The Optimization of IPTV Service Through SDN In A MEC Architecture, Respective Simulations In OMNeT ++

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Abstract

The aim of this paper is to present the 'Power' of SDN Technology and MEC Technic in improving the delivering of IPTV Service. Those days, the IPTV end –users are tremendous increased all over the world , but in the same time also the complains for receiving these prepaid real time multimedial services like; high latency, high bandwidth, low performance and low QoE/QoS. On the other end, IPTV Distributors need a new system, technics, network solutions to distribute content continuesly and simultaneously to all active end-users with high-quality, low-latency and high Performance, thus monitoring and re-configuring this 'Big Data' require high Bandwidth by causing difficult problems by offering it affecting in the same time the price and QoE/QoSperformance of delivered service.

For this reason, we have achieved to optimize the IPTV service by applying SDN solution in a MEC Architecture (Multiple-Access Edge Computing). In this way , through MEC Technology and SDN, it is possible to receive an IPTV service with Low Latency, High Performance and Low Bandwidth by solving successfully all the problems faced by the actual IPTV Operators. These improvements of delivering IPTV service through MEC will be demonstrated by using the OMNet +++ simulator in an LTE-A mobile network. The results show clearly that by applying the MEC technique in the LTE-A network for receiving IPTV Service through SDN Network, the service was delivered with latency decreased by >90% (compared to the cases when the MEC technique is not applied), with PacketLoss of almost 0 and with high performance QoE. In addition these strong Contributions, the 'Big' innovation achieved in this work through simulations is that the quality of delivered IPTV Service did not change according to the increasing of the end-users.This latency of delivering the video streaming services did not change. This means that the IPTV Service providers will increase their benefits by ensuring in the same time also the delivering of service with high quality and performance toward innumerous end users. Consequently, MEC Technology and SDN solution will be the two right and "smart" network choices that will boost the development of the 5th Mobile generation and will significantly improve the benefit of Video Streaming services offered by current providers worldwide (Netflix, HULU, Amazon Prime, YouTube, etc).

Keywords: SDN, MEC, IPTV, Latency.

1. INTRODUCTION

Nowadays, the end users' demands for Multimedia services like TV, Video, Audio, Text, Graphics and Data have increased tremendously. Most researchers believe that IP-based TV service, also known as IPTV service, is a key opportunity for operators around the world to increase their profits by offering video over IP networks. The IPTV service itself supports QoS (Quality of

Service), QoE(Quality of Experience or Perception made by end users), Security, Interoperability and the Level of Reliability required by IP networks. Since IPTV service is a paid service to receive all live broadcast TV channels, this service providers are focused on providing it in real time with very high QoE and low cost. To achieve this goal, IPTV Operators need a system to distribute content to high-quality, low-latency video subscribers located in different geographical locations. On the other hand, IPTV end users are expecting to get the service inhigh quality, low cost and low latency. To ensure the provision of IPTV service in accordance with the requirements of end users, providers require high bandwidth and an efficient way of sending video content from the source to end users.

In addition, typical IPTV networks have to deliver video content continuously and simultaneously to thousands of end users, thus monitoring and re-configuring these systems poses a very difficult problem. Multimedia services, which were previously provided by network operators and on a dedicated network, have now migrated to an open Internet. This way of gaining service is called OTT (Over-The-Top). OTT systems(Nam et al ., 2016) such as 'Nettflix', 'Youtube' and 'Hulu' for video service delivery can be challenging because end users, IPTV service providers and network operators (ISPs) do not have an overview of network point-to-point conditions. In this case, IPTV service providers do not have the ability to simultaneously realize the change of ISPs and the logic of that small part of the network that currently covers the user. In addition to that, the content server, once it starts broadcasting a channel, very rarely switches to another node. Consequently, due to unstable network conditions, end users often encounter re-buffering to the end of a video and in turn, the obtained service quality is low. That is how the idea for this research came up, the aim of which is to solve all the above problems using a new network technology, the SDN (Software Defined Networking) one. (Sezer et al., 2016).

SDN technology (Hysenbelliu et al ., 2015) is an ideal solution for managing IPTV networks because it has the potential to detect video quality problems within the network core with its new sizes and mechanisms. This 'Smart Solution' called SDN, supports Key Performance Indicators (KPIs) and contains powerful network monitoring and reconfiguration features. Through it, IPTV service providers may choose the best content server when an end user requiresto receive video service, enabling the optimization of obtaining IPTV service and monitoring of network conditions. Furthermore, the implementation of SDN technology for obtaining IPTV service reduces the OpEx/CapEx provision costs and provides an improved service (Hasan et al., 2020). Unlike traditional networks, the SDN network:

- separates the data plan that passes full-speed traffic from the control plan, which makes decisions about how to pass scalable traffic over a long period of time
- already provides a well-defined interface between the shared control plan and that of the data, including a set of abstractions for network devices that hide many details within them
- migrates the control plan logic to a centralized logic controller (SDN Controller) that employs the global view of network resources and applications requirements ‘knowledge to create and optimize general rules.

2. IMPLEMENTATION OF SDN TECHNOLOGY IN SMC IPTV ISP DATA CENTER

In this section, we have implemented SDN Technology in the real IPTV service provided, called ‘SMC IPTV ISP’, which is always interested in the continuous improvement of IPTV service (Esmeralda et al ., 2017-Esmeralda et al., 2018). Following the implementation of the SDN technology through Virtualization, the QoEof received IPTV Service, from end users’ point of view, increased significantly, the cost of receiving IPTV service decreased, and through the implementation of the GPU processor (Esmeralda et al., 2017) the service performance increased as well. The demand for such services has increased terribly, thus in turn increasing at the same time the bandwidth required in the network, the service delivery delays, as well as difficulties in the management and check of enormous data. In addition to that, the increase of IoT users, 5G services, mobile and social media users has led to an increase in the number of devices on the network as well as an increase in the central cloud traffic (increases the load and computing capabilities). This causes problems for real-time applications where latency is a very important factor. This introduces MEC (ETSI et al., 2016, Sabella at al., 2016) technology, which shifts resources across network boundaries and close to mobile and IoT users by providing real-time video streaming services with low latency and bandwidth, as well as high performance. Based on the principles of SDN technology, SMC IPTV ISP has built an optimized network architecture for providing low latency and high QoE IPTV service. This architecture is based on SDN and MEC technologies where MEC technology is made of three layers: the MEC Application, the MEC Platform itself and the Abstract layer.

