Modelling And Simulation Of Voice Over Internet Protocol (Voip) Over Wireless Local Area Network (WLAN)

Ifijeh Ayodele Hephzibah, Fagbohun O. Oludare

ABSTRACT: The adoption of Voice over Wireless Local Area Network is on tremendous increase due to its ease, non-intrusive and inexpensive deployment, low maintenance cost, universal coverage and basic roaming capabilities. However, deploying Voice over Internet Protocol (VoIP) over Wireless Local Area Network (WLAN) is a challenging task for many network managers, architects, planners, designers and engineers, hence the need for a guideline to design, model and simulate the network before deployment. This work analyzed parameters such as latency, jitter, packet loss, codec, bandwidth, throughput, voice data length and de-jitter buffer size, which quantify the quality of degradation over the network. The analytical mathematical E-model was used to predict the readiness of the existing network to support VoIP. The Transmission Rating Factor R was calculated as 85.08 indicating a high speech quality and excellent user satisfaction. Riverbed Modeller Academic Edition was used to model and simulate the network. Results from this project work show that VoIP can be successfully deployed on WLAN with perceived high speech quality, user's satisfaction, low delay and high throughput.

Keywords: Voice over Internet Protocol (VoIP), Wireless Local Area Network (WAN).

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1. INTRODUCTION

Prior to the deployment of any telecommunication network performance analysis and predictions are necessary to be carried out to efficiently design and manage the networks, thereby analytically make plans for future network expansions. To effectively exploit network resources, network engineers, architects, developers and administrators must appropriately design the network by modeling, evaluating and measuring the performance of components and devices of the proposed network by using a certified simulation tool [1, 2]. Modelling is the representation of a real world object or system in a mathematical framework for the purpose of studying it [3]. VoIP often referred to as IP Telephony converges multiple forms of communication such as voice, video and data into a single network. These multiple forms of IP packetized communication are transmitted over privately managed IP-based network [4]. VoIP technology emerged with several advantageous features over the old traditional Public Switched Telephone Network (PSTN). However, the deployment of the real time packet switched network such as VoIP has proved to be very challenging for organizations to successfully merge and unify their voice and data networks hence, the need to adequately model and simulate the network to represent the traffic behavior as real as possible. The VoIP technology despite its numerous advantages over the PSTN technology is characterized by quality of service issues such as delay and jitters. Modeling and simulation of the VoIP network enables the evaluation Quality of Service parameters and characteristics performance issue.

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VoIP can be deployed over several data network such as the Internet, Local Area Networks (LAN) or Wireless networks. This paper models and simulates VoIP over the Wireless LAN. Wireless LANs originally called LAWNs (Local Area Wireless Network) are based on the IEEE 802.11 standards. IEEE 802.II –based Wireless LAN with a bandwidth of up to 54 Mb/s in the 5GHz band, using OFDM Modulation scheme, typically provides short-range coverage such as in offices and homes. This technologies are used as a packet-switched IP network, hence making them an accepted option for supporting VoIP. Riverbed Modeler Academic Edition is a suite of protocols and technologies with sophisticated development environment used to model several networks and technologies which includes VoIP, TCP, OSPFv3, MPLS, IPv6 etc. The Modeler compares the impact of different technology designs on end-to-end behavior by analyzing the network. Designs are tested and demonstrated before production, thereby increasing research and development productivity of network. Wireless protocols and technologies are also developed, while evaluating the enhancements to standards-based protocols [49]. This work presents a simulated model on the deployment of VoIP into the existing WLAN. This simulated model can further be implemented in any enterprise over a Wireless Network.

