ENSC 427: Communication Networks

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OPNET Final Project

Analysis of Voice over IP performance on Wi-Fi networks

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List of Acronyms

DS0: Digital Signal 0

DS1: Digital Signal 1

ETE: End-to-End

IEEE: Institute of Electrical and Electronics Engineers

IP: Internet Protocol

LAN: Local Area Network

MOS**:** Mean Opinion Score

PCM: Pulse Code Modulation

PPP: Point-to-Point Protocol

QoS: Quality of Service

VoIP: Voice over Internet Protocol

Wi-Fi: Wireless Fidelity

WLAN: Wireless Local Area Network

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1. Abstract

Voice over IP (VoIP) has revolutionized the telephone industry. VoIP transcends limitations of traditional phone service (most notably expensive long distance phone calls) and brings the world closer together. Public access Wireless Fidelity (Wi-Fi) networks, also known as Hotspots, are increasing in popularity and are now widely available in North America. They allow for the high speeds required to utilize VoIP, while providing the freedom of mobility. The focus of this project is on analyzing the quality of VoIP calls using Wi-Fi networks and comparing it to a typical wired broadband internet connection

2. Introduction

2.1 Background

VoIP involves digitization of voice streams and transmitting the digital voice as packets over IP-based packet networks like the internet, LANs (Local Area Networks) and WLANs (Wireless Local Area Networks) [1].

In Wi-Fi networks packets are transmitted over a wireless IEEE 802.11 network. IEEE 802.11 is a set of standards carrying out WLAN computer communication in the 2.4, 3.6 and 5 GHz frequency bands [2]. It is important to determine whether VoIP transmission over a wireless network can provide a QoS (Quality of Service) comparable to that of the existing cellular networks [1]. Some parameters affecting the QoS of a VoIP on WLAN include jitter, packet loss, packet End-to-End (ETE) delay and MOS value. The project aims to measure these factors in a VoIP system implemented in the IEEE 802.11g standard.

Packet ETE delay is a property of packet-switched networks which can be used to measure network performance of the traffic on the network. It is generally the time it takes for a packet to reach its destination.

Jitter is the variation rate in the delay of received packets and it determines if information has been delivered smoothly and hence affects the quality of a voice call greatly [3].

MOS value is a measure of voice quality and it is expressed in one number between 1 to 5, worst and best respectively [4]. The following table provides a summary of the MOS value and their respective definitions (see table 1).

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

Table 1: MOS values [4]

Packet loss occurs when one or more packets routed across a network do not reach their destination. It can be caused by a number of factors including signal degradation over the network medium, over saturated network links, corrupted packets rejected in transit, faulty networking hardware, faulty network drivers or normal outing routines [5].

2.2 Motivation

Traditionally, mobile users were forced to use expensive and restrictive voice plans (limited number of minutes per month and during blackout periods) to place and receive phone calls. Recently, the speed of mobile internet networks has been increasing, however, the only downside is that currently data is charged by usage and is typically not affordable enough to replace voice data plans.

Furthermore, as mobile technology advances, the number of mobile phones with embedded Wi-Fi modules has increased dramatically over the last few years. Using Wi-Fi networks, it is now possible to harness the flexibility and power of VoIP. This means the user is free to use any VoIP provider, such as Skype, to make both local and long distance calls for a fraction of the cost.

2.3 Project goals

The main goal this project was to see if VoIP over Wi-Fi networks were a suitable replacement for traditional mobile phone usage. To do this, we tested some scenarios which roughly emulate typical mobile phone users.

3. Simulation Design

3.1 Voice application

The profile attribute dialogue box indicates the type of VoIP service used along with duration and repeatability of traffic (see figure 1). A wireless VoIP call is carried out between a caller/callee pair. The caller routs packets according to the Voice over IP Call (PCM Quality) attribute chosen in the profile attributes table. The destination preference in the caller's attribute dialogue box is set to support the mentioned VoIP service.

The callee of the network is set to also support the designated VoIP service in its own attribute box. The encoder scheme for the voice call is chosen as G.711 as shown in Figure 2. The G.711 represents the PCM samples used in voice transmission.

