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Throughput Performance Analysis for VOIP over UMTS Networks

Atyaf Mohamed Abdalwahad Mohamed[1], Dr. Khalid Hamid Bilal[2] and Dr. Mustafa Mohamed alhassan[3]

[1]Department of Communication, Faculty of Engineering, Al-Neelain University, Khartoum, Sudan
[2]University of Science and Technology, Khartoum, Sudan
[3]Department of Communication, Faculty of Engineering, Al-Neelain University, Khartoum, Sudan

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Abstract

Now days 3G/UMTS Network had been the challenge in the Network environment. The goal of the project is to study the cellular system of mobile the third generation network UMTS and what the application and services that the UMTS network offers. What is the requirements for VoIP to be implemented on the UMTS, and the operation of VoIP in the UMTS Network. The experiment had been carried out using OP-NET software based on throughput and mean opinion score (MOS) parameters. We studied and discussed the throughput and mean opinion score performance for VoIP in UMTS network in different scenarios, and then compare and analyzing the results to evaluate the performance of VoIP. The results show that the proper adaptation of QoS provides the best performance.

Keywords: *VoIP, OPNET, UMTS, Throughput, MOS.*

1. Introduction

Telecommunication had been the researchers discuss and make a numerous changes in the recent years, especially in the development of mobile broadband. Nowadays communication is going to be popular due to wireless communication; cellular technology is going to be emerged rapidly. Based on the demand of the users, next generation cellular system is being able to provide a collection of applications for users. The Universal Mobile Telecommunication (UMTS) System is one of the features of cellular communication that is able to offer the different types of services as per user contentment. Voice and video conferencing have a great deal of demand over the cellular network. Cellular networks are also emerging with an improved level of Quality of Service (QoS) and better mobility and improvement in voice and video transmission over different technologies .To obtain better performance to satisfy the users, proper adaptation and improve the quality of service (QoS) is required [1].

The Universal Mobile Telecommunications System (UMTS) is an umbrella term that encompasses the third generation (3G) radio technologies developed and maintained by the 3GPP (3rd Generation Partnership Project). UMTS is a network standard used throughout much of the world as an upgrade to existing GSM mobile networks. The technology described in UMTS is sometimes referred to as Freedom of Mobile Multimedia Access (FOMA) [3] [4] or 3GSM.

UMTS is a component of the International Telecommunications Union IMT-2000 standard set and compares with the CDMA2000 standard set for networks based on the competing CdmaOne technology. UMTS uses wideband code division multiple access (W-CDMA) radio access technology to offer greater spectral efficiency and bandwidth to mobile network operators. [2], a technology that shares much with CDMA networks used throughout the world, though it is not compatible with them. [5]

1.1 Architecture of the UMTS Network

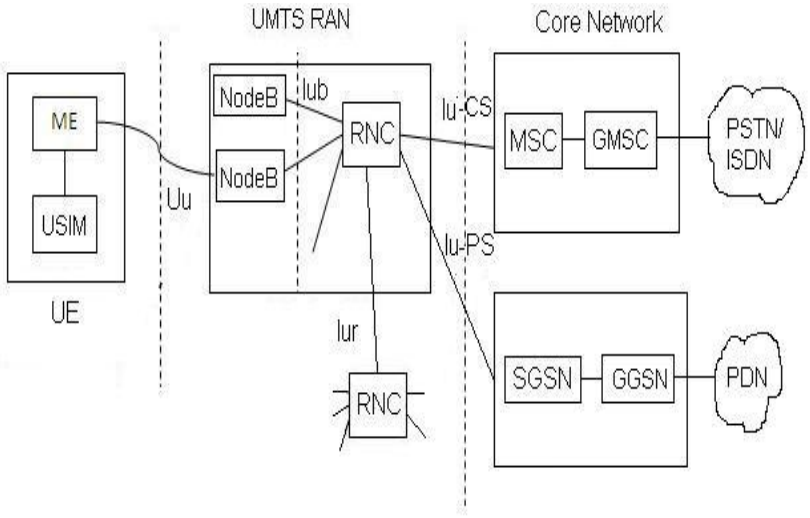


Figure1: UMTS Network Architecture [6]

The Universal Mobile Telecommunications System represents a complete system, which includes:
User equipment (UE): It is a terminal, which help a user to access the network. Also it involves mobile equipment and universal service identification module (USIM). Terminal equipment must have an application programming interface (API). USIM provides personal mobility.

