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PERFORMANCE ANALYSIS OF INFRASTRUCTURAL WIRELESS NETWORK WITH AND WITHOUT VOIP

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ABSTRACT

Evolution of the Internet and the wide growth of computer networks have made the Internet part of our everyday life. This accounts for the reason why the interest and demand on different internet-based applications has increased Steinmetz & Wehrle (2005). Voice over Internet Protocol (VoIP) technology has become a potential alternative to and supplement of the traditional telephony systems over the Public Switched Telephone Network (PSTN), providing a versatile, flexible and cost-effective solution to speech communications. Goode (2002). Koforidua Technical University spends a large chunk of its revenue on public switched network telephone. Since there is an already existing wireless network supporting file transfers, streaming and other functionalities, it would be cheaper to incorporate VoIP that could enable students and administrators communicate at a low cost. In this paper, we use OPNET modeler to simulate the effects of VoIP on a wireless network infrastructure. The simulation results obtained showed that implementing VoIP over and existing wireless network infrastructure does not have any negative effect on the system.

KEYWORDS: VoIP, Internet, PSTN, wireless, network

INTRODUCTION

An infrastructural wireless network is a network in which devices communicates with each other by first going through an Access Point (AP). In infrastructure mode, wireless devices can communicate with each other or can communicate with a wired network. When one AP is connected to a wired network and a set of wireless stations, it is referred to as a Basic Service Set (BSS). An Extended Service Set (ESS) is a set of two or more BSSs that form a single sub network. Most corporate wireless LANs operate in infrastructure mode because they require access to the wired LAN in order to use services and resources on the network.

To support transmission of voice, data, and video, several wired information network infrastructures have evolved throughout the past century. Wireless networks allow a mobile telecommunications terminal to access these wired information network infrastructures. At first glance, it may appear that a wireless network is only an antenna site connected to one of the switches in the wired information infrastructure, which enables a mobile terminal to be connected to the backbone network. In reality, in addition to the antenna site, a wireless network may also need to add its own mobility-aware switches and base station control devices to be able to support mobility when a mobile terminal changes its connection point to the network. Therefore, a wireless network has a fixed infrastructure with mobility aware switches and point-to-point connections, similar to other wired infrastructures, as well as antenna sites and mobile terminals. (https://www.lifewire.com)

A wireless telecommunications device, such as a cordless telephone, can connect to the PSTN infrastructure by replacing the wire attachment with radio transceivers. However, for the mobile terminal to change its point of contact (antennas) the PSTN switches must be able to support mobility. Switches in the PSTN infrastructure were not originally designed to support mobility. To solve this problem, the cellular telephone service providers add their own fix

Infrastructure with mobility-aware switches. The fixed infrastructure of the cellular telephone service provider is an interface between the base stations and the PSTN infrastructure that implements the requirements to support mobility

PROBLEM STATEMENT
Koforidua Technical University is spends huge proportion of its scarce resource as bills on public switched network telephone. There is an already existing wireless network supporting file transfers, streaming and other functionalities but do not have the VoIP feature that could enable students and administrators communicate with no additional cost. To minimize the amount spent on bills for telephony, the wireless network should have a VoIP protocol such as SIP / H.323 extension. Singh & Schulzrinne (2000). From Goode (2002), VoIP can provide substantial savings on your telephone service by allowing you to use an IP network to make phone calls instead of the traditional telephone companies' public switched telephone network (PSTN). However, the introduction of VoIP may introduce additional overhead on the existing wireless infrastructure. Garg & Kappes (2003). The study therefore aims to study the effect the introduction of VoIP will have on the existing wireless infrastructure.

OBJECTIVE OF RESEARCH
GLOBAL/MAIN OBJECTIVE
To measure the performance of a wireless infrastructural network with VoIP and without VoIP

SPECIFIC OBJECTIVE
1. To analyse packet losses in a network with and without VoIP.
2. To analyse the entire data traffic received by the wireless LAN for each Basic service set (BSS) in a network with and without VoIP. (Network Load)
3. To analyse the entire data traffic submitted to the wireless LAN. (Load)

Review of related Work
A lot of work has been done in the area of VoIP but most of these research has been skewed towards the area of quality of service (QoS). Das et al (2003), Looked at the performance optimization of VoIP calls over wireless links. In this paper the researchers concluded that VoIP call set-up performance can degrade significantly even for moderately high air-link frame error rates (FERs). Alexander et al (2009), Focused on voice quality of VoIP with the introduction of The Secure Real-Time Transport Protocol (SRTP). Tariq, here the authors looked at the additional overhead that SRTP was going to introduce on the voice quality of VoIP. Analyses of the performance of VoIP codecs over BE WiMAX networks was focused on by Azad et al (2013). In this work the researchers sought to find the effects of VoIP codecs on the performance of BE WiMAX networks and concluded that varying the jitter buffer size and packetization time affects the quality of voice over the best effort network. Che et al (2009) studied how VoIP performance can be affected by different routing behaviours which include Routing Information Protocol (RIP), Open Shortest Path First (OSPF) and Enhanced Interior Gateway Routing Protocol (EIGRP).
This paper attempts to fill the void by analysing the overhead that VoIP introduces on a whole wireless network infrastructure.

