Group #4

ENSC 427: COMMUNICATION NETWORKS

Analysis of VoIP Performance over Wi-Fi Networks

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FINAL REPORT

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ABSTRACT

Voice over Internet Protocol (VoIP) is an increasingly popular technology that has been proposed to be an alternative to public switched telephone networks. With Wi-Fi technology, VoIP can become more easily accessible as an increasing number of consumers are accessing Wi-Fi features on their mobile devices. With higher demand, some metropolitan areas have even proposed the idea of implementing City-wide Wi-Fi, which would increase the number of Wi-Fi hot-spots around the city. However, a major concern of providing VoIP over Wi-Fi networks is Quality of Service (QoS). Through simulations of VoIP over Wi-Fi networks in OPNET, this project will analyze the performance of VoIP through multiple case scenarios.

1. INTRODUCTION

2.1. Wi-Fi

Wireless fidelity, more commonly known as Wi-Fi, is a wireless network technology which allows users to access the Internet remotely. Wi-Fi is available in most high traffic areas such as airports, coffee shops and university campuses. Today, there is an ever growing demand for these hotspots. According to AT&T statistics, the number of people using their domestic hotspots has more than tripled from 3.4 million to 10.5 million in Q108 to Q109 respectively [2]. The reason for this sudden increase of demand for Wi-Fi accessibility is partly due to the overload problems in 3G networks. Many municipalities are recognizing this need for more Wi-Fi hotspots and have considered the idea of City-wide Wi-Fi. The concept of City-wide Wi-Fi involves blanketing an entire city with Wi-Fi. This concept was even proposed by the city of Vancouver back in 2007 for the Olympic Games. However, since Wi-Fi is limited in range, in order to upgrade to City-wide Wi-Fi, a city would need to install plenty of Wi-Fi access points. In Vancouver's case, the estimated cost was about ten million dollars [3].

2.2. VoIP

Voice over Internet Protocol (VoIP) allows users to make and receive calls by connecting through the IP network. It has been introduced as an alternative to public switched telephone networks (PSTN). A popular communications means that has spawned off of VoIP popularity is the proprietary protocol: Skype. Skype allows users to talk to other users on the Skype network over an IP network connection, free of cost.

Some of the reasons behind the growing popularity of VoIP are the low costs, the efficient bandwidth, and the flexibility [5]. However, a major concern of this network is the Quality of Service (QoS) of using VoIP over a Wi-Fi network.

2.3. Overview of Wi-Fi and VoIP Technology

Wi-Fi uses the IEEE802.11a, 11b, and 11g standards which provide a large throughput but limited range of service. This enables Wi-Fi to handle large file transfers but limits its service to a small area. To consider City-wide Wi-Fi, multiple access points need to be set up in the area.

The following table outlines some parameters of the Wi-Fi technology.

Table 1: Wi-Fi Specifications [1]

	Wi-Fi
Technology	IEEE802.11a, 11b, 11g
Throughput	4.4-6.6 Mbps/ 12.4-24.7 Mbps
Coverage	Local-Area

The protocols that VoIP uses are: Session Initiation Protocol (SIP) and H.323. The QoS can be observed by voice quality and transit delay. Voice quality degradation can occur from packet loss due to errors or buffer overflows, and dropping late arrival packets in the Real-time Transport Protocol (RTP) jitter buffer. Transit delay is caused by voice coding/compression, distance, and router buffering. There are also many voice codecs that could be used for VoIP; however, for the scope of this project, we will only be using IP telephony which uses the G.729A standard. This standard compresses pulse-code modulation (PCM) to 8 kbps and is generally used in large area IP telephony [4].

2.4. Project Scope

In this project, we will analyze the effects on packet loss, jitter, end-to-end delay and packets received/sent for calling pairs using VoIP over Wi-Fi. We will also simulate scenarios comparing single versus multiple, and fixed versus mobile calling pairs to provide better insight into VoIP QoS.

3. OPNET SIMULATION GUIDELINE

3.1. Overall Design of the Network

The overall design of the project will focus on a small simulated community the size of Simon Fraser University. However, due to limitations in resource and time, our simulations will only involve a maximum of four users within a campus sized network.

The model used for our analysis in OPNET, is the Wi-Fi Local Area Network model and it was modified accordingly for each case scenario. The following table details the parameters modified for the Application and Profile definitions. All other settings were left as default.

Table 2: Application and Profile Configurations

	Application Configuration Attributes
Application Definition	Set to “Default”
	Modify “Voice over IP Call (PCM Quality)”
	Voice: IP Telephony
	Profile Configuration Attributes
Profile Configuration	Set to “Sample Profiles”
	Number of rows: 6
	Profile name: Client
	Application name: Voice over IP Call (PCM Quality)
	Start time(seconds): uniform (0,10)

For the calling pairs and the router, the following attribute settings were made. The settings were modelled after a previous VOIP project’s Caller/Callee set-up [6].

