

Master Thesis
Electrical Engineering
Thesis no: MEE 10:51
June 2010



OPNET simulation of voice over MPLS

With Considering Traffic Engineering

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This thesis is submitted to the School of Engineering at Blekinge Institute of Technology in partial fulfillment of the requirements for the degree of Master of Science in Electrical Engineering. The thesis is equivalent to 20 weeks of full time studies.

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ABSTRACT

Multiprotocol Label Switching (MPLS) is an emerging technology which ensures the reliable delivery of the Internet services with high transmission speed and lower delays. The key feature of MPLS is its Traffic Engineering (TE), which is used for effectively managing the networks for efficient utilization of network resources. Due to lower network delay, efficient forwarding mechanism, scalability and predictable performance of the services provided by MPLS technology makes it more suitable for implementing real-time applications such as voice and video. In this thesis performance of Voice over Internet Protocol (VoIP) application is compared between MPLS network and conventional Internet Protocol (IP) network. OPNET modeler 14.5 is used to simulate the both networks and the comparison is made based on some performance metrics such as voice jitter, voice packet end-to-end delay, voice delay variation, voice packet sent and received. The simulation results are analyzed and it shows that MPLS based solution provides better performance in implementing the VoIP application.

In this thesis, by using voice packet end-to-end delay performance metric an approach is made to estimate the minimum number of VoIP calls that can be maintained, in MPLS and conventional IP networks with acceptable quality. This approach can help the network operators or designers to determine the number of VoIP calls that can be maintained for a given network by imitating the real network on the OPNET simulator.

Keywords: Multiprotocol Label Switching (MPLS), Traffic Engineering (TE), Voice over Internet Protocol (VoIP) and Optimized Network Engineering Tool (OPNET)

ACKNOWLEDGEMENT

We are grateful to our parents, whose blessings have been enormous and their support and encouragement gave us ability and strength to complete this task.

We would like to dedicate our thesis work to our family members.

It is a great honor and privilege to thank our honorable thesis supervisor Mr. Yong Yao at, School of Computing, Blekinge Institute of Technology, who had guided our thesis work with his scholarly advice. Without his guidance and encouragement this task would have been unachievable. We express our heartfelt gratitude to him.

We would like to thank our friends who supported us during our master's thesis.

We would also like to thank Patrik Arlos, Mikeal Åsaman, Anders Nelsson and BTH staff.

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ACRONYMS

BGP	Border Gateway Protocol
CR-LDP	Constraint Based Label Distribution Protocol
CR-LSP	Constraint Based Label Switch Path
FEC	Forward Equivalence Class
IETF	Internet Engineering Task Force
IS-IS	Intermediate system to Intermediate system
IPv4	Internet Protocol version 4
LDP	Label Distribution Protocol
LER	Label Edge Router
LIB	Label Information Base
LSR	Label Switching Router
LSP	Label Switch Path
MPLS	Multiprotocol Label Switching
OSPF	Open Shortest Path First
QoS	Quality of Service
RSVP	Resource Reservation Protocol
RTP	Real Time Protocol
RTTP	Real Time Transport Protocol
TCP/IP	Transmission Control Protocol/ Internet Protocol
TE	Traffic Engineering
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network

1 INTRODUCTION

Now-a-days Internet is playing a vital role in most of the people's life due to wide variety of applications and services provided on Internet. The increased number of Internet users made the popular services television and telephone to use the Internet as a medium to reach their customers. These services are provided by convergence of voice and data communications over single network infrastructure. Providing the real-time applications on Internet is a challenging task for the conventional IP networks as it uses best-effort services which doesn't provide guarantee of services and Traffic Engineering (TE). Moreover IP networks offer minimum predictability of services which is unacceptable for the applications like telephony and multimedia services [1].

Multi-Protocol Label Switching (MPLS) is an emerging technology which plays an important role in the next generation networks by providing Quality of Service (QoS) and TE. MPLS networks provide high performance packet control and forwarding mechanism, which forwards the packets based on the labels [2]. It overcomes the limitations like excessive delays and high packet loss of IP networks by providing scalability and congestion control. Due to the low latency and low packet loss during routing of packets MPLS is considered ideal for VoIP applications.

1.1 Aims and Objectives

The main aim of this thesis work is to investigate the performance of real-time voice traffic in IP networks and MPLS networks by including Traffic Engineering in MPLS.

- Designing two network models for IP and MPLS considering same network topology and analyzing the results by considering the same performance parameters.
- Selecting the performance parameters such as end-to-end delay, throughput, voice jitter, packets sent and packets received
- Studying the performance of real-time voice communications in IP network model and MPLS network model with TE.
- All the results are analyzed and shown graphically.

1.2 Scope of the Thesis

This thesis covers the technical issues and factors that need to be considered for the implementation of VoIP in the computer networks. It discusses the challenging issues that need to be faced by computer networks to transmit the VoIP applications. It gives the description about the routing mechanisms, functionalities and design parameters of the MPLS and IP networks. Moreover, a short description of MPLS signaling protocols such as Constraint-based routing-Label Distribution Protocol (CR-LDP) and Resource Reservation Protocol (RSVP) are discussed which are used for implementing TE in MPLS networks.

Deep architecture details for MPLS and IP are not discussed in this thesis. It covers basic background details of MPLS and IP that were considered important and relevant to our thesis. The in-depth algorithms and details of encoder schemes used for VoIP application in both network models are not considered in thesis.

1.3 Research Questions

The following research questions are answered during the thesis process.

- What are the problems related implementing the real-time applications in the networks?
- What is the effect on the performance of VoIP when Traffic Engineering is considered?
- What are the limitations of IP networks when used to transfer voice applications?
- What is the performance of voice over MPLS compared with Traditional IP?
- When considering similar network topology and designing parameters for the MPLS and IP network, which technology provides more number of VoIP calls in the network?

1.4 Expected Outcomes

The thesis work mainly describes the limitations of IP networks in transferring the voice packets compared to MPLS networks. The possible outcomes are the simulation results obtained from OPNET which shows the performance parameters such as throughput, jitter, end-to-end delay, packets send and packets received for the both MPLS and non-MPLS networks. The simulation results also provide graphical comparison of the networks.

1.5 Research Methodology

For the thesis work both qualitative and quantitative approaches are used. First a detailed review of literature is done from studying the case studies and ethnographies. Then we focus the state of problem and how it can be solved. A simulation study is performed by using OPNET modeler to get statistical results or data. By analyzing the results we provide solution for the problem statement of the thesis. For this a network model is designed and results are collected to compare the performance of voice over MPLS with IP network.

1.6 Related Works

In [1], the authors make a comparative analysis of MPLS over non-MPLS networks. The MPLS signaling protocols (CR-LDP, RSVP and RSVP-TE) are studied with considering Traffic Engineering. Their functionality and classification are discussed.

Simulations of MPLS and non-MPLS networks are presented. The performance comparison is made with respect to the parameters such as packet loss, throughput and end-to-end delay on the network traffic. Simulation is done by using QualNet 4.0 simulator.

In [2], the paper mainly focuses on the analytical models to measure efficiency of voice applications when transferred over IP network and on MPLS networks. In this paper, network models are presented which satisfy quality of service (QoS) requirements and traffic engineering standards. The author uses mathematical expressions for evaluating the models for both IP and MPLS networks.

In [13], the main objective of the paper is to calculate minimum number of VoIP calls that can be established in an enterprise IP network. The designing of the real-world network model on the OPNET simulator is presented. The paper discusses the engineering factors need to be considered when implementing the VoIP application in the IP network. A network model is built in the simulator. The model is simulated for calculating the number of VoIP calls that it can maintain. The design parameters such as configuring of VoIP, selection of encoder schemes and the assumptions need to be considered for implementing VoIP applications in OPNET simulator are considered from this paper and are adopted in our simulation model.

In [3], the paper mainly focuses comparison of MPLS signaling protocols CR-LDP and RSVP. A network model is built on the NS-2 simulator for simulation purpose. The paper concludes that CR-LDP gives better operation than RSVP, since RSVP has drawback in its

scalability issues. According to this paper we choose CR-LDP signaling protocol to implement TE in MPLS network in our simulation.

1.7 Comparison of related work to our work

For the next generation networks MPLS is an emerging technology. Most of the research is done in this area to evaluate how well the performance can be improved when MPLS is added on traditional IP networks. Most previous work focus on by performance comparison of network traffic (without considering video and VoIP traffic) between both MPLS and non-MPLS networks by simulation. The related work for comparing the real-time voice communications between MPLS and non-MPLS had done either through analytical models or from theoretical analysis [2], [5], [7], and [12]. Our contribution is that we design a network model using OPNET modeler for comparing VoIP traffic on both MPLS and non-MPLS networks.

In this thesis work we mainly focus on comparing performance of the real-time voice communication in both MPLS and conventional IP networks. Further we investigate the possible affect on performance of voice communication when TE is considered. Simulation is done on IP network and MPLS network using OPNET simulator. The performance of VoIP application is compared by considering voice jitter, end-to-end delay, throughput, packets send and packets received as performance parameters. Simulation results are analyzed and shown in graphical manner. It is to be realized that MPLS is not a replacement of IP but it is designed to add a set of rules to IP so that the traffic can be classified and policed.

1.8 Thesis Outline

In this section we are going to describe outline of the thesis.

- Chapter 2 provides a detailed description of VoIP with its characteristics, as well as how the VoIP packets are forwarded in IP Network.
- Chapter 3 presents a detailed description of MPLS network, MPLS functionality and how packets are forwarded in MPLS network is described.
- Chapter 4 presents a brief description of simulators. The reason for us to choose OPNET is presented. Design of IP and MPLS networks model is presented.
- Chapter 5 exhibits the simulation result and corresponding analysis.
- Chapters 6 present conclusions and future work.

