QOS BASED-QROUTING AND OPTIMIZED PACKET SCHEDULER FOR 5G IP NETWORK

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ABSTRACT

Technological development and the current era of the Internet are a concern for telecommunications operators and are prompting researchers in the field to find long-term solutions to ensure user satisfaction in terms of Quality of Service (QoS) on each of the different existing and future applications.
Indeed, the expected developments in the 5G networks era present a very high level of requirements, both in technological and quality terms. In this vision, our work in this paper consists of proposing optimization solutions at the routing and scheduling level of IP packets from approaches based on QoS control. Then, the performance of the new proposed techniques is evaluated within 5G IP Network as IMS and Internet.

Keyword: *5G IP Network, QoS, IP routing, Reinforcement learning, Q-Routing, Scheduling*

1. INTRODUCTION

It is reported that the priorities for the future Internet research under Horizon 2020 (H2020) must aim for impact in products, services, capabilities and benefits in about 10 years from now. As for technical aspects, if we assume that the network convergence and cloud have already happened and look forward, we will view the future Internet not as network, cloud, storage or devices, but as the execution environment for smart applications, services, interaction, experience and data. [1]
Hence, it is also expected, according to the Public Private Partnership (PPP) program proposed by the European Union (EU) [1], that with the unprecedented growing users' demands, the network does require scalable, reliable, cost- and energy-efficient solutions for the creation of value added services, transported through differentiated QoS guarantees, and a wide range of QoS options for customers.

Thus, we have chosen to devote our work to the routing and management of queues (in particular at the level of the scheduling mechanism) because these two functions are complementary and are fundamental to the proper functioning of the network because they interact directly with the traffic to be handled, and therefore with the QoS control of the latter. Indeed, the general objectives of these two concepts are to maximize the useful throughput of the network; meet the performance indicator thresholds to ensure the quality of service expected by users: low network transit delay as well as acceptable jitter, low probability of packet loss, guaranteed availability of a route or of the network, fairness between users; to minimize the economic cost to use; and finally to avoid congestion.
The first routing optimization approach will be an algorithm based on the Q-Routing technique with the aim of overcoming the limitations of the classical routing protocols based on the Bellman-Ford and Dijkstra algorithms by meeting the needs expected by the various services.
As for the new scheduling technique, it will be the result of the combination of the principles of the two schedulers Priority Queuing (PQ) and Weighted Fair Queuing (WFQ) aiming to best meet the requirements in terms of quality of service for each type of application running through the network.

The implementation of our two optimization approaches will be done by developing our algorithms and implementing them in routers. Then, we will evaluate and analyze their performances by comparing them with existing techniques.

2. OVERVIEW OF 5G TELECOMMUNICATION NETWORK

2.1 Global architecture of 5G Network

The new architecture of the 5G telecommunication network is composed of a new radio access called Next Generation RAN (NG-RAN) and a core network called 5G Core (5GC) [2] [3]. Figure 1 shows this overall architecture.

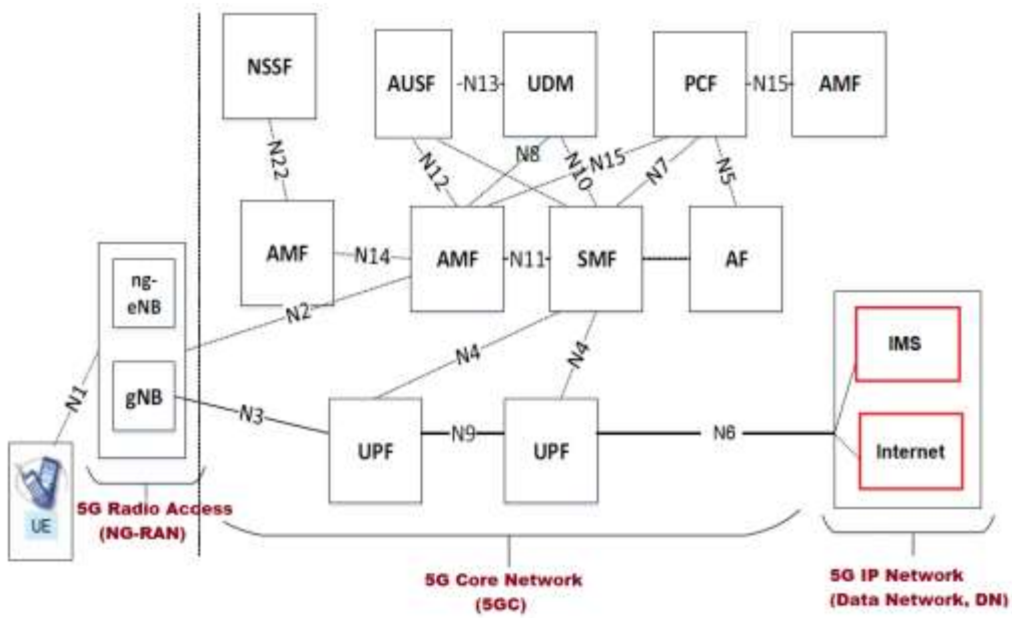

Fig -1: 5G Architecture with interfaces between entities

In the 5G Radio Access part, the user equipment or User Equipment (UE) communicate with the base stations either by a 5G radio link or by a 4G radio link. We have two types of base station:

- The gNB (Next Generation Node Base Station) which is the new radio eNodeB specific to 5G communication;
- The ng-eNb (Next Generation-eNb also called eLTE-eNB) is an evolution of eNB which supports connectivity to EPC and 5GC.

On another side, we can see that the 5GC communicates with the Data Networks (as IMS and Internet) with UPF entity under N6 Interface.
The application of our research relates to the block in red (DN or Data Network) of the 5G architecture in Figure 1.
In the next paragraphs, we will talk about the future Internet for 5G and about IMS and its architecture.

2.2 5G Internet

The evolution of Internet technologies has converged on an all-IP packet-switched system, which has shaped the way we live, work, learn and play.
It turns out that the engines of the future Internet are all kinds of services and applications, from low bitrates (sensor data and IoT) to higher speeds (high definition video streaming), which must be compatible to support various latencies and different devices.
Thus, the 5G Internet mainly revolves around five main characteristics [1]:

- The Internet of Things: simultaneously interconnect different miniaturized devices such as sensors or not and through different access technologies, for different scenarios and use cases;
- Network reconfiguration and virtualization support: mainly concern SDN (Software Defined Network) and NFV (Network Function Virtualization) technologies;
- Mobility: this is mobile access to services supporting interoperability in the wide deployment of heterogeneous access technologies such as 3G / LTE and WiMAX, as well as WiFi;

- Quality of Service control: defines tools and techniques for predictable, measurable and differentiated quality guarantees for applications according to their characteristics and requirements by guaranteeing sufficient resources (bandwidth) and by controlling delay parameters, jitter and packet loss;
- The resource over-provisioning approach: describes a generic mechanism, capable of integrating SDN and NFV for effective control of resource over-reservation to support differentiated QoS on the 5G Internet.

2.3 IMS architecture

Coming from the specifications of release 5 of 3GPP (Third Generation Partnership Project), IMS is a new architecture standardized by the telecommunications world and based on new concepts, new technologies, new partners and a new ecosystem. It supports real-time (voice, video, conference, etc.) and non-real-time (Push to Talk, Presence, instant messaging, etc.) application sessions on an all-IP network.
IMS provides a multi-service, multi-access, secure and reliable IP network:

- We speak of "multi-services" because all types of services are delivered by a core network supporting different levels of QoS can be offered to the user;
- We speak of "multi-access" because any broadband access network, fixed and mobile can interface with IMS

The IMS architecture can be structured into four important layers as shown in Figure 2 below.

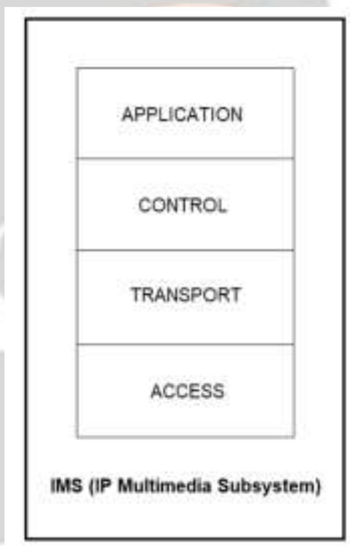

Fig -2: IMS Architecture

- **Access layer:** it represents any broadband access such as: UTRAN (UMTS Terrestrial Radio Access Network) / eUTRAN (evolved UTRAN) / NG-RAN, xDSL, wired network (cable), wireless IP network, WiFi ...

- **Transport layer**: it is an IP network that can integrate QoS mechanisms with MPLS, DiffServ, RSVP, etc. The transport layer therefore consists of “Edge router" at access and "Core router" in transit connected by a transmission network. Different transmission stacks can be considered for the IP network: IP / ATM / SDH, IP / Ethernet, IP / SDH …

- **Control layer**: it consists of session controllers responsible for routing signaling between users and invoking services. These nodes are called CSCFs (Call State Control Function). IMS therefore introduces a session control environment on the packet domain

- **Application layer**: introduces the applications (value-added services) offered to users. The application layer consists of application servers (AS) and MRFs (Multimedia resource function) called IP media servers (IP MS or IP Media Server).

3. FUNCTIONAL ARCHITECTURE OF A ROUTER

3.1 General architecture of a router

Routers are the basic equipment in IP networks. The queue is a central component in the architecture of routers. Indeed, a router is made up of asynchronous processes intended to:

- Assemble the received packets (input queue);
- Check the integrity of the package;
- Determine the destination interface;
- Transmit packets (outgoing queues)

Figure 3 shows the functional block diagram of a router.

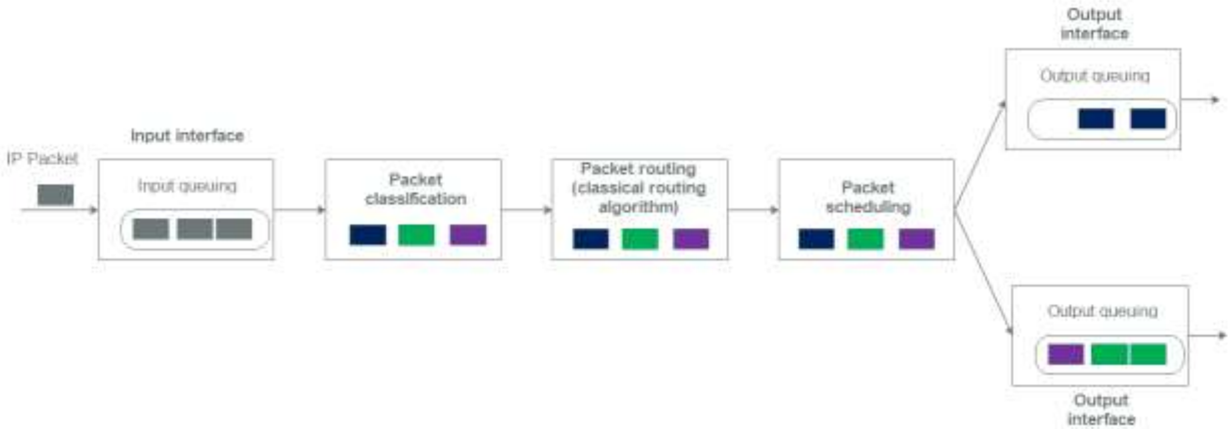

Fig -3: Router architecture

This architecture includes all routing functions and queue management functions to process the routing of IP packet to its destination.