Table 3: Attribute Settings for Caller/Callee and Access Point (Router)

	Caller	Callee (Receiver)
Applications:		
Supported Profiles	Add “Client” profile	None
Supported Service	None	Voice over IP Call (PCM Quality), Supported
	Access Point Attributes	
Wireless Lan Parameters:		
Physical Characteristics	Extended Rate PHY (801.11g)	
Data Rate	54 Mbps	

3.2. Topologies of Simulation Cases

Case 1: Single Pair - Wireless Mobile

Figure 1: Single Pair - Wireless Mobile Nodes

Case 2: Single Pair - Wireless Mobile (10 mW)

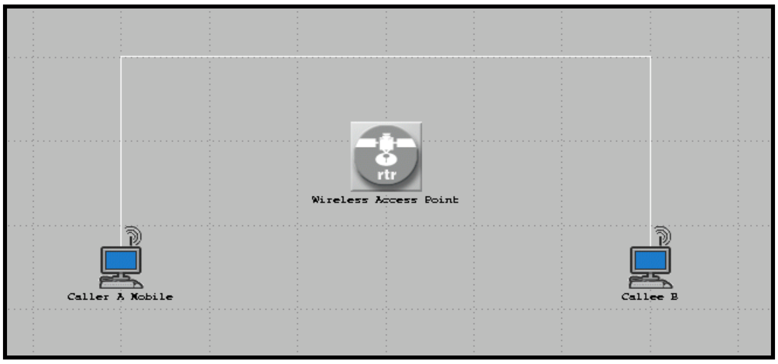

Figure 2: 10mW Power Setting for Mobile Pair

In the first and second scenarios, we envisioned that the network was utilized by one mobile Caller and a stationary Callee shown in Figure 1 and Figure 2. The mobile caller would traverse our predefined trajectory with a velocity of 50 km/hr, which granted us the ability to test maximum speed within a campus environment should the caller be in a vehicle. It was imperative to also remove any myths that the VoIP protocol was susceptible to peer to peer communication when nodes very placed very close together yet far from an access point.

The only difference between the first and second cases was that fact that the power doubled from 5mW to 10mW to see the nominal gains that could be visualized.

Case 3: Single Pair - Wireless Fixed

Figure 3: Single Pair – Wireless Fixed Nodes

Case 4: Single Pair - Wireless Fixed (10mW)

Figure 4: 10mW Power Setting for Fixed Pair

A typical wireless setup was simulated. This was done by having a fixed Caller node and a fixed Callee node. This setup implied that no factors in terms of connectivity would be compromised as time theoretically hits infinity. Similar to our first two setups we tested with the default power settings of 5mW and doubled it for the next trial to 10mW.

Both Figure 3 and Figure 4 demonstrate the scope of our cases. The distance separating the two nodes is 0.725 meters while their radial distance from the Wireless Access Point is approximately 400 meters. Our graphs shown in the subsequent sections will illustrate the exact gains of our investigation.

Case 5: Two Pairs - Wireless Fixed (10mW)

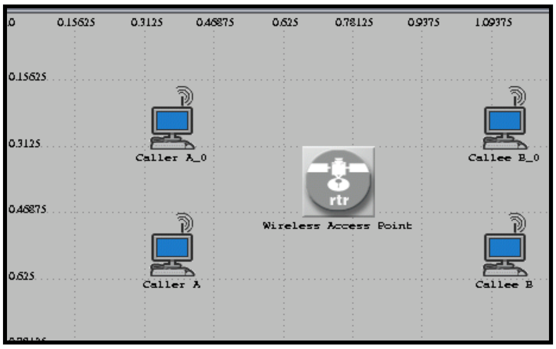

Figure 5: Two Fixed Calling Pairs

To simulate the modularity of our system we decided to increase the node count by a factor of two shown in Figure 5. Due to the changes we have chosen, we should expect double the load volume and higher amounts of delay. Furthermore, the nodes are all placed at a radius of 400 meters from the Wireless Access Point.

Case 6: Two Pairs - Wireless Fixed (Short Range)

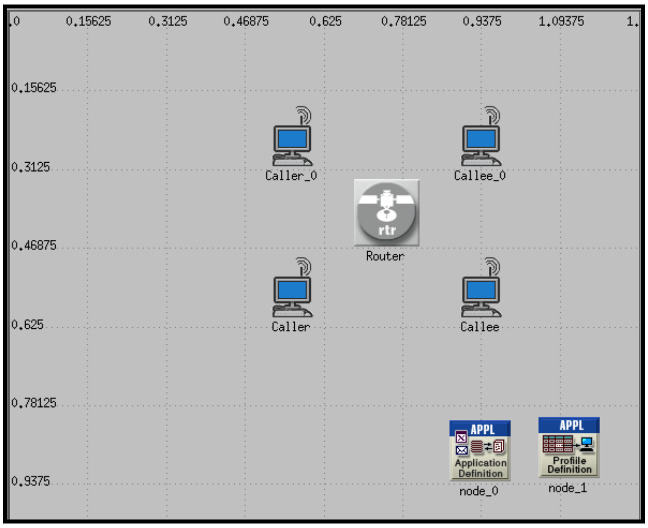

Figure 6: Network for two stationary calling pairs 100m away from Access Point

The above topology is for simulating VoIP using two stationary Callers and Callees over Wi-Fi. Both the Callers and Callees are placed 100 meters away from the router, which is a relatively short range. The simulations in OPNET were set to run for 3 minutes.

Case 7: Two Pairs - Wireless Fixed (Long Range)

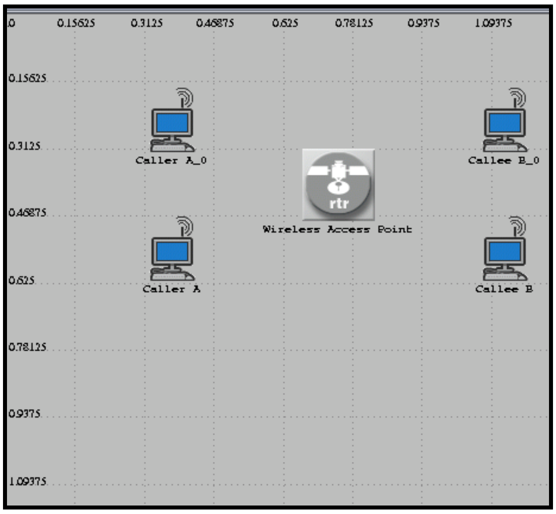

Figure 7: Two Fixed Calling Pairs 400m away from Access Point

This case uses the same set-up as Case 6 except the two stationary pairs are placed further away (400 meters) from the Access Point. This simulation was also set up to run for 3 minutes and we will observe the differences in packet loss and jitter from changes in range.