UTRAN(UMTS Terrestrial Radio Access Network): Which includes one or more of the RNS. Each RNS involves a radio network controller (RNC) and one or more Node (B).

Radio network controller (RNC): RNC is responsible for managing radio resources of a set of cells. It is equivalent of BSC networks GSM / GPRS, but more independent.

Node B: Provides transmission and reception of signals in one or more cells in the network. It is also responsible for monitoring the power level of the cells.[7]

Core Network: (Mobile Application Part) :The 3G/ UMTS core network architecture is a migration of that used for GSM with further elements , whowever to enable the additional functionality demanded by users . UMTS core network is split into two parts:
Circuit switched elements: These are primarily based on the GSM network entities and carry data in a circuit switched manner, which include the following:

Mobile switching center (MSC): It is basically the same as that in GSM, and it manages the circuit switched calls.

Gateway MSC (GMSC): It is the interface to the external networks. [8]

Packet switched: These are designed to carry packet data, which enables much higher network usage as the capacity can be shared and data is carried as packets which are routed according to their destination. It includes the following:

- Serving GPRS Support Node (SGSN): It is a packet switch and router in the domain of PS core network. Also it controls the access between a mobile station and the network and routes packets to the correct BSC / RNC .And it serves as the mobility management, such as registering the location of the adjustment region routing and paging. SGSN also carry out the functions of privacy such as authentication and encryption.
- Gateway GPRS Support Node (GGSN): It is act as a packet router in the domain , and it is the gateway between routing IP packets to the mobile network UMTS and IP packet routing in fixed networks of the Internet. Also it is storage user's data for active UE , and implement the functions of privacy such as firewalls and filtering. [7]

Voice over IP (VoIP) is a group of technologies for the delivery of voice communications and multimedia over Internet Protocol (IP) networks. Other terms commonly associated with VoIP are IP telephony, Internet telephony, broadband telephony, and broadband phone service. Mobile

VoIP works with a cell phone's or other Internet service is send voice calls as digital signals over the Internet using voice over IP technology. Digital data transmission using VoIP is also typically faster, as the data is spread out over multiple packets; each of them is taking the fastest route to its intended destination.

The purpose of this thesis is to study the UMTS Network features and services, however to evaluate the QoS performance of the 3G/UMTS network based on the performance metrics of Throughput and mean opinion score(MOS) .

Throughput: is known as the rate packets that were successfully delivered through the channel communication to the destination, it is usually measured in bits per second (bit/s or bps).

Mean Opinion Score (MOS) : is attesting that provides a numerical index to illustrate the quality of service in the phone networks observed by users or destinations reception.

Table 1: The classification of the MOS

MOS	QUALITY	Impairment
5	Excellent	insignificant
4	Good	Perceptible
3	Fair	lightly annoying
2	Poor	Annoying
1	Bad	extremely annoying

2. Methodology

To evaluate the performance of VoIP application in the UMTS network in the packet switched part of the network by monitoring the VoIP Traffic across the network we have used OPNET Modeler, in our simulation; and designed different scenarios for conference and measure the performance of throughput ratio according to the Codec's. G729A and GSM Codec, as well as meet the goal to analysis the quality in 3G/UMTS wireless networks. To evaluate the results for successful transmission, and comparing the VoIP performance for different number of nodes in the network, this can be done by capturing the packet sent and packet received of VoIP Traffic and calculating the value of throughput for the VoIP traffic.