Different forms of VoIP
Computer-to-Computer Connections
Computer-to-Computer connections provide users with an easy way to make phone and video calls. These connections are typically free to use. Additionally, set-up fees are cheap or non-existent, as all users need is Internet (and of course use of audio transmitting hardware such as microphone and speakers). Aside from that, users will also need an account through a service provider such as Google Voice or Skype.

Computer-to-Any-Phone Connections
Most providers don't charge users for computer-to-computer connections because they're not responsible for setup, and calls would be made to other in-network subscribers; however, here a computer-to-phone connection uses a provider's software/service to connect to any out-of-network phone. In this process, the digital data is then converted to be sent through telephone lines. While there may be a cost, it is substantially less, especially for international calls. For example, you can use Skype or Google Voice to dial out to an out of network user, even landlines. This method has a little more to it than the previous, as users no longer simply connect directly from their computer; however, users may feel a little more comfortable using a phone.

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Mobile Softphones & Apps

Where the previous two methods worked from stationary devices, mobile softphones and apps allow for greater mobility without sacrificing quality or ease of use. With mobile softphones and apps, users are able to make VoIP calls, as well as utilize premium features, directly from their mobile device. While these apps vary in price, they come with full VoIP capability that is accessible and user friendly (and in some cases don't require a fixed provider). Additionally, these apps are typically available on various platforms including Android, iPhones, Windows, etc. With increased compatibility and interoperability, mobile softphones and apps allow users the same flexibility as computer-to-computer or computer-to-phone connections; however, here there is a simplified setup that integrates expanded mobility.

Any Landline Phone via ATA

Analog Telephone Adapters (ATAs) are devices that make calls over the Internet by connecting an analog telephone to a digital network or VoIP system through either a computer or a network device. In doing so, ATAs convert your voice into data to be sent over the Internet. In using an ATA, users' phone systems will also be fit with a host of VoIP features (i.e. caller ID, call waiting, call transfer, etc.). As with all methods of VoIP connection, users will need to register with a provider for service. Additionally, users need to either port their existing number or get a new number depending on the provider selected. (https://getvoip.com)

IP Phones

These specialized phones look just like normal phones with a handset, cradle and buttons. But instead of having the standard RJ-11 phone connectors, IP phones have an RJ-45 Ethernet connector. IP phones connect directly to the router and have all the hardware and software necessary right on board to handle the IP call. Soon, Wi-Fi IP phones will be available, allowing subscribing callers to make VoIP calls from any Wi-Fi hot spot.

DEVICES USED IN VOIP TECHNOLOGY

The architecture of Internet telephony is similar to traditional telephone networks in many ways, but it also has some significant differences. Most fundamentally, Internet telephony is different from traditional telephone networks in that it, naturally, runs over the Internet, or more generally over IP networks. The most significant consequence of having this underlying network is that it provides transparent connectivity between any two devices on the network. Whereas devices in traditional networks are restricted to communicating with those devices to which they are directly connected, and the telephony protocols themselves must handle all location and routing features, Internet telephony can rely on an underlying infrastructure, which provides all these capabilities automatically. The most common devices used in the Internet telephony are end systems, gateways and signaling servers.

- End systems are electronic devices with which clients or users place and receive calls. These end systems responds and initiate to signaling, and receive and transmit media. These devices also maintain the track of calls and their status.

- Gateways are devices that allow calls to be placed to and from other telephone networks.

- Signaling servers handle the application-level control of the routing of signaling messages. They are typically used to perform user location services; a signaling server can maintain information about where a user can currently be found and forward or redirect call setup requests to the appropriate current location. Signaling servers are the devices, which, from the point of view of feature-creation, are most similar in functionality to service control or switching points in the circuit-switched network; they can programmatically direct, block, or alter call-signaling messages based on their own internal logic.

HOW VoIP WORKS

VoIP works in the following ways:

1. Initially at the source, Analogue to digital convertors (ADC) converts the analogue signals into digital signals.
2. Then there is speech compression. Traditional telephone Networks use pulse code modulation (PCM) at 8K samples per second. Sklar (2001). 12-bit samples are compressed and expanded ('compounded') by a nonlinear

ISSN: 2277-9655
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look-up table into 8-bit words giving a transmitted rate of 8 Kbit/s. The compression typically used by an Internet phone today is of the order of 16 to 1 (128 Kbit/s to 8kbit/s). Sklar (2001).