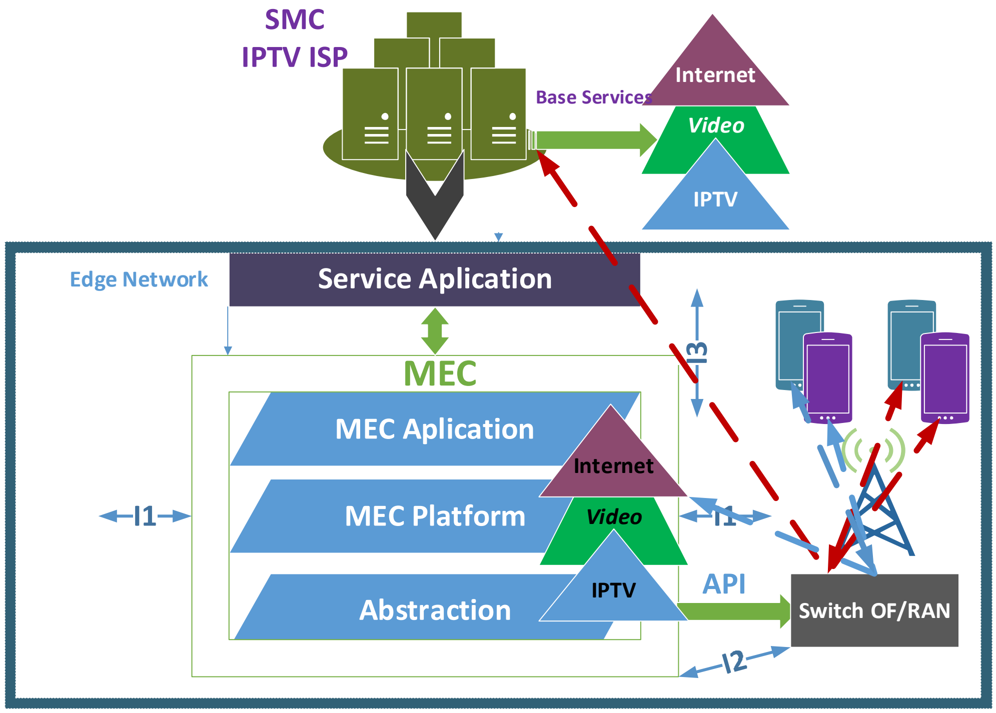

FIGURE 1: IPTV Service Optimization Architecture according to MEC based on SDN.

As seen in Fig 1, a copy of the services provided by the SMC IPTV ISP data center has been transferred to the end-user boundaries, i.e., to the MEC Platform. The aim is to receive an IPTV service with less delays and low latency, as well as high QoE and performance. Also, in this architecture, there are appeared two ways of receiving the IPTV service:

a) The first way is the traditional way (in red) where the end user wants to watch a real time TV channel or a streaming video by making the internet request to the real content server owned by the SMC IPTV service provider ISP. Whereas, the Routing part of the requests coming to the network is realized according to the principles of the SDN solution provided by the OpenFlow Switch.

b) The second way is the blue one, where two mobile users perform video streaming via HTTP as a specific MEC application. At the moment of starting video streaming, the above architecture has the possibility to program the Routing patches in such a way that the Video application is downloaded from the MEC Server and not from the real server of the SMC IPTV ISP Internet service provider. The MEC network also adjusts the streaming speed according to RAN status.

Consequently, the end user's deliver the optimized und enhanced streaming services like Video, IPTV with low delay and high QoE.

3. THREE SCENARIO SIMULATIONS WITH OMNET++ SIMULATOR

The main purpose of this part is to simplify the demonstration of receiving the already optimized and improved IPTV service using the SDN Network Solution and MEC Technology principles. Such simulations will be performed through theOMNeT ++ simulator platform. This simulator uses the NED language (Network Element Description language) to define the network structure and the incoming or outgoing data is placed in the '.ini' file (commonly called omnetpp.ini), supports all types of Wireless, and wired protocols.

As a first scenario it will be simulated the benefit of IPTV service from an end user via the internet using real time streaming protocols like RTP and RTCP.

As a second scenario it will be simulated the benefit of IPTV streaming service by applying the SDN solution and using the 'OpenFlow' controller.

Finally, as a third simulation scenario it will be realized the benefit of IPTV service through the implementation of MEC Technology based on an LTE-A network.

Some of the most important parameters of the transmission link for multimedia services such as Video, Audio, IPTV, etc. in real time are point-to-point end delays, Bit Error Rate (BER), service benefit latency, Jitter, SNIR and PacketLoss.

3.1 Scenario 1: Receiving IPTV service through internet using real-time streaming protocols such as RTP and RTCP

These are the manuscript preparation guidelines used as a standard template. Author must follow Simulation of video streaming data transmission (or IPTV service) uses the RTP protocol in the INET framework. Specifically, our simulation will be performed on the OMNeT ++ platform, version 4.2.2 and with the INET 2.2 framework. In the simulation that we will perform for the transmission of the Video service according to the H.264 coding standard, the following steps need to be performed:

1. The network is created in the INET framework of OMNeT ++, and specifically it is called 'Sk1IPTV'.
2. In this step, there are created the sending and receiving modules, namely the specific RTP hosts called 'Sender' and 'Receiver', which are responsible for sending, and receiving H.264 video data. These two modules complete the packaging and disassembly process.
3. It will be created video data routing elements, namely 3 IPv4 Routers that support wireless, PPP, Ethernet and external interfaces
4. Fourthly, the submodule connections are created with each other, which in this case are Ethernet connections as shown in Figure 2.

In this way, the network topology is with success created and is ready for simulation via the Sk1IPTV.ned file, as well as also the parameters of the 'Sender', 'Receiver' modules and other submodules by successfully generating the Sk1IPTV.ini file. Communication has also been logically established between the 'Receiving and Sending' modules, which will exchange IPTV video streaming with each other.

The simulation process appears through a special interface called Tkenv, which shows in detail the specific events, time, number of actions of modules and submodules, etc. This interface also verifies the validity of the network topology, which visually shows in real time, which modules receive or send data packets, as it is show in Figure 3.

In this first simulation scenario, we will consider the evaluation of some very important parameters that directly affect the quality of images of IPTV service received by end users; the evaluation of the Point-to-point delay of delivering the video streaming service, Queue packet time and Packet loss. The main purpose of these simulations is to highlight the improvement of the quality of IPTV service from the implementation of new network solutions such as SDN and MEC technology.

