A Study and Simulation of VoIP in MPLS Network

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Abstract -- **The key necessity of today's world is information gathering, processing and distribution that depend on two major technologies: Computer & Communications. There has been a merger of these two technologies giving birth to computer communication era. Multiprotocol Label Switching (MPLS) is an emerging technology and plays an important role in the next generation networks by providing Quality of service (QoS). It overcomes the limitations like excessive delays and high packet loss of IP networks by providing scalability and congestion control. The key feature of MPLS is its Traffic Engineering (TE) which is used for effectively measuring the performance of the networks and efficient utilization of network resources. MPLS provides lower network delay, efficient forwarding mechanism, scalability and predictable performance of the services which makes it more suitable for implementing real-time applications such as Voice and video. In this paper, performance of Voice over Internet Protocol (VoIP) application is implemented in MPLS network and conventional Internet Protocol (IP) network. NS-2 (Network Simulater-2) is used to simulate the both networks and the comparison is made based on the metrics such as Voice packet delay, voice packet lost probability, throughput, voice packet send and received.**

Keywords: MPLS,LDP, LSP, VoIP, Traffic Engineering, Forward Equivalence Class (FEC), NS-2.

I. INTRODUCTION

THE key necessity of today's world is information gathering, processing and distribution that depend on two major technologies: Computer & Communications. There has been a merger of these two technologies giving birth to computer communication era. Early computer networks carried continuous bit streams over physical links in a technique called circuit switching. This was well suited to transmit voice or real time data from a single sender to a single receiver (unicast communications). In this kind of network a single physical link failure had dramatic consequences, leading to the interruption of all communications that was using the failed link. The Internet today is a datagram packet switched network that fixes this drawback by cutting data into small chunks called packets. These packets are individually routed through the network, so two packets from the same communication are individually handled in the network.

Therefore if a link fails, packets can be rerouted to avoid the failed link and communications are not interrupted. This feature of a datagram switched network is called resilient because it hides network failures from the end users. On the other hand it is more difficult to manage flows of data in a datagram packet switched network than in a circuit switched network because each packet is handled individually. The convergence of voice and data communications over a single network infrastructure is expected to happen over IP-based networks. Traditional IP networks offer little predictability of service, which is unacceptable for applications such as telephony, as well as for emerging and future real-time applications [1].

Multicast traffic is usually distributed over a tree structure. A single failure will affect all members at the downstream of the failure point. A couple of preplanned fault-tolerant multicast schemes have been studied previously [12–14]. Some of them use link protection [12]. IP technology is dominating over every communication system in the globe because it supports a vast variety of services and it's compatible with almost everything. Because of that, terms such as Everything over IP (EoIP), ALL IP and Next-Generation Network (NGN) started to appear in the network world. "The future of voice communications, much like most everything else, is going with some type of Internet Protocol (IP) implementation." [2]. NGN is a telecommunication core and access network architectural change and its idea is that one network can be able to transport all types of data and services by encapsulating them into packets, similar to those used on the Internet.

Today, the Voice over Internet Protocol (VoIP) is a very growing service because of the increasing utilization of IP networks in modern enterprises. Its major benefit is the reduction of the operational and infrastructural cost that is vital for disaster management planning. Amer et. al [3] presented a literature review on voice transmission techniques over IP networks and discusses the advantages of the Multi-Protocol Label Switching (MPLS) backbone for transmitting voice packets. This could be a case of an emergent communication case where there is a need for utilizing available infrastructure along with the VoIP protocol. In this paper the authors have focused on the identification of the constraints that affects the quality of service provisioning to the end users and they have suggested a set of modifications to improve it.

An IP/MPLS network that could handle Internet traffic and L2-

based connections (or circuits) offers providers several compelling benefits [4].

Reduced overhead: Multiple networks and their attendant operational and infrastructure costs can be collapsed into one.

Simplified transitioning: Providers with frame relay or ATM networks can transition existing L2 circuits to IP/MPLS networks without impacting end-to-end service.

Expanded services portfolio: IP/MPLS providers can offer frame relay and ATM backbone connectivity along with new and emerging services such as transparent LAN services (TLS) and virtual leased lines (VLL).

Multiprotocol packet forwarding: Providers can transit L2-encapsulated network layer packets that would not normally be routable across the IP/MPLS network. IPv6, IPX, or SNA packets, for example, can be transparently carried across the IPv4 backbone network.

Greater efficiency: Statistical multiplexing allows service providers to offer a suite of services more efficiently with packet-based IP/MPLS networks than over L2 or time-division multiplexing- (TDM-) based networks.

Single-interface access: IP/MPLS providers can bundle L2 connectivity and Internet services over a single user interface. *Increased flexibility:* Minimizing the degree of integration between the customer and provider networks allows a provider to deliver only L2 connectivity and lets users exercise independent routing policies. With so much to offer, tunneling L2 circuits through.