A. VoIP Components

To successfully deploy VoIP, an Internet Protocol Private Branch Exchange (IP PBX) running VoIP protocol must first be installed into the organization network server. Session Initiation Protocol (SIP) phones a VoIP protocol phone and other codecs will be installed in all workstations, laptops, desktops, PDAs existing on the network. The major components of the wireless connectivity are the wireless access points (AP). Access points are devices that serve multiple stations directly and act as bridge between the IEEE 802.11- based wireless network and the wired network. They comprise a radio transceiver, communication and encryption software, and a IEEE 802.11-based wireless Network Interface Card (NIC). A single access point in an open-space free of obstruction can cover a circle centered diameter of about 100m. The larger the distance between the computer and the access point, the poorer the signal and the slower the connection, hence large offices often deploy several access points with overlapping ranges. VoIP protocol defines the mechanisms that users in a communication session must follow, depending on the service the end user requires. The protocols are meant to establish end-to-end voice call [20, 13]. From VoIP perspectives, there are two types of protocols; Signaling Protocols and Media Transport Protocols. Signaling protocols provides session setup, control and teardown allowing call information to be carried across network boundaries. Auxiliary function related to setting up and maintaining calls are performed with these protocols [20]. The two most popular signaling protocols are SIP and H.323. Media Transport Protocol is the second type of protocol used in VoIP in the transportation of voice. The protocol is responsible for actually carrying the digitized voice in the form of packets. It ensures the transfer of voice packets through the network. The most popular transport protocols are Real-time Transport Protocol (RTP) and Real-time Control Protocol (RTCP). However, for VoIP over WLAN, the IEEE802.11 Medium Access Control (MAC) protocol defines the method for communication over wireless stations [4, 17]. The MAC protocol ensures that multiple stations can transmit data on the same wireless channel without causing any interference. There are two modes for accessing the shared wireless channel for transmitting data. Distributed Coordination Function (DCF), based on Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) the first mode is a distributed protocol requiring no centralized entity for coordinating the channel access. The second mode Point Coordination Function (PCF) supports time-sensitive traffic flows. PCF is a centralized protocol and requires the existence of a centralized coordinator entity called a point coordinator residing inside an access point and decides which station can transmit in a given time period. Data transmitted over the network without being compressed uses a lot bandwidth [7] hence, codecs is required to compress speech prior to being transmitted. The tuning of a codec for a particular type of network is very important [8]. Voice codec is one of the most critical component of a VoIP system. It converts the input speech signal into digital form, transmit the signal to the receiver and reconstruct the original speech signal. This paper evaluates various VoIP codecs such as G.711, G.723.1, G.729A, G.728, G.726, AMR and GSM codecs. Quality of service (QoS) is a very significant aspect for VoIP applications. While the cost of deploying VoIP is considerably less, there are significant performance issues that affect the voice quality of the network. These parameters that quantify the quality degradation over a certain connection are latency, jitter, packet loss, codec, bandwidth, throughput, voice data length and de-jitter buffer size. This paper applied practical simulation tool to design, model and analyze the new VoIP network. Riverbed Modeler simulation package is used in this paper to investigate throughput, delay bounds and to verify and analyze results. Riverbed Modeler provides a virtual environment for modeling, analyzing, and predicting the performance of the VoIP network to be deployed [5].

B. Related Works

Several researches have been done on VoIP deployment since its advent. Though VoIP over WLAN is one of the most recent applications in the high-speed packet-switched networks, nevertheless several essential researches are being carried out to ensure a good Quality of Service. Detailed simulation approach for deploying an end – to end VoIP component from sender to receiver was implemented in [6] and [7] using the OPNET network simulator. The implementation of a practical simulation tool to design and analyze communication networks were described in [1], [2] and [8]. The papers focused on the implementation methods and the applications of the designed simulator supporting VoIP. The innovation of the new model in [2] consists in modeling the user behavior instead of the aggregated traffic. Modeling and aggregation of VoIP streams were investigated in [8]. Development and validation of models for the multiplexed process was presented based on statistical analyses of VoIP traffic in [9], [10], [11] and [12]. The models can be used for simulation of any IP network architecture, wire line or wireless. The analysis in [11] was performed by importing the data derived from the post processing into Matlab software. A theoretical approach to model an aggregated flow of VoIP connections on packet basis created due to tele-traffic parameters were presented in [13], [14] and [15] to evaluate the performance of VoIP services with different Codecs. Simulations were conducted using the one hybrid UMTS in [13] and the Network Simulation 2 in [14] and [15]. In [16], [17], [18] and [19] various mathematical models for measuring VoIP qualities such as Mean Opinion Score (MOS), E-model, and Perpetual Evaluation of Speech Quality (PESQ) score were analyzed. Experimental speech quality measurements under wired and wireless scenarios were compared with the mathematical speech predictor. Findings indicate that WLAN QoS parameters have a high variability in real-world environments, with a significant effect on application performance.

2. Materials and Methods

This project work modelled VoIP over a managed carrier IP networks over a Wireless Local Area Network of Ekiti State University (EKSU), Ado-Ekiti, Nigeria. The methodology to be deployed in this project is;

1. Conduct a pre-deployment assessment, audit of existing network and a general survey of the existing infrastructure available in the university campus. This assessment will also predict if the existing network can support VoIP and identify the potential benefits that can be derived from the intending VoIP system, in comparison with the existing PSTN or LAN network.
2. Determine VoIP characteristic requirement such as; VoIP codec, traffic flow, call distribution and assumptions.
3. Define performance thresholds and future growth capacity.
4. Design the network to determine the optimal VoIP system configuration requirements.
5. Simulate the designed network using OPNET to measure and analyze delay, packet loss and jitters.
6. Deploy steps and test result to confirm if the expected VoIP quality meets the acceptable metrics for the proposed environment using a mathematical model

such as R-Factor, Mean Opinion Score (MOS) and the E - Model.