2 BACK GROUND

2.1 Voice over Internet Protocol (VoIP)

Real-time applications such as voice and video over Internet made it the most desirable and cost-effective service to everyone. The VoIP is also known as Internet Telephony, where the communication data is transported by using Real Time Protocol (RTP). RTP consists of data and a control part. The control part is known as Real Time Control Protocol (RTCP) [12]. VoIP is transmitted by using the combination of RTP/UDP/IP protocols. Although TCP/IP is a reliable communication protocol suite, it is not used in real-time communications due to that its acknowledgement/retransmission feature may lead to excessive delays [14]. Further, voice communications are less tolerant to delays and TCP/IP is not suitable. Therefore RTP adopts UDP to provide end-to-end transmission of real-time data where RTCP is used for monitoring the link.

2.1.1 Traffic Characteristics and Requirements of VoIP

There are many factors that affect the quality of voice e.g., the choice of codec, packet loss, jitter, and delay [14], 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 [13]. This packet delay can be divided into three contributing components they are described as follows:

- Encoding, compression and packetization delay occurs at the sender. In the G.711 codec the delay introduced for encoding and packetization are 1 ms and 20 ms respectively. The delay at the sender considering above two delays along with compression is approximated to a fixed delay of 25ms.
- Buffering, decompression, depacketization and playback delay result in the delay at receiver. The total delay due to these factors is approximated to a fixed delay of 45ms.
- The delay considering from sender and receiver we can calculate the network delay which should not exceed 80ms or $(150-25-45)$. This delay actually is the sum of the delays given from propagation, transmission and queuing delay in the network.

It means that the network delay from the source to receiver shouldn't exceed 80 ms in order to provide acceptable quality for established VoIP call. The required bandwidth for a VoIP call is 64 kbps.

Packet loss in the network is another characteristic that is undesirable for a VoIP application. It leads to the loss of sensible information between the conversations of the VoIP call which is disagreeable by the users and the service providers. Packet loss causes the voice breaks and voice skips. During the voice transmission codec can correct lost voice packets. In order codec algorithm to be effective only a single packet need to be lost during the short period of conversation. The voice jitter (delay variation) can be reduced by using the buffer at the receiving device.

In the next sub-section we present IP Networks and how the packets are routed.

2.2 IP Networks

Internet Protocol (IP) allows a global network among an endless mixture of systems and transmission media [1]. The main function of IP is to send the data from the source to destination. Data is constructed as a series of packets. All the packets are routed through a chain of routers and multiple networks to reach the destination. In the Internet, router takes independent decision on each incoming packet. When a packet reaches a router, it forwards the packet to the next hop depending on the destination address present in the packet header. The process of forwarding the packets by the routers is done until the packet reaches the destination.

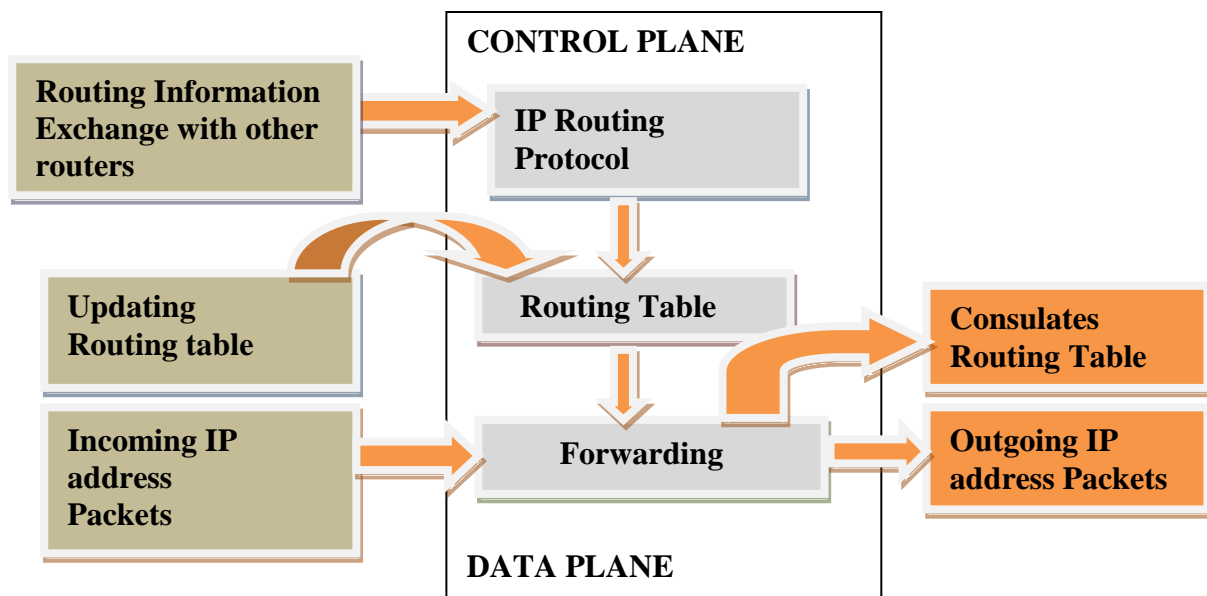


Figure 1 Traditional Router

In conventional routing, to build routing tables each router runs IP routing protocols like Border Gateway Protocol (BGP), Open Shortest Path First (OSPF) or Intermediate System-to-

Intermediate System (IS-IS)[2]. These protocols enable the routers to build the forwarding table shown in Fig.1.

For forwarding the packets and controlling the routing tables, data plane and control plane are the main components. The data plane is a forwarding component which is responsible for forwarding packets from input interface to output interface on router. In the data plane forwarding decisions are made by consulting the routing table.

The control plane is the controlling component which is responsible for construction and maintenance of routing table. The control plane uses the information from the routing protocols such as Open Shortest Path First (OSPF), Intermediate System to Intermediate System (IS-IS) and Border Gateway Protocols (BGP) in building and updating the routing table. These two planes are integrated in the traditional routers.

2.3 Implementation of VoIP Application in IP Networks

It is very challenging to implement the real-time application like VoIP in the conventional IP network. IP mostly work on the best-effort service which doesn't guarantee the delivery of the services. The following factors describe the limitations of IP networks to implement VoIP applications.

- Routing in IP is designed to calculate the shortest path towards the destination but not the best path (mentioned in section 3.4.2).
- In IP networks routing is done in the Network layer which is slower than the switching.
- Most links in IP networks are either under-utilized or over-utilized caused by its routing process, which results in congestion for over-utilized links.
- IP networks are not scalable and TE is difficult to implement (explained in the section 3.4.2) in the IP networks.

The VoIP application requires the guarantee of services with predictable minimum delay and low packet loss. This can be achieved by implementing based on the MPLS networks. In MPLS networks, the packets are transmitted on the Label Switched Path (LSPs). LSP are set based on constraints like the bandwidth availability, administration policies etc. These paths are the virtual connections which are used to transmit the packets reliably; this is desirable for transmitting the VoIP traffic.

MPLS architecture, functionality, routing and MPLS TE are presented in the chapter 3.

3 MPLS NETWORK

Multiprotocol Label Switching (MPLS) provides high performance packet control and forwarding mechanism for routing the packets in the data networks [2]. It has evolved into an important technology for efficiently operating and managing IP networks because of its superior capabilities in providing traffic engineering (TE) and virtual private network (VPN) services [9]. It is not a replacement for the IP but an extension for IP architecture with including new functionalities and applications. The main functionality is to attach a short fixed-label to the packets that enter into MPLS domain. Label is placed between Layer2 (Data Link Layer) and Layer3 (Network Layer) of the packet to form Layer 2.5 label switched network on layer 2 switching functionality without layer 3 IP routing [9] . Packets in the MPLS network are forwarded based on these labels.

3.1 MPLS Header

MPLS operates by defining a label inside MPLS “Shim header” that is placed on the packet between layer-2 and layer-3 headers. The 32-bit MPLS header is organized as in Fig.2.



Figure 2 MPLS Header

The header consists of 20-bit label which is used to identify the Label Switched Path (LSP) to which the packet belongs in the MPLS domain. The labels on the packets are established by using Forwarding Equivalency Class (FEC). Following the label field there are 3 bits EXP field which is called as Traffic Class field (TC field) this is used for Quality of Service (QoS) related functions. Next field is called stack field which is 1 bit field and this is used to indicate bottom of label stack. TTL (Time to Live) field is of 8-bit which has similar function that of TTL field in IP header.

3.2 MPLS Architecture

The MPLS domain is described as “a contiguous set of nodes which operate MPLS routing and forwarding”. It is divided into MPLS core which consists of Label Switch Routers (LSRs) and MPLS edge which consists of Label Edge Routers (LERs).

The main terminologies of MPLS technology are described as follows:

- **Label Switch Router (LSR)** - Any router which is located in the MPLS domain and forwards the packets based on label switching is called LSR. When an LSR receives a packet, it checks the look-up table and determines the next hop. Before forwarding the packet to next hop, it removes the old label from the header and attaches new label.
- **Label Edge Router (LER)** - A packet enters into MPLS domain through LER which is called Ingress router. Packet leaves the MPLS domain through LER which is called Egress router. LER has an ability to handle L3 lookups and is responsible for adding or removing the labels from the packets when they enter or leave the MPLS domain.
- **Label Distribution Protocol (LDP)** - It is a protocol in which the label mapping information is exchanged between LSRs. It is responsible in establishing and maintaining labels.
- **Forward Equivalence Class (FEC)** - It is considered as the set of packets which have related characteristics and are forwarded with the same priority in the same path. This set of packets is bounded to the same MPLS label. Each packet in MPLS network is assigned with FEC only once at the Ingress router.
- **Label Switched path (LSP)** - LSP is the path taken by all the packets in the MPLS domain. Multiple LSPs are present in MPLS domain, which originates at ingress router and traverses one or more core LSRs and terminates at Egress router. These paths are established by the signaling protocol (LDP) in MPLS domain.