3.2 New proposed architecture

As part of our objective, we first want to propose a new architecture that will allow us to integrate our two optimization approaches. This is shown in Figure 4.

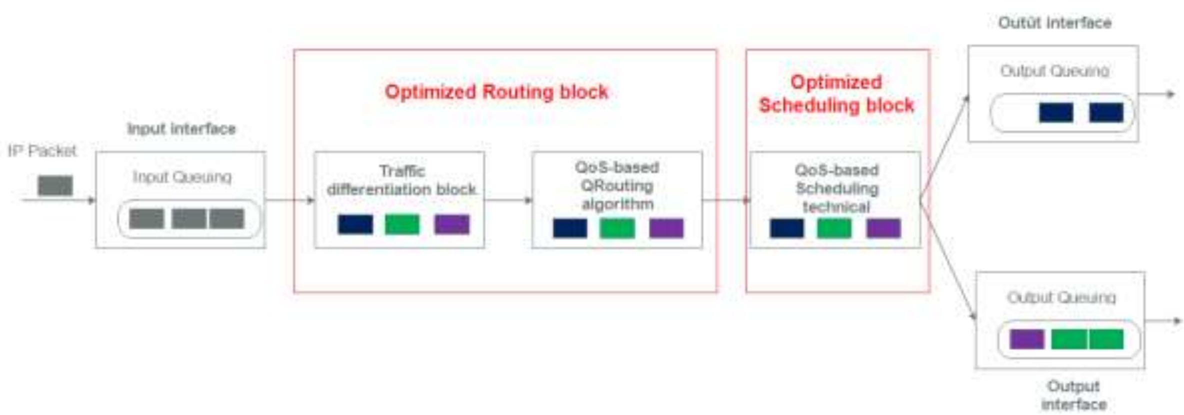

Fig -4: New functional architecture

The optimized blocks are:

- The traffic differentiation block: this block corresponds to the classification function in the classical architecture (in figure 3). However, unlike the latter, this block goes upstream of the path selection, that is to say before the routing block because our goal is to differentiate each type of traffic well before the routing decision;

- The QoS-based Q-Routing algorithm: this is the routing function optimized by our approach based on reinforcement learning combined with a threshold policy on QoS metrics;
- The scheduling block: each differentiated traffic that has been routed is then found in the scheduling block where they will receive processing based on our new QoS-based solution.

3.3 Thresholds for VoIP and Video Performance Metrics

The following two tables (01) and (02) summarize the threshold values of the network parameters that can qualify the performance of two critical applications, which are VoIP and video, on three levels of service: good, medium and bad. Parameters described are transit delay, jitter and packet loss.

Table -1: Thresholds for VoIP performance metrics

Level of service	Good	Medium	Bad
Transit delay	Δ < 150ms	150ms < Δ < 400ms	Δ > 400ms
Jitter	$G < 20ms$	$20ms < G < 50ms$	$G > 50ms$
Packet loss	$P < 1\%$	$1\% < P < 3\%$	$P > 3\%$

Table -2: Thresholds for Video performance metrics

Level of service	Good	Medium	Bad
Transit delay	Δ < 150ms	150ms < Δ < 400ms	Δ > 400ms
Jitter	$G < 20ms$	$20ms < G < 50ms$	$G > 50ms$
Packet loss	$P < 0{,}5\%$	$0{,}5\% < P < 2{,}5\%$	$P > 2{,}5\%$

4. NEW IP ROUTING ALGORITHM BASED ON REINFORCEMENT LEARNING

Reinforcement learning is one of the techniques of machine learning, which is one of the optimization solutions most used by researchers. This concept is also used in several research works in the field of network routing, mainly based on the Q-Learning algorithm.

Indeed, several routing algorithms in networks have been proposed in particular Q-Routing [4] on which several other ideas have been extended such as Predictive Routing (PQ-Routing) or Q-Neural Routing [5] [6] . The goal of this work is to have adaptive traffic control in a dynamically unstable environment (in terms of network load level and network topology).

Also, other routing applications based on reinforcement learning are also studied, such as Adhoc networks, WSN (Wireless Sensor Network), Bluetooth networks, SDN… [7] [8] [9] [10]

4.1 Vision and objective of our routing optimization

In this work, our vision is to differentiate the different types of traffic flow of each application and to respond to the quality of service corresponding to each of them while making the routing decision.

Thus, we try to go beyond the limits of the routing algorithms based on Q-Routing already proposed with regard to the following points:

- Identify the type of traffic before routing,
- Take into account the type of traffic in the routing decision in order to ensure the level of QoS that corresponds to it.
- Define the performance metric for which each type of traffic is the most sensitive and ensure that a certain threshold level of this parameter is not exceeded to have a guaranteed high level of QoS.

- Make the best use of all network resources by considering all possible alternatives in the routing decision and thus balance the various network loads.

4.2 Principle of the routing algorithm

Our algorithm is built on the foundations of the Q-Neural Routing policy [6] (itself an improvement of the Q-Routing algorithm).

Our contribution is about the differentiation of types of traffic and take each QoS requirement into account in the routing algorithm. Indeed, our approach thus aims at the following two main objectives:

- Set up a threshold policy for each type of traffic to respond to the corresponding level of QoS;
- Take into account this threshold policy when making the routing decision

Then, we will study the three main types of traffic present on networks:

- Data represented by applications such as FTP, Web requests, mailing, Telnet, etc. (which are non-real-time applications)
- Voice as in the case of VoIP, VoNR (for IMS in 5G), which is a real-time application
- The video found in videoconferencing, online video games and video on demand (VoD), which is another real-time application.

Table 3 below will summarize the modeling of our routing problem with the concept of Q-Learning.

Table -3: Q-Learning modeling of the routing problem

Q-Learning matching	Network element
Agent	Network node (router)
State	Position of current router *x* in the network at time *t*
Action	Choose from among its neighbors the router with the best path for each type of traffic considered
Reward	Sum of end-to-end transit times to the destination taking into account the threshold policy
Optimal policy to be determined	Q-Routing policy with threshold policy on each node

4.3 Mathematical modeling of the approach

The following figure (05) illustrates the topology of the approach modeling.

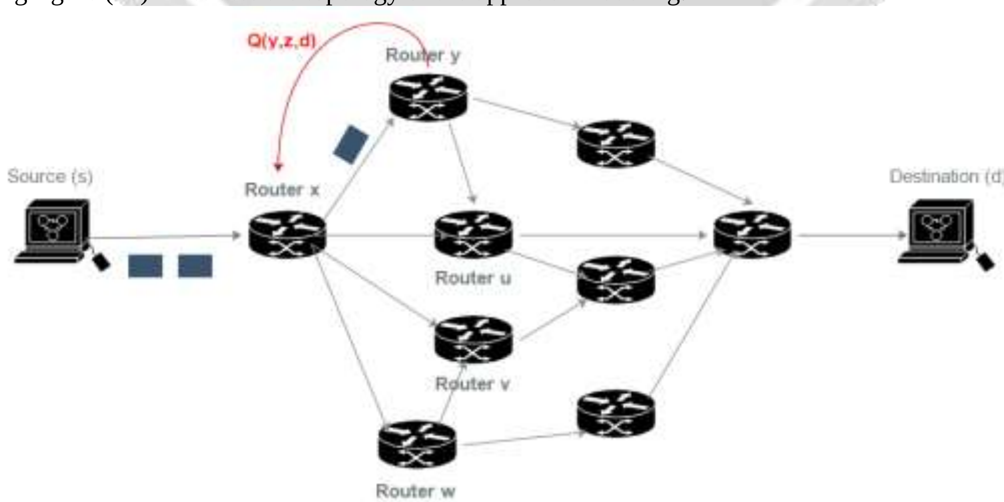

Fig -5: Modeling topology

We will consider the following hypotheses:

- ***P:*** the packet to send
- ***s:*** the source node (source router)
- ***N:*** set of routers in the network
- $\boldsymbol{x} \in \boldsymbol{N}$ **:** the current router directly connected to the router *s*
- ***E:*** set of network states defining the position of the current router
- $\boldsymbol{x(t)} \in \boldsymbol{E}$ **:** state of router ***x*** at time ***t***
- $\boldsymbol{V_x}$ **:** set of neighboring routers of ***x***
- $\boldsymbol{y} \in \boldsymbol{V_x}$**:** ***y*** is a neighbor router chosen by ***x***
- $\boldsymbol{d}$: the destination node (destination router)
- ***A:*** set of possible actions to choose one of the neighbors of the current router
- $\boldsymbol{a_{x_t}(y, d)} \in \boldsymbol{A}$ **:** action of choosing the neighbor ***y*** to go to the destination ***d*** in the state $\boldsymbol{x_t}$
- $\boldsymbol{w_x}$ **:** the waiting time in the queue of any router ***x***
- $\boldsymbol{\tau_{xy}}$ **:** the transmission time of the packet ***P*** between two adjacent nodes ***x*** and **y**
- $\boldsymbol{R}$**:** set of rewards returned for each chosen route
- $\boldsymbol{r(x, y, d)} \in \boldsymbol{R}$ **:** reward obtained by ***x*** from its neighbor ***y*** by choosing the latter to go to the destination ***d***
- $\boldsymbol{Q(x, y, d)}$**:** value function called "Q-value" of a router ***x*** to go to destination ***d*** through router ***y*** and it designates the end-to-end routing time from ***x*** to ***d***

Following the reinforcement learning and Q-Routing principles, each reward is defined by the following formula [4]:

$$r(x, y, d) = w_y + \tau_{xy} + T_y \qquad \textbf{(1)}$$

With:

$$T_y = \min_{z \in V_y} Q(y, z, d) \qquad \textbf{(2)}$$

The estimate of Q-value is given from the following formula [4]:

$$Q(x, y, d) = Q(x, y, d) + \eta\{r(x, y, d) - Q(x, y, d)\} \qquad \textbf{(3)}$$

Where η ∈] 0.1 [, represents the learning rate, with a constant value throughout the various routing decisions

4.3.1 Threshold policy

The principle of our approach is to use a specific policy that we will call "Threshold Policy". This is a strategy to ensure that the QoS metric threshold is not exceeded so that the performance of the application under consideration is reached.

