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# Packet Delay Variation Performance Analysis for VOIP Over UMTS Networks

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**Abstract**

With the widespread availability of advanced mobile phones and Pocket PCs, the need for VoIP applications on these mobile platforms is tangible. In this thesis, we evaluate Voice over IP service over UMTS network in terms of feasibility and quality. In order to easily implement an IP-based service on UMTS technology, we used the GPRS service, which provided by the UMTS operators. Following this we build different scenarios of VoIP using op-net software with different number of nods and cells to measure the performance of Packet Delay variation and mean opinion score (MOS) in voice over IP . We studied and discussed the Packet Delay variation and mean opinion score performance for VoIP in UMTS network in the 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, Packet Delay Variation, MOS.*

## 1. Introduction

Nowadays communication is going to be popular due to wireless communication; cellular technology is emerged rapidly, based on the demand of the users. Cellular systems accommodate a large number of mobile units over a large area within a limited frequency spectrum. A cellular radio system provides a wireless connection to the public telephone network for any user location within the radio range of the system.[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 [3]. 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 Cdma One technology [2]. The technology described in UMTS is sometimes referred to as Freedom of Mobile Multimedia Access (FOMA) [4] or 3GSM. 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]

The UMTS System Concept enables network operators to provide universal mobile telecommunications coverage in diverse as multi-operator, multi-vendor, It is provides for interworking with other telecommunications systems, including private networks such as office systems. Multiple advanced the UMTS System Offers many services such as high bit rate, multimedia services, and the service requirement of the users in a consistent and personalized manner wherever he is located. The UMTS system concept supports adaptive terminals able to access different radio access technologies, with the capability of being upgraded with the passage of time and support end to end negotiation to establish and maintain certain communication capability.[6]

With the widespread availability of advanced mobile phones and Pocket PCs, the need for VoIP applications on these mobile platforms is necessary. VoIP technology has become prevalent today due to its lower cost than traditional telephony and its ability to support new value-added services. Additionally, the increasing availability of wireless internet access has led to research studies examining the combination of wireless network access with voice over IP. [7]

Voice over Internet Protocol (VoIP) is a group of technologies and protocols that using the digital data network to sent the voice call, which analogue signal conversion into a digital signal so that it can over the network. By offering Voice over IP moving toward through network, where voice, data, and video all travel along the digital data network. Digital data transmission using VoIP is also faster, as the data is spread out over multiple packets; each of them is taking the shortest and fastest path to reach the destination. Other terms commonly associated with VoIP are IP telephony, Internet telephony, broadband telephony, and broadband phone service.

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

Packet Delay Variation: Is the delay in the expected time for the arrival of packets required for the destination, whoever the sent packets over the network do not arrive at the same time but at different times, and this difference is called Packet Delay Variation.

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

We have used OPNET Modeler, in our simulation; we have designed different scenarios for voice conference and measure the performance of delay in voice packet according to the Codec's. G729A and GSM Codec, as well as meet the goal to analysis the quality voice over the internet protocol in 3G/UMTS wireless networks. Then we will evaluate the results improving the QoS to support in 3G network for successful transmission.

## 3. Mathimatical Model

(A) Mathematical equation of delay

$$\boldsymbol{PDV} = \{\sum_{i-1}^{n}([\mathbf{t'(n)} - \mathbf{t(n)}] - \boldsymbol{\mu})\mathbf{\hat{}2}\}/\mathbf{n} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (\mathbf{1})$$

Where:

PDV is the packet delay variation, μ is the average delay of *n* selected packets.

(B) Mathematical equation of mean opinion score (MOS):