2. After conversion of voice packets into data packets, RTP (real time protocol) are used for time stamping and content identification of user datagram protocol (UDP) voice packets.
4. Signaling system then performs its work and it does the following tasks:
(a) Try to find out the destination IP address
(b) After finding destination IP address and it establishes communication with that party.
(c) After negotiating the Internet protocol performs voice compression, buffer Length and time stamping of packets and starts communication.
4. Finally, at the receiving end, packets have to be disassembled for data extraction and for converting; the data into analogue voice signal and send those signals to the soundcard of the respective device.

Methodology
The research employed simulation based on OPNET modeler. The simulator was used to analyse the performance of a wireless network infrastructure without VOIP and compared with the performance with the incorporation of VOIP.

The existing System
Currently the existing infrastructure wireless of Koforidua Technical University depicts the architecture of public switched telephone network.
There are seven access points on the institution's wireless infrastructure (spectra network). Radio waves are transmitted from the spectra's access point in Accra and received by the main access point located at the Central Classroom (CCB) block. The main access point at CCB repeats the signal using a MACROTIK wireless router to the School of Business and Management Studies (SBMS). From the SBMS the signal is received by the access point located at the Faculty of applied science (FBMS) and it's also routed to the computer science department.
At the computer science department, the access point distributes to two other access points. These access points are located at the Library and the Faculty of engineering. There is a sub access point at the Administration Block that receives its signals from the CCB access point.
The central access point that is located at CCB does assigning of IP address to clients. The IP address assigned to the clients are found in the 192.168.1.0 pool. IP lease is for 24 hours.
There is no VoIP extension on any of the IP address.
The institution has to be employ the service of Vodafone, which provides services for the voice calls.
The following are the locations that form the core of spectra wireless of the institution:
- Central Class Block (CCB)
- Faculty of Business and Management Studies (FBMS)
- Computer Science Block (CSD)
- Faculty of Applied Science and Technology (FAST)
- Faculty of Engineering (FOE)

SIMULATION OF THE SPECTRA WIRELESS OF KOFORIDUA TECHNICAL UNIVERSITY.

THE PROPOSED SYSTEM
In the proposed system, we add a VoIP extension to the existing wireless network.
Internet service from SPECTRA is received through radio waves to the CCB block, being the main distribution point for the institution's wireless network. The proposed system would follow the same architecture of the existing system but would have support for VoIP Calls with PGM quality .The main workstation would be located at CCB and that would distribute the signal in a point-to-point mode to all other workstations. All workstations would support the file heavy transfer, streaming, video conferencing and VoIP call protocols that are found in the profile configuration of the network .The workstations are grouped into two giving them a different BSS ID. This is reported in figure 1.

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Figure 1. The proposed system BSS ID setup

Device specifications for the proposed system

- Wireless Local Area Network (WLAN) Ethernet router
- WLAN mobile workstation (advanced)
- Ethernet server
- Application configpalatte
- Profile configpalatte

Device-by-device configuration

WLAN Ethernet router: the wireless Ethernet router in this setup functions as the main distributor of wireless signals to all the nodes on the network. It classify each node into a different BSS ID group to prevent interference. The routers support a data rate of 48mbits per second. Other parameters are submitted in figure 2a and 2b

Figure 2a. Basic router configuration

ISSN: 2277-9655
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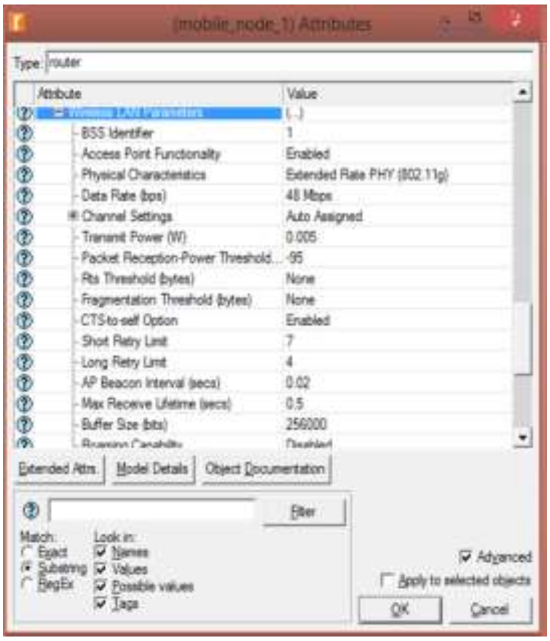

Figure 2b. Advanced router configuration

Wlanmobile workstation advanced: the workstations in this context represent a phone, computer or any other media that can be used to connect to the wireless network .Each workstation has support for all application profile which support the VoIP call application. This means that each workstation have inherited a VoIP extension therefore VoIP calls can be made on these workstations. Other configurations are reported in figures 3a and 3b