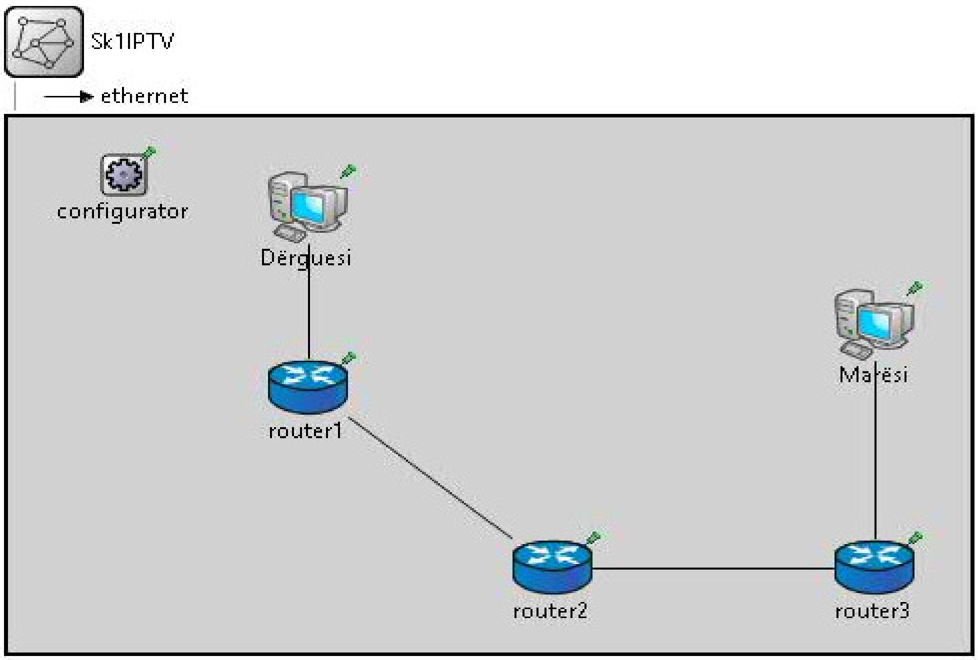

FIGURE 2: Transmission of H.264 video data between a Sender and a receiver (.ned file).

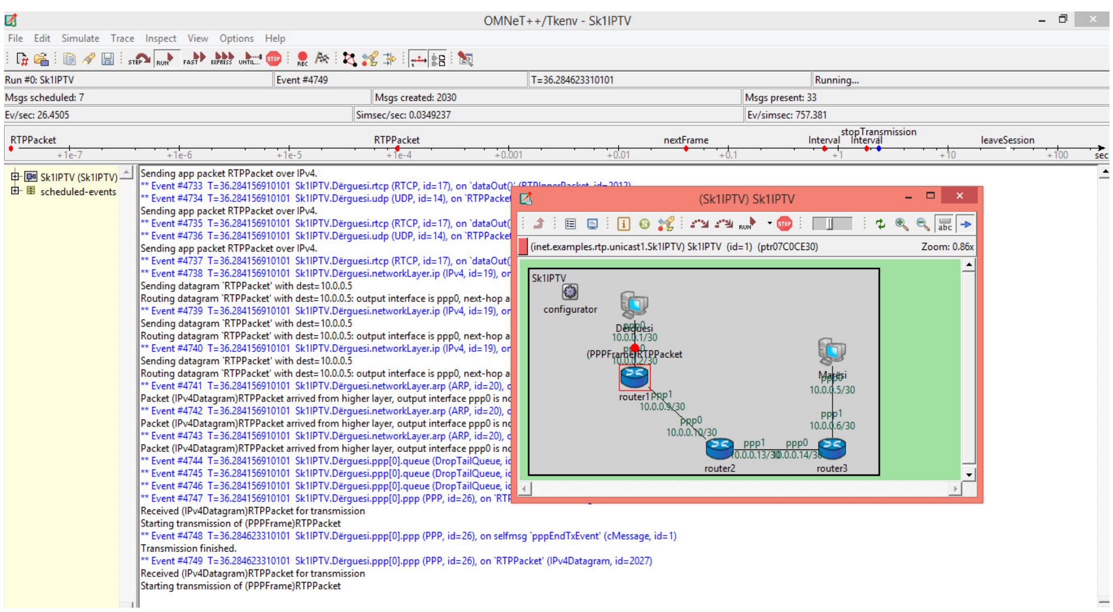

FIGURE 3: The final result in the Tkenv interface.

a) Estimation of the Point to Point delay

According to the network topology created in fig. 2, the network is configured in the .ned file with the corresponding modules and submodules, as well as their parameters are defined by creating the .ini file

The first simulation is performed for 300s and the Video sent by the Sending Module is lego_video.mpg.gdf, with a size of 54,528 Bytes and Bandwidth = 10Mbps. Final simulation data generation in OMNeT ++ IDE is realized in three forms; vector data, scalar data and histogram data. Then, the final delay between 'Sender' and Recipient while obtaining video streaming is as in the figure below:

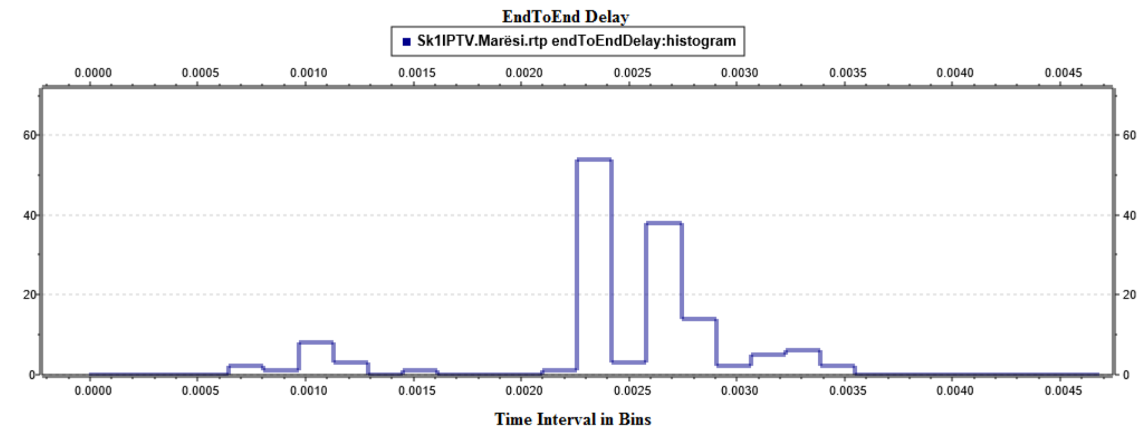

FIGURE 4: Estimation of point-to-point delay in delivering IPTV service.

The final delay is received in the RTP Receiving Module, which reaches a maximum value of 54 seconds, in the bins interval (0.00225804484, 0.0024194981733333) and no delay is displayed from the RTP Sender module.

b) Estimation of packet waiting time for transmission

As shown in Figure 5, the longest packet waiting time occurred on Router 1, and specifically 93 seconds in the bins interval (1.346E-5 .. 1.346E-5.)