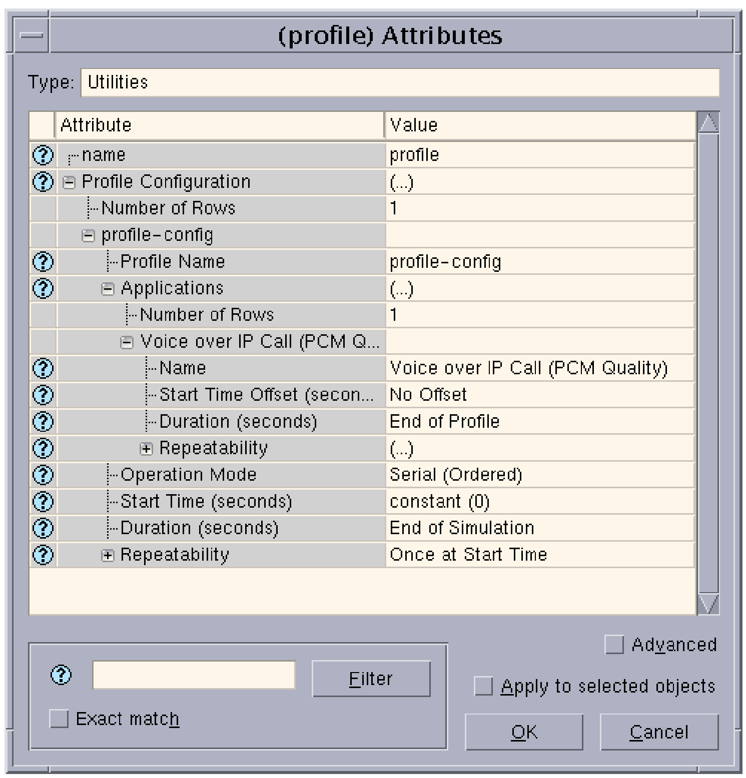
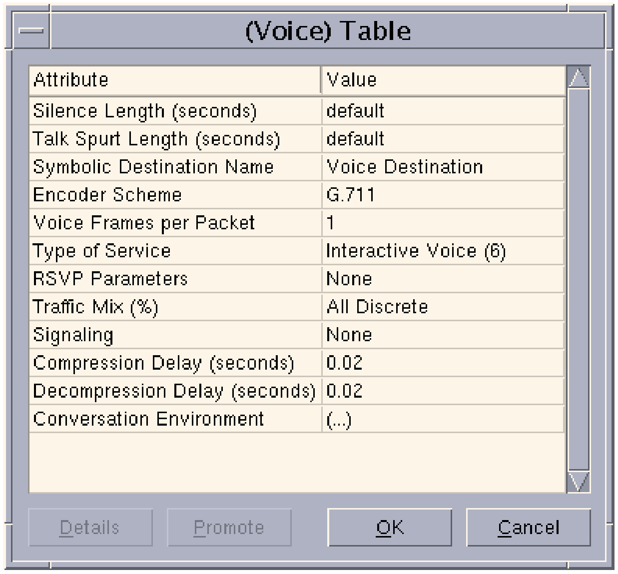


Figure 1: Profile attribute dialogue box


(Voice) Table

Attribute	Value
Silence Length (seconds)	default
Talk Spurt Length (seconds)	default
Symbolic Destination Name	Voice Destination
Encoder Scheme	G.711
Voice Frames per Packet	1
Type of Service	Interactive Voice (6)
RSVP Parameters	None
Traffic Mix (%)	All Discrete
Signaling	None
Compression Delay (seconds)	0.02
Decompression Delay (seconds)	0.02
Conversation Environment	(...)

Figure 2: VoIP application configuration

3.2 Ethernet Scenario

The first scenario we created was the Ethernet scenario that will be used as the baseline for all of our other scenarios. To create this scenario, we used two Ethernet workstation objects and an Ethernet/SLIP gateway. The workstations were connected to the gateway using 100Mbit Ethernet links and the gateway was connected to the IP (Internet Protocol) backbone using a PPP (Point-to-Point Protocol) link. The completed scenario is shown below (see figure 3).

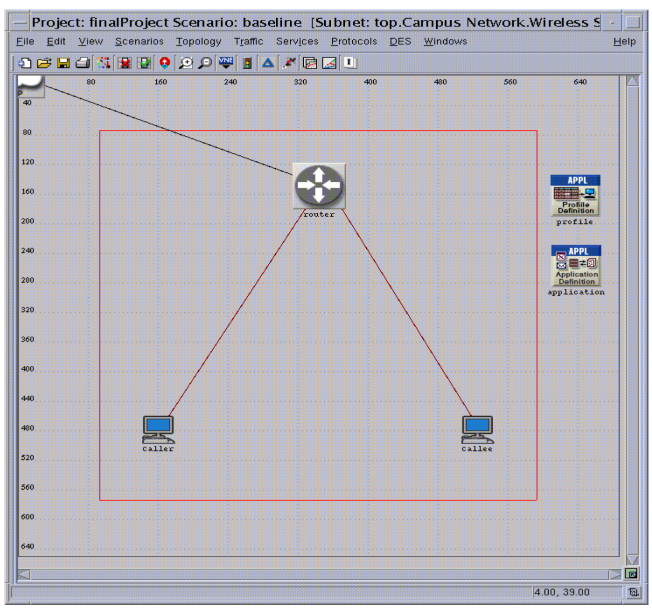


Figure 3: Baseline Ethernet Scenario

3.3 Wi-Fi (fixed) Scenario

To create the fixed Wi-Fi scenario, we used the OPNET Wireless Deployment Wizard to add a subnet to the workspace. This subnet contains two wireless workstation nodes and a wireless router connected to the IP backbone using a PPP link. The completed scenario is shown below (see figure 4).

