Review of Computer Engineering Studies
Vol. 10, No. 1, March, 2023, pp. 7-13
Journal homepage: http://iieta.org/journals/rces

Comparison of Flow Forwarding Between Software-Defined and Legacy Networks Based on Fixed Routing and QoS Conditions

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https://doi.org/10.18280/rces.100102

Received: 10 February 2023
Accepted: 23 March 2023

Keywords:
SDN, routing policies and protocols, network performance, OpenVSwitch, QoS, flow prioritization

ABSTRACT

The following article presents a comparison of flow forwarding between traditional and software-defined networks (SDN), focusing on the impact of routing protocols and policies. The present work evaluates the efficiency of SDN routing (bandwidth and packet loss) against the performance obtained in traditional networks and estimates the performance variation when using ad-hoc software implementations such as OpenVSwitch [1].

1. INTRODUCTION

Traditional routing protocols such as OSPF [2], BGP [3], RIP [4] and EIGRP [5] have been extensive and complete developments resulting in complex and rigid systems that are difficult to adapt to the current requirements of user services through the Internet. This rigidity reduces the possibility of use these protocols in networks with high volume of complex data types to be transmitted. The appearance of SDN [6] has introduced new concepts to solve this kind of problems.

Due to the increase in unforeseen failures in communication networks, it has become crucial to predict and know the approximate maximum time it takes the network to avoid or minimize packet and data loss.

Routers, with their traditional network routing protocols, require considerable convergence time decreasing bandwidth and increasing packet loss. This convergence time and its impact are present in SDN too so they will be critically important to benchmark networks performance's.

2. ANALYSIS SCENARIO

The chosen scenario is generally used to present routing protocols problems over traditional networks and to identify solutions defined through Traffic Engineering (TE) and Quality of Service (QoS). Those solutions will include routing protocols, QoS policies and load balancing mechanisms. The topology of the scenario is shown below in Figure 1.

The network traffic flows from VM1 and VM2 (Virtual Machine 1 and 2) to the through PE1 (Border Router) to enter to AREA 0 (shaded in yellow) according to the queuing system. A brief description of the types of routing queues and, particularly the one chosen for the development of this work, is detailed as follows.

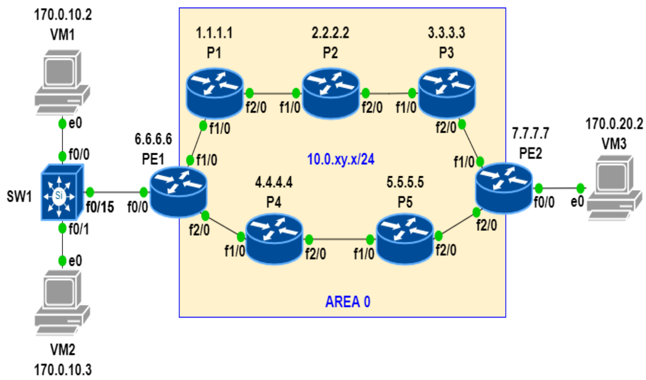
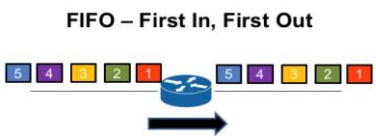


Figure 1. Analysis scenario

2.1 Queue types

The queuing system will be defined with one or more flows that may or may not enter a congested network. Queuing mechanisms are determinant regarding the performance of data networks. The different existing queuing mechanisms will have different bandwidths, delays, jitters and packet losses depending on the network congestion. In case of QoS, the following methods are used:

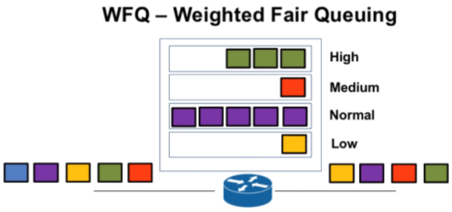
FIFO Queuing (First In First Out) [7]: There are no priorities or classification of types of traffic. The first packet to enter this queuing system is the first to leave, as shown in Figure 2.



Source: Cisco Systems

Figure 2. FIFO Queuing

WFQ (Weighted Fair Queuing): This queuing method generates a different queue with its priority for each flow type. When a queue is empty of packets, the system continues with the next priority queue as shown in Figure 3.



Source: Cisco Systems

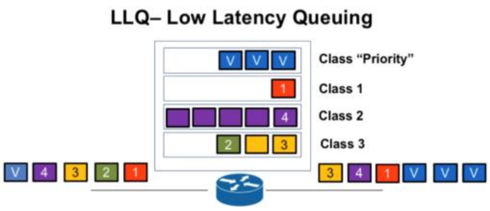
Figure 3. WFQ mechanism

CBWFQ (Class Based Weighted Fair Queuing): Because WFQ has scaling limitations when the traffic per link increases, CBWFQ incorporates the use of a weighted Round Robin algorithm where it establishes different attention times for each queue depending on the guaranteed bandwidth for each of them, as shown in Figure 4.