The methodology for this work is summarized in the flow chart in figure 2.1;

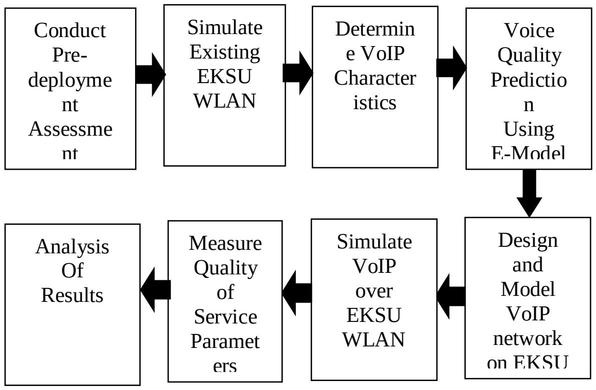

Figure 2.1: *Methodology Block Diagram*

2.1 Ekiti State University (EKSU) WLAN MODEL

Ekiti State University WLAN is quite a large network. To enable simulation using the Riverbed Modeler, the network is divided into 13 concourse subnets in figure 2.2. Each of twelve of the concourse subnet is connected to a central bridge and then connected via 100baseT to six access points (APs). Scenario 1: In figure 2.2, the main subnet which is the Network Operating Centre is connected through Fibre Demonstration Data Interface (FDDI) to the twelve other subnets which represents each faculty, the Administrative building and lecture theatre.

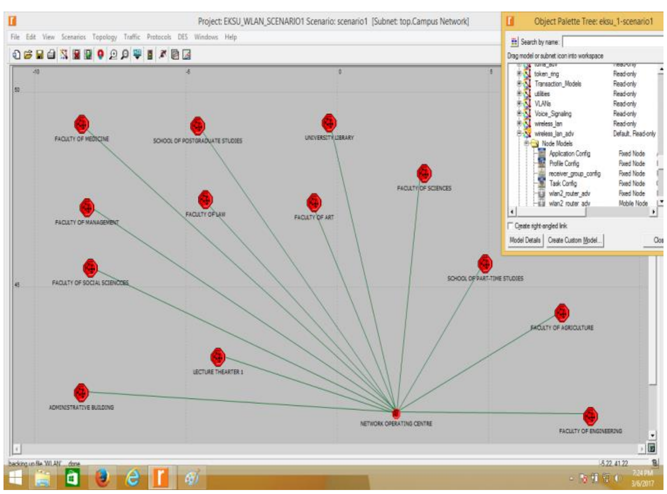

Figure 2.2: *EKSU WLAN Model Scenario 1*

In the Network Operating Centre, the network is connected to the WLAN Ethernet 16 switch and then to the server in figure 2.3. Server attributed were configured for WLAN PHY and Data and IP protocols. Application and Profile Support Services were also defined to enable wireless transmission.

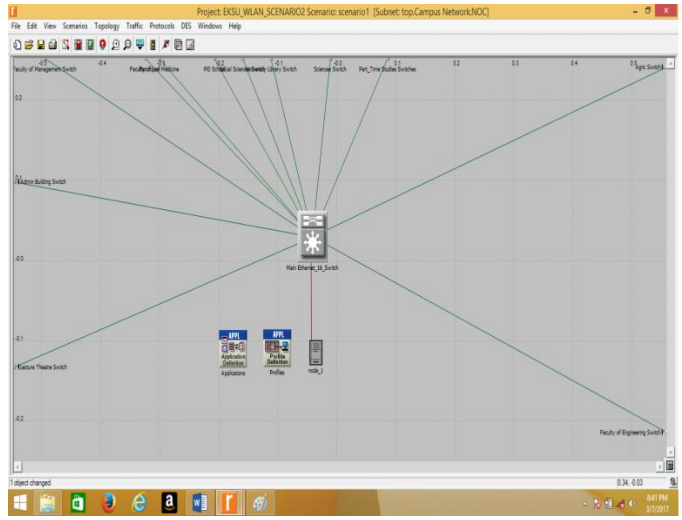

Figure 2.3: Main Subnet, Data Centre

Scenario 2: In this scenario, WLAN Ethernet Router is connected in a star connection through 100baseT to six access points in the various faculties. This scenario in figure 2.4 is deployed in the twelve subnets. IP protocols, WLAN parameters, Application and Profile Support Services were also configured to enable wireless transmission. Each subnet in figure 2.2 when expanded or opened will result in the scenario in figure 2.4. Individual statistics are chosen in scenario to measure the delay, throughput, dropped packets, sent and received packets of the WLAN. The network is simulated for 30 minutes and the results examined.