3. Mathematical Model

Mathematical equation of throughput ratio is given by:

$$\text{throughput} = \frac{\sum \text{total traffic}}{\text{time (sec)}} \text{.....(1)}$$

Mathematical equation of mean opinion score (MOS):

The MOS is calculated using a non-linear mapped R factor:

$$\text{MOS} = 1 + 0.035R + 7 \times 10^{-6}[R(R-60)(100-R)] \text{.....(2)}$$

Where:
MOS is the mean opinion score

$$R = 100 - Is - Ie - Id + A$$

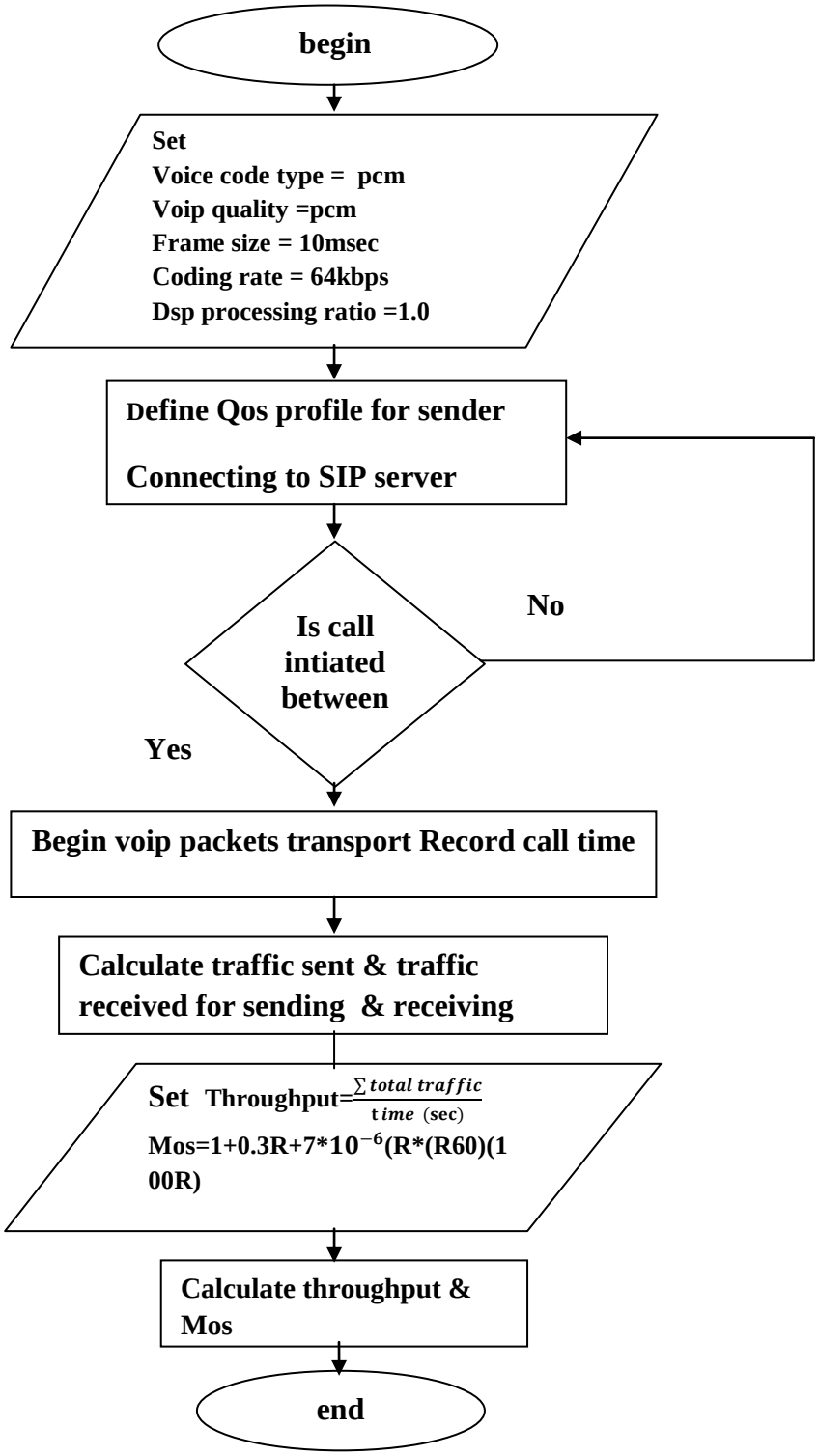
Is: voice signal impairment effects.