Case 8: Two Pairs - Wireless Mobile Users

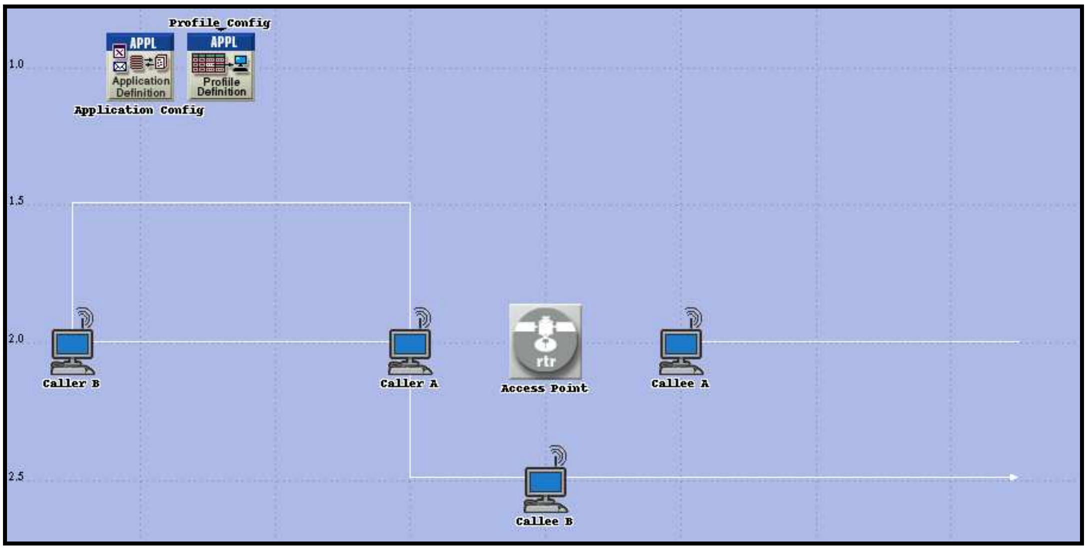

Figure 8: Topology for Two Mobile Calling Pairs

To more realistically analyze VoIP on a Wi-Fi network, we must consider mobility. With VoIP available on a wireless network, users will have the ability to move around within the network.

As shown in figure 8 above, we have two calling pairs, A and B. Pair A was placed initially at a location to give it ideal performance results, only 0.5 km away from the router. The pair was set up to mirror each other's movement at a speed of 60km/hr: starting close to the router, then moving further away, and then returning to their initial position. Note that we increased the maximum speed of the users to limit our simulation time to 5 minutes. The trajectory setting for Caller A is shown in figure 9 below.

Trajectory name: SetA1

	X Pos (km)	Y Pos (km)	Distance (km)	Altitude (km)	Traverse Time	Ground Speed	Wait Time	Accum Time	Pit
1	1.500000	2.000000	n/a	0.000000	n/a	n/a	2m00.00s	2m00.00s	
2	0.250000	2.000000	1.250000	0.000000	1m15.00s	000000	00.00s	3m15.00s	
3	1.500000	2.000000	1.250000	0.000000	1m15.00s	60.00000	00.00s	4m30.00s	

Figure 9: Caller A's Trajectory

Calling pair B, on the other hand, is set up to only have one caller move and the other stationary. This set-up was chosen to analyze the effect of changing distance from a router and from the receiver. The trajectory set-up for Caller B is shown below.

Trajectory name: SetB

	X Pos (km)	Y Pos (km)	Distance (km)	Altitude (km)	Traverse Time	Ground Speed	Wait Time	Accum Time
1	0.000000	0.000000	n/a	0.000000	n/a	n/a	30.00s	30.00s
2	0.000000	-0.500000	0.500000	0.000000	30.00s	000000	00.00s	60.00s
3	1.250000	-0.500000	1.250000	0.000000	1m15.00s	000000	00.00s	2m15.00s
4	1.250000	0.500000	1.000000	0.000000	1m00.00s	000000	00.00s	3m15.00s
5	3.750000	0.500000	2.500000	0.000000	2m15.03s	60	00.00s	5m30.03s

Figure 10: Caller B's Trajectory

We also set an initial wait time for each calling pair to analyze if there is any impact as users start moving. Pair A had an initial 2 minute wait time, while pair B started moving at 30 seconds.