We will define this threshold according to the type of traffic and according to their need in terms of expected level of quality.

Depending on the three types of traffic we consider, we will define three different threshold policies to be taken into account during the routing decision:

- The threshold policy defined for the voice, $\boldsymbol{\pi_{\Delta_V}}$ considering the delay metric $\boldsymbol{\Delta_V}$;
- The threshold policy defined for the video, $\boldsymbol{\pi_P}$ considering the packet loss metric **P**;
- The threshold policy defined for the data, $\boldsymbol{\pi_{\Delta_D}}$ considering the delay metric $\boldsymbol{\Delta_D}$.

4.3.2 Routing policy

Let's pose:

- $\boldsymbol{\Pi}$: set of all possible routing policies;

- $\boldsymbol{\pi} \in \boldsymbol{\Pi}$: is the routing policy that we will adopt in our new algorithm by checking the threshold policy; it is defined by:

$$\begin{aligned} \pi &: E \to A \\ x_t &\to a_{x_t}(y,d) \big| \pi_\Delta or\ \pi_P \end{aligned} \quad \textbf{(4)}$$

- $\boldsymbol{Q^{\pi}}$: denotes the value function obtained by adopting the policy π, defined by:

$$\begin{aligned} Q^\pi &: (A,\Pi) \to R \\ (a_{x_t}(y,d),\pi) &\to Q^\pi(x_t,y,d) \end{aligned} \quad \textbf{(5)}$$

Taking into account the threshold policy, the optimal routing policy noted $\boldsymbol{\pi}^*$ is the one that minimizes the value of $\boldsymbol{Q^{\pi}}$. It is defined by the following formula:

$$\pi^* = \min_{y \in V_x} \{Q^\pi(x,y,d)\} \quad \textbf{(6)}$$

To set up our algorithm, we will consider four types of packet denoted T (P):

- T (P) = 0: this is a reinforcement packet
- T (P) = 1: this is a data packet
- T (P) = 2: this is a voice packet
- T (P) = 3: this is a video packet

Figure 6 shows the details of this new routing policy which is the very essence of our work

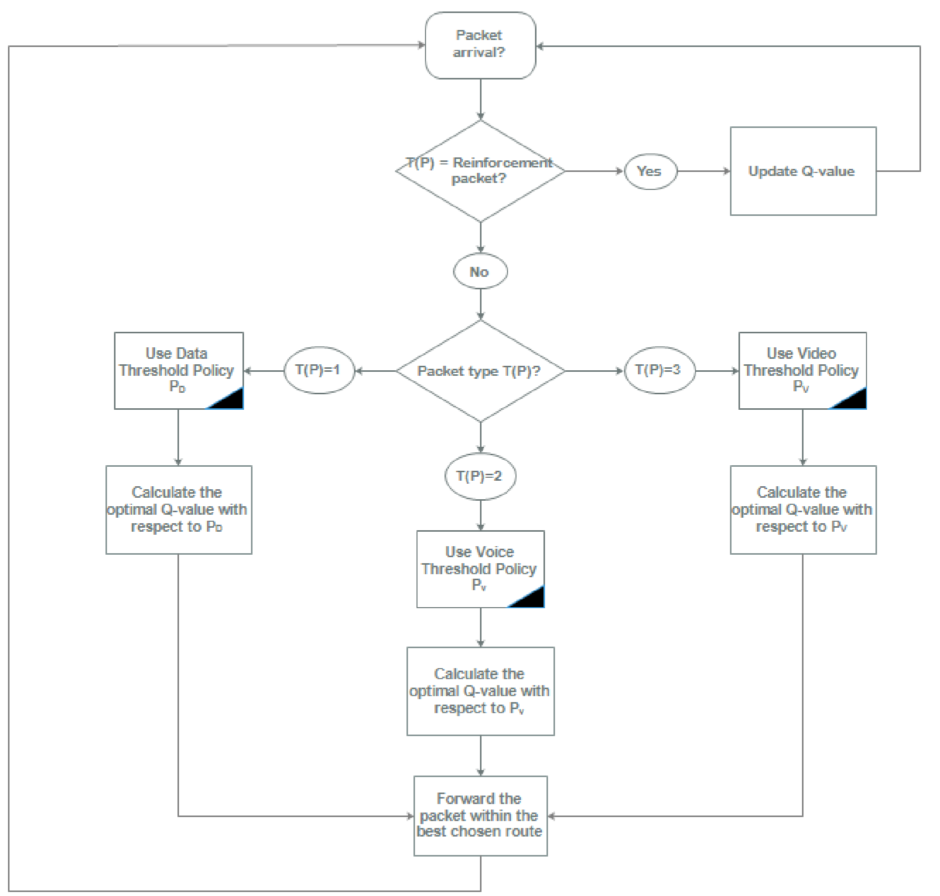

Fig -6: New routing policy

4.4 Presentation of the routing algorithm

Figure 7 illustrates the general process of our new routing algorithm from receiving a packet at each router in the network.

It includes the step of transmitting packets to the corresponding neighbor, the step of reinforcement by sending the reinforcement signal to the source router and finally the step of updating the Q-values as well as the contents of router's Q-table.

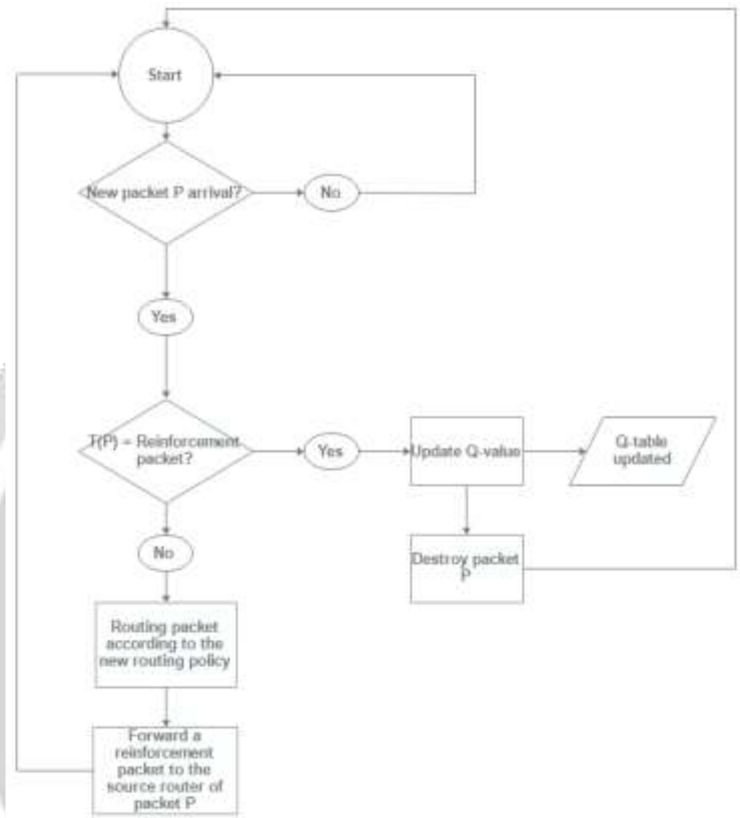

Fig -7: General description of our algorithm

4.5 Results of the routing algorithm applied in IMS over 5G architecture

Our research work consists in applying our new routing algorithm for better network performance of the IMS, at the transport layer of the architecture.

Figure 8 illustrates all the changes to be made to the IMS architecture with our optimization approach.

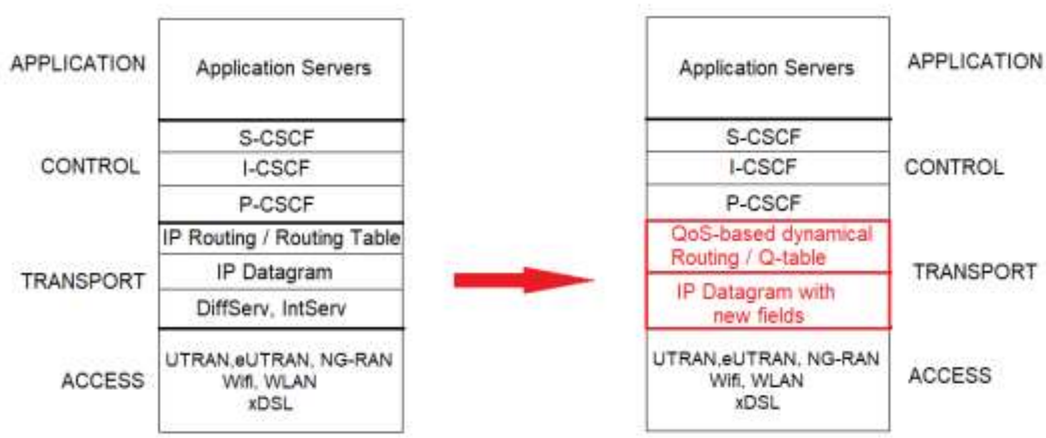

Fig -8: Optimized IMS architecture at the transport layer

To validate and analyze the performance of our new routing algorithm, we will consider three main traffics: FTP application for data, VoNR for voice and video.
Then, we have implemented the following four scenarios:

- analyze the general performance of the solution in a context of a normal functional network in terms of the quality of service of the various applications;
- check the tolerance of the algorithm against failures in the network;
- check the tolerance of the algorithm with respect to delays in the network;
- compare the performance of our algorithm with respect to existing routing protocols, in particular RIP and OSPF.

We will consider the topology in the figure 9 below to our simulation.