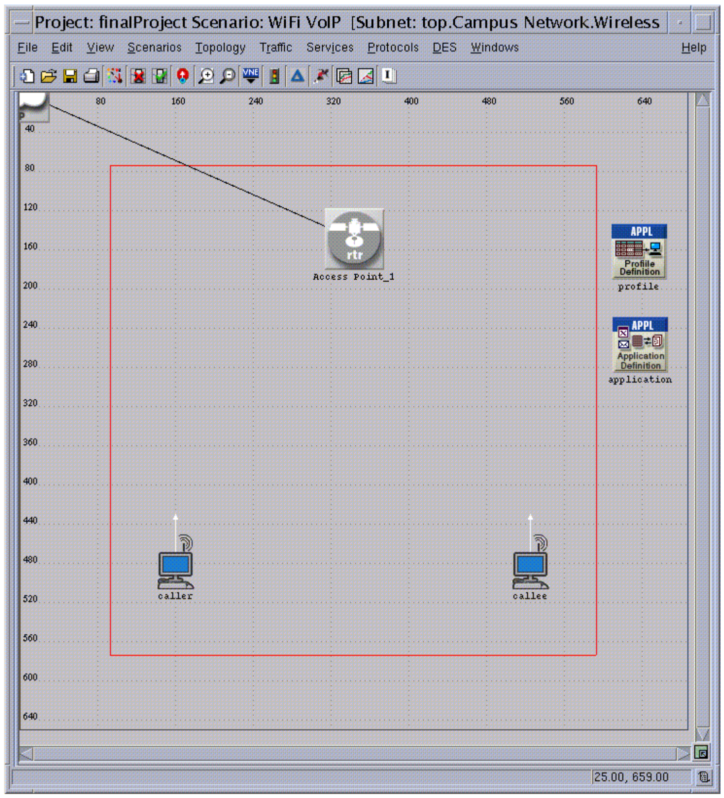


Figure 4: Wi-Fi scenario with two fixed workstations

The wireless workstations were place a fixed distance of 375 meters away from the router. This is the boundary distance for which network deterioration occurs due to the wireless range (see table 2 for node placement details).

Node	X position (meter)	Y position (meter)
Router	344	150
Caller	160	480
Callee	525	480

Table 2: Node positions

3.4 Wi-Fi (mobile) Scenario

One of the most important features of Wi-Fi networks is the mobility it offers the user. The mobile Wi-Fi scenario was created to roughly simulate a user moving around. To create this scenario, we duplicated the fixed Wi-Fi scenario and moved the caller (initially) closer to the router. The completed scenario is shown below (see figure 5).

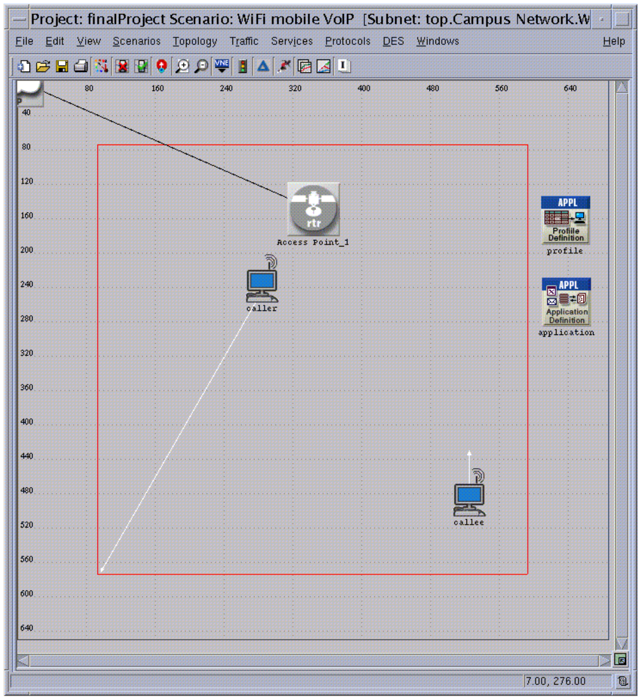
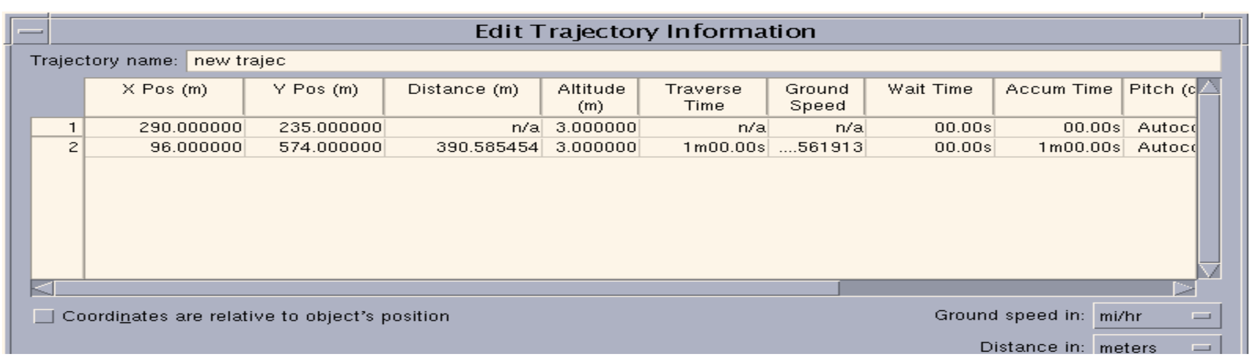


Figure 5: Wi-Fi (mobile) Scenario

Next we created a trajectory for the caller to follow. This path begins 100m away router and moves diagonally in the opposing direction. The details of the trajectory are shown in figure 6.



Edit Trajectory Information

Trajectory name: new trajec

	X Pos (m)	Y Pos (m)	Distance (m)	Altitude (m)	Traverse Time	Ground Speed	Wait Time	Accum Time	Pitch (c
1	290.000000	235.000000	n/a	3.000000	n/a	n/a	00.00s	00.00s	Autoc(
2	96.000000	574.000000	390.585454	3.000000	1m00.00s	561913	00.00s	1m00.00s	Autoc(

Coordinates are relative to object's position

Ground speed in: mi/hr

Distance in: meters

Figure 6: Trajectory of mobile node

3.5 Two-subnet network (with and without background load) Scenario

In this setup, the caller is located in wireless subnet_0 and sends the VoIP call to the callee located in the wireless subnet_1 (see figure 7). The caller and callee are set to support the VoIP PCM call as they did in the Wi-Fi VoIP scenario. The background load is introduced to observe the variations in the results produced and to compare and contrast those results with the ones obtained in the situation with no background load, i.e. the two-subnet VoIP over Wi-Fi. The details of the link load are described in the load profile section below.