Source: Cisco Systems

Figure 4. CBWFQ mechanism

LLQ (Low Latency Queueing): It is an extension of the CBWFQ mechanism that adds an additional PQ queue with the highest priority over the others as shows in Figure 5. LLQ queuing (CBWFQ + PQ) is currently the most recommended method for VoIP, IP telephony and video conferencing.



Source: Cisco Systems

Figure 5. LLQ mechanism

This work uses the CBWFQ (Figure 4) and LLQ (Figure 5) queues. Our experiment will prioritize the UDP flow taking as a premise to obtain the minimum loss of packets. For this reason, the queue model considered was the one that allows assigning absolute priorities, the LLQ protocol.

Although UDP is used in applications that can be treated as Best Effort and as a transport for real-time applications, this work will not focus on these characteristics within the experimentation.

3. EXPERIMENTS

Two scenarios were designed where each of them will incorporate an improvement over the previous one. These two scenarios were considered for both traditional and SDN networks.

Traditional networks scenario: A single TCP stream will be sent to test the available bandwidth for network saturation. In the following test, a TCP flow and a UDP flow are sent along the path that the routing protocol has defined. Finally, both flows are sent again adding QoS policies.

SDN scenario: We will force the sending of a TCP flow through the longest route with the same objective as in the case of traditional networks. TCP and UDP streams are then sent in order to improve packet loss mitigation. The UDP stream will be sent by the shortest path.

4. TEST AND RESULTS

4.1 Traditional networks scenario

4.1.1 Test without QoS, with Iperf3 [8], single stream over GNS3 [9], OSPF protocol

A first TCP stream is sent over port 5201 from VM1 to VM3. With this configuration we obtain that the capacity of the system is approximately 17 Mbps at the moment of starting the packet loss. This result will set the bandwidth limit as a parameter in the subsequent performance tests of this scenario. The OSPF protocol chose the path PE1–P4–P5–PE2 within "Area 0" as shown in Figure 6.

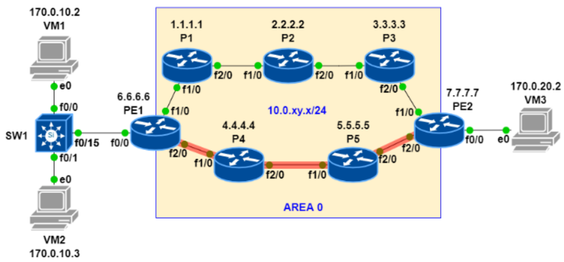


Figure 6. Route chosen by the OSPF protocol

Figure 7 shows the sending of the TCP flow generated in Iperf from VM1 to VM3 through port 5201 in VM1.

```
VM1: iperf3 -c 170.0.20.2 -p 5201 -t 60
Server listening on 5201
Accepted connection from 170.0.10.2, port 48222
[  5] local 170.0.20.2 port 5201 connected to 170.0.10.2 port 48222
[ ID] Interval          Transfer     Bandwidth
[  5]   0.00-1.00   sec  1.96 MBytes  16.5 Mbits/sec
[  5]   1.00-2.00   sec  2.05 MBytes  17.2 Mbits/sec
..................
[ ID] Interval           Transfer     Bandwidth       Retr
[  5]   0.00-60.09  sec   125 MBytes  17.4 Mbits/sec   81          sender
[  5]   0.00-60.09  sec   124 MBytes  17.3 Mbits/sec               receiver
```

Figure 7. Iperf3 outcome

Results: Figure 8 shows the TCP flows sent. We can appreciate the maximum capacity of the system (Bandwidth).

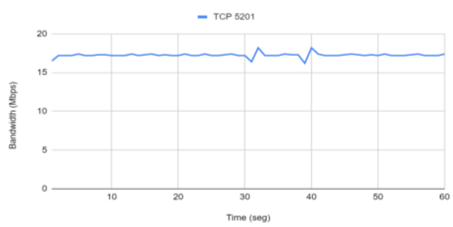


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Figure 8. Bandwidth performance test

4.1.2 Test without QoS, with Iperf3, double streams over GNS3, OSPF protocol

Two flows were sent, one from VM1 to VM3 and one from VM2 to VM3. UDP streams were sent through port 5060 and TCP through port 5201 respectively. According to the OSPF protocol, the chosen route is the same as in the previous case, PE1–P4–P5–PE2.