Since the Internet was opened to commercial traffic in 1992, it has grown rapidly from an experimental research network to an extensive public data network. This has increased traffic volume in a geometric progression, and makes it more difficult to support quality of service (QoS) on the Intemet. For the purpose of solving such problem, two research areas are addressed in particular; traffic engineering and high-speed packet forwarding. Gaeil et.al [5] proposed a MPLS simulator, which supports label swapping operation, LDP, CR-LDP, and various options of label distribution function defined in MPLS standards.

A hybrid approach for routing flows in IP networks is used [6] to achieve an optimal network configuration maximizing bandwidth usage (optimality), minimizing re-routing upon failure (reliability) and reducing the signaling overheads resulting from a full IP tunneling (scalability). In this paper the authors formulated the routing of flows in IP networks as a service differentiated model where the IP flows are classified into high bandwidth demanding (HBD) and low bandwidth demanding (LBD) flows at the ingress of the network and handled differently into the core using a hybrid IGPCMPLS approach where the LBD flows are routed over the existing IGP paths while the HBD flows are carried over MPLS bandwidth-guaranteed tunnels. This model can be deployed in heterogenous network environments where HBD flows carrying real-time traffic and LBD flows transporting best-effort traffic are handled differently over a common transport network implementing traffic prioritization in response to natural or man-made emergencies.

Insufficient Quality of Service (QoS) of the emerging multimedia applications is a growing concern that has led the need for research and study. Jas et.al [7] investigated the impact of increased video traffic on QoS parameters and their correlation in next generation network. Multi-protocol Label Switching (MPLS) and Differentiated Services (DiffServ) integration is very useful strategy for today's traffic. MPLS Traffic Engineering (TE) plays an important role in the implementation of network services with QoS guarantees. The aim of this simulation study is to underline how MPLS TE and MPLS Diffserv integration improve the performances of today's networks, and identify opportunities for improvement, and development of new mechanisms to ensure QoS features in future networks.

II. MULTIPROTOCOL LABEL SWITCHING (MPLS)

MPLS is an emerging technology and most of the research is done in this area to evaluate how the performance networks can be improved when MPLS is added on Traditional IP networks. MPLS is a scalable, protocol-independent transport. In an MPLS network, data packets are assigned labels. Packet-forwarding decisions are made solely on the contents of this label, without the need to examine the packet itself. This allows one to create end-to-end circuits across any type of transport medium, using any protocol. The primary benefit is to eliminate dependence on a particular OSI model data link layer technology, such as Asynchronous Transfer Mode (ATM), Frame Relay, Synchronous Optical Networking (SONET) or Ethernet, and eliminate the need for multiple layer-2 networks to satisfy different types of traffic. MPLS belongs to the family of packet-switched networks. Multiprotocol label switching (MPLS) is an extension to the existing Internet Protocol (IP) architecture. By adding new capabilities to the IP architecture, MPLS enables support of new features and applications. In MPLS short fixed-length labels are assigned to packets at the edge of the MPLS domain and these pre assigned labels are used rather then the original packet headers to forward packets on pre-routed paths through the MPLS network [8].

Multi Protocol Label Switching (MPLS) provides a framework for doing more flexible traffic engineering via its explicit routing

capability. Robert et.al [11] provides, MPLS routing models with two different objectives that utilize MPLS explicit routing are presented and discussed. The objectives of this paper were to minimize the network cost and maximize the minimum residual link capacity. The model that maximizes the minimum residual link capacity is found to perform substantially better, in terms of network throughput and packet loss.

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.2.1 [9].
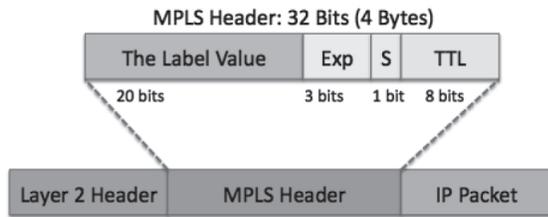

Figure 1. MPLS Header.

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. The tail consist 8-bit TTL (Time to Live) field which had similar function that of TTL field in IP header.

MPLS Architecture: The MPLS Architecture is divided between the *Control Plane* and the *Data or Forwarding Plane*. The components and processes critical operation of MPLS network in the Forwarding Plane figure 2 [9].
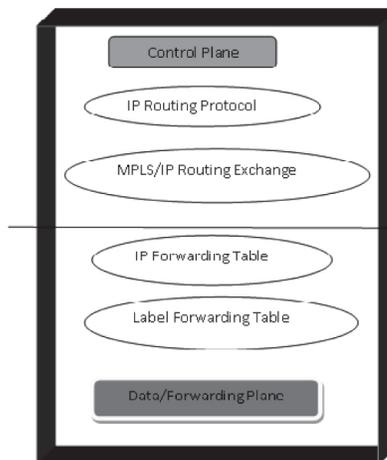

Figure 2. MPLS Architecture.