3.2. Simulation Results and Discussion

Case 1: Single Pair - Wireless Mobile

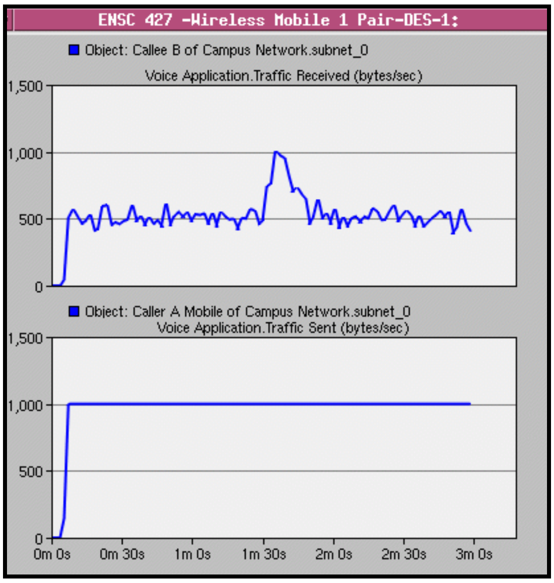

Figure 11: Received and Sent Data From the Nodes

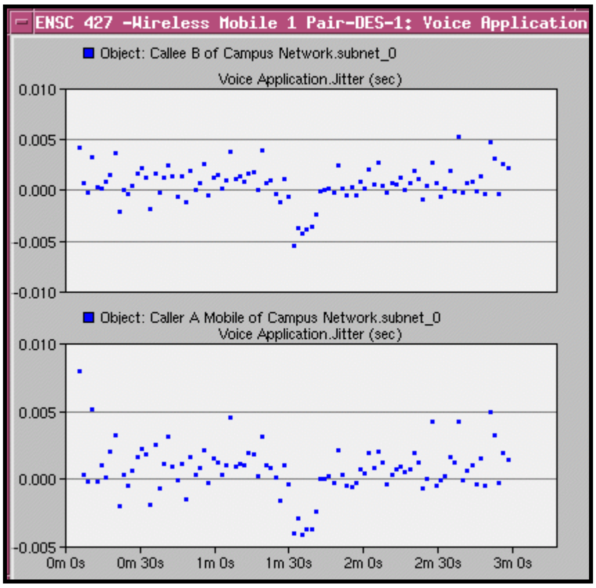

Figure 12: Jitter Visualized From the Nodes

From our simulation results of Case 1 is shown in Figure 11. We can see that as the mobile node traverses the trajectory and gets closer to the access point, the quality of the signal increases until the transmission ratio is 1:1. However, as long as the node maintained a distance of approximately 400 meters, the signal is displaying effectively half of the data that is sent.

From Figure 12, the jitter results are as we expected, because as the signal strength is relatively good, there is minimal jitter.

Case 2: Single Pair - Wireless Mobile (10 mW)

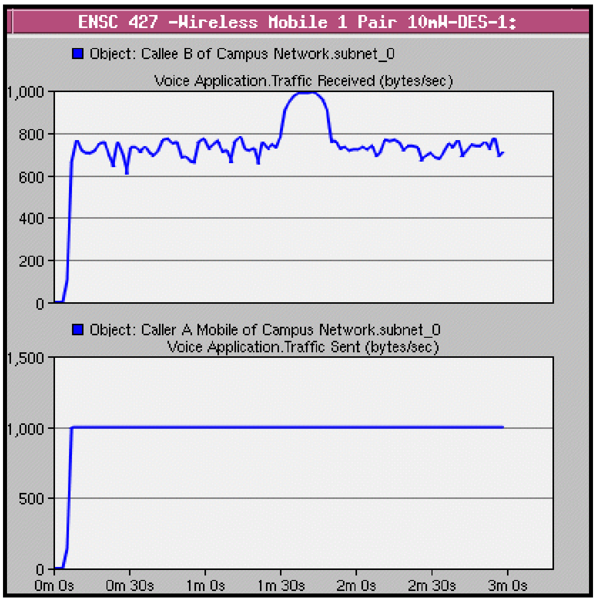

Figure 13: 10mW Mobile Node Data Transmission

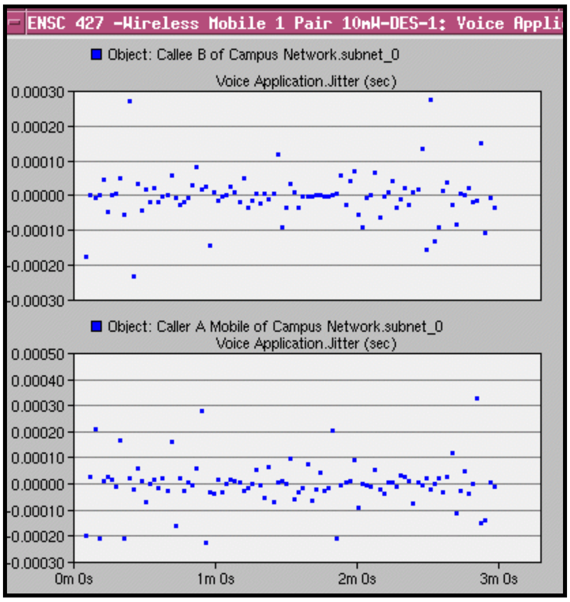

Figure 14: Jitter Results from Both Nodes

Like the previous simulation, we analyzed the effects of how a higher power of 10mW would affect the mobile aspect. To our amazement a gain of 25% (750 bytes/second) was realized and a sustained rate of 1000 bytes/second was shown for approximately 20 seconds. This is shown in Figure 13. Jitter was also reduced significantly for both the sending and receiving nodes as shown in Figure 14. Hence, we can conclude that as we increase the power to levels on par with today's standard of 72mW, greater sustained connectivity levels can be attained.

