Performance Evaluation of Quality Parameters in VOIP and PSTN Systems

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Abstract— **In the present scenario of Communication, research is going on to transmit voice, data, video, etc at high speeds. One of the way to achieve this is to use IP networks based on VOIP Technology. VOIP based systems play an important role for fast data communication. Thus, VOIP based systems have a great rising future in the telecom field. As of now PSTN systems are being used for voice communication but with the advent and development of VOIP based systems, it will gradually decrease and will come to an end. In this paper various QOS parameters are discussed briefly and performance of PSTN and VOIP systems are evaluated by analyzing their QOS parameters. Simulation analyses of both systems are done using OPNET MODELER 14.5 software. Different scenarios are considered for both systems and its different QOS parameters like delay, Mean Opinion Score, Packet loss ratio and Jitter are examined and analyzed on the basis of which the usefulness of Systems is decided and its impact on the QOS parameters is evaluated.**

Keywords-VOIP,PSTN,MOS,End-to-End,Delay,Jitter,PacketLoss

I. INTRODUCTION

VoIP (Voice over Internet Protocol) is an advanced technology that has a great potential to develop new telecommunication with much lower cost and better QoS. This paper presents a detailed simulation approach for deploying VoIP successfully. The simulation uses the OPNET 14.5 network simulator which has gained a considerable popularity in both academia and industry. Nowadays, there is a growing trend in real time voice communication using Internet protocol (IP). Voice over Internet Protocol (VoIP) is a technology that allows delivery of voice communications over the Internet or other packet switched networks rather than the traditional Public Switched Telephone Network (PSTN). Many VoIP applications are available on the internet such as Skype, Viber, Tango, and Yahoo messenger. In VoIP the analogue voice signal of the transmitter is converted into digital form before compression and encoding it into stream of IP packets for transmission to the receiver over IP network. At the receiving end, Digital to Analogue Converter (DAC) works on regenerating the original analogue voice signal after reassembling the received IP packets [1]. There are many authors who have worked on various Quality of Service (QOS) parameters using different service classes in different network types. Here a basic knowledge and comparison of PSTN and VOIP systems is given, then using OPNET Modeler 14.5 software two scenarios was designed for PSTN systems and for VOIP systems and there Quality of service parameters were analyzed. The reason behind it is that it would reflect the advantage of VoIP over the traditional PSTN and thus proving that VoIP would be an ideal candidate for the modern technology in network communication. In the perspective of a good design of VoIP, the most important factor would be the quality of service (QoS) provided to all users on the network, while considering medium-to-high traffic loads that is most likely to occur in reality. However, factors that might influence the overall performance of VoIP are bandwidth, end-to-end delay, jitter, packet lost rate and utilization [2]. This paper is an attempt to address some important factors and presents an approach based on simulation. Here different scenarios are considered for both systems and analysis is done on its QOS parameters. The analysis is done using OPNET 14.5 network simulator. OPNET Simulator contains a vast amount of models of commercially available network elements, and has various real-life network configuration capabilities. This makes the simulation of real-life network environment close to reality. Other features of OPNET include GUI interface, comprehensive library of network protocols and models, source code for all models, graphical results and statistics, etc.

The rest of the paper is organized as follows. Section II gives meaning of QOS parameters used in VOIP systems and a brief review of the related work. Section III presents designing and description of OPNET-based Network models of PSTN and VOIP systems. Section IV presents Simulation results and analysis. Finally, section V concludes the paper.

II. IMPORTANT VOIP QOS PARAMETERS

Before going further it is important to know the   meaning of Quality of Service Parameters used in VOIP. Here a brief definition of QOS parameters is given.

2.1. Packet Loss

First parameter which is considered is Packet Loss. It is defined as Failure of one or more packets to reach their destination across the network is recognized as packet loss. The packet loss depends a lot on packet size. Packet loss is related to end-to-end delay and is present in almost every network and is extremely noticeable with real time streaming technology such as Skype and online gaming. If packet size becomes higher, the delay becomes shorter. Packet loss can severely degrade the voice quality. Higher packet loss can be reduced by decreasing packet size. The possible causes that lead to packet loss, such as channel congestion, corrupted packets rejected in-transit and poor networking hardware. To properly recover the loss packets, reliable network transport protocol such as TCP are used to insure an acceptable and stable transmission. Packet loss is due to congestion in network, altering and inconstant delay and indigent network. Factors which cause packet loss are bit error, lack of queuing space, indecent capacity of buffer and de-buffer and end *to* end transmission delay. If a packet is lost during transmission it

cannot be recovered and will end up causing the voice being unsatisfactory, resulting in disturbed quality of conversation.

