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MPLS as Backbone for Site to Site VPN Networks in VOIP Applications

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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 in MPLS network and conventional Internet Protocol (IP) network. OPNET modeler is used to simulate the both networks and the comparison is made based on the metrics such as Voice jitter, Voice packet end-to-end delay, voice delay variation, voice packet send 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 maintained for a given network by imitating the real network on the OPNET simulator.***

Keywords: Delay, jitter, MPLS, VoIP and Optimized Network Engineering Tool (OPNET)

1. Introduction

1.1 Background

Multiprotocol Label Switching (MPLS) is a Layer-2 switching technology. MPLS-enabled routers apply numerical labels to packets, and can make forwarding decisions based on these labels. The MPLS architecture is detailed in RFC 3031.

MPLS reduces CPU-usage on routers, by allowing routers to make forwarding decisions solely on the attached label, as opposed to parsing the full routing table.

Labels can base on a variety of parameters:
- Destination IP network
- Source IP address
- QoS parameters
- VPN destination
- Outgoing interface
- Layer-2 circuit

MPLS is not restricted to IP, or any specific Layer-2 technology, and thus is essentially protocol-independent. Labels are applied to and removed from packets on edge Label Switch Routers (edge LSRs). Only edge routers perform a route-table lookup on packets. All core routers (identified simply as LSRs) in the MPLS network forward solely based on the label.

As a packet traverses the core MPLS network, core routers will swap the label on hop-by-hop basis.

1.2 The MPLS Label:

Two forms of MPLS exist:
- Frame Mode MPLS – utilizes a 32-bit label that is injected between the Layer-2 and Layer-3 headers.
- Cell Mode MPLS – used with ATM, and utilizes the VPI/VCI fields ATM header as the label as shown in figure 1 below.

Label	Experimental	Bottom-of-Stack	TTL
20 bits	*3 bits*	*1 bit*	*8 bits*

Figure 1: MPLS Header

- Label (20 bits) :
- Experimental (3 bits) – This field is officially undefined, but is used by Cisco as an IP precedence value.
- Bottom-of-Stack (1 bit) – This field indicates the last label, as multiple labels are supported in the same packet. A value of 1 identifies the last label in the stack.
- TTL (8 bits) – This field indicates the number of router this label can 'live' through.

1.3 IP Networks

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

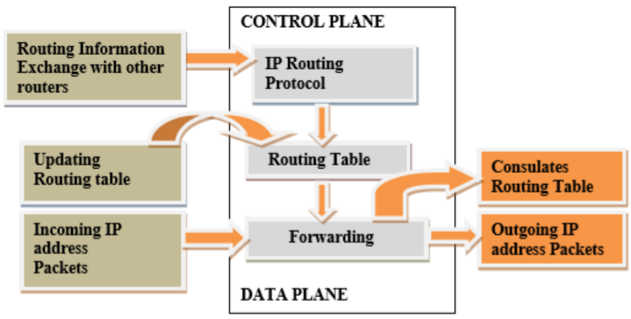

Figure 2: Traditional Router

1.4 Voice over Internet Protocol (VoIP)

Implementation of the real time applications such as voice and video in Internet made it the most desirable and cost-effective service to everyone. The VoIP is also known as Internet Telephony. VoIP is the real-time data and it is transported in Internet by using Real Time Protocol (RTP). RTP consists of data and a control part. The control part is called as Real Time control protocol (RTCP) .VoIP packet is transported by using the set of RTP/UDP/IP protocols. Although TCP/IP is a reliable communication protocol suite it is not used in real-time communications due to the fact that it uses acknowledgement/retransmission feature which would lead to excessive delays. Since voice communications are less tolerant to delays TCP/IP are not suitable.RTP is used with UDP to provide end-to-end transmission of real-time data where RTCP is used for monitoring of the link.

2. Methodology

The network is implemented by using same topologiesafter the network implementation; start to configure the attributes for different routings types, four parameters (voice packet end-to end Delay, Voice jitter and throughput) has considered to evaluate the network performance for VOIP according two routing technology.

3. Network Scenario

The network components used in the models running on OPNET device used in the network are 40 end device two Ethernet switch directly connected to end devices and application server work as a VOIP Servers. To represent and supporting Voice transaction between workstations and switch, the switch (Ethernet 32) are used. The IP packets arriving on the input interface are switched to the appropriate output interface based on packet destination IP address.

This workstation requires a fixed amount of time to route each packet, as determined by the "IP Forwarding Rate" attribute of the node. Packets are routed on a first-come-first-serve basis and may encounter queuing at the lower protocol layers, depending on the transmission rates of the corresponding output interfaces. And all of designs have same topologies and the different inside routing technique as shown in figure 3,4 below for MPLS Network and normal IP routing network.