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

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

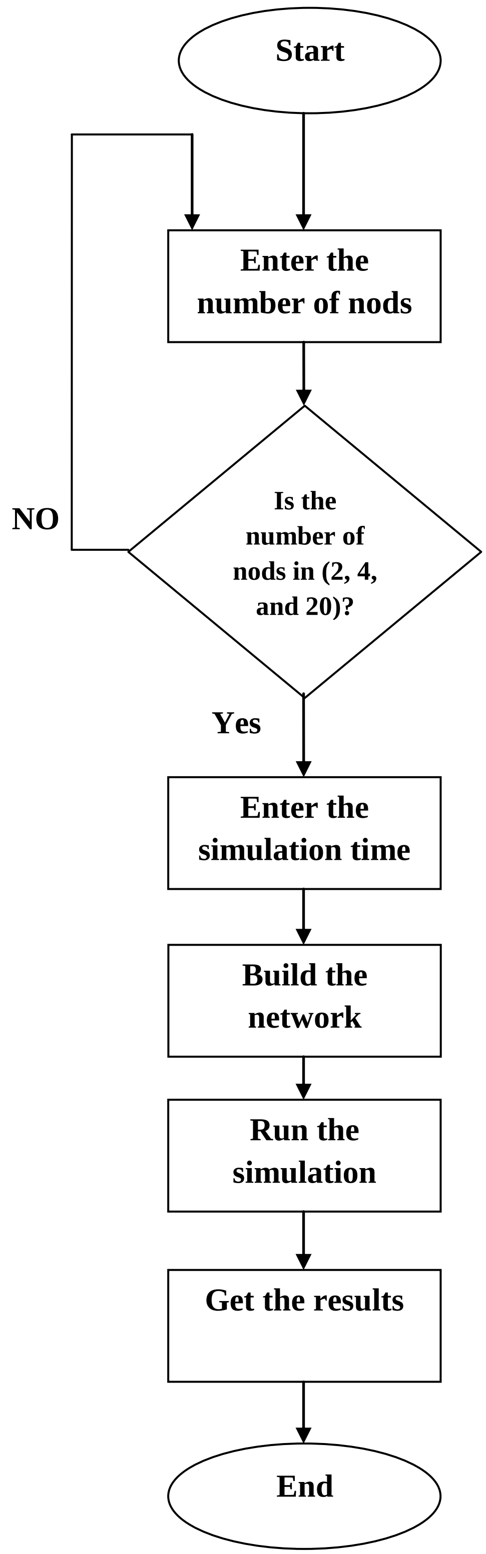
Where:

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. Simulator environment

The tool used for simulations is OPNET Modeler as it provides the results very closer to the real time environment. The models were created by selecting the nodes and links from the object palette such that to reduce the losses and impairments effects.

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

**Table 2: The environment that has the simulation depending upon**

| **Parameter** | **value** |
|---|---|
| Network | UMTS |
| Area | 0.5-5KM |
| Number of nods | 2 , 4, And 20 |
| Number of cell | 1,5 |
| Cell radius | 0.5KM |
| Power | 1 |
| Frequency | 900GHZ |
| Simulation time | (10-50)M ,(1-10)H & 24H |
| Data rate | 240Kps |
| Maximum Bit rate for uplink & downlink | 64Kps |
| 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 |
| Start time offset (seconds) | Constant (0) |
| Duration (seconds) | End of simulation |
| UMTS PDCP Compression | Disabled |
| UMTS RLC Processing time | 0.0015 |