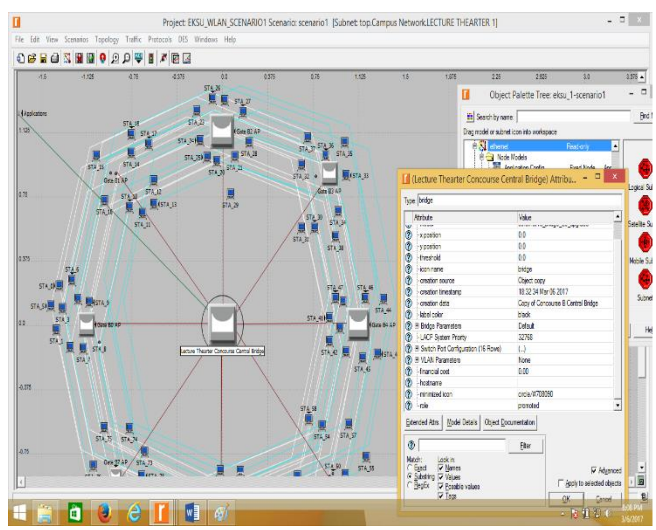

Figure 2.4: Scenario 2 – Faculty WLAN Connectivity

2.2 NEW VoIP MODEL FOR EKSU WLAN

The new VoIP network is deployed on the existing WLAN of the Ekiti State University. The calculation in session above indicates the number of simultaneous VoIP calls a single AP running DCF can support. The maximum number of VoIP sessions (*N*) a wireless access point can support using G711 codec is 10. This connection will give a good and fine quality of service, which can be measured by the loss, delay, jitters experienced in the network. As soon as the eleventh client is added the quality of service drops thereby increasing the loss, delay and jitters. To deploy VoIP on the existing modeled EKSU WLAN, some features such as gateways, IP Phones and IP PBX will be connected to the existing network. Voice parameters will also be configured to enable voice calls. The IP phones and telephones are linked to both the IP network and PSTN to generate the voice packets only. Protocols, Codecs and traffic patterns are specified. Traffic and voice calls are generated by the hosts and phone nodes.

2.2.1 VoIP Model Scenario 1:

In the Network Operating Centre in figure 2.2, a gateway is connected to the network to enable voice communication from the PSTN network. Voice application and profile support service are configured. In figure 2.5, a SIP gateway is connected to the WLAN Ethernet Server, to enable voice communication from PSTN telephones from the various faculties where PSTN is available and for future back-up services. The SIP Server is also connected to the SIP gateway to provide the soft Internet Protocol Private Branch Exchange (IP-PBX). Voice applications such as the Proxy Service is enabled, maximum simultaneous calls setting was set to unlimited to allow voice calls from other buildings. VoIP Codecs are also selected and configured to provide effective compression capabilities to save network bandwidth. The GSM AMR codec is used in this project work as against the popularly used G.711 codec. From previous calculation and simulation in this work, the GSM codec has the highest throughput when compared to the G.711 codec though the GSM AMR codec has a higher

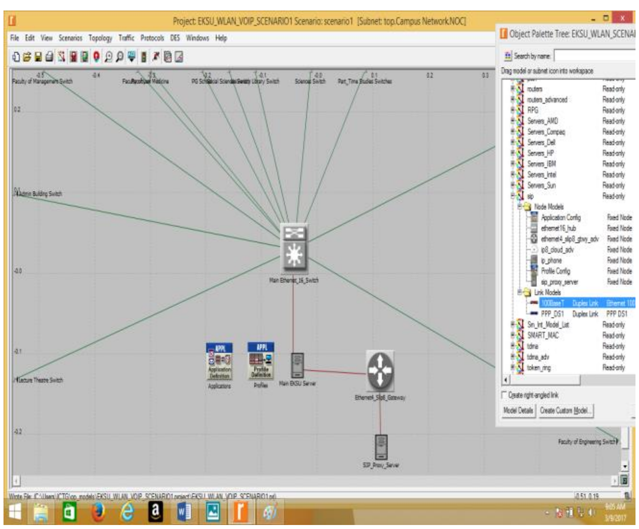

Figure 2.5: *EKSU VoIP WLAN Model Scenario*

delay than the G.711, the GSM AMR codec will produce better voice satisfaction.