The MPLS architecture describes the mechanisms to perform label switching, which combines the benefits of packet forwarding based on Layer 2 switching with the benefits of Layer 3 routing. Similar to Layer 2 networks (for example, Frame Relay or ATM), MPLS assigns labels to packets for transport across packet- or cell-based networks. MPLS is also known as 2.5 layer networks which combines the feature of Packet switching and circuit switching. The forwarding mechanism throughout the network is label swapping, in which units of data (for example, a packet or a cell) carry a short, fixed-length label that tells communicating nodes along the packets path (FEC) how to process and forward the data.

Based on switching between different architectures MPLS domain architecture is split into two separate components: the forwarding component (also called the data plane) and the control component (also called the control plane). The forwarding component uses a label-forwarding database maintained by a label switch to perform the forwarding of data packets based on labels carried by packets. The control component is responsible for creating and maintaining label-forwarding information (referred to as bindings) among a group of interconnected label switches. Every MPLS node must run one or more IP routing protocols (or rely on static routing) to exchange IP routing information with other MPLS nodes in the network. In this sense, every MPLS node (including ATM switches) is an IP router on the control plane.

III. VOIP AND MPLS

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 means replacing our traditional circuit switching or time division multiplex (TDM) based PBX systems with LAN switches that use IP/Ethernet handsets in conjunction with telephony servers to provide an IP-PBX which is primarily used for savings on cabling costs, the migration to IP-based solutions, in either a pure LAN switch or a hybrid IP-TDM configuration, has now become a foregone conclusion. The same cannot be said of wide area VoIP implementations [10]. The ability to support multiple traffic categories and provide separate performance parameters for each has made MPLS a natural fit for wide area VoIP services.

Designing A Voice Service Over MPLS: The design process may involve coordination between the voice and data staffs. The overall design process can be defined with following steps [10].

i). The first step is to decide which voice calls will be carried over the MPLS network. The obvious answer is voice traffic that goes between sites that are connected to the MPLS network.

ii). Next, we have to do a voice traffic study. We isolate the voice traffic that will be carried over the MPLS network, identify the busy hour, determine the amount of traffic that must be carried during that period, and compute the number of trunks that will be required to support it with an acceptable level of blocking (i.e. the P-Grade of service).

iii). Once we know the number of trunks that are required, it's time to shift into VoIP mode. We first determine the bit rate required for each voice trunk including all of the packet overhead. The variables in that computation are the voice encoding used (e.g. G.711, G.729A, etc.) and the size of the voice sample carried in each packet.

iv). Once we know the number of bits required per trunk and the number of trunks, we multiply them together to determine the capacity required for real time traffic.

Alvarez, S [9] discussed the Transferring of voice packets over a public network such as Internet has come with some drawbacks such as delay. They have also suggested that this type of packets must be processed in real time which can be useful. A certain set of standards has been created to raise the quality of the VoIP service. 'MPLS' can be considered as one of the solution which can improve the network by reducing packet process time and empowers QoS.

IV. SIMULATIONS AND RESULTS

Network simulators tools provides better simulating environment Compared to the real implementation of networked computers, routers and data links etc which requires more cost and time involved in setting up an entire test bed , network simulators are relatively fast and inexpensive. These tools enables network engineers, researchers to test scenarios, test cases that might be particularly difficult or expensive to emulate using real hardware for example , simulating a scenario with several nodes or experimenting with a new protocol in the network designing and implementing high bride topologies network communication.

In this paper performance of Voice over Internet Protocol (VoIP) application is implemented in MPLS network and conventional Internet Protocol (IP) network. NS-2 (Network Simulater-2) is used to simulate the both networks and the comparison is made based on the metrics such as packet delay, packet lost probability, throughput, packet send and received. Simulation is divided into following parts:

a) At the initial part the simulation of 10 nodes are created in MPLS Network and the VoIP traffic is send from source node (voice Node 0) to destination (voice Node 7) in both networks (MPLS and Traditional IP networks).The performance of VoIP traffic in the both networks are compared by using performance parameters like Packet lost probability and throughput etc. The simulation results obtained and are analyzed to determine the efficient technology used for transmitting VoIP traffic.

b) In this phase we have estimated the performance of both networks in terms of the voice calls that can be maintained in the IP and MPLS networks. This approach can help the network designers to estimate the number of calls, in a real network like Voice or Multimedia data. This is done by designing the real network in the NS-2.

c) The trace files created in the simulations are read and the results are evaluated in terms of graphs and animations.

Proposed Methodology: NS-2 Simulation environment is used in this paper, In the simulation IP networks and MPLS networks are being created with the use of MPLS signaling protocol like LDP.