In order to estimate the delivering time of packets during the transmission of video streaming from the Sender module to the Receiver Module, we will take again in consideration the 'Histogram data'.

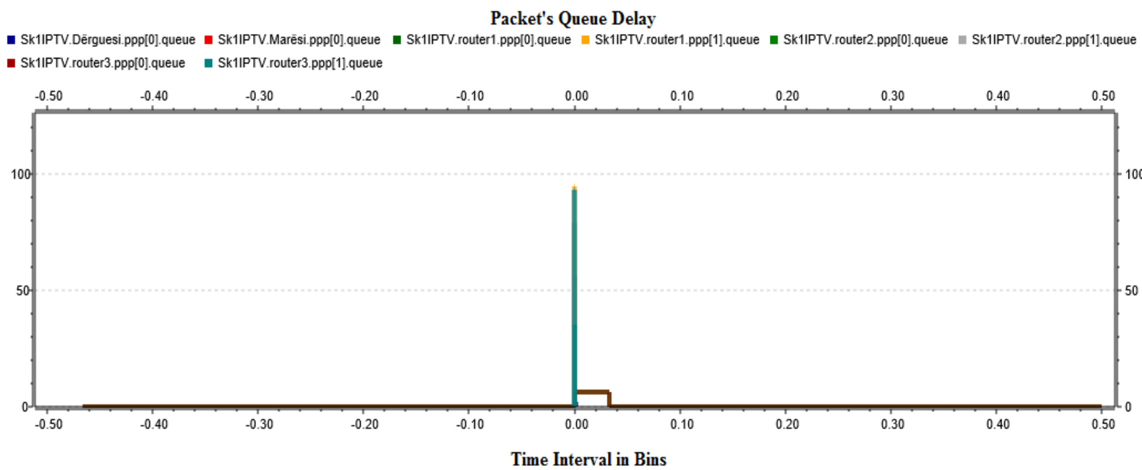

FIGURE 5: Estimation of the packets'queuing delay.

c) Estimation of Packet Loss

One of the most important parameters during transmission of video streaming is also that of PacketLoss, which the smaller it is, the better the quality of service and performance is delivered by the end users.To get an accurate definition of this parameter, we have estimated the number of packets sent from the udp Sender Module to the rtp Receiver Module (in bytes), as well as the number of packets received from the rtp Receiver Module (in bytes). Logically their difference determines the value of packet loss. From the generated conclusions file, we considered the scalar data. As shown in the figure below, the number of packets sent by the udp sender is 55802 Bytes,while the number of packets received by 'Receiver rtp' is 53410 Bytes. Consequently, the number of lost packets is 55802 - 53410 = 2392 packets. Referring to the following formula for calculating packet loss in percentage, then for the first simulation scenario, this packet loss P (H) is 4.28%.

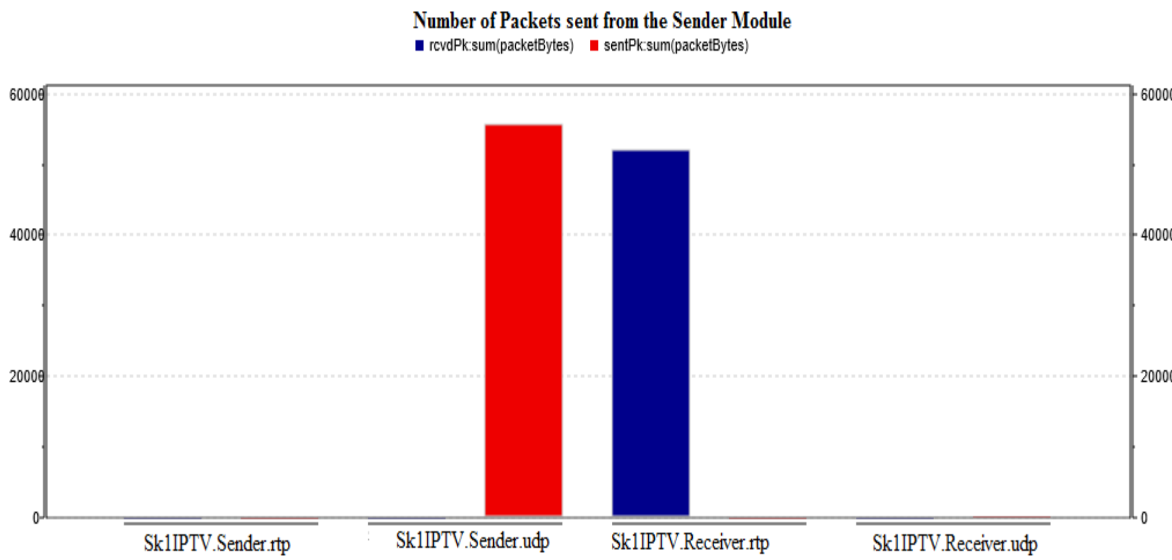

FIGURE 6: Evaluation of packets sent by the Sender module and that ones received from the Receiver Module.

P (H) = P (h) / N

Where P (h) is the number of packets lost

N is the total number of packages sent by the Sender

3.2 Scenario 2: Receiving from IPTV Service using SDN Solution

In the OMNeT 4.2.2 simulator that we will use to simulate the delivering of video streaming service using the SDN solution, we need to integrate OpenFlow components. OpenFlow assists in the development and use of SDN technology by providing high flexibility in routing network leaks, as well as allowing you to change the behavior of a portion of the network without affecting whole data traffic. This is achieved by separating the control plan in the OpenFlow Switches network from the data plan. In this way, using the INET framework, OpenFlow Switches and a Controller, we have managed to build the network topology in OMNet. The controller is responsible for all streaming data routing decisions as well as changing packet-forwarding rules in Switch. To realize the simulation of receiving the IPTV service on an SDN network, we have used a Controller, a Server that carries video data according to the RTP protocol configurations, two OpenFlow Switches and end users who will receive IPTV streaming service; Client1 and Client2. The steps followed for building the network topology in NED language and all the other actions are same as in the first scenario, in thus a way the NED topology is SDN1.ned as it is showed in Figure 7.

In the simulation that we will perform we have taken as a behavior of the Controller that of 'Forwarding' because the Controller module itself has complete knowledge of the whole network, thus realizing the sending of packets to the OpenFlowSwitch_it, which is positioned in the path of required for data transmission to the end user.

The simulation is presented through the Tkenv interface, which shows in detail the specific events, time, number of actions of modules and submodules, etc. This interface, as shown in Figure 8, shows in detail which modules and submodules in the SDN1.ned network topology exchange information with each other.

