Comparison in the Applicability of MPLS When Using Different Dynamic Routing Protocols

Ivan Nedyalkov
Dept. of Communication and Computer Technologies, South–West University “Neofit Rilski”, Blagoevgrad, Bulgaria
Email: i.nedqjlkov@gmail.com

Abstract—The purpose of this paper is to examine how the Multiprotocol Label Switching (MPLS) technology affects the latency in an IP network when using different dynamic routing protocols. For the purposes of the study, a virtual IP network was created, in which was sequentially configured routing protocols. For the purposes of the study, a virtual IP

network was created, in which was sequentially configured to work with Routing Information Protocol (RIP), Enhanced Interior Gateway Routing Protocol (EIGRP) and Open Shortest Path First (OSPF). The studies were performed without configured and then with configured MPLS. Techniques, methods and tools used in the monitoring of IP networks were used during the study of the network.

Index Terms—IP network, Latency, MPLS, Time delay, Virtual network, VoIP

I. INTRODUCTION

Communication networks are growing and especially IP networks, as the most used now and established as a total hegemon over other communication networks. Therefore, this network must provide high transmission speeds, Quality of Service (QoS), low latency values, offering new and new services to users and more [1]-[4]. The Multiprotocol Label Switching (MPLS) technology largely meets these requirements. MPLS allows communication network operators and service providers to build next-generation smart networks. Networks using this technology provide a wide variety of new services within an infrastructure. Therefore, before introduction of the technology in a specific network, it is necessary to carry out studies for this specific network like what is the most appropriate protocol for dynamic routing; what would be the possible delay in the network, for the specific traffic that is exchanging in the network and more. The best way to conduct such a study is by creating a real experimental network. However, the creation of such a network requires significant financial investment in equipment and infrastructure - the network must be fully implemented in advance and then the study must be carried out. It is more practical to create a small-scale virtual network that will be similar to the future real network and carry out the studies in it [5]-[8].

The aim of this paper is to study a virtual IP network that will consistently use the following three dynamic routing protocols: Routing Information Protocol (RIP), Enhanced Interior Gateway Routing Protocol (EIGRP) and Open Shortest Path First (OSPF). These dynamic protocols are chosen because they are the most commonly used dynamic routing protocols in IP networks of different sizes and numbers of network devices. The IS-IS (Intermediate System to Intermediate System) dynamic routing protocol is not used because it is mainly used in the core network of IPSSs (Internet Service Providers). In the present work, routing between autonomous systems (AS) is not a part of the study. Therefore, the BGP (Border Gateway Protocol) dynamic routing protocol is not used too.

For each of the protocols, the network will be monitored. The delays in the network will be observed with and without using the MPLS technology for each of the three protocols. Only voice traffic exchanges in the virtual network. Finally, a summary of the results will be made and which of the three protocols, for the specific traffic, is the most appropriate to use with the MPLS technology will be noted.

II. RELATED WORKS

In [9] the authors presented a model of an IP network developed by them, which has the ability to work with various dynamic routing protocols. The developed model of the IP network uses the MPLS technology. The network is intended for exchanging of different types of multimedia traffic - voice and video. The purpose of the developed model is to study the load of the network elements, to observe the values of the delay, the change of the bandwidth and other parameters. The network model is created using the program Opnet Modeler v14.5.

In [10] the authors made a comparative study between two virtual IP networks, one using MPLS technology and the other not. The exchanged traffic in both networks is voice and FTP traffic. The monitored parameters in both networks are packet loss, end-to-end delay, jitter and others. The studies were performed using QoS and in the absence of QoS.

In [11] the authors made a comparative study between a network using MPLS technology and when not using the technology. The aim of the work is to monitor the performance of the network by monitoring the following parameters: jitter, packet loss, delay and others. The exchanged traffic in the virtual network is video and text. The virtual network was created using the GNS3 platform.
In [12] the authors presented an algorithm for real-time monitoring of the performance of tunnels created by MPLS TE (Traffic Engineering). Jitter, delay, packet loss and others are the monitored parameters in real-time. Through the received data in real time, the algorithm can assess the performance of the monitored network or detect problems. The application of the proposed algorithm is studied on a real network.