2.2.2. VoIP Model Scenario 2:

In this scenario, the SIP gateway is connected to each faculty subnet of figure 2.4 to enable communication to and fro the PSTN. IP phones and PSTN telephones are also connected to enable voice connection on the network and for future expansion. SIP proxy protocols are enabled and configured on the gateway. In figure 2.6, the IP-PBX is connected or configured into the network. Ten IP phones are connected to PSTN switch to enable communication to and fro the network.

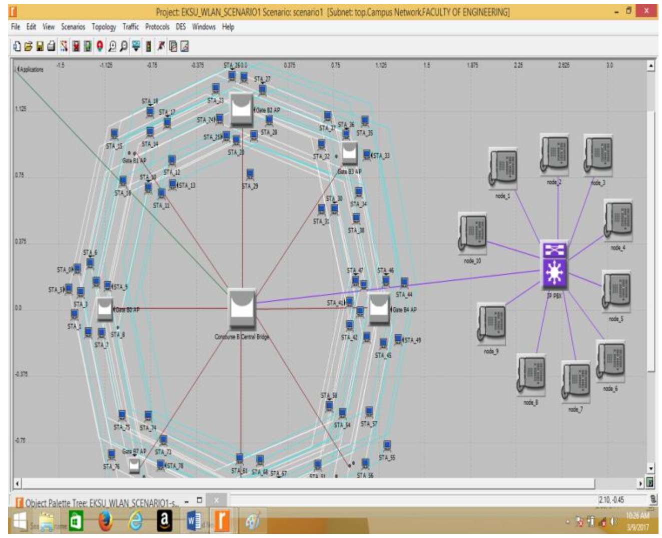

Figure 2.6: *VOIP Model Scenario 2*

Soft switch parameters such as Codec mapping, IP Header Size (byte), Voice Frame per packet (frame/packet), De-jitter Buffer are configured. The scenario in figure 2.6 are configured in the each faculty subnet, however, two IP-PBX and twenty IP phones are deployed and configured in the Administrative Subnet. The EKSU networks of scenario 1 is now adjusted to the new VoIP model and hence simulated, by running Discrete Event Simulation (DES) on the simulator. The simulation duration is set for 100seconds. Individual statistics are chosen on the simulator to measure end to end delay, throughput, traffic sent and received. The simulator will take about 1minute to execute.

3. RESULTS

In this session results from the simulated EKSU WLAN and VoIP over EKSU WLAN are viewed and analyzed. The results shows measurements for jitter, packet delay variation, packet end-to-end delay, traffic received and traffic sent.

3.1 Measured Quality of Service in Existing EKSU WLAN

The quality of service requirements for WLAN such as delay, throughput, media access delay, were measured. Delay represents the end to end delay of all the packets received by the wireless LAN MACs of all WLAN nodes in the network and forwarded to the higher layer. This delay includes medium access delay at the source MAC, reception of all the fragments individually, and transfer of the frames via AP if access point functionality is enabled. The delay and media access delay in figure 3.1 shows 0.03secs (30ms) and 0.005secs (5ms) respectively.

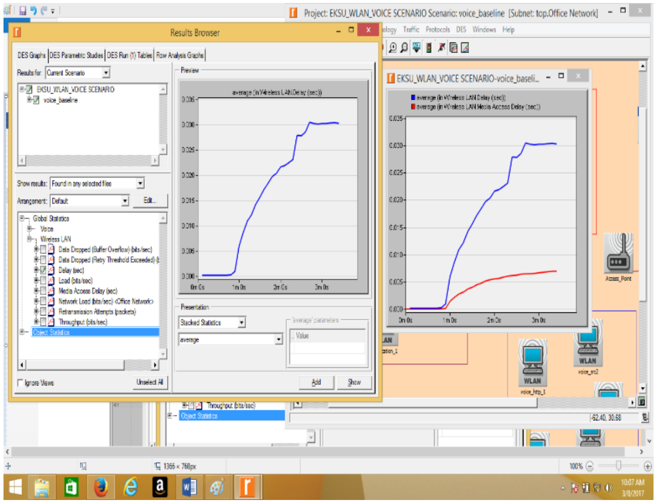

Fig 3.1a: *EKSU WLAN Delay and Media Access Delay*

4.3 VOICE QUALITY PREDICTION

The voice quality of the VoIP network was predicted using the E-model. The E-model is used to evaluate delay, packet loss and speech quality for different network scenarios. The transmission rating factor R, is calculated in section 3.3.5 as;

$$R = R_O - I_S - I_D - I_{e-eff} + A$$

$$\boldsymbol{R = 94.74 - 1.41 + 15.5 - 23.75 + 0 \ = \ 85.08}$$

The calculated value 85.08 of the Transmission Rating Factor R, is of high speech quality and satisfactory user quality, hence the voice network if deployed on the WLAN will give a satisfactory speech quality.