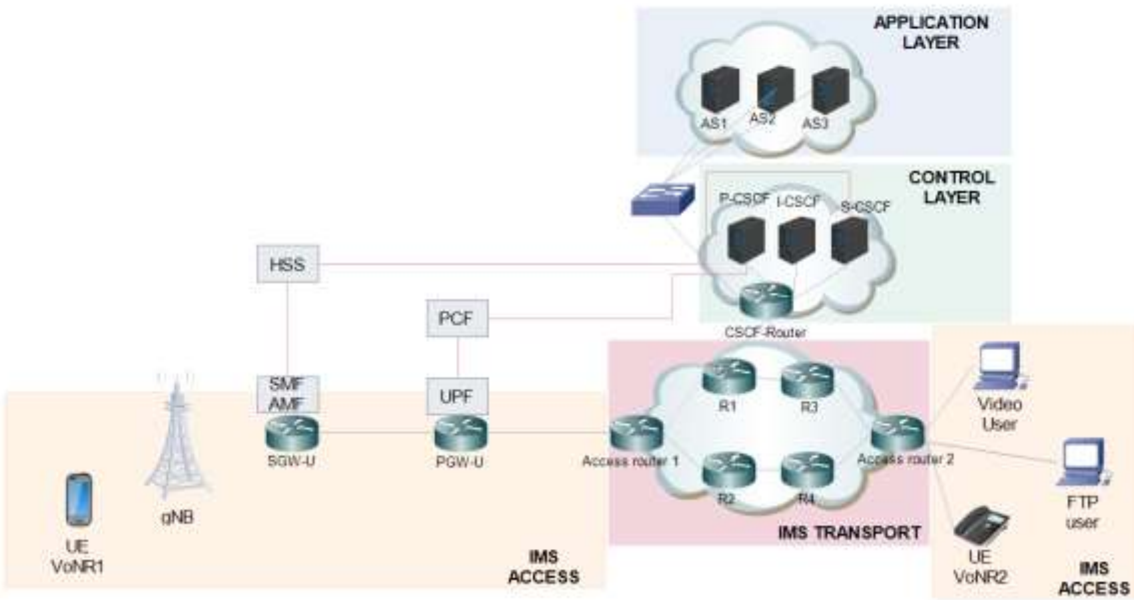

Fig -9: IMS/5G Architecture

4.5.1 General performances of the routing algorithm

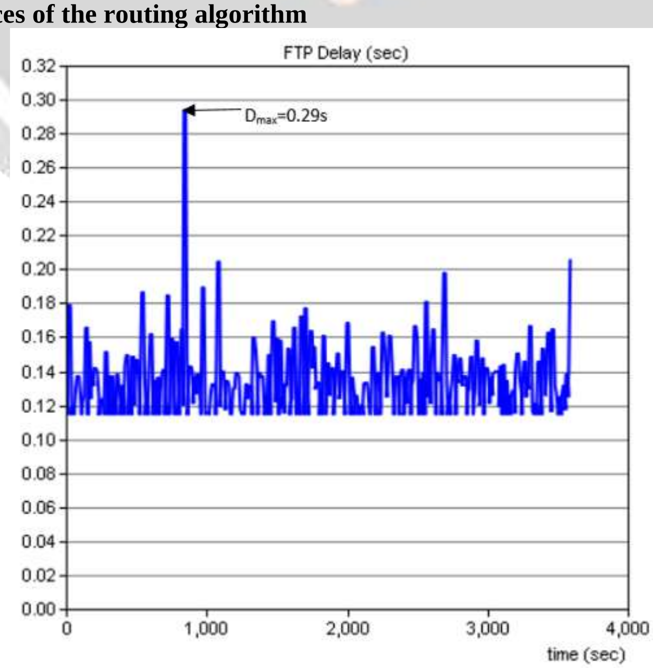

Fig -9: FTP delay

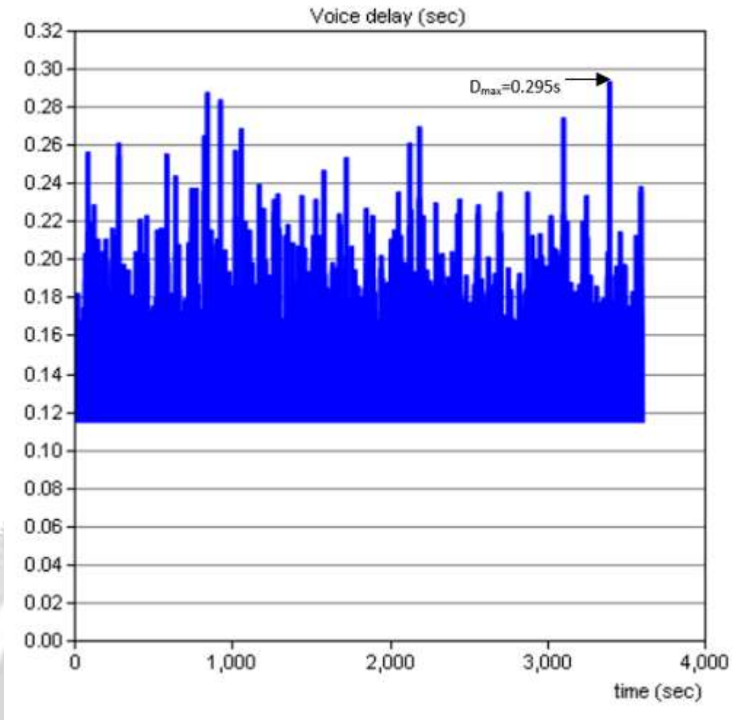

Fig -10: Voice delay

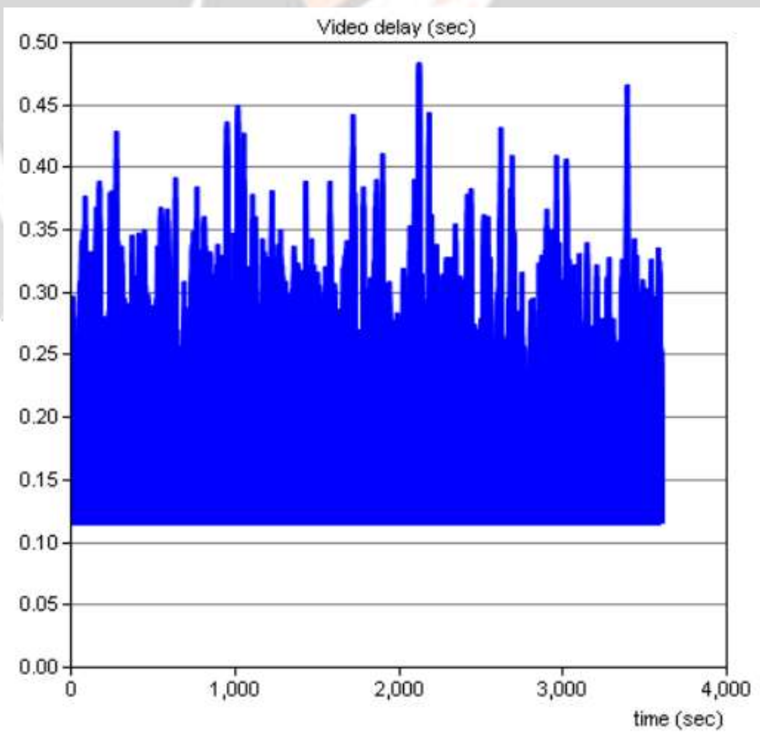

Fig -11: Video delay

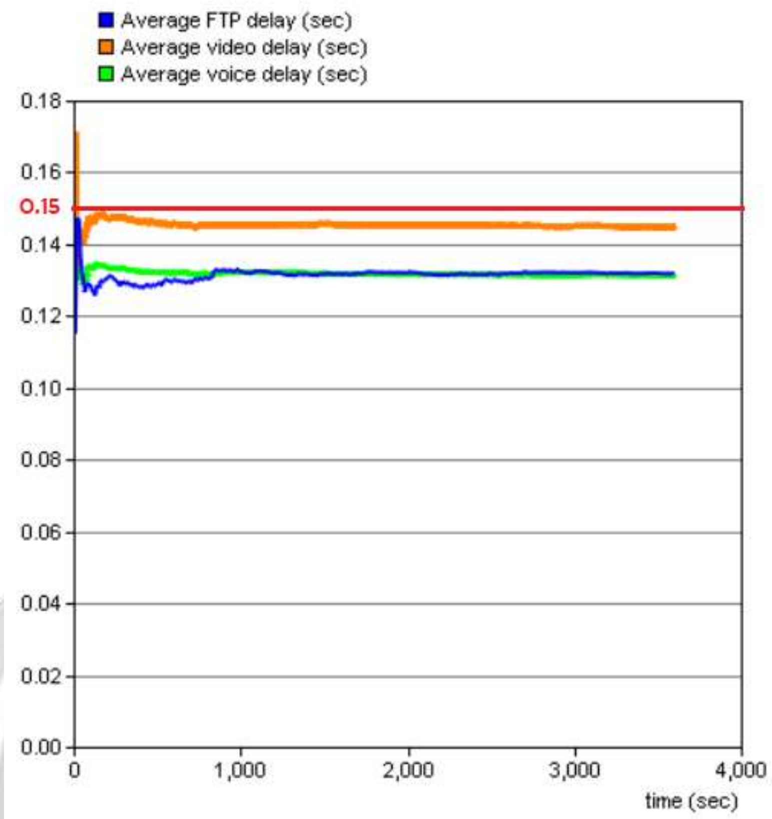

Fig -12: Average delay for FTP, voice and video packets

- Figures 9, 10 and 11 show us the delay results respectively for FTP, voice and video applications using our new routing algorithm. For FTP and voice results, we can see that the maximum delay obtained is approximately 0.29s (=290ms). If we refer to the thresholds for VoIP performance metrics in table 1, we can say that it doesn't exceed the threshold equal to 400ms that defines a bad level of service. And for FTP which is not demanding in terms of transit delay, we can say that the quality of service is more than satisfactory. In another hand, in the figure 11, video packet delays are in an interval of values between 0.12s (=120 ms) and 0.48s (= 480ms). We can also say that, overall, the quality of the video application is satisfactory compared to the thresholds in the table 2.

- In addition, Figure 12 illustrates the average delay values of the three FTP, voice and video applications within our network. We can see that the results are all below the minimum threshold (= 150ms) to ensure a good level of service quality for real-time applications such as voice and video.

4.5.2 Network failures tolerance results

To do this, we will present the load rate of each of the two links "Router_acc_1> R1" and "Router_acc_1> R2" presented in the topology of Figure 9 in both cases before and after a failure of the link through R2.

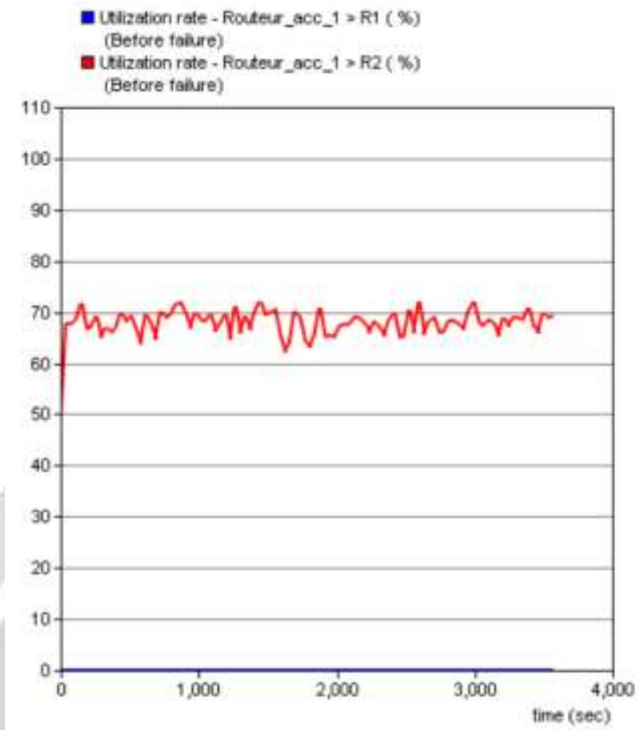

Fig -13: Utilization rate of routes to R1 and R2 before failure

During the simulation, we can observe in figure 13 that, before the failure, our algorithm chose the route passing the router R2 for the routing of the packets, i.e. with a load rate of about 70%. The path passing through router R1 is not used.