Case 3: Single Pair - Wireless Fixed

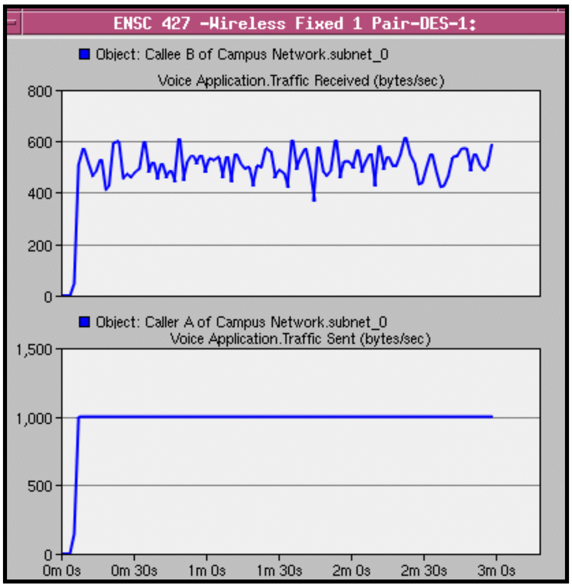

Figure 15: Fixed Node Pair Transmission Rates

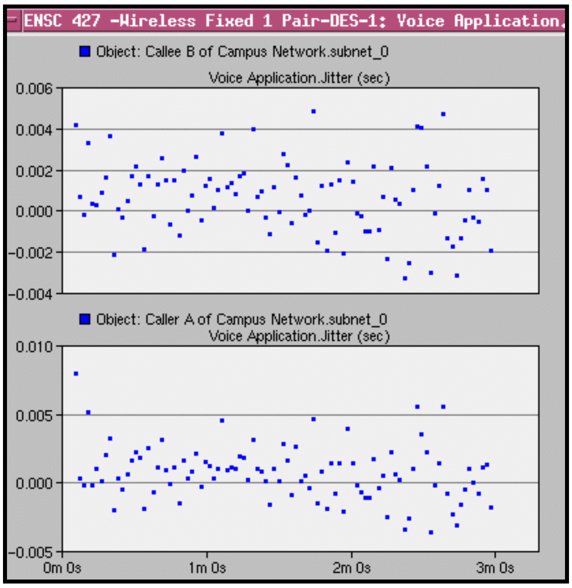

Figure 16: Fixed Pair Jitter

From Figure 15, we observe that fixed nodes within a wireless network will experience the most consistent level of service throughout their connection cycle. Again with a radial distance of 400 meters from the Access Point, the values are very similar to the initial and final values of our mobile scenario. This gave us an average readout of approximately 500 bytes/second. As for the jitter realized, we saw that the jitter was very close to zero with very negligible delay shown in Figure 16.

Case 4: Single Pair - Wireless Fixed (10mW)

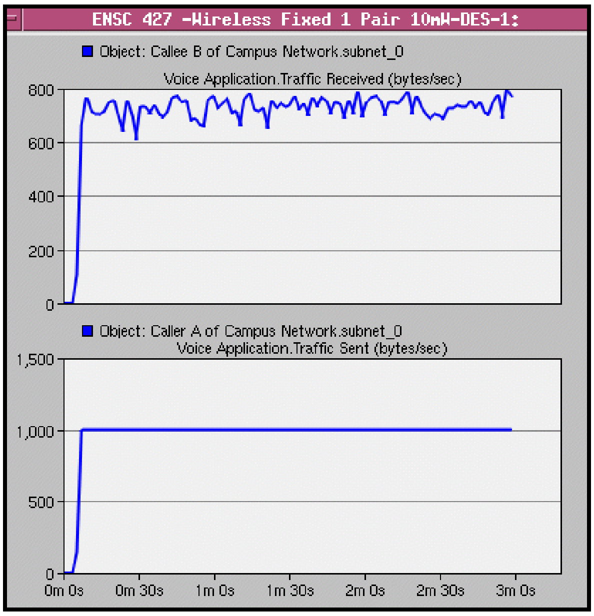

Figure 17: 10mW Fixed Node Transmission Rates

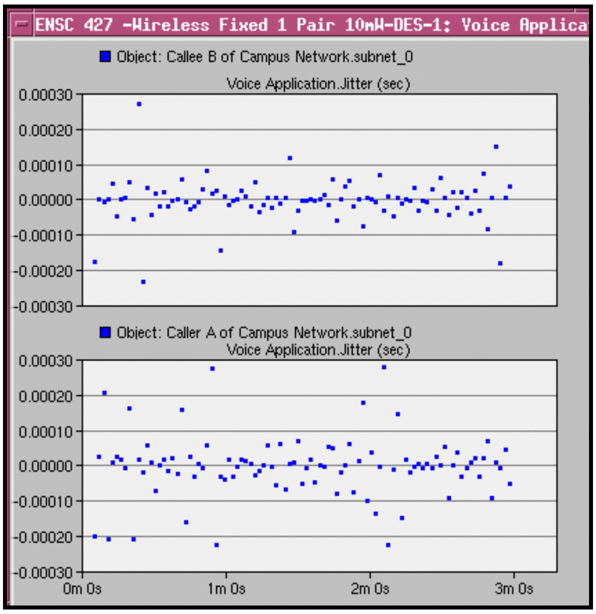

Figure 18: 10mW Fixed Node Jitter Levels

Again our expectations were realized as we did see a higher transmission ratio for the nodes. In Figure 17, an average of 750 bytes/second was seen, granting us results consistent to that of the initial and final values of the mobile nodes. With higher transmission power from the Access Point, the jitter levels also decreased significantly with smaller values. This observation is shown in Figure 18.