Additional researches close to the subject of the presented work are in [13]-[15].

III. VIRTUAL NETWORK TOPOLOGY

Fig. 1 shows the topology of the virtual network. The virtual network is created by using the GNS 3 platform [16]-[18]. This platform is used because of the many advantages it offers such as: integration with IP network monitoring programs (Wireshark), possibility to work with disk images of real operating systems of real network devices, possibility to connect to real networks and many other options. The work with real disk images of real network devices is realized by emulation. Device emulation is the imitation ( emulation) of a device's hardware. This allows the users to start and work with real images of real working network devices. Thanks to these capabilities, the implemented virtual networks in GNS 3 are identical to real ones.

IV. METHODOLOGY

The study is carried out as follows: each of the virtual machines establishes calls with each of the other virtual machines for the period of the study. The studies begin when the network is initially configured with the corresponding dynamic protocol (RIPv2, EIGRP or OSPF) without the MPLS technology to be configured. The studies are then repeated, with configured MPLS. The results from Wireshark [19]-[24], Colasoft Ping Tool and Colasoft Capsa 11 Free are mainly used. In addition, mathematical distributions for the packets arrival times are made. These distributions are used for additional evaluation, as well as for obtaining a visual idea of how the time delay changes [25], [26].

V. RESULTS WHEN USING RIP v2

A. Results without Configured MPLS

After the study and monitoring of the virtual network, it is found that the voice traffic between VM1 and Asterisk passes through R3, R2, R5, and vice versa. The voice traffic between VM2 and Asterisk goes through R2, R5 and vice versa. The voice traffic between VM3 and Asterisk goes through R4, R5 and vice versa. The voice traffic between VM4 and Asterisk goes through R1, R5 and vice versa. Only service traffic, such as RIP updates, CDP (Cisco Discovery Protocol – it is used to share information with other, directly connected Cisco devices, such as the version of the operating system and other similar information), and other service protocols, passes through the other links.

Fig. 2 shows summarized results for the voice flow that passes through R2 and R3 (from VM1 to Asterisk and vice versa). As it can be seen from the presented results for both directions there is no packet loss, the average value of the jitter is below the allowable value of 30ms, according to [27], [28]. The results for the same voice stream, which passes through the link between R2 and R5, are almost identical.
Fig. 2. Summarized results for the R2 – R3 link.

Fig. 3. Summarized results for the R4 – R5 link.

Fig. 4. Summarized results for the R1 – R5 link.

Fig. 5. Summarized results for the R1 – R5 link.

Fig. 6. Round-trip delay for the route from VM1 to the Asterisk PBX.

Fig. 7. Round-trip delay for the route from VM2 to the Asterisk PBX.

Fig. 8. Round-trip delay for the route from VM3 to the Asterisk PBX.

Fig. 9. Round-trip delay for the route from VM4 to the Asterisk PBX.

Fig. 3 shows summarized results for the voice flow that passes through R4 and R5 (from VM3 to Asterisk and vice versa). Fig. 4 shows summarized results for the voice flow that passes through R1 and R5 (from VM4 to Asterisk and vice versa). Fig. 5 shows summarized results for the voice flow that passes through R2 and R5 (from VM2 to Asterisk and vice versa). As it can be seen from the presented results, they are almost similar to the results in Fig. 2.

Fig. 6 shows the round-trip delay (RTD) in the connection between VM1 and Asterisk (R3, R2, R5 and vice versa). As it can be seen from the results, except for a few moments, the RTD varies between 20ms and 40ms or an average of 27ms. This is because R2 is adjacent to R5 and the traffic passes only between the two routers, unlike the traffic generated by VM1, which passes through three routers, resulting in an increase in RTD values.