2.2. Latency

Another parameter which is considered is Latency. Latency is the average time it takes for a packet to travel from its source to its destination. A person who is speaking into the phone called the source and the destination is the listener at the other end. This is one-way latency [4]. Ideally, the delay should be as low as possible but if there is too much traffic on the line (congestion), or if a voice packet gets stuck behind a bunch of data packets (such as an email attachment), the voice packet will be delayed to the point that the quality of the call is compromised [5]. The Maximum amount of latency that a voice call can tolerate one way is 150 Milliseconds (0.15 sec) but is preferred be 100 Milliseconds (0.10 sec) [6].

VoIP refers to the integration of telephone services with the growing number of other IP-based applications; a digital telephone service that uses the public Internet as well as private networks instead of the traditional telephone network. A VoIP system simply converts analog signals such as telephone calls into digital IP packets and distributes these packets across Internet or any other packet-switched network.

VoIP is currently considered one of the most important technologies for telecommunications. VoIP is expected to accelerate the development of rich multimedia services. Some of the most obvious characteristics that make VoIP a very popular telephony alternative as compared to PSTN Systems are listed below

> Very efficient use of the data bandwidth, equipment and transmission lines. VoIP technology allows the transmission of more than one telephone call and data over the same broadband connection. Additionally, speech can be encoded using different algorithms in accordance with channel capacity and quality requirements, permitting a more controlled and efficient use of the channel.

> Generally, the cost of VoIP services is low when compared to traditional telephony and features offered.

> Location independence. Only an Internet connection is needed to get a connection to a VoIP provider or server.

> Integration with other services available over the Internet, including video conversation, message or data file exchange in parallel with the conversation, audio conferencing, managing address books, and notification of availability.

2.3. Delay

The third parameter taken into consideration is Delay. It is the total transit time for packets in a data stream to arrive at the endpoint and it is inevitable in communication system. Delay time is one of the most important factors in determining the quality of a call. Echoing has been a major problem that is caused by delay. Typically, for getting optimum VoIP call quality, delay must be less than 150ms.

2.4. Jitter

Last parameter considered is Jitter. It is the name given to the variation in delay that occurs between two successive packets. It is mainly observed because the packets have to be multiplexed upstream in the network. It occurs when the packets need to be queued. Several types of traffic traverse the IP network Jitter is one of the most common VoIP problems Jitter is the undesired time delay for the packets sending end to receiving end in VoIP. The jitter are affected by computer usage, the length and quality of the Ethernet cables and some other issues. The delay is inevitable and high levels of jitter causes large numbers of packets to be discarded by the jitter

buffer in the receiving IP phone or gateway. This will result in severe distortion in call quality.

In the next section designing of networks in OPNET MODELER 14.5 is discussed.

III.NETWORK MODEL DESIGN AND DESCRIPTION

Our simulation approach uses OPNET simulator for network modeling. OPNET simulator is an authoritative communication system simulator developed by OPNET Technologies. Firstly, network model of PSTN System is designed considering two scenarios one for office network and another for campus network. The simulation model for the PSTN office network under study is illustrated in Figure 1. The network model is a scenario of an office network where PSTN switches are connected as on client server model via PSTN E1 trunks. Five switches are working as client to the central switch as server. There are some important features in OPNET Software that are used to simplify and facilitate the work with it. In this research, some of these objects that are used are Application Configuration, Profile Configuration and QOS Attribute Configuration as show in Fig.1:

3.1. Application Configuration Object: It is an object used to define and configure all applications in the network according to the user requirements. OPNET has most common applications like: HTTP, E-mail, video, File transfer, Voice, database.

