International Journal of Emerging Technology in Computer Science & Electronics (IJETCSE) ISSN: 0976-1353 Volume 24 Issue 5 – APRIL 2017.

PERFORMANCE ANALYSIS OF VoIP IN MPLS CORE NETWORK USING VIRTUAL ROUTING INSTANCES (VRF)

[1] D. BADRINARAYANAN, [2] T.DINESH, [3] R.GANAPATHY, [4] A. JAGADEESHWARAN
[1]Asst. Professor, [2 3 4]Final Year UG students, Department of Electronics and Communication Engineering, Panimalar Institute of Technology, Chennai.
Email id: badrinarayanadev@gmail.com, renedinesh@gmail.com, ganapathy24081994@gmail.com.

Abstract--- **Real-time and multimedia applications have grown during when compare to last few years. Such applications require guaranteed bandwidth in a packet switching networks. More over these applications require that the guaranteed bandwidth remains available when a node or a link in the network fails. Multiprotocol Label Switching (MPLS) networks cater to these requirements without compromising scalability. Guaranteed service and protection against failures in an multiprotocol label switching network requires backup paths to be present in the network. Such backup paths are computed and installed at the same time a primary is provisioned. Our focus will be on performance study of internet protocol (IP) & Multi-protocol Label Switching networks in data as well as voice traffic & finally comparing the results for data & voice.**

I. INTRODUCTION

These days, organizations conduct their business operations anywhere at any time. The explosion of social media, the affordable sales of tablets, personal computers and smart phones has lead to “always on” connectivity level. These scenarios have driven the need to improve and apply more techniques to make services more reliable. Voice over internet protocol is a transmission technology that offers a cost-effective and reliable communication tool for data and voice transmission. This technology uses the Internet Protocol (IP) to transmit voice as packets over an IP network. Using Voice over internet protocol protocols, voice communications can be accomplished on any IP network regardless, Internet, Intranets or Local Area Networks (LAN). However, Voice over internet protocol exhibits bounded Quality of Service (QoS) requirement such as low delay, jitter and packet loss. Ensuring the optimum QoS parameters is a must to implement Voice over internet protocol, thus implementing Multiprotocol Label Switching is one of the popular technique now a days. mulit-protocol label switch network has been proven to perform better than non-mulit-protocol label switch network for Voice over internet protocol. mulit-protocol label switch is a tunneling technology used in many service provider networks as it offers better routing delivery in a packet switching network. It is a good packet switching technology that ensures Quality of Service (QoS), useful for multimedia applications, reliable and efficient use of network resources.

Transmission of data and voice over a single connection has raised the security issue which led to the combinations of Voice over internet protocol and virtual private network (VPN) technologies to offer better delivery. Virtual private network technology uses a public telecommunication infrastructure, such as internet but provide secure access to the organization’s network. Delivering real time traffic over data network has been a major challenge to researchers. Voice over internet protocol using conventional routing has high call drops and low voice quality due to delay and packet loss. mulit-protocol label switch-based virtual private network is the best solution for all scales of companies currently deployed virtual private networks to public or private site-to-site communication. Routing protocols forwarded packets on the network and plays an important elements that ensure the performance of data transmission on the internet. Enhanced Interior Gateway Protocol (EIGRP) and Open Shortest Path First (OSPF) are the most used routing protocols that being used. Combination of Voice over internet protocol application on the virtual private network and utilizing the mulit-protocol label switch network seems to offer great transmission quality. Thus, the focus of this paper is to evaluate the performance of the variant interior routing protocol for Voice over internet protocol application on Border Gateway Protocol (BGP) -mulit-protocol label switch virtual private network. The next section is the related works follows by the methodology. Section IV is the results and section V is the conclusion.

