Zaria Journal of Electrical Engineering Technology, Department of Electrical Engineering, Ahmadu Bello University, Zaria – Nigeria. Vol. 12 No. 1&2, September 2023.

Development and Performance Analysis of Routing Algorithm for Voice over Internet Protocol Long Term Evolution Network

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Keywords: –
Internet Protocol
Long Term Evolution
Quality of Service
Voice Over Internet Protocol.

Article History: –
Received: July 22, 2023.
Reviewed: Aug. 30, 2023
Accepted: Sept. 10, 2023
Published: Dec. 28, 2023

ABSTRACT

Voice over Internet Protocal (VoIP) has become accessible to consumers in recent years due to the increase in voice call applications and the increasing capabilities of the internet. Consumers can access the internet almost anywhere. Long Term Evolution (LTE) network is a standard for wireless data communications intended to significantly improve the previous technology standard. Unlike the precedent technology that supports circuit switching, LTE only supports packet switching. The development and performance analysis of routing algorithm for VoIP LTE was carried out to determine the network Quality of Service (QoS). VoIP LTE model network was developed firstly, followed by routing algorithm based on the network model and simulated on Network simulator2 (Ns2)., It has been found that throughput with obtained value 10.52Mbps against the International Union Telecommunication standardization sector (ITU-T) (0-50)Mbps range, end to end delay obtained value 0.06ms against (0-150)ms range and jitter with obtained value 7.00ms against (0-20)ms fall in the good range of QoS while packet loss with obtained value 1% against the (0.5-1.5)% range falls in the acceptable range of ITU-T QoS standards. The results show that VoIP LTE is a good service for both local and international application because the network offers good QoS.

1. INTRODUCTION

The progressive improvement in the functionalities of mobile devices and the emergence of Long Term Evolution (LTE) wireless technology aim to providing a high quality data network as well as voice communications that can co-exist over Internet Protocol (IP) network. According to Ralf (2018), Voice over IP (VoIP) is simply defined as the digitized voice traffic intrinsically transmitted using a data network to make telephone calls. This differs from using a traditional analogue circuit switched public network, as now the data has been split into packets. These packets can take any route to reach the *destination*. Packetized data travel through a virtual circuit which differs from a circuit switched network in that the circuit does not need to be reserved for the entire duration of the call between the sender and the receiver with packet switching. Thus the channels may be utilized more by sharing with other users than compared to circuit switching. However, the data packet can arrive out of sequence, experience delay or even may never arrive as a consequence of traffic congestion and buffer overflows. These are some of the major disadvantages of sharing traffic across a virtual network that VoIP traffic has to contend with. On the other hand, the advantages offered include the multiple routing of the VoIP traffic ensuring a cheaper and often free of cost flow of traffic between the different intra-packet network components such as the routers and switches. Transmitting digital data in the format of packets signifies that all types of digitized data such as voice, video, fax, music and telephony have the opportunity to be carried together utilizing a shared common network at any given time.

The aim of this research work is to develop a routing algorithm for Voice over Internet Protocol Long Term Evolution network, and analyse its performance. The aim is achieved though the following objectives:

- To develop a VoIP over LTE network topology model and investigate its performance
- To develop and implement routing algorithm that will guarantee user QoS when using VoIP system.
- To measure and analyse VoIP QoS through the measurement of jitter, packet loss, delay and throughput by creating two different network topology with varying network load.
- To compare the results that shall be obtained in the analysis with VoIP call quality requirement given by the international telecommunication union telecommunication standardization sector. (ITU-T) for LTE.

2. LITERATURE REVIEW

In order to gain sufficient understanding and information on Voice over Internet Protocol Long Term Evolution Network, and other related information relevant to this work, it is worthy of note to review existing research papers, projects and base technologies that support or have a relationship with this subject area. The most notable literatures that have a direct relationship or are very similar to this work were chosen and reviewed among the several academic work.