Fig. 7 shows the RTD in the connection between VM2 and Asterisk (R2, R5 and vice versa), and Fig. 9 shows the RTD in the connection between VM4 and Asterisk (R1, R5 and vice versa). Fig. 8 shows a slight deterioration of the RTD values, but in general, the results are similar to those of Fig. 7, as well as the average RTD value of 27ms.
Fig. 10. Mathematical distribution.

Fig. 11. Total traffic by bytes.

Fig. 12. Summarized results for the R2 – R3 link.

Fig. 13. Summarized results for the R4 – R5 link.

Fig. 14. Summarized results for the R1 – R5 link.

B. Results for Configured MPLS

After studying the network, it is found that the traffic again passes through the same links as in Section V - A. Fig. 12 shows summarized results for the voice flow that passes through R2 and R3 (from VM1 to Asterisk and vice versa). As it can be seen from the results there are improvements in the values of the delta parameter (the delta shows the time difference between the receipt of the previous packet from the stream and the received now packet). There are also improvements in the maximum values of the jitter in the two directions, as well as in the other parameters directed to Fig. 2. The average value of the jitter continues at a permissible value of 30ms. The results for the same voice flow, which passes through the link between R2 and R5 are almost identical.

Fig. 13 shows summarized results for the voice flow that passes through R4 and R5 (from VM3 to Asterisk and vice versa). As it can be seen, the results are almost identical to those in Fig. 12 except for the values of the maximum jitter and the mean value of the jitter in the reversed direction. In addition, there is again an improvement in the values of the other parameters compared to the same observed parameters in Fig. 3. This is because there is no MPLS configured.

Fig. 14 shows summarized results for the voice flow that passes through R1 and R5 (from VM4 to Asterisk and vice versa). As it can be seen from the presented results, there is again a significant improvement in the values of the parameters, compared to those in Fig. 4.
Fig. 15. Summarized results for the R2–R5 link.

Fig. 16 shows the RTD in the connection between VM1 and Asterisk (R3, R2, R5 and vice versa). Fig. 17 shows the RTD in the connection between VM2 and Asterisk (R2, R5 and vice versa). Fig. 18 shows the RTD in the connection between VM3 and Asterisk (R4, R5 and vice versa). Fig. 19 shows the RTD in the connection between VM4 and Asterisk (R1, R5 and vice versa). As it can be seen from the results, except for a few moments where the RTD is high, no significant difference is observed with the results from Section V-A. The averaged RTD values remain the same.

Fig. 20 shows the mathematical distribution of arrival times between packets for the link between R2 and R5. Again, the results for the other links are the same as the results for this link and therefore they are not presented.

As it can be seen from the distribution, the delay between the individual packets remains constant and does not change for the duration of the study.

C. Analysis of the Results When Using RIP

The obtained results of the study when using RIP are as follows: the use of MPLS technology, together with the RIP v2 protocol, leads to significant improvements in voice flow parameters. The use of the MPLS technology does not significantly improve the RTD in the virtual network. It is almost constantly. This is due to the topology of the virtual network, because there is no clearly defined MPLS core network.

VI. RESULTS WHEN USING EIGRP

A. Results without Configured MPLS

After the carried out study and monitoring of the virtual network with configured EIGRP, the following results came up: the voice traffic passes through the same devices and links as in Sections V–A and V–B.

Fig. 21 shows summarized data for the voice flow that passes through R2 and R3 (from VM1 to Asterisk and vice versa). As it can be seen from the results, the values of the parameters are much better than the results for RIP without MPLS and RIP with MPLS. This improvement is due to the use of EIGRP and its working principle. There is only one lost packet in the forward direction (from VM1 to Asterisk). The results for the same voice stream, which passes through the link between R2 and R5 are almost identical.
Fig. 21. Summarized results for the R2–R3 link.