Ie: impairment losses suffered due to the network and codec's.

Id: impairment delays particularly mouth-to-ear delay.

R: mean opinion score factor.

Computer Model:



4. Simulation Environment

The simulation of VoIP application over the UMTS Network with the OPNET Software was completed under the following UMTS Network parameters.

Table 2: Simulation Parameters

Voice codec type	**PCM**
Name	G.711
Frame size(sec)	10msec
Lock ahead size(sec)	0 sec
DSP Processing ratio	1.0
Coding rate (bits/sec)	64 kbps
Speech activity detection	Disabled
Voice over IP Call	PCM Quality
Start time offset (seconds)	Constant (0)
Duration (seconds)	End of simulation
UMTS PDCP Compression	Disabled
UMTS RLC Processing time	0.0015
UMTS CPICH transmission power	1.0
Path loss Model	Outdoor to indoor environment
UMTS Node-b ID	Auto assigned

5 . Simulation

After setting the simulation parameters for the UMTS network with VoIP Application , the screenshots from the OPNET software shows UMTS Network scenario with different number of nodes ,and configuration

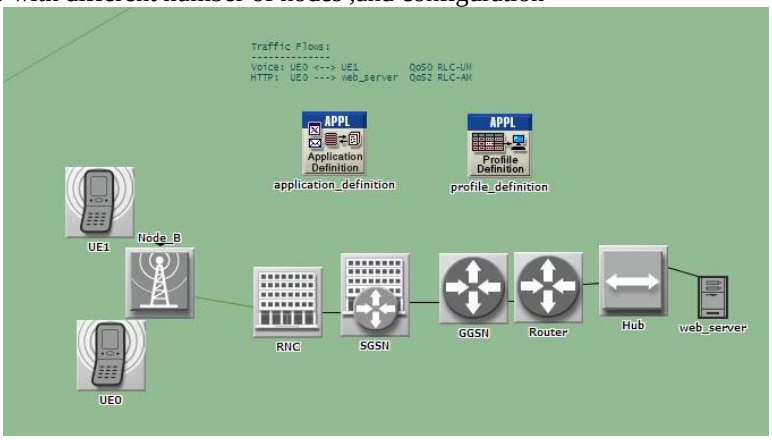


Figure 2 : Network Scenario with two Nodes

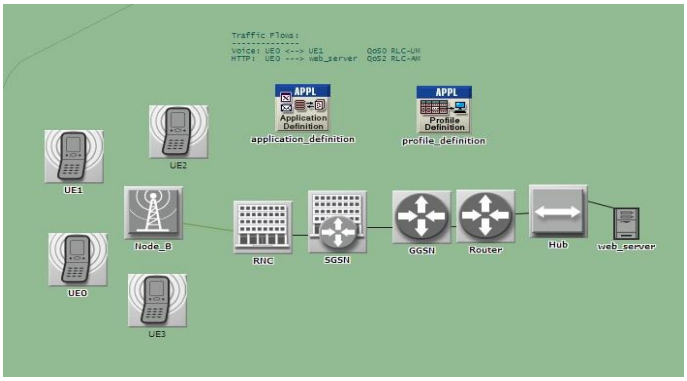


Figure 3: Network Scenario with 4 Nodes

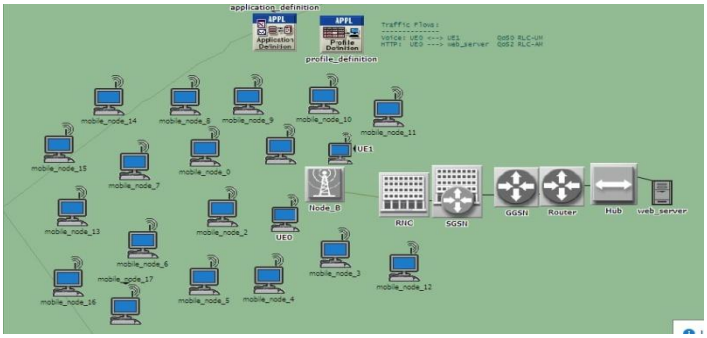


Figure 4: Network Scenario with 20 Nodes

6. Results

The simulation was completed with the OPNET Software and the following results was extracted from the software for three scenarios. Results for 2 Nodes

