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Analysing the usage of Network Resources over a Next Generation Network Using MPLS

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ABSTRACT: Traffic Engineering is a way of propagating data over the network in place of management view, existence of resources and the current and required traffic. It also supports the network supplier to make the best utilization of existing resources. Different utilization of internet needs various levels of facilities to be provided, for example voice traffic needs less delay and very less delay variation. Video traffic requires high bandwidth, etc. Hop-by-hop mechanism is utilized to send a packet in a network employing IP protocol. Routing protocols are employed to make routing tables, to discover a route which has the lesser cost, with respect to its metrics to every destination in the network. This method results in the over-usage of some connections while other connections remain unemployed and are under-used, which causes to the network congestion. MPLS does not send data depending on destination address instead it forwards data according to the labels. Utilizing MPLS network, resources can be analyzed by routing data by less congested route instead of the shortest route utilized in routing protocols. These new routes are produced by hand or by various signalling protocols. MPLS provides support to many characteristics i.e. traffic engineering, VPNs and QoS etc. By using MPLS in traffic engineering we can increase the use of network resources building it more effective. In this research paper a comparison evaluation is done depending on parameters of traffic engineering i.e. effective utilization of bandwidth, throughput and delay etc. for various kind of traffic in their movements throughout the network for both MPLS-TE and conventional IP network. RIVERBED simulator is employed to model the comparison results.

KEYWORDS: Multiprotocol layer switching RIVERBED Modeler, Traffic Engineering, and Virtual Private Network.

I. INTRODUCTION

Next Generation Network (NGN) which is a packet-based network can provide services including Telecommunication Services and can also make use of multiple broadband, QoS enabled transport technologies and in which service-related functions are independent from underlying transport-related technologies. NGN gives revolution in the mobile networks, the capability to assure the seamless mobility with end-to-end QoS present an essential criterion of the success in the NGN. There are many different factors like low throughput and high latency, jitter and delay, which degrade the service level by affecting the QoS, so, here we discuss the requirements to ensure the continuity of end-to-end QoS in such an environment. The design of the multiservice networks is possible by the development of MPLS technology that enables to meet the different needs from various flows which they transport. The new MPLS protocol defined by the IETF with two main aims:

1. To allow a fast routing of IP packages by replacing the function of routing by a switching function much faster due to the substitution of the traditional tables of routing by much smaller matrices of commutation.
2. To facilitate engineering network by providing to the operators the control of the routing of the data which was very complex for the traditional protocols of routing like OSPF. MPLS technology is very crucial for the evolutionary virtual private networks (VPNs) and quality of service (QoS), to meet the future growth the efficient usage of exiting networks is allowed and the speedily correction of mistakes of the failure of host. The use of switching label technology by MPLS to give a seamless service of transport of data for customers was done. Any type of traffic (data, voice, video……etc) can be transported by using it. MPLS is an effective solution for the present day's problems in networks, e.g. speed, scalability, traffic engineering and quality of service (QoS) management. To meet the requirements related to services and bandwidth management for the next generation IP based networks MPLS is a capable tool. The amplification of capabilities of large scale IP networks and the routers forwarding speed is done by it.

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From some time the use of internet is done everywhere and it required different types of new capabilities that can performs the business and enterprise network requirements. The speed and bandwidth is the requirement of these varieties of applications. For existing internet infrastructure the increasing growth in users and volume of traffic is a big problem. Despite these starting challenges and to assemble the service and bandwidth needs through the next generation networks MPLS will have to play an essential role in packet forwarding, switching and routing.

II. MULTI-PROTOCOL LABEL SWITCHING (MPLS)

MPLS was demonstrated to address the IP flaws; it offers extra facilities to the applications employing IP. As demand for multimedia services is greater, traffic engineering has become a necessary requirement for the network service suppliers as it makes the basis of some performance parameters. MPLS offers the solution to the problems of traffic engineering i.e. QOS, speed, network congestion and delay etc. MPLS sends data through labels attached to every packet, these labels are allocated among all the nodes making the network. Constraint-based Routed Label Distribution Protocol (CR-LDP) and Resource Reservation Protocol (RSVP) are the two label distribution which offers support for Traffic Engineering. The first router does the routing lookup, similar to IP routing based protocol, but rather than next hop it discovers the destination router and also a previously determined route from its current location to the final router. The router uses a label (or shim) on the data packets depending on this information. Now other routers as described utilize this label to send traffic further in the network without doing any extra IP lookups. At the final router the packet is forwarded by normal IP routing and label is discarded.