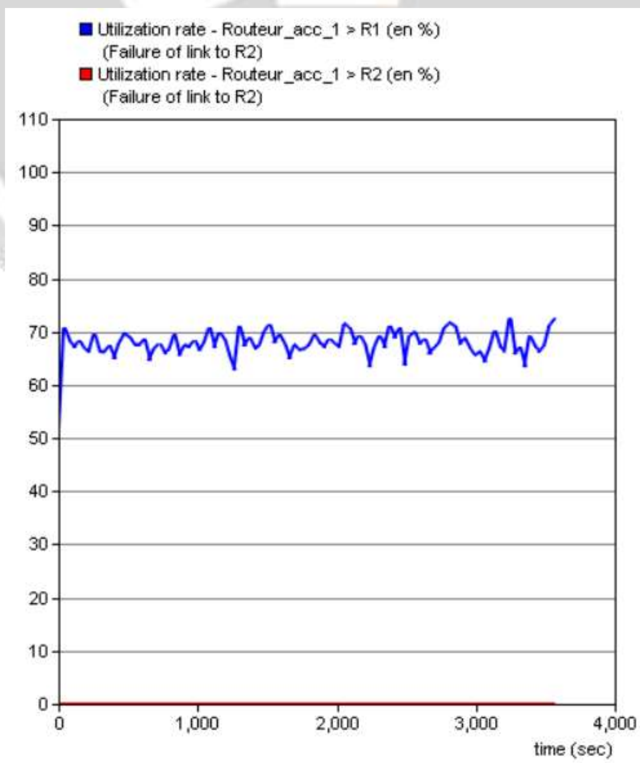

Fig -14: Utilization rate of routes to R1 and R2 after failure

Then, when we introduced a failure at the level of the link between the access router 1 and the router 2, we can say from figure 14 that the traffics are automatically routed to the path passing through the router R1.
We have a utilization rate equal to 70% too. So, this result also means that there was no loss of packet.

In other words, these results demonstrate that our routing algorithm is reactive to the various failures that may arise within the network by looking for another alternative route when one exists.

4.5.3 Network delay tolerance results

During this scenario, we will configure a delay ***D = 1.4s*** on the main link passing through router R2 which will affect the voice and the FTP data.

It is to know that the thresholds that we have set in our algorithm are respectively ***1s*** and ***0.4s*** for FTP and voice.
We will then compare the delays obtained for each of the two traffics, in the three cases before and after adding the delay (with and without our threshold policy)

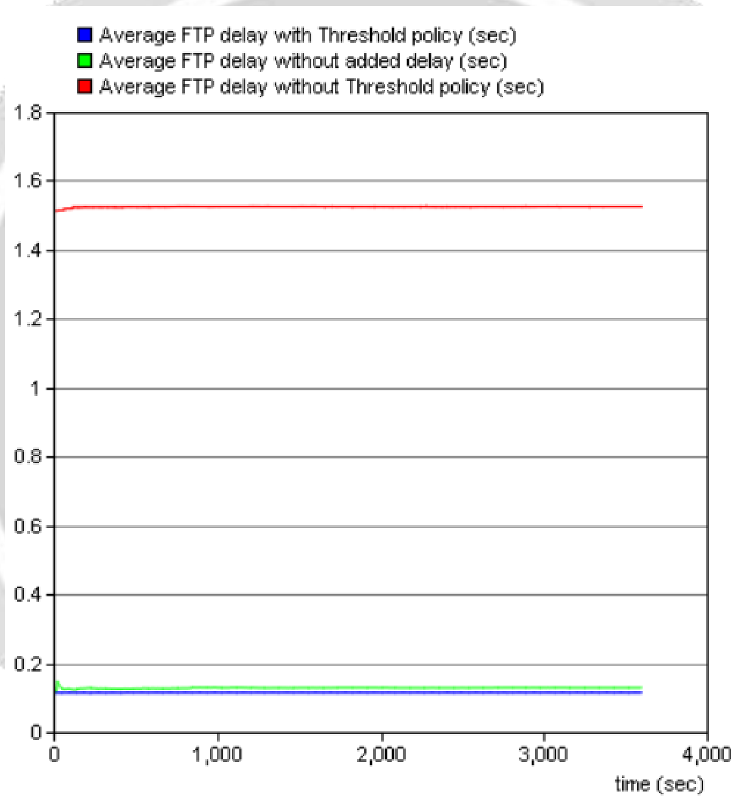

Fig -15: Average FTP delay before and after latency (D =1.4s)

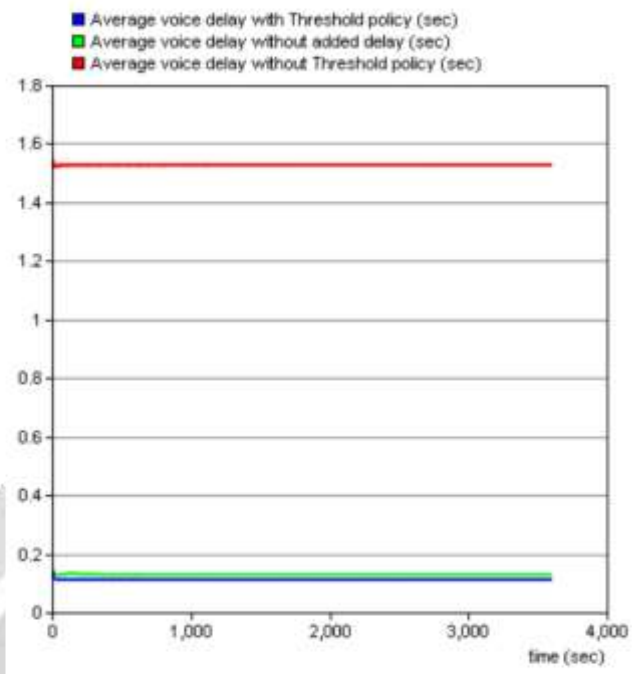

Fig -16: Average voice delay before and after latency (D =1.4s)

From these figures 15 and 16, we can quickly notice that without the threshold policy, the FTP delay and the voice delay largely exceed the respective minimum set thresholds 1s and 0.4s.
The green curves represent the values in figure 12 where no delay has been added. In these results, we can say that with our threshold policy (blue curves), delays after adding latency are quite lower than previously observed delays (green curves).

This can be explained by the fact that our routing algorithm transferred the route for the FTP and voice packets to the one that goes through the router R1 when it detected that the thresholds were exceeded but it left the main route (going through R2) to video packets for which the fixed threshold is not affected.

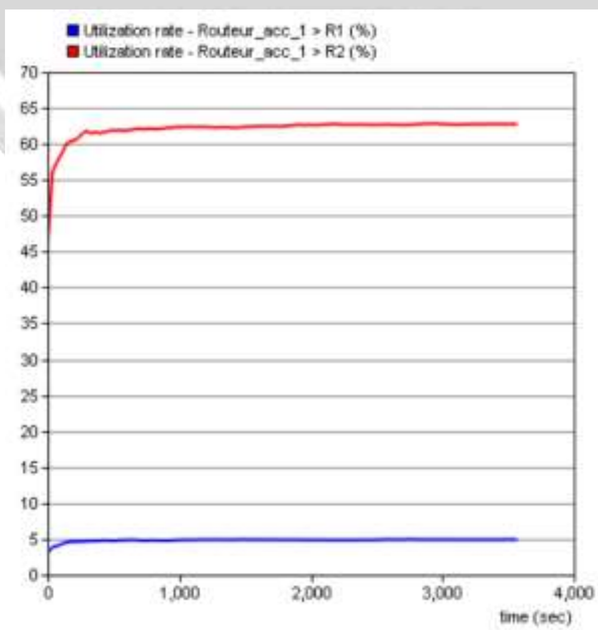

Fig -17: Utilization rate of links through R1 and R2 after latency (D =1.4s)

The figure 17 shows the load on the links passing respectively through R1 and R2 during our scenario which will verify the previous comments.
Indeed, we can see that the route through the previously unused router R1 (see figure 13) is now responsible for about 5% of the routing of the FTP and voice packets to this route. The load passing through router R2 has decreased compared to that in figure 13, i.e. at an average rate of 65% for video packets.

4.5.4 Performance comparison results against RIP and OSPF routing protocols

In this scenario, we will compare the performance (in term of delay) of our approach compared to the two routing protocols RIP and OSPF on the three previous traffics, in a context of normal functional network then in a context of network having a significant latency for the voice.