Table 3: Results for 2 Nodes

Simulation Time(min)	MOS	Throughput
6	2.95	0.967321058
8	2.94	0.977206797
16	2.95	0.977830562
30	2.98	0.978439828
60	3.02	0.978798587
300	3.15	0.979381443
600	3.21	0.979665072
Average	3.02	0.976949049

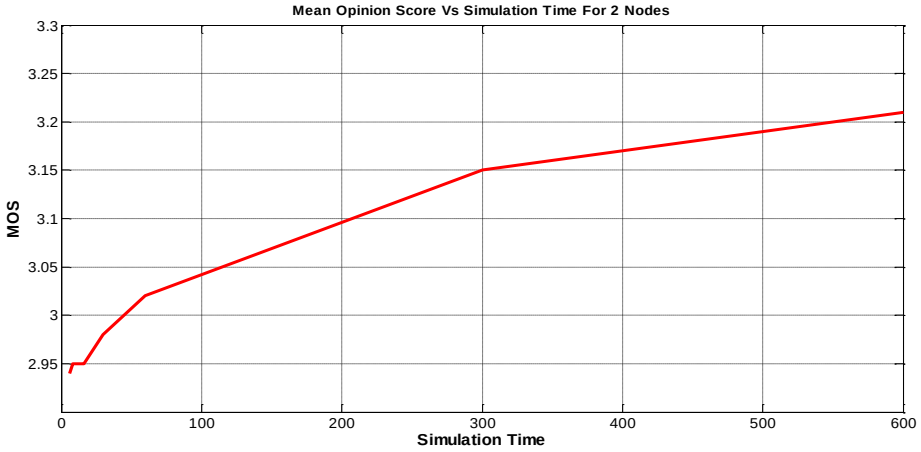


Figure 5: Simulation Time vs MOS for 2 Nodes

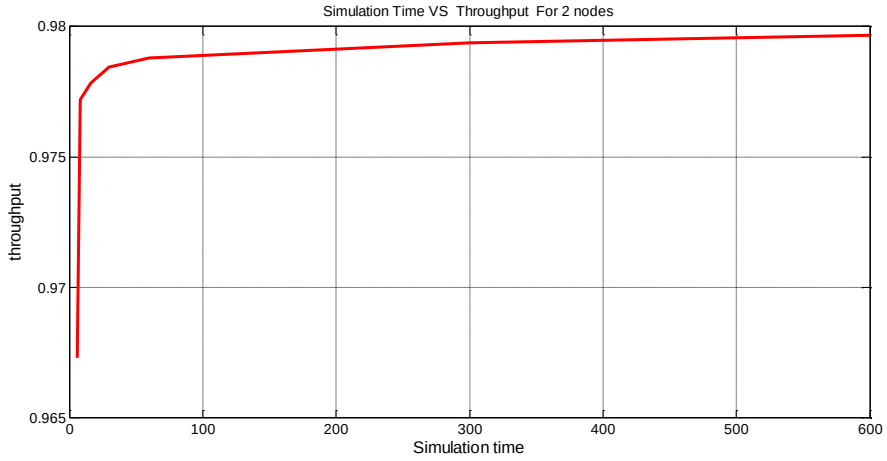


Figure 6: Simulation Time vs Throughput for 2 Nodes

Results For 4 Nodes

Table 4: Results for 4 Nodes

Simulation Time	MOS	Throughput
6	2.49	0.886455696
8	2.52	0.920816327
16	2.55	0.895505618
30	2.61	0.926235741
60	2.74	0.925283019
300	2.83	0.919548872
600	2.95	0.892369478