II. RELATED WORKS

Many researchers have centered on Voice over internet protocol proposing techniques and the right combination for optimum performance. Business organizations have utilized Voice over internet protocol virtual private network over mulit-protocol label switch network in delivering data, video and voice traffic. comparison of network infrastructure models between ip and mulit-protocol label switch shows that mulit-protocol label switch perform better. Various parameters have been tested aiming to improve the performance of Voice over internet protocol on the internet. Examples of analysis parameters are voice end-to-end delay voice jitter, voice packet delay variation and voice packet send and received, packet loss, throughput and mean opinion score (MOS). In Voice over internet protocol application coder-decoders (Codec) also influence the transmission performance however GNS3 is accepted as most used codec in Voice over internet protocol application. Besides that, security is an avoidable criterion in Voice over internet protocol application, thus virtual private network is one of the most widely adopted. The implementation of Voice over internet protocol also depends on routing protocols. Comparison between well known routing protocols such as Routing information protocol (RIP), OSPF and EIGRP shows that determining the best routing protocols are complex. For example, different network topology shows different characteristic of convergence time and different queuing delay.

III. METHODOLOGY

The simulation of Voice over internet protocol on mulit-protocol label switch virtual private network is conducted using OPNET Modeler 14.5. The codec selected for the Voice over internet protocol simulation is GNS3 encoder scheme and Interactive Voice with delay, throughput and reliability for establishing the Voice over internet protocol calls. The bgp-mulit-protocol label switch virtual private network network topology for the Voice over internet protocol simulation. The network consists of Label edge routers (LERs) of Ingress and Egress, two Label switch routers (LSRs) of (site 1 _PE, site 2 _PE), two Voice over internet protocol stations (Voice over internet protocol_West and Voice over internet protocol_East), one video client and one video server

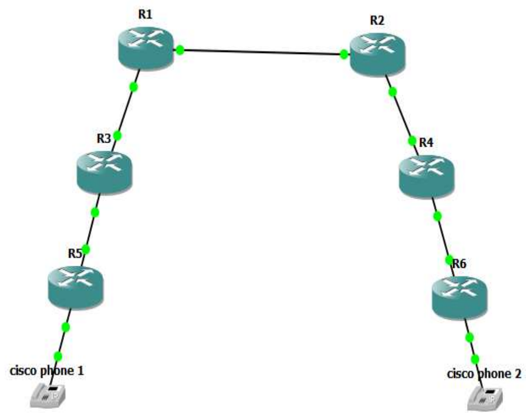


Fig. 1 Simulation Network Topology

The BGP-MPLS VPN simulations are conducted by varying the two targeted interior routing protocol namely EIGRP and OSPF use the same setup as in Fig 1. The rate of a VoIP call is fixed at 500, 2500 and 4000 calls/hour. Average call duration is set to 5 minutes and the voice flow duration is set to 2.5 hours. The simulations are targeted to measure the voice packet end-to-end delay, voice jitter and mean opinion score as to define the overall VoIP quality in both scenarios during the three VoIP scenarios.

IV. SIMULATION RESULT

The results are arranged as follows; VoIP traffic delay, VoIP Jitter and VoIP MOS. The mean value is calculated from the collected parameter.

A. *VoIP Packet End-to-end Delay*

Fig. 2 shows the result of VoIP Traffic delay for 500 calls/hour using two different interior routing protocols OSPF. The delay of started to increase at 900s, while the OSPF delay remains small and constant. The minimum and maximum VoIP traffic delay for EIGRP was 1.49ms and 478.64s, for OSPF was at 10.9ns and 176µs.

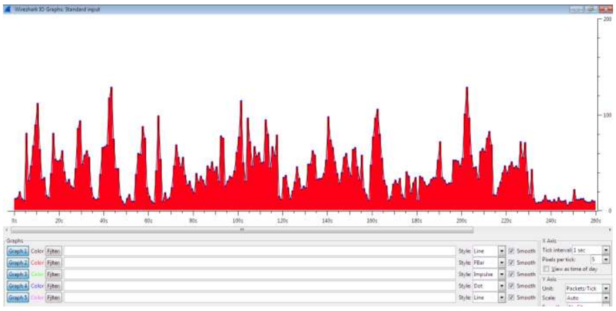


Fig 2. VoIP Traffic Delay (sec) for 500 VoIP calls/hour

Fig.3 shows the VoIP traffic delay for 2500 VoIP calls/hour. The delay of started to increase at 950s, but OSPF delay remains the same. The minimum and maximum VoIP traffic delay for was 0.42µs and 42.85s. Meanwhile the minimum and maximum VoIP traffic delay for OSPF was 2.81ns and 0.54ms.