Figure 9 shows the output generated by Iperf3 of the UDP flow sent from VM1 to VM3.

```
VM1: iperf3 -c 170.0.20.2 -p 5060 -t 60 -u -b

Server listening on 5060

Accepted connection from 170.0.10.2, port 48294

[  5] local 170.0.20.2 port 5060 connected to 170.0.10.2 port 48294
[ ID] Interval           Transfer     Bandwidth       Jitter    Lost/Total Datagrams
[  5]   0.00-1.00   sec   840 KBytes  6.88 Mbits/sec  7.074 ms  50/155 (32%)
[  5]   1.00-2.00   sec  1.02 MBytes  8.52 Mbits/sec  7.376 ms  53/183 (29%)
[  5]   2.00-3.00   sec  1.01 MBytes  8.46 Mbits/sec  6.915 ms  55/184 (30%)
[  5]   3.00-4.00   sec   992 KBytes  8.13 Mbits/sec  8.013 ms  58/182 (32%)
[  5]   4.00-5.00   sec  1.04 MBytes  8.72 Mbits/sec  6.878 ms  51/184 (28%)
.........
[  5]  57.00-58.00  sec  1.03 MBytes  8.65 Mbits/sec  6.894 ms  51/183 (28%)
[  5]  58.00-59.00  sec  1008 KBytes  8.26 Mbits/sec  7.960 ms  57/183 (31%)
[  5]  59.00-60.00  sec  1.07 MBytes  8.98 Mbits/sec  6.780 ms  48/185 (26%)
[  5]  60.00-60.06  sec  16.0 KBytes  2.36 Mbits/sec  6.590 ms  0/2 (0%)

[ ID] Interval           Transfer     Bandwidth       Jitter    Lost/Total Datagrams
[  5]   0.00-60.06  sec  85.7 MBytes  12.0 Mbits/sec  6.590 ms  3282/10962 (30%)
```

Figure 9. Iperf3 output for UDP flow

Figure 10 shows the output generated by Iperf3 of the TCP flow sent from VM2 to VM3.

```
VM2: iperf3 -c 170.0.20.2 -p 5201 -t 60

Server listening on 5201

Accepted connection from 170.0.10.3, port 45102

[  5] local 170.0.20.2 port 5201 connected to 170.0.10.3 port 45102
[ ID] Interval           Transfer     Bandwidth
[  5]   0.00-1.00   sec   895 KBytes  7.33 Mbits/sec
[  5]   1.00-2.00   sec   713 KBytes  5.84 Mbits/sec
[  5]   2.00-3.00   sec   795 KBytes  6.50 Mbits/sec
[  5]   3.00-4.00   sec   803 KBytes  6.59 Mbits/sec
[  5]   4.00-5.00   sec   820 KBytes  6.71 Mbits/sec
[  5]   5.00-6.00   sec   817 KBytes  6.70 Mbits/sec
.........
[  5]  57.00-58.00  sec   731 KBytes  5.99 Mbits/sec
[  5]  58.00-59.00  sec   775 KBytes  6.34 Mbits/sec
[  5]  59.00-60.00  sec   798 KBytes  6.54 Mbits/sec
[  5]  60.00-60.04  sec  35.4 KBytes  6.51 Mbits/sec

[ ID] Interval           Transfer     Bandwidth       Retr
[  5]   0.00-60.04  sec  42.9 MBytes  5.99 Mbits/sec  148             sender
[  5]   0.00-60.04  sec  42.7 MBytes  5.96 Mbits/sec                  receiver
```

Figure 10. Iperf3 output for TCP flow

Results: In Figure 11 we can see the difference in the bandwidth used by each of the flows. The UDP flow tries to gain the maximum bandwidth allowed by the system, but the TCP flow limits its maximum reach.

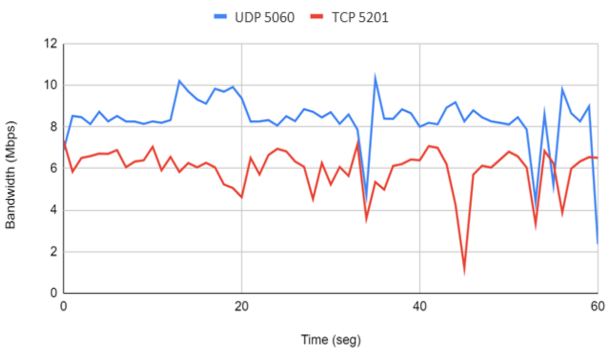

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Figure 11. Bandwidth used by UDP and TCP flows

On the other hand, the percentage of packets lost for the UDP flow reaches peaks around 60% as shown in Figure 12.

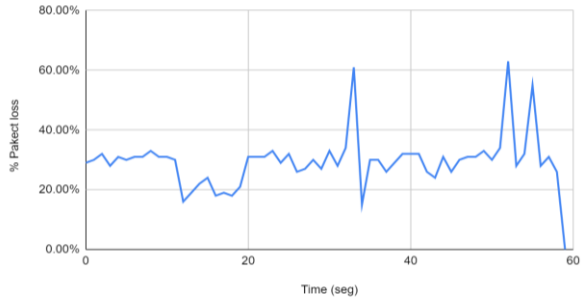

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Figure 12. Packet loss in UDP flows

4.1.3 Test with QoS, with Iperf3, double streams over GNS3, OSPF protocol

In the previous tests with UDP and TCP flows without QoS (experiment 4.1.1) considerable packet losses were observed, especially in the UDP flow. This configuration will try to correct these losses by applying QoS policies on the chosen queuing mechanism.