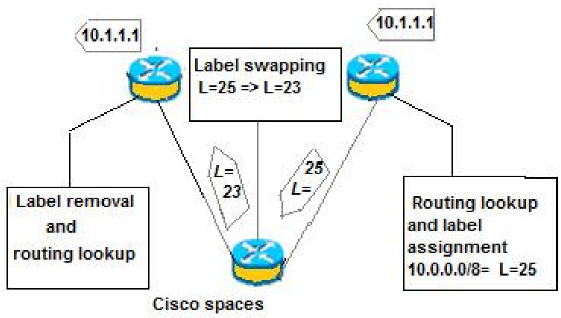

Figure 1. MPLS forwarding

III. MPLS NETWORK INFRASTRUCTURE

MPLS data forwarding or broadcasting of packets needs a label switched path or LSP which is a unidirectional tunnel available between the routers in a MPLS network model. The label edge router (LER) is a router which chooses the initial route and introduces a packet in an MPLS LSP.

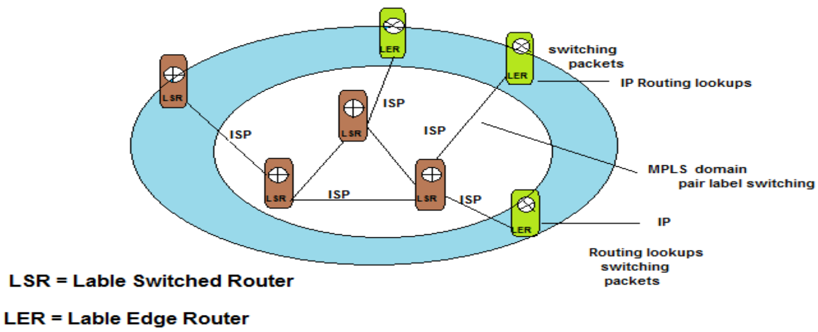

Figure 2. MPLS Network infrastructures

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MPLS network switching in the middle of LSP is done by Label switching router (LSR). The LSP final router also called as egress router eliminates the label. A label distribution protocol is utilized to allocate address/label mappings between neighbouring nodes. MPLS label format utilizes a 32-bit label field, which consists the under described fields. In MPLS, each packet has a label with them. This label is a shim header field. The shim header is inserted between link layer header and IP header of the packet. These packet headers build a MPLS stack. The described figure explains a MPLS stack consisting various headers and its location in the packet.

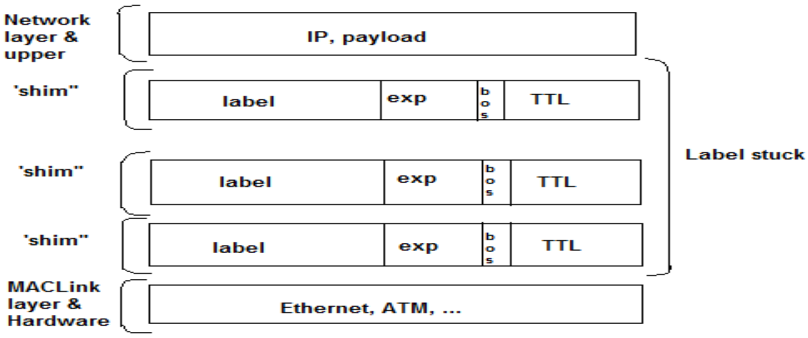

Figure 3. Position of MPLS stack in Network Protocol stack

IV. PROPOSED METHODOLOGY

Simulation is divided in two tasks to satisfy the aim of the paper.

Task 1: In this phase of the simulation the VoIP traffic is transport from source (VoIP_West) to destination (VoIP_East) in the two networks (MPLS and Traditional IP networks).The primary task is to compare the performance of VoIP traffic in the both networks by the usage of performance metrics, i.e., voice jitter, packet End-to-End delay, packet loss and throughput. The simulation results got are analyzed to examine the effective technology used for transmitting VoIP traffic.