## 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.

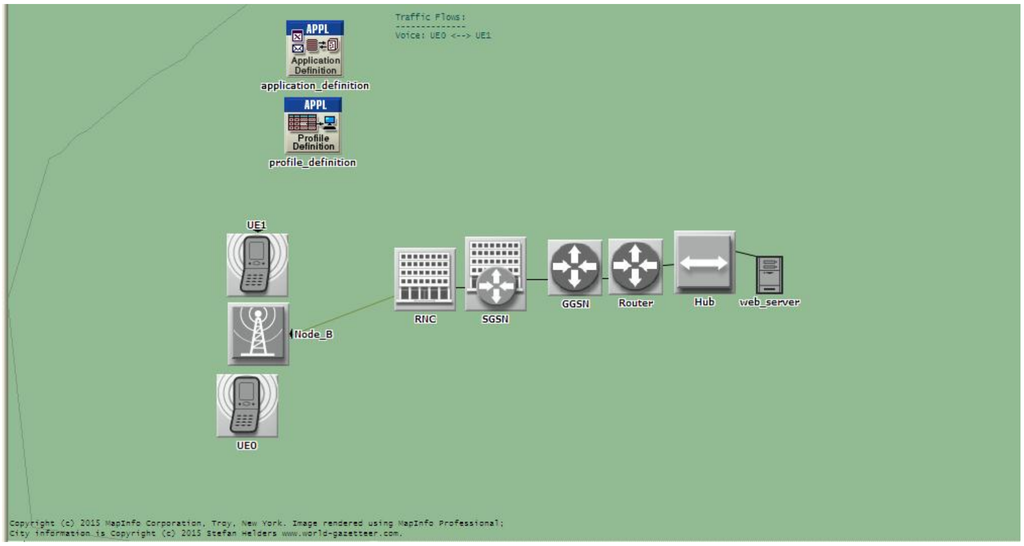



**Figure 1: The simulation of two nods**

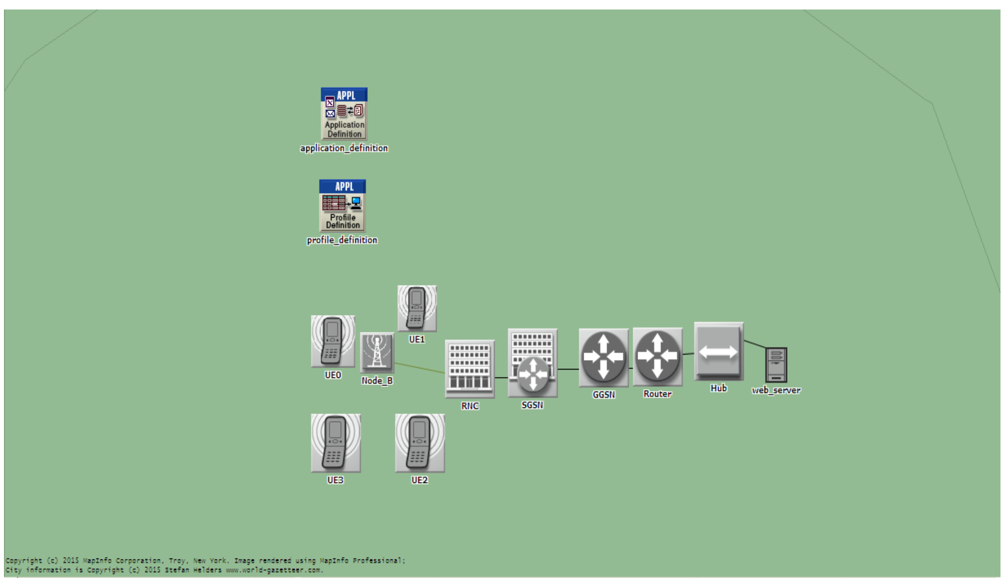



**Figure 2: The simulation of four nods**

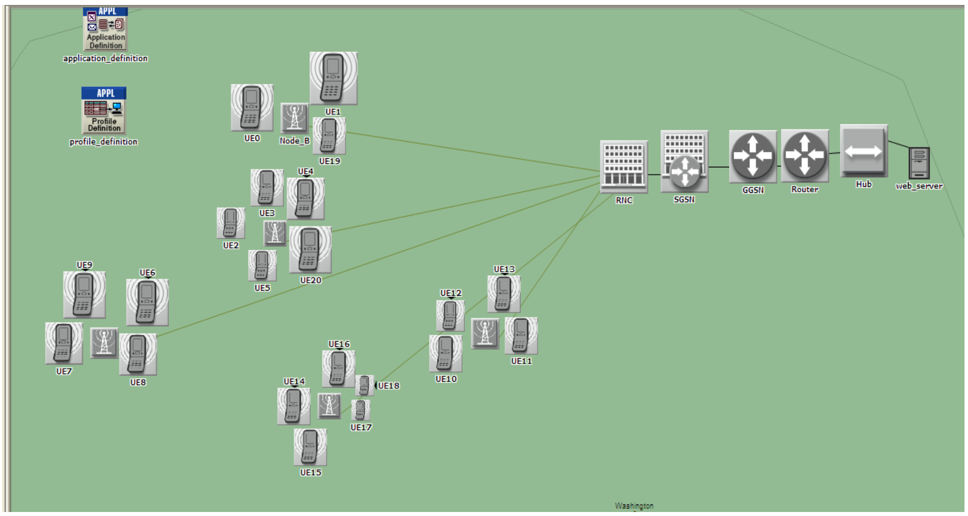


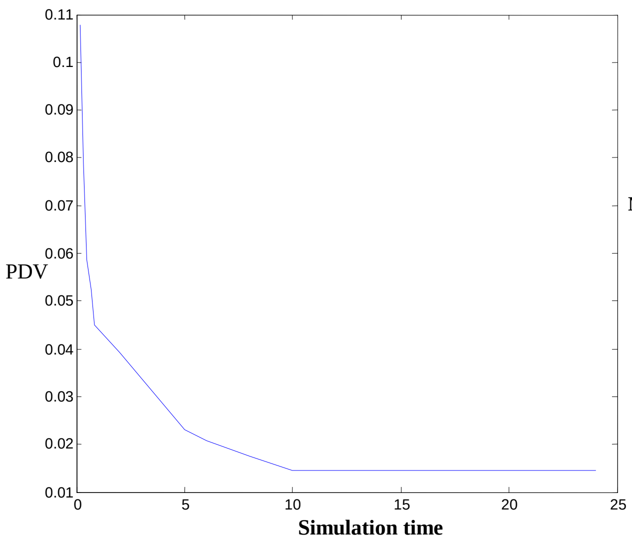

**Figure 3: The simulation of twenty nods**

## 6. Results

Results for 2 nodes

**Table 3: The results of two nodes**

| SIMTIME | delay variation | MOS |
|---|---|---|
| 10m | 0.1078146 | 2.960107 |
| 20m | 0.0788871 | 2.979722 |
| 30m | 0.0585779 | 2.999362 |