According to (Oluwadamilola et al. 2016), LTE is designed as an all-IP-network. This implies that all applications on the network medium share the same channel. That is guaranteeing QoS of delay intolerable multimedia applications now become very crucial to fulfill the benefits attached to LTE network in the Third Generation Partnership Project (3GPP). The paper did not go into in depth study of how good is the QoS when in terms of Voice over IP Long Term Evolution Network.

In (Mohammed E. and Amin B. A., 2014), Atoll simulator was used to simulate LTE network and to study its performance. It was found that the network has many advantages over the previous second and third generation networks. The signal coverage by throughput in uplink and downlink, coverage by signal level, coverage by signal to interference ratio in uplink and downlink and overlapping zone in addition to the limited resources are good and steady at speed during transmission. The research focuses on LTE network study considering uplink and downlink transmission.

In (Mahdi et al, 2017), the paper presented the early findings related to VoIP traffic transmitted through WiFi, WiMAX and WiFi-WiMAX networks. Initially, two scenarios were designed using optimized network technology OPNET software where both generation and termination of the VoIP calls take place in an environment of homogenous networks such as WiFi and WiMAX. Another scenario was later added where calls were generated at the WiFi network and terminated at WiMAX networks and vice-versa. It was found in the research that the average jitter value of the WiFi-WiMAX scenario did not perform as expected even at some points time, it exceeded that of WiFi. The mean opinion score MOS of the WiFi-WiMAX network was found to exhibit a much higher performance than that of the WiFi and or WiMAX. Similarly, the packet end-to-end delay of WiFi-WiMAX remains close to that of WiFi and is much higher than expected.

2.1 LTE

The advent of mobile devices and has a great influence on our lives and has impacted both public and private sectors. Various evolutions have been witnessed in advancing series of mobile Telecommunication system. One of the recent is the LTE of the UMTS, (Isaam, T. 2009).

LTE is the access part of the Evolved Packet System (EPS). It is a wireless broadband designed to support roaming internet access via cellular and handheld devices. According to (Christine D. et al, 2013), the development of LTE was due to the huge increase in the number of smart devices users and an increase in the number of low latency tolerance applications like online gaming, video-conferencing, video streaming and VoIP that has zero tolerance for jilter or delay. This rapid increase had lead the 3GPP to work on the LTE on the way towards 4th Generation network (4G).

The 3GPP in their release 8 improved on its standardization work on initial EPS by introducing Evolved Packet Core EPC and Evolved UMTS Terrestrial Radio Access Network (RAN), (Aderson et al, 2011).The EPC is the latest evolution of the 3GPP core network architecture. EPC is based on packet switching only. It is an all IP-network according to Anderson et al (2011), which aids the end-to-end architecture for supporting smart and mobile network devices.

2.2 Voice over Internet Protocol (VoIP)

Traditional voice calling uses Packet-Switched Telephone Networks (PSTN), which uses minutes to determine costs. Making international calls are more expensive which hinder long distance communication abilities. VoIP was developed to provide voice communication anywhere in the world that has access to the internet. VoIP uses internet as a backbone, as opposed to the telephone network, and compresses data packets while they are transmitted. Because of this compression, more data is able to be handled and more calls can be carried across the line. VoIP also enables multiple users to connect at the same time without much additional data, for this reason, it is a suitable way to have international business meetings.

Studies have shown that around 50% of voice conversations are silent. To avoid wastage of bandwidth in data communication channels, VoIP fills these silent spaces with data which increases the efficiency, (Unuth, U. 2015). A well-known example of VoIP is Skype and the number of users continue to exponentially increase, as shown in figure 2.1, (Richter, F. 2017).

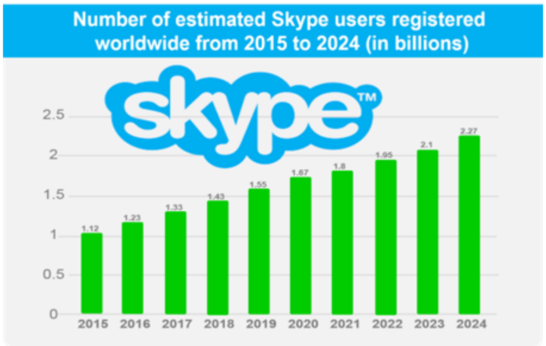

Fig. 1: Skype User Statistics, (Richter, F. 2017).