4.4 RESULTS FOR NEW VoIP MODEL FOR EKSU WLAN

The newly modelled VoIP network over EKSU WLAN in figure 4.7 was simulated in the previous chapter. This section views and analyse the results for average voice jitter (sec), average voice packets end-to-end delay (sec), average voice packet delay variation, voice traffic received (bytes/sec), average traffic received and average traffic sent in packets/sec and the average voice Mean Opinion Score (MOS) to enable the assessment of the quality of voice in the network.The results shows that the newly modeled VoIP over EKSU WLAN will give a satisfactory user or voice network when the deployed. The results from the simulation also confirms the voice quality prediction derived from the calculation of the transmission rating factor of the E-model. The prediction gives a user satisfactory voice quality which is also seen from the result of the simulations. The end to end delay in the result is 0.13sec at 3minutes after transmission, the result also shows 92% of the voice sent were received and only 8% were lost as a result of variations. The highest value captured however is 4.3 which represents a good voice quality as calculated using the E-model.

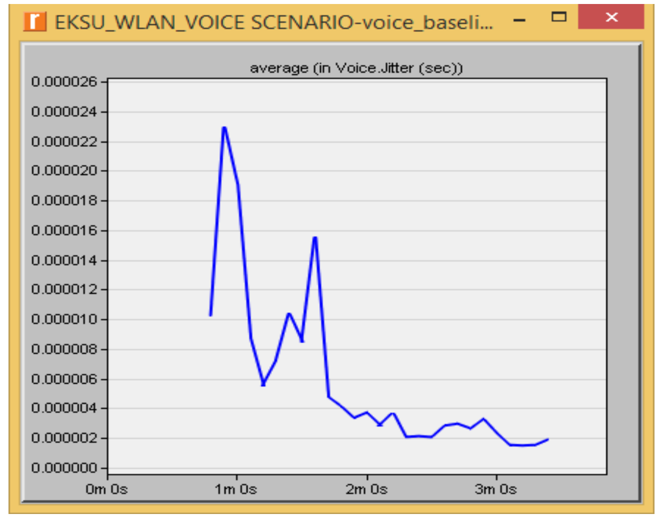

Fig4.9a: *Voice Jitter*

Average Voice Jitter (sec): in figure 4.9a, if two consecutive packets leave the source node with time stamps t1 & t2 and are played back at the destination node at time t3 & t4, then Jitter is (t4 –t3) – (t2 –t1). Negative jitter indicates that the time difference between the packets at the destination node was less than that at the source node.

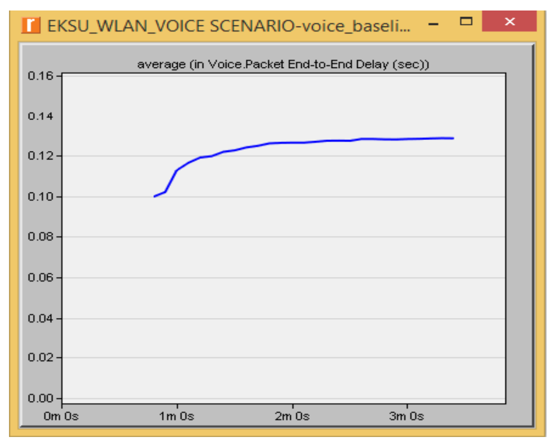

Fig4.9b: *Voice End-to-End Delay*

Average Voice Packets End-to-End Delay (sec) in figure 4.9b is the time at which the sender node gave the packet to RTP to the time the receiver got it from RTP. Encoding delay on the sender node is computed from the encoder scheme. Decoding delay on the receiver node is assumed to be equal to the encoding delay. Compression and decompression delays come from the corresponding attributes in the voice application configuration. The end to end delay in figure 4.9b is 0.13sec at 3minutes after transmission.

Average Voice Packet Delay Variation is the variation among end to end delays for voice packets. End to end delay for a voice packet is measured from the time it is created to the time it is received. In figure 4.9c, the delay variation as at 2minutes of transmission is 0.00055, and 0.00062 at 3minutes, giving a delay variation of about 11%.