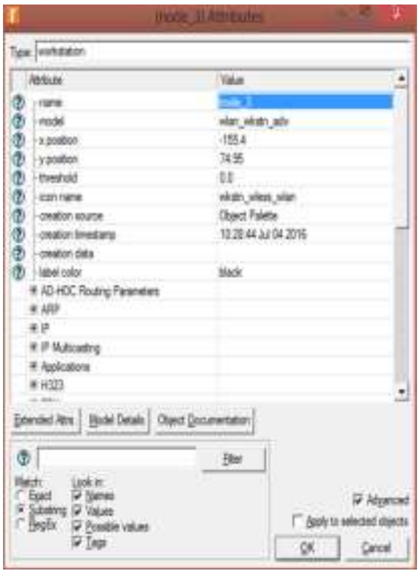

Figure 3a. Basic workstation configuration

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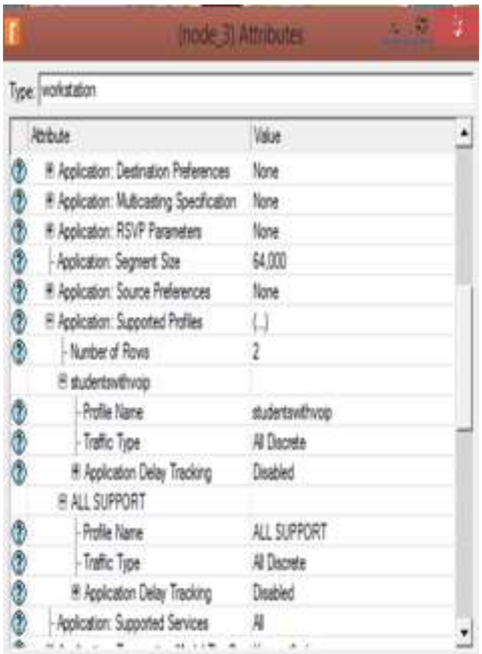

Figure 3b. Advanced workstation configuration

Application configpallate
The application configuration pallate contains all the applications that the work station, routers, servers and more importantly the profile configuration can support on a network. The total number of applications is 24 with VoIP and 22 without VoIP. All other parameters are reported in figures 4a and 4b

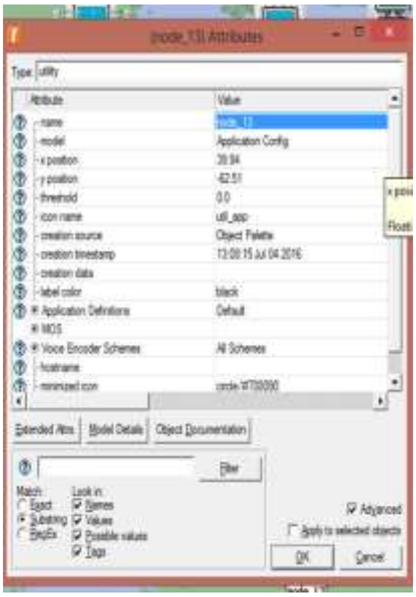

Figure 4a. Basic configpallate configuration

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Impact Factor: 4.116
CODEN: IJESS7

Profile configpallate

The "Profile Config" node can be used to create user profiles. These user profiles can then be specified on different nodes in the network to generate application layer traffic. The application defined in the "Application Config" objects are used by this object to configure profiles. Applications are created using the "Application Config" object before using this object. This is shown in figure 5. Figure 6 shows VoIP application support for profiles and figure 7 show the individual profiles.

