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Investigating the Performance of VOIP over WLAN in Campus Network

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ABSTRACT

Voice will remain a fundamental communication media that cuts across people of all walks of life. VoIP, a new technology has been increasingly popular in recent times due to its affordability and reduced cost in making calls over broadband Internet. This paper uses simulation method to specifically investigate the performance of VoIP over wireless LAN for an increased number of VoIP calls, the use of different coding scheme and increased number of workstations in video conferencing. With this, a determination of the actual number of VoIP calls that each wireless Access point can adequately support with enhanced voice quality was made alongside with the coding scheme that yields the best quality of service in wireless LAN.

KEYWORDS: Voice Over Internet Protocol (VOIP), Public switched telephone Network (PSTN), Jitter, Internet,

1. INTRODUCTION

Wireless Local Area Network (WLAN) has become one of the most important technologies in networking[1]. It has helped to simplify networking by enabling multiple computers to connect, share resources without the use of extensive wiring. WLAN can be found in different areas of applications: airport, cafeteria, educational institutions, companies, hospitals etc[2] . WLAN helps computers users to share computer resources such as broadband Internet connections and networked printers. With wireless networking we can achieve the same speed and capability as wired-line network without difficulties of layering and drilling into walls or putting up wires/Ethernet cables through office buildings or homes.

In most of our educational institutions (colleges, Universities), campus networks are used to connect different academic buildings, administrative blocks, libraries, laboratories and school hostels. In this information age where people need to be up to date with the happenings around them, universities are building wireless LAN as integral part of their networks. The Wireless LAN enables staff and students to access information anywhere within the campus with no restrictions. Lecturers and students with their wireless devices like laptops and PDAs can use the Internet including real-time communication. With the benefit of increased productivity, network scalability, flexibility and lower cost, wireless LAN is now an affordable network model for Voice over Internet Protocol (VoIP) implementation. Universities install VoIP software to enable lecturers and students to make cheap voice, video and conference calls at lower rates.

Voice over Internet Protocol (VoIP) is a technology that makes it possible for users to make telephone calls over the Internet or intranet networks. The technology does not use the traditional Public Switched Telephone Network (PSTN); instead calls are made over an internet protocol data network. VoIP has great benefits of increased saving, high quality voice and video streaming and several other added value services. Examples of VoIP software are: Skype, Google talks and windows live messenger[1].

There are a lot of metrics that determine the performance of VoIP when deployed into a network. These metrics are the number of clients present in the network, the type of compression/decompression (codec) scheme and VoIP quality determinant (data loss, consistent delay characteristics otherwise known as jitter and latency which leads to echo in the system. Much research has been carried out on quality of service (QOS) which is crucial in real time communication especially VoIP and how it affects its performance in Wireless LAN. This paper extends this to include VoIP compression/decompression (codec) scheme, increasing the number of client in the wireless environment and most importantly how increasing the number of clients in video conferencing affect VoIP system in Wireless LAN.

2. LITERATURE REVIEW

Before a network is deployed with voice over internet protocols (VoIP) system, it is important to measure its readiness to support it[3] . This is done by injecting real traffic into the network and measuring for delay, jitter and loss. Modelling and simulation can also be used.

Bur Goode[4] studied the quality of voice call through compression/decompression algorithm by using subjective testing under controlled condition to determine mean opinion score (MOS). Vauhatupa T et al[5] **,** used N-2 network simulator and Mean Opinion Score (MOS) to evaluate the performance of IEEE 802.11b wireless LAN to support multi-hop VoIP services. The authors discovered that, the number of hop between VoIP transmitters in IEEE 802.11b wireless LAN has effects on call quality. Di Wu[1] evaluated the performance of VoIP traffic characteristics over Ethernet LANs. Kotz D & Essien K[6] presented a comprehensive study of network activities of over two thousand user in large productive wireless network. Saleh & Alkhoraidl[3] used simulation models to study the deployment of VoIP. Angel Cuaevas et al[7] investigated the number of VoIP calls IEEE 802.11b/EEE 802.11e wireless LAN can support. Mona Habib & Nirmala Bulusu[8] studied the possible shortcomings of wireless network in the area of Quality of Service (QoS) and security as compared to Ethernet LAN standard. The Lin Cai et al [9]] investigated the recent advances in QoS enhancement mechanism of Voice over IP (VoIP) in wireless LAN medium Access Control (MAC) layer. They discovered that IEEE 802.11 WLAN can only support limited number of voice connection due to MAC protocol inefficiency.

Raghuraman Rangarajan et al[10] presented Access Control List (ACL) based scheme to ensure quality of service (QoS) for VoIP call admission. Pailo Dini et al[11] studied and evaluated the relationship between voice call quality perceived by user and basic network using ITU-T E-modelling. Moncef Elaoud et al[12] identified a suitable evaluation metrics to quantify the performance of voice calls.