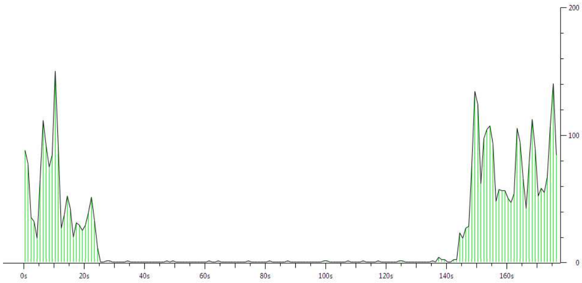


Fig. 3. VoIP Traffic Delay (sec) for VoIP calls/hour

Fig.4 illustrates the delay for 4000 VoIP calls/hour with of EIGRP started to increase at 980s, while there was not much different for OSPF delay. The minimum and maximum VoIP traffic delay for EIGRP was 1ps and 3.64s, while OSPF was 5.95ns and 0.16ms.

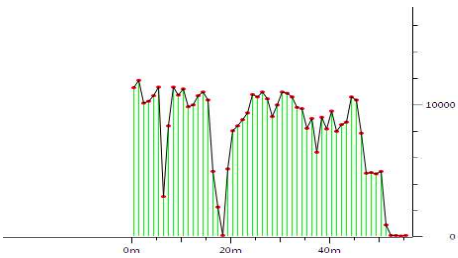


Fig 4. VoIP Traffic Delay (sec)

Table 1 show the mean delay obtained during the simulation. The delay experience under OSPF is smaller than the delay of as the interior routing protocol. It also indicates that as the number call rate is increased, the end-to-end delay also increased on both routing protocol.

VoIP CALLS/HOUR	OSPF-DELAY
500	78.95µs
2500	0.36ms
4000	0.12ms

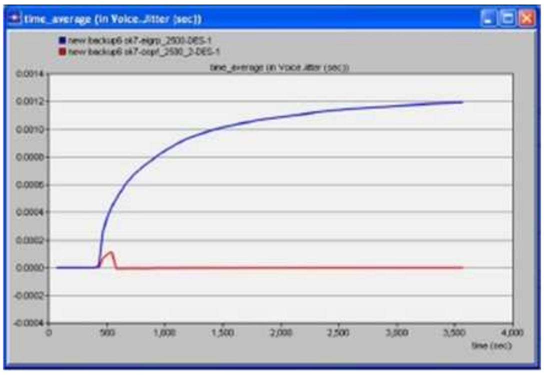


Fig.2500 VoIP call/Hour

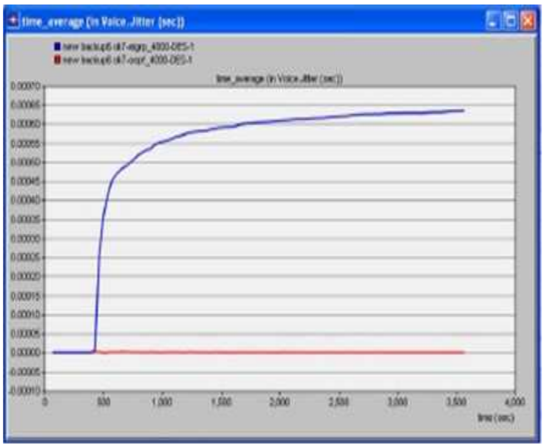
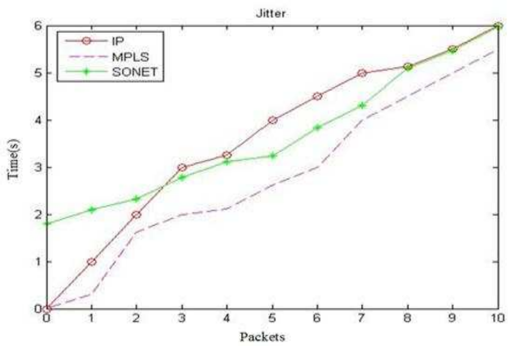
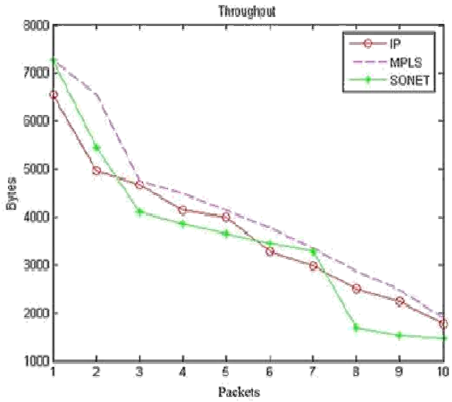