3.3 MPLS Functionality

The Fig.3 shows MPLS router which operates on two functional blocks

- The separation of control plane and forwarding plane (data plane) components
- Label-swapping forwarding algorithm

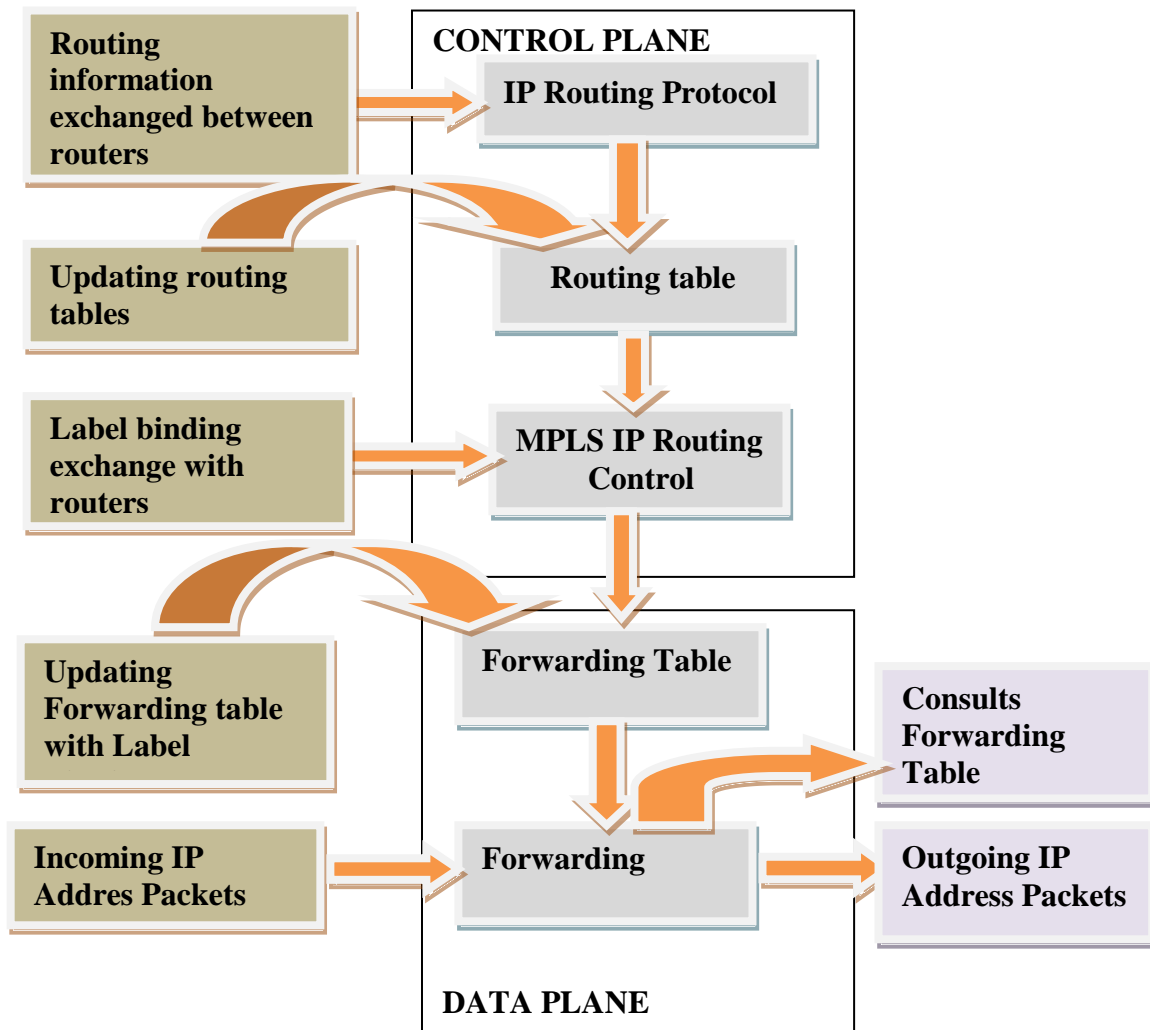


Figure 3 Label Switch Router

The control plane (control component) maintains and controls the forwarding table by learning the network topology from the routing protocols such as OSPF, IS-IS and BGP. In MPLS each router maintains a label information table that is used for updating the forwarding table, based on this forwarding table the forwarding decision is done. Control plane is responsible for building the MPLS IP routing control by updating the label bindings which are exchanged between the routers. Therefore when a packet arrives at the router, the forwarding decision is taken by the data plane (forwarding component) by consulting the forwarding table, which is maintained by control plane. The packets are then forwarded towards the appropriate node according to the forwarding decision.

In MPLS routers control plane and data plane are separated entities. This separation allows the deployment of a single algorithm that is used for multiple services and traffic types [9].

The label-swapping forwarding algorithm explains how the packets are routed in the MPLS domain. It is described in the following steps.

- When a packet enters the MPLS domain, a label of short fixed-length is inserted in the packet header by the ingress router. FEC is identified from the label.
- The packets belonging to one particular FEC are forwarded through the same path through the MPLS network. Even though all these packets do not have the same destination address.
- The path on which the packets are forwarded to the next hop in the network is LSP.
- Every hop in MPLS network forwards the packets based on the label but not on IP address. This is done until the packets reach the final hop in MPLS network and then the label is removed by egress router and normal IP forwarding continues.
- The Ingress and Egress routers are located in MPLS edge domain. The routers within the MPLS core domain are LSR's which is shown in Fig.4.

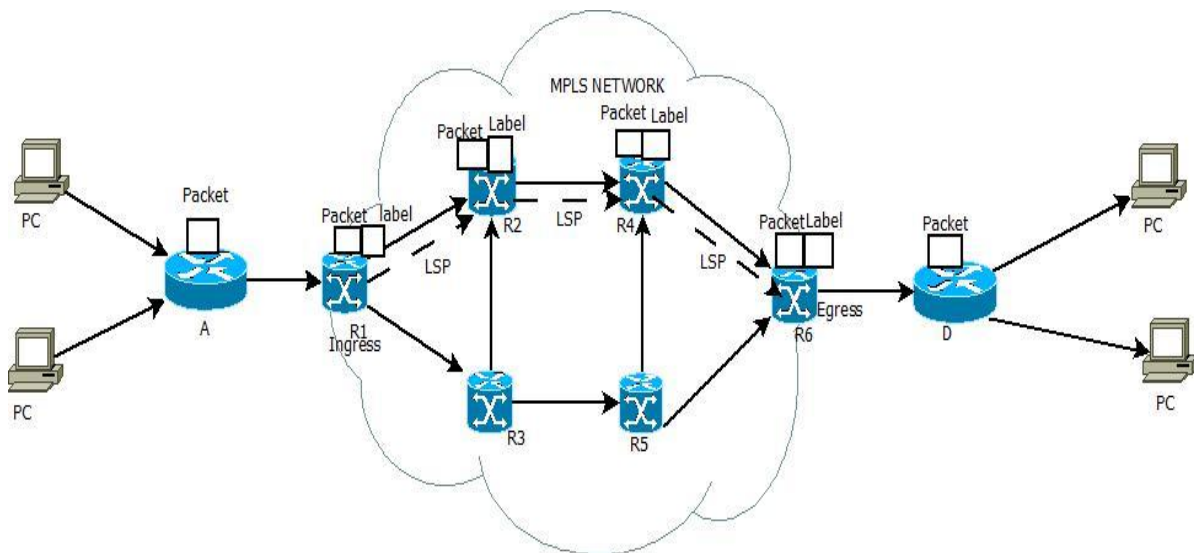


Figure 4 MPLS Network

The following is the brief description of MPLS routing:

MPLS uses signaling protocols to establish the paths. Label Distribution Protocol is the signaling protocol and the established paths are called Label Switched path. Routers that support MPLS are Label Switched Routers (LSRs). The LSRs which are located at the edges of MPLS are called Label Edge Routers (LERs). All the packets enter or exit the MPLS domain through LERs. In Fig.4 'R1' and 'R6' are the LER's. 'R1' is the ingress router which maps the incoming traffic into the MPLS domain. 'R6' is the egress router through which the packets exit from the MPLS domain. An LSP originates at ingress router and travels through one or more LSRs and terminates at egress router.

When packet enters the MPLS domain, labels are inserted in their headers by Ingress router and the packets are mapped on to the LSP using Forwarding Equivalence Class (FEC). All the packets which match a particular FEC are forwarded on the same LSP. The FEC is described by the set of attributes e.g. destination IP, type of service, etc. The core LSRs (which are 'R2', 'R3', 'R4', 'R5' in the Fig.4) forward the packets based on label information not on the IP address. When a router receives the packet, it checks label information base (LIB) instead of routing table and determines the next hop in MPLS domain.

Finally the egress router 'R6' removes the label from the packet header and forwards the packet to the next hop based on IP address. And then the conventional IP forwarding of packets continues. The MPLS signaling protocols used for traffic Engineering is presented in section 3.5.

3.4 Traffic Engineering

Traffic Engineering (TE) is a mechanism that controls the traffic flows in the networks and provides the performance optimization by optimally utilizing the network resources [6]. Several key features of TE are resource reservation, fault-tolerance and optimum resource utilization [3].

3.4.1 Important Factors Needed for TE

- Distribution of topology information
- Path selection
- Directing traffic along the computed paths

- Traffic management

Distribution of topology information: It requires a mechanism to advertise the current information about the links for the nodes, so that the nodes can build a map about the network topology. It is crucial that the information about the link or node failures have to be rapidly propagated through the networks. This makes the problem to be fixed quickly [4].

Path selection: Path information between the nodes is computed in the network while path selection. The shortest path with minimum links is selected. The other constraints like bandwidth and delay are also considered when selecting the path.

Directing traffic along the computed paths: Traffic is forwarded along the particular calculated path between source and destination node. Typically this is achieved by forwarding table.

Traffic management: Traffic management deals with the process of forwarding the traffic with the predictable quality. The parameters, e.g., bandwidth, delay, jitter and packet loss are the main concern for the traffic management.

3.4.2 Challenges in IP Network for TE

It is very challenging for providing TE in conventional IP networks. The packets in the IP networks are forwarded based on Open Shortest Path First (OSPF) protocol which chooses the shortest path from source to destination. These shortest paths [6] can save network resources but may result in the following problems.

