM model are better than data traffic because of using user datagram protocol (UDP) for voice and video transmission. UDP is fast connection less protocol, less overhead than TCP (UDP OH = 20 byte, TCP OH = 40 byte), not using flow control to control traffic flow and do not send any acknowledgement message from destination to source to ensure correct data received. Voice and video delay and delay variation results are similar because both of them using UDP as transport protocol. Average data delay of IPoATM is larger than that of IPoFR, because ATM protocol header has larger length than Frame Relay protocol header.

The Results of MPLSoATM model show that, Results of voice and video traffics are better than data traffic because of using user datagram protocol (UDP) for voice and video transmission which is fast connection less protocol, less overhead than TCP (UDP OH = 20 byte, TCP OH = 40 byte), not using flow control to control traffic flow and do not send any acknowledgement message from destination to source to ensure correct data received. Voice and video delay and delay variation results are almost the same because both of them using UDP as transport protocol. It is clear from the results that MPLSoATM model give better average end-to-end delay and delay variation results for data, voice and video traffic compared to IPoATM model. Good results are obtained because MPLS and ATM networks both of them support QOS parameters. MPLS using a 3 bit COS field for differentiation among data, voice and video traffics, ATM network using different class of service (COS) for different traffic types. But in case of IPoATM model IP protocol do not support QOS parameters and ATM cloud using unspecified bit rate (UBR) which equivalent to IP best effort (no QOS) for all traffic types (data, voice and video). Beside that MPLS by using labels and ATM by using permanent virtual circuits (PVCs) are considered the fastest traffic switching technologies which contribute to less end-to-end delay and delay variation results.

The Results of IPoMPLS model show that, Results of voice and video traffics are better than data traffic because of using user datagram protocol (UDP) for voice and video transmission which is fast connection less protocol, less overhead than TCP (UDP OH = 20 byte, TCP OH = 40 byte), not using flow control to control traffic flow and do not send any acknowledgement message from destination to source to ensure correct data received. Voice and video delay and delay variation results are almost the same because both of them using UDP as transport protocol. It is clear from the results that IPoMPLS model give better average end-to-end delay and delay variation results for data, voice and video traffic compared to IPoATM model. And almost the same results as MPLSoATM model. Good results are obtained because MPLS network support QOS parameters. MPLS using a 3 bit COS field for differentiation among data, voice and video traffics, so give the highest priority to voice then video and the lowest priority to data traffic. ATM Network also support QOS parameters as in MPLS Network, so similar results are obtained in both cases.