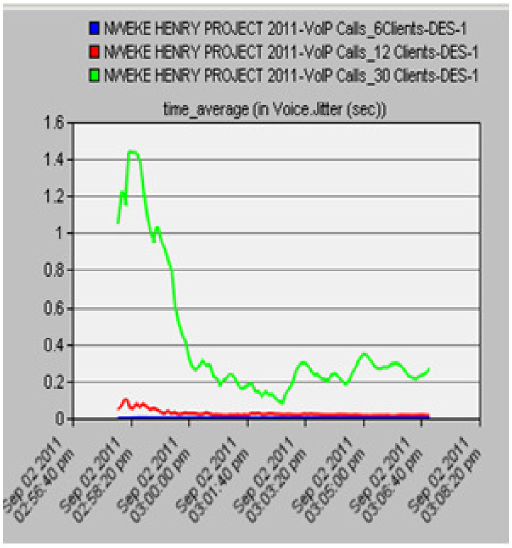
3. RESEARCH METHODOLOGY

The simulation methodology was used to investigate the performance of VoIP over Wireless LAN in Campus Network. Simulation enables network designers and engineers to test a designed model on a platform which imitates the real environment. The proposed network or system can be created, modified and studied in order to propose the behaviour of the network designed hence predicting its strength and weakness before implementing the model in a real environment[13]. According to Di Wu [1] simulation is the most used research methodology for investigating network performance, modelling, design, analysis and evaluation. Network with large number of nodes will be costly to build without knowing how it will perform in real physical environment. So, simulation is used to deal with the performance of physical quality of VoIP network system before it is deployed.

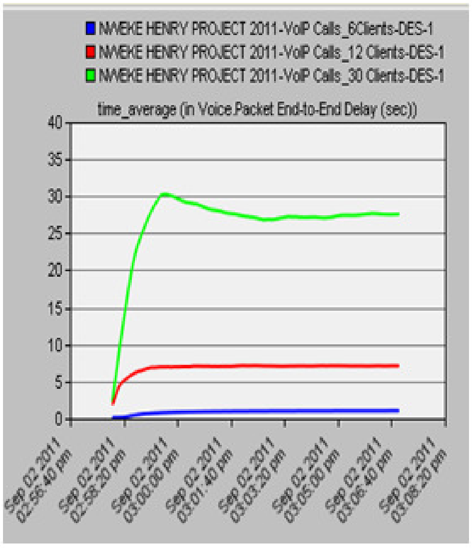
4. SIMULATION RESULT AND ANALYSIS

The OPNET environment was used to configure the VoIP application and components such as VoIP traffic, workstation, wireless router, Application Definition and Profile Definition. Each of these components was configured to obtain statistics in order to investigate the performance of VoIP in WLAN. The simulation was set for 10 minutes (600 seconds) and VoIP traffic began at 60 seconds after it started. The simulation lasted for 600 seconds. A presentation and analysis of the graphical results obtained during the simulation is shown. Section 1 discussed the effect of increasing the workstation from 6 to 30 workstations; section 2 discussed the use of different coding schemes with voice frame packet set to 2 and 5 while the section 3 compared the effect of increasing the number of clients in video conferencing over wireless LAN.

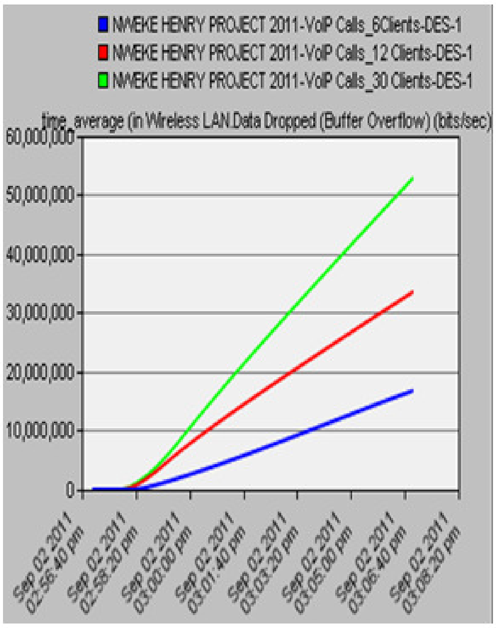
Section 1 Scenario 1: Increasing the VoIP workstations in the network