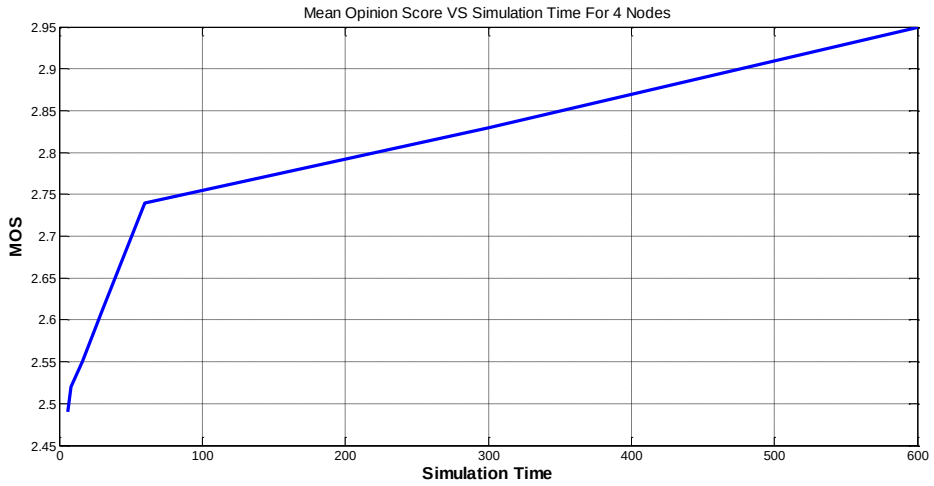


Figure 7: Simulation Time vs MOS For 4 Nodes

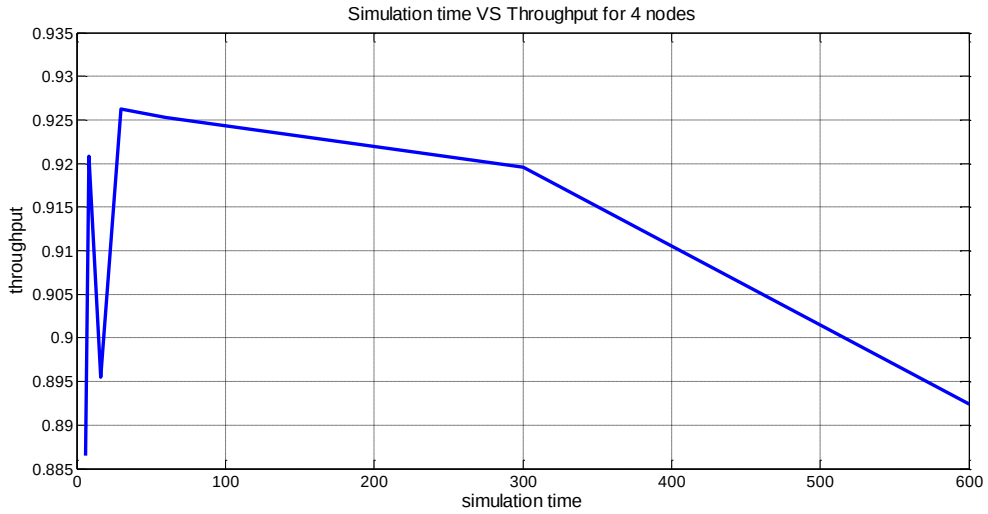


Figure 8: Simulation Time vs Throughput for 4 Nodes

Results for 20 Nodes

Table 5: Results for 20 Nodes

Simulation Time	MOS	Throughput
6	2.32	0.783467742
8	2.35	0.858943089
16	2.46	0.894921875
30	2.49	0.899242424