Case 5: Two Pairs - Wireless Fixed (10mW)

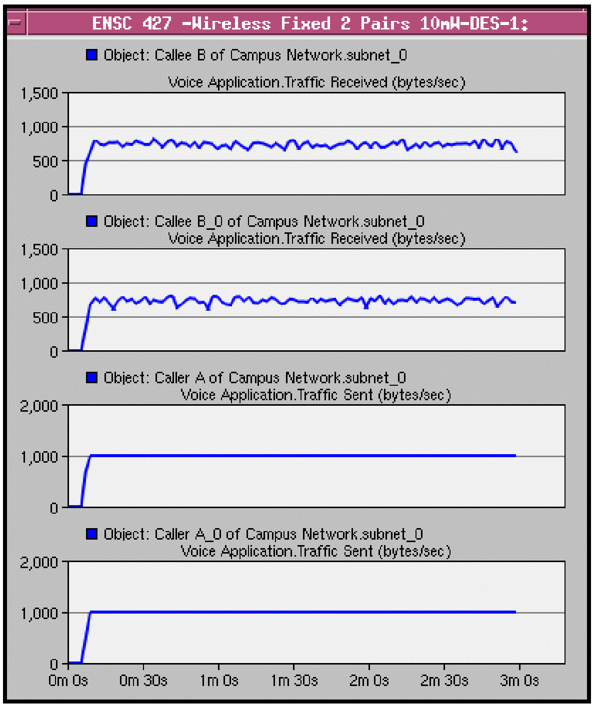

Figure 19: Transmission Rates for Two Fixed Pairs

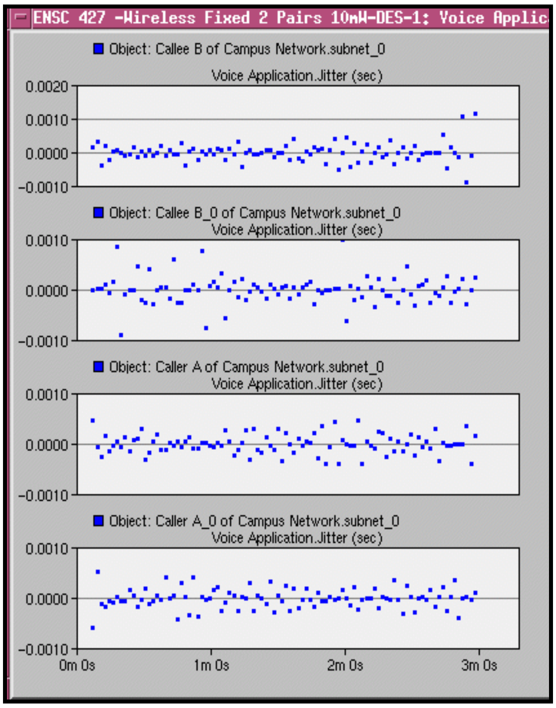

Figure 20: Jitter Readouts for the Calling Pairs

Utilizing the same settings as the prior scenario, two extra nodes were added. This scenario was chosen to ensure that the wireless network would be adaptable as the client base increases. In essence our results correlated with the single fixed pair nodes with a power setting of 10mW. Figure 19, illustrates this point explicitly as the average receiving rate to be 750 bytes/second. Figure 20 also shows that jitter remains quite low despite the doubling of the client base.

Case 6: Two Pairs - Wireless Fixed (Short Range)

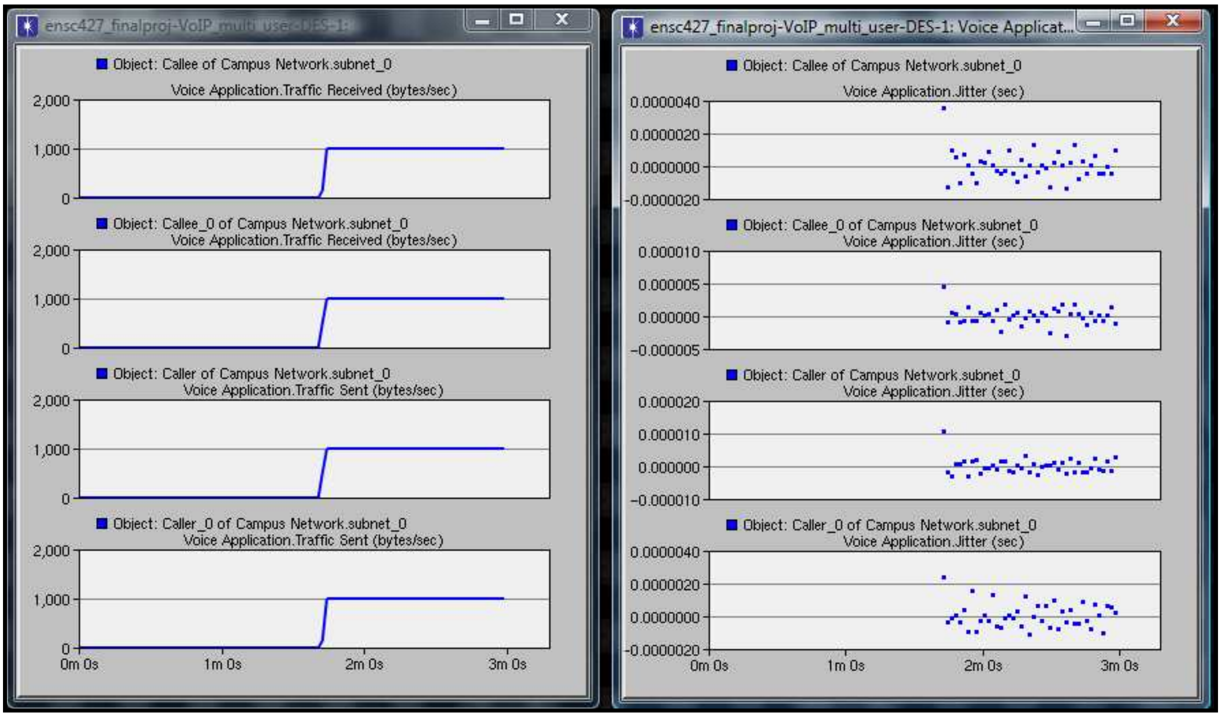

Figure 21: Caller and Callee Traffic Sent/Received (left) and Jitter (right)