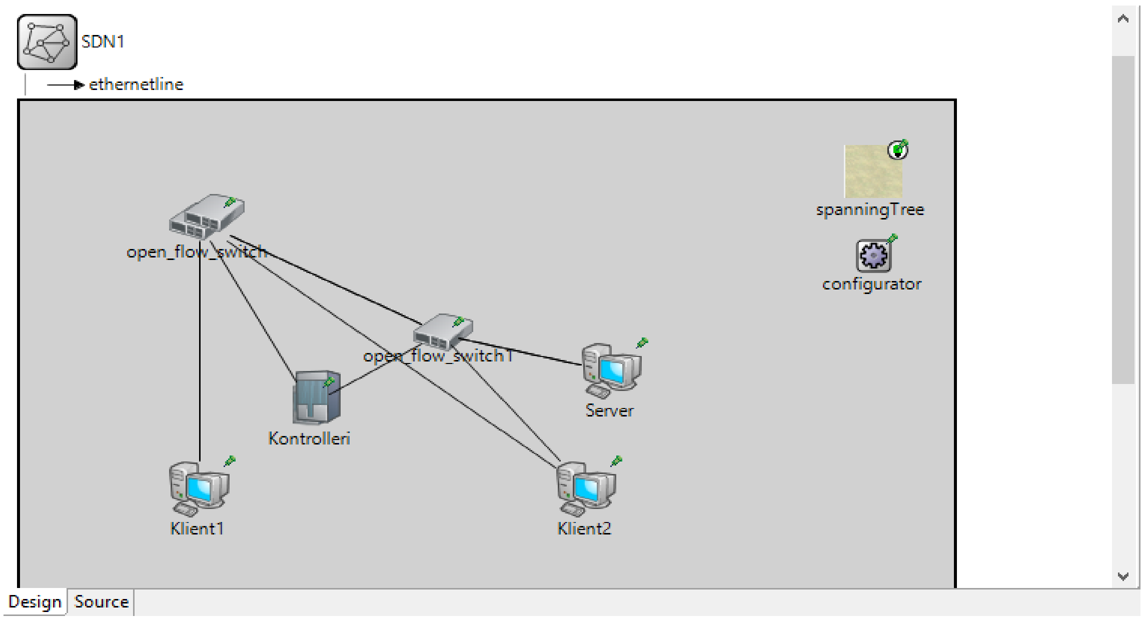

FIGURE 7: Building the SDN network topology according to the NED language (SDN1.ned).

The initial parameters considered for the SDN1 network are: Bandwidth 100 Mbps, delay is 0.000001s, Clients are standard Hosts, Throughtput 10µs, Server is an RTP Host and the uploaded video is the same as in the first scenario; lego_video.mpg.gdf with size 54528 Bytes. This video is encoded in H.264 format. The simulation time is 600 s.

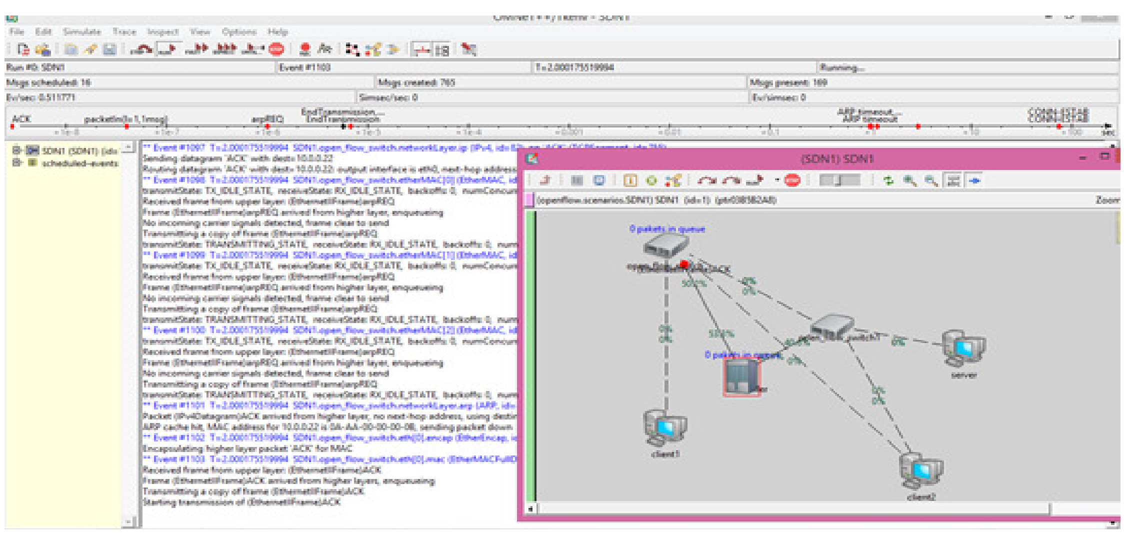

FIGURE 8: The final result in the Tkenv interface.

The parameters we will evaluate in this second scenario are:

a) Average RTT (Round Trip Time)
This time, called RTT is the same as end to end delay. This parameter estimates the time it takes for packets to be transmitted from a given 'Sender' to a 'Receiver'. The smaller this delay, the better the quality of service that the Customer perceives (So the QoE increases). SDN solution has high expectations to enable the delivering of Video Streaming service with small RTT.

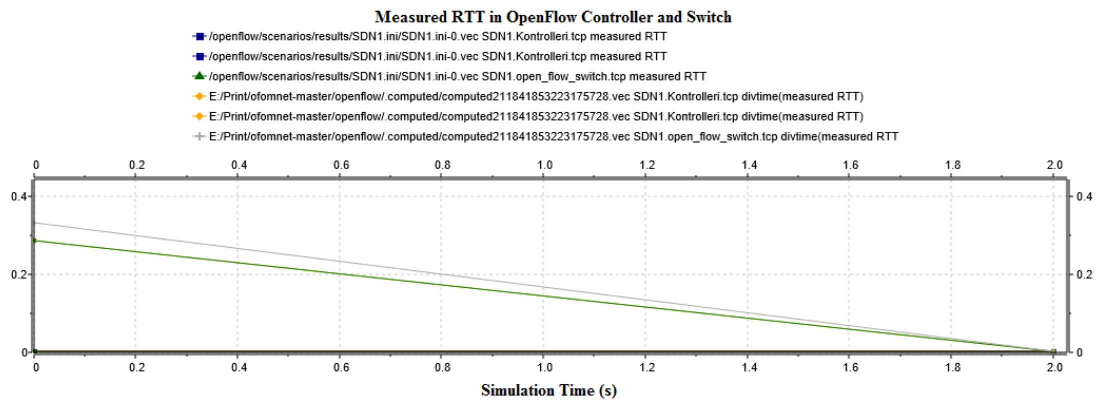

FIGURE 9: Estimation of RTT average time for OpenFlow Controller and Switches.