- The shortest paths from various sources overlap at some links where network congestion may occur.
- Longest path between the nodes stays underutilized, though the capacity of traffic exceeds the capacity of shortest path between the nodes.
- Equal-cost multipath of the links and load sharing makes difficult to implement the TE in IP networks.
- Equal-cost multipath of links happens when a source needs to transfer the traffic to destination through the paths with equal-costs. Since IP forwards the packets based on destination address, it uses a link among the equal-cost multipath links irrespective of utilization factor of the links. So that traffic may not be forwarded on the link which is less busy or which is forwarding less traffic compared to other paths.
- Load sharing is the other factor which is difficult to implement in IP networks among multiple paths with different costs.

In order to implement TE effectively, Internet Engineering Task Force (IETF) has introduced MPLS technology.

3.4.3 Traffic Engineering in MPLS Network

The main objective of considering TE is to efficiently use the available network resources and increase service quality of applications on the Internet. The motivation behind MPLS TE is Constraint Based Routing (CBR). It considers bandwidth, policies and network topology for establishing a path (path refers to LSPs) in MPLS domain to forward the packets. In contrast, IP routing uses OSPF that calculates the shortest path between the nodes, it do not consider if that path has enough resources.

Constraint Based Routing

Constraint Based Routing (CBR) is also known as Constrained shortest Path First (CSPF) is an extension of shortest path algorithms. CBR computes the expected path in MPLS, based on the constraints e.g., minimum amount of required bandwidth in a link, end-to-end delay and administrative policy. The corresponding selection procedure is to involve removing the path which have insufficient bandwidth or not satisfy the required constraints. Only the paths which satisfy the required criteria given by the administrative policy are selected and a shortest path algorithm is run on these paths to find a shortest path from Ingress to Egress router in MPLS network. The path given by CBR routing may be a longer but lightly loaded path which is better than heavily loaded shortest path [4]. CBR is extensively used in the MPLS TE for distributing the traffic evenly and increasing the network performance.

For establishing the LSPs from ingress to egress routers and implementing TE in MPLS network, signaling protocols are used. They are developed by IETF.

Before getting into details of the RSVP and CR-LDP signaling protocols, we firstly present the requirements of signaling protocols in MPLS TE.

3.4.4 Requirements of Signaling Protocols used in MPLS TE

It is essential that signaling protocols used in MPLS TE need to contain the following characteristics:

- **Robustness:** Even in the presence of network congestion or failure the delivery of signaling messages need to be reliable in the network.
- **Scalability:** If the size of MPLS network is large, each node needs to support a large number of LSPs. It is required that the signaling system is scalable in order to deliver the required performance, even though there are large number of LSPs and nodes.

- **LSP establishment/teardown/maintenance:** The signaling protocol has to provide LSP establishment, teardown and maintenance in the MPLS network.
- **LSP priority/preemption:** In MPLS network it is possible that high priority LSPs may teardown lower priority LSPs, when there are not enough resources available for both.
- **Alternative path setup and rerouting capability:** To deliver the dependable services the path optimization, resilience and failure recovery are required in the networks. The failure situations have to be identified and recovered quickly with the minimum control.

3.5 Signaling Protocols in MPLS TE

In the MPLS network the LSPs are established and the labels are distributed on each hop along the LSPs before the packets are forwarded. There are two methods to establish LSPs in the MPLS network:

- Control driven LSP.
- Explicitly routed LSP.

Control driven LSP is also called as hop-by-hop LSP which is set using LDP protocol. The control driven LSPs is set by the process of each LSR determining the next hop for the LSP based on its IP forwarding table. It sends the label request to the next hop to establish LSP. This process is continued till the LSP reaches the edge router (i.e., egress router) in MPLS domain.

Explicitly routed LSPs is also called as constraint based routed LSPs (CR-LSPs). CR-LSPs are set by specifying the route for LSP in the setup message. This setup message travels all the hops along the specified path. At each hop, a label request is sent to the next indicated hop along the LSP [11]. The difference between the control driven LSPs and CR-LSP is that a packet in control driven LSPs may follow the path where default IP routing might have been used. Whereas the CR-LSPs are set by network administrator or a network management application on which the traffic is sent which is independent of the LSP computed by IP forwarding. In this way CR-LSPs are used for TE in MPLS. Two protocols are used to set CR-LSPs in MPLS that are:

- Constraint based routed LDP (CR-LDP) and
- Resource Reservation Protocol(RSVP)

3.5.1 Constraint based routed LDP (CR-LDP)

The main idea of CR-LDP is to enhance the LDP protocol to work over on explicit route. CR-LDP is used to establish CR-LSPs, transporting various parameters for resource reservation and providing options for CR-LSP robustness features [1]. CR-LDP and LDP protocols are hard state protocols. It doesn't require periodic refreshing of information to maintain the CR-LSP. The signaling messages are sent only once for establishing the CR-LSPs. In CR-LDP approach, UDP is used for peer discovery and TCP is used for session advertisement, notification and LDP messages.

CR-LSPs in the MPLS network are set by using Label Request message. This message is the signaling message which contains the information of the list of nodes that are along the constraint-based paths. In the process of establishing the CR-LSP the label request message is sent along the constraint-based path towards the destination. If the path meet the requirements given by network operator or network administrator, all the routers present in path distribute the labels by means of Label Mapping message. The distribution of labels is started from the destination and move in reverse direction towards the source, the final label that reaches the source indicates that CR-LSP is established.

For the failure recovery process CR-LDP uses TCP as transport mechanism to ensure that recovery process is started timely based on the policies specified by the network operator [11].

3.5.2 Resource Reservation Protocol (RSVP) approach

RSVP is the soft state protocol. It uses Path and Resv commands to establish path. The CR-LSPs established by RSVP signaling protocol in MPLS network is described by the following steps.

- The ingress router in the MPLS network selects a LSP and sends the Path message to every LSR along that LSP. This Path message indicates that this is the desired LSP used to establish as CR-LSP.
- The LSRs along the selected LSP reserve the resources and that information is send to Ingress router using the Resv message.
- In which the Path and Resv messages are send periodically to refresh the state maintained in all LSRs along the CR-LSP [11].

3.5.3 Comparison of RSVP and CR-LDP

The comparison of RSVP and CR-LDP signaling protocols are as follows.

- RSVP is the soft-state protocol. It requires periodic refreshing of Path and Resv messages to maintain the state in all LSRs along the CR-LSP. The signaling traffic (Path and Resv messages) increases as the number of CR-LSPs increases in the MPLS network. Due to this, RSVP provides poor scalability in the MPLS network.
- In contrast, CR-LDP provides better scalability. It is a hard state protocol and establishes the CR-LSP based on Label request and Label mapping messages. Once the CR-LSP is established it will not be teardown until a specific request is made.
- The other difference is that CR-LDP uses TCP to transport its signaling messages. When a failure occurs the error message is sent using reliable transport mechanism this can ensure the fast failure notification. Whereas the RSVP doesn't guarantee fast failure notification, because RSVP lack in reliable transport infrastructure [1].
- RSVP was created before CR-LDP for different purposes. Due to above reasons mentioned RSVP is not suitable for TE in MPLS networks. Among the two protocols CR-LDP is best suited in MPLS TE.

4 SIMULATION

4.1 Simulation Tools

Simulation is the process of testing a designed model on a platform which imitates the real environment. It provides the opportunity to create, modify and study the behavior of proposed design. It helps the user to predict its strengths and weakness before implementing the model in real environment. Some of the popular simulators used to simulate the data networks are

- OMNet++
- OPNET Modeler
- NS2

OMNet++

OMNet++ is the discrete event environment programmed in C++. This is used to simulate computer networks. To use OMNet is not straightforward, as it requires in learning of number of tutorials, demos and walking through large web based documentation. Although it provides MPLS, LDP and RSVP-TE modules, it provides poor documentation for those modules [15].

NS2

NS-2 is the simulator targeted at network research. The user interfaces with the simulator by using object-oriented script language OTcl on the UNIX systems (although it is possible to install NS-2 in windows). In order to build the simulation topologies on NS-2, OTcl language is required to be familiar with. Moreover in NS-2, MPLS and RSVP-TE modules are not available as standard libraries, as they are implemented from third party. The documentation is not available for all modules and the source code is required to read, has to learn how to interface with it and results are not generated automatically [15].

OPNET Modeler

OPNET provides several modules for the simulation comprising a vast universe of the protocols and network elements [15]. It has gained popularity in academia as it is offered for free of cost to institutions. The users don't need to have deep programming knowledge to use OPNET. The user can directly concentrate in building and analyzing model for simulation.

The main feature of OPNET is that it provides various real-life network configuration capabilities that make the simulation environment close to reality [13]. The advantages of OPNET compared to other simulators are GUI interface, comprehensive library of network protocols and models, graphical interface for results viewing, availability of documentation for the user to develop the network models etc.

4.2 OPNET SIMULATION

Simulation is divided into two tasks to fulfill the purpose of the thesis.

In this part of the simulation, the VoIP traffic is send from source (VoIP_WEST) to destination (VoIP_EAST) in the two networks (MPLS and Traditional IP networks).The main task is to compare the performance of VoIP traffic in both IP and MPLS networks with respect to metrics, i.e., voice jitter, packet end-to-end delay, packet loss and throughput. The simulation results are analyzed to determine the efficient network used for transmitting VoIP traffic.

In this part, an approach is made to estimate the approximate minimum number of calls that can be maintained in both IP and MPLS networks. This approach is accomplished by designing the real-life network in the OPNET. In the simulation, we use the end-to-end delay metric to estimate the approximate minimum number of calls maintained in both networks.

4.2.1 Assumptions

It is difficult to predict the traffic behavior in the network as the traffic in network instantaneously varies from source to destination. We prefer to simulate the conventional IP and MPLS models by considering the worst case scenario, to estimate the minimum number of VoIP calls that a network can support with acceptable quality. We consider the background traffic excluding the VoIP traffic to be as 50% of link capacity, as explained in [16] 60% link capacity is the max-utilization allowed of a link to protect it from bursts.

4.3 Network Design

The simulation of both IP and MPLS networks are deployed in the OPNET Modeler 14.5. The simulations consist of two scenarios with considering the same network topology.

- Scenario 1 on the basis of MPLS network with TE.
- Scenario 2 on the basis of IP network without TE.