Voice Traffic Received (bytes/sec) is the average number of bytes per second forwarded to all voice applications by the transport layers in the network. **Voice Traffic Sent (bytes/sec)** is the average number of bytes per second submitted to the transport layer by all voice applications in the network. In figure 4.9d, of the 130000 bytes/sec sent in 1minute of transmission to the transport layer, 120000 bytes/sec were received, that is 92% of the voice sent were received and only 8% were lost as a result of variations.

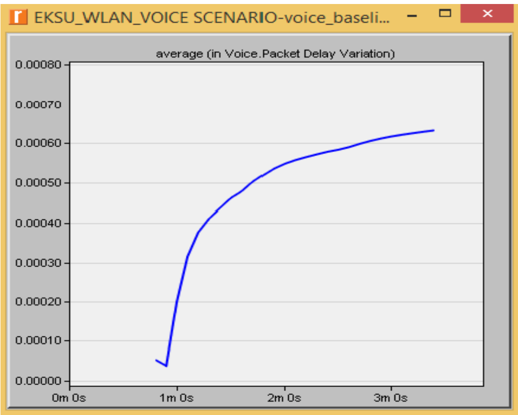

Fig4.9c: *Voice Packet Delay*

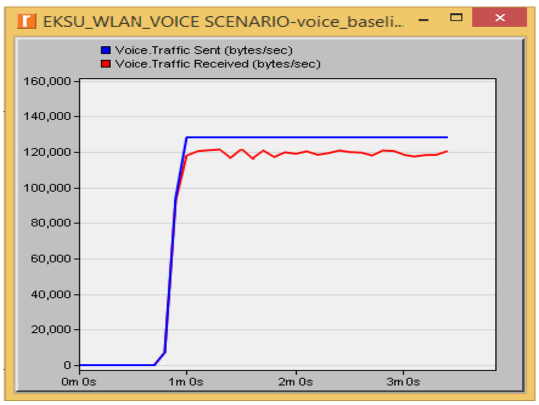

Fig4.9d: *Traffic Sent/Received (bytes/sec)*

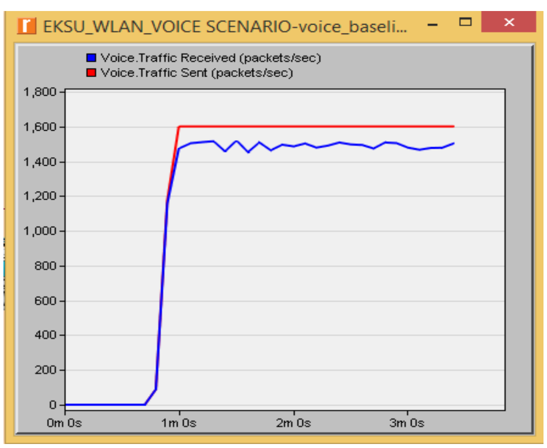

Fig4.9e: *Voice Sent/Received (packets/sec)*

Figure 4.9e shows the ***average traffic received and average traffic sent in packets/sec.*** The average traffic received or sent is the average number of packets per second forwarded or submitted to all voice applications by the transport layer in the network. The received packets as seen in figure 4.9e is 1500 packets/sec, while 1600 packets/sec were initially sent.

The Average Voice Mean Opinion Score (MOS) is a methodology used to access the quality of voice in the network. By default this global statistic captures the minimum MOS value collected in the network, a behavior which can be modified by changing the statistics. The highest value captured however is 4.3 which represents a good voice quality as calculated using the E-model.

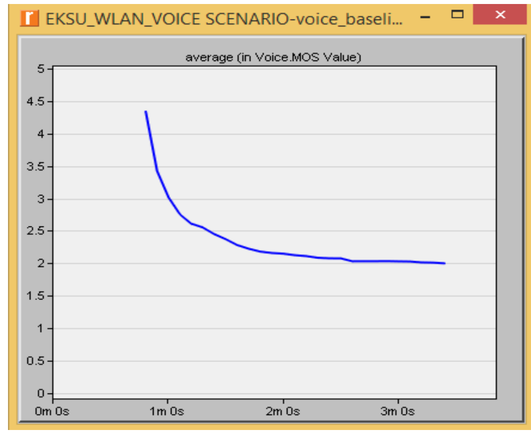

Fig4.9f: *Voice MOS Value*

4.4 The Effect of Deploying VoIP on Existing EKSU WLAN

The existing EKSU WLAN has all the prerequisites required to deploy VoIP on its network. This session compares and measure the difference between the QoS of WLAN and QoS of VoIP on WLAN. WLAN data dropped (bits/sec) is compared with Voice Traffic sent and received (bytes/sec). The difference between the traffic sent and receive would give the voice dropp in the network. The traffic sent is 120,000bits/sec (120000 * 8 = 960000 bytes/sec) while traffic received is 130000 bits/sec (130000*8=1040000 bytes/sec) in figure 4.10. The difference gives a 7% voice drop, while the data dropp in the WLAN is as low as 2000 bits/sec.