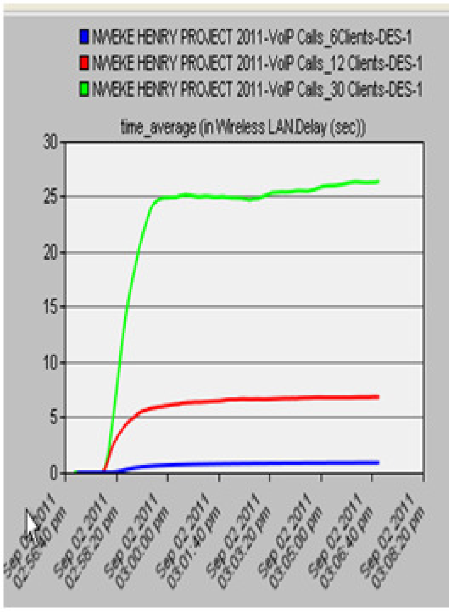
(a) Voice Jitter



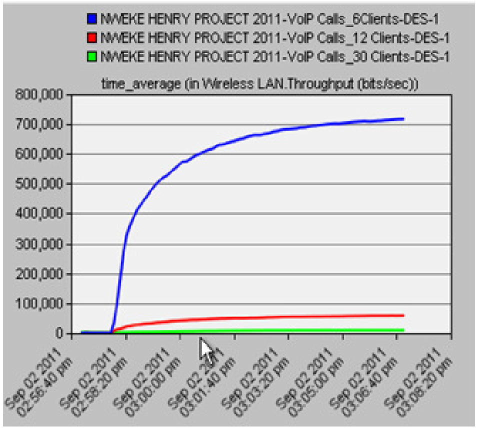
(b) End-to-End Delay



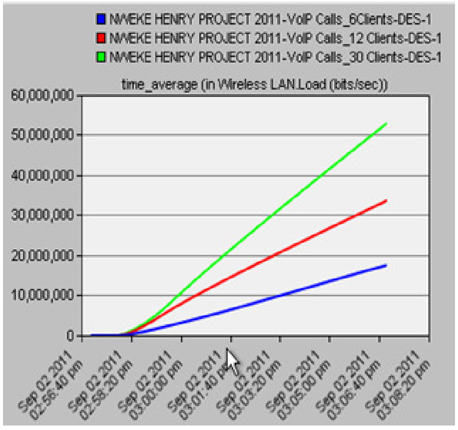
(c) Data dropped



(d) Wireless LAN Delay



(e) Throughput



(f) Wireless LAN Load

Figure 1: Performance of VoIP over Wireless LAN with increase of clients from 6 to 30

Figure 1(a-f) shows the simulation result of increasing the number of number of workstations participating in simultaneous VoIP calls in the network. Figure 1a shows the voice jitter. From the graph, there is an increase in voice jitter from 2.6ms to 1.45second. The voice jitter threshold for VoIP network is about 1.0ms. So there is high increase in jitter as more number of workstation is added to the network. This increase in jitter leading to packet arriving at different time will make the voice difficult to understand. The use of less number of clients (e.g. two clients) will make the jitter less and better performance of VoIP application in wireless LAN.

Figure 1b shows the voice packet End-to-End delay in the network. The voice packet End-to-End delay changed from 1.05 second (6 clients), 7 seconds (12 clients) to 30 second (30 clients). The acceptable Voice packet End-to-End delay threshold is 80ms which has been exceeded by the scenario. Therefore, increase in the network clients has great impact on the performance of VoIP over wireless LAN.

Figure 1c shows the Data dropped rate in the network. Increasing the number of clients will lead to higher data been dropped due to over flow of network traffic as seen in the graph above (figure 1c). From the graph, data dropped show an increase difference of 34Mb/s which is quite high for normal data dropped rate.

There is decrease in throughput and increase in wireless LAN delay as the VoIP clients is increased from six to thirty. Figures 1d and e show the Wireless LAN delay and Throughput of increasing the VoIP clients in the network. Throughput (average rate of successfully delivered data) data declined from 700, 000 bits per second, 59, 000 bits per second to 5, 000 bits per second while the Wireless LAN delay increases from 0.89 seconds (6 clients), 6.8 seconds (12 clients) to a high value of 26 seconds when thirty VoIP clients was added to the network.

Table 1 shows the summary of the experimental results in using different number of VoIP clients in the campus Wireless network.

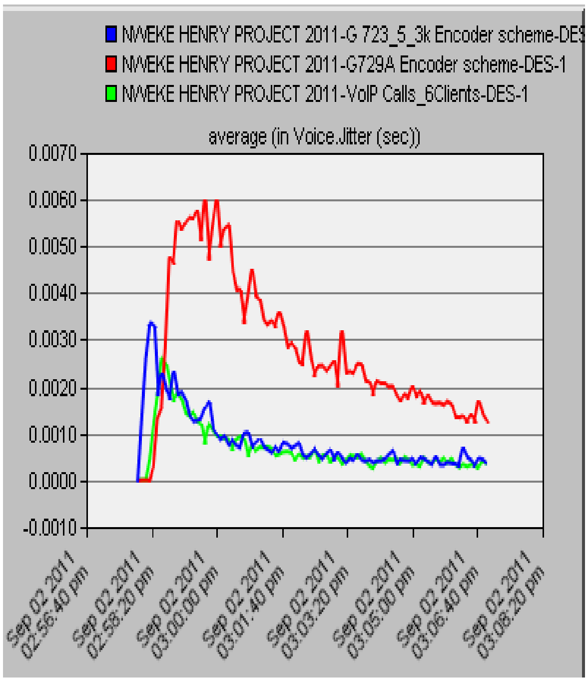
Table 5.1: Summary of experimental results (wireless VoIP clients)

No of Clients	Voice Jitter	Packet End-to End Delay	Wireless LAN Delay	Voice Traffic Sent (Packet/s)	Voice Traffic Received (Packet/s)	Packet loss rate (pps)
6	0.0026sec	1.05sec	0.89sec	5,004	4,798	4.12%
12	0.6 sec	7.0 sec	6.9	35,246	28, 562	18.96%
30	1.45 sec	30 sec	25 sec	51, 734	36,688	29.08%