4.3.1 MPLS Simulation model

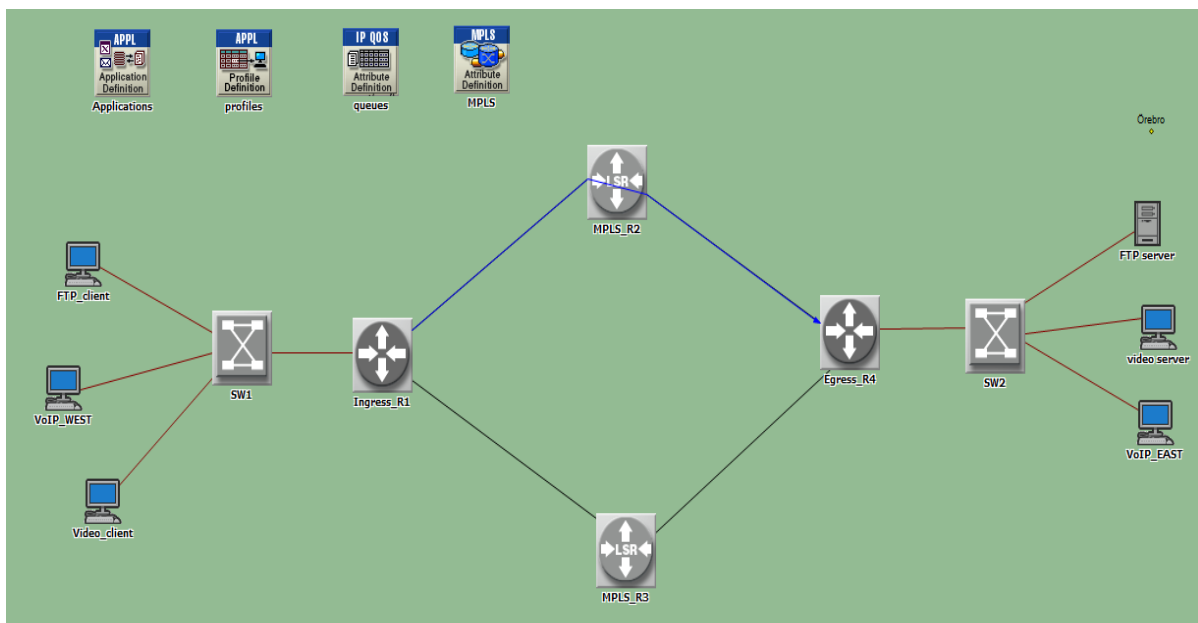


Figure 5 MPLS Simulation Model

Fig.5 shows the MPLS network based scenario which consists of the following network elements

- Two LERs (Ingress_R1 and Egress_R4)
- Two LSRs (MPLS_R2 and MPLS_R3)
- Two VoIP stations (VoIP_West and VoIP_East)
- Two switches (SW1 and SW2)

DS3 links (44.736 Mbps) and 100Mbps links are used for respectively connecting all the routers connecting workstations to the two switches.

TE is implemented in the above simulation model by using CR-LDP signaling protocol, which is configured in OPNET by defining FECs in MPLS definition attribute and also set LDP parameters in the routers. The CR-LSP which is established can be visible in the Fig.5 as

a blue colored link from Ingress_R1 to Egress_R4 through router MPLS_R2. When the network congestion occurs, the traffic directed along CR-LSP path is evenly distributed in the MPLS network. This decreases the affection of the network congestion and increases the efficiency in utilizing the network resources.

In this scenario VoIP traffic is send from VoIP_WEST to VoIP_EAST. The VoIP calls are established by configuring the application and profile definition attributes (explained in the next section). We simulate both networks IP and MPLS in order to obtain packet end-to-end delay, voice jitter, packet sent and packet received values.

In the MPLS network model (shown in Fig.5) there exits two paths which are Ingress_R1<->MPLS_R2<->Egress_R4 and Ingress_R1<->MPLS_R3<->Egress_R4. Both paths are equidistant from source to destination. When the packets are routed, an IP network uses only one of the paths and doesn't consider the other path, as both are shortest paths. Since TE is implemented in the MPLS network, the network load is evenly distributed and makes MPLS an efficient technology. In IP network model (Fig.6) the two paths are defined as IP_R1<-> IP_R2<->IP_R4 and IP_R1<->IP_R3 <-> IP_R4.

4.3.2 Conventional IP Simulation Model

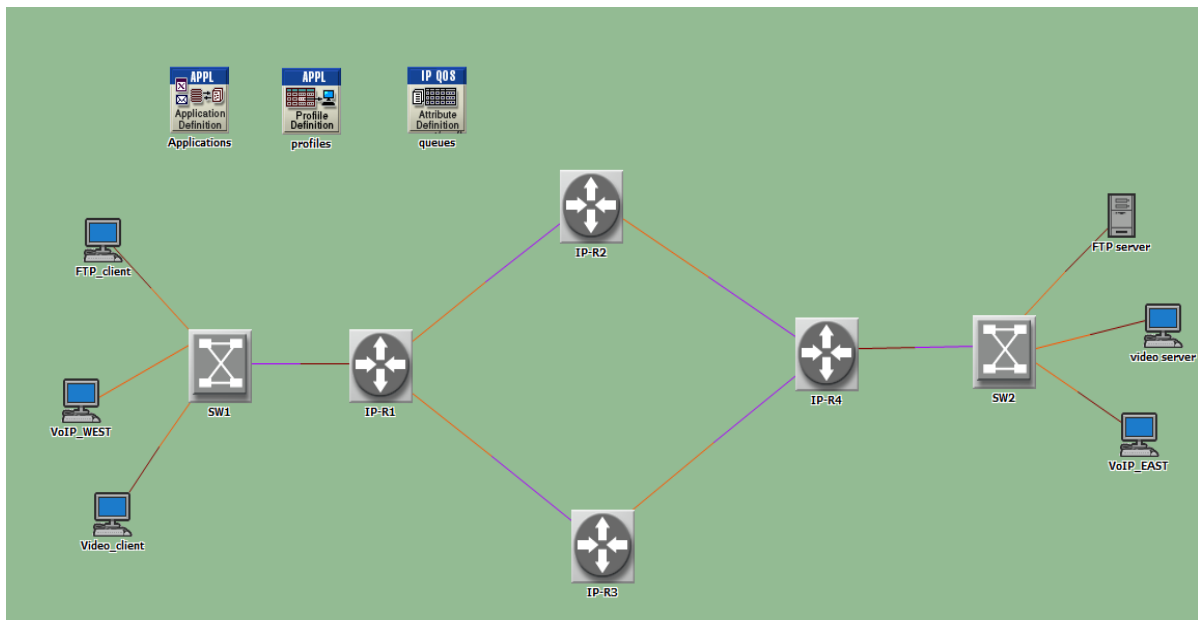


Figure 6 IP Simulation Model

Fig.6 shows the simulation scenario based on the conventional IP network without TE. In this scenario MPLS routers are replaced with normal IP routers. MPLS definition attribute, hence is not considered and the packets are routed using OSPF protocol (which doesn't take capacity constraints). The VoIP traffic is transmitted between the VoIP_West and VoIP_East and the procedure for setting VoIP calls is similar to that of MPLS scenario.

4.4 Modeling of VoIP traffic in OPNET

In order to model an application in OPNET, an object is available which is called *application definition attribute*. This attribute consists of predefined applications which can be modified as per the requirements of the user. Some of the predefined applications in application definition attribute are HTTP, E-mail, Video, FTP, Voice, Database etc.

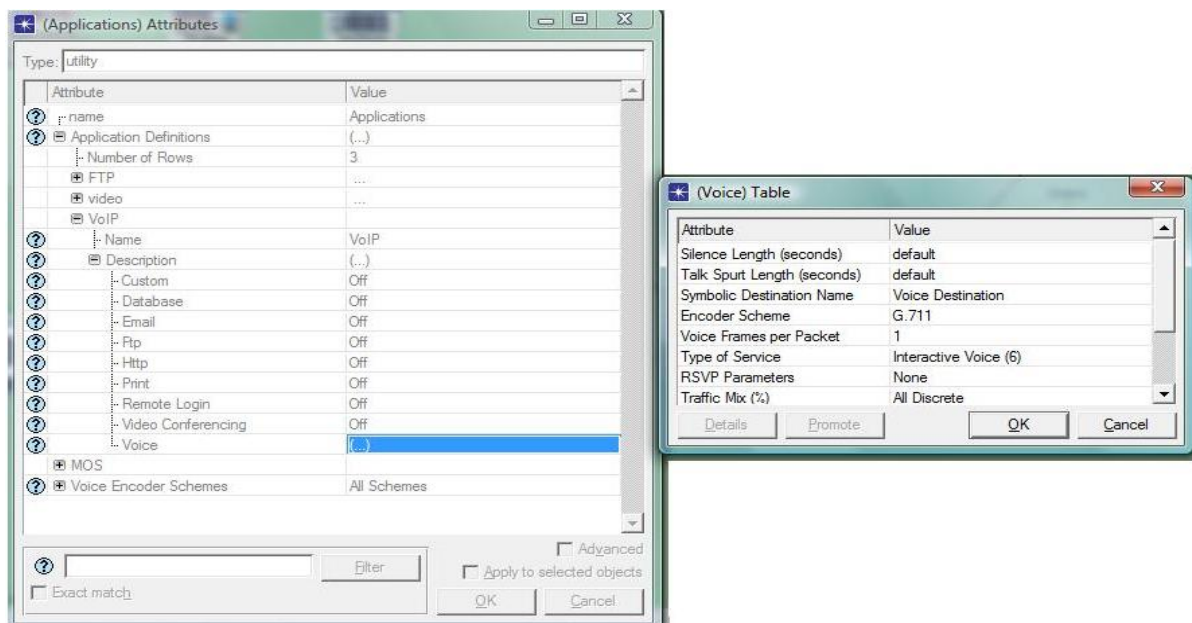


Figure 7 Application Definition

Fig.7 shows the Application Definition attribute used in our simulation model. Three applications (FTP, Video and VoIP) are modeled in our simulation by using the Applications attributes. FTP and Video applications are modeled in order to introduce background traffic in the simulation. The Voice application is modeled by configuring the (Voice) Table (see in Fig.7). The VoIP application uses G.711 encoder scheme and Interactive Voice (6) as the type of service for establishing the VoIP calls.