As seen in Figure 9, the biggest delay is shown by the OpenFlow Switch, which goes specifically to 0.332894725869979 s.

Compared to the point-to-point delay estimated from the first scenario (54 s) without applying the SDN solution, this delay is reduced by 61.62%, which increases direct the quality of service delivered from the end user's (QoE).

b) RTO (Recovery time Objective)
RTO time is a paramter that measures the time that data is invalid or inaccessible to the network. Mostly this time will be estimated at SDN Controller and OpenFlow Switch at TCP level. Similar to PaketLoss in the first simulation scenario, this time should be as short as possible in order to deliver with high-performance the real-time streaming services like IPTV. As you can see the result in the figure below, this time for the OpenFlow Controller and Switches is almost the same, specifically it achieves the value 0.65625s.

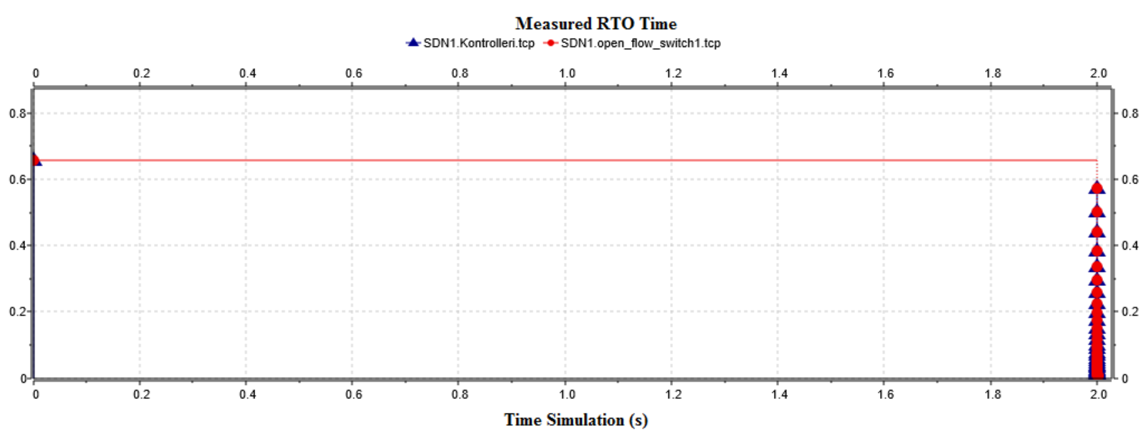

FIGURE 10: RTO estimation for OpenFlow Controller and Switches.

c) Estimation of PacketLoss

To estimate the number of packet's Bytes being lost (Packetloss), specifically to Client1 and Client2 in this time interval, we will calculate the number of bytes of packets sent by the RTP Server and those that actually have been successfully received by end users. As shown in Figure 11, the number of bytes of packets sent by the Video Streaming Server is 2432.

While the number of packet's bytes successfully received by Client1 and Client2 are:

Client1 has successfully received 2368 bytes of packets
Client2 has successfully received 2432 bytes of packets

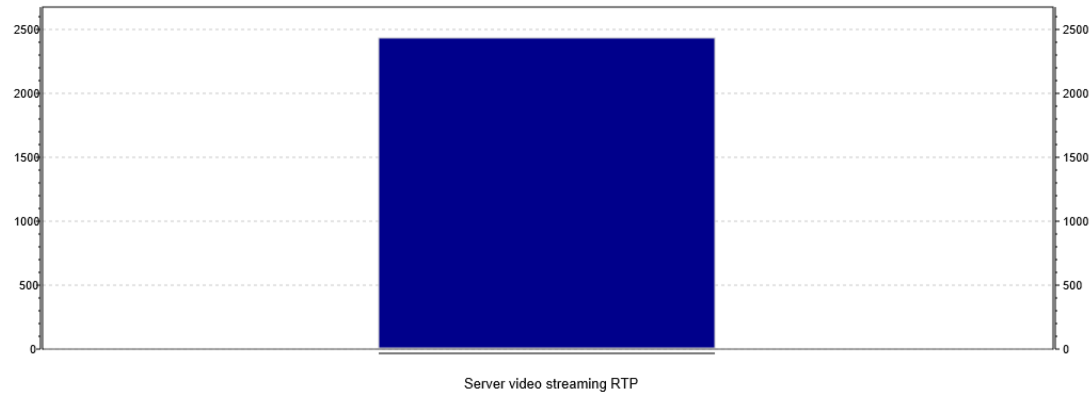

FIGURE 11: The number of packets' bytes sent by the Video streaming Server.

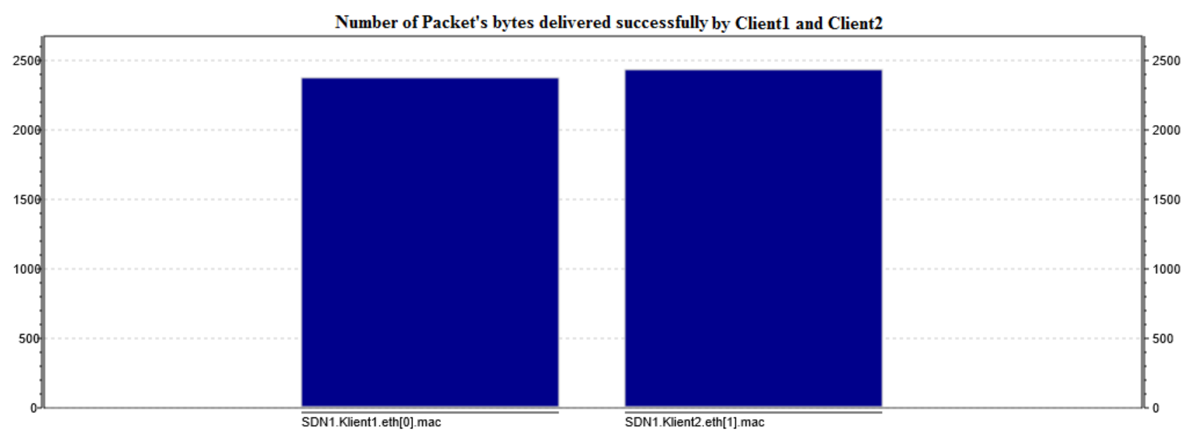

FIGURE 12: The number of packets' bytes successfully delivered by Client1 and Client2.

As shown in the Figure 12, Client2 has successfully received 2432 packets' bytes, thus in Client2 there is no PacketLoss. Referring to the formula for calculating packet loss (First Scenario; Estimation of packet loss) in percentage, then packet loss in percentage referred to Client1 is: P (H) = 2.631%.