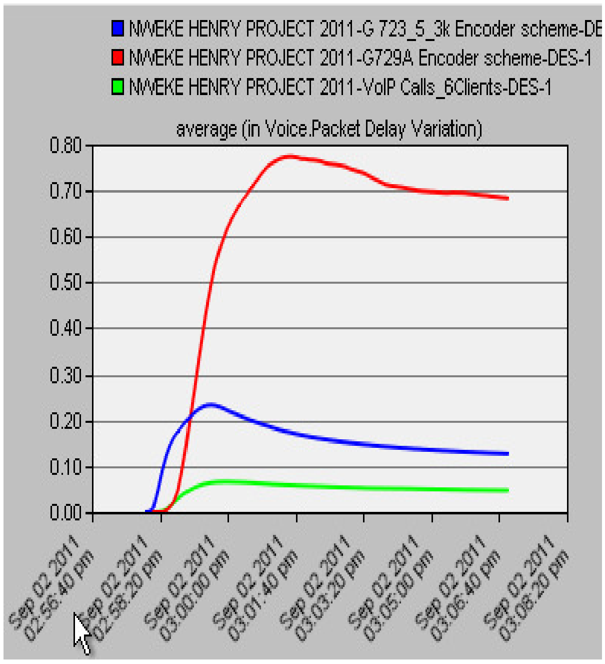
From the table, the percentage of packet loss increased as the number of wireless VoIP clients are added to the network. It is worthy to note that the simulation result has shown that the network can only support fewer number of VoIP clients (2 to 4 clients) for more enhanced performance.

Section 2 Scenario 2: Use of Different Encoder Scheme

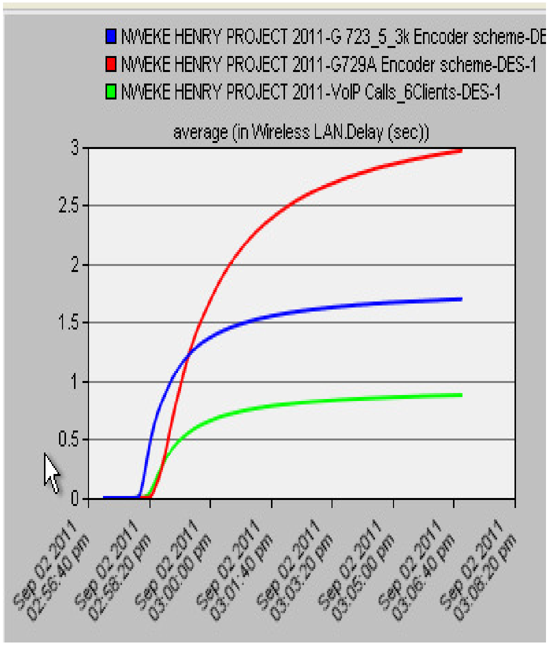
Scenario 2 simulation was carried out to measure the impact of encoder scheme on VoIP over wireless LAN. The simulation changed the encoder scheme from G.711, G.723 (5.3K) to G.729 and different voice frame per packet was used. Figure 2 a-f shows the simulation result obtained during the experiment.