Task 2: In this phase, an approach is made to estimate the approximate minimum number of calls that can be arranged in the both networks. This approach can be used to determine the number of calls, in a real network. This is done by designing the real network in the RIVERBED. We use the End-to-End delay performance metric got from the simulation to determine the approximate minimum number of calls maintained in both networks.

Assumptions

It is complicated to predict the traffic behavior in the network as the traffic in network varies from source to destination at anytime. We will simulate the conventional IP and MPLS models by determining the worst case scenario i.e. since we require to estimate the minimum number of VoIP calls that a network can support with acceptable quality. We determine the background traffic excluding the VoIP traffic to be as 50% of link capacity, as mentioned in [16] 60% link capacity is the max-utilization allowed of a link to save it from bursts.

Network design

The simulation of both IP and MPLS networks are hired in the RIVERBED Modeler 14.5. The simulations are established using two scenarios.

1. Scenario 1 made of simulation of MPLS network with TE
2. Scenario 2 made of simulation of IP network without TE.

Both the networks are simulated by determining common topology.

V. MPLS SIMULATION MODEL

Figure 4 shows the MPLS network model which made of the following network elements: 2 LERs (Ingress_R1 and Egress_R4) , 2 LSRs (MPLS_R2, MPLS_R3), 2 VoIP stations (VoIP_West and VoIP_East) Two switches (SW1 and SW2) and DS3 links are used to join all the routers and 100 Mbps links are used for connecting workstations to the two switches.

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TE is implemented in the above simulation model by using CR-LDP signaling protocol, which is configured in RIVERBED by defining FECs in MPLS definition attribute and setting LDP parameters in the routers. The CR-LSP which is set up can be visible in the Fig.4.1 as a blue colored link from Ingress_R1 to Egress_R4 through router MPLS_R2. When congestion happens in the network, the traffic is directed along CR-LSP path so that the traffic is evenly distributed in the MPLS network. This manages the congestion in the network and enhances the efficiency in utilizing the network resources.

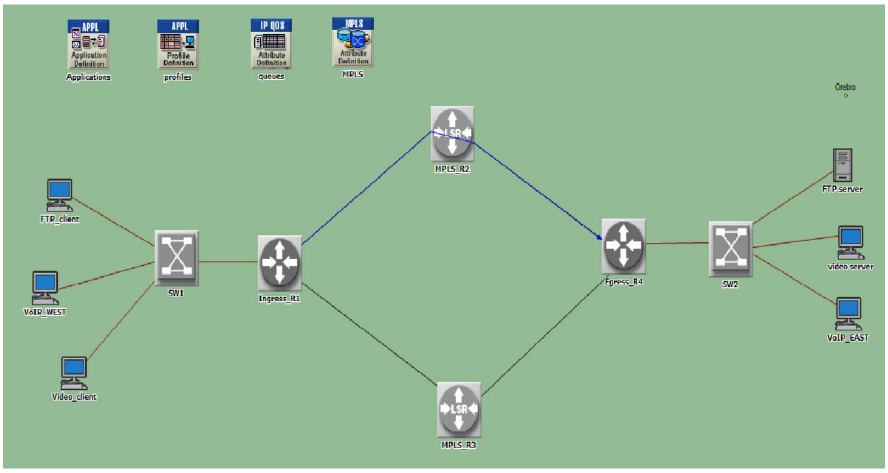

Figure 4: MPLS Simulation model

In this scenario VoIP traffic is transported from VoIP_West to VoIP_East. The VoIP calls are set up in the above model by configuring the *Application definition* and *profile definition* attribute (mentioned in the next section). We simulate both scenarios in order to get packet end-to-end delay, voice jitter, packet sent and packet received values.

VI. COMPARISON OF PERFORMANCE METRICS

The results depicted in the Fig 5, Fig 6, Fig 7 and Fig 8 is the performance metrics achieved for MPLS and traditional IP networks. From the graphs it is realized that there is an increment in the performance when the VoIP traffic is transferred utilizing MPLS technique. For every scenario the simulation duration is 420 sec. The VoIP traffic begins at the 100th sec and finishes at the 420th sec of the simulation time. In both scenarios VoIP calls are joined at specific time intervals such as for every two seconds beginning from 100th sec till 420th sec.