After configuring the VoIP application in Application Definition there is a necessity to define which work station will be using this VoIP application. In our case VoIP_East and VoIP_West are the workstations (shown in Fig.5 and Fig.6) that will run the VoIP application. The behavior of the work station is described by its *Profile* which is defined by using the Profile Definition. Fig. 8 shows the Profile Definition object used in our simulation. It describes the start time of the simulation, which is set to 100 (offset'60'+start time'40') seconds and the VoIP application is repeated continuously till the end of the simulation. It means that VoIP calls are established between workstations VoIP_West and VoIP_East starting at 100 seconds and the calls are added continuously till the end of simulation.

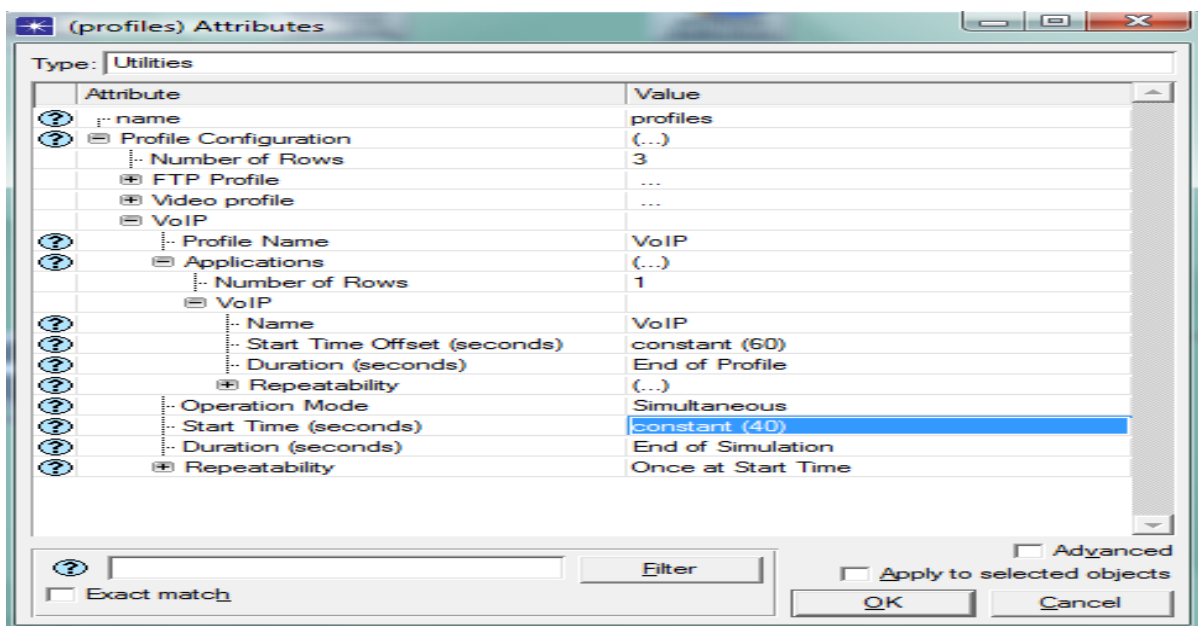


Figure.8.Profile Definition

In OPNET, the calculating of the minimum number of the calls which can be maintained in the given network is referred to the shown in Fig.8 the profile definition. This profile is configured in a way that each VoIP call is added after a fixed interval of time and the process of adding the call is repeated till the end of simulation.

The first VoIP call is established at the 100th second of the simulation, then for every 2 seconds a VoIP call is added to the simulation. The addition of VoIP call is done by repeating the VoIP application for every 2 seconds in the profile definition. This process is repeated till the end of simulation. In this way the VoIP calls are added continuously at fixed interval to

the given network model, this approach makes it possible to determine the number of calls that can be maintained in the given network.

5 RESULTS AND ANALYSIS

5.1 Comparison of Performance Metrics

The results are shown in the Fig.9, Fig.10, Fig.11 and Fig.12. They are respectively associated with metrics exhibition in both MPLS and conventional IP networks based scenarios. We observe that there is an increase in the performance when the VoIP traffic is transmitted using MPLS technology.

For each scenario the duration of the simulation is 420 seconds. The VoIP traffic starts at the 100th second and ends at the 420th second of the simulation time. In both scenarios VoIP calls are added at fixed time intervals i.e., for every two seconds. The addition of VoIP calls are started from 100th second till 420th second of simulation.

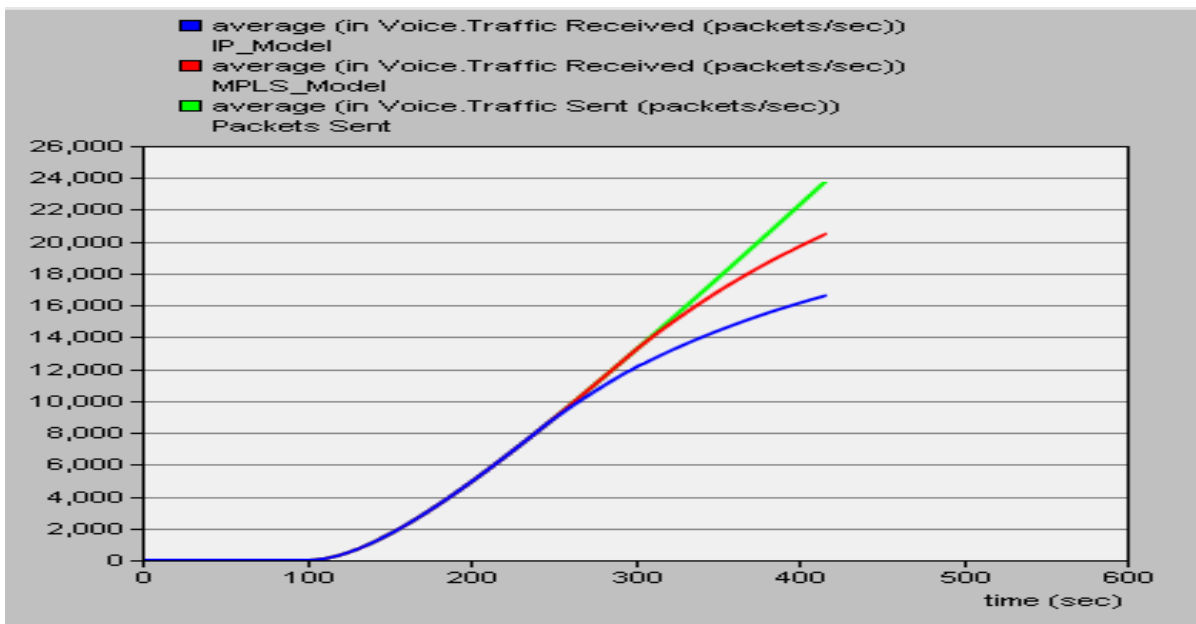


Figure 9 Voice Packet Send and Received

The Fig.9 gives the average number of packets sent and received in both MPLS and conventional IP networks. Simulation result shows that MPLS model gives more throughput than the IP model.

The two scenarios are simulated considering the background traffic (explained in section 4.2.1). Fig.9 shows that voice packets start to drop from 240 second in the IP network, whereas from 300 second in MPLS network.

In the simulation, the early packet drop in IP network indicates that it cannot establish the VoIP calls with acceptable quality after 240 seconds. The VoIP calls established after 240 seconds exhibit packet loss. This cause loss of information and results in voice breaks and voice skips.

The voice packet loss in MPLS network starts at 300 seconds. MPLS delivers the packets with high transmission speed and lower delays. There is TE implemented in the MPLS network which temporarily reduces the congestion. Due to these factors the packet drop in MPLS networks starts at 300 second where as in IP network the packet drop starts at 240

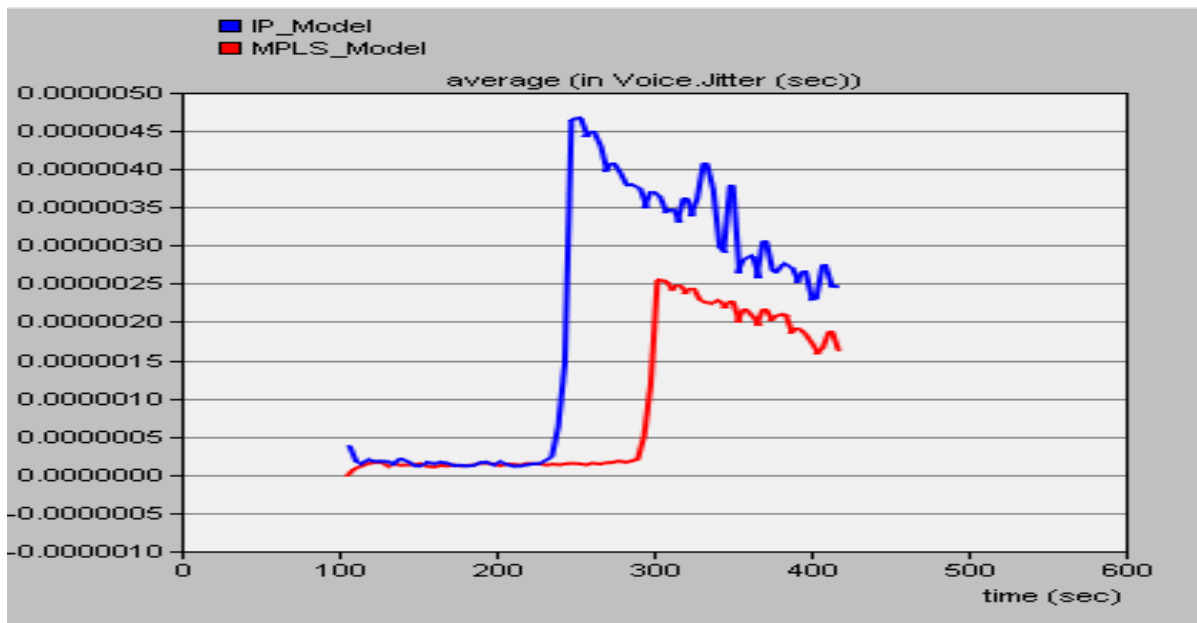


Figure 10 Voice Packet Jitter

seconds, this increases the throughput in the MPLS network.