In this test, a preferential delivery service is configured for the applications that needed it, ensuring sufficient bandwidth, controlling latency and reducing data loss.

Two flows were sent, one from VM1 to VM3 and one from VM2 to VM3. UDP streams were sent through port 5060 and TCP through port 5201 respectively. According to the OSPF protocol. The chosen route is the same as in the previous case. Figure 13 shows the UDP flow from VM1 to VM3 generated under Iperf3.

```
VM1: iperf3 -c 170.0.20.2 -p 5060 -t 60 -u -b

Server listening on 5060

Accepted connection from 170.0.10.2, port 48296

[  5] local 170.0.20.2 port 5060 connected to 170.0.10.2 port 48296
[ ID] Interval           Transfer     Bandwidth       Jitter    Lost/Total Datagrams
[  5]   0.00-1.00   sec   720 KBytes  5.90 Mbits/sec  8.962 ms  3/93 (3.2%)
[  5]   1.00-2.00   sec   840 KBytes  6.87 Mbits/sec  7.920 ms  0/105 (0%)
[  5]   2.00-3.00   sec   864 KBytes  7.08 Mbits/sec  7.056 ms  0/108 (0%)
[  5]   3.00-4.00   sec   832 KBytes  6.81 Mbits/sec  7.840 ms  0/104 (0%)
[  5]   4.00-5.00   sec   872 KBytes  7.15 Mbits/sec  6.913 ms  1/110 (0.91%)
[  5]   5.00-6.00   sec   848 KBytes  6.95 Mbits/sec  8.062 ms  1/107 (0.93%)
[  5]   6.00-7.00   sec   840 KBytes  6.88 Mbits/sec  8.250 ms  0/105 (0%)
[  5]   7.00-8.00   sec   872 KBytes  7.15 Mbits/sec  8.600 ms  0/109 (0%)
[  5]   8.00-9.00   sec   848 KBytes  6.94 Mbits/sec  7.942 ms  0/106 (0%)
[  5]   9.00-10.00  sec   848 KBytes  6.95 Mbits/sec  8.632 ms  0/106 (0%)
.........
Ospf
[  5]  53.00-54.00  sec   840 KBytes  6.88 Mbits/sec  7.243 ms  0/105 (0%)
[  5]  54.00-55.00  sec   848 KBytes  6.95 Mbits/sec  8.853 ms  1/107 (0.93%)
[  5]  55.00-56.00  sec   840 KBytes  6.87 Mbits/sec  7.931 ms  2/107 (1.9%)
[  5]  56.00-57.00  sec   832 KBytes  6.82 Mbits/sec  9.011 ms  3/107 (2.8%)
[  5]  57.00-58.00  sec   840 KBytes  6.88 Mbits/sec  7.532 ms  1/106 (0.94%)
[  5]  58.00-59.00  sec   848 KBytes  6.95 Mbits/sec  8.901 ms  0/106 (0%)
[  5]  59.00-60.00  sec   832 KBytes  6.80 Mbits/sec  6.157 ms  9/113 (8%)
[  5]  60.00-60.06  sec  8.00 KBytes  1.23 Mbits/sec  6.431 ms  0/1 (0%)
- - - - - - - - - - - - - - - - - - - - - - - - -
[ ID] Interval           Transfer     Bandwidth       Jitter    Lost/Total Datagrams
[  5]   0.00-60.06  sec  50.0 MBytes  6.98 Mbits/sec  6.431 ms  140/6399 (2.2%)
```

Figure 13. Iperf3 output for UDP flow

Figure 14 shows the output generated by Iperf3 of the TCP flow sent from VM2 to VM3.

```
VM2: iperf3 -c 170.0.20.2 -p 5201 -t 60

Server listening on 5201

Accepted connection from 170.0.10.3, port 45110

[  5] local 170.0.20.2 port 5201 connected to 170.0.10.3 port 45110
[ ID]   Interval         Transfer     Bandwidth
[  5]   0.00-1.00   sec  1.50 MBytes  12.6 Mbits/sec
[  5]   1.00-2.00   sec  1.21 MBytes  10.1 Mbits/sec
[  5]   2.00-3.01   sec  1.17 MBytes  9.73 Mbits/sec
[  5]   3.01-4.00   sec  1.17 MBytes  9.87 Mbits/sec
[  5]   4.00-5.00   sec  1.17 MBytes  9.80 Mbits/sec
[  5]   5.00-6.00   sec  1.17 MBytes  9.80 Mbits/sec
[  5]   6.00-7.00   sec  1.19 MBytes  10.0 Mbits/sec
[  5]   7.00-8.00   sec  1.18 MBytes  9.87 Mbits/sec
[  5]   8.00-9.00   sec  1.17 MBytes  9.80 Mbits/sec
[  5]   9.00-10.00  sec  1.19 MBytes  9.99 Mbits/sec
[  5]  10.00-11.00  sec  1.17 MBytes  9.80 Mbits/sec
[  5]  11.00-12.00  sec  1.19 MBytes  9.98 Mbits/sec
[  5]  12.00-13.01  sec  1.18 MBytes  9.81 Mbits/sec
.........
[  5]  56.00-57.00  sec  1.19 MBytes  9.97 Mbits/sec
[  5]  57.00-58.00  sec  1.17 MBytes  9.76 Mbits/sec
[  5]  58.00-59.00  sec  1.20 MBytes  10.1 Mbits/sec
[  5]  59.00-60.00  sec  1.18 MBytes  9.89 Mbits/sec
[  5]  60.00-60.08  sec  79.2 KBytes  7.88 Mbits/sec
- - - - - - - - - - - - - - - - - - - - - - - - -
[ ID] Interval           Transfer     Bandwidth       Retr
[  5]   0.00-60.08  sec  70.2 MBytes  9.81 Mbits/sec  52              sender
[  5]   0.00-60.08  sec  70.0 MBytes  9.77 Mbits/sec                  receiver
```