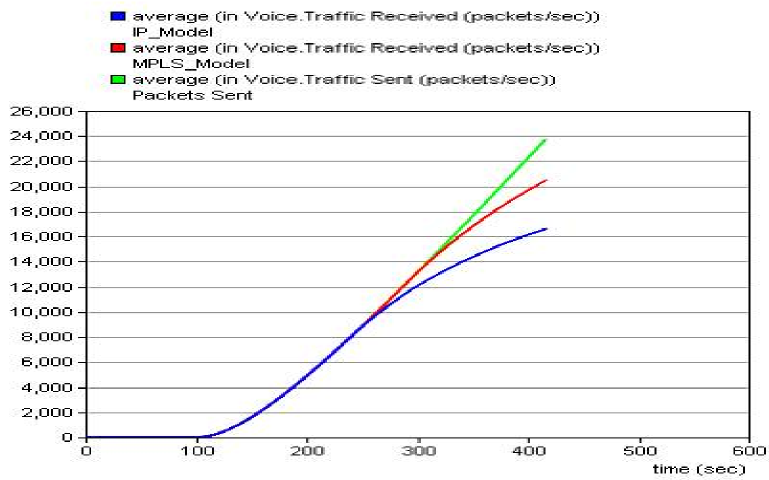

Figure 5: voice packet send and received

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The Fig 5 provides the average no. of packets forwarded and obtained in both MPLS and traditional IP networks. By the simulation end it is realized that MPLS model provides more throughput as compared to the IP model.

The MPLS and IP models simulation are performed considering the background traffic. It is noted from the Fig 5 that voice packets begin to drop from 240 sec in the IP network while in MPLS voice packets are began to loss from 300 second. By the simulation the early packet drop in IP network shows that it cannot set up the VoIP calls with suitable quality after 240 sec. The VoIP calls set up after 240 sec feels loss of information because of the packet loss which cause voice skips and voice breaks. The Voice packet drop in MPLS network begins at 300 sec because of the fact that MPLS deliver the packets with low delay, high transmission speed and furthermore the Traffic Engineering is enforced in the MPLS network which locally decreases the congestion. Because of these factors the packet drop in MPLS networks begins at 300 sec while in IP network.

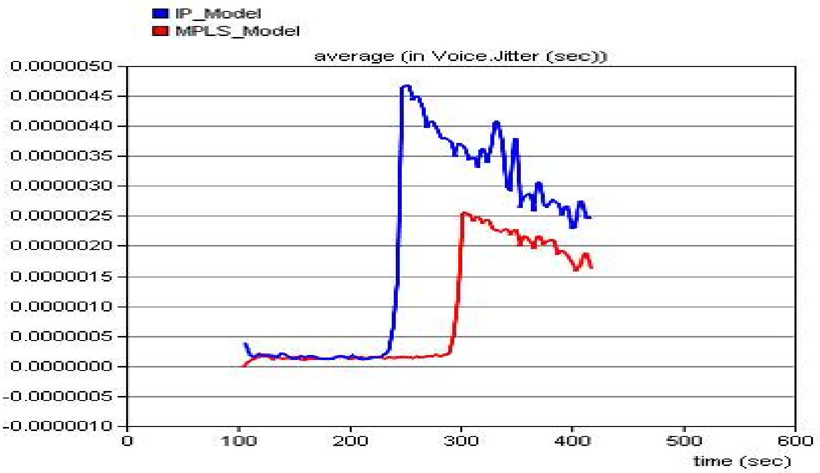

Figure 6: Voice packet jitter

The packet drop begins at 240 sec, this increases the MPLS network throughput. The Fig 6 illustrates the Voice packet jitter of IP and MPLS network model. It is observed that Voice Jitter begins to increase at 240 sec in IP network for MPLS network it begins to increase at 300 sec. The voice packet delay variation depicted in Fig 7 has same variations in graphs as described here.