The Fig .10 shows the voice packet jitter of MPLS and IP network model. It is noticed that voice jitter starts to rise at 240 sec in IP network. In contrast, for MPLS network it at 300 second.

It is observed from the above figure that there is an increase in the jitter for IP and MPLS model around 240 and 300 second respectively. This is due to that, at this time instant there is a voice packet drop which can be noticed from the Fig.9. This variation at these time instants can be seen for the remaining performance metric. The voice packet delay variation shown in Fig.11 has same variations in graphs as explained here.

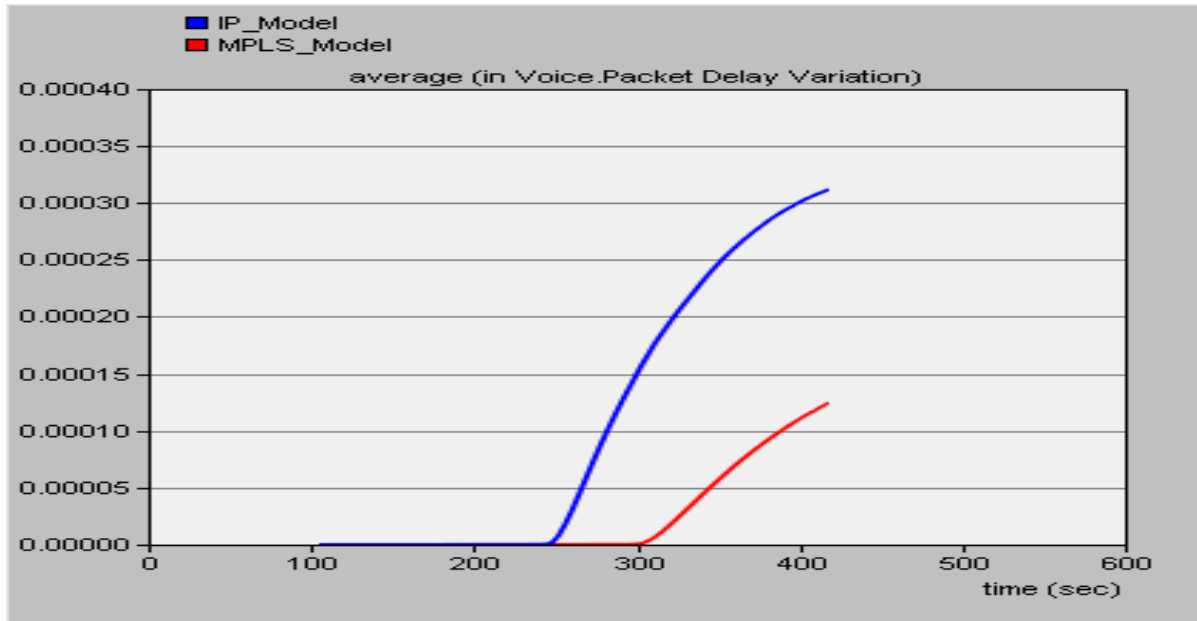


Figure 11 Voice Packet Delay Variation

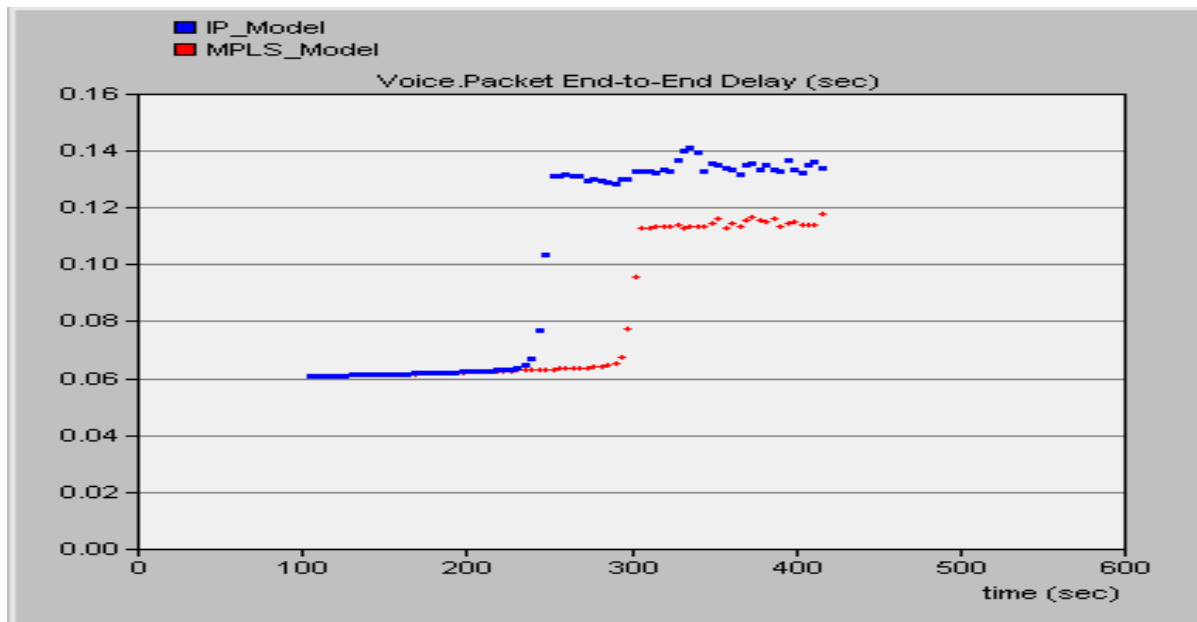


Figure 12 Voice Packet End-to-End Delay

The Fig.12 shows the packet end-to-end delay of MPLS and IP network model. As explained in the section 2.1.1, the end-to-end delay in a network is not advised to increase above the threshold value of 80 milliseconds. So that established VoIP calls are of acceptable quality. From the Fig. 12 it is noticed that end-to-end delay in IP network exceeds the threshold at 240 sec. whereas for MPLS network it reaches at 300 seconds. The IP network reaches the threshold early than MPLS network; this is due to TE is implemented in MPLS network.

MPLS uses CR-LSPs for controlling the temporary congestion. In simulation CR-LSPs is set from Ingress_R1 to Egress_R2 through R3 which is shown by ‘blue’ path in Fig.5. In the MPLS network the network resources are efficiently utilized compared to IP network.

5.2 Calculating the number of VoIP calls

As explained in the section 2.1.1 the delay in the network must not exceed the threshold value of 80ms to maintain the minimum number of VoIP calls with acceptable quality. The numbers of VoIP calls that can be maintained in the MPLS and IP networks are estimated using the end-to-end delay graph. Fig. 13 shows the end-to-end delay statistics of the IP network and Fig.14 shows the end-to-end statistics of MPLS network model.

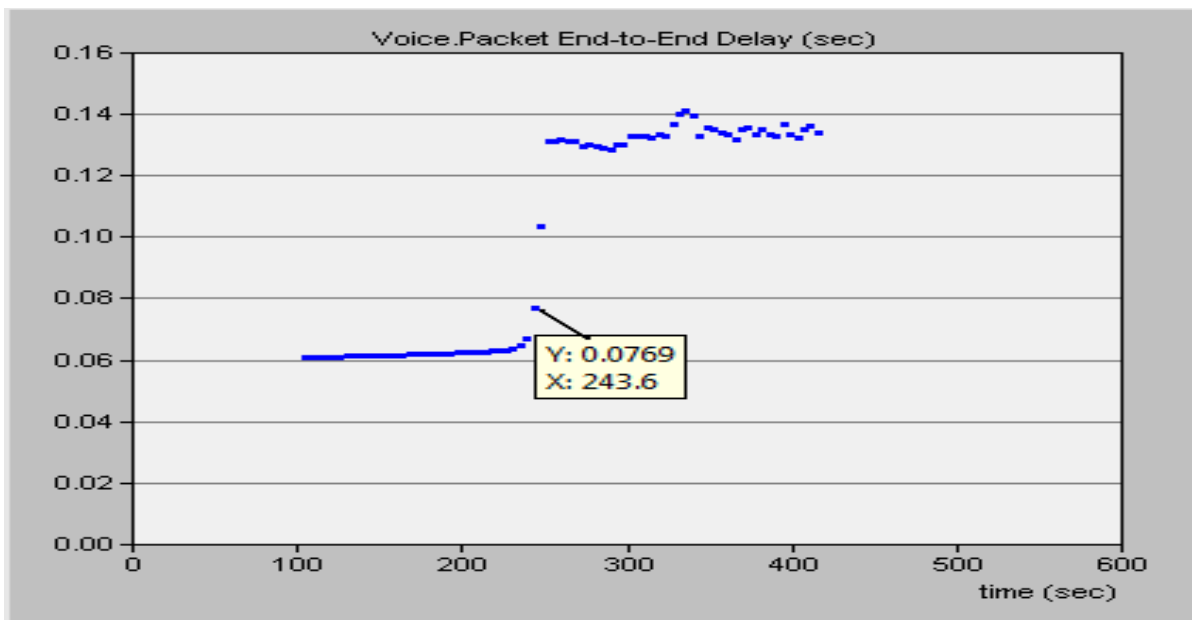


Figure 13 IP Networks End-to-End Delay

From the Fig.13 it is noticed that in conventional IP network the end-to-end delay crosses the threshold value of 80ms at 243 seconds, whereas in MPLS network the end-to-end delay crosses threshold value at 298 seconds shown in Figure.14.

In section 4.4 it is explained that in the OPNET the VoIP calls are added to the network by configuring the Application Definition and Profile Definition. The configuration is done so that each call is added for every 2 seconds and the addition of calls start from 100 second of the simulation to end of simulation.(100 second is considered because in the simulation, the start time of VoIP application from source to destination is at 100 second) .