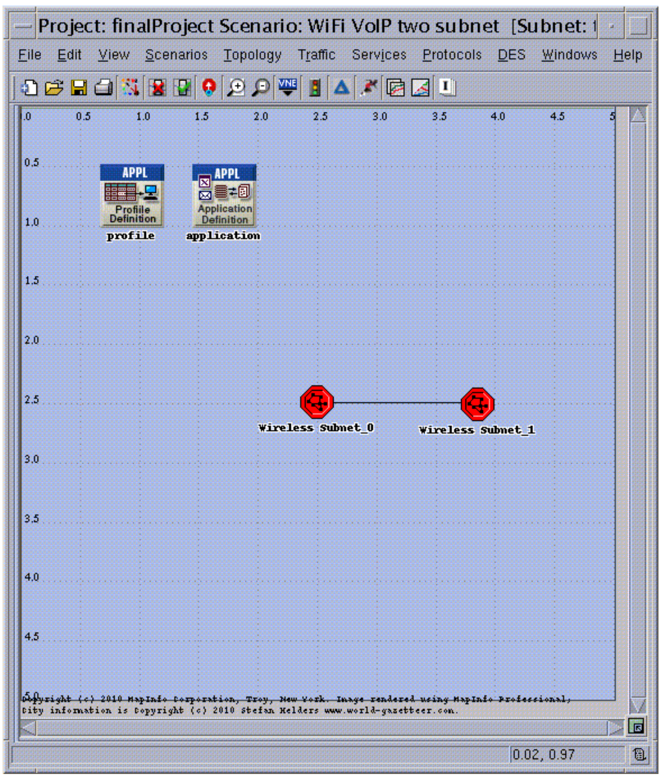


Figure 7: Wi-Fi two-subnet network including the caller/callee pair

The following two figures show the internal layout of wireless subnet_0 and wireless subnet_1 respectively (figures 8 and 9).

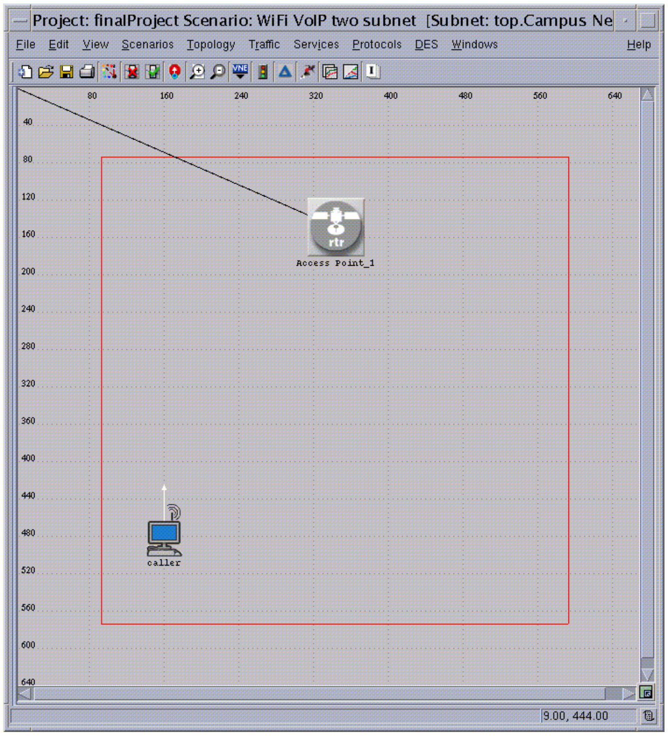


Figure 8: wireless subnet_0 layout

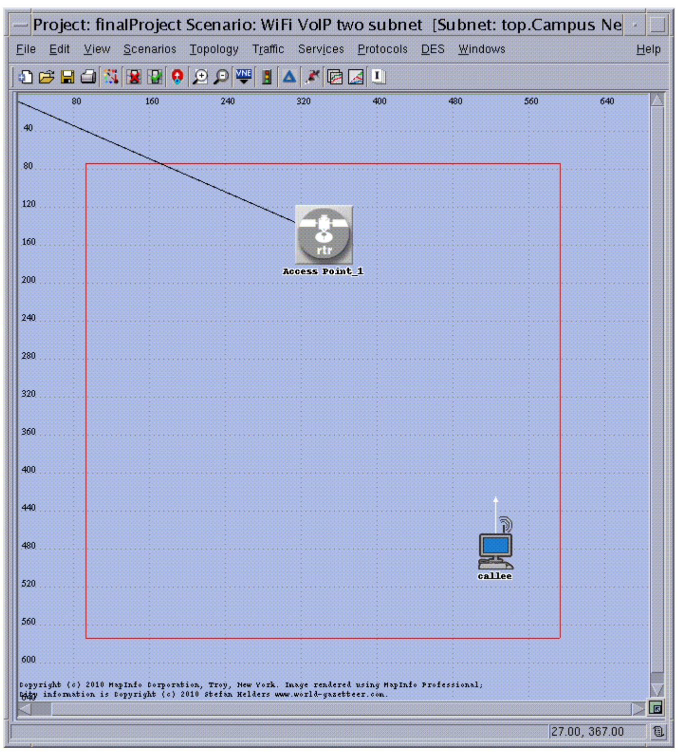


Figure 9: wireless subnet_1 layout

3.6 Load Profile

The link load was created to simulate the effects of network congestion on VoIP calls. We did this by creating a load profile for the DS1 link between the two subnets described above (see figure 7). The minimum speed required for our VoIP calls were 64 Kbps (or a DS0 link). Since DS1 links are made up of 24 DS0 links [6] the maximum load required to affect the quality of the call was 23*DS0 or roughly 1.472Mbps. Therefore, we created 6 uniform time intervals of 30 seconds each starting at a load of zero, and gradually increasing to 1.472Mbps. Finally, we took it one interval further until we had reached complete saturation of the DS1 link. The following figure shows the link load profile used in the simulation (figure 10).