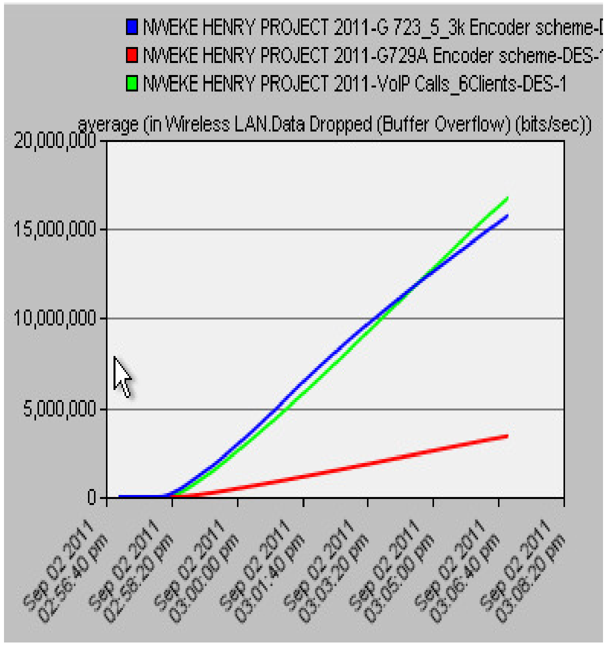


(a) Voice Jitter

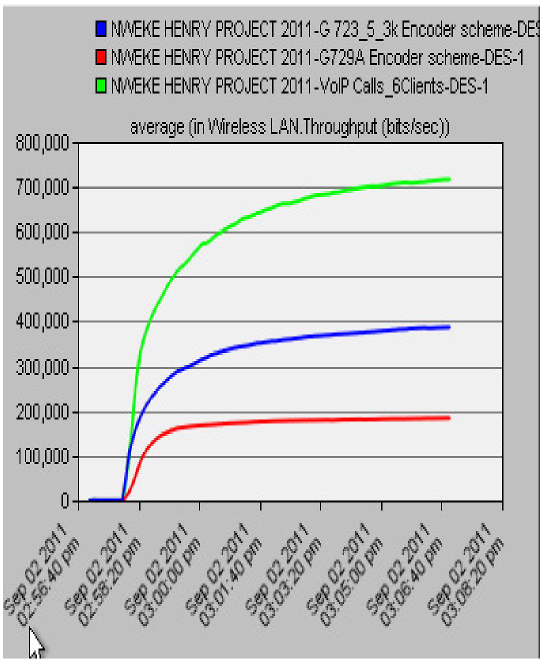


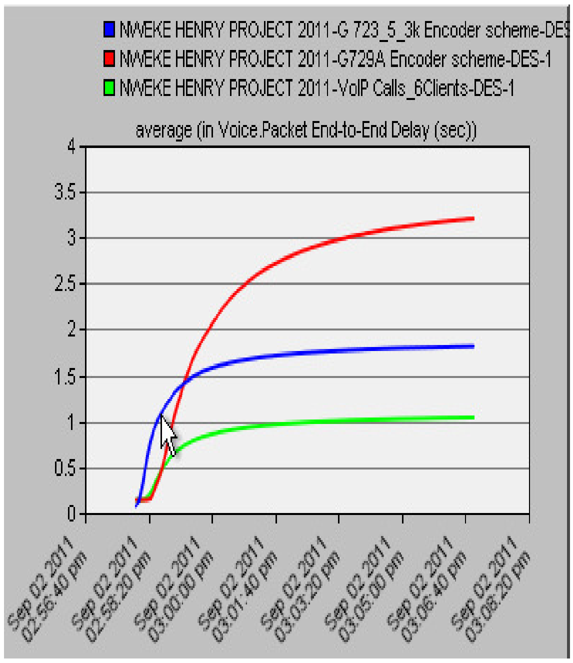
(b) Voice Packet Delay





(c) Packet End-to-End Delay **(d) Data dropped rate**





(e) Wireless LAN delay **(f) Throughput**

Figure 2 Performance of VoIP using different Codec scheme (voice frame per packet = 5)

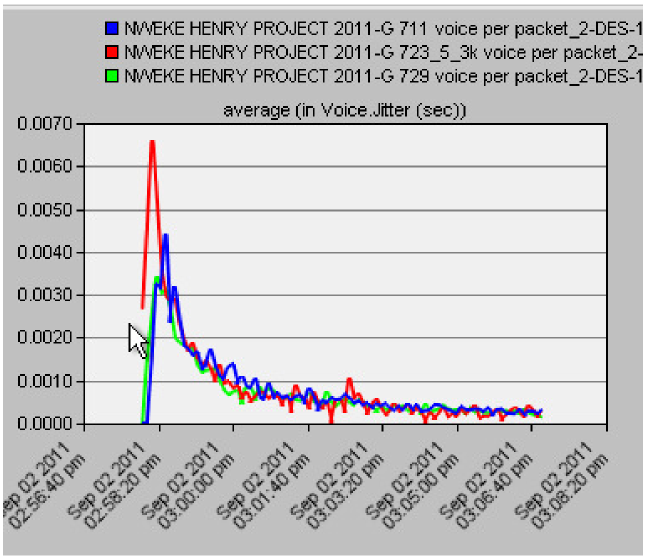
Figures 2a and 3a shows the Voice jitter comparison using different codec scheme. In figure 2a the voice frame per packet was set to “5”, while it was set to “2” in figure 3a. G.729 encoder scheme has the highest voice jitter value of 60ms while G.711 has the lowest 25ms. Each of the encoder scheme compared yielded a normal jitter value of less than 1ms. When the packet frame per packet was set to 2 (figure 3a), the G.729 encoder scheme jitter decreased to 32ms and G.723 (5.3K) encoder scheme has the highest encoder scheme value of 66ms. All the jitter values are still within acceptable value boundaries.

Figures 2b and 3b shows that the voice packet delay variations, G.729 codec scheme yielded the highest voice packet delay variation in the network having a value around 780ms and G.711 given the lowest value of 5ms. When the voice frame per packet was decreased to 2, G.729 still maintained a high jitter value of 260ms, G723 (5.3K) with a value of 240ms and G.711 given the lowest value of 80ms.

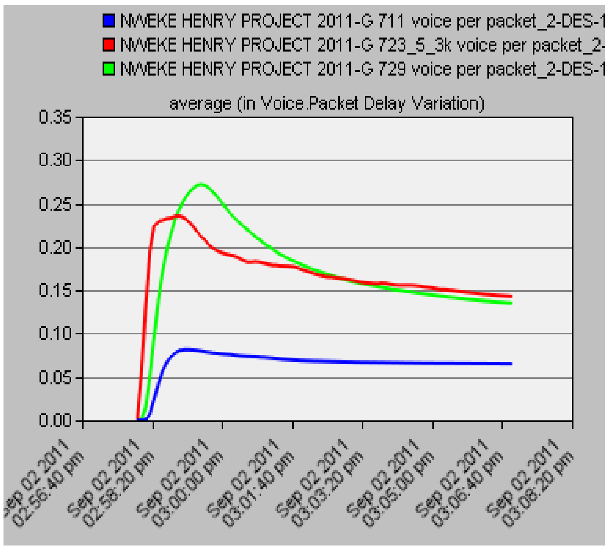
The network gave a very high end to end delay when different codec scheme was used with G.729 given the highest end to end delay of 3.75 second (figure 2c) and G.711 with the lowest value of 1.00 seconds. These values are considered unacceptable for good network performance; the acceptable end to end delay is below 200ms. Even the decrease of the voice frame packet to 2 does not give an acceptable end to end delay as G.711, G.723 (5.3K) and G.729 encoder schemes yield 1.4sec, 2.5sec and 1.8sec respectively (figure 3c). So G.723 (5.3k) has the highest end to end delay of 2.5 seconds. This shows that the acceptable number of clients must be below six (6) to achieve better performance for VoIP over WLAN.

