International Journal of Advanced Trends in Computer Applications
www.ijatca.com

Evaluating the performance of network traffic for providing real time applications in an Internet Protocol network and Multi Protocol Label Switching Network

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Abstract: *Multiprotocol Label Switching (MPLS) is an evolving standards-approved technology to increase the speed of the network traffic flow and making the network easier to manage. In MPLS a specific path is set up for a given sequence of packets, identified by a label put in each packet, thus saving the time needed for a router to look up the address to the next node to forward the packet. MPLS is called multiprotocol because it works with the Internet Protocol (IP),Asynchronous Transport Mode(ATM), frame relay network protocols.*

It ensures the efficient and reliable delivery of the Internet services with high transmission speed and lower delays. The main feature of MPLS is the concept of Traffic Engineering (TE), which is used for effectively managing the networks for efficient utilization of network resources. This technology is more suitable for implementing real-time applications such as voice and video due to lower network delay, efficient forwarding mechanism, scalability and predictable performance of the services provided by MPLS. In this paper the performance of voice over internet protocol in MPLS and non MPLS network is compared. Here we consider non-MPLS network as conventional IP network. In order to simulate the network we use Optimized Network Engineering Tool 14.5 and comparison is made on various parameters such as Ethernet delay, voice jitter, voice packet end to end delay.

Keywords: *MPLS, LSP, TE, ABE, RRP, QoS , LSP*

1. Introduction

Today Internet is playing an indispensable role in most of the people‘s life due to a large number of applications and services provided by the internet. This results into an increased number of Internet users. It is a challenging task to provide the real-time applications on the traditional IP networks as it doesn‘t provide guarantee of services and Traffic Engineering (TE). When a packet arrives at each node, it consults a pre computed routing table to decide which interface to exit this packet. Routing tables are built using routing protocols or through manual configuration. Moreover Internet Protocol networks provides minimum predictability of services which is unacceptable for the applications like telephony and multimedia services [8].

With traditional IP routing, it is impossible to provide a mechanism for load balancing across unequal cost path (with the only exception of Cisco proprietary EIGRP), because there is always a single best path towards a destination, while taking into account multiple path metrics. All packets from source to destination adopt the best path through independent look up table at each intermediate node. In extreme situations, the best path has to carry a large volume of traffic, so that packets may get drops or inherit a certain level of latency, whereas the bandwidth along the not-so best paths remains idle.

MPLS based solution provides better performance in implementing the real time applications such as VoIP application. [1]. Multi-Protocol Label Switching (MPLS) is an emerging technology which utilizes Resource Reservation Protocol (RRP) and Path selection based on Available Bandwidth Estimation (ABE) to securely manage traffic from the source to Multi console MPLS VPN cloud [1]. MPLS networks plays an important role in the next generation computer networks by providing Quality of Service (QoS) and traffic engineering. MPLS networks provide high performance packet control and forwarding mechanism, which forwards the packets based on the labels [6]. It overcomes the limitations like excessive delays and high packet loss of IP networks by providing scalability and congestion control. So MPLS networks are considered ideal for multimedia applications.

2. Research Methodology

The research work involves the detailed study of the literature of IP and MPLS network. 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. For this work both qualitative and quantitative approaches are used. In this paper, different network scenarios are implemented to gain a better knowledge about the characteristics of diverse applications. Different evaluation methods are chosen based on different parameters. The theoretical knowledge gained is implemented in the Optimized Network Engineering Tool research simulator. A network model is designed and results are collected to compare the performance of voice over MPLS network with IP network.

3. VoIP implementation in Internet Protocol Networks and MPLS Networks

It is a very challenging task to implement the real-time applications like VoIP in the conventional IP network. Internet protocol networks work on the best-effort service which doesn‘t guarantee the delivery of the services. It is difficult to implement VoIP application.

In an IP network, routing is designed to calculate the shortest path towards the destination but not the best path. The routing in an IP networks is done in the Network layer which is slower than the switching.. In IP networks, most links are either over-utilized or under-utilized caused by its routing process, which results in congestion for over-utilized links.

It is difficult to implement Traffic Engineering in an IP network. The Voice over Internet Protocol application requires the guarantee of services with a minimum delay and a low packet loss. This can be achieved by using MPLS networks. MPLS-based networks use existing Internet Protocol mechanisms for addressing elements and routing traffic. MPLS adds connection oriented capabilities to the connectionless IP architecture.

In Multi Protocol Label Switching networks, the data packets are sent based on the Label Switched Path (LSPs). These LSPs are set based on constraints like the available bandwidth, administrative policies etc. These paths are the virtual connections which are used to send the packets reliably. This is desirable for transmitting the Voice traffic.