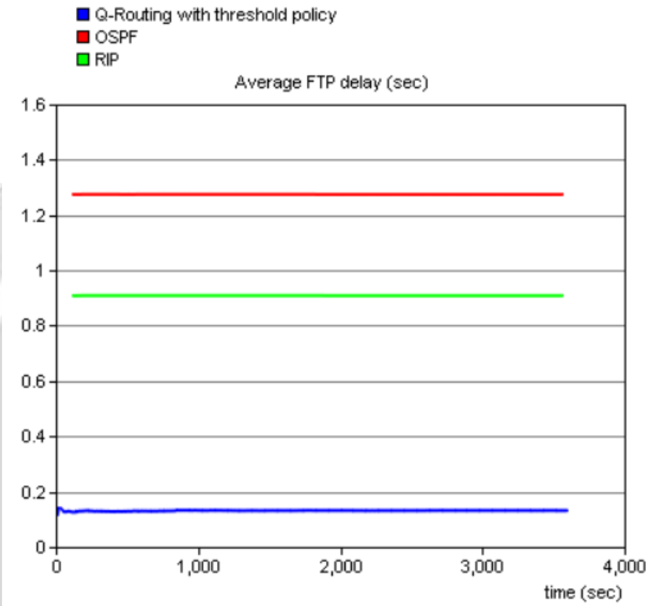

Fig -18: Comparison of average FTP delay in a normal network

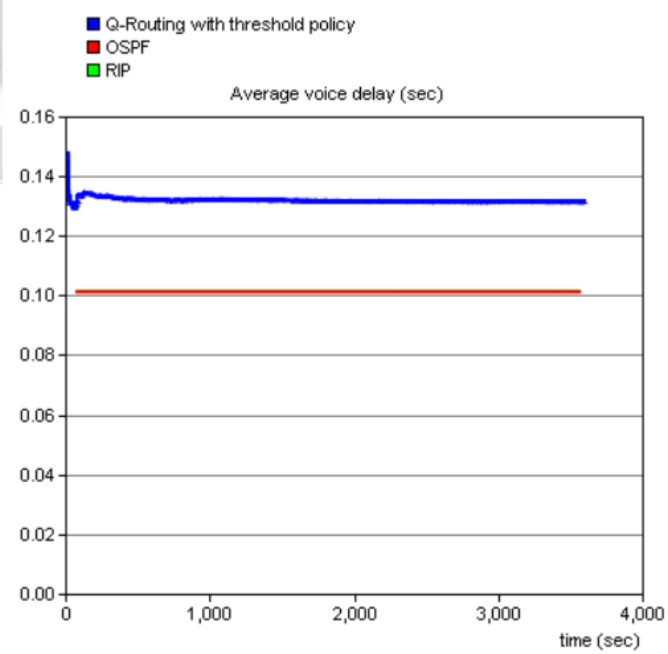

Fig -19: Comparison of average voice delay in a normal network

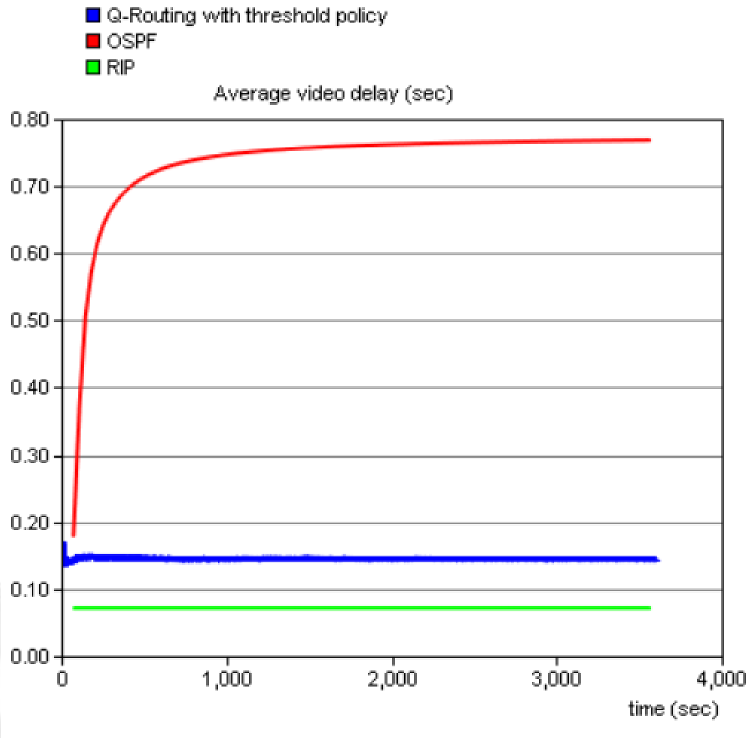

Fig -20: Comparison of average video delay in a normal network

- In the FTP results in figure 18 we observe the difference in delays between the three algorithms such as $D_{RIP} \approx 0.9s$, $D_{OSPF} \approx 1.3s$ while with our algorithm the average delay obtained is 0.13s. This shows us a better performance of our algorithm compared to the two protocols RIP and OSPF.

- For the voice traffic in figure 19, we can see that the delay with our algorithm ($D \approx 0.13s$) is slightly higher than the delays with RIP and OSPF ($D_{RIP} \approx D_{OSPF} \approx 0.10s$) either with a difference of 0.03s. However, this difference is not significant compared to the quality of the voice since the delay fits well in the table of threshold values (table 1)

- In figure 20 for video traffic, we have $D_{RIP} \approx 0.08s$, $D_{OSPF} \approx 0.77s$ and with our solution $D \approx 0.15s$. If the routing with RIP is the best for the video, our algorithm is placed in second place with a difference of 0.07s. However, both algorithms largely satisfy the requirement for good QoS in reference of Table 2.

In the following scenario, we are mainly interested in voice traffic which is the most sensitive in terms of latency. Thus, the results in Figure 21 below are in a latency-added network context. We have injected into the link between the access router Router_acc_1 and the router R2 a latency of 1.4s.

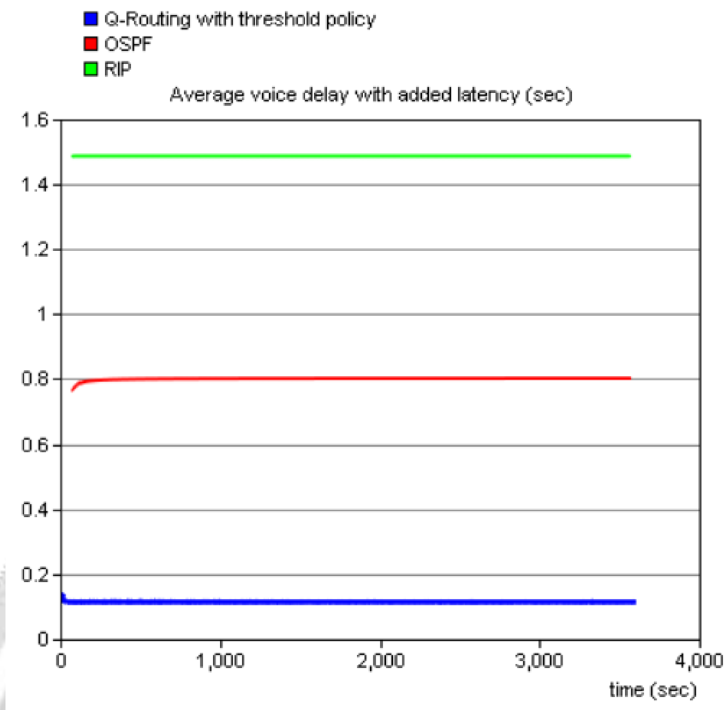

Fig -21: Comparison of average voice delay in a network with added latency (D =1.4s)

In the figure 21, we have the following results: $D_{RIP} \approx 1.5$s, $D_{OSPF} \approx 0.8$s and for our solution $D \approx 0.11$s.
From these results we can say that our algorithm with threshold policy largely tolerates any latency encountered in the network compared to RIP and OSPF while satisfying the quality of voice service.

Although with a rather high delay of 0.8s, we can still say that the OSPF algorithm reacts much more than that RIP as we can see the big difference between the two delays. Indeed, the metric used by RIP during the routing decision is based on the number of nodes traveling the path to the destination. Although the delay on this route has increased, the number of nodes remains unchanged which explains why the same decision routing has been taken. As for OSPF, the routing decision is based on the bandwidth unlike our solution which is based on the state of the network influencing the performance parameters of the various traffics.

5. NEW PACKET SCHEDULING TECHNIQUE BASED ON PQ AND WFQ

5.1 Vision and objective of the approach

The development of scheduling techniques aims to control the resources sharing, to isolate classes of service and to reduce the waiting time of packets in queues.
The problem lies in the fact that although several techniques have been proposed, each principle can be beneficial for a certain type of traffic but penalizing for others.
Hence, we will always consider the three traffics classified respectively by higher priority level as follows: voice, video and data.

Our main goal is to find a scheduling solution that would handle all types of traffic satisfactorily, i.e.:

- Ensure low latency for voice traffic
- Ensure a low loss rate and a fairly low latency for video traffic
- Ensure the transmission of data traffic in an appropriate and fair manner, with no particular requirements in terms of delay or loss.

To achieve these goals, we were inspired by the different advantages and disadvantages of the two schedulers PQ (Priority Queuing) and WFQ (Weighted Fair Queuing).

5.1.1 Priority Queuing (PQ) Scheduler

Priority Queuing or PQ scheduling technology ensures fast processing of time-demanding service packets. Figure 22 below describes the principle of PQ.

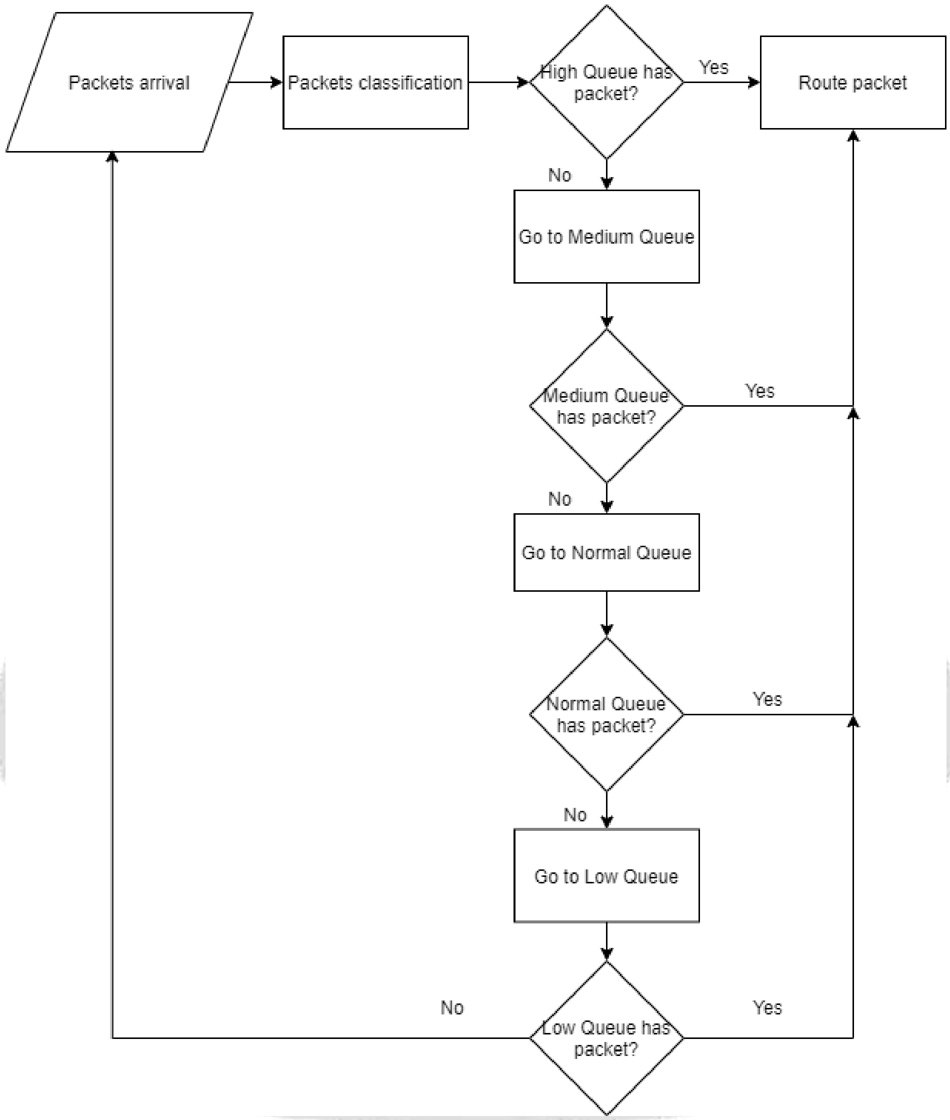

Fig -22: Principle of PQ scheduler

PQ's motivation is found in the low packetization time (time required for the codec to assemble the samples that will form a packet). However, PQ involves degradation in performance. There may be a problem when the high priority traffic is very heavy; indeed, normal packets may be rejected due to the low priority queue size exceeded.