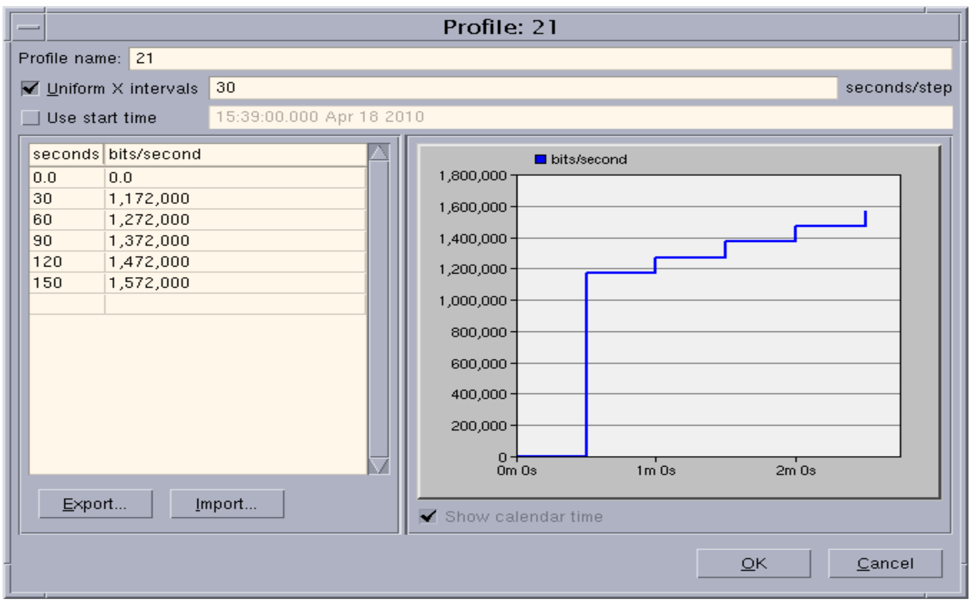


Figure 10: Load Profile

4. Simulation Results

4.1 Jitter

From the graph of jitter it can be seen that there is no noticeable jitter for the baseline scenario. In the Wi-Fi mobile scenario, jitter diminishes in between 20 and 30 seconds into the simulation, and in the VoIP scenario on the other hand jitter continues to occur until the end of the simulation. The maximum value of jitter experienced for the Wi-Fi VoIP scenario is close to 0.12 ms. The variations in the jitter appear to be higher for the Wi-Fi VoIP scenario because in the mobile scenario the trajectory traversed by the caller somewhat reduces the jitter as expected. The following figure shows the jitter experienced by the Wi-Fi VoIP, Wi-Fi mobile and baseline scenarios (figure 11).

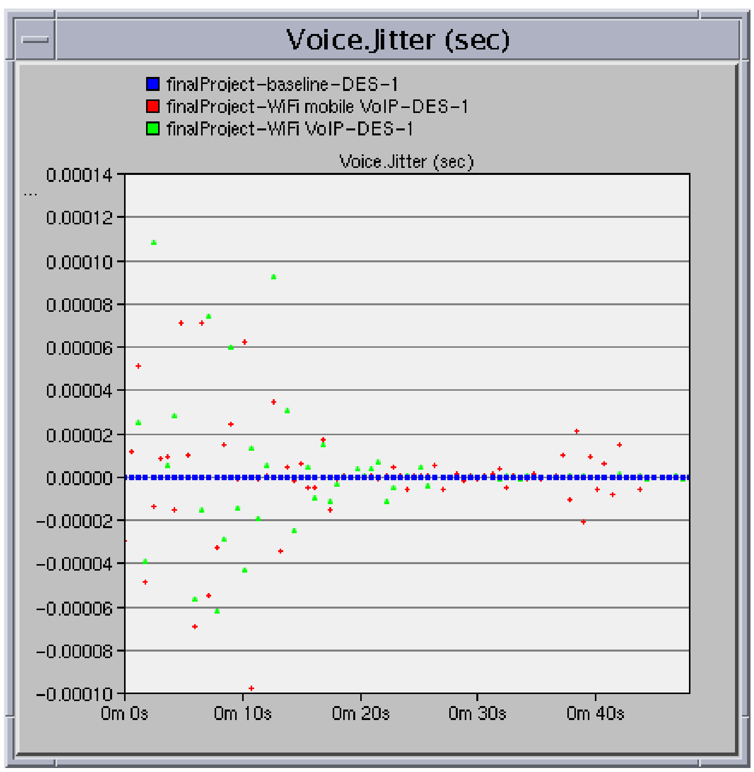


Figure 11: Jitter results

For the load scenario, the jitter is quite high as expected. This is because applying background load causes a significant delay in the arrival of packets to the destination (the callee node). The plot of the jitter experienced by the two-subnet scenarios with and without load is shown below (figure 12).

