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

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***Abstract*—The purpose of this paper is to examine how the Multiprotocol Label Switching (MPLS) technology affects to 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 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 IPSs (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.

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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.

VM1 to VM4 are virtual machines. The virtual machines are subscribers to the IP telephone exchange - Asterisk. In the virtual network, only voice traffic exchanges between these subscribers.

R1 to R5 are routers that are disk image emulations of real routers.

S1 to S5 are switches, more precisely these are simulation models of switches.

There are no additional configurations for QoS, load balancing of the traffic or route prioritization in the virtual network, because the network under study is small and does not require the application of such settings.

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 begins 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].

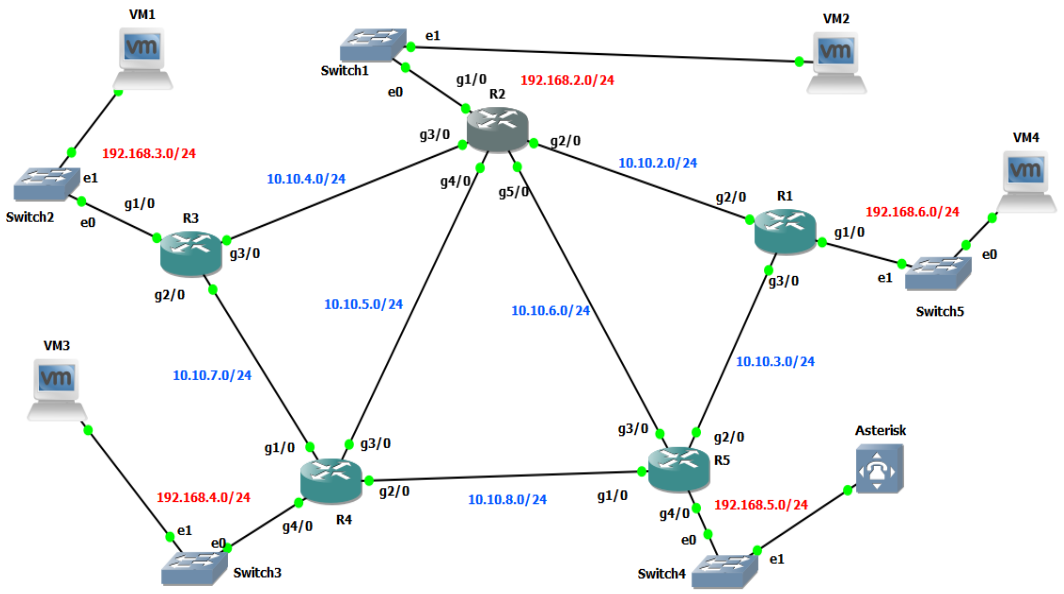

Fig. 1. Topology of the virtual network.

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.

Forward		**Reverse**	
192.168.3.2:8000 → 192.168.5.4:12944		192.168.5.4:12944 → 192.168.3.2:8000	
SSRC	0xbf6c73fa	**SSRC**	0x74b09d86
Max Delta	358.54 ms @ 114286	**Max Delta**	400.55 ms @ 114304
Max Jitter	28.59 ms	**Max Jitter**	31.38 ms
Mean Jitter	8.01 ms	**Mean Jitter**	7.73 ms
Max Skew	-703.97 ms	**Max Skew**	-597.47 ms
RTP Packets	107961	**RTP Packets**	107980
Expected	107961	**Expected**	107980
Lost	0 (0.00 %)	**Lost**	0 (0.00 %)
Seq Errs	0	**Seq Errs**	0
Start at	587.762919 s @ 57344	**Start at**	587.781422 s @ 57346
Duration	2159.53 s	**Duration**	2159.63 s
Clock Drift	31 ms	**Clock Drift**	224 ms
Freq Drift	8000 Hz (0.00 %)	**Freq Drift**	8001 Hz (0.01 %)

Fig. 2. Summarized results for the R2 – R3 link.

Forward		**Reverse**	
192.168.4.2:8000 → 192.168.5.4:14224		192.168.5.4:14224 → 192.168.4.2:8000	
SSRC	0x9c38a0c6	**SSRC**	0x265bc826
Max Delta	375.02 ms @ 327110	**Max Delta**	380.55 ms @ 313386
Max Jitter	30.20 ms	**Max Jitter**	29.01 ms
Mean Jitter	8.11 ms	**Mean Jitter**	8.19 ms
Max Skew	-854.93 ms	**Max Skew**	-933.83 ms
RTP Packets	116301	**RTP Packets**	116286
Expected	116301	**Expected**	116286
Lost	0 (0.00 %)	**Lost**	0 (0.00 %)
Seq Errs	0	**Seq Errs**	0
Start at	2803.977003 s @ 284863	**Start at**	2804.376054 s @ 284884
Duration	2326.79 s	**Duration**	2326.24 s
Clock Drift	-505 ms	**Clock Drift**	-134 ms
Freq Drift	7998 Hz (-0.02 %)	**Freq Drift**	8000 Hz (-0.01 %)

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

Forward		**Reverse**	
192.168.6.3:8000 → 192.168.5.4:18632		192.168.5.4:18632 → 192.168.6.3:8000	
SSRC	0x8ecc217a	**SSRC**	0x5e927ca4
Max Delta	351.55 ms @ 312305	**Max Delta**	390.02 ms @ 336457
Max Jitter	31.25 ms	**Max Jitter**	31.86 ms
Mean Jitter	8.18 ms	**Mean Jitter**	4.41 ms
Max Skew	-679.29 ms	**Max Skew**	-658.92 ms
RTP Packets	116232	**RTP Packets**	116236
Expected	116232	**Expected**	116236
Lost	0 (0.00 %)	**Lost**	0 (0.00 %)
Seq Errs	0	**Seq Errs**	0
Start at	2840.114833 s @ 285373	**Start at**	2840.123834 s @ 285375
Duration	2324.79 s	**Duration**	2324.93 s
Clock Drift	140 ms	**Clock Drift**	58 ms
Freq Drift	8000 Hz (0.01 %)	**Freq Drift**	8000 Hz (0.00 %)

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

Forward		**Reverse**	
192.168.2.3:8000 → 192.168.5.4:15884		192.168.5.4:15884 → 192.168.2.3:8000	
SSRC	0x26c814ae	**SSRC**	0x084a8dbf
Max Delta	390.02 ms @ 668435	**Max Delta**	381.05 ms @ 620264
Max Jitter	31.58 ms	**Max Jitter**	31.80 ms
Mean Jitter	4.30 ms	**Mean Jitter**	8.25 ms
Max Skew	-775.93 ms	**Max Skew**	-619.78 ms
RTP Packets	116239	**RTP Packets**	116231
Expected	116239	**Expected**	116231
Lost	0 (0.00 %)	**Lost**	0 (0.00 %)
Seq Errs	0	**Seq Errs**	0
Start at	2827.090686 s @ 566485	**Start at**	2827.317714 s @ 566515
Duration	2325.08 s	**Duration**	2324.71 s
Clock Drift	57 ms	**Clock Drift**	140 ms
Freq Drift	8000 Hz (0.00 %)	**Freq Drift**	8000 Hz (0.01 %)

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

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). Horizontal axis stands for the time (hour, minute and seconds) at which the RTD is measured, and vertical axis stands for the value of the RTD in ms. As it can be seen from the results, except for a few moments where the RTD is very large, it varies between 30ms and 60ms or an average of 37ms.