5.1.2 Weighted Fair Queuing (WFQ) Scheduler

Figure 23 shows the WFQ scheduling technique.

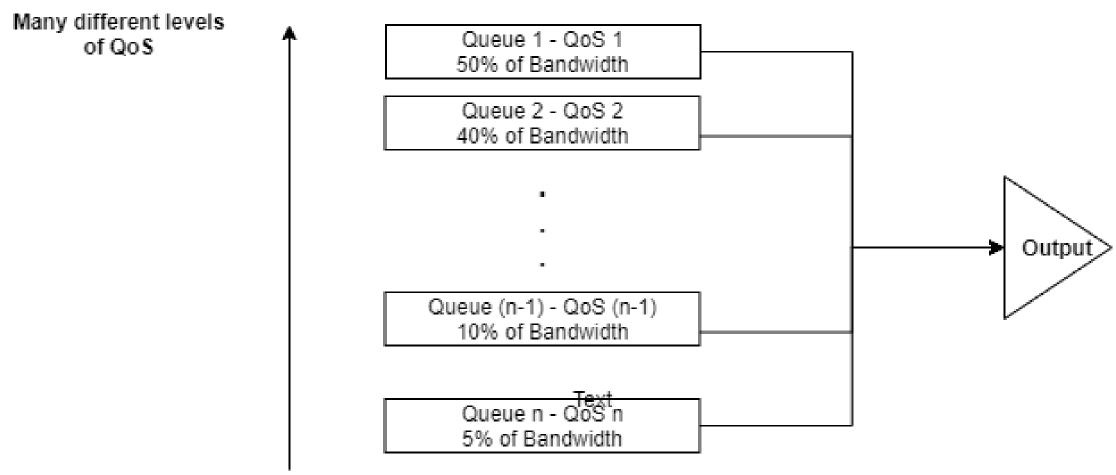

Fig -23: Principle of WFQ scheduler

The WFQ scheduler helps achieve a degree of fairness between different applications. It is possible to obtain a bandwidth per connection as well as time limits. Also, the gusts are smoothed out and there is no need to do any upstream traffic control.

Its weakness lies in the complexity of the stamp calculations for each packet and the small time limits require a large reserve of bandwidth. It is necessary to have a state per connection and it is necessary to classify the packets before serving them

5.2 Presentation and principle of the new scheduling approach

Our approach consists in putting in stages the two techniques of PQ and WFQ in order to better process our applications. Indeed, to meet the requirements in terms of delay we will use PQ to process priority voice and video traffic. In addition, although there has been a starvation for video packets, we have also seen that PQ provides a low rate of packet loss. However, this loss parameter is the one that is most considered to have a high level of quality of service for videos. Thus, this advantage must be taken into account.

In order to overcome the problem of starvation, to prevent voice traffic from monopolizing processing, we are going to put upstream of one of the PQ queues a WFQ scheduler specifically the WF^2Q + version to derive its advantages for our two real time applications (voice and video).

Our PQ scheduler will therefore have two queues:

- The first high priority queue for voice and video traffic
- The second low priority queue for data traffic.

Regarding the architecture of WF^2Q +, we will also have two queues, one for voice and the other for video.

Since voice is classified as priority, we will set a higher weighting to voice packets compared to video packets.

In detail, on leaving the routing block, the various packet streams are routed to an output interface. Each egress interface has one or more queues. In our case, we will have four queues and two schedulers staged. The voice and video traffic will be placed in the two queues of the WFQ scheduler and the data traffic will be placed in the low priority queue of the PQ scheduler.

At WFQ level, weighted scheduling between voice and video traffic will be performed so as not to monopolize high priority voice traffic. Thus, they each arrive equally at the high priority queue of PQ.

At the PQ level, voice and video traffic will therefore be optimally prioritized with respect to data traffic, which allows them to be processed quickly with a minimum delay and loss rate.

The principle of our new scheduling solution is described in figure (24).

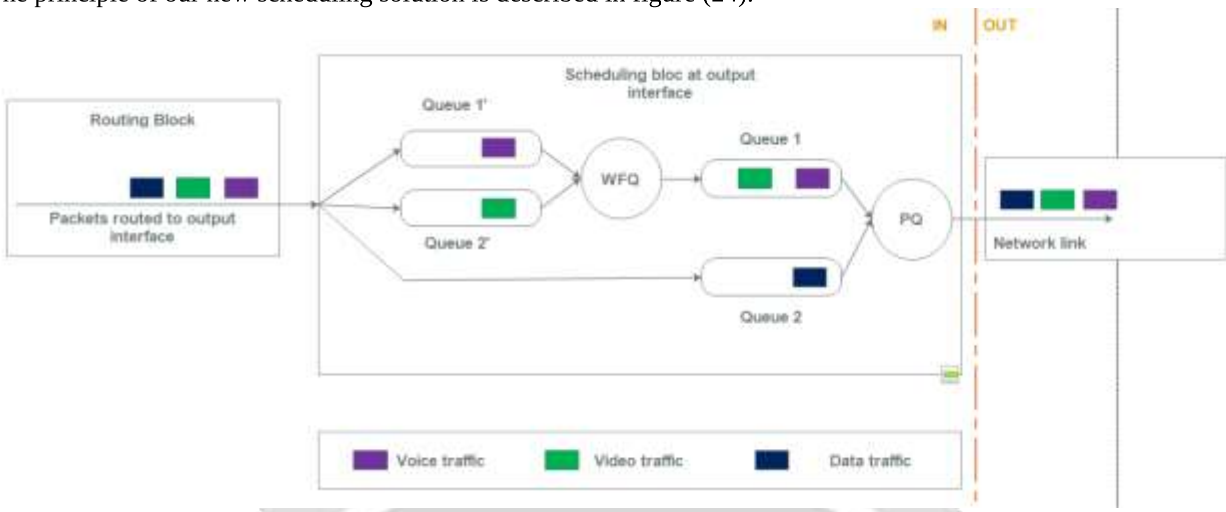

Fig -24: Principle of our optimized scheduling technique

5.3 Results simulations on optimization of the scheduling technique applied to the 5G Internet

To verify and analyze the performance of our new scheduling technique, we will compare our solution with the existing schedulers which are FIFO, PQ and WFQ.

We will take into account the following output parameters:

- the average end-to-end times for voice and video traffic
- the number of packets received for each application

5.3.1 Comparison of real time applications average delay

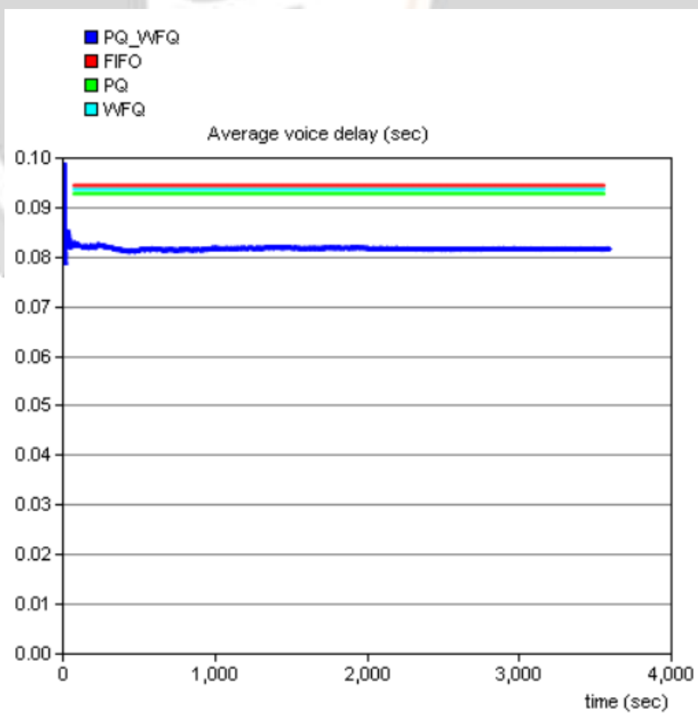

Fig -25: Average voice delay

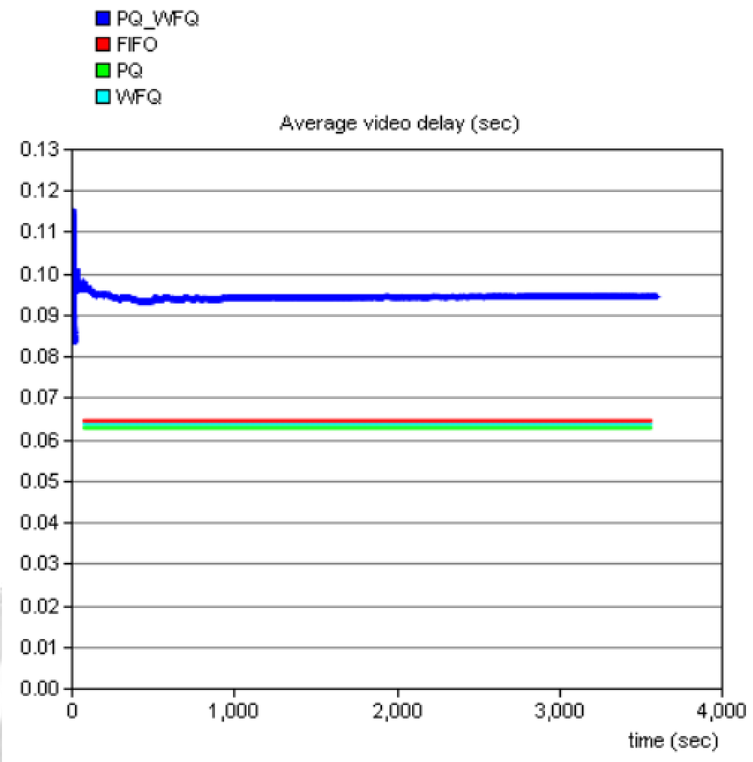

Fig -26: Average video delay

- For the voice packets in figure 25, we can observe that the delays with the three schedulers FIFO, PQ and WFQ are appreciably equal being $D_{FIFO} \approx D_{PQ} \approx D_{WFQ} \approx 0.093$s, whereas with the combination of PQ and WFQ the average delay obtained is about 0.081s. This shows us that for the VoIP application, our solution presents a better performance of our algorithm compared to existing schedulers, although all the curves offer an excellent level of quality of service, according to table 1.