Figure 14. Iperf3 output for TCP flow

Results: Compared to the previous test, it can be seen how both flows are stabilized and how the packet loss of UDP flows is significantly reduced. It can also be seen how UDP flows take full advantage of the 7 Mbps, leaving the remaining 10 Mbps to TCP flows (Figure 15). In GNS3, the total 17 Mbps of bandwidth has been defined as a parameter.

Own source

Figure 15. Bandwidth used by UDP and TCP flows with QoS

Figure 16 shows the considerable decrease of 40% in the average packet loss, which results in a final value of 4%.

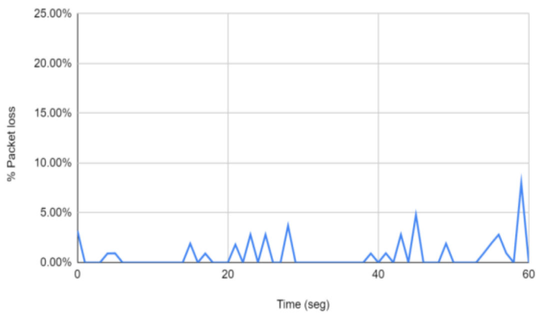

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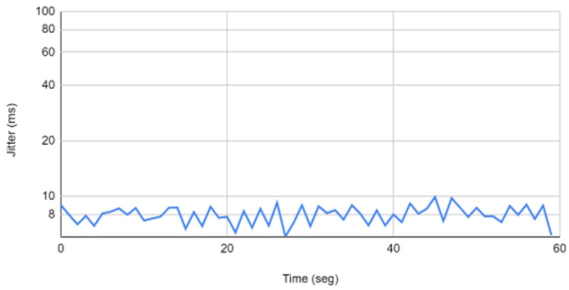
Figure 16. Packet loss in the flow under UDP protocol with QoS

Figure 17 shows the result of jitter (delay variation) for the UDP flow.



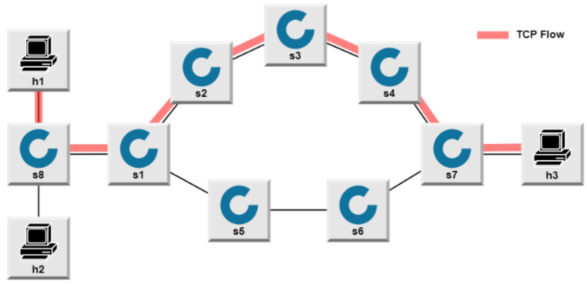
Own source

Figure 17. Jitter of the UDP flow with the application of differentiated services

4.2 SDN networks scenario

4.2.1 Tests with Iperf3, single flow over Mininet [10], rule of minimum number of skips

TCP streams are sent through port 5201 from h1 to h3 along the fixed path defined as s1–s2–s3–s4–s7. OSPF is not used. Figure 18 shows the scenario together with the limitation of interfaces.



Own source

Figure 18. SDN topology scenario for the TCP flow

To match the configuration of the tests developed in traditional networks, the maximum capacity of the system has been determined at 17 Mbps (Figure 19).

```
for i in {2..6}
do
tc qdisc add dev s$i-eth1 root tbf rate 17Mbit latency 50ms burst 1540
tc qdisc add dev s$i-eth2 root tbf rate 17Mbit latency 50ms burst 1540
done

for i in {1,7,8}
do
tc qdisc add dev s$i-eth1 root tbf rate 17Mbit latency 50ms burst 1540
tc qdisc add dev s$i-eth2 root tbf rate 17Mbit latency 50ms burst 1540
tc qdisc add dev s$i-eth3 root tbf rate 17Mbit latency 50ms burst 1540
done
```

Figure 19. Limiting SDN Bandwidth on Mininet

As in the previous case, the bandwidth limit is decisive for the following performance tests. The following rules were established with priority 99:

- Decrement of TTL
- Modify ethernet address
- Exit by interface

Figure 20 shows the configuration code for these rules in the Mininet simulator.

```
#s1
ovs-ofctl -Oopenflow13 add -flow s1 priority=99,eth,ip,in_port=s1 -eth1,actions='dec_ttl,
          mod_dl_src:00:00:00:00:00:12,mod_dl_dst:00:00:00:00:00:21,output:2'

ovs-ofctl -Oopenflow13 add- flow s1 priority=99,eth,ip,in_port=s1 -eth2,actions='dec_ttl,
          mod_dl_src:00:00:00:00:00:11, mod_dl_dst:00:00:00:00:00:81,output:1'

#s2
ovs-ofctl -Oopenflow13 add- flow s2 priority=99,eth,ip,in_port=s2 -eth1,actions='dec_ttl,
          mod_dl_src:00:00:00:00:00:22,mod_dl_dst:00:00:00:00:00:31,output:2'

ovs-ofctl -Oopenflow13 add -flow s2 priority=99,eth,ip,in_port=s2 -eth2,actions='dec_ttl,
          mod_dl_src:00:00:00:00:00:21, mod_dl_dst:00:00:00:00:00:11,output:1'

...............
#s8
ovs-ofctl -Oopenflow13 add -flow s8 priority=99,eth,ip,nw_dst=172.0.20.2,actions='dec_ttl,
          mod_dl_src:00:00:00:00:00:81,mod_dl_dst:00:00:00:00:00:11,output:1'

ovs-ofctl -Oopenflow13 add -flow s8 priority=99,eth,ip,in_port=1,nw_dst=172.0.10.2,actions='dec_ttl,
          mod_dl_src:00:00:00:00:00:82,mod_dl_dst:00:00:00:00:00:01, output:2'

ovs-ofctl -Oopenflow13 add-flow s8 priority=99,eth,ip,in_port=1,nw_dst=172.0.10.3,actions='dec_ttl,
          mod_dl_src:00:00:00:00:00:83,mod_dl_dst:00:00:00:00:00:02,output:3'

ovs-ofctl -Oopenflow13 add-flow s8 priority=80,actions=NORMAL
```

Figure 20. Mininet setup rules

Results: It is observed that the bandwidth remains within the limits established by the parameters specified in the rules (Figure 21).

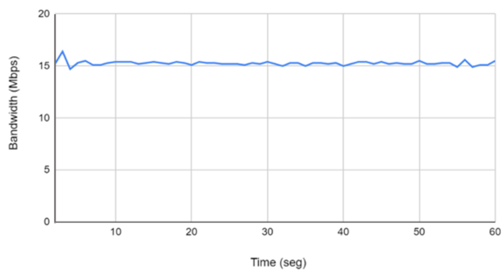


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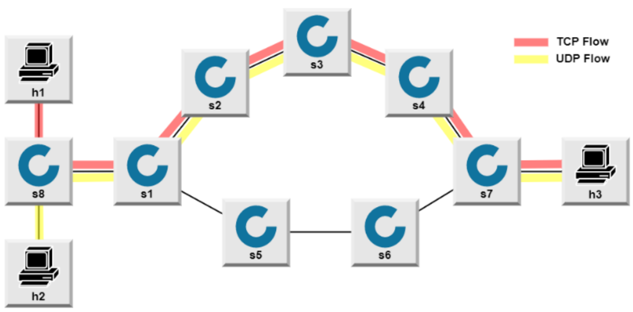
Figure 21. Bandwidth of the TCP flow

4.2.2 Tests with Iperf3, double stream over Mininet, traffic sent by the longest route

Two flows were sent, one TCP from h1 to h3 through port 5201 and another UDP from h2 to h3 through port 5202. The same route is used as in the previous case (Figure 18) as shown in Figure 22.



Own source

Figure 22. SDN topology scenario for the TCP and UDP flows through the same route

In Figure 23, we observe the bandwidths occupied by each of the flows.

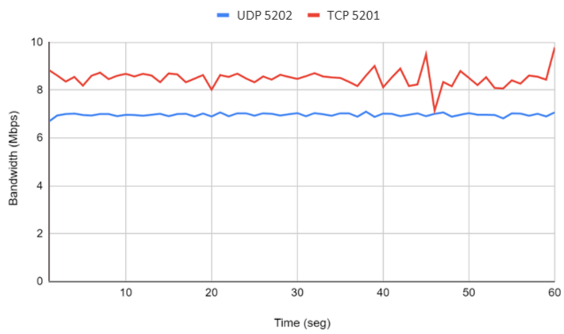


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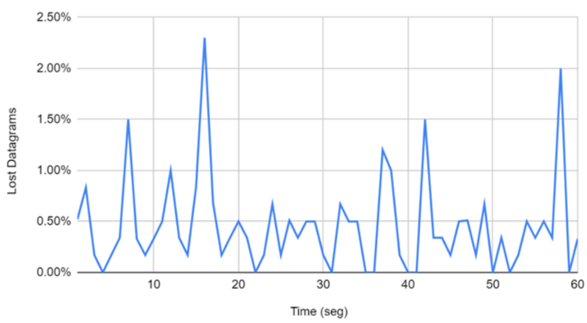
Figure 23. Bandwidth of TCP and UDP flows

Results: The number of packets lost in UDP flow is considerably lower for the same flow observed in Figure 16 despite the fact that, in this case, the two flows were sent by the longest route (Figure 24).