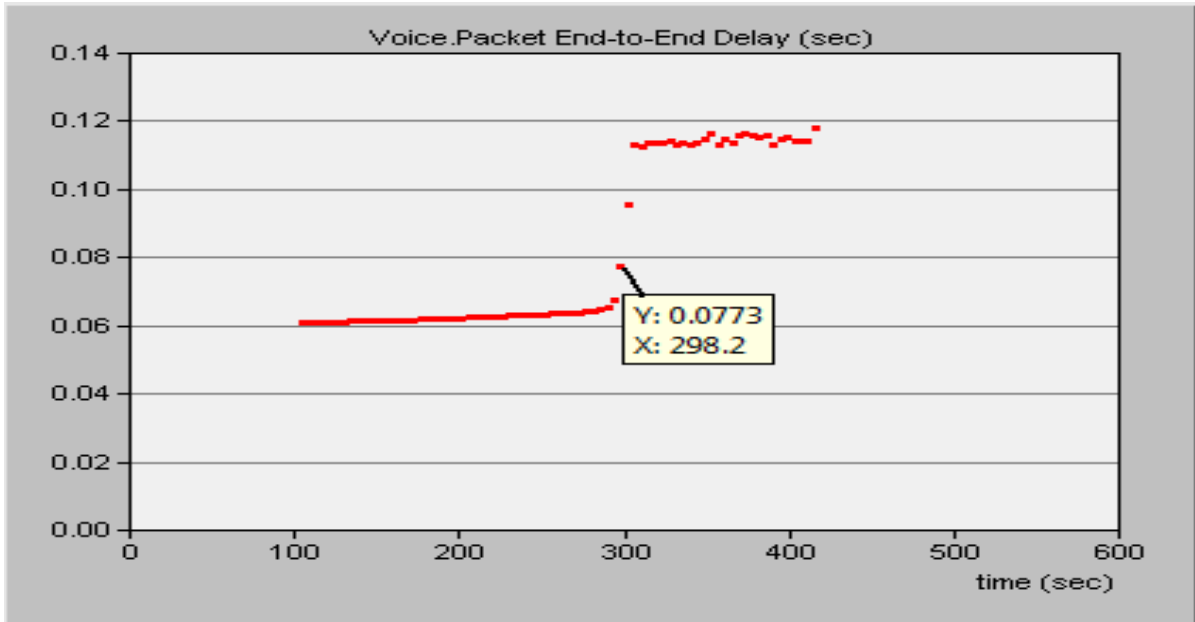


Figure 14 MPLS networks End-to-End Delay

In each scenario the total number of calls established is given by calculating total simulation time i.e., from 100 to 420 seconds. Since for every 2 seconds one call is added to the network so the total number of calls maintained in the network is $(420-100)/2 = 160$ VoIP calls. (These are the number of VoIP calls established in each scenario i.e., for IP and MPLS model)

Number of calls maintained in both networks with acceptable quality:

In IP network around 240th second (see in Fig.13) the threshold is reached i.e., 80 millisecond, the VoIP calls calculated are from 100 to 240 seconds, as we are adding a VoIP call for every 2 seconds.

Total number of VoIP calls maintained with acceptable quality in IP networks is:
 $(240-100)/2 = 70$ VoIP calls with acceptable quality.

In MPLS network the total number of VoIP calls maintained with acceptable quality is:
 $(300-100)/2 = 100$ VoIP calls with acceptable quality.

In MPLS around 300th second (see in Fig.14) the end-to-end delay threshold is reached which is at 80ms.

The calls calculated in the both network models are varied depending on the traffic conditions and the network topology.

6 CONCLUSIONS AND FUTURE WORK

6.1 Conclusion

The main objective of the thesis is based on the performance analysis of conventional IP network and MPLS network in respect of VoIP traffic. The performance analysis is followed by presenting an approach in OPNET to estimate the minimum number of VoIP calls that can be maintained in the MPLS and IP networks. The performance analysis in both networks is made on focusing on the performance metrics such as voice jitter, voice packet delay variation, voice end-to-end delay, Voice packet send and received.

Our research started by literature review made on the state of art on MPLS, TE and IP. The literature review helped us to answer three of our research questions. Based on the simulation results it can be concluded that MPLS provides best solution in implementing the VoIP application (Internet Telephony) compared to conventional IP networks because of the following reasons

- Routers in MPLS takes less processing time in forwarding the packets, this is more suitable for the applications like VoIP which posses less tolerant to the network delays.
- Implementing of MPLS with TE minimizes the congestion in the network. TE in MPLS is implemented by using the signaling protocols such as CR-LDP and RSVP.
- MPLS suffers minimum delay and provides high throughput compared to conventional IP networks.

In the beginning of the thesis we identified four research questions whose solutions are found after the completion of the thesis. The solution for the research questions are listed below.

6.2 Research Questions and Answers

R1) what are the problems related implementing the real-time applications in the networks?

From the literature study and from the simulation results (shown in Fig. 9, 10, 11 and 12) we found that the real-time applications are less tolerant to network delays and packet loss. There will be degradation in the VoIP service if it is implemented in a network which exhibit high delays and packet loss.

R2) what is the effect on the performance VoIP application when Traffic Engineering is considered?

From the literature study we have learned that Traffic Engineering is a key factor in improving the performance of Internet services or applications. From the simulation results we can conclude that TE minimizes the congestion and provides the scalability in the networks. It helps in managing the network effectively by efficiently utilizing the available network resources.

R3) what are the limitations of IP networks when used to transfer voice applications?

By studying the state-of-art of the IP architecture and functionality, we have identified some crucial factors why IP networks are unsuitable for implementing VoIP applications. IP networks offer little predictability of services and exhibits high packet loss and more delays which are unacceptable for the real-time applications.

R4) what is the performance of voice over MPLS compared with Traditional IP?

The simulation results show that the performance of VoIP application is increased when it is implemented in MPLS network with considering TE. OPNET modeler 14.5 is used for simulation. The performance metrics obtained from the simulation shows that due to the added features like TE and MPLS mechanism of forwarding packets make MPLS a better choice in transferring the VoIP communication.

R5) considering similar network topology and designing parameters for the MPLS and IP network, which technology provides more number of VoIP calls in the network?

An approach is made in OPNET to calculate maximum number of calls that can be maintained in MPLS and IP network. It is concluded that using MPLS network more number of VoIP calls can be established when compared to IP networks.

6.3 Future work

This thesis work mainly focused on the performance comparison of VoIP traffic between IP and MPLS network including TE. The future work of the thesis can be carried to study the performance of MPLS Traffic Engineering signaling protocols CR-LDP and RSVP when VoIP application is implemented in them. It would be interesting if one take different codec's into consideration while establishing a VoIP application. The work can be further extended to study the performance of video applications on CRLDP and RSVP signaling protocol.

7 BIBLIOGRAPHY

- [1] Mahesh Kr. Porwal., Anjulata Yadav., S. V. Charhate, “Traffic Analysis of MPLS and Non MPLS Network including MPLS Signaling Protocols and Traffic distribution in OSPF and MPLS,” International Conference on Emerging Trends in Engineering and Technology, ICETET, July 2008.
- [2] Nader F.Mir., Albert Chien, “Simulation of Voice over MPLS communications Networks,” IEEE ICSS’02,conference.
- [3] Rahman M.A.,Kabir A.H., Lutfullah K.A.M., Hassan M.Z. and Amin M.R., “Performance analysis and the study of the behavior of MPLS protocols,” International Conference on Computer and Communication Engineering ICCCE, 13-15 May 2008, Page(s):226 – 229
- [4] Haris Hodzic, Sladjana “Traffic Engineering with Constraint Based Routing” Zoric 50th International Symposium ELMAR-2008, 10-12 September 2008, Zadar, Croatia.
- [5] Junaid Ahmed Zubairi Ph.D., *Member, IEEE* “Voice Transport Techniques over MPLS”
- [6] Xipeng Xiao, Alan Hannan, and Brook Bailey, Global Center Inc. Lionel M, NI, Michigan State University. “Traffic Engineering with MPLS in the Internet”
- [7] David Wright, University of Ottawa “Voice over MPLS Compared to Voice over Other Packet Transport Technologies” IEEE Communications Magazine • November 2002.
- [8] Md. Rahman ,Kabir A.H., Lutfullah K.A.M., Hassan M.Z. and Amin M.R., “Performance analysis of MPLS Protocols over conventional,”
- [9] Liwen He, Paul. Botham “Pure MPLS Technology”. The Third International Conference on Availability, Reliability and Security, IEEE.
- [10] Daniel O. Awduche, “MPLS and Traffic Engineering in IP Networks”, IEEE Communications Magazine December 1999.
- [11] Anoop Ghanwani, Bilel Jamoussi, and Don Fedyk, Peter Ashwood-Smith. “Traffic Engineering Standards in IP Networks Using MPLS ” IEEE Communication Magazine December 1999.
- [12] Jong-Moon Chung, Elie Marroun, Harman Sandhu, and Sang-Chul Kim “VoIP over MPLS Networking Requirements” ICN 2001, LNCS 2094, pp. 735–744, 2001.

- [13] K.Salah and A.Alkhoraidly “An OPNET-based simulation approach for deploying VoIP”, INTERNATIONAL JOURNAL OF NETWORK MANAGEMENT Int. J. Network Mgmt 2006; 16: 159–183.
- [14] BUR GOODE “Voice Over Internet Protocol (VoIP)” PROCEEDINGS OF THE IEEE, VOL. 90, NO. 9, SEPTEMBER 2002.
- [14] Marcos Portnoi, Joberto S.B.Martins “TARVOS – an Event-Based Simulator for Performance Analysis, Supporting MPLS, RSVP-TE, and Fast Recovery”
- [15] Bernard Fortz, Jennifer Rexford, Mikkel Thorup “Traffic Engineering With Traditional IP Routing Protocols”
- [16] MPLS Model Description. OPNET Documentation.
- [17] Configuring Applications and Profiles. OPNET Documentation.
- [18] Simulation Methodology for Deployment of MPLS. OPNET Documentation.
- [19] Umesh Lakshman and Lancy Lobo “MPLS Configuration on Cisco IOS Software” published by Cisco Press Oct 17,2005, Copy right 2006