2.3 Network Simulator 2 (Ns2)

Network Simulator (Version 2), widely known as Ns2, is simply a discrete event driven network simulation tool for studying the dynamic nature of communication networks.

It is an open source solution implemented in C++ and Otcl programming languages, (Salima et al, 2015). Ns2 provides a highly modular platform for wired and wireless simulations supporting different network element, protocol. For example, routing algorithms, Transmission Control Protocol (TCP), User Datagram Protocol (UDP), File Transfer Protocol (FTP), traffic, and routing types.

Ns2 is used in this research to simulate the routing algorithm that has been developed for VoIP over LTE network because of its following features and advantages:

It is popularly used in academia and industries.

It is an open source software that:

i. Provides greater accuracy and speed of testing
ii. Supports large number of external protocols
iii. Programming can be done using tcl script and/or C++
iv. Provides with visualization tools
v. Complex scenarios can be easily tested
vi. More ideas can be tested in a smaller timeframe

In general, Ns2 provides users with a way of specifying network protocols and simulating their corresponding behaviors. Result of the simulation is provided within a trace file that contains all occurred events. Ns2 is a REAL simulator developed by UCB (University of Carolina Berkley) in 1989.

The Platform required to run network Simulator are as follows;

1. Unix and Unix like systems
   i. Linux, Free BSD, SunOS/Solaris
2. Windows 95/98/NT/2000/XP/7/8.1 (requires Cygwin).

2.4 Quality of Service (QoS) Parameters

VoIP QoS is determined by many factors such as jitter, throughput, end-to-end delay and packet loss (Khaled et al, 2019). These factors for each VoIP usage scenario was compared. The network quality parameters are summarized in table 1 as given by the ITU-T.

Table 1: ITU-T PRECEPT Voice Quality for LTE (James et al, 2007)

QoS Parameters	Good	Acceptable	Poor
End-to-end delay (ms)	0-150	150-300	>300
Jitter (ms)	0-20	20-50	>50
Packet loss	0-0.5%	0.5%-1.5%	>1.5
Throughput (Mbps)	0-50	50-144	>144

In this section, the steps and methodology adopted in order to achieve the set goals for this work are stated and explained in detail. The processes carried out here are mainly influenced by the literature reviewed in previous section.

The routing algorithm for VoIP over LTE was developed and implemented on Ns2 software. Awk programming script was used to process and evaluate the trace files generated from the algorithm after running it on the Ns2.

3.1 LTE Network Model

The model for the LTE network is shown in figure 2. It consists of 2 servers, 3 Access Gateways (AGWs), 3 Enhanced Base Transceivers (EBTs) and 9 User Equipment's (UEs). On server0, there are two AGWs. Each AGW has an EBT. Two users are on EBT (0) and three on EBT(1). On server 1, there is one AGW and one EBT with four users.

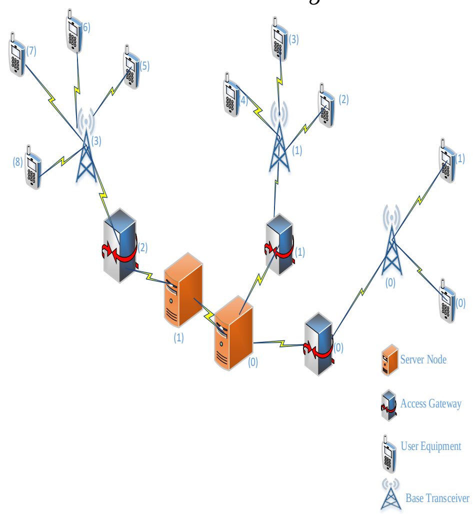

Fig. 2: LTE network model

3.2 Routing Algorithm

Input:

S ← number of servers;
A_{GW} ← number of AGW;
E_{BT} ← number of EBT;
E_{EU} ← number of EUs;

Output:

Throughput, end to end delay, pkt loss & computed jitter

Initialization:

Set pkt_size = 480bytes; *% as per CISCO convention*

Set pkt_interval = 0.03ms *% asper CISCO convention*

Initialize trace files and animation file as well

For *tr = 0 to* E_{EU}***Do***

Set_throughput_trace(tr) ;
Set_pkr_loss_trace(tr);
Set_e2edelay_trace(tr);
Set_animate (out.anim);

EndFor

Configuration:

For i = 0 to S **Do**

ns ← setServer(*server(*i)) *% connect S nodes*

For j = 0 to A_{GW} **Do** *% connect AGW nodes to S nodes*

ns ← *connectGateWays(AGW_j server_i) ;*
ns ← setOrientation(AGW_j server_i orient right-up);

For k = 0 to E_{BT}**Do**
ns ← set_EBT_to_AGW (EBT_k AGW_j) ;
ns ← Attach_AGW_to_EBT(AGW_j EBT_k);
Endfor
Endfor
Endfor

For n = 0 to E_{EU} Do
If n <2 then
Ns← connect(UE(n) EBT(0));
Ns← connect(EBT(0) UE(n));
Else If n >= 2 **&&** n < 5 **then**
ns ← connect (UE(n) EBT(1)) ;
ns ← connect (EBT(1) UE(n));
Else
ns ← connect(UE(n) EBT(2));
ns ← connect(EBT(2) UE(n));
EndIf
EndFor

For *ud = 0 to* E_{EU}***Do***
Set_up_UDP([new agent])
Attach_agent(UE(ud) ,udp(ud));
EndFor
For *cb = 0 to* E_{UE}***Do***
Set_pkt_pttn([new application]);
Init_pkt(cbr(cb) ,pkt_size ,pkt_interval);
EndFor

While (im <= E_{EU} **) Do**
Set_sink(sink_im , [new Agent]);
Attach_agent(UE_im , sink_im);

EndDo
// UE(0) to UE(2)
Connect_EUs(udp_0 ,sink_2);
Connect_EUs(udp_2 ,sink_0);
// UE(3) to UE(5)
Connect_EUs(udp_3 ,sink_6);

Connect_EUs(udp_5 ,sink_3);
// set-up group chat
For gch = 1 to $E_{EU} - 1$ **Do**

Group_chat_config(cbr_num,num_sink,pkt_size,pkt_interval, get_UDP_for_GROUP( ) **);**

EndFor

For n = 0 to E_{EU}**Do**

Compute delay $(T_d(n) - T_s(n))$ // where$T_d(j)$ and $T_s(j)$ time for transmission and receipt respectively.

For t = 0 toT_{sim} **Do**

Compute_throughput($p\_size_i(t)$);

Compute_pkt_loss($p_{discard_i}(t)/p\_size_i(t)$) ; // P_discard$_i$(t) and p_size$_i$(t) are the size of discarded packets and the size of all packets that have arrived

EndFor

EndFor

End

4 Result and Discussion

This section shows the result of the implementation of the routing algorithm on Ns2. The Ns2 output trace files were analysed and evaluated using Awk programming script. The QoS; Throughput and Packet loss plotted on Xgraph of the Ns2 and interpreted below.

Throughput

The simulation graph shows different throughput levels on both one-to-one call and group chat. The simulation is consistent with theory as group chat has introduced three times the traffic than two one-to-one calls, and the overlaps of both scenarios also indicated that there are no effects from each scenario on throughput as shown in figure 3.