| 40m | 0.0523035 | 3.009484 |
| 50m | 0.0449875 | 3.022075 |
| 1H | 0.0476633 | 3.035406 |
| 2H | 0.039224 | 3.08155 |
| 3H | 0.0243392 | 3.123666 |
| 4H | 0.0335879 | 3.130718 |
| 5H | 0.0229664 | 3.160769 |
| 6H | 0.020709 | 3.175825 |
| 7H | 0.0204622 | 3.217008 |
| 8H | 0.0175758 | 3.225224 |
| 9H | 0.0171872 | 3.209574 |
| 10H | 0.014597 | 3.21933 |
| 24H | 0.015044 | 3.307865 |



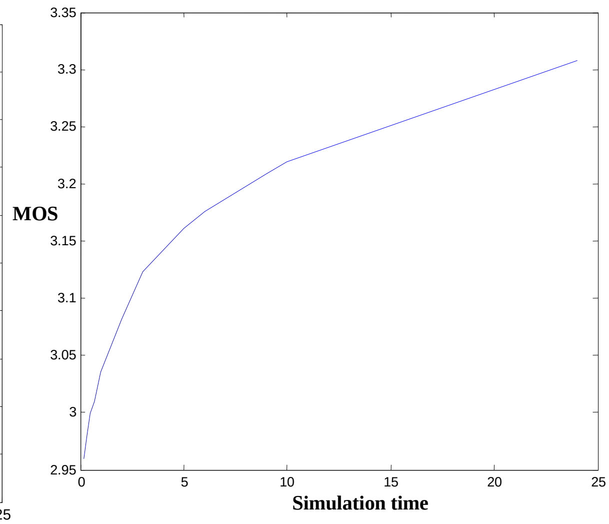
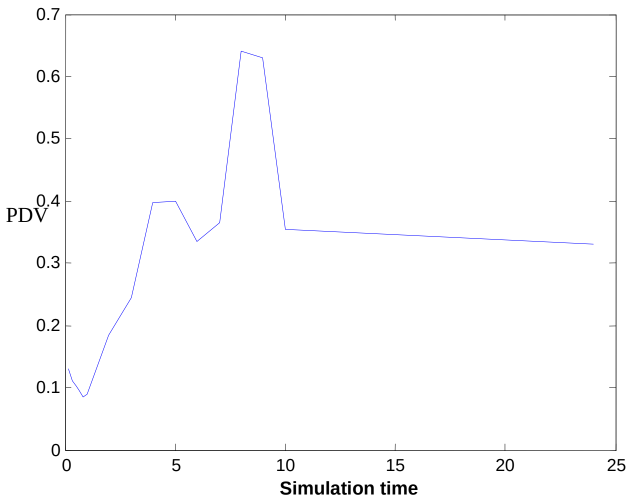




**Figure 4: The delay variation for two nodes** **Figure 5: The MOS for two nodes**

Results for 4 nodes:

**Table 4: The results of four nodes**

| SIMTIME | delay variation | MOS |
|---|---|---|
| 10m | 0.129895 | 2.947964 |
| 20m | 0.110948 | 2.971245 |
| 30m | 0.102901 | 2.989744 |
| 40m | 0.096516 | 3.002197 |
| 50m | 0.085701 | 3.014589 |
| 1H | 0.089562 | 3.032768 |
| 2H | 0.184515 | 3.079201 |
| 3H | 0.244227 | 3.124257 |
| 4H | 0.398283 | 3.119 |
| 5H | 0.399998 | 3.15946 |
| 6H | 0.33428 | 3.174862 |
| 7H | 0.364249 | 3.217402 |
| 8H | 0.639899 | 3.22439 |
| 9H | 0.630013 | 3.209043 |
| 10H | 0.353355 | 3.219586 |
| 24H | 0.331157 | 3.30787 |



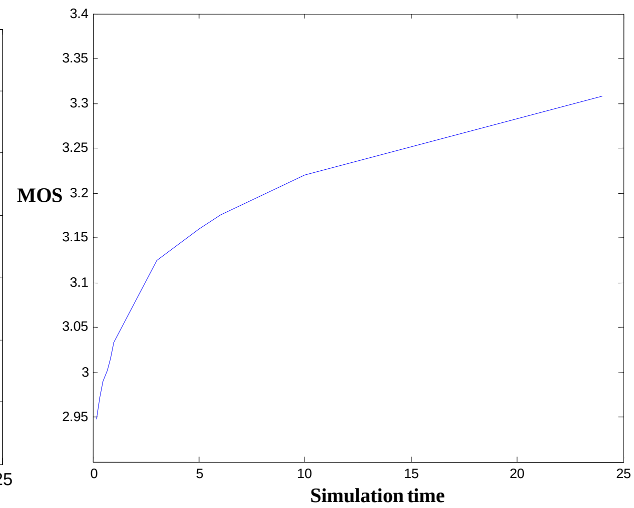




**Figure 6: The delay variation for four nodes**

**Figure 7: The MOS for four nodes**

Results for 20 nodes:

**Table 5: The results of 20 nodes**

| SIMTIME | delay variation | MOS |
|---|---|---|
| 10m | 0.016072 | 2.964587 |
| 20m | 0.0215 | 2.981722 |
| 30m | 0.022688 | 2.995191 |
| 40m | 0.026337 | 3.003791 |
| 50m | 0.026564 | 3.014355 |
| 1H | 0.028389 | 3.032355 |
| 2H | 0.034308 | 3.074199 |
| 3H | 0.02913 | 3.115963 |
| 4H | 0.033682 | 3.123679 |
| 5H | 0.025586 | 3.15253 |
| 6H | 0.030601 | 3.167717 |
| 7H | 0.051629 | 3.209737 |
| 8H | 0.227422 | 3.218188 |
| 9H | 0.118746 | 3.209527 |
| 10H | 0.267008 | 3.21229 |
| 24H | 0.550407 | 3.307908 |

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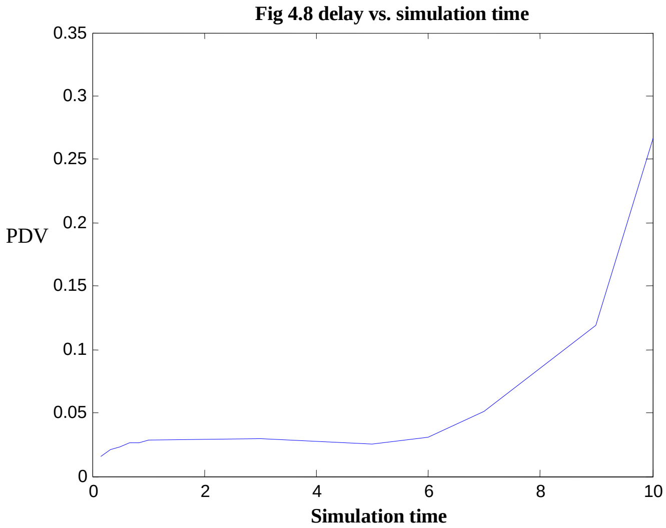