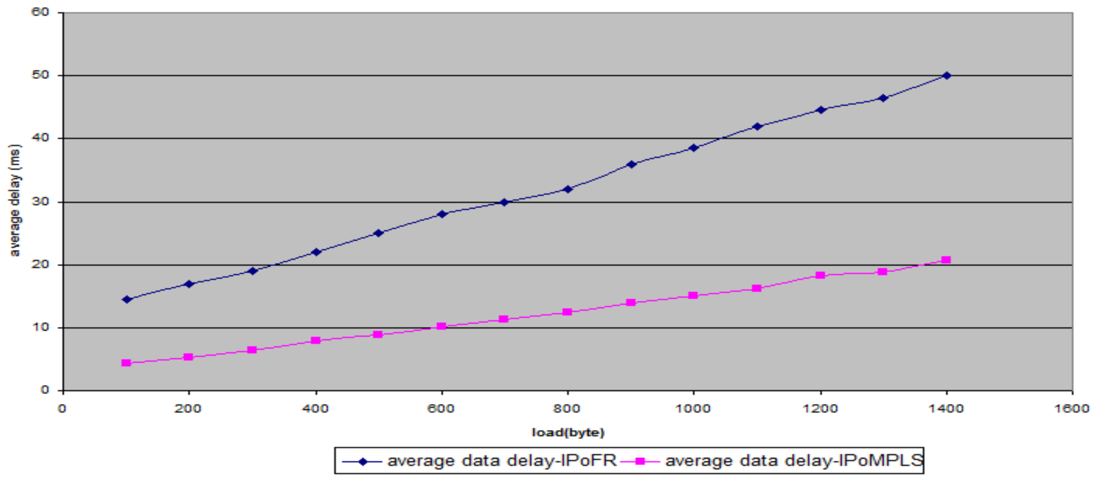

Figure 16: Comparison of IPoFR and IPoMPLS average end-to-end data delay

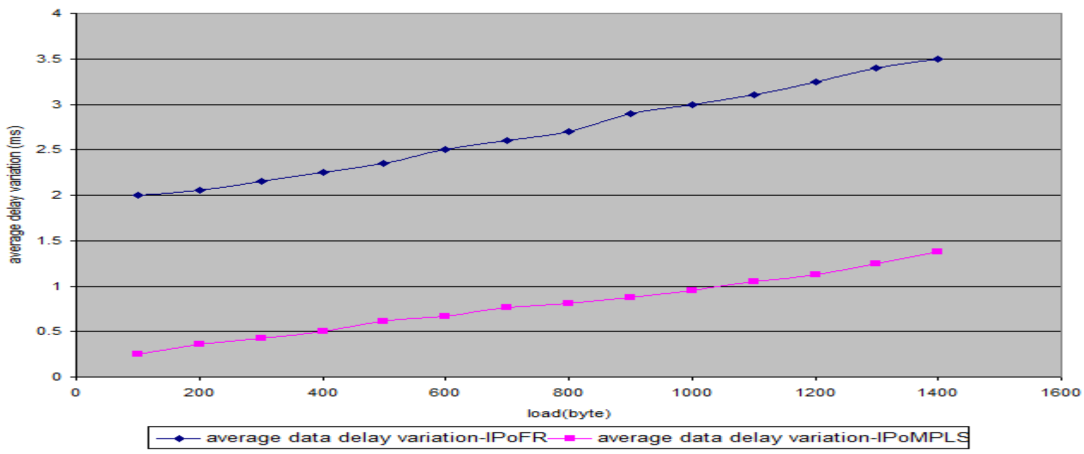

Figure 17: Comparison of IPoFR and IPoMPLS average end-to-end data delay variation

4.2 Field Experiments

The practical experiments on both IPoFR and IPoMPLS Networks show that, average end to end delay and delay variation results of IPoMPLS model is better than that of IPoFR as shown in figures (16, 17). The process of data transmission within MPLS network that shown in figure (13) is quite different from data transmission within Frame Relay (FR) network. At the incoming edge of MPLS network LER1 (ingress router) receives the data packet from R1 (IP router) and appends Virtual Private Network (VPN) label V1 and Label Distribution Protocol (LDP) label L2 and forwards packet to the first LSR (core router) within MPLS network. All core routers switching packets depending on its VRF (VPN routing and forwarding) tables. LDP uses VRF tables to distribute labels among core routers through

LSPs (label switching pathes). LER2 (egress router) remove labels from packet and map it to its destination IP address. IP routers (R1 & R2) using destination IP address and routing table that contain routes (network IP addresses) to find best path to destination (S2). Looking up at VRF tables in case of MPLS network for certain label is faster than look up in routing table in case of Frame Relay network(with R1 & R2) for certain destination IP address. MPLS network with its fast packet switching, QoS support for different traffic types and redundant LSPs is better than FR network that do not support QoS and LSPs. so Enhanced results of IPoMPLS can summarized in the following reasons:

- In MPLS forward packets based on something called a label instead of using the destination IP address, this provide fast switching and less time consuming.
- L3 MPLS-VPN provides full mesh connectivity among remote sites by using only one link for each site.
- In case of IPoFR model, every Router has its own routing table and look up in it to find the path to destination depend on the destination IP address. This look up process consuming time for every packet to reach its destination.
- In case of IPoMPLS model by using L3 MPLS-VPN to connect remote sites, every Router give a label to each route in its routing table and look up the MPLS Label Table instead of routing table to select the best path to destination. This label look up process is faster than routing table look up and not consuming time.

5. Conclusions

From Comnet III Simulation models, IPoFR, IPoATM, MPLSoATM and IPoMPLS the average end to end delay and delay variation Results show that, MPLSoATM and IPoMPLS methods are most suitable for voice and video transmission among Remote Sites. The two methods give similar results and have the same QoS. But in case of IPoMPLS, MPLS Network nodes are shared among several customers that are connecting their remote sites through one SPN (Service Provider Network). This issue is a security problem rather than using a virtual circuit (VC) across private network (over access link and through SW1 & SW2 to R1 & R2 to emulate VPN function. In case of MPLSoATM (Overlay model), the provider (ATM or FR Switch) does not participate in customer routing. It provides only point to point links (PVC or SVC) for customer data transport. Also using MPLSoATM for data transmission requires full mesh of virtual circuits between all customer sites for optimal connectivity. N sites need N (N-1)/2 circuits so increase complexity of network. Beside the previous drawbacks of MPLSoATM method, ATM devices are very expensive. Traditional IP networks have many limitations such as routing tables which can be complex and time consuming. Thus, it offers little predictability of service, which is unacceptable for triple play services. MPLS technology has been proposed to overcome these limitations, to speed up the traffic flow and also can provide a QOS for real time applications by using labels. From practical experiment on MP Intranet by using L3 MPLS-VPN to transfer data between two remote sites it can be clear from the results that, L3 MPLS-VPN give the best results compared to other methods and overcome the drawbacks of MPLSoATM and IPoATM methods (the service provider routers participate in customer routing, does not require the creation of virtual circuits and create different VPN for every customer for secure data transmission). Figure (16) and (17) show the comparison between IPoFR method and

IPoMPLS method with using L3 MPLS-VPN for data transmission between S1 and S2. The evaluation come in favor of using L3 MPLS-VPN as a wide area network (WAN) to connect remote sites where average end to end delay is better than that of IPoFR by 41.5 % and average end to end delay variation is better than that of IPoFR by 39.28 % at 1400 byte load value.

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