Fig. 7 shows the RTD in the connection between VM2 and Asterisk (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. 8 shows the RTD in the connection between VM3 and Asterisk (R4, 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.

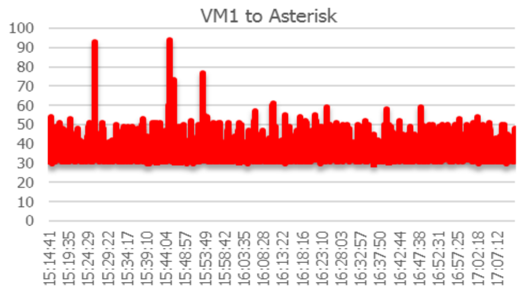

Fig. 6. Round-trip delay for the route from VM1 to the Asterisk PBX.

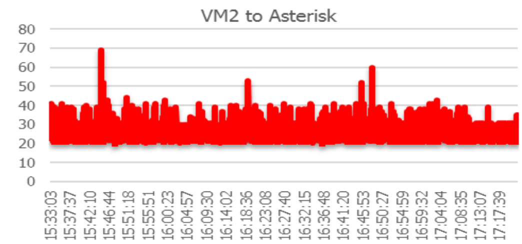

Fig. 7. Round-trip delay for the route from VM2 to the Asterisk PBX.

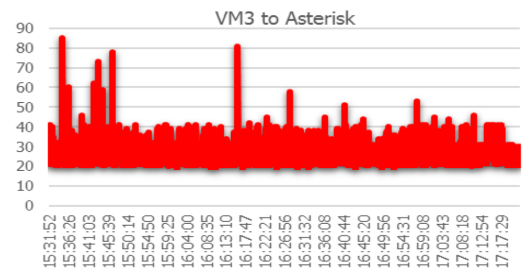

Fig. 8. Round-trip delay for the route from VM3 to the Asterisk PBX.

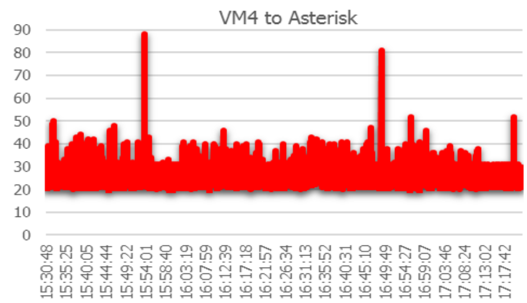

Fig. 9. Round-trip delay for the route from VM4 to the Asterisk PBX.

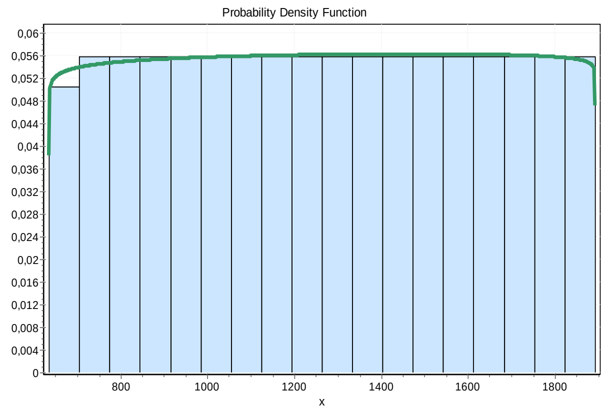

Fig. 10. Mathematical distribution.

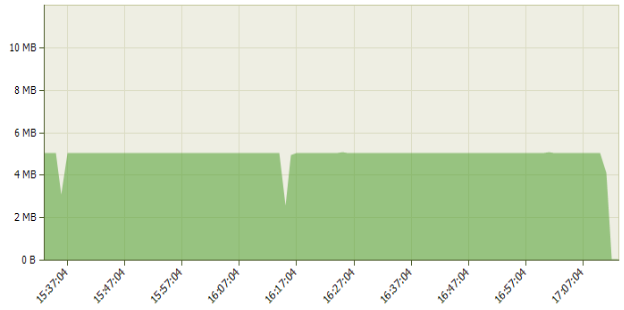

Fig. 11. Total traffic by bytes.

Fig. 10 presents the mathematical distribution for the arrival times between packets. This distribution is for the link between R2 and R5. Horizontal axis *X* stands for the times of arrival of the individual packets for the whole period of the captured VoIP stream and vertical axis *f*(*x*) stands for the delay of the received packet compared to the previous packet. The results for the other links are similar to these results and therefore are not presented. It was chosen to present this link because the traffic from two virtual machines - VM1 and VM2 - passes through it. This will give a better idea of how the delay changes to the most loaded link. As it can be seen from the distribution, the delay between the individual packets is constant, i.e. no change for the duration of the study for RIP protocol.

Fig. 11 shows the value of the generated traffic measured at the input of the Asterisk. As it can be seen from the results, the value of traffic is constant –5MB/s. The crashes in traffic are due to the moments when the virtual machines break up the already established connections and start new call setups with other subscribers. For the studies of the other protocols, the total generated traffic remains the same, i.e. 5MB/s. Therefore, the other results will not be presented.

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.

Forward	192.168.3.2:8000 → 192.168.5.4:18790	**Reverse**	192.168.5.4:18790 → 192.168.3.2:8000
SSRC	0x9570f461	**SSRC**	0x1bfbe9e7
Max Delta	289.54 ms @ 96230	**Max Delta**	326.54 ms @ 96247
Max Jitter	26.19 ms	**Max Jitter**	27.89 ms
Mean Jitter	7.96 ms	**Mean Jitter**	8.25 ms
Max Skew	-652.08 ms	**Max Skew**	-538.58 ms
RTP Packets	91086	**RTP Packets**	91099
Expected	91086	**Expected**	91099
Lost	0 (0.00 %)	**Lost**	0 (0.00 %)
Seq Errs	0	**Seq Errs**	0
Start at	253.982000 s @ 24255	**Start at**	254.008003 s @ 24257
Duration	1822.05 s	**Duration**	1822.16 s
Clock Drift	13 ms	**Clock Drift**	26 ms
Freq Drift	8000 Hz (0.00 %)	**Freq Drift**	8000 Hz (0.00 %)