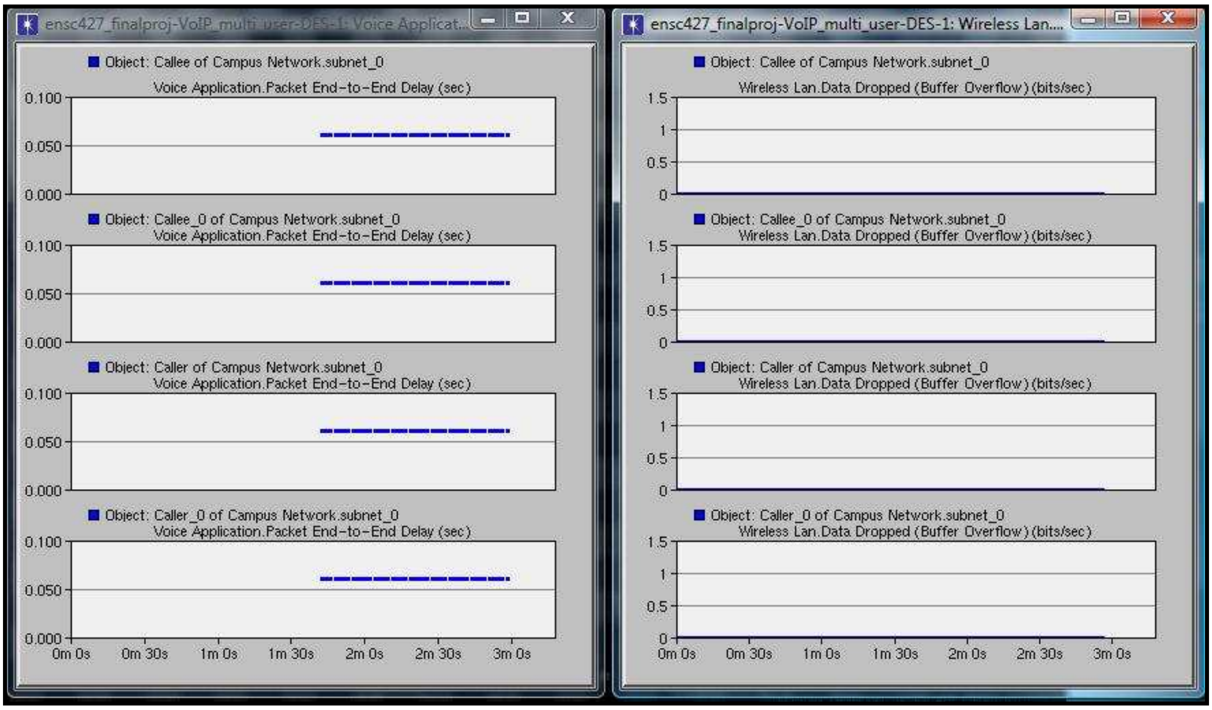

Figure 22: Caller and Callee ETE Delay (left) and Data Dropped (right)

As observed in Figure 21, the traffic sent by both Callers was received by both Callees with no packet loss. The Jitter observed was also reasonable. Moreover, as expected, the packet end-to-end delay is constant at about 0.060 sec and there was no data dropped as shown in Figure 22. Therefore, the Callers and Callees were placed close enough to the Access Point such that no data loss is observed.

Case 7: Two Pairs - Wireless Fixed (Long Range)