3.2. Profile Configuration Object: It is an object that can be used to create users profiles, profile can contain one or more applications and each application can be configured by the starting time and ending time. In this research, there is only one profile to all users called VoIP_ profile.

3.3. The QOS Attribute node: It is a mean of attribute configuration details that assess protocols at the IP layer.
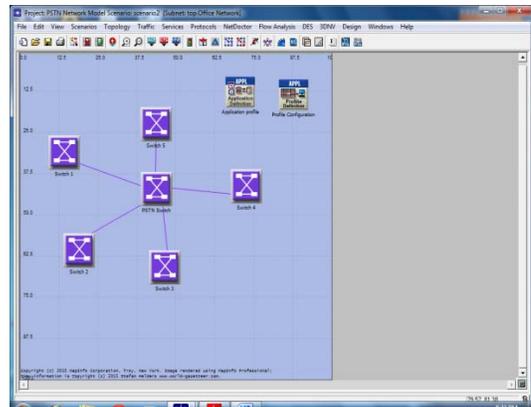

Figure 1. The Simulation Model of PSTN office Network.

The Application Configuration of PSTN office Network is given in Figure 2. In the Configuration Applications and voice encoder schemes are defined. Similarly in case of Profile Configuration Profile name and its applications are defined which is given in Figure.3.
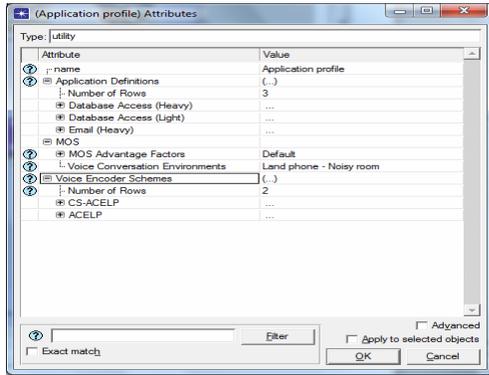
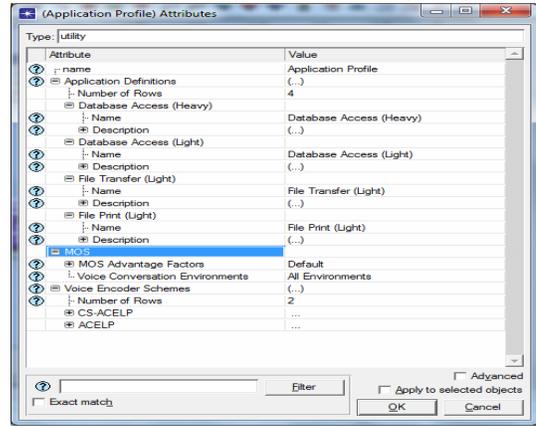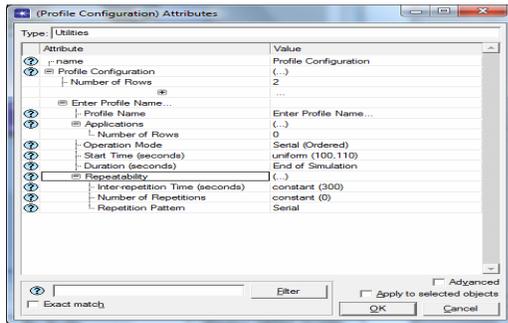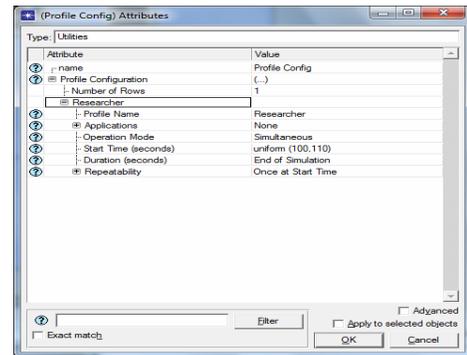Figure 2. Application Configuration of PSTN office Network

Figure 3. Profile Configuration of PSTN office Network

Next network model is a scenario of a large campus network which was given in Figure.4.
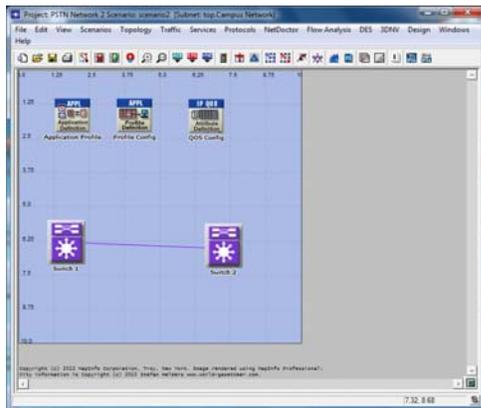

Figure 4. The Simulation Model of a PSTN Campus Network.