Figures 2d and 3d shows the Wireless Data dropped rate in the network. G.711 encoder scheme has the highest Data dropped rate with a value around 15,000,000bit/sec and G.729 yielded the lowest Data dropped rate below 5,000,000 bits/sec. When the voice frame per packet is set to 2, G.723 encoder scheme Data dropped rate increase to around 23,000,000 bits/sec (figure 3d). This shows that voice frame per packet values has significant effect on the encoder scheme use in Voice over Internet protocol in WLAN. The higher the voice frame per packet, the better the performance.

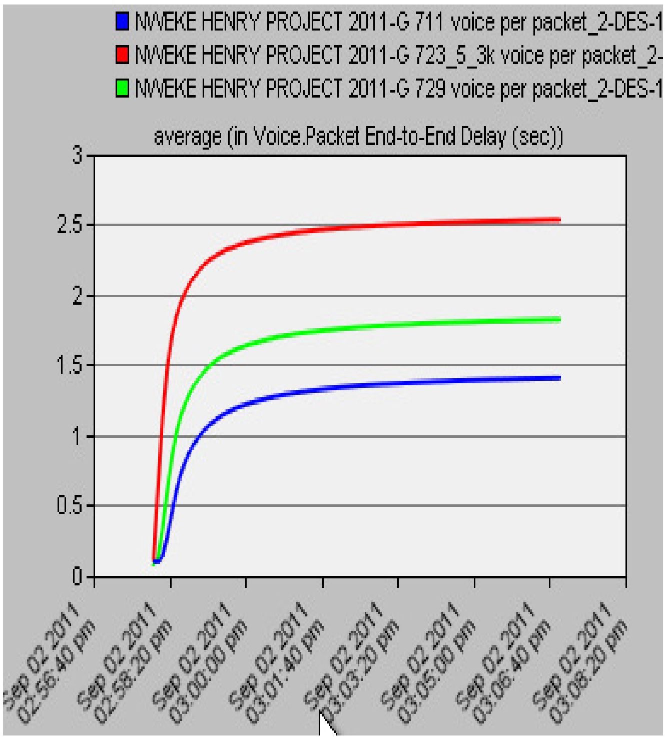
Figures 2e and 3e shows the Wireless LAN delay of VoIP using different codec scheme. The voice frame per packet is set to 5 and 2 respectively. From figure 2a, G.729 encoder scheme has the highest wireless LAN delay (3 seconds) and G.711 has the lowest delay (0.8 seconds). When the voice frame per packet is set to 2, the wireless LAN delay of G.723 (5.3k) increased from 1.8 seconds in figure 2 e to 2.5 seconds in figure 3e. Because we are dealing with voice, the delay is very high as it will significantly affect the quality of VoIP performance. This figures also shows that voice frame per packet of each encoder scheme has great effect on their performance. In each of the simulation, encoder scheme G.711 has the highest throughput. Figures 2 f and 5.3f shows throughput of 700,000 bits/sec and 500,000bits/sec respectively. Therefore, G.711 encoder scheme has the highest rate of throughput among the three investigated. This gives the reason for it acceptability in the Voice over IP implementation.