The results from these simulations are used for comparison between the IP and MPLS networks. Both simulations are based on the common topology as shown in figure 4.1. and figure 4.2. The topology is formed by 8 MPLS (LSR 0 to LSR 7) nodes and 2 IP nodes (n2 and n9) and is distributed. The topology links the source to destination through the MPLS domain. The network consists of total 10 nodes. The Traffic connection was set up between node 0 and node 9 using UDP with CBR of 1000 byte packets and 5ms inter-arrival time which is having mixed node (voice and non voice nodes) environment. node 0 and node 7 are voice nodes.

All links were setup as duplex with 100 ms delay and using Drop Tail Queuing, which serve packets on a First Come First Serve (FCFS) basis. The MPLS network simulation topology is similar to the IP network topology with only difference being that nodes 0 through 8 are MPLS capable, which allow non-shortest path links to be used. The output trace file from the simulation is being used to evaluate the performances of the network in terms of Packet Loss probability, Throughput at the destination node, link utilization and total number of packets received at voice node.
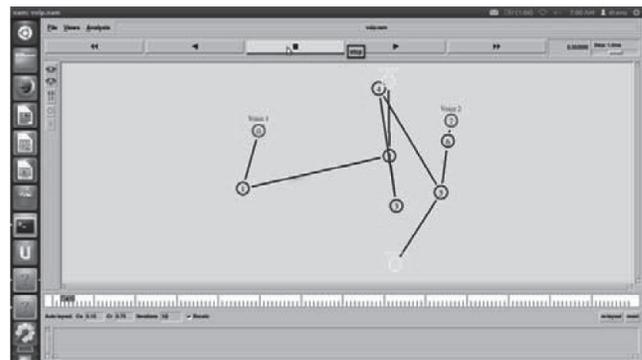

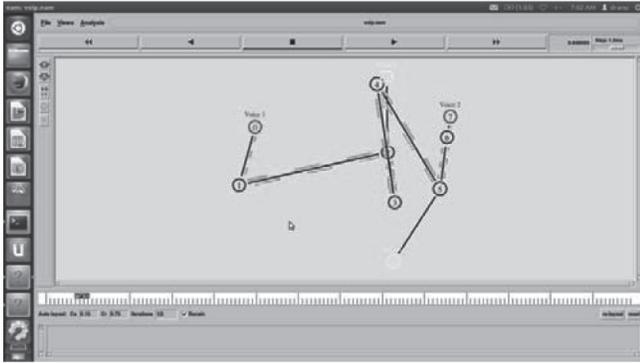Figure 3. IP Networks with Voice Nodes.

Figure 4. MPLS Networks with Voice Nodes.

Results On Simulations: The results are shown below that define performance measures obtained for MPLS and conventional IP networks. From the graphs it is observed that there is an increase in the performance when the VoIP traffic is transmitted using MPLS technology compare to IP technology results are shown below:

(i) Calculation of Packet Lose probability in MPLS and IP Network for VoIP
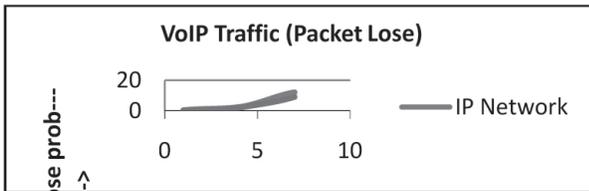
The result shows that the Packet lose probability in IP network for VoIP Traffic is more than in MPLS Network.

(ii) Calculation of throughput in MPLS and IP Network for VoIP
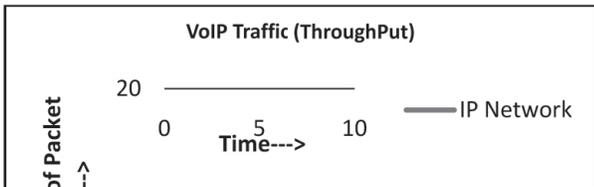
The result shows that the performance of the network for VoIP data traffic is improved with MPLS network over IP network. So, the results in MPLS Networks are considered to be Improved the performance of High speed data networks Like Real time data of Voice and Multimedia Traffic.

V. CONCLUSION

In this paper a study on VoIP is being performed and the performance is evaluated in terms of packet loss probability, throughput, Voice Calls received. NS-2 is used as networks simulation tool in which MPLS and IP networks have been created. The network performance in IP and MPLS network is analyzed through animations and graphs and the trace files are read which is focusing on the performance metrics such as

Packet Loss probability of Voice, Voice packet delay variation, Throughput, Voice End-to-End delay, Voice packet send and received. The paper also used to understand the current scenarios of network performance evaluation and enhancement it also includes the study on MPLS, TE and IP.

VI. REFERENCES

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