In this model a PSTN based network is created using two PSTN switches which has TDM Switching. The Switches are connected via OC_12 PSTN Trunks and traffic are generated in them using Application Profile, Profile configuration and QOS configuration. The switches are at two different locations of a campus. The application configuration is used to define the tier information utilized in the network model. In this application configuration various tier attributes are defined which are given in Figure .5.

Figure.5. Application Configuration of PSTN Campus Network

Similarly, Profile Configuration describes the activity applications used by users for PSTN network of campus through a period of time. Profile configuration as defined for this network is given in Figure.6.

Figure.6. Profile Configuration of PSTN Campus Network

Next QOS attributes are defined in QOS configuration for analyzing the QOS parameters which is given in Figure.7.
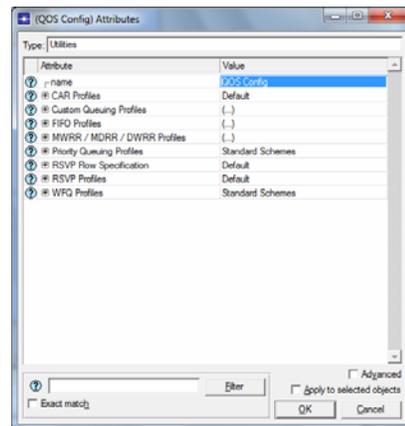

Figure.7 QOS Configuration of PSTN Campus Network

After designing PSTN model we have simulate the results which are discussed in next section. Now in case of VOIP systems we will take the case of a LAN network of a small office where initially numbers of clients are set to be 2 nodes and gradually they are increased to 150 nodes. Thus by increasing the nodes what is the variation of the

QOS parameters i.e., jitter, packet delay, etc is shown. Similarly another model considered is of LAN with FTP server. This is done to evaluate the performance with more realistic simulation situation on the quality of calls. The network model of VOIP LAN is given in Figure.8.
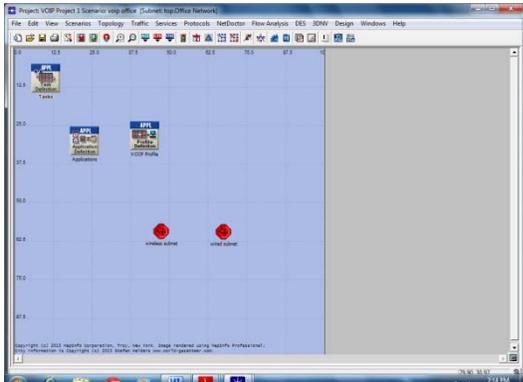

Figure.8 Network Model of VOIP LAN.

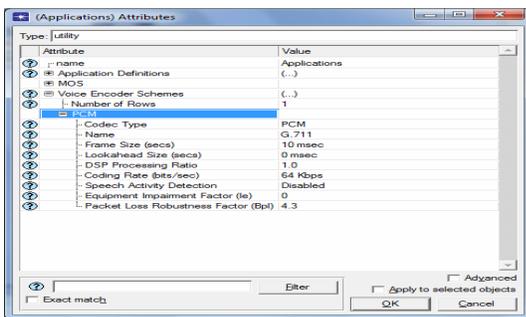The Application profile for LAN VOIP is given in Figure.9.