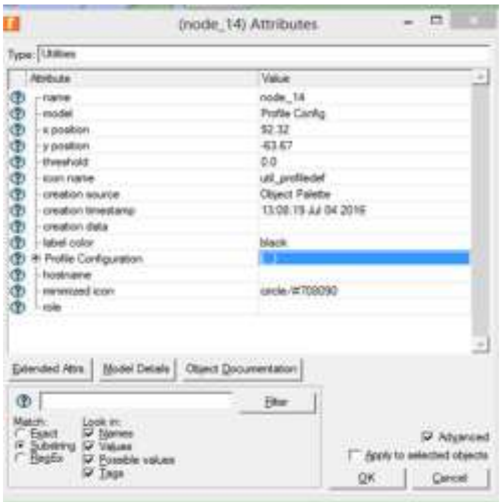

Figure 5. The profile configpallate

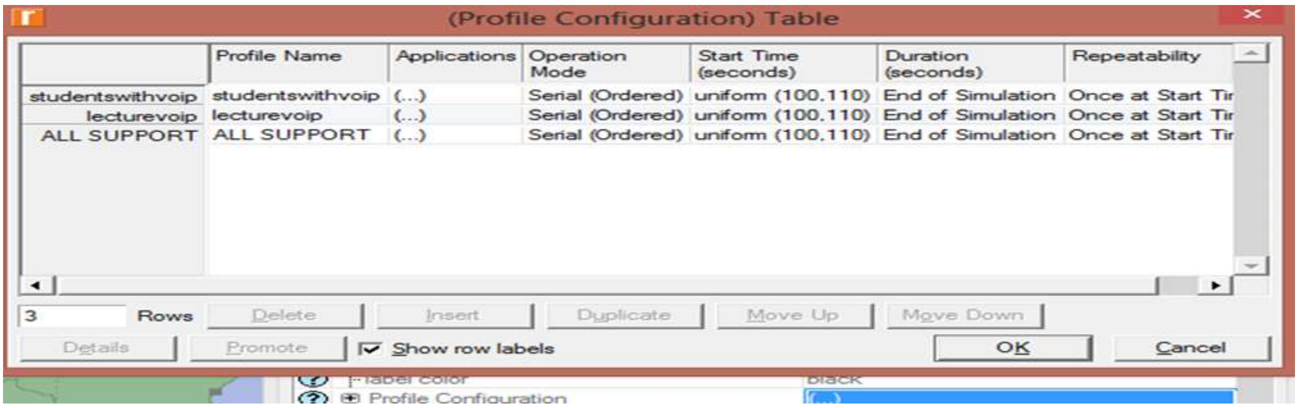

Figure 6. VoIP application support for profiles

ISSN: 2277-9655
Impact Factor: 4.116
CODEN: IJESS7

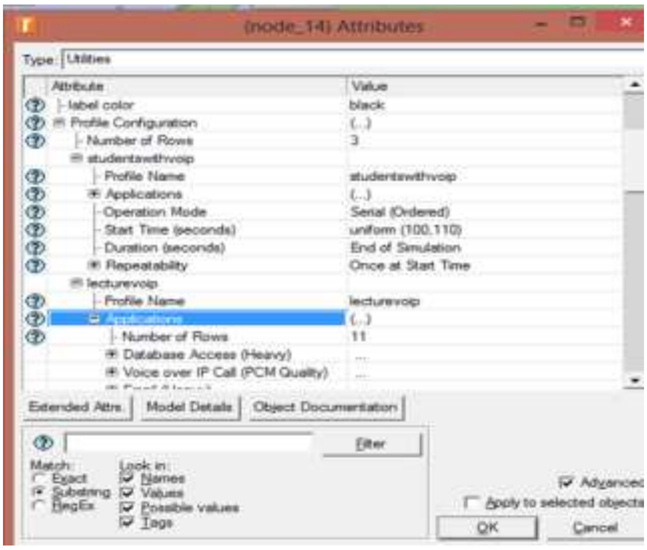

Figure 7. Individual profiles

SIMULATION RESULTS DISCUSSION

Data dropped

In this criteria, we determined the total higher layer data dropped in bits/sec by the all the WLAN Media Access Control Addresses (MACs) in the network because of consistently failing retransmissions. We also looked at the statistic reports of the number of the higher layer packets that are dropped because the MAC couldn't receive any acknowledgements (ACKs) for the (re)transmissions of those packets or their fragments, and the packets' short or long retry counts reached the MAC's short retry limit or long retry limit, respectively. Figure 8 shows the results from the simulation.