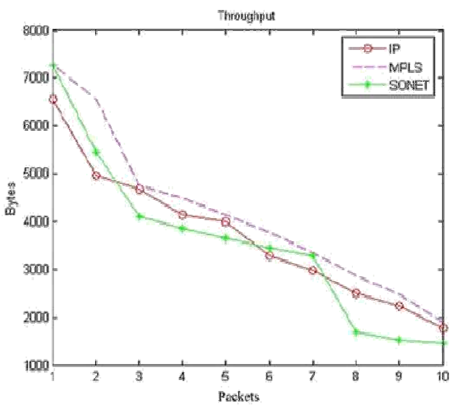
Fig.5 4000 VoIP call/Hour

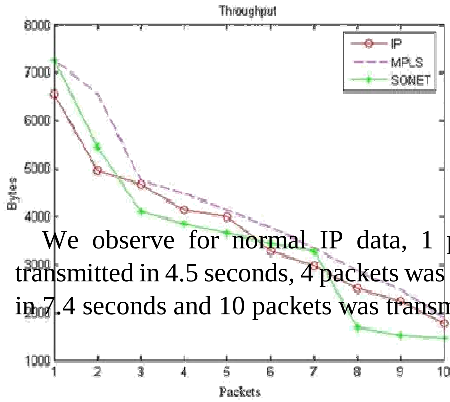
V. SIMULATION RESULTS



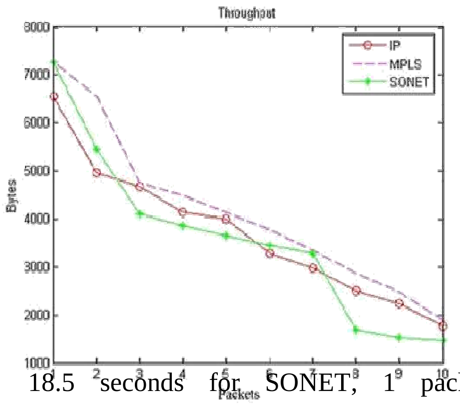


we observe IP data forwarding, 1 packet was transmitted in 1 second, 4 packets was transmitted in 3.2 seconds and 10 packets was transmitted in 6 seconds for SONET 1 packet was transmitted in 2.1 seconds, 4 packets was transmitted in 3 seconds and 10 packets was transmitted in 6 seconds and for MPLS 1 packet was transmitted in 0.2 seconds, 4 packets was transmitted in 1.8 seconds and 10 packets was transmitted in 5.5 seconds. From the above values we can say that the Jitter value is reduced for MPLS when compared with normal IP data forwarding and SONET. In the below figure.5.2 if we observe for normal IP data, 1 packet was transmitted with 6553 bytes, 4 packets was transmitted with 4147 bytes and 10 packets was trans-mitted with 1771 bytes for SONET, 1 packet was transmitted with 7281 bytes, 4 packets was transmitted with 3855 bytes and 10 packets was transmitted with 1456 bytes and for MPLS, 1 packet was transmitted with 7281 bytes, 4 packets was transmitted with 4488 bytes and 10 packets was transmitted with 1883 bytes. In the below figure.