Figure.9 Application Attributes of VOIP LAN Network

The Profile configuration of LAN Network is given in Figure 10.
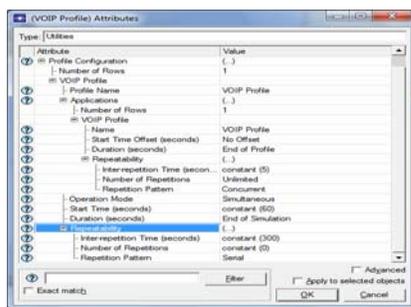

Figure.10 Profile Configuration of VOIP LAN Network

Now, a VOIP network with the size of 100 X 100 m in dimension and categorized as office in OPNET. Two stations are located 15m apart from each other and are connected through 10 Base T line to the Ethernet switch. The topology is setup in such way to achieve an LAN environment for the small office to establish VoIP communications. The model developed is shown in Figure.11
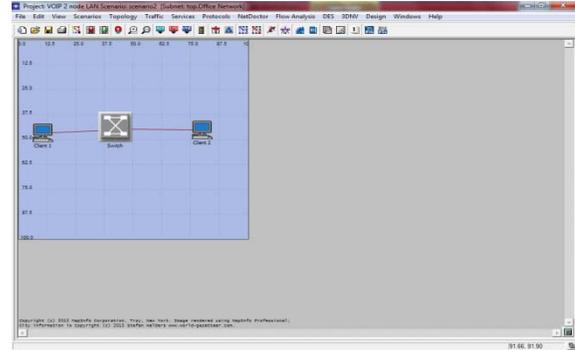

Figure.11 Network Model of VOIP LAN with two nodes

Next the number of nodes/ stations are increased to 20 then 50 and then to 150 nodes. As the number of nodes is increased the communication is done using CISCO router. Models with 20 and 150 nodes are shown in Figure 12 and Figure 13 respectively.
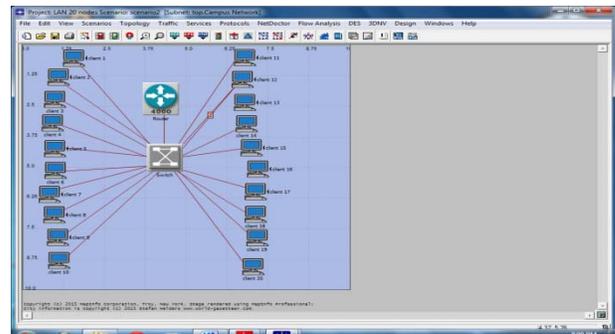

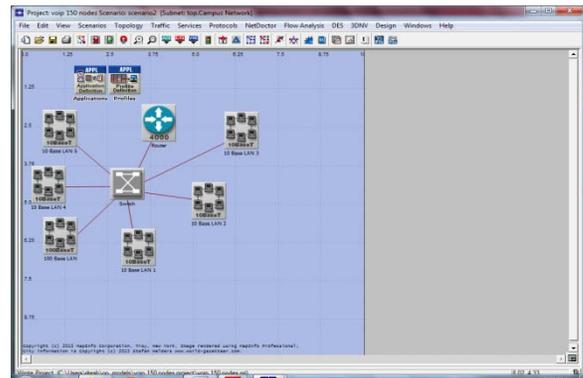Figure.12 Network Model of VOIP LAN with twenty nodes

Figure.13 Network Model of VOIP LAN with 150 nodes

Next case which is simulated is under the FTP scenario, a **File Transfer Protocol** (**FTP**) model is added to the Ethernet switch. By adding the FTP, we would expect the traffic load to increase and reduce in bandwidth usage since the work stations are making a VoIP call and accessing the FTP simultaneously [17]. The model is given in Figure.14.
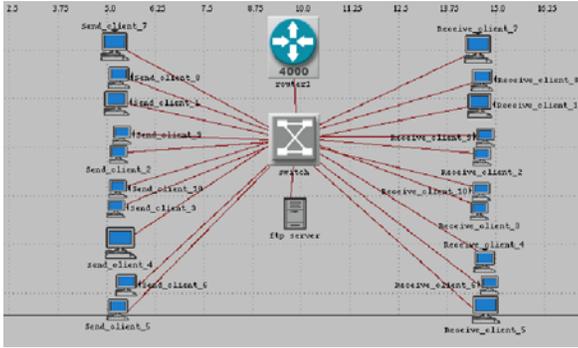
Figure.14 FTP in VOIP

The simulation results of PSTN and VOIP is discussed in next section.