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

Forward	192.168.4.3:8000 → 192.168.5.4:10084	**Reverse**	192.168.5.4:10084 → 192.168.4.3:8000
SSRC	0x0e5e88f2	**SSRC**	0x3fbf0510
Max Delta	280.54 ms @ 98215	**Max Delta**	327.54 ms @ 98229
Max Jitter	21.57 ms	**Max Jitter**	22.79 ms
Mean Jitter	8.09 ms	**Mean Jitter**	4.23 ms
Max Skew	-586.72 ms	**Max Skew**	-594.67 ms
RTP Packets	91199	**RTP Packets**	91174
Expected	91199	**Expected**	91174
Lost	0 (0.00 %)	**Lost**	0 (0.00 %)
Seq Errs	0	**Seq Errs**	0
Start at	295.318213 s @ 26930	**Start at**	295.775772 s @ 26954
Duration	1824.25 s	**Duration**	1823.74 s
Clock Drift	13 ms	**Clock Drift**	14 ms
Freq Drift	8000 Hz (0.00 %)	**Freq Drift**	8000 Hz (0.00 %)

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

Forward	192.168.6.3:8000 → 192.168.5.4:19464	**Reverse**	192.168.5.4:19464 → 192.168.6.3:8000
SSRC	0x4b034141	**SSRC**	0x47fdc375
Max Delta	299.04 ms @ 100856	**Max Delta**	339.04 ms @ 100875
Max Jitter	25.12 ms	**Max Jitter**	29.20 ms
Mean Jitter	8.12 ms	**Mean Jitter**	8.07 ms
Max Skew	-584.59 ms	**Max Skew**	-405.04 ms
RTP Packets	91103	**RTP Packets**	91085
Expected	91103	**Expected**	91085
Lost	0 (0.00 %)	**Lost**	0 (0.00 %)
Seq Errs	0	**Seq Errs**	0
Start at	303.219914 s @ 29071	**Start at**	303.615465 s @ 29092
Duration	1822.30 s	**Duration**	1821.77 s
Clock Drift	27 ms	**Clock Drift**	13 ms
Freq Drift	8000 Hz (0.00 %)	**Freq Drift**	8000 Hz (0.00 %)

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

	Forward	Reverse
	192.168.2.3:8000 → 192.168.5.4:16934	192.168.5.4:16934 → 192.168.2.3:8000
SSRC	0x9779fb1a	0x554b0373
Max Delta	315.04 ms @ 197682	312.54 ms @ 197686
Max Jitter	37.28 ms	24.36 ms
Mean Jitter	4.03 ms	8.18 ms
Max Skew	-934.21 ms	-539.21 ms
RTP Packets	91175	91196
Expected	91175	91196
Lost	0 (0.00 %)	0 (0.00 %)
Seq Errs	0	0
Start at	304.588750 s @ 57138	304.594251 s @ 57139
Duration	1824.09 s	1824.13 s
Clock Drift	13 ms	12 ms
Freq Drift	8000 Hz (0.00 %)	8000 Hz (0.00 %)

Fig. 15. Summarized results for the R2 – R5 link.

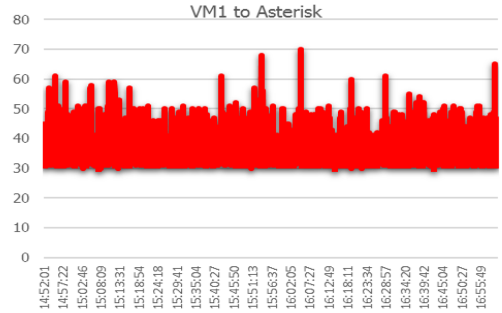

Fig. 16. Round-trip delay for the route from VM1 to the Asterisk PBX.

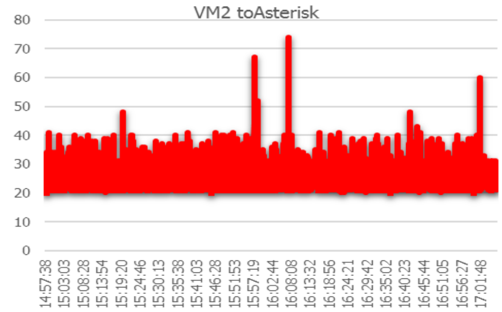

Fig. 17. Round-trip delay for the route from VM2 to the Asterisk PBX.

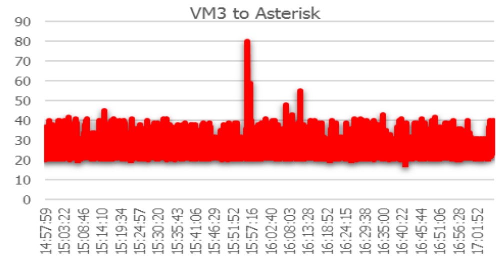

Fig. 18. Round-trip delay for the route from VM3 to the Asterisk PBX.

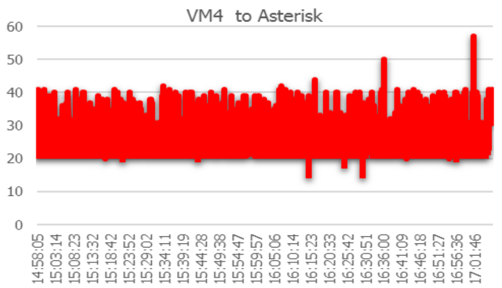

Fig. 19. Round-trip delay for the route from VM4 to the Asterisk PBX.

Fig. 15 shows summarized results for the voice flow that passes through R2 and R5 (from VM2 to Asterisk and vice versa). Again, there is an improvement in the parameters compared to those in Fig. 5, except for the maximum value of the jitter, which in the forward direction is slightly inflated.

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.

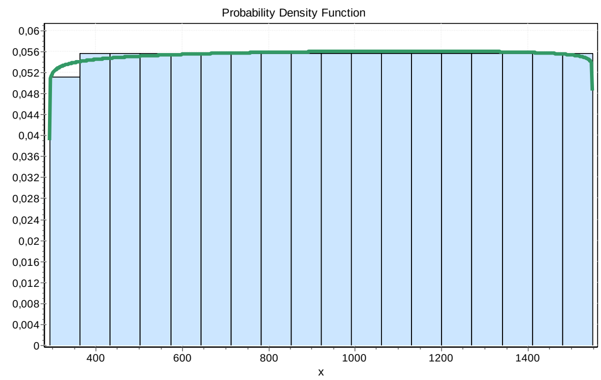

Fig. 20. Mathematical distribution.

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.