Comparing this value with the value of packet loss from Scenario 1 (P (H) = 4.28%) where delivering IPTV streaming service is realized without applying the SDN solution, then we conclude that IPTV streaming service through SDN solution is delivered with a relatively low PacketLoss, decreasing by 61%.

3.3 Scenario 3: Delivering IPTV Service using MEC Technology

Recently, the demands for Video streaming services have increased significantly. Powerful service providers such as 'Netflix', 'Hulu', 'Amazon Prime Video', etc. are facing major problems with network capacity, fast bandwidth consumption, large amount of data management, resource efficiency and saving of required QoS service quality. Requests for Video Streaming services have become a necessity not only for Wireless end users, but also for mobile ones. Consequently, the delivering of these services in real time needs to be provided with low Latency and high QoE. Last year, there were reported numerous problems with the delay in getting streaming services from Netflix service users. In this way, it is necessary to realize the placement of resources of these services (eg Content) and the realization of data processing and management as close as possible to the end user, finally at the age of the network, in order to reduce delays, jitter and increase the quality of service QoE. MEC technology enables the deployment of a computing platform in the Cloud (called MEC Host) as close as possible to mobile users on 4G and 5G networks in order to receive the service with Low Latency and high performance. In this way, for this third part of the simulations we have chosen to realize the benefit of the Video streaming service using MEC technology in an LTE-A mobile network. The main purpose is to evaluate the benefit of delivering IPTV video streaming service with low latency using MEC technology combined with SDN and NVF network solutions. To realize this simulation scenario we will use the simulator OMNeT ++, version 5.1.1. The MEC architecture will be integrated within the LTE-A network through the SimuLTE framework (Giovanni et al., 2016) enabling the evaluation of the performance of MEC services according to the real conditions of a network infrastructure. The main reason why we chose the LTE network for the implementation of MEC technology and to finalize the delivering of video streaming with very low latency, is because the MEC in interaction with the developed standards of LTE technology will be a by very important component for building 5G mobile network.

To build the MEC Host model in the SimuLTE framework integrated into OMNet, we have referred to the work done by the authors in (Giovanni et al., 2018). As shown in the figure below, Host MEC is built as a composite module, which has within it four sub-modules; Simple ME Platform Module, Simple ME Application Module, Video Streaming Server Module and PFGTP Endpoint Module.

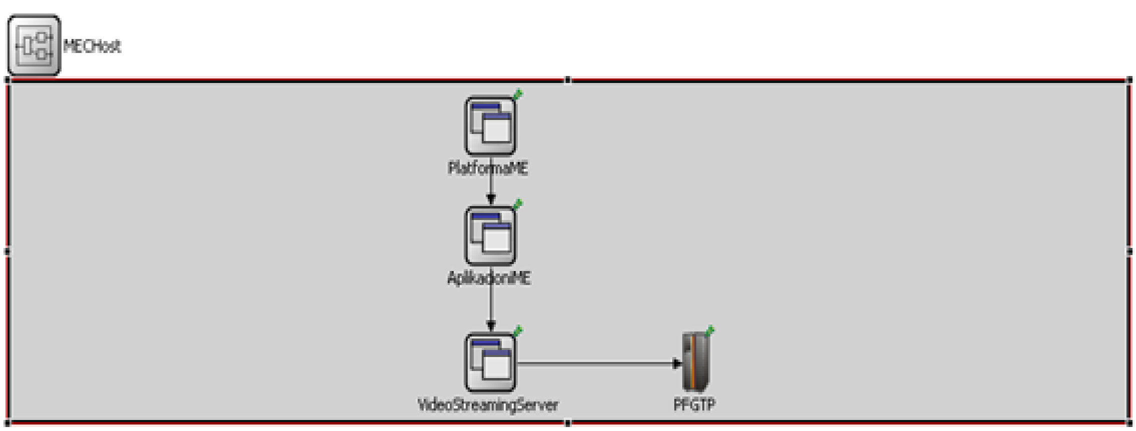

FIGURE 13: Building MEC Host in SimuLTE.

The ME Application Module is the module that receives video streaming requests from EU end user's, which go first to the video streaming udp server via the PFGTP endpoint. After that, the Video streaming server will communicate with the base radio station called eNB, which receives requests for video streaming from the respective EU. The GTP module can be placed inside the EPC part of the LTE network and communication between them is tunneled via the GTP protocol. This module performs the encapsulation and de-capsulation of data packets within the GTP package.

In this way, we will build the LTE network in OMENeT by placing the MEC Host above directly on the eNB base station as shown in the figure below.

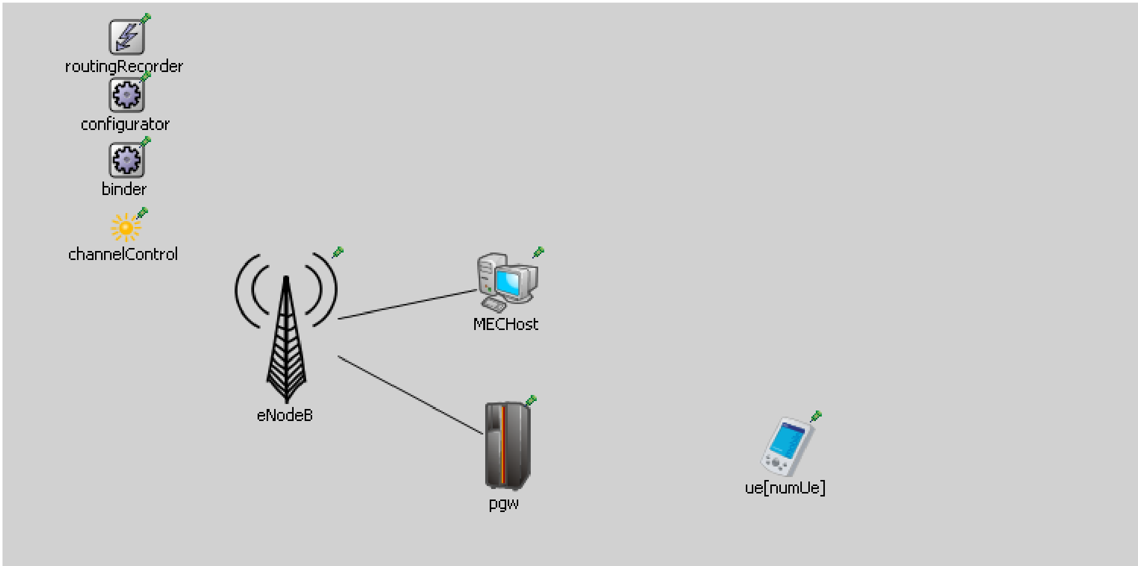

FIGURE 14: Construction of LTE network topology in OMNeT based on MEC Host connection directly to eNB base station.