4. Opnet Network Model

4.1 The Conventional Internet Protocol Network Model:

The Figure 1 shows the simulation scenario based on the conventional IP network without considering the concept of Traffic Engineering. In this scenario normal IP routers are used. MPLS definition attribute, is not considered and the packets are routed using OSPF protocol . The Voice traffic is transmitted between the Voice node and node 18.Node 18 is configured to support voice services. VoIP calls are set up between these nodes.

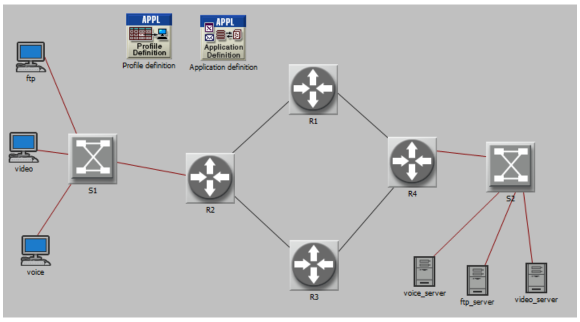

Figure 1: Conventionl IP Network

4.2 MPLS Network Model

The Figure 2 shows the MPLS network based scenario which consists of the network elements such as two LERs (Ingress_R1 and Egress_R4), two LSRs (MPLS_R2 and MPLS_R3), two VoIP stations (Voice computer node and Voice server) and two switches (SW1 and SW2)

DS3 links (44.736 Mbps) and 100Mbps links are used for connecting all the routers connecting workstations to the two switches. Traffic Engineering 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 Figure 2.

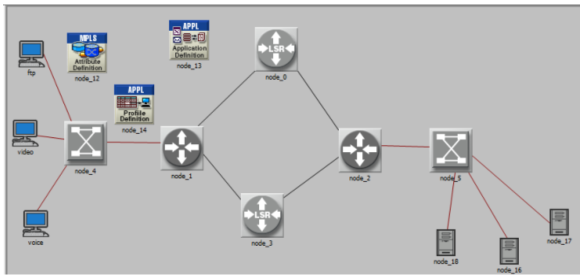

Figure 2 : MPLS Network Model

5. Simulation

The simulator used here is Optimized Network Engineering Tool. The main feature of OPNET is that it provides various real life network configuration capabilities that make the simulation environment close to reality [6]. The advantages of Optimized Network Engineering Tool 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.

In this paper, the simulation is done by transmitting, the VoIP traffic from source to destination in the two networks (MPLS and Traditional IP networks).The main aim is to compare the performance of VoIP traffic in both IP and MPLS networks with respect to different performance metrics such as voice jitter, ethernet delay, voice packet end to end delay. The simulation results are analyzed to determine the efficient network used for transmitting VoIP traffic.

An application definition attribute is used to model an application in OPNET. This attribute consists of predefined applications which can be modified as per the user's requirements. Some of the predefined applications in application definition attribute are Voice, Video, FTP, Email, HTTP, Database etc.

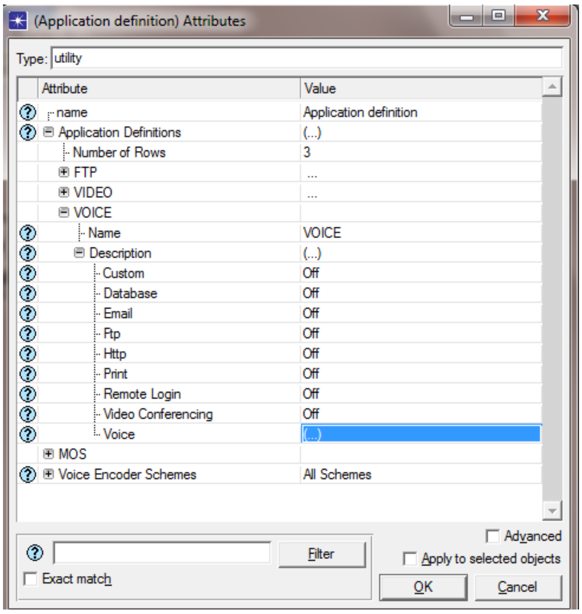

Figure 3: Application Definition Attribute

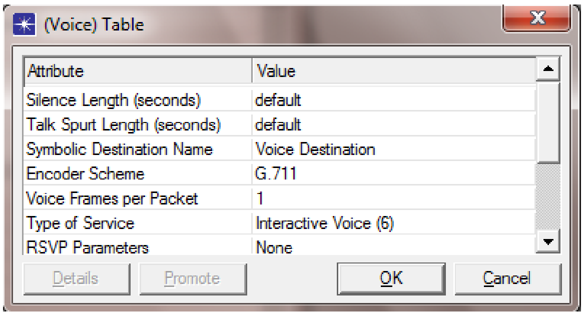

Figure 4: Voice Table

Figure 3 shows the Application Definition attribute used in our simulation model. Three applications such as 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 Figure 4). 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 workstation named voice and server voice are the workstations (shown in Figure1 and Figure 2) 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

Figure 5 shows the Profile Definition object used in our simulation. It describes the start time of the simulation, which is set to 100 seconds and the VoIP application is repeated continuously till the end of the simulation. It means that VoIP calls are established between workstations voice and server voice starting at 100 seconds and the calls are added continuously till the end of simulation.