Forward		Reverse	
192.168.3.2:8000 → 192.168.5.4:11336		192.168.5.4:11336 → 192.168.3.2:8000	
SSRC	0xa0c02076	SSRC	0x525795d8
Max Delta	192.01 ms @ 259106	Max Delta	210.01 ms @ 259118
Max Jitter	21.01 ms	Max Jitter	20.06 ms
Mean Jitter	8.15 ms	Mean Jitter	8.29 ms
Max Skew	-436.41 ms	Max Skew	-652.31 ms
RTP Packets	94347	RTP Packets	94318
Expected	94348	Expected	94318
Lost	1 (0.00 %)	Lost	0 (0.00 %)
Seq Errs	1	Seq Errs	0
Start at	1965.373216 s @ 198787	Start at	1965.709759 s @ 198806
Duration	1887.15 s	Duration	1886.92 s
Clock Drift	29 ms	Clock Drift	-682 ms
Freq Drift	8000 Hz (0.00 %)	Freq Drift	7997 Hz (-0.04 %)

Fig. 21. Summarized results for the R2 – R3 link.

Forward		Reverse	
192.168.4.3:8000 → 192.168.5.4:19032		192.168.5.4:19032 → 192.168.4.3:8000	
SSRC	0xc3d02539	SSRC	0x6a054488
Max Delta	220.01 ms @ 260542	Max Delta	242.01 ms @ 260554
Max Jitter	19.72 ms	Max Jitter	23.06 ms
Mean Jitter	8.15 ms	Mean Jitter	8.31 ms
Max Skew	-874.20 ms	Max Skew	-413.90 ms
RTP Packets	94322	RTP Packets	94346
Expected	94323	Expected	94346
Lost	1 (0.00 %)	Lost	0 (0.00 %)
Seq Errs	1	Seq Errs	0
Start at	1995.764807 s @ 200689	Start at	1995.770308 s @ 200691
Duration	1887.24 s	Duration	1887.11 s
Clock Drift	-681 ms	Clock Drift	29 ms
Freq Drift	7997 Hz (-0.04 %)	Freq Drift	8000 Hz (0.00 %)

Fig. 22. Summarized results for the R4 – R5 link.

Forward		Reverse	
192.168.6.3:8000 → 192.168.5.4:17612		192.168.5.4:17612 → 192.168.6.3:8000	
SSRC	0x7c956b63	SSRC	0x07a7655f
Max Delta	180.02 ms @ 391705	Max Delta	150.02 ms @ 391720
Max Jitter	15.67 ms	Max Jitter	16.68 ms
Mean Jitter	8.03 ms	Mean Jitter	8.10 ms
Max Skew	-379.94 ms	Max Skew	-329.92 ms
RTP Packets	90279	RTP Packets	90286
Expected	90279	Expected	90286
Lost	0 (0.00 %)	Lost	0 (0.00 %)
Seq Errs	0	Seq Errs	0
Start at	3970.338512 s @ 391691	Start at	3970.379017 s @ 391693
Duration	1805.84 s	Duration	1805.95 s
Clock Drift	24 ms	Clock Drift	15 ms
Freq Drift	8000 Hz (0.00 %)	Freq Drift	8000 Hz (0.00 %)

Fig. 23. Summarized results for the R1 – R5 link.

Fig. 22 shows 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 shows 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.

Forward		Reverse	
192.168.2.3:8000 → 192.168.5.4:18138		192.168.5.4:18138 → 192.168.2.3:8000	
SSRC	0x15d60c7b	SSRC	0x02a8d408
Max Delta	230.01 ms @ 521482	Max Delta	152.01 ms @ 547828
Max Jitter	20.71 ms	Max Jitter	17.24 ms
Mean Jitter	4.13 ms	Mean Jitter	8.12 ms
Max Skew	-691.49 ms	Max Skew	-819.31 ms
RTP Packets	94920	RTP Packets	94889
Expected	94920	Expected	94889
Lost	0 (0.00 %)	Lost	0 (0.00 %)
Seq Errs	0	Seq Errs	0
Start at	1994.329615 s @ 401773	Start at	1994.518639 s @ 401778
Duration	1898.83 s	Duration	1898.50 s
Clock Drift	29 ms	Clock Drift	-776 ms
Freq Drift	8000 Hz (0.00 %)	Freq Drift	7997 Hz (-0.04 %)

Fig. 24. Summarized results for the R1 – R5 link.

Fig. 24 shows summarized results for the voice flow that passes through R2 and R5 (from VM2 to Asterisk and vice versa). There is significant improvement in the values of the parameters compared to Fig. 21 and Fig. 22.

Fig. 25 shows the RTD in the connection between VM1 and Asterisk (R3, R2, R5 and vice versa). Fig. 26 shows the RTD in the connection between VM2 and Asterisk (R2, R5 and vice versa). Fig. 27 shows the RTD in the connection between VM3 and Asterisk (R4, R5 and vice versa). Fig. 28 shows the RTD in the connection between VM4 and Asterisk (R1, R5 and vice versa). The RTD continues to maintain the pattern of change in Section V despite slight improvements in instantaneous values. The average values remain the same as in Section V.

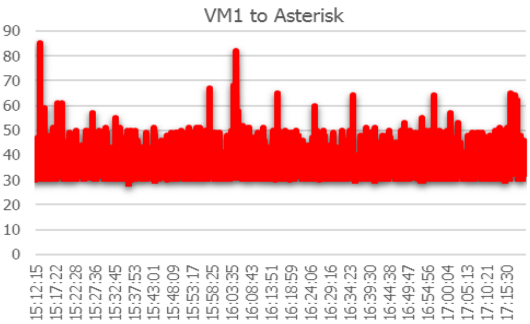

Fig. 25. Round-trip delay for the route from VM1 to the Asterisk PBX.

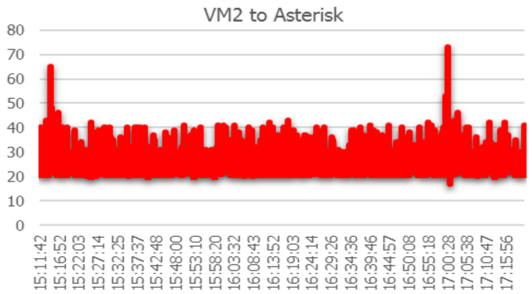

Fig. 26. Round-trip delay for the route from VM2 to the Asterisk PBX.

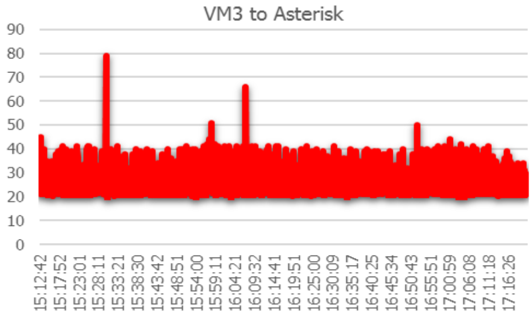

Fig. 27. Round-trip delay for the route from VM3 to the Asterisk PBX.