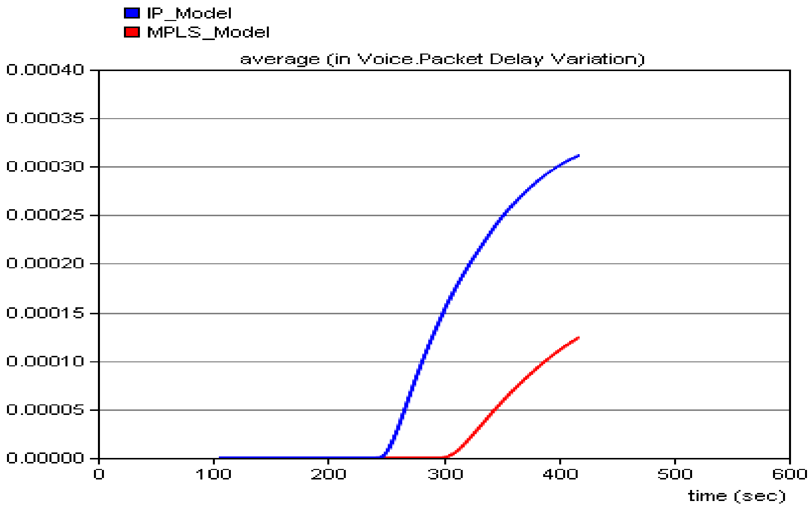

Figure 7: Voice packet delay variation

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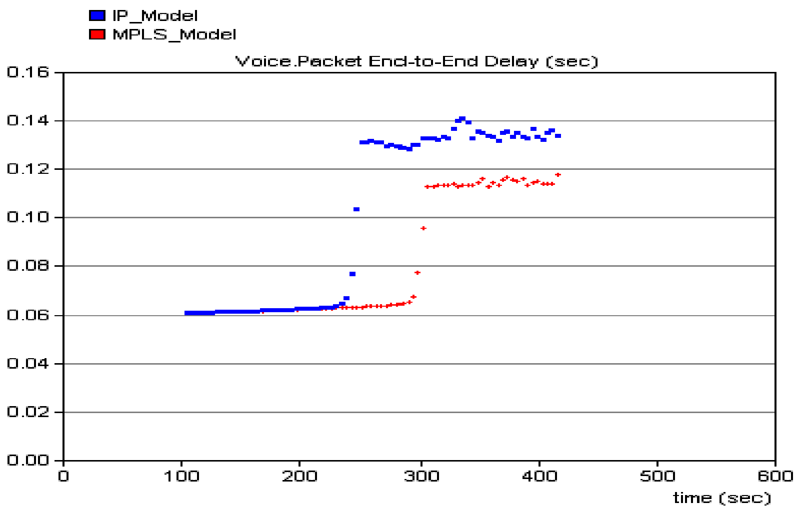

Figure 8: Voice packet End-to-End Delay

The Fig 8 explains the end-to-end delay of packets of IP and MPLS network model. The network End-to-end delay shouldn't exceed above the threshold value of 80 ms in for establishing VoIP calls are of appropriate quality. From the Fig 8 it is observed that end-to-end delay in IP network increases the threshold at 240 second and the MPLS network arrive the end-to-end delay threshold at 300 sec. The IP network arrives the threshold quicker than MPLS network, is because of that TE is implemented in MPLS network. MPLS utilizes CR-LSPs for managing the local congestion. In simulation CR-LSPs is set from Ingress_R1 to Egress_R2 through R3 which is represented by „blue‟. In the MPLS network, the network resources are effectively used in comparison of IP network.

VII. CONCLUSION

The main objective of the paper 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 RIVERBED 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.

REFERENCES

[1] Abusin, A.A., Alam, M.D.J. and Abdullah, J. Testing and Analysis of VoIPv6 (Voice over Internet Protocol V6) Performance Using FreeBSD. International Journal of Communications, Network and System Sciences. 5: p. 298-302, 2012

[2] Al-Ani, M.S. and Haddad, R.A.A. IPv4/IPv6 Transition. International Journal of Engineering Science and Technology,. 4 (12): p. 4815-4822, 2012

[3] Ali, A.N.A. Comparison study between IPV4 & IPV6. International Journal of Computer Science Issues, 9(3): p. 314-317, 2012

[4] AlWehaibi M, Kadoch A, and ElHakeem A, "Packet Loss Probability for DiffServ Over IP and MPLS Reliable Homogeneous Multicast Networks," *Information Processing Letter*, Volume 90, Issue 2, pp. 73 - 80, April 2004

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(An ISO 3297: 2007 Certified Organization)