Fig. 22. Summarized results for the voice flow that passes through R4 and R5 (from VM3 to Asterisk and vice versa). The results are similar to those in Fig. 21. Again, there is an improvement in the values of the parameters compared to the parameters for the same connection in RIP without MPLS and RIP with MPLS.

Fig. 23. Summarized results for the voice flow that passes through R1 and R5 (from VM4 to Asterisk and vice versa). There are significant improvements in the values of the parameters.
Fig. 29. Mathematical distribution.

Fig. 29 presents the mathematical distribution of arrival times between packets for the link between R2 and R5. The dependence is the same as in section V.

B. Results with Configured MPLS

Again, the voice flow flows through the same network devices and links as in Section VI – A.

Fig. 30 presents the summarized results for the voice flow that passes through R2 and R3 (from VM1 to Asterisk and vice versa). As it can be seen from the results, the use of the MPLS leads to a significant improvement in the parameter values. The combination of EIGRP and MPLS further improves the values of the voice flow parameters, in contrast to the combination of RIP and MPLS. Again, the results for the same voice stream, which passes through the link between R2 and R5, are almost identical to the results from Fig. 30 and therefore are not presented.

Fig. 31 shows the summarized results for the voice flow that passes through R4 and R5 (from VM3 to Asterisk and vice versa). The results are similar to those in Fig. 30. Again, there is a significant improvement in the parameter values caused by the use of MPLS.

Fig. 32 shows the summarized results for the voice flow that passes through R1 and R5 (from VM4 to Asterisk and vice versa). The increased value of delta in both directions is only one time. This is evident from Fig. 33. As it can be seen from it, the instantaneous values of delta in both directions are between 60ms and 70ms. The same applies to the values of the jitter; the maximum value shown in both directions is only one time. The instantaneous values of the jitter in forward and reverse direction do not exceed 15ms-16ms, as shown in Fig. 34.

Fig. 35 shows the summarized results for the voice flow that passes through R2 and R5 (from VM2 to Asterisk and vice versa). The results are almost identical to those in Fig. 24. An analysis similar to that for the link between R1 and R5 shows that there was again a significant improvement in voice flow parameters when activating MPLS - the results were similar to those in Fig. 33 and Fig. 34, even better.
C. Analysis of the Results when Using EIGRP

When using EIGRP, there are improvements in voice flow parameters. The use of MPLS technology together with EIGRP further improves these parameters, despite the small size of the virtual network - only a few routers. Mathematical distributions and the graphs for the RTD shows that the delay is still constant despite of using EIGRP with MPLS compared to RIP with MPLS. Again, this is due to the topology of the virtual network, because there is no clearly defined MPLS core network.

VII. RESULTS WHEN USING OSPF

A. Results without Configured MPLS

Again, the voice traffic flows through the same network devices and links as in the previous Sections VI – A and VI – B.

Fig. 41 shows the summarized data for the voice flow that passes through R2 and R3 (from VM1 to Asterisk and vice versa). When using OSPF, there is an additional improvement in the observed parameters - most notably in the delta parameter. The average jitter values remain constant (similar to those in RIP and EIGRP). Once again, the results for the same voice stream, which passes through the link between R2 and R5, are almost identical to the results from Fig. 41 and therefore are not presented.

Fig. 41. Summarized results for the R2 – R3 link.
Fig. 42 shows the summarized results for the voice flow that passes through R4 and R5 (from VM3 to Asterisk and vice versa). Here, too, there is an improvement in delta values over RIP and EIGRP. The maximum values of the jitter continue to vary in the range of 15ms to 20ms. The average jitter value also remains almost constant.

Fig. 43 shows summarized results for the voice flow that passes through R1 and R5 (from VM2 to Asterisk and vice versa). The results are almost identical to those in Fig. 32. An analysis similar to that for the link between R1 and R5 shows that there was again a significant improvement in voice flow parameters—the results were similar to those in Fig. 33 and Fig. 34.