IV. SIMULATION RESULTS AND ANALYSIS

The PSTN and VOIP systems are modeled and simulation is done on it. In case of PSTN model of office network of client server model when the traffic is increased quality of voice calls is good. The Simulated time and memory usage taken in case of office network is given in Figure 15.
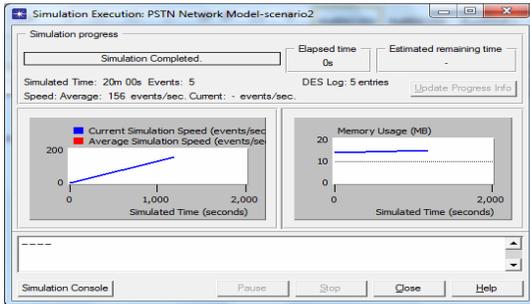

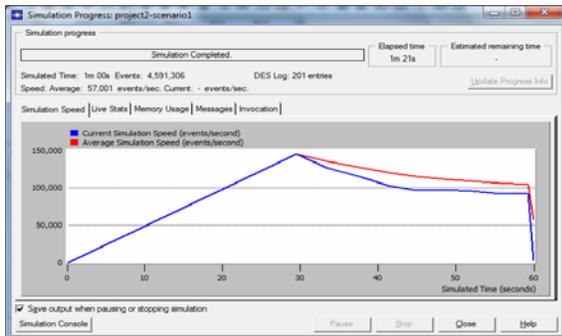Figure.15   Simulation Time and Memory usage of PSTN network.

Figure.16 Simulation Time and Memory usage of PSTN campus network

Similarly for PSTN campus network the Simulation time taken and memory usage done is given in Figure 16.From the simulation done it was found that in case of voice signal quality, one of the primary differences between the PSTN and the VoIP network is that the PSTN provides a dedicated voice channel of consistent bandwidth for each voice call, whereas a VoIP network provides best-effort voice packet delivery consistent with IP network behavior. It means that PSTN voice channels are designed with the voice signal in mind (i.e., they have just the right amount of bandwidth and the right frequency response to minimally support a conversational quality voice signal)

while IP networks, were never really designed for real-time, dedicated bandwidth applications like voice. This difference affects virtually all aspects of noise and distortion avoidance for VoIP implementations and VoIP/PSTN integration. Another difference is that PSTNs network provide call setup and management intelligence in the core of the network (via SS7 signaling and central office processing) whereas VoIP networks have pushed this intelligence to the edge of the network where it resides in VoIP endpoints such as personal computers or IP/Ethernet telephones. This affects the voice quality as the network core is no longer tightly controlled or regulated. It also means that in case of PSTN systems call setup time varies with distance. For long distance calls the setup time is more it depends upon routing how the call is opened, up to how much depth the calls are opened. If lower depth is opened then the setup time is more. In case of PSTN systems measurement of jitter, delay is complex and call quality basically depends upon routing of call. Delay in case of PSTN calls depends upon how the call is routed and up to which depth the call is opened and whether all levels of the calls are properly opened and routed. In case of VOIP systems we have taken a small office network with two nodes and gradually the nodes are increased to 150 nodes. In this case the effect on jitter in VoIP system with two nodes, 20 nodes and 150 nodes are shown below in Figure 17, Figure 18 and Figure 19 respectively.
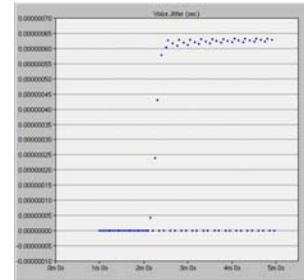

Figure 17. With two Nodes

Figure 18. With 20 Nodes

Figure 19. With 150 Nodes

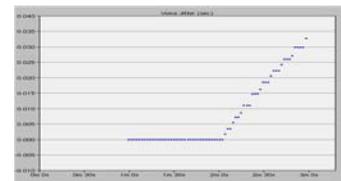From above figures it is found that the voice jitter increases as the number of nodes is increased. When the number of nodes is in between 2 to 20 nodes, the jitter is very small and unnoticeable. When the number of nodes in the LAN increased to 150, the voice jitter is 1000 times larger than a LAN with 20 nodes. Also the voice jitter in a LAN with 150 nodes seems to increase at a constant rate at

the end of the simulation. Next is the packet delay variation in VOIP systems as discussed below.