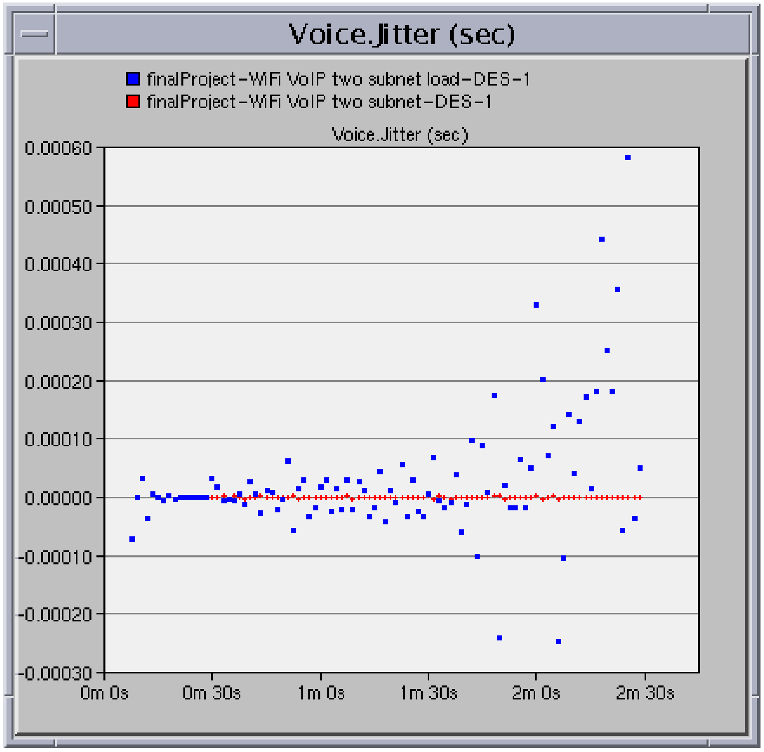


Figure 12: Jitter for load scenario

4.2 MOS Values

The plot of the average MOS value for the Wi-Fi VoIP, Wi-Fi mobile and baseline scenarios is provided below in figure 13.

From the graph it can be seen that the MOS value is quite high for the baseline scenario as expected. The high MOS value indicates the quality of the call is fair and at the same time imperfections can be perceived at the callee. The MOS value for the other two curves is close to 3 and this indicates that the quality of the call received can be annoying. In this project we used a full PCM codec; had we used a more compressed codec, we may have obtained better voice quality and hence a higher MOS value.

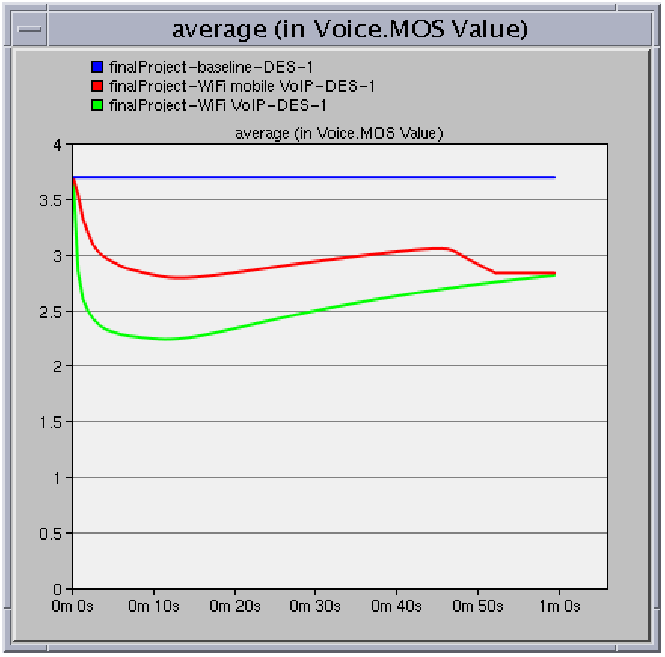


Figure 13: MOS values

The average MOS value for the scenario with no traffic load starts off at slightly below the 3.5 value (see figure 14) and ends at slightly above that value and remains constant thereafter. The introduction of load causes the MOS value to undergo abrupt changes in the beginning and end of the simulation. Therefore on an averaged scale, the quality of voice call is higher than the case with background load.

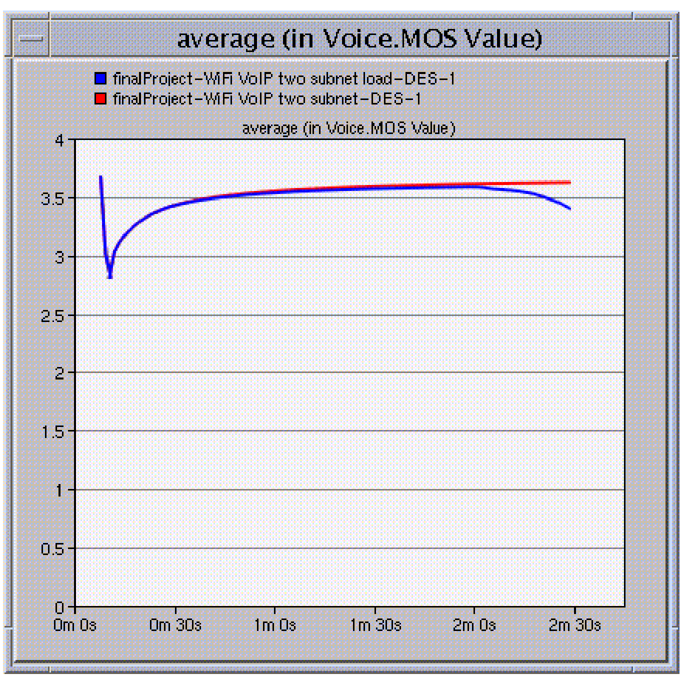


Figure 14: MOS values for load scenario

4.3 Packet Loss

Packet loss was measured by examining the total number of packets received during the voice call with the ideal value being 100 packets per second (we compared the results to this golden value). The following plots (Figures 15 and 16) show the number of packets received per second as a function of call duration.