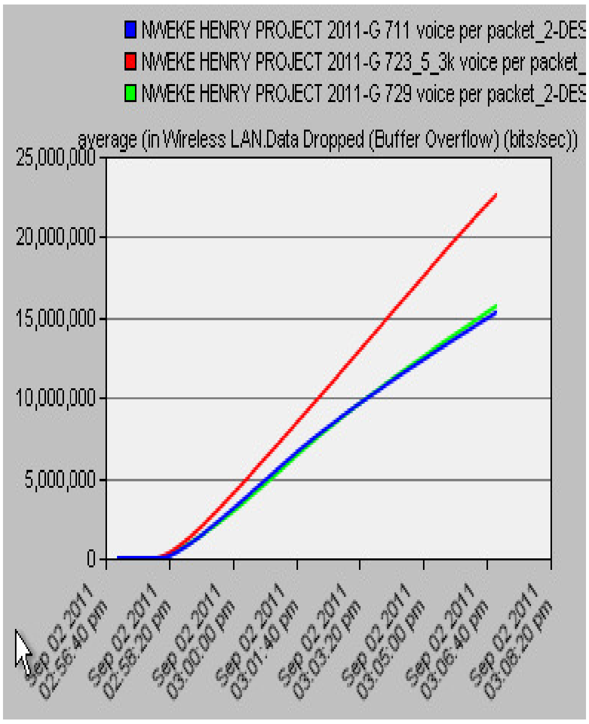
(a) **Voice Jitter**



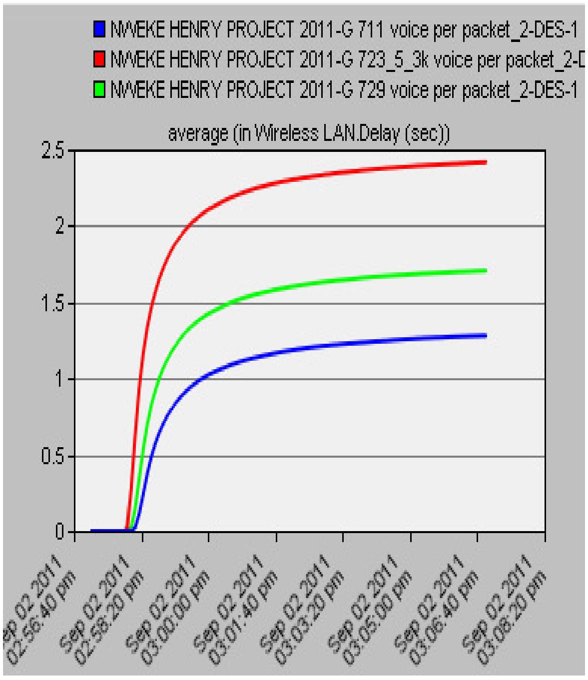
(b) **Packet delay Variation**



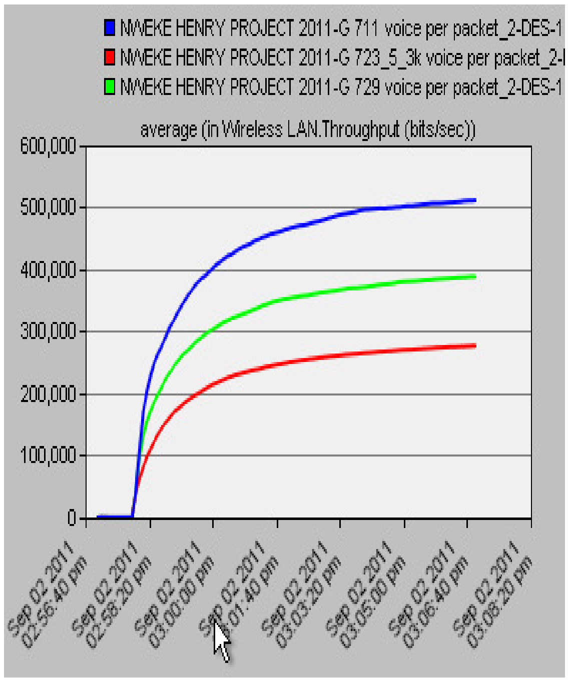
(c) End-to-End Delay



(d) Data dropped rate



(e) Wireless LAN Delay



(f) Throughput.

Figure 3: VoIP performance using different codec scheme (voice frame per packet = 2).

Table 5.2 shows the summary of experimental result obtains during the simulation.

Table 5.2: Summary of experimental result using different codec scheme

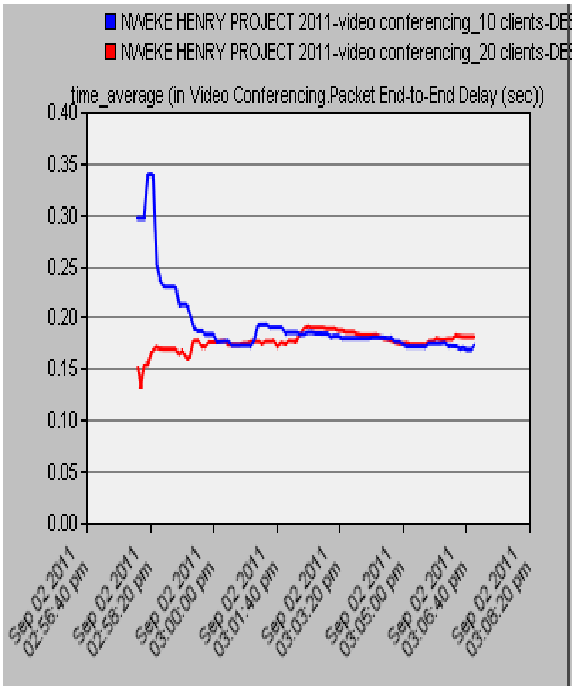
Voice Codec Scheme		**Voice Jitter**	**Voice Packet delay variation**	**Packet End-to-End delay**	**Wireless LAN delay**	**Voice traffic Sent(packet/sec)**	**Voice traffic received (packet/sec)**	**Packet loss rate (pps)**
Voice frame per packet= 5	**G.711**	0.0025	0.005	1.00	0.80	10,348	9,988	3.48%
	G.723 (5.3K)	0.0032	0.260	1.75	1.55	20,036	19,966	0.35%
	G.729	0.0060	0.780	3.50	3.00	10,256	9,896	3.51%
Voice frame per packet= 1	**G.711**	0.0045	0.080	1.40	2.40	12,896	11,992	
	G.723 (5.3k)	0.0066	0.240	2.50	1.20	44,522	44,522	0.00%
	G.729	0.0034	0.260	1.80	1.70	20,036	19,892	0.72%

Scenario3: Use of high numbers of clients in video conferencing

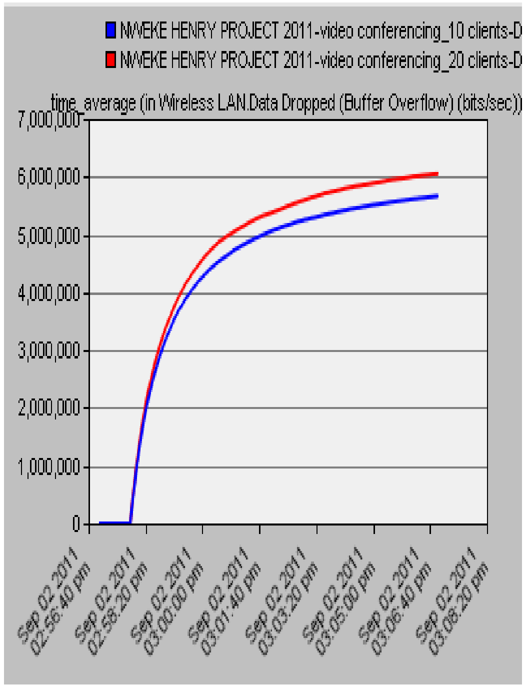
This scenario was carried out to measure the impact of video conferencing over Wireless LAN in campus network. The scenario measures the effects of network parameters on the increase of wireless workstations for student and lecturers using wireless devices like laptops. The parameters measured include End-to-End delay, Data dropped rate, Wireless LAN delay and Throughput.