We observe for normal IP data, 1 packet was transmitted in 4.5 seconds, 4 packets was transmitted in 7.4 seconds and 10 packets was transmitted in



18.5 seconds for SONET, 1 packet was transmitted in 4.5 seconds, 4 packets was trans-mitted in 7.3 seconds and 10 packets was transmitted in 17.4 seconds and for MPLS, 1 packet was transmitted in 4.5 seconds, 4 packets was transmitted in 6.6 seconds and 10 packets was transmitted in 17 seconds. From the above values we can say that the Latency value is reduced for MPLS when compared with normal IP data forwarding. In the above figure.5.7 if we observe IP voice packet forwarding, 1 packet was transmitted in 11 seconds, 4 packets was transmitted in 37 seconds and 8 packets was transmitted in 80 seconds for IP data packet forwarding 1 packet was transmitted in 1 second, 4 packets was transmitted in 3.2 seconds, 8 packets was transmitted in 5.2 seconds for MPLS data forwarding 1 packet was transmitted in 0.2 seconds, 4 packets was transmitted in 1.8 seconds, 8 packets was transmitted in 4.2 seconds for MPLS voice packet forwarding 1 packet was transmitted in 7 seconds, 4 packets was transmitted in 28 seconds and 8 packets was transmitted in 77 seconds. If we compare above data and VOIP values for jitter, the data values are comparatively less when compared with VOIP values. In the above figure.5.8 if we observe MPLS voice packet forwarding, 1 packet was transmitted with 1456.33 bites, 4 packets was transmitted with 120.48 bites and 8 pack-ets was transmitted with 612.47 bites for MPLS data packet forwarding 1 packet was transmitted with 7281.66

International Journal of Emerging Technology in Computer Science & Electronics (IJETCSE)
ISSN: 0976-1353 Volume 24 Issue 5 – APRIL 2017.

bites, 4 packets was transmitted with 4488.69 bites, 8 packets was transmitted with 2874.34 bites and for IP voice packet forwarding 1 packet was transmitted with 1347.44 bites, 4 packets was transmitted with 1256.52 bites and 8 packets was transmitted with 500.47 bites for IP data packet forwarding 1 packet was transmitted with 6553.5, 4 packets was transmitted with 4147.7 bites and 8 packets was transmitted with 2501.33 bites. If we compare above data and VOIP values for throughput, the VOIP values are comparatively less when compared with data values.

Ethernet: 13 | Fibre Channel | FDDI | IPv4: 11 | IPv6: 4 | IPX | JXTA | NCP | RSVP | SCTP | TCP: 79 | Token Ring | UDP: 44 | USB | WLAN

Ethernet Endpoints

Address	Packets	Bytes	Tx Packets	Tx Bytes	Rx Packets	Rx Bytes
CadmusCo_e6:3e:fb	225 762	16 791 442	225 762	16 791 442	0	0
c4:0c:05:24:00:01	226 578	16 852 909	507	47 529	226 071	16 805 380
CadmusCo_fb:c8:2f	219 533	16 315 108	194 662	14 475 438	24 871	1 839 670
c4:0d:06:cc:00:01	220 058	16 356 270	25 205	1 873 458	194 853	14 482 812
0a:00:27:00:00:00	142	15 419	142	15 419	0	0
IPv6mcast_01:00:02	59	8 791	0	0	59	8 791
Broadcast	258	25 032	0	0	258	25 032
CDP/VTP/DTP/PAgP/UDLD	101	36 764	0	0	101	36 764
DEC-MOP-Remote-Console	9	693	0	0	9	693
IPv6mcast_16	10	900	0	0	10	900
IPv4mcast_16	10	540	0	0	10	540
IPv6mcast_01:00:03	18	1 532	0	0	18	1 532
IPv4mcast_fc	18	1 172	0	0	18	1 172

Fig.8 Ethernet endpoint

Each row in the list shows the statistical values for exactly one endpoint. Name resolution will be done if selected in the window and if it is active for the specific protocol layer (MAC layer for the selected Ethernet endpoints page). Limit to display filter will only show conversations matching the current display filter. Note that in this example we have GeoIP configured which gives us extra geographic columns. See Section 10.10, "GeoIP Database Paths" for more information. The Copy button will copy the list values to the clipboard in CSV (Comma Separated Values) or YAML format. The Map button will show the endpoints mapped in your web browser. Endpoint Types lets you choose which traffic type tabs are shown. See Section 8.5, "Endpoints" above for a list of endpoint types.