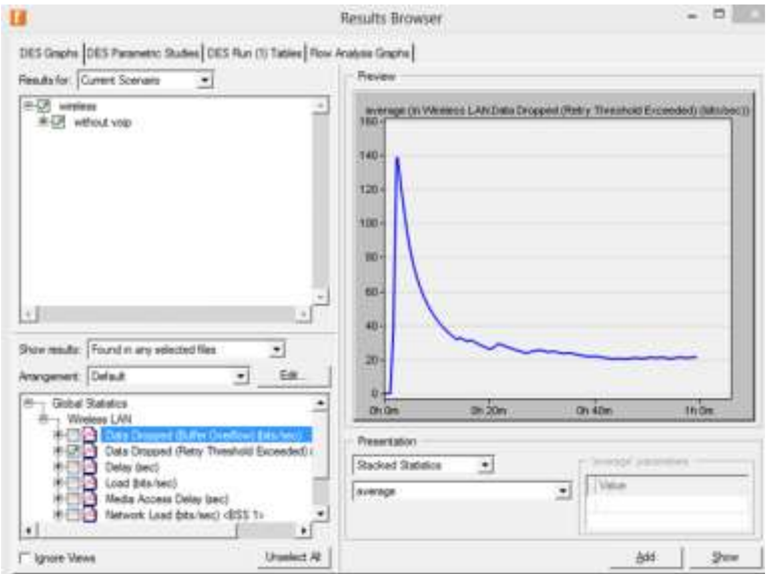

Figure 8. Simulation result of the network without VoIP

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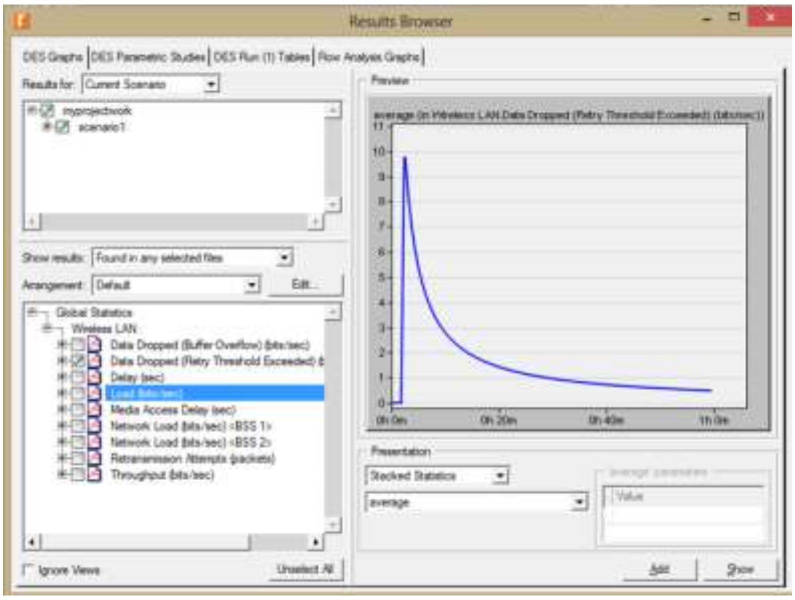

Figure 9. Simulation result of the network with VoIP

X axis = time in mins, y axis = bits/sec (speed)

From figure 8, the network without VoIP the data dropped at its peak when the time was 0mins at 139 bits per second whilst figure 9, the network with the VoIP had a maximum data drop of 9.5 bits/second at time 0mins.
After time 0min, the network without VoIP, had a steady drop of data until at time 16 mins. After time 16 to 35 mins, the data drop rate became unsteady until finally assuming a flat rate of 20bits per second after time 40min. The flat rate of 20bits per second was the minimum data lost on the network after the complete 1 hour simulation time.
Meanwhile on the network with VoIP, the data dropped continued in a steady motion and took a flat rate at time 50min with the bit rate at 5bits/sec. the continuous steadiness of the network with VoIP is the presence of advanced protocols such as the real time transport protocol and the real time transport control protocol which makes sure that all data sent on the VoIP network is received by all nodes in real time and effectively. Schulzrinne et al. (2003).

Load bits/sec:
This analysis is based on the total load (in bits/sec) submitted to the wireless LAN by all higher layers by all WLAN nodes of the network. The statistics are submitted in figure 10 and figure 11.

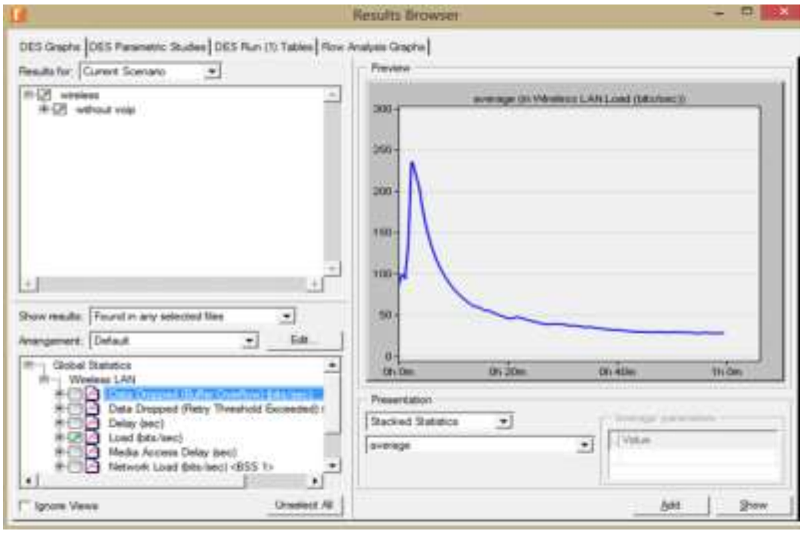

Figure 10. Simulation result of the network without VoIP

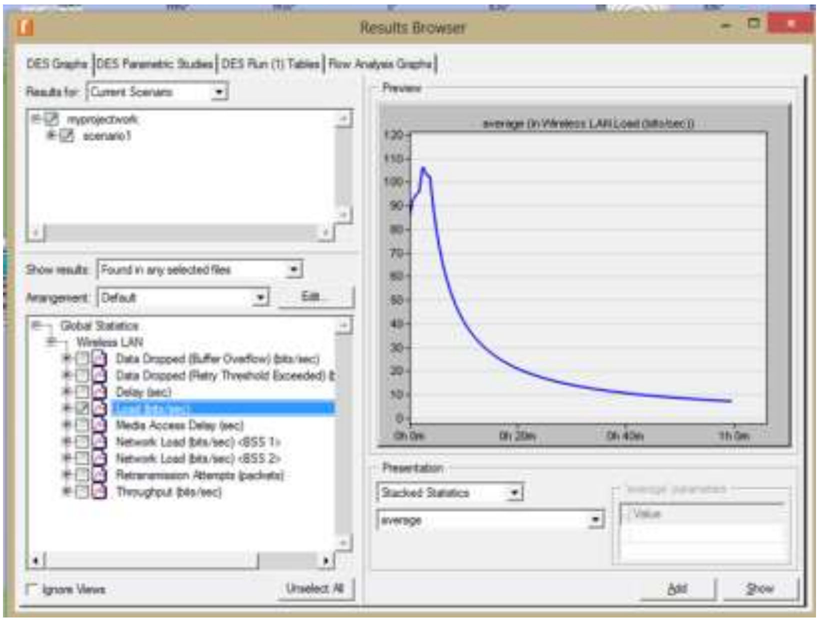

Figure 11. Simulation result of the network with VoIP

X axis = time in mins y axis = bits/sec (speed)