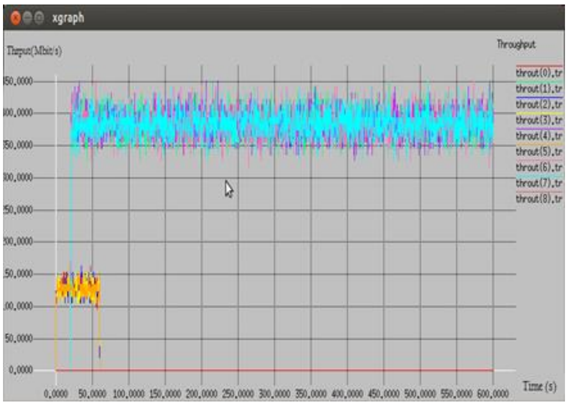

Fig.3: Throughput

Packet Loss

The simulation graph shows that a significant packet loss appeared when group chat was established but there was a very low packet loss for the one to one call. The results are consistent with theory as the group chat scenarios has introduced extreme amount of traffic into LTE network. The overlaps of both scenarios also indicated that there are no effects from each scenario on packet loss rate as shown in figure 4.

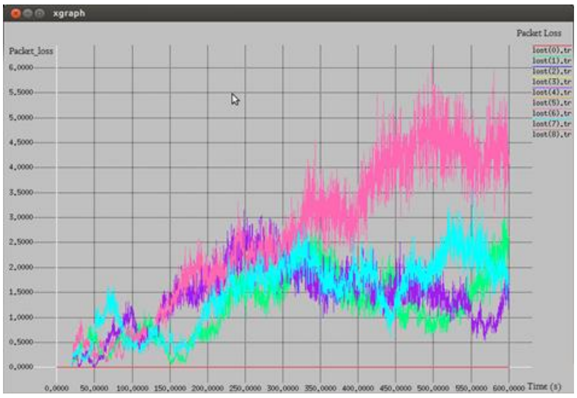

Fig. 4: Packet loss

4.1 Obtained Result

From the computation as shown in table 2, and having compared with ITU-T PRECEPT for voice quality for LTE in table 1, Jitter, Delay and Throughput are found to fall in good range of QoS standards for both scenario one and scenario two when network load was low and also increased, Packet loss fall into the acceptable range of QoS standards.

Table 2: Statistical Results

Qos Parameter	Scenario One	Scenario Two
End-to-end delay (ms)	0.06	0.05
Jitter (ms)	7.00	14.40
Packet loss (%)	1.00	1.00
Throughput (Mbps)	10.52	7.90

Table 3: Literature Comparison

(Mahid et al, 2007) Literature Methodology			Methodology	
Simulator	Opnet		Simulator	Ns2
Scenario	Two		Scenario	Two
Number of user Equipment	Four		Number of user Equipment	Nine
Network type	WiFi/WIMAX		Network type	LTE
User Equipment type	Mobile		User Equipment type	Stationary

CONCLUSION

This work presented a routing algorithm that has been developed and simulated on Ns2 considering Voice over Internet Protocol Long Term Evolution network. The algorithm has been successfully implemented and simulated on the Ns2 and the data for the various Quality of Service parameters; throughput, packet loss, end to end delay and jitter collected from the trace file have been evaluated successfully. The following conclusions were deduced from the result:

Voice over Internet Protocol Long Term Evolution network model was developed using two servers, three access gateways, three enhanced based transceivers and nine user equipments.

Routing algorithm was developed and successfully simulated on Ns2.

The obtained result was compared with VoIP call quality requirement given by the International Telecommunication Union Telecommunication standardization sector (ITU-T). Based on the comparison; throughput with obtained value 10.52Mbps against the ITU-T (0-50)Mbps range, end to end delay obtained value 0.06ms against ITU-T (0-150)ms range and jitter with obtained value 7.00ms against (0-20)ms fall in the good range of QoS while packet loss with obtained value 1% against the ITU-U (0.5-1.5)% range falls in the acceptable range after comparing with the ITU-T QoS standards. From the QoS obtained results and having compared with ITU-T QoS standards, VoIP LTE therefore is a good service for both local and international application because the network offers good Quality of Service.

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