60	2.51	0.89739777
300	2.53	0.887272727
600	2.59	0.873476703

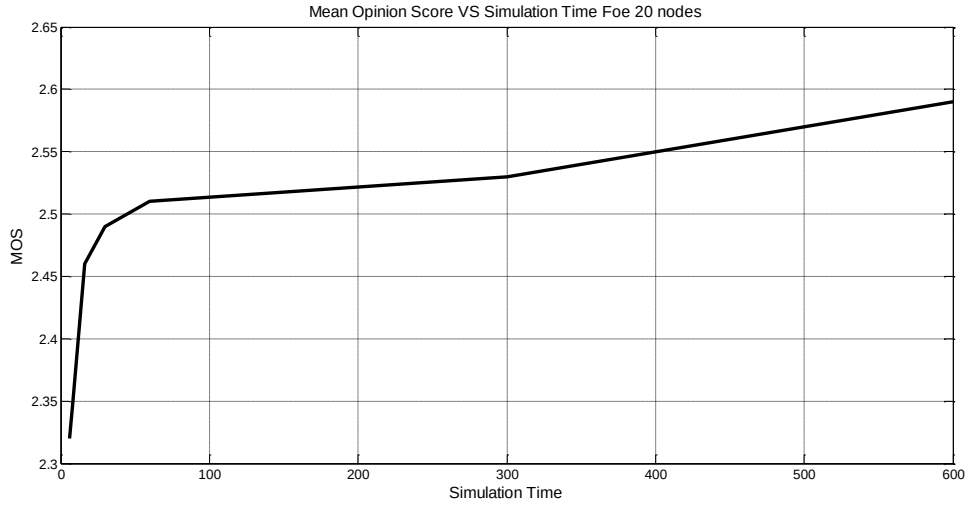


Figure 9: Simulation Time vs MOS For 20 Nodes

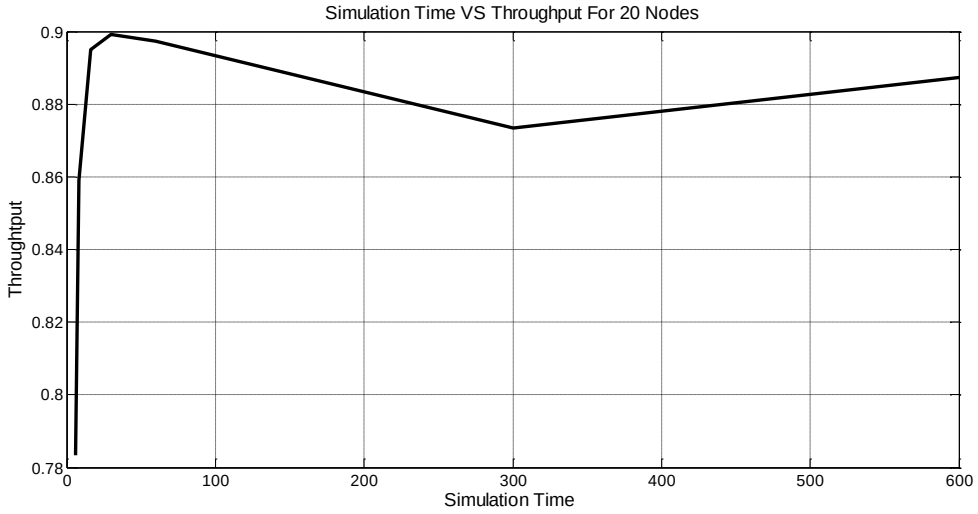


Figure 10: Simulation Time vs Throughput for 20 Nodes

7. Result Discussion

For 2 nodes: The results from table (3) we observe the throughput performance during time of the simulation is in high ratio and the ratio of the throughput is increasing proportionally with the simulation time according an average value, the values of the mean opinion score is also increase proportionally with the time of simulation with average value, which is very good. From the graph on figure(5) of the mean

opinion score we see that the line is representing an exponential function .The graphical throughput representation on figure(6) shows that the values of throughput is high and close to each other, in the end the line approach to steady value.

For 4 nodes : The results from table (4) and figures(7,8) the mean opinion score is also increasing proportionally with time while the throughput ration is decreasing with time , as time goes on the amount of traffic sent will be increased which will lead to probability of losing packets an so the throughput ratio will be decreased on time .

For 20 nodes: The results from table (5) and figures (9, 10) the mean opinion score value is less and the throughput ratio is less due to large number of nodes in the network which affects the performance of VoIP.

8. Conclusion

The simulation is done by creating a virtual environment for the UMTS network to simulate the network and the application within it, also to study and analyze the performance of the VoIP Application of the UMTS network has been completed using the OPNET software under the parameters shown in table(2).The simulation results from the results we conclude that the performance of the VoIP is affected by number nodes , more nodes means more traffic on network , and more load witch lead to congestion in the network the congestion degrades the quality of service and the throughput ratio will not be high.

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