From the figure 10 which is the network without VoIP, the network load developed a little burst when the data rate was 100 bits/ sec. This burst is as a result of attenuation on the wireless network. Aguayo et al. (2004). Link-level measurements from an 802.11 b mesh network. After the burst, the recorded maximum amount of load was at the time 2 mins and the rate was 240 bits per second. After this point, the load decreased to 50bits/sec with time 15 before taking a flat rate of 30bits/sec whilst on the network with VoIP. With figure 11, the burst occurred a little above the 100 mark and took a continuous steady drop from 100 bits /sec to 20 bits/ sec before taking a flat rate of 5bits/sec. With these results we can be conclude that the load of the network without VoIP is lower as compared to the network which didn't have the VoIP support.

ISSN: 2277-9655
Impact Factor: 4.116
CODEN: IJESS7

Network load (bits/sec) <BSS1>
The statistic represents the total data traffic (in bits/sec) received by the entire WLAN BSS from the higher layers of the MACs that is accepted and queued for transmission. This statistic doesn't include any higher layer data traffic that is rejected without queuing due to full queue or large size of the data packet. Any data traffic that is relayed by the AP from its source to its destination within the BSS is counted twice for this statistic (once at the source node and once at the AP), since such data packets are double-loads for the BSS because both the source node and the AP have to contend for their transmissions via the shared medium. The results are submitted in figure 12 and figure 13.

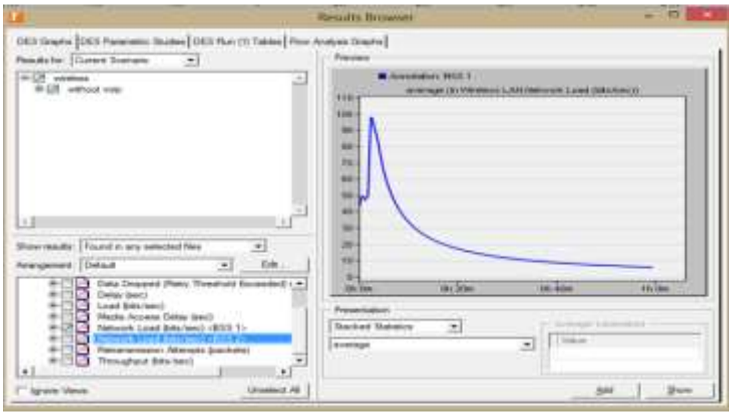

Figure 12. Simulation result of the network without VoIP

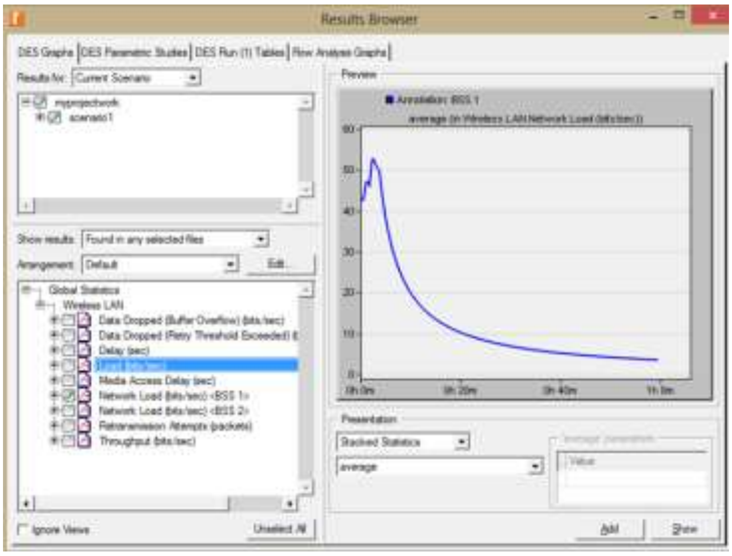

Figure 13. Simulation result of the network with VoIP

X axis = time and the y axis = bits/sec
The network load in the basic set service 1 (BSS 1) of the network without VoIP recorded 95 bits per second as the highest amount of load that is sent from the router to the nodes and the nodes to the router, it experienced a little surge/burst at 50bits/sec when the time is a little above 0 mins .
Both networks experience a steep slope of load falling from its peak until it reached a flat rate. The flat rate recorded in figure 12 was 5bits/sec and same was recorded in figure 13. This means the introduction of VoIP in the wireless network does not affect the network load on individual routers that serves as access points to all other nodes on the network even if it does the result is negligible .