Fig. 44 shows summarized results for the voice flow that passes through R2 and R5 (from VM2 to Asterisk and vice versa). There is a slight deterioration due to a momentary value of the delta. A more detailed analysis of the data for the studied voice flow revealed that the actual instantaneous values of delta are about 40ms (similar to the analysis of the results for Fig. 32).

**Fig. 42.** Summarized results for the R4 – R5 link.

<table>
<thead>
<tr>
<th>Forward</th>
<th>Reverse</th>
</tr>
</thead>
<tbody>
<tr>
<td>192.168.5.4:14960</td>
<td>192.168.4.3:14960</td>
</tr>
<tr>
<td>SSRC 0x8080808</td>
<td>SSRC 0x8080808</td>
</tr>
<tr>
<td>Max Delta 66.01 ms @ 149677</td>
<td>Max Delta 66.01 ms @ 149677</td>
</tr>
<tr>
<td>Max Jitter 26.04 ms</td>
<td>Max Jitter 26.04 ms</td>
</tr>
<tr>
<td>Mean Jitter 7.90 ms</td>
<td>Mean Jitter 7.90 ms</td>
</tr>
<tr>
<td>Max Slow 238.61 ms</td>
<td>Max Slow 238.61 ms</td>
</tr>
<tr>
<td>RTP Packets 12915</td>
<td>RTP Packets 12915</td>
</tr>
<tr>
<td>Expected 91919</td>
<td>Expected 91919</td>
</tr>
<tr>
<td>Lost 0 (0.00 %)</td>
<td>Lost 0 (0.00 %)</td>
</tr>
<tr>
<td>Seq Errs 0</td>
<td>Seq Errs 0</td>
</tr>
<tr>
<td>Start at 1304302030 @ 1304302030</td>
<td>Start at 1304302030 @ 1304302030</td>
</tr>
<tr>
<td>Duration 2020.09 s</td>
<td>Duration 2020.09 s</td>
</tr>
<tr>
<td>Clock Drift 24 ms</td>
<td>Clock Drift 24 ms</td>
</tr>
<tr>
<td>Freq Drift 8000 Hz (0.00 %)</td>
<td>Freq Drift 8000 Hz (0.00 %)</td>
</tr>
</tbody>
</table>

**Fig. 43.** Summarized results for the R1 – R5 link.

**Fig. 44.** Summarized results for the R2 – R5 link.

Fig. 45 shows the delay in the connection between VM1 and Asterisk (R3, R2, R5 and vice versa). Fig. 46 shows the RTD in the connection between VM2 and Asterisk (R2, R5 and vice versa). Fig. 47 shows the RTD in the connection between VM3 and Asterisk (R4, R5 and vice versa). Fig. 48 shows the RTD in the connection between VM4 and Asterisk (R1, R5 and vice versa). As it can be seen from the obtained results for the RTD, they are similar to the results presented so far in Sections V and VI—the dependence is still the same, regardless of the routing protocol. The average values of the RTD are again the same.

**Fig. 45.** Round-trip delay for the route from VM1 to the Asterisk PBX.

**Fig. 46.** Round-trip delay for the route from VM2 to the Asterisk PBX.

**Fig. 47.** Round-trip delay for the route from VM3 to the Asterisk PBX.

**Fig. 48.** Round-trip delay for the route from VM4 to the Asterisk PBX.
Fig. 49 presents the mathematical distribution of arrival times between packets for the link between R2 and R5. There is a slight change in the form here, but regardless of this the trend continues - a constant time delay.

B. Results with Configured MPLS

Here again the voice traffic passes through the same network devices and links as Section VII – A.

Fig. 50 shows the summarized data for the voice stream that passes through R2 and R3 (from VM1 to Asterisk and vice versa). Fig. 51 shows the summarized data for the voice stream that passes through R4 and R5 (from VM3 to Asterisk and vice versa). Fig. 52 shows the summarized data for the voice stream that passes through R1 and R5 (from VM4 to Asterisk and vice versa). Fig. 53 shows the summarized data for the voice stream that passes through R1 and R5 (from VM2 to Asterisk and vice versa). On-depth analysis, through the Wireshark functionality for voice streams analysis (graphical representation of the change of the parameter values for each second of the entire call period), shows that the higher values of the delta parameter are only one time. The instantaneous values are close to or lower than those in Fig. 32. The same applies to the values of the jitter.