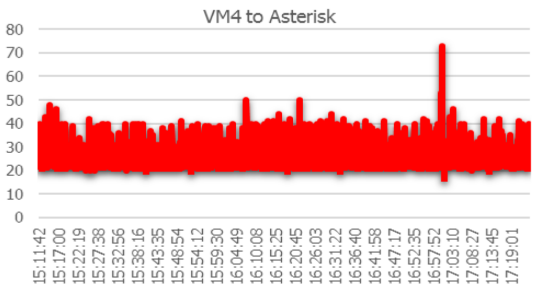

Fig. 28. Round-trip delay for the route from VM4 to the Asterisk PBX.

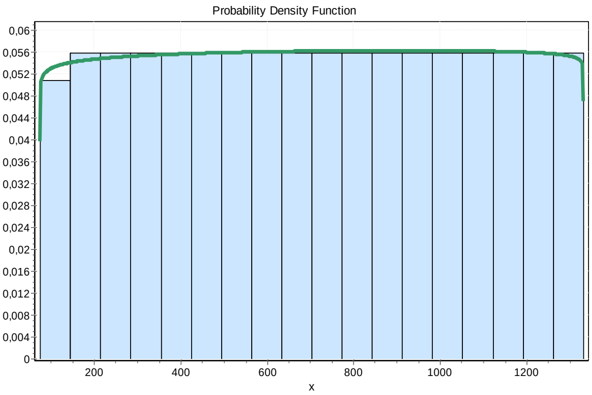

Fig. 29. Mathematical distribution.

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.

Forward		Reverse	
192.168.3.2:8000 → 192.168.5.4:14958		192.168.5.4:14958 → 192.168.3.2:8000	
SSRC	0xa2c4d440	**SSRC**	0x6bf16dab
Max Delta	140.02 ms @ 257767	**Max Delta**	130.02 ms @ 260291
Max Jitter	15.36 ms	**Max Jitter**	15.81 ms
Mean Jitter	8.24 ms	**Mean Jitter**	8.08 ms
Max Skew	-338.38 ms	**Max Skew**	-222.79 ms
RTP Packets	108961	**RTP Packets**	108946
Expected	108961	**Expected**	108946
Lost	0 (0.00 %)	**Lost**	0 (0.00 %)
Seq Errs	0	**Seq Errs**	0
Start at	2506.852338 s @ 257732	**Start at**	2507.202882 s @ 257752
Duration	2179.45 s	**Duration**	2178.99 s
Clock Drift	14 ms	**Clock Drift**	14 ms
Freq Drift	8000 Hz (0.00 %)	**Freq Drift**	8000 Hz (0.00 %)

Fig. 30. Summarized results for the R2 – R3 link.

Forward		Reverse	
192.168.4.3:8000 → 192.168.5.4:13474		192.168.5.4:13474 → 192.168.4.3:8000	
SSRC	0x98eb967c	**SSRC**	0x5680b498
Max Delta	125.52 ms @ 262976	**Max Delta**	140.02 ms @ 260448
Max Jitter	15.59 ms	**Max Jitter**	16.81 ms
Mean Jitter	7.96 ms	**Mean Jitter**	8.38 ms
Max Skew	-430.31 ms	**Max Skew**	-342.81 ms
RTP Packets	108947	**RTP Packets**	108955
Expected	108947	**Expected**	108955
Lost	0 (0.00 %)	**Lost**	0 (0.00 %)
Seq Errs	0	**Seq Errs**	0
Start at	2533.747871 s @ 260414	**Start at**	2533.755372 s @ 260415
Duration	2179.21 s	**Duration**	2179.30 s
Clock Drift	14 ms	**Clock Drift**	13 ms
Freq Drift	8000 Hz (0.00 %)	**Freq Drift**	8000 Hz (0.00 %)

Fig. 31. Summarized results for the R4 – R5 link.

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.

Forward		Reverse	
192.168.6.3:8000 → 192.168.5.4:17802		192.168.5.4:17802 → 192.168.6.3:8000	
SSRC	0x16e00bef	**SSRC**	0x7b08924d
Max Delta	337.02 ms @ 620305	**Max Delta**	302.04 ms @ 619687
Max Jitter	27.45 ms	**Max Jitter**	27.78 ms
Mean Jitter	8.02 ms	**Mean Jitter**	8.20 ms
Max Skew	-668.94 ms	**Max Skew**	-489.44 ms
RTP Packets	182976	**RTP Packets**	182979
Expected	182976	**Expected**	182979
Lost	0 (0.00 %)	**Lost**	0 (0.00 %)
Seq Errs	0	**Seq Errs**	0
Start at	4761.403847 s @ 491553	**Start at**	4761.400347 s @ 491552
Duration	3659.80 s	**Duration**	3659.76 s
Clock Drift	66 ms	**Clock Drift**	36 ms
Freq Drift	8000 Hz (0.00 %)	**Freq Drift**	8000 Hz (0.00 %)

Fig. 32. Summarized results for the R1 – R5 link.

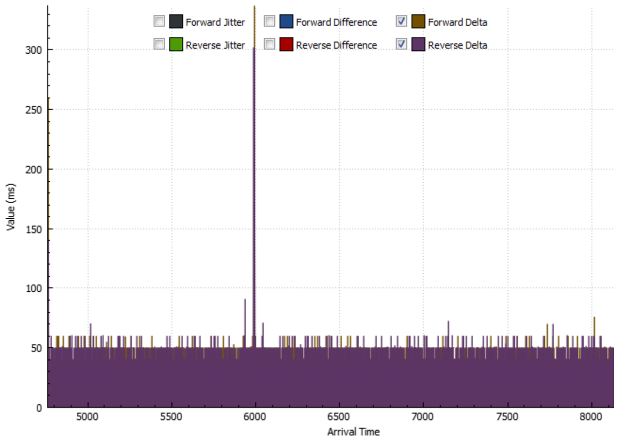

Fig. 33. Forward and reversed delta for the R1 – R5 link.

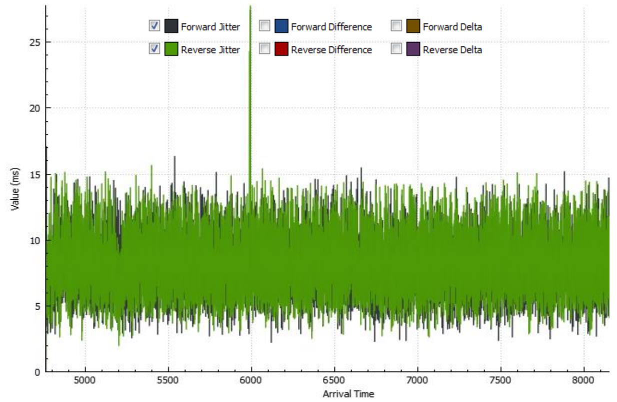

Fig. 34. Forward and reversed jitter for the R1 – R5 link.