4. Network Architecture

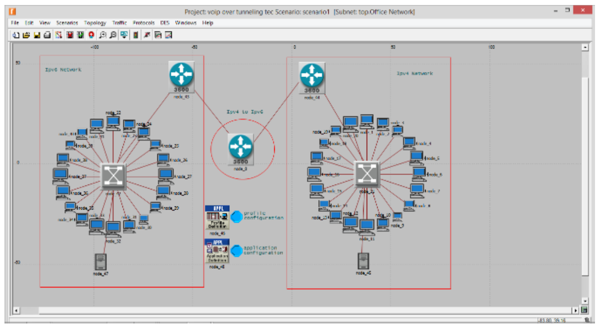

Figure 3: MPLS Network scenario

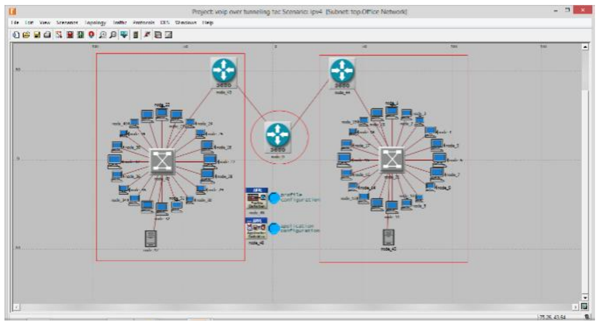

Figure 4: IP Routing Network scenario

5. Results and Discussion

The simulation ran for 1 hour (3600 sec), sufficient to overview of the network's behavior. The results of the network scenarios are shown in Fig. 5, 6, 7 and 8IP Routing represented in blue, MPLS Traffic represented in red.

A. The delay obtained from sender and receiver we can calculate the network delay which should not exceed 80ms or (150-25-45). The network delay is the sum of the delays given from propagation, transmission and queuing delay in the network.

Delay in the Figure 5 below shows the comparison of delay The MPLS Technique has a Lower Delay than Normal IP Routing

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

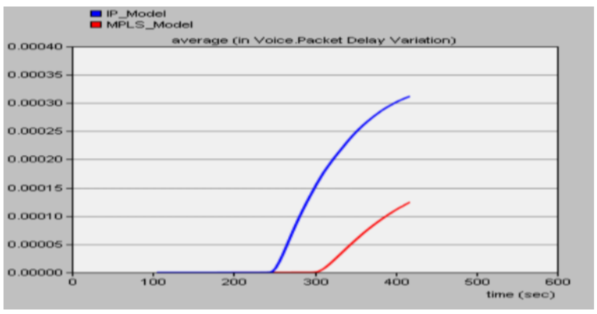

Figure 5: Voice Packet delay variation

B. Voice packet jitter in the Figure 6below shows the comparison of jitter the ip routing had a frequently value of jitter and the MPLS its more stable or less jitter value .

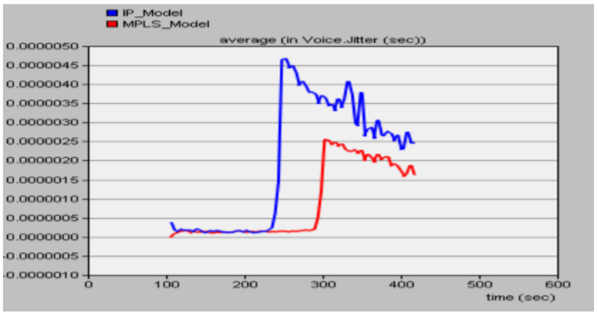

Figure 6: Voice Packet Jitter

C. Voice packet sent and receive

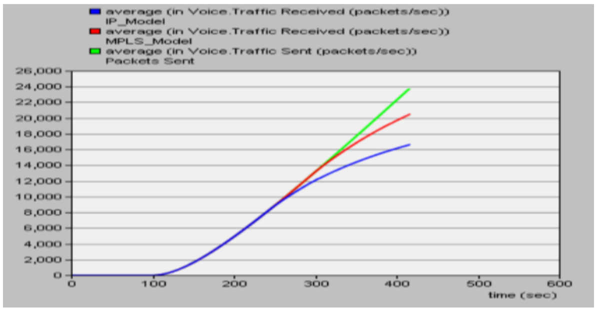

Figure 7: Voice Packet sent and receive

We saw that the normal ideal value of packet sent and received in green color and the other value representing that IP routing its less value of data packet sent and received

D. Voice Packet End-to-End Delay

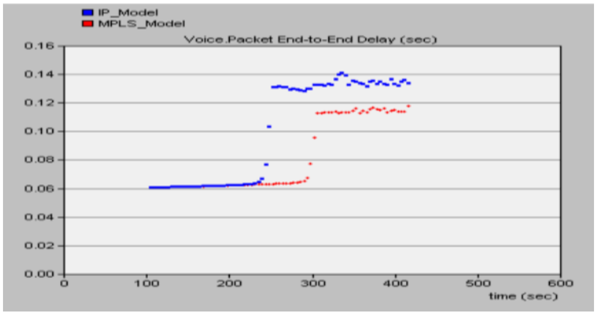

Figure 8: Voice Packet End-to-End Delay

6. 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.

The author made a comparative analysis of MPLS over Non-MPLS networks and shows MPLS have a better performance over traditional IP networks.

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