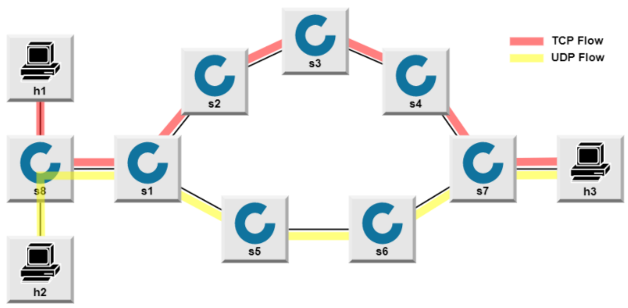


Own source

Figure 24. Number of packets lost from the UDP flow

4.2.3 Tests with Iperf3, double stream over Mininet, rules: minimum number of skips for UDP flow and TCP flow sent by the longest route

The flows were divided as follows: the TCP flow is sent through the route s1–s2–s3–s4–s7 through port 5202 and the UDP flow through the route s1–s5–s6–s7 through the port 5201 (Figure 25).



Own source

Figure 25. SDN topology scenario for the TCP and UDP flows through different routes

Set up of flows in Mininet (Figure 26):

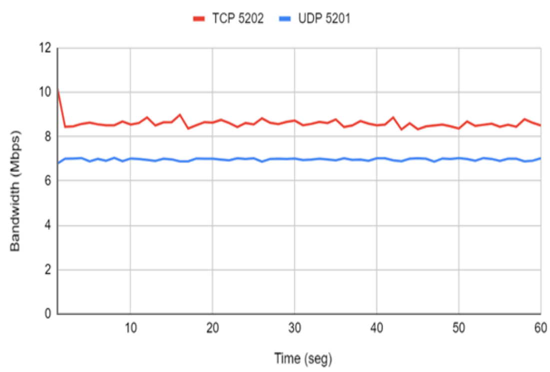
- Flows with priority 100 on s1 and s7 to correspond to TCP/UDP ports
- TCP flow: 5202 port, s1–s2–s3–s4–s7 route
- UDP flow: 5201 port, s1–s5–s6–s7 route

```
#s1
#5202 upper route
ovs-ofctl -Oopenflow13 add -flow s1 priority=100,eth,ip,tcp,in_port=s1 -eth1,tcp_dst=5202,
         actions='dec_ttl,mod_dl_src:00:00:00:00:00:12,mod_dl_dst:00:00:00:00:00:21,output:2'

ovs-ofctl -Oopenflow13 add -flow s1 priority=100,eth,ip,tcp,in_port=s1 -eth2,tcp_src=5202,
         actions='dec_ttl,mod_dl_src:00:00:00:00:00:11,mod_dl_dst:00:00:00:00:00:81,output:1'

#5201 lower route
ovs-ofctl -Oopenflow13 add -flow s1 priority=100,eth,ip,udp,in_port=s1 -eth1, udp_dst=5201,
         actions='dec_ttl,mod_dl_src:00:00:00:00:00:13,mod_dl_dst:00:00:00:00:00:51,output:3'

ovs-ofctl -Oopenflow13 add -flow s1 priority=100,eth,ip,udp,in_port=s1 -eth3, udp_src=5201,
         actions='dec_ttl,mod_dl_src:00:00:00:00:00:81,mod_dl_dst:00:00:00:00:00:11,output:1'

#s7
#5202 upper route
ovs-ofctl -Oopenflow13 add -flow s7 priority=100,eth,ip,tcp,in_port=s7 -eth2, tcp_dst=5202,
         actions='dec_ttl,mod_dl_src:00:00:00:00:00:71,mod_dl_dst:00:00:00:00:00:03,output:1'

ovs-ofctl -Oopenflow13 add -flow s7 priority=100,eth,ip,tcp,in_port=s7 -eth1, tcp_src=5202,
         actions='dec_ttl,mod_dl_src:00:00:00:00:00:72,mod_dl_dst:00:00:00:00:00:42,output:2'

#5201 lower route
ovs-ofctl -Oopenflow13 add -flow s7 priority=100,eth,ip,udp,in_port=s7 -eth3, udp_dst=5201,
         actions='dec_ttl,mod_dl_src:00:00:00:00:00:71,mod_dl_dst:00:00:00:00:00:03,output:1'

ovs-ofctl -Oopenflow13 add -flow s7 priority=100,eth,ip,udp,in_port=s7 -eth1, udp_src=5201,
         actions='dec_ttl,mod_dl_src:00:00:00:00:00:73,mod_dl_dst:00:00:00:00:00:62,output:3'
```