In this way, the delivering of video streaming service from the EU could be realized with low latency, very low packet loss and high performance.

We have built the .ned file and .ini configuration file. In the first case of the simulation, we will get the same 5MiB video size (5.24288 MB), with packet length of 500B, packet delivery interval of 10ms and 20 EU.

The simulation results are in .vec and .sca files.

In general, the concept of "Latency" indicates the delay between a source and a certain destination caused by a certain network (which can be a mobile network, a wireless network, etc.). Since this parameter is a very important aspect in the development of the 5G mobile generation where it is expected that this parameter called 'latency' will be much lower compared to the previous 4G generation, it is really necessary to determine exactly what this delay is and where it depends. Thus, the latency classification in the network referring to 3GPPis divided into;

• **Latency according to the control plan** - which includes the delay of the transition of end users from the state of 'Quiet' to the 'Active' in both, the Radio network and the core network.

• **User plan latency** - is the time it takes for a valid packet in the IP layer of the RAN node or EU end user to transfer and be valid again in the RAN or EU. As explained above, the RAN node is the node that provides the Radio access interface to the Core network.

In our simulation case, we will evaluate the Latency according to the user plan which constitutes the time it takes for a packet from the moment it becomes valid at the source and until it is valid at the destination node.

As seen in the Figure below, the latency of delivering video service from 20 EU according to MEC technology is 0.004 s.

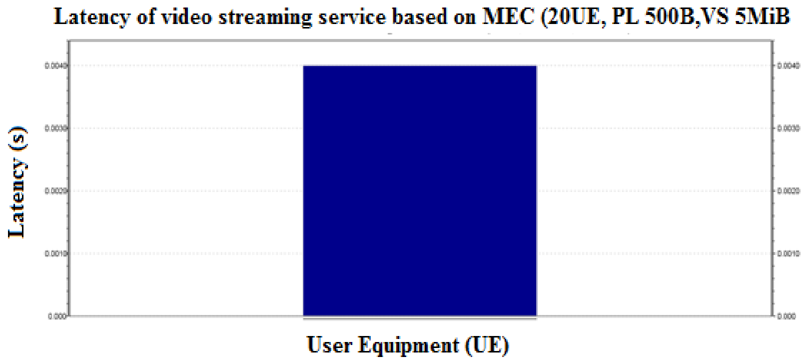

FIGURE 15: Latency of delivering Video Streaming service based on MEC technology for 20 UE.

Comparing it to the result of two other scenarios, where we have not applied MEC technology, Latency has been decreased by >90% (0.332s based to scenario 2 and by 53,996s based to scenario 1).

From the analysis of the simulation results files, we noticed that in the case of applying the MEC technology for the delivering of the Video Streaming service, the packet loss is 0. This means that the QoE quality of the service perceived by the end users is very high.

As seen in Figure 16, with the increase in the number of End Users, specifically from 20 to 100 UE, and from 100 to 1000 UE with unchanged packet length of 500 B, unmodified packet delivery interval 10ms, Service benefit latency video streaming is again approximately 0.004, therefore it does not change. In addition to that, PacketLoss is 0.

These results show that MEC Technology and SDN will be the two right and "smart" network choices that will boost the development of the 5th Mobile generation and will significantly improve the benefit of Video Streaming services offered by current providers worldwide (Netflix, HULU or Amazon Prime Video).

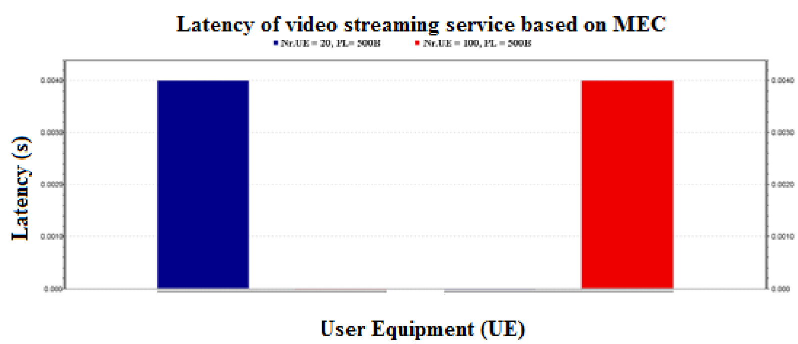

FIGURE 16: Comparison of the Latency of delivering Video Streaming service based on MEC technology for 20 UE and 100 UE.

4. CONCLUSION

This paper addresses the improvement and optimization of delivering IPTV service through the application of SDN solution and network virtualization. We have also built an optimized IPTV service delivery architecture based on MEC and SDN technology, in which streaming data was moved to small clouds as close as possible to the end users. Consequently, video streaming services like IPTV, VoDetcwere delivered with low Latency and Bandwidth, and in the same time with High Performance QoE.

To evaluate the optimizations of IPTV service based on the SDN network solution and according to the MEC technique, we have built three simulation scenarios in OMNet ++, through which we have obtained the following results:

1. The final RTT delay of delivering IPTV service via SDN, compared to delivering the IPTV service through RTP and RTCP streaming protocols has been reduced by 61.62%.
2. Network PacketLoss, estimated according to the SDN solution, reduced by 61%
3. By applying the MEC technique in the LTE-A network for the provision of the video streaming service, the service was delivered with latency reduced by >90 % ***(0.332s based to scenario 2 and by 53,996s based to scenario 1)***, with PacketLoss of almost 0 and with high performance QoE
4. Also, with the increase in the number of End Users, specifically from 20 to 100 UE, and from 100 to 1000 UE with unchanged packet length of 500 B, unmodified packet delivery interval 10ms, the latency of delivering video streaming services does not change. In addition to that, PacketLoss is 0. This is the greatest innovation achieved where the economic benefits of IPTV Service Providers will be growing up tremendous, while also increasing in the same time the quality of the delivered IPTV Service.

5. FUTURE WORKS

This research paper has brought about results which are considered as innovations or as resources to be developed in other studies in the future. Among the main improvements, the following may be mentioned:

1. Application of MEC technique by giant Video Streaming service providers such as 'Netflix', 'Hulu', 'Amazon Prime', Youtube etc., where the Content could be placed as close as possible to the users at the borders with the network to deliver the IPTV video streaming service with low Latency, QoE quality and high performance.

2. Furthermore, a future work could be the construction of optimizing algorithms that will enable the benefit of IPTV service through SDN based on MEC principles

3. Implementation and production of Virtual Set box (V-Set Box) for the provision of IPTV and VoD based services to reduce the cost of providing such services and increase the speed of providing new services to end users.

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