IPv4 Endpoints

Address	Packets	Bytes	Tx Packets	Tx Bytes	Rx Packets	Rx Bytes	Latitude	Longitude
192.168.56.20	225 770	16 791 946	225 770	16 791 946	0	0	-	
192.168.56.10	225 746	16 785 904	0	0	225 746	16 785 904	-	
192.168.66.100	220 635	16 393 540	195 832	14 557 938	24 803	1 835 602	-	
192.168.66.30	220 125	16 357 430	24 775	1 833 642	195 350	14 523 788	-	
192.168.56.30	460	28 744	0	0	460	28 744	-	
192.168.66.255	22	5 406	0	0	22	5 406	-	
192.168.56.255	54	8 802	0	0	54	8 802	-	
192.168.56.1	58	4 472	58	4 472	0	0	-	
224.0.0.22	10	540	0	0	10	540	-	
172.16.16.14	28	1 960	28	1 960	0	0	-	
224.0.0.252	18	1 172	0	0	18	1 172	-	

IPv4 End Points

VI. CONCLUSION

In today's highly demanding world the need for a good network service is very important and challenging for the service providers to full fill all the needs of the customers in all the ways. But they have come up with this new MPLS technology that could facilitate efficiency and QoS, this paper has given the simulating test results of the three such parameters like jitter, latency and throughput of IP and MPLS under two conditions (one using data and other using voice) and comparatively MPLS has given a better performance in both the conditions which allows us to come to a conclusion that it is one of the best network technologies existing and also has also got scope for expansion of its services over the years.

VII. FUTURE SCOPE

The following project has the facility to be employed in GMPLS, with WDM, flexible grid also which is to be deployed in the fourth coming years which will enable higher degree of addressing and auto-configuration mechanism.

VIII. REFERENCES

[1] Comparative analysis of IP, ATM and MPLS with their QoS (Rameshwar T. Murade).
[2] PA-POS-OC3 Packet OC-3 Port Adapter Installation and Configuration.
[3] Evolution of Multiprotocol Label Switching (Arun Viswanathan).
[4] An efficient engineering approach based on flow distribution and splitting in MPLS network (G. Mohan).
[5] Multi-Protocol Label Switching (MPLS) Mr. Manish G.
[6] Analysis, Review and Optimization of SONET/SDH technology for today and further aspests (Gourav Varma).
[7] Difference between SONET and OBS on the basis of block diagram
[8] (Mr. Bhu-pesh Bhatia).
[9] High performance Backbone technology (Naoaki yamanaka).
[10] MPLS and Traffice Engineering in IP networks (Daniel O. Awdcuche).
[11] A Review Paper on Voice over Internet Protocol (Rahul Singh, Ritu Chauhan). K. M. McNeill, M. Liu and J. J. Rodriguez, An Adaptive Jitter Buffer PlayOut Scheme to Improve VoIP Quality in Wireless Networks, IEEE Conf. on BAE Systems Network Enabled Solutions, Washington, 2006.
[12] Comparative Analysis of Traditional Telephone and Voice-over-Internet Protocol (VoIP) Systems. Hui Min Chong and H. Scott Matthews Department of Civil and Environmental Engineering Carnegie Mellon University Pittsburgh, PA USA.
[13] Voice Over IP as a Model for Multi-Services Networking by (E. Michael Staman).
[14] P. M. Athina., A. T. Fouad and J. K. Mansour, Assessing the Quality of Voice Communications Over Internet Backbones, IEEE/ACM Transactions on Networking, Vol. 11, No. 5, Oct. 2003.
[15] IEEE Multipath routing with adaptive playback scheduling for Voice over IP in Service Overlay Networks. Sarnoff Symposium, 2008 IEEE: 1AS5¸. 28 AS¸ 30 April 2008