Own source

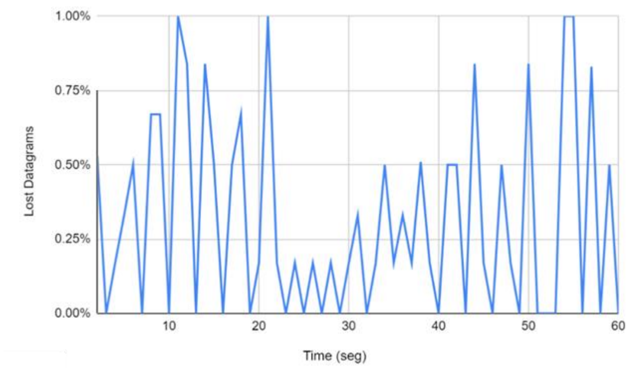
Figure 26. Mininet setup of UDP and TCP flows

Results: Figure 27 shows that the bandwidth is not affected in either of the two flows. Also, in Figure 28, we appreciate UDP packet loss is significantly reduced to 1%.



Own source

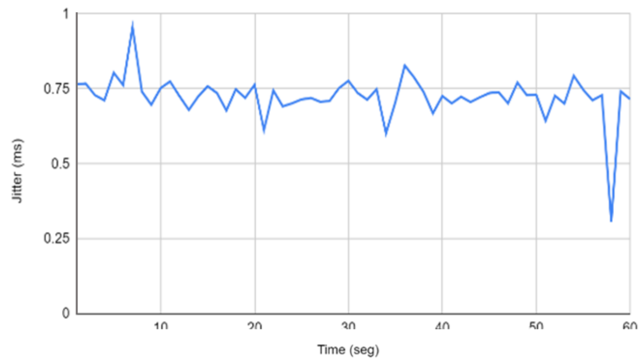
Figure 27. Bandwidth of both streams



Own source

Figure 28. Lost packets for the UDP flow

Figure 29 shows the result of the jitter (delay variation) for the UDP flow.



Own source

Figure 29. UDP flow jitter

4.3 Considerations for future experiments

In heterogeneous networks, both traditional and SDN, their developments have been much more complex. This brings many challenges to organize, manage and optimize network resources in a more effective way. One possible way to solve these problems is to incorporate more intelligence into networks as proposed by the Knowledge Plane approach (KP) [11] by applying Machine Learning (ML) [12] and cognitive techniques. However, the KP has not been prototyped or implemented at the time of writing this present work.

In traditional networks, each router or switch node can only see and act on a small portion of the network; If we need to control the entire network, it is very complex to learn from each node since they only have a small partial view of their environment. Future developments in SDN networks are expected to make it easier to learn the entire network as a whole [13].

5. CONCLUSIONS

To sustain end-to-end QoS it will be very important to dig into the dynamic behavior of networks through measured and monitored parameters. Among the parameters that determine whether the level of service offered is met, the most important are packet loss and jitter [14-16].

Following these considerations mentioned before, we can see that as we have simplified the schemes and configurations defined at the beginning, we have obtained conclusions that confirm the objectives established.

The conclusions obtained are detailed below:

- The idea of a routing based on optimal routes seems to lead to a balanced routing that meets committed QoS parameters for all flows, links, routes and available resources in the network.
- Networks will be able to adapt to future changes using metrics to control available bandwidth, lost packets in prioritized flows, or necessary jitter. This will be achieved by dynamically modifying the rules in the routing tables through the static or dynamic configuration of the required network behavior patterns.
- SDN networks inherently allow us to apply custom-developed algorithms without being tied to routing protocols such as OSPF. In the present work we were able to observe that in SDN networks over traditional networks with Differentiated Services, if we send different flows through different routes, we will improve performance parameters such as packet loss or jitter.
- Bandwidth could be used much more efficiently by incorporating few rules into SDN. This task was simple and did not require the application of specific protocols for each section of the network.
- Performance in SDN networks can be increased with a simple division of traffic, obtaining a delay variation of 0.75 msec on average (Figure 29) against one of 8 msec for traditional networks (Figure 16).
- With the application of Differentiated Services in traditional networks, the average packet loss was significantly reduced from 45% to 4%. In the case of SDN networks, when the flows were divided into two different routes, this decrease was from 2% to 1%.

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NOMENCLATURE

BGP	Border Gateway Protocol
EIGRP	Enhanced Interior Gateway Routing Protocol
ex	Lower hierarchy link x
fx/y	Top hierarchy link x/y
GNS3	Graphical Network Simulator-3 software
hx	Host x
IP	Internet Protocol
Iperf3	Network emulator software
KP	Knowledge Plane
Mbps	Megabits per second
ML	Machine Learning
Mininet	Network emulator software
msec	Miliseconds
OSPF	Open Shortest Path First
Px	Legacy Router x
Pex	Legacy Border Router x
QoS	Quality of Service
RIP	Routing Information Protocol
sx	SDN switch x
SWx	Legacy Switch x
TCP	Transmission Control Protocol
TE	Traffic Engineering
TTL	Time To Live
UDP	User Datagram Protocol
VMx	Virtual Machine x
VoIP	Voice over IP
x.x.x.x/y	IPV4 Address