ISSN: 2277-9655
Impact Factor: 4.116
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Network load (bits/sec) <BSS 2>

The statistic represents the total data traffic (in bits/sec) received by the entire WLAN BSS from the higher layers of the MACs that is accepted and queued for transmission. This statistic does not include any higher layer data traffic that is rejected without queuing due to full queue or large size of the data packet. Any data traffic that is relayed by the AP from its source to its destination within the BSS is counted twice for this statistic (once at the source node and once at the AP), since such data packets are double-loads for the BSS because both the source node and the AP have to contend for their transmissions via the shared medium. The result for the analysis are reported in figure 14 and figure 15.

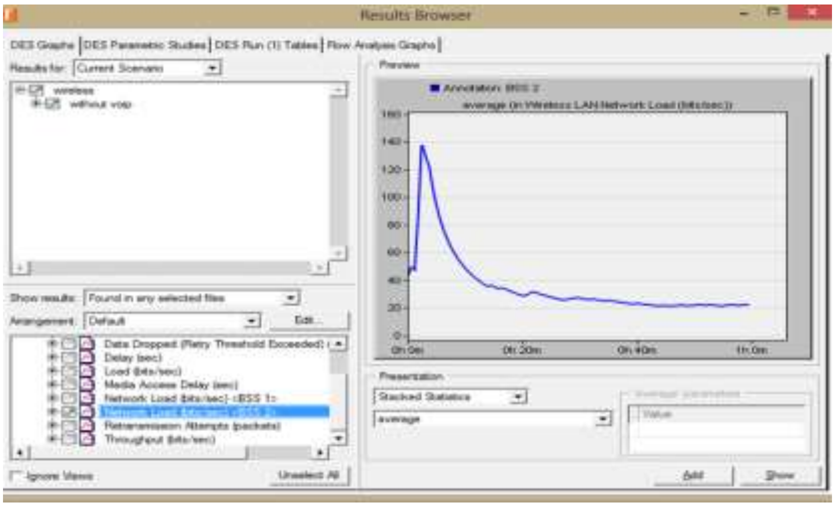

Figure 14. Simulation result of the network without VoIP

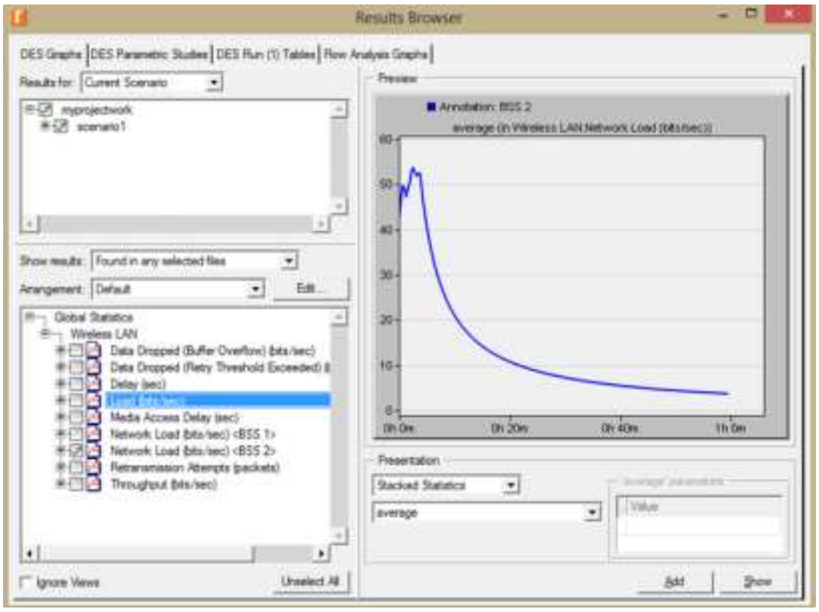

Figure 15. Simulation result of the network with VoIP

X axis = time and the y axis = bits/sec

the network load in the basic set service 2 (BSS 1) of network 1 recorded 138bits per second as the highest amount of load that is sent from the router to the nodes and the nodes to the router , it experienced a little surge/burst at 50bits/sec when the time is a little above 0 min's which is negligible . The burst occurred at this point due interference by other electromagnetic waves. Tong (2016).

The slope in figure 14 is less steeper than that of figure 15 because in figure 15, the VoIP protocol known as media gateway control protocol, divided the functions required for other applications the on the network therefore maintain a steady slope. Arango et al. (1999).

CONCLUSION

From the simulations made, we can conclude that the VoIP extensions can be introduced on networks with a wireless infrastructure without affecting its performance. This is so because, the data needed to be sent across the VoIP network is maintained with less data drops compared to the network without VoIP. In addition, the amount of data that can be sent on an individual node to ensure performance, is less on the VoIP network due to protocols such as real-time transfer protocol (RTP) and real-time transport control protocol (RTCP) that ensures that all packets are sent through the layers of the network in real-time.

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