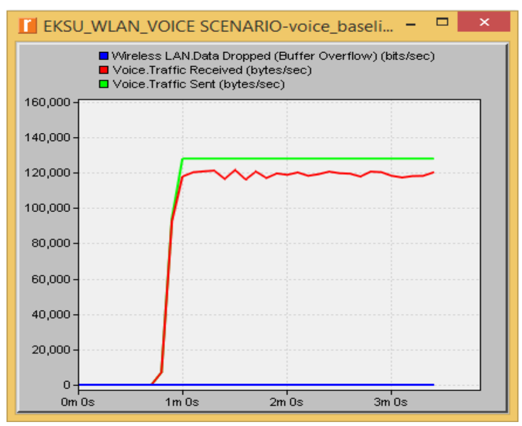

Figure 4.10a: *VOIP-WLAN Packet Drop*

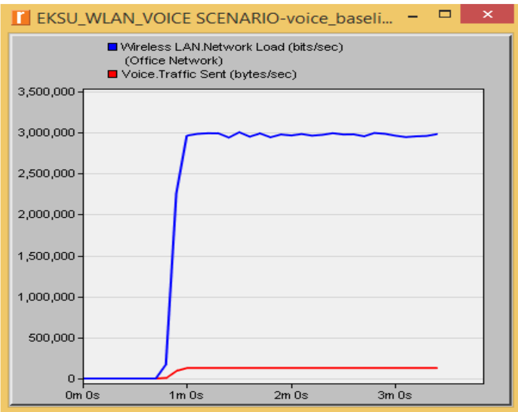

Figure 4.10b: *VOIP-WLAN Load*

Figure 4.10b compares the WLAN network load to the Voice Traffic sent per minutes. The WLAN load is much more than the voice traffic load. In 1minute of transmission, the WLAN load is 3Mb/s while the voice load is 960Kb/s. The WLAN is throughput is not affected by the voice application, as the voice throughput is only 66% that os the WLAN in figure 4.10c. The voice packet end to end delay is however higher than the WLAN delay in figure 4.10d. The voice packet delay however is 120ms which is below the ITU recommended value. The voice delay can be further reduced to improve quality of service.

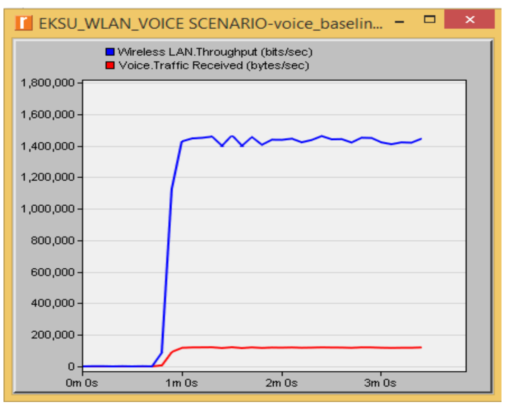

Figure 4.10c: *VOIP-WLAN Throughput*

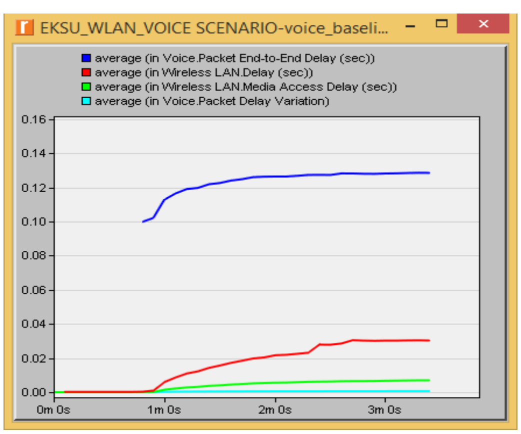

Figure 4.10d: *VOIP-WLAN Delay*

In conclusion, it is observed that voice application has a better throughput but still has some losses and longer delays.

5. Track for Future Work

The deployment of VoIP over WLAN is on the increase; however the quality of service remains a setback, hence more work can still be done on improving the quality of service of VoIP over the wireless network. More research work can be carried out on measures of reducing channel access delay on the network, internal and external interferences, jitter, and packet loss etc., thereby improving voice quality over the network. Lots of research works are ongoing on the modeling and simulation of VoIP over Ethernet, this work proposes to modeling and simulate the deployment of VoIP over WLAN using a suitable network simulation tool such as OPNET.

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