(i) With 2 nodes
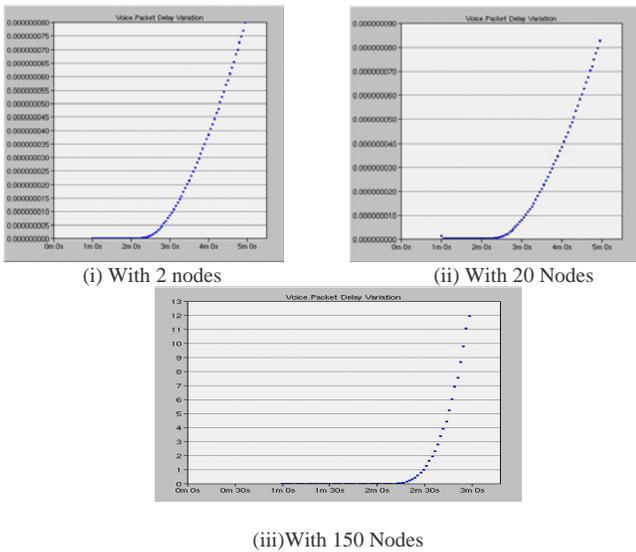

(ii) With 20 Nodes

(iii)With 150 Nodes

Figure 20 Packet Delay Variations in VOIP Systems

As seen from Figure.20 the packet delay variations are similar to voice jitter results. When the number of nodes is small, we have a very small and unnoticeable delay variation. The magnitude increases as the number of nodes increases, as the nodes are increased to 150, the delay variation exceeded by 1 second and reaches to maximum of 12 seconds. The delay variation seems to increase exponentially with the number of calls. The third QOS parameter analyzed is packet end to end delays as given in Figure.21.

(i) With 2 Nodes

(ii) With 20 Nodes

(iii) With 150 Nodes

Figure.21 Packet End to End Delays in VOIP Systems

In this case when the number of nodes is small, between 2 nodes to 20 nodes, we have the identical ETE delay around 140ms. ETE delay increases as the number of client nodes and calls are added into the network. As the nodes increased, ETE delay increases to around 160ms. In the simulation with 150 nodes, approximately 60 seconds

after VoIP calls started, ETE delay rapidly increased by over 200ms and reach to 12 seconds.
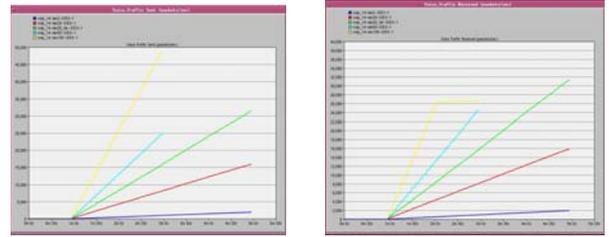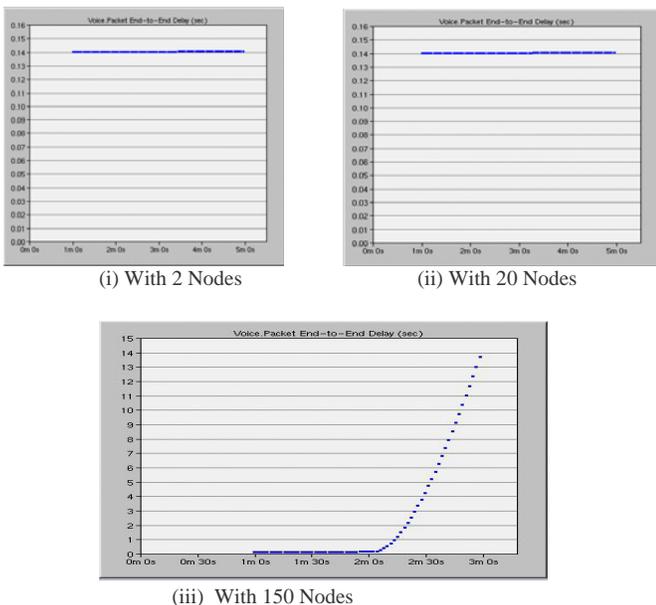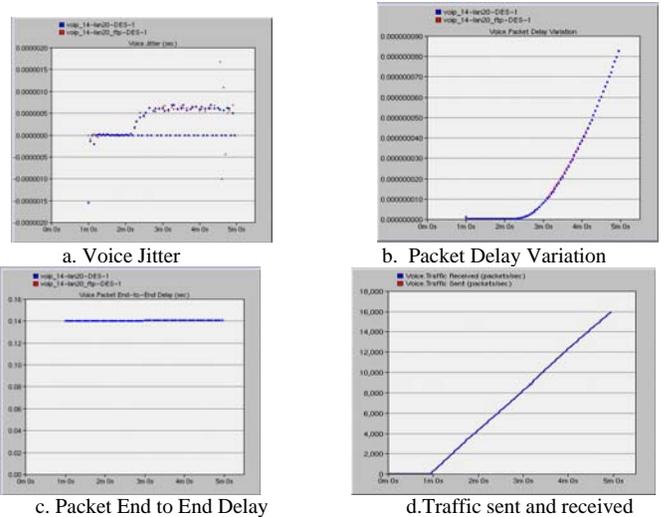