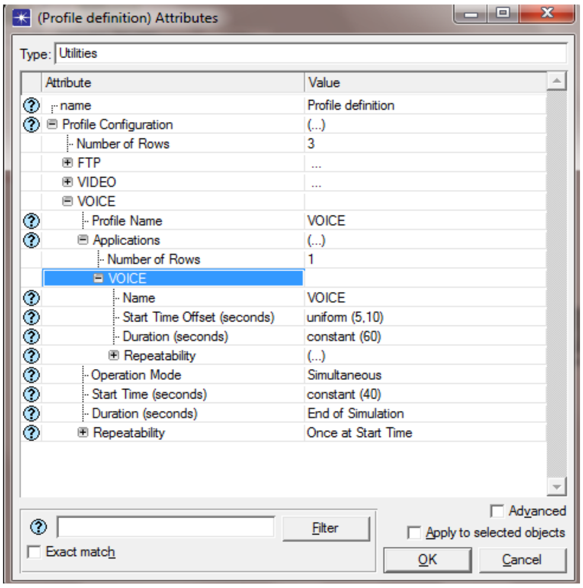

Figure 5: Profile Definition

6. Results

i. Ethernet delay

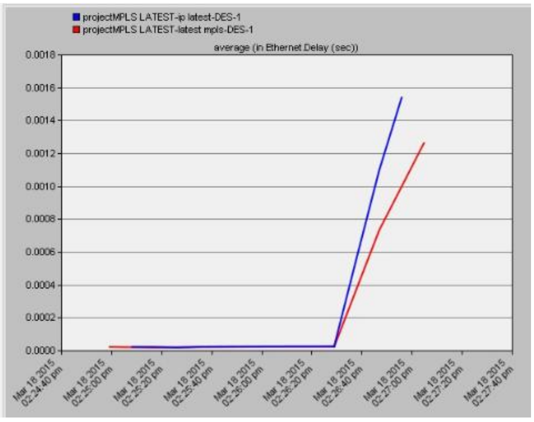

Figure 6: Ethernet delay

As seen from figure 6, the Ethernet delay provided in IP network is more as compared to MPLS network

ii.Voice Jitter

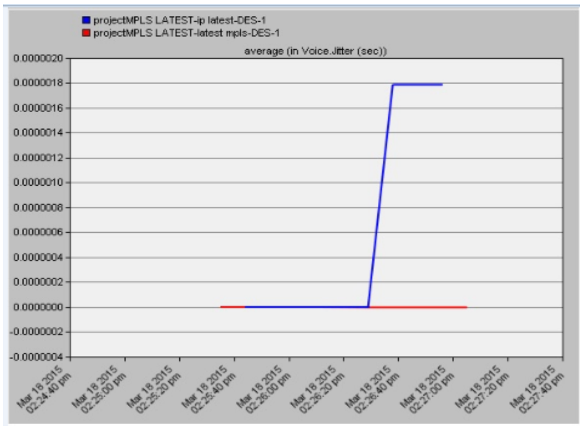

Figure 7: Voice Jitter

As it is clear from the figure 7 that the jitter in MPLS network is less than IP network

iii.Voice packet end to end delay

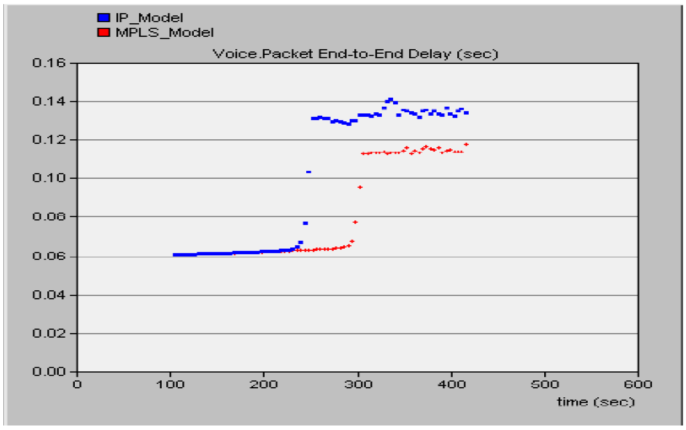

Figure 8 : Voice Packet End to End delay

The voice packet end-to-end delay in a network should not increase above the threshold value of 80 milliseconds in order to establish VoIP calls of good acceptable quality. From the Figure 8 it is observed that the voice packet end-to-end delay in an IP network exceeds the threshold value at 240 second. whereas in MPLS network it reaches at 300 seconds. The threshold value is reached early in IP network as compared to MPLS network. This is because the concept of Traffic Engineering is implemented in MPLS network.

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Author Profile

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