Forward		Reverse	
192.168.2.3:8000 → 192.168.5.4:18122		192.168.5.4:18122 → 192.168.2.3:8000	
SSRC	0x011975ff	SSRC	0x3713f101
Max Delta	280.04 ms @ 1731259	Max Delta	110.51 ms @ 2132398
Max Jitter	21.93 ms	Max Jitter	16.93 ms
Mean Jitter	4.20 ms	Mean Jitter	8.14 ms
Max Skew	-1039.53 ms	Max Skew	-269.32 ms
RTP Packets	171454	RTP Packets	171478
Expected	171454	Expected	171478
Lost	0 (0.00 %)	Lost	0 (0.00 %)
Seq Errs	0	Seq Errs	0
Start at	8488.817582 s @ 1731180	Start at	8489.075114 s @ 1731201
Duration	3430.03 s	Duration	3429.69 s
Clock Drift	-617 ms	Clock Drift	29 ms
Freq Drift	7999 Hz (-0.02 %)	Freq Drift	8000 Hz (0.00 %)

Fig. 35. Summarized results for the R2 – R5 link.

Fig. 36 shows the RTD in the connection between VM1 and Asterisk (R3, R2, R5 and vice versa). Fig. 37 shows the RTD in the connection between VM2 and Asterisk (R2, R5 and vice versa). Fig. 38 shows the RTD in the connection between VM3 and Asterisk (R4, R5 and vice versa). Fig. 39 shows the RTD in the connection between VM4 and Asterisk (R1, R5 and vice versa). For all obtained results, there are slight improvements in the values compared to the results without the use of MPLS. However, the average values are the same as in Sections V and VI – A.

Fig. 40 presents the mathematical distribution of arrival times between the packets for the link between R2 and R5. As it can be seen from the distribution, the dependence is the same as in the virtual network with EIGRP and MPLS – the delay is constant.

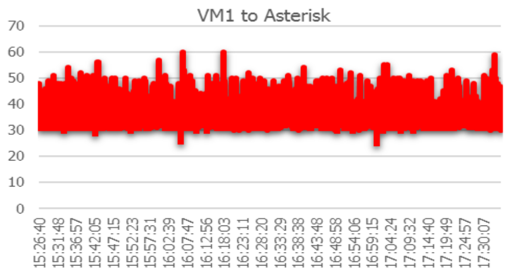

Fig. 36. Round-trip delay for the route from VM1 to the Asterisk PBX

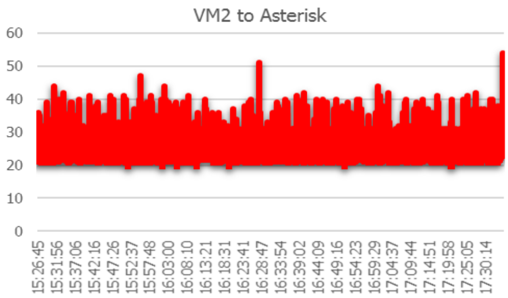

Fig. 37. Round-trip delay for the route from VM2 to the Asterisk PBX.

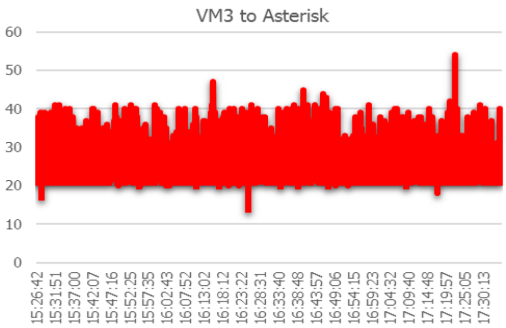

Fig. 38. Round-trip delay for the route from VM3 to the Asterisk PBX.

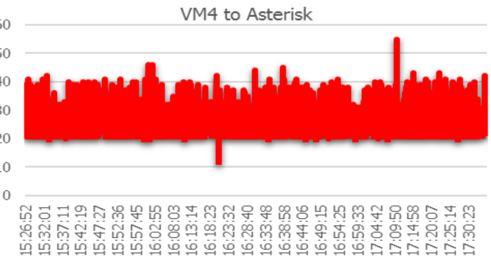

Fig. 39. Round-trip delay for the route from VM4 to the Asterisk PBX.

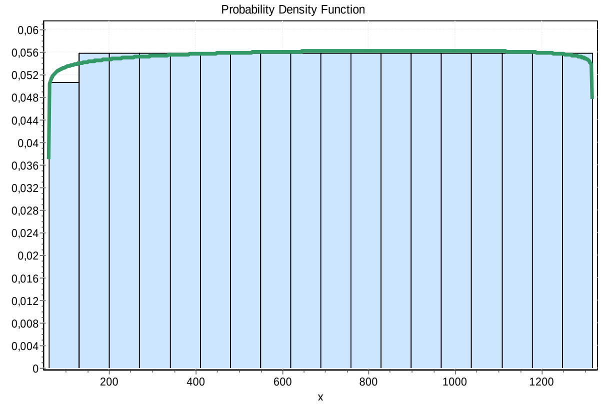

Fig. 40. Mathematical distribution.

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.

Forward		Reverse	
192.168.3.2:8000 → 192.168.5.4:19302		192.168.5.4:19302 → 192.168.3.2:8000	
SSRC	0x10a54b33	SSRC	0x37acfe57
Max Delta	110.01 ms @ 125987	Max Delta	60.01 ms @ 130996
Max Jitter	15.70 ms	Max Jitter	15.57 ms
Mean Jitter	7.99 ms	Mean Jitter	8.07 ms
Max Skew	-332.68 ms	Max Skew	-157.99 ms
RTP Packets	91927	RTP Packets	91915
Expected	91928	Expected	91915
Lost	1 (0.00 %)	Lost	0 (0.00 %)
Seq Errs	1	Seq Errs	0
Start at	1270.992614 s @ 125951	Start at	1271.447672 s @ 125977
Duration	1838.79 s	Duration	1838.37 s
Clock Drift	5 ms	Clock Drift	23 ms
Freq Drift	8000 Hz (0.00 %)	Freq Drift	8000 Hz (0.00 %)

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).