Fig. 54 shows the RTD in the connection between VM1 and Asterisk (R3, R2, R5 and vice versa). Fig. 55 shows the RTD in the connection between VM2 and Asterisk (R2, R5 and vice versa). Fig. 56 shows the RTD in the connection between VM3 and Asterisk (R4, R5 and vice versa). Fig. 57 shows the RTD in the connection between VM4 and Asterisk (R1, R5 and vice versa). Excluding high one-time RTD values, the results are similar to those in Section VII – A. The trend of changing the RTD from Section V and Section VI is the same here as well. The average values are again the same as before.

Fig. 58 shows the RTD in the connection between VM1 and VM2.
As expected, when we use RIP, the parameters of the voice flow are the worst, which is due to its principle of operation. With the activation of the MPLS technology, the values of the voice flow parameters significantly are improved. The time delay values remain almost the same whether MPLS is configured and not.

The use of EIGRP without configured MPLS leads to a further improvement of the voice flow parameters compared to RIP v2. Enabling MPLS further improves the voice flow parameters. The network delay remains almost the same as with RIP v2.

The use of OSPF improves the parameters of the voice flow even more, but the activation of the MPLS technology does not lead to further improvements of the monitored parameters.

Some of the obtained results coincide with the results obtained by other researchers - when using OSPF with MPLS, the IP network is additionally loaded. As a result, there is no improvements in network performance.

Regardless of the use of MPLS technology, the RTD values remain almost constant. This is due to the choice of the topology of the studied network - there is no clearly defined MPLS core network. All routers are both ingress (puts the label in front of the IP packet) and egress (removes the label from the IP packet). As a result, the full capabilities of the MPLS technology are not used. This is the disadvantage of the used topology.

In future works, the topology will be different - to have a clearly defined MPLS core network. Additionally, QoS will be configured.

Despite the small size of the virtual network (only five routers) - the activation of MPLS technology for RIP and EIGRP helps to improve the parameters of voice flow.

Mathematical distributions show that the delays in the created virtual network are constant.

In summary, in real networks similar in size to the studied virtual network, the most suitable dynamic protocol for working with MPLS is EIGRP.

CONFLICT OF INTEREST

The author declares no conflict of interest.

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Ivan Iv. Nedyalkov was born in the town of Berkovitsa, Bulgaria on 02.11.1986. He completed his Ph.D. degree in Technical University – Sofia, Bulgaria in October 2016. The field of the study was in power electronics and energy storage.

From August 2008 - September 2012 he worked as an Installer of security systems and CCTV, technical support for “SOT 161” in Sofia, Bulgaria. From September 2013 – July 2013 he worked as a part-time lecturer at the Technical University – Sofia, Bulgaria. From September 2012 – September 2019 he worked as assistant prof.t at the University of Telecommunications and Post, Sofia, Bulgaria. From September 2018 till now he has been working at the South – West University “Neofit Rilski”, Blagoevgrad, Bulgaria – firstly as a chief assistant and from December 2020 as an associate professor, His field of research covers: power converters for supercapacitor charging, simultaneously charging and voltage equalization over series connected supercapacitor cells, monitoring of IP - based networks, characterization of the traffic flows in IP networks. Dr. Nedyalkov is a member of Bulgarian Union of Electronic, Electrical Engineering and Telecommunications (CEEC), CEEC is a member of the Federation of the Scientific-Technical Unions in Bulgaria (FNTS), which is itself a permanent member of the European Federation of Engineering National Associations (FEANI) and CEEC is also a member of the Convention of National Societies of Electrical Engineers in Europe (EUREL).