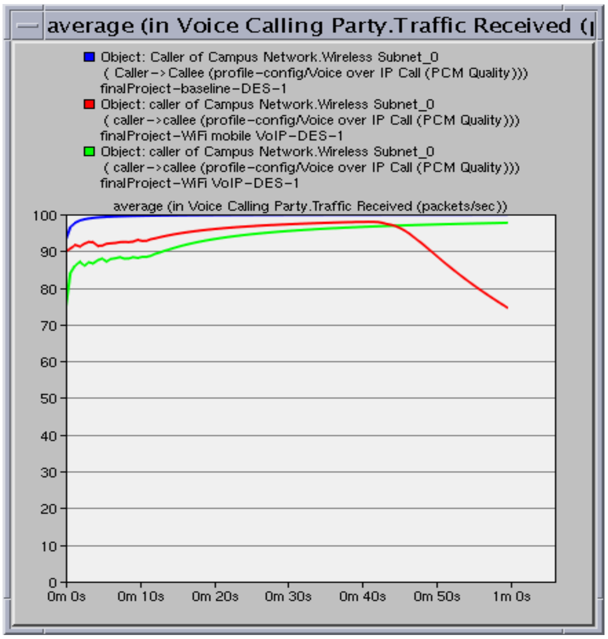


Figure 15: Packet Loss

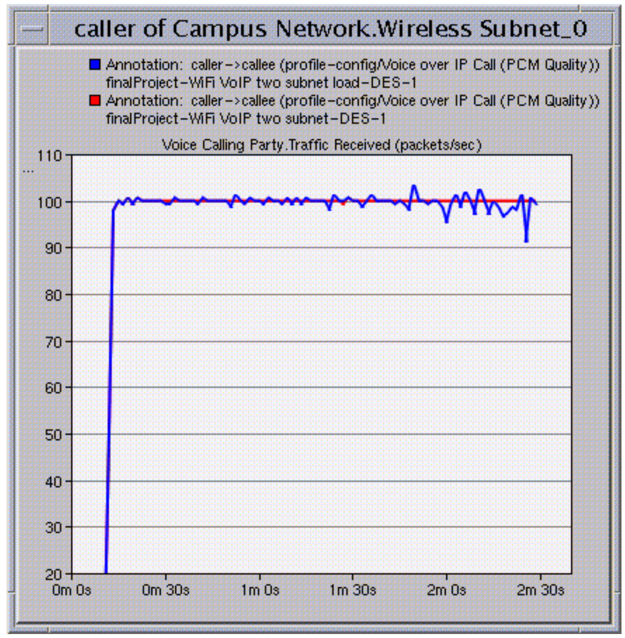


Figure 16: Packet Loss, two sub networks

As we can see, in the baseline scenario there is a perfect score of 100 packets/sec during the majority of the call. There are a few outliers during the first few seconds of the call, but this can be attributed to the call setup.

Looking at the Wi-Fi fixed scenario, we notice a slightly lower score of approximately 80 packets/sec as expected. As time goes by, this score gradually increases to the near baseline levels of 98 packets/sec.

Now, examining the results from the mobile Wi-Fi scenario, we can see that initially the packet loss is less than the fixed Wi-Fi scenario with a score of 90 packets/sec. This is due to the fact that the node is initially closer to the router then the fixed scenario. As the node begins to move away from the router, we can see a gradual increase of packets dropped compared to the fixed Wi-Fi scenario. At approximately 45 seconds into the simulation, the graphs of the fixed and mobile Wi-Fi scenarios intercept. This is the point at which the distance of the mobile caller matches that of the fixed caller. Past this point, we can see a rapid increase in packet loss as the Wi-Fi range threshold is surpassed.

Finally, for the traffic load scenario, we look at the affects that link load has on packet loss. Initially, both the loaded and unloaded cases had roughly the same score of 100 packets/sec, but as the background load was increased, the number of packets dropped also increased.

4.4 End-To-End Delay

ETE delay was measured using the voice packet, ETE global statistic. These results are provided below in figures 17 and 18.