Figure 4 (a-d) shows the simulation result of the experiment. Figure 4a shows the End-to-End delay of video conferencing when the number of clients is increased from 10 to 20. The ideal End-to-End delay in wireless network is around 80ms to have acceptable video conferencing calls. From figure 4a, both scenarios exceed the stipulated threshold. When the number wireless clients are ten (10), End-to-End delay started from 300ms and increased to about 340ms and increase in the wireless clients to twenty shows End-to-End delay starting from 150ms with a slight increase. In both scenarios, the End-to-End delay is unacceptable as it is beyond the acceptable threshold. This indicates the network can only support between 2 to 6 clients for good quality video conferencing calls.

Figure 4b showed the data dropped rate in the network. The impact of increasing the workstations indicates a high data dropped rate due to over flow of network traffic. The data dropped rate only show a slight increase when the number of workstation is increased (200kb/s). There is also a steady decrease in the number of data dropped rate as the number of workstations is reduced.



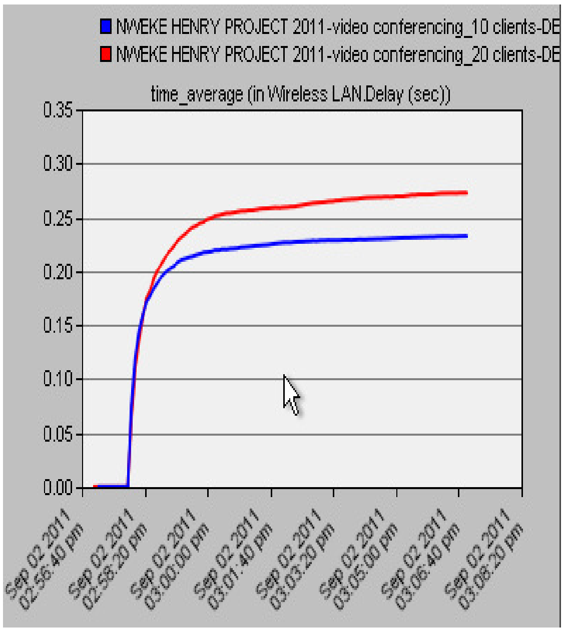
(a) Packet End-to-End delay

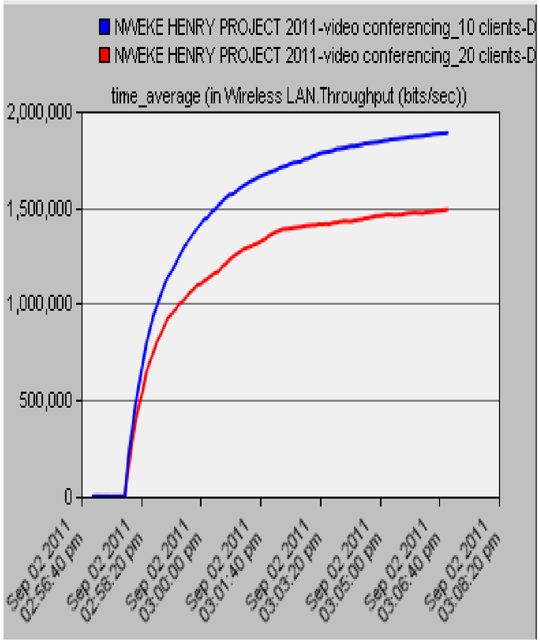


(b) Data Dropped Rate

Figure 4a shows the wireless LAN delay of packet received and sent by the wireless nodes across the network. The figure indicates the wireless delay difference with ten workstations and twenty workstations. The average delay difference is about 60ms.

Figure 4d shows the wireless Throughput which indicates the sending and receiving of success packet in the network. The average Throughput show an increase from 1.5Mb/s (20 clients) to about 1.8Mb/s (10 clients), an increase of 300kb/s. It shows slightly normal Throughput in both scenarios. The decrease in Throughput when the number of workstations is increased to 20 is due to the volume of packet sent to the wireless access point for distribution the clients, delay and data dropped in the network





(c) Wireless LAN delay **(d) Wireless LAN Throughput**

Figure 4 : Increasing the number of clients in video conferencing

5. Limitations

Despite the effort to obtain accurate results, some limitation may not be ruled out in the research. The paper considers the performance of VoIP over IEEE 802.11g wireless LAN and VoIP related services. OPNET tries to represent real-life network traffic scenario, it may not be as perfect as real-life network. So, when applying this to real-life network, effort should be made to obtain more realistic model for better quality of service for all the services and scenarios considered in the dissertation.

6. Conclusion

Voice communication no doubt remains a generally accepted means of communication. Investigating the quality of service based on different criteria is therefore important. This paper has provided a detailed explanation of the results of the simulation experiment and justifies it using graphs and tables. It also measures such factors like increase in wireless LAN clients, encoder scheme and use of high number of clients which has great impact on the performance of VoIP over wireless LAN.

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