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[5] Anouari, T. and Haqiq, A. Performance Analysis of VoIP Traffic in WiMAX using various Service Classes. International Journal of Computer Applications,. 52 (20): p. 29-34, 2013
[6] Ayokunle, O.O. Integrating Voice over Internet Protocol (VoIP) Technology as a Communication Tool on a Converged Network in Nigeria. International Journal of Information and Communication Technology Research,. 2 (11): p. 829-837, 2014
[7] Bernard Fortz, Jennifer Rexford, MikkelThorup “Traffic Engineering With Traditional IP Routing Protocols”
[8] Brak, S.E., Bouhorma, M., Brak, M.E. and Bohdhir, A. Speech Quality Evaluation based Codec for VoIP over 802.11p. International Journal of Wireless & Mobile Networks, 5(2): p. 59-69, 2014
[9] Chen, W., Wu, Q., Lin, Y. and Lo, Y. Design of SIP Application Level Gateway for IPv6 Translation. Journal of Internet Technology, 2004. 5 (2): p. 147-154
[10] Chuah M, Medepalli K, Park S, Wang J, "Quality of Service in Third- Generation IP-Based Radio Access Networks", Bell Labs Technical Journal, vol. 7, issue no. 2, pp. 67–89, 2002.
[11] Dawood, H.A. IPv6 Security Vulnerabilities. International Journal of Information Secirity Science, 2012. 1(4): p. 100-105.
[12] Dey, S and Shilpa, N. Issues in IPv4 to IPv6 Migration. International Journal of Computer Applications in Engineering Sciences, 2011. 1(1): p. 9-13.
[13] Durdagi, A. and Buldu, A. IPv4/IPv6 security and threat comparisons. Procedia Social and Behavioral Sciences, 2010. 2: p. 5285-5291.
[14] Dutta, C. and Singh, R. Sustainable IPv4 to IPv6 Transition. International Journal of Advanced Research in Computer Science and Software Engineering, 2012. 2 (10): p. 298-305.
[15] Falk, T.H. and Chan, W. Performance Study of Objective Speech Quality Measurement for Modern Wireless VoIP Communications. EURASIP Journal on Audio, Speech, and Music Processing, 2009. Vol. 2009: p. 1-11, doi:10.1155/2009/104382.
[16] Filsfils C, Evans J, "Engineering a multiservice IP backbone to support tight SLA's", Computer Networks, vol. 49, pp. 131–148, 2002.
[17] Fineberg V, Sinicrope D, Phelan T et al, "The MPLS UNI and end-to-end QoS", Business Communications Review, pp. 27- 32, Dec 2004.
[18] Gaeil A, Woojik C, "Design and Implementation of MPLS Network Simulator Supporting LDP and CR-LDP", Eighth IEEE International Conference on Networks (ICON'00), Chungnam National University, Taejon, South Korea, Sep 2000.
[19] Guangyi Liu, Xiaokang Lin, “MPLS Performance Evaluation in Backbone Network”, IEEE International Conference, Tsinghua Univ., Beijing, 2002.
[20] Handley, M. Why the Internet only just works. BT Technology Journal, 2006. 24 (3): p. 119-129.
[21] HarisHodzic, SladjanaZoric, “Traffic Engineering with Constraint Based Routing in MPLS Networks”, *Proc of 50th InternationalSymposium ELMAR*-2008, Zadar, vol. 1, pp. 269-272, 2008
[22] Hunt R, "A review of quality of service mechanisms in IP-based networks - integrated and differentiated services, multi-layer switching, MPLS and traffic engineering", Computer Communications, vol. 25, pp. 100–108, 2002.
[23] Ina M , “MPLS DiffServ-aware Traffic Engineering”, White Paper, Part No. 200048-001, Juniper Networks, Inc, 2004.
[24] Ismail, M.N. Performance analysis between IPv6 and IPv4: voice over IP implementation in Campus Network. International Journal of Academic Research, 2012. 4 (5): p. 29-40.
[25] Jamalipour A, Lorenz P, "End-to-end QoS support for IP and multimedia traffic in heterogeneous mobile networks", Computer Communications, vol.29, pp. 671–682, 2006.
[26] Jose L Marzo, EusebiCalle , CaterinaScoglio, Tricha Anjali, “Adding QoS Protection in Order to Enhance MPLS QoS Routing”, Proceedings of IEEE ICC, University de Girona, Spain, May 2003.