	Forward		Reverse
	192.168.4.3:8000 → 192.168.5.4:14860		192.168.5.4:14860 → 192.168.4.3:8000
SSRC	0xea8e887e	**SSRC**	0x28c6dcb3
Max Delta	60.01 ms @ 134467	**Max Delta**	110.51 ms @ 129492
Max Jitter	20.04 ms	**Max Jitter**	16.28 ms
Mean Jitter	7.95 ms	**Mean Jitter**	8.14 ms
Max Skew	-468.04 ms	**Max Skew**	-252.67 ms
RTP Packets	91919	**RTP Packets**	91925
Expected	91919	**Expected**	91925
Lost	0 (0.00 %)	**Lost**	0 (0.00 %)
Seq Errs	0	**Seq Errs**	0
Start at	1304.220309 s @ 129457	**Start at**	1304.220309 s @ 129459
Duration	1838.75 s	**Duration**	1838.69 s
Clock Drift	24 ms	**Clock Drift**	7 ms
Freq Drift	8000 Hz (0.00 %)	**Freq Drift**	8000 Hz (0.00 %)

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

	Forward		Reverse
	192.168.6.3:8000 → 192.168.5.4:19926		192.168.5.4:19926 → 192.168.6.3:8000
SSRC	0x5d95aff1	**SSRC**	0x3f715ddf
Max Delta	88.00 ms @ 704358	**Max Delta**	170.02 ms @ 477989
Max Jitter	16.52 ms	**Max Jitter**	15.51 ms
Mean Jitter	8.16 ms	**Mean Jitter**	8.24 ms
Max Skew	-426.47 ms	**Max Skew**	-349.44 ms
RTP Packets	134811	**RTP Packets**	134818
Expected	134811	**Expected**	134818
Lost	0 (0.00 %)	**Lost**	0 (0.00 %)
Seq Errs	0	**Seq Errs**	0
Start at	4749.193535 s @ 477956	**Start at**	4749.191535 s @ 477955
Duration	2696.54 s	**Duration**	2696.62 s
Clock Drift	17 ms	**Clock Drift**	14 ms
Freq Drift	8000 Hz (0.00 %)	**Freq Drift**	8000 Hz (0.00 %)

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

	Forward		Reverse
	192.168.2.3:8000 → 192.168.5.4:17090		192.168.5.4:17090 → 192.168.2.3:8000
SSRC	0xfad8b9ce	**SSRC**	0x489ac9c1
Max Delta	100.01 ms @ 643311	**Max Delta**	220.03 ms @ 643325
Max Jitter	16.21 ms	**Max Jitter**	15.24 ms
Mean Jitter	3.88 ms	**Mean Jitter**	8.14 ms
Max Skew	-302.92 ms	**Max Skew**	-362.62 ms
RTP Packets	75622	**RTP Packets**	75598
Expected	75622	**Expected**	75598
Lost	0 (0.00 %)	**Lost**	0 (0.00 %)
Seq Errs	0	**Seq Errs**	0
Start at	3186.658584 s @ 643282	**Start at**	3186.857610 s @ 643297
Duration	1512.63 s	**Duration**	1512.22 s
Clock Drift	14 ms	**Clock Drift**	40 ms
Freq Drift	8000 Hz (0.00 %)	**Freq Drift**	8000 Hz (0.00 %)

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.

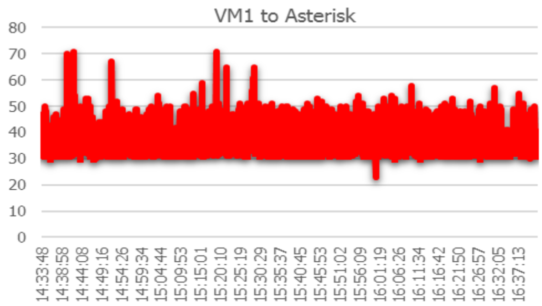

Fig. 45. Round-trip delay for the route from VM1 to the Asterisk PBX.

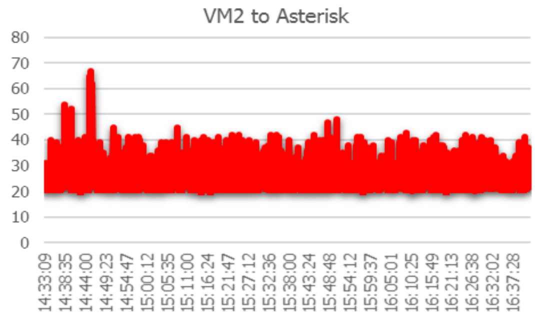

Fig. 46. Round-trip delay for the route from VM2 to the Asterisk PBX.

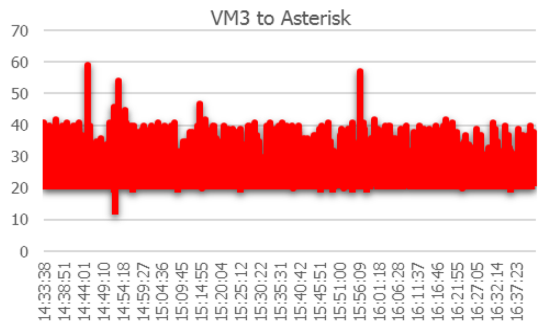

Fig. 47. Round-trip delay for the route from VM3 to the Asterisk PBX.

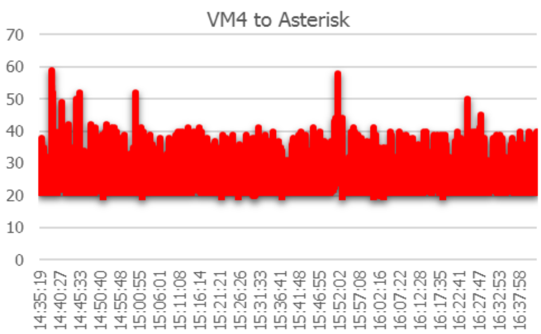

Fig. 48. Round-trip delay for the route from VM4 to the Asterisk PBX.

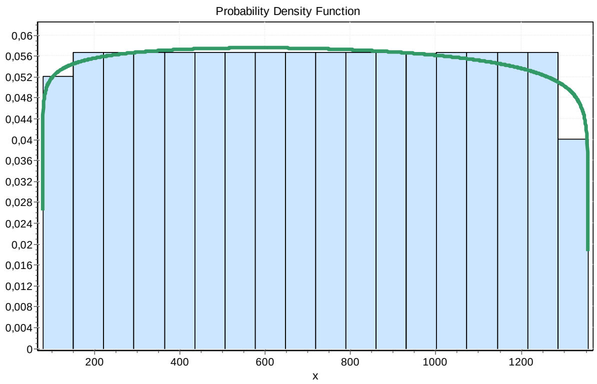

Fig. 49. Mathematical distribution.

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.

	Forward	Reverse
	192.168.3.2:8000 → 192.168.5.4:14228	192.168.5.4:14228 → 192.168.3.2:8000
SSRC	0x3ea6bb15	0x06ea24b6
Max Delta	129.52 ms @ 9099	180.02 ms @ 9089
Max Jitter	14.74 ms	17.16 ms
Mean Jitter	8.06 ms	8.05 ms
Max Skew	-377.58 ms	-354.05 ms
RTP Packets	100295	100293
Expected	100295	100293
Lost	0 (0.00 %)	0 (0.00 %)
Seq Errs	0	0
Start at	97.815685 s @ 9070	98.011210 s @ 9079
Duration	2006.19 s	2006.07 s
Clock Drift	12 ms	19 ms
Freq Drift	8000 Hz (0.00 %)	8000 Hz (0.00 %)

Fig. 50. Summarized results for the R2 – R3 link.