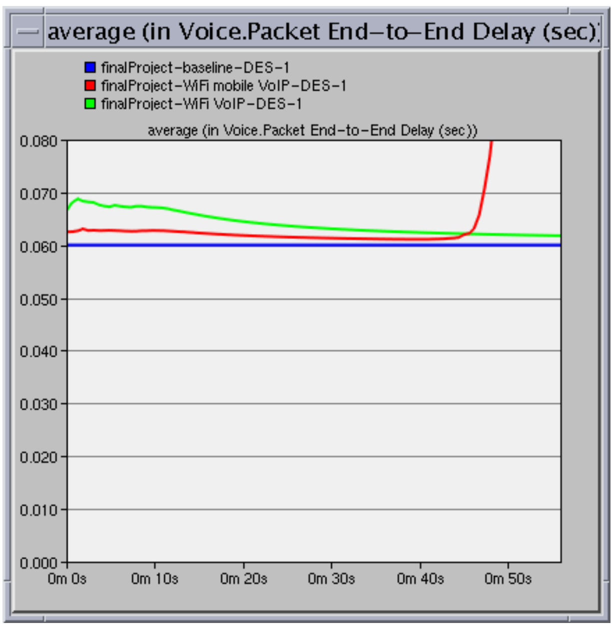


Figure 17: End-to-End delay

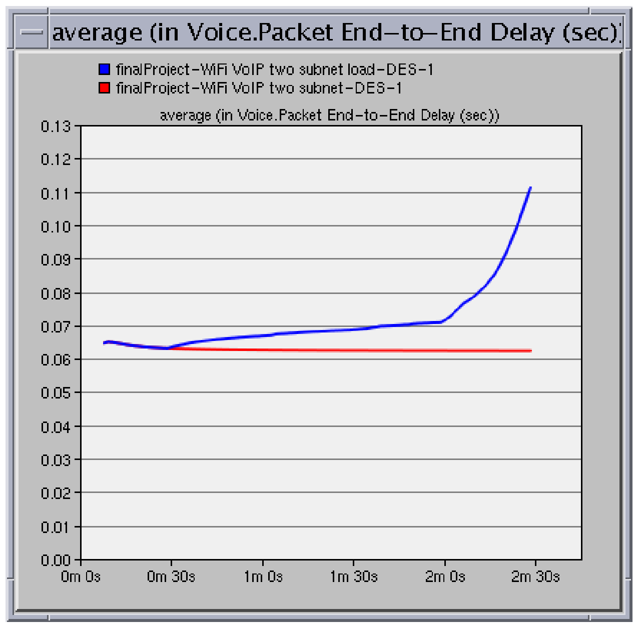


Figure 18: End-to-End two sub networks

Examining the baseline scenario, we notice that we have a constant delay of 0.060 seconds. This value will be considered the ideal value for our comparisons of the other scenarios.

For the fixed Wi-Fi scenario, we have an ETE delay of approximately 0.070 seconds. This is to be expected because the higher packet loss and propagation delay through the air compared to the copper wire of the Ethernet model. As time goes by, this ETE delay decreases to a near ideal value of approximately 0.063 seconds.

Next, we examine the results from the mobile Wi-Fi scenario. Initially this scenario has a slightly lower ETE delay (of approximately 0.063) compared to the fixed Wi-Fi scenario. This is because the caller is initially positioned closer to the router then in the fixed scenario. As the caller moves away from the router, we can see a gradual incline in the ETE delay. At approximately 45 seconds into the simulation, the graphs of the fixed and mobile Wi-Fi scenarios intercept. This is the point at which the distance of the mobile caller matches that of the fixed caller. As the caller exceeds the limit of the wireless network range, the ETE delay increases exponentially.

Now, we look at the link load scenario. Initially, the two cases have the same of ETE delay of approximately 0.065 sec. After 30 seconds (when the first traffic load is applied to the link) we can see that the ETE delay of the no load case remains constant whereas the delay of the case with the load begins to increase. The rate of change of the delay for the loaded case continues to increase after each increment in the load profile. The delay remains at a level of 0.070 seconds which is comparable to the Wi-Fi scenarios. However, we can see from the plot that when we reach the final load profile, the delay exponentially increases.

5. Conclusion

In this project we simulated four different scenarios to measure the performance of VoIP over IEEE 802.11g wireless networks. To determine the quality of a VoIP call we obtained plots for a range of network QoS parameters including packet loss, ETE delay, jitter and MOS.

The plots for the packet ETE delay indicate that introducing both background load and mobility of nodes has the effect of increasing the packet ETE delay as simulation time progressed. The MOS plots show that the overall voice quality for VoIP calls is generally acceptable and can be used in place of mobile phones or even land lines. The jitter was within the acceptable limits and using a Wi-Fi network did not adversely affect jitter compared to the Ethernet network assuming the user was within a reasonable range of the router. Finally, we found that packet loss for the Wi-Fi VoIP calls were comparable to that of the baseline scenario.

This project only touches the surface of VoIP communications. This field of research is rapidly growing every day. Advances in both wireless networks technology as well as sophisticated new voice codecs will allow for an increase in quality and user satisfaction. One example of a newer technology is the 802.11n specification that was recently developed. This standard will allow for a wider range of wireless communication.

6. List of References

[1] L. Cai Yang Xia, X. (Sherman) Shen, L. Cai , J. W. Mark, "VoIP over WLAN: Voice capacity, admission control,QoS, and MAC" , International Journal of Communications , p. 491–508, 2006.

[2] "Wi-Fi", [online], Available; http://en.wikipedia.org/wiki/Wi-Fi. [Accessed: April 1, 2010].

[3] "Packet delay variation," Feb. 17, 2010. [Online]. Available: http://en.wikipedia.org/wiki/Packet_delay_variation. [Accessed: Mar 16, 2010].

[4] "Mean opinion score", [online], Available; http://en.wikipedia.org/wiki/Mean_opinion_score. [Accessed: Mar 06, 2010].

[5] "Packet loss", Feb. 7, 2010. [Online]. Available: http://en.wikipedia.org/wiki/Packet_loss. [Accessed: Mar 14, 2010].

[6] "Digital Signal 1" Mar 15, 2010. [Online]. Available: http://en.wikipedia.org/wiki/Digital_Signal_1. [Accessed: April 14, 2010].

[7] S. Dixit, R. Prasad, "Wireless IP and building the mobile Internet" , Published Boston, MA : Artech House, c2003.

[8] A. Raake, "Speech Quality of VoIP: Assessment and Prediction", 1 ed. New York, NY: Wiley, 2006.

[9] Y. Lin, W. Chen, C. Gan, "Effective VoIP Call Routing in WLAN and Cellular Integration", IEEE Communication Letters, vol. 9, no. 10, October 2005.