



Performance Analysis of Voice over Multiprotocol Label Switching Communication Networks with Traffic Engineering

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Abstract: - 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].

Key Words: - MPLS, IP networks, VOIP

I. Introduction

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.

II. Overview

We discuss 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.

III. 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].

IV. 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.

V. 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.

VI. 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]. **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.

MPLS Simulation model

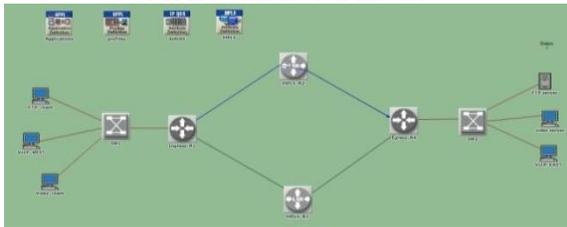


Figure 1 MPLS Simulation Model

Fig.1 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.1) there exists 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.2) the two paths are defined as IP_R1-> IP_R2->IP_R4 and IP_R1->IP_R3-> IP_R4.

Conventional IP Simulation Model

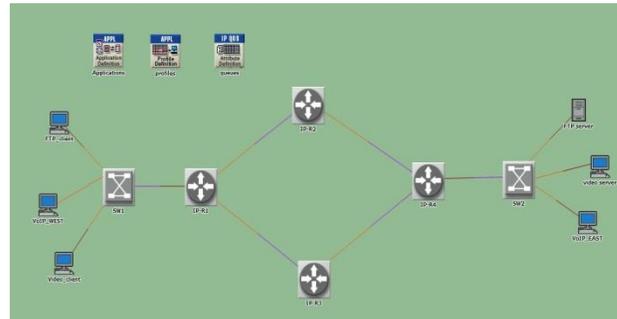


Figure 2 IP Simulation Model

Fig.2 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.

VII. RESULTS AND ANALYSIS

Comparison of Performance Metrics

The results are shown in the Fig.3, Fig.4, Fig.5 and Fig.6. 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.

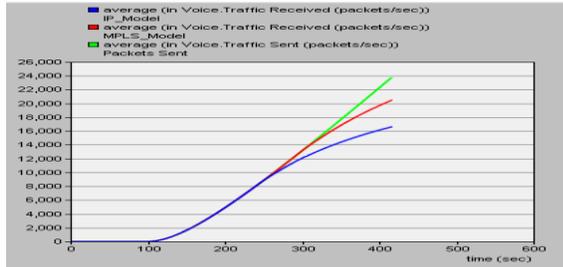


Figure 3 Voice Packet Send and Received

The Fig.3 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.3 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 seconds, this increases the throughput in the MPLS network.

The Fig .4. 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.3. This variation at these time instants can be seen for the remaining performance metric. The voice packet delay variation shown in Fig.5 has same variations in graphs as explained here.

Figure 4 Voice Packet Jitter

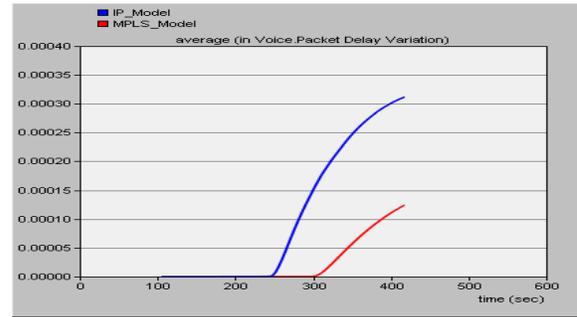


Figure 5 Voice Packet Delay Variation

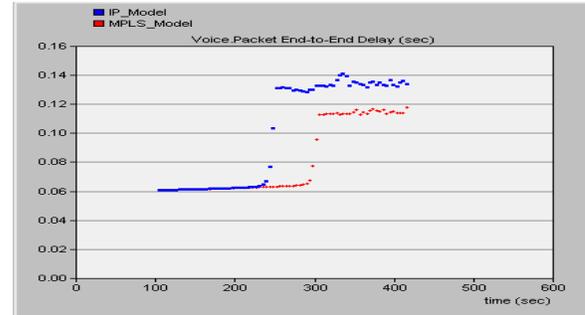


Figure 6 Voice Packet End-to-End Delay

The Fig.6 shows the packet end-to-end delay of MPLS and IP network model. As explained in the above section, 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. 6 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. From the Fig.7 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.8.

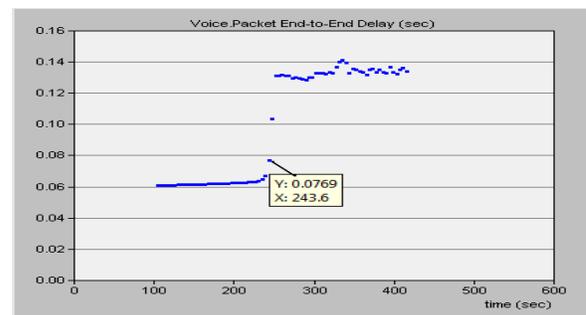


Figure 7 IP Networks End-to-End Delay

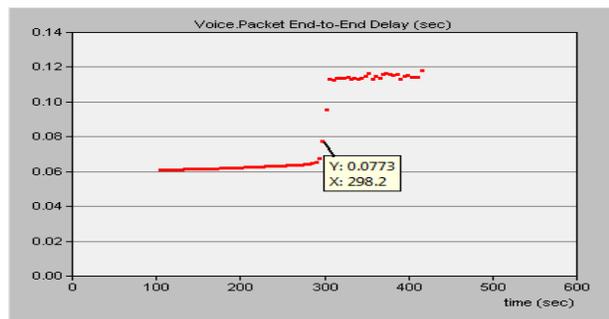


Figure 8 MPLS networks End-to-End Delay

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.7) 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.8) 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.

VIII. CONCLUSIONS AND FUTURE WORK

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 poses 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.

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