Figure.22 Traffic Sent and Traffic Received in VOIP Systems

Figure.22 shows the traffic sent and received with the voice application in VOIP systems. For the simulation with less than 50 nodes, the traffic received is consistent with the traffic sent. For the 150 nodes simulation, packets are lost when the numbers of calls get too high. As the number of VoIP clients and calls increases, the voice jitter, delay variation, and ETE delay become significant factors to the calls quality simulation, packets are lost when the numbers of call gets too high. Now in next case by adding an FTP server and FTP traffic into the network, we would like to produce a more realistic simulation result. In this case, adding FTP traffic in the network does not affect the voice jitter, delay variation or end to end delay. Traffic sent is identical to traffic received. Since this FTP simulation is done on a LAN with only 20 nodes. There is not much VoIP connection and calls. We assume that there is enough bandwidth to meet the demands of both the VoIP connection and FTP traffic. Therefore, the FTP packets are not dropped and the results are shown in Figure.23.

a. Voice Jitter

b. Packet Delay Variation

c. Packet End to End Delay

d.Traffic sent and received

Figure 23.VoIP calls in LAN with FTP server

The Quality of Service parameters Jitter, delay variations and ETE delay increased as a result of the increased distance. However the value is still very small when compared to the simulation with large number of nodes. The ETE delay is relatively stable around 150ms throughout the simulation. Calls quality is within the acceptable range.

V. CONCLUSIONS

In this paper a brief overview of QOS parameters is given. Both the systems are evaluated by simulating its models using OPNET

MODELER. Two scenarios for both PSTN and VOIP systems are considered and the performance of its QOS parameters is evaluated. As the analysis is studied we have seen that as the no. of nodes is increased jitter, packet delay increases and hence in case of VOIP systems, as long as the end-to-end delay does not exceed about 120-180 milliseconds, end users will report no conversational impairment but if it exceeds this value then packet loss and end to end delay increases which degrade the voice quality in VOIP systems. So jitter buffers are included in VoIP systems. On the basis of the results it was found that VOIP systems have a more stable and less delay connection than PSTN systems connection. Similarly another scenario is taken where FTP traffic is impaired but no much difference is shown in its results of QOS parameters. Thus imparting more traffic does not affect much in case of VOIP systems. As far as voice quality is concerned, it is best for PSTN systems but its call setup time for long distance is more. On the other hand the cost of VOIP systems is much lower than PSTN systems particularly for long distance communication. Further, VOIP systems will provide a stable connection and hence once deployed will produce better results for transmitting video, data, etc. with improved efficiency and lesser cost. Thus, VOIP based Systems is a future alternative to replace the traditional PSTN based systems due to its good network quality, better efficiency and lower infrastructural and operating cost.

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