- As for the video packets, the delay obtained with our technique observed in figure 26 is approximately 0.095s and presents a difference of approximately 0.03s more than the delays with the other schedulers which are $D_{FIFO} \approx D_{PQ} \approx D_{WFQ} \approx 0.065$s. However, the difference is slight and our result nevertheless fulfills the conditions for better video quality by referring to table 2.

- From these two figures, we can then say that the solution of combining the two techniques PQ and WFQ improves even more the quality of the voice which we put in high priority and also ensures a high level of quality for the video, although this is in medium priority. In addition, the voice delay ($D_{voice} \approx 0.081$s) and the video delay ($D_{video} \approx 0.095$s)) obtained differ only slightly, verifying our objective to have more equity between the two applications.

5.3.2 Comparison of received packets for each application

The received packets parameter is needed to determinate the number of packets loss when the router is forwarding them into the destination.

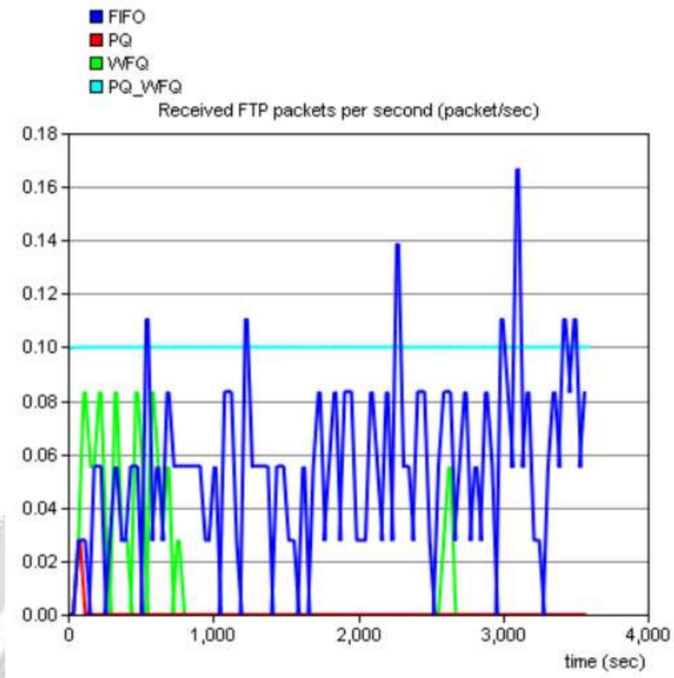

Fig -27: Received FTP packets

From the curves in Figure 27, we can observe that the PQ scheduler (curve in red) lets almost no FTP packets through because this application has been classified in the queue with the lowest priority. The WFQ scheduler causes it to pass some FTP traffic (fairly low WFQ weight) over time because in the meantime the voice and video traffic are transmitted with greater weight. Then, we can see that the FTP packets are regularly transmitted with the FIFO scheduler because if they are the first to arrive they are the first to leave without distinction of their class. Finally, for our solution PQ_WFQ we can notice that the FTP traffics are constantly received due to the fact that although they are of low priority, they are still transmitted because the system lets them pass at a low rate so as not to dominate facing the other two applications but without being totally neglected.

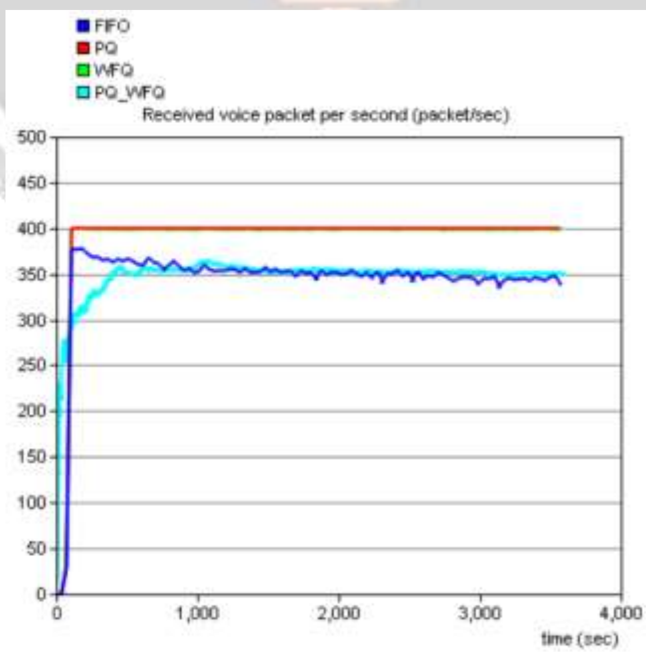

Fig -28: Received voice packets

In figure 28, we can observe that the voice packets received with the PQ and WFQ schedulers are the most numerous and are approximately equal to $N_{PQ} \approx N_{WFQ} \approx 400$paquets / sec. This can be explained by the fact that with PQ the voice is transmitted as a priority as long as the corresponding queue is not empty. As for WFQ, the weight assigned to voice traffic is much greater than that of video and FTP. Then there is our curve (in cyan blue) which is slightly higher than with FIFO scheduler with $N_{(PQ+WFQ)} \approx 350$paquets/sec and $N_{FIFO} \approx 349$paquets/sec. Our solution therefore obtains the advantages offered by the combination of the two techniques PQ and WFQ.

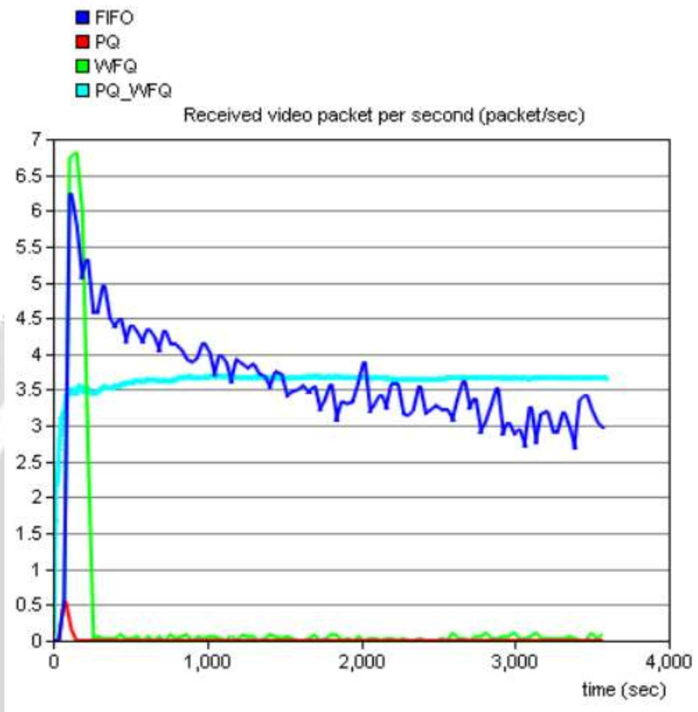

Fig -29: Received video packets

From the curves of this figure 29 we can say that the video packets are transmitted with the FIFO, WFQ and PQ_WFQ schedulers while with PQ it is almost null because the video queue passes after the voice queue is processed. However we can notice, that the curve with our solution is better because it is more constant. Indeed, the principle of our algorithm which allows video traffic to pass with fairness with voice traffic. Hence, its performance is better than others.

6. CONCLUSION

In this work we have proposed two main approaches for optimized 5G IP networks.
The first approach is at the routing level, by proposing a new adaptive routing algorithm based on the reinforcement learning technique, actually on the principle of Q-Routing and its derivatives. We have given a new extension of this algorithm by defining the notion of threshold policy.
The simulation results obtained showed us that the Q-Routing technique combined with our threshold policy has an advantage over conventional RIP and OSPF routing protocols in terms of quality of service for any type of application, but also in terms of terms of reactivity and adaptation of the network in case of possible failure or latency events.
The second approach is to achieve better management and better processing in the scheduling of packets. It then consists of putting in stages the two schedulers WFQ and PQ. The choice of this combination aims to benefit from the advantages of both techniques; and to compensate for the disadvantage of one by using the principle of the other. The comparison between our solution and FIFO, PQ and WFQ schedulers has demonstrated that the results obtained were largely satisfactory in terms of delay and fairness between the different applications.

7. REFERENCES

[1]. J. Rodriguez, « Fundamentals of 5G mobile networks », John Wiley and Sons, 2015
[2]. Y. Wu, H. Huang, C. Wang, Y. Pan, « 5G Enabled Internet of Thing », CRC Press, 2019
[3]. E. Dahlman, S. Parvall, J. Sköld, « 5G NR: The Next Generation Wireless Access », Academic Press, 2018
[4]. A. Boyan, M. L. Littman « Packet Routing in Dynamically Changing Networks: A reinforcement learning approach », Cognitive Science Research Group Bellcore, 1994
[5]. S. P. M. Choi, D.Y. Chang, « Predictive Q-Routing: A Memory-based Reinforcement Learning Approach to Adaptive Traffic Control», Department of Computer Science, University of Science and Technology, Hong Kong, 1996
[6]. S. Hoceini « Techniques d'Apprentissage par Renforcement pour le Routage Adaptatif dans les Réseaux de Télécommunication à trafic irrégulier », PhD Thesis, 2004
[7]. Z. Mammeri, « RL-based Routing in Network: Review and classification of approaches », In Open Access Journal, IEEE Access, 2019
[8]. CY. Fan, CC. Hsu, WY. Wang, « An innovative Routing Algorithm with Reinforcement Learning and Pattern Tree Adjustment for Wireless Sensor Network », In Pan JS, Chen SM., Nguyen N.T. (eds), Computational Collective Intelligence, Technologies and Applications, Lecture Notes in Computer Science, Vol 6423, Springer, ICCCI, 2010
[9]. R. Olsson, J. Egeland « Reinforcement Learning Routing Algorithm for Bluetooth Mesh Networks », Thesis, KTH School of Electrical Engineering and Computer Science, Stockholm, 2018
[10]. C. Fang, C. Cheng, Z. Tang, C. Li « Research on Routing Algorithm Based on Reinforcement Learning in SDN », in Journal of Physics: Conf. Ser 1284012053, 2019