	Forward	Reverse
	192.168.4.3:8000 → 192.168.5.4:13410	192.168.5.4:13410 → 192.168.4.3:8000
SSRC	0xb8f63aaf	0x4b0afd6b
Max Delta	220.03 ms @ 218297	168.02 ms @ 248503
Max Jitter	17.78 ms	16.21 ms
Mean Jitter	7.94 ms	8.22 ms
Max Skew	-489.07 ms	-329.06 ms
RTP Packets	114294	114299
Expected	114294	114299
Lost	0 (0.00 %)	0 (0.00 %)
Seq Errs	0	0
Start at	2158.699766 s @ 218278	2158.721268 s @ 218280
Duration	2286.19 s	2286.10 s
Clock Drift	28 ms	31 ms
Freq Drift	8000 Hz (0.00 %)	8000 Hz (0.00 %)

Fig. 51. Summarized results for the R4 – R1 link.

	Forward	Reverse
	192.168.6.3:8000 → 192.168.5.4:16110	192.168.5.4:16110 → 192.168.6.3:8000
SSRC	0x18763b27	0x29d9c99d
Max Delta	180.02 ms @ 13964	131.01 ms @ 119098
Max Jitter	17.11 ms	15.87 ms
Mean Jitter	7.91 ms	8.16 ms
Max Skew	-447.06 ms	-347.06 ms
RTP Packets	100297	100294
Expected	100297	100294
Lost	0 (0.00 %)	0 (0.00 %)
Seq Errs	0	0
Start at	147.823790 s @ 13950	147.860795 s @ 13952
Duration	2006.22 s	2006.09 s
Clock Drift	20 ms	13 ms
Freq Drift	8000 Hz (0.00 %)	8000 Hz (0.00 %)

Fig. 52. Summarized results for the R1 – R5 link.

	Forward	Reverse
	192.168.2.3:8000 → 192.168.5.4:19612	192.168.5.4:19612 → 192.168.2.3:8000
SSRC	0x8480d312	0x22e1ebef
Max Delta	220.03 ms @ 439355	250.03 ms @ 439342
Max Jitter	18.83 ms	16.50 ms
Mean Jitter	3.97 ms	8.12 ms
Max Skew	-564.19 ms	-463.16 ms
RTP Packets	115154	115152
Expected	115154	115152
Lost	0 (0.00 %)	0 (0.00 %)
Seq Errs	0	0
Start at	2158.710140 s @ 439314	2158.912166 s @ 439322
Duration	2303.47 s	2303.32 s
Clock Drift	22 ms	23 ms
Freq Drift	8000 Hz (0.00 %)	8000 Hz (0.00 %)

Fig. 53. Summarized results for the R2 – R5 link.

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.

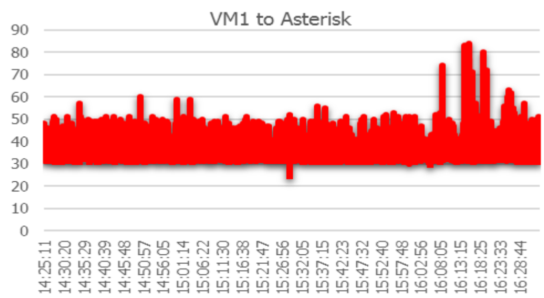

Fig. 54. Round-trip delay for the route from VM1 to the Asterisk PBX.

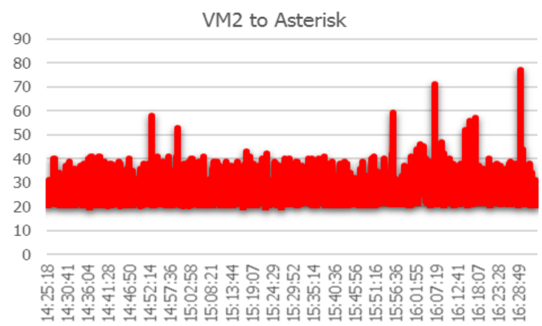

Fig. 55. Round-trip delay for the route from VM2 to the Asterisk PBX.

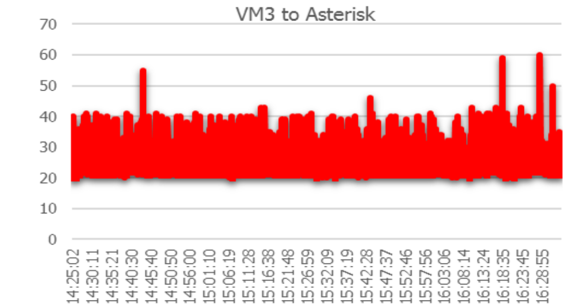

Fig. 56. Round-trip delay for the route from VM3 to the Asterisk PBX.

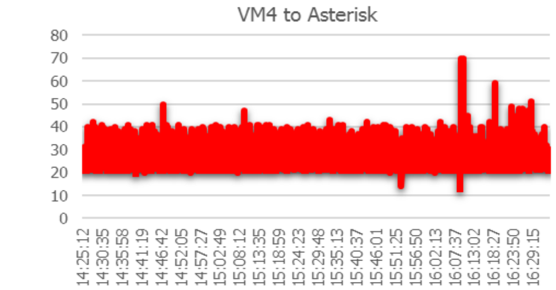

Fig. 57. Round-trip delay for the route from VM4 to the Asterisk PBX.

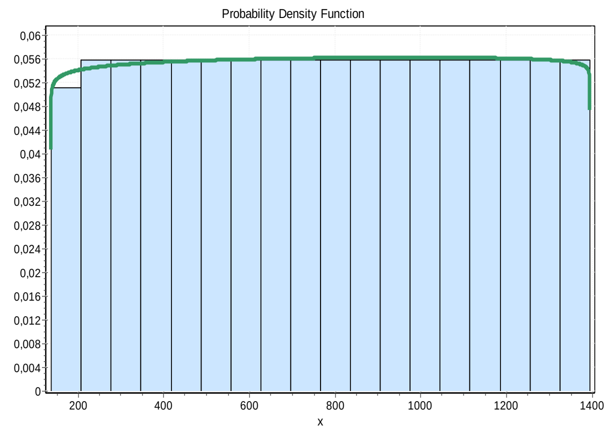

Fig. 58. Mathematical distribution.

Fig. 58 presents the mathematical distribution of arrival times between packets for the link between R2 and R5. As it can be seen, there is no difference with the results obtained so far - the time delay is constant.

C. Analysis of the Results Using OSPF

When we use OSPF, there are further improvements in voice flow parameters compared to RIP and EIGRP. The use of MPLS technology together with OSPF does not lead to any significant further improvements in the values of the monitored parameters. The mathematical distributions and graphs for the delay show that it has remained constant. This is due to the topology of the virtual network.

VIII. Conclusion

The created virtual network is working and through it voice traffic exchanges.

Known methods and techniques for monitoring of IP networks, as well as well-known tools for monitoring of IP networks, have been used during the study of the virtual network.

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 RIPv2. Enabling MPLS further improves the voice flow parameters. The network delay remains almost the same as with RIPv2.

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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