**Figure 8: The delay for twenty nodes**

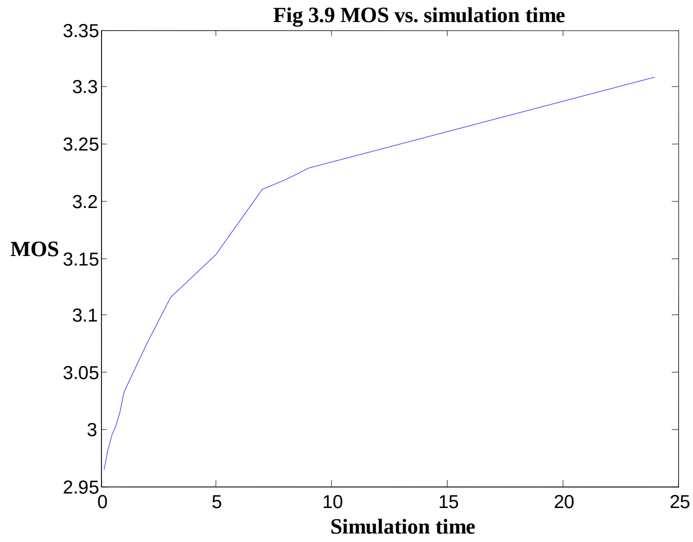


**Figure 9: The MOS for twenty nodes**

## 7. Result Discussion

Tables (3, 4, 5) shows the simulation time that been used and the packet delay variation in the defined time and the MOS associated with it.

Figure (4) shows the packet delay variation for the two nodes verses the simulation time we can see that the packet delay variation is decreasing as a decreasing exponential function when the simulation time is increasing.

Figure(5) shows the MOS for the two nodes verses the simulation time we can see that the MOS is increasing as increasing exponential function when the simulation time is increasing. We observed that the quality of the network is in between fair and good network quality.

Figure(6) shows the packet delay variation for the four nodes verses the simulation time we can see that from 0 up to one hour the packet delay variation is decreasing as a decreasing exponential function, from 1H up to 4H the it is increasing as increasing exponential function, from 4H up to 5H the it is remain constant, from 5Hup to 6H it is decreasing by 0.065718,from 6H up to 9H it is increasing as increasing exponential function, from 9H up to 10H the delay is decreasing by 0.276657 and from 10H to 24H it was decreased by 0.022198. The average is 0.280969 ±0.145658 (The stander deviation).

Figure(7) shows the MOS for the four nodes verses the simulation time we can see that the MOS is increasing as increasing exponential function when the simulation time is increasing. We observed that the quality of the network is good.

Figure(8) shows the packet delay variation for the twenty nodes verses the simulation time we can see that from 0 up to 2H it is increasing by 0.004497and from 2H up to 10H it is increasing as an increasing exponential function. The average is 0.094379 ± 0.086893 (The stander deviation).

Figure(9) shows the MOS for the twenty nodes verses the simulation time we can see that the MOS is increasing as increasing exponential function when the simulation time is increasing. We observed that the quality of the network is good.

## 8. Conclusion

The goal of the project is to study the VoIP application over the UMTS Network. The experiment had been carried out using OP-NET software based on packet delay variation and mean opinion score (MOS) parameters. The parameter which was taken in our simulation was shown in table (2). After the execution of the simulator we get the results in terms of tables and graphs. We studied and discussed the performance for VoIP in UMTS network in different scenarios, and then compare and analyzing the results to evaluate the performance of VoIP. From the results we observed the following:

- When the number of nods increases its causes that the value of packet delay variation is increased.
- The delay affected by the data rate, as much the data rate increased the delay is decreased.
- We observed that the quality of the network is good.

## References

[1] Kamil Sh. Zigangirov," Theory of Code Division Multiple Access Communication", Institute of Electrical and Electronics Engineers, 2004.
[2] Creativentechno, https://creativentechno.wordpress.com/2012/01/03/the-generations-1g-2g-3g-4g/.
[3]Techopedia.com,http://www.techopedia.com/definition/5092/universal-mobile-telecommunications-system-umts.
[4] Marola Yousri Yousif Masood , 2Dr. Amin Babiker A/Nabi Mustafa 3Dr. Ahmed Salah Abdallah "UMTS vs LTE Coverage by signal level and overlapping zone A Comparative Study" IJRDO-Journal Of Electrical And Electronics Engineering, VOL 2 ISSUE 2 February 2015.
[ 5] Kaaranen, Ahtiainen, Laitinen, Naghian, Niemi" UMTS Networks – Architecture, Mobility and Services"2nd edition, Wiley 2005.
[6]V. Mancuso, I. Tinnirello ,UMTS CoreNetwork ,http://www.tti.unipa.it/~ilenia/course/13-umts-core.pdf.
[7] Homayoun,Derakhshanno," Voice over IP over GPRS" Stockholm, April 30th 2008.