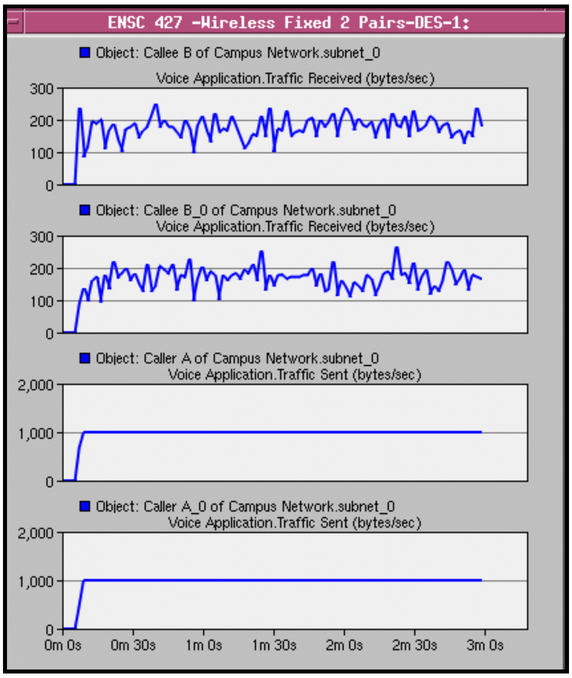

Figure 23: Caller and Callee Traffic Sent/Received

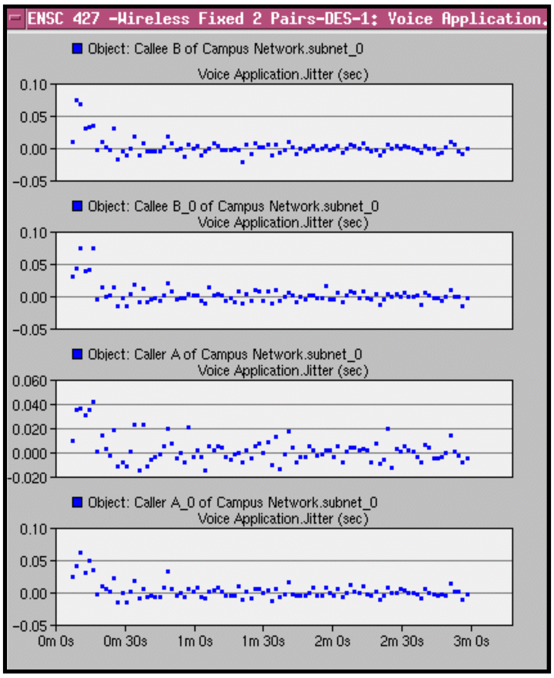

Figure 24: Caller and Callee Jitter

When increasing the range of the Callers and Callees from the Access Point we noticed a large amount of packet loss. As observed above in Figure 23, there was about 880 bytes/sec of packet loss. We expected some packet loss as we increased the range of the Caller and Callees from the Access Point because distance affects the quality of service. As distance increases, packet loss is more frequent. Figure 24, shows a greater variable range of jitter.

Case 8: Two Pairs - Wireless Mobile Users

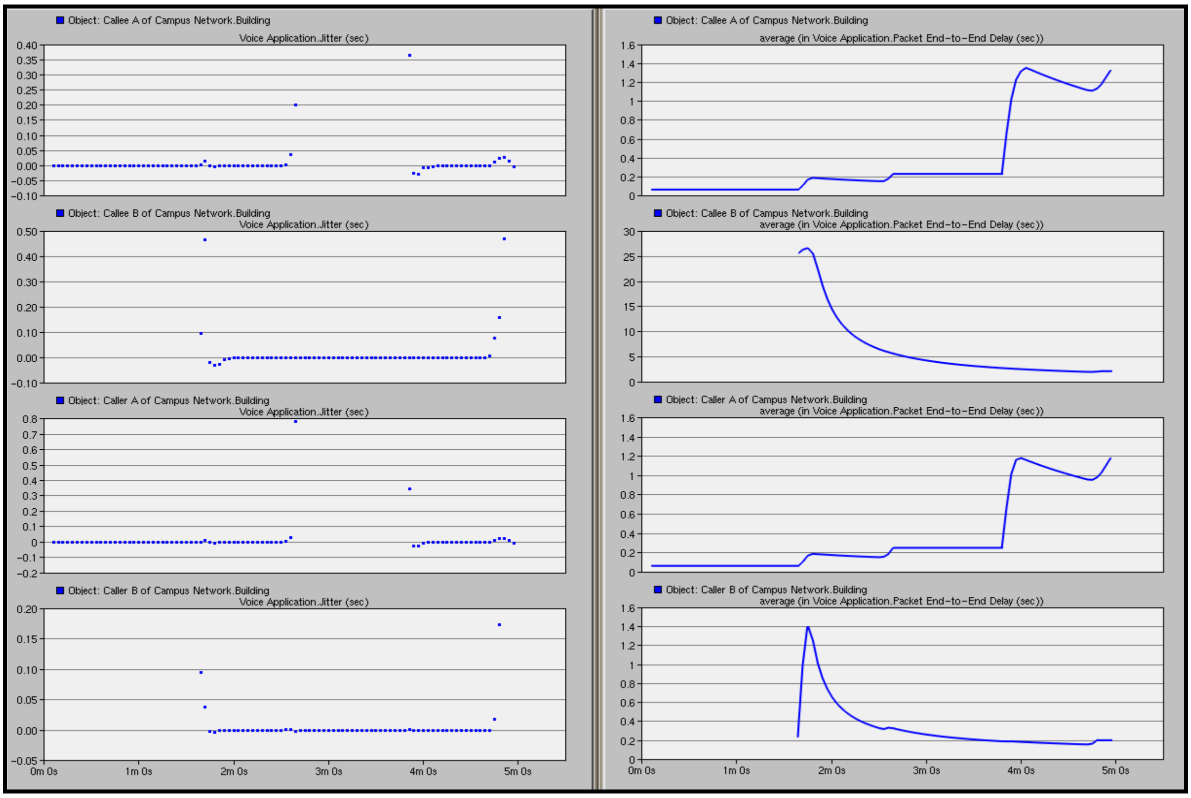

Figure 25: Jitter (left) and Averaged End-to-End Delay (right) for Mobile Pairs

Figure 25, shows the results for jitter for each Calling pair. Notice that jitter exists in VoIP calling when the pairs are within the network range. Jitter also increases when Callers/Callees move into and out of the network range. The maximum jitter experienced is just below 0.8 seconds, which exceeds the acceptable amount. However, this only occurs as calling pairs move out of the network range, in which a call would disconnect anyways. The affect on other calling pairs, as a pair moves out of range is below 25ms which does not exceed an acceptable jitter amount. Distance greatly affects the end-to-end delay in a Wi-Fi network. We can see how end-to-end delay corresponds to jitter as shown above (right).

The following figures show the results for data traffic sent and received. All pairs starting sending data at around 10 seconds. Remember that Pair B starts moving at 30 seconds and Pair A starts moving at 2 minutes. Notice again that once one pair moves in or out of the network range, data for the other calling pair is buffered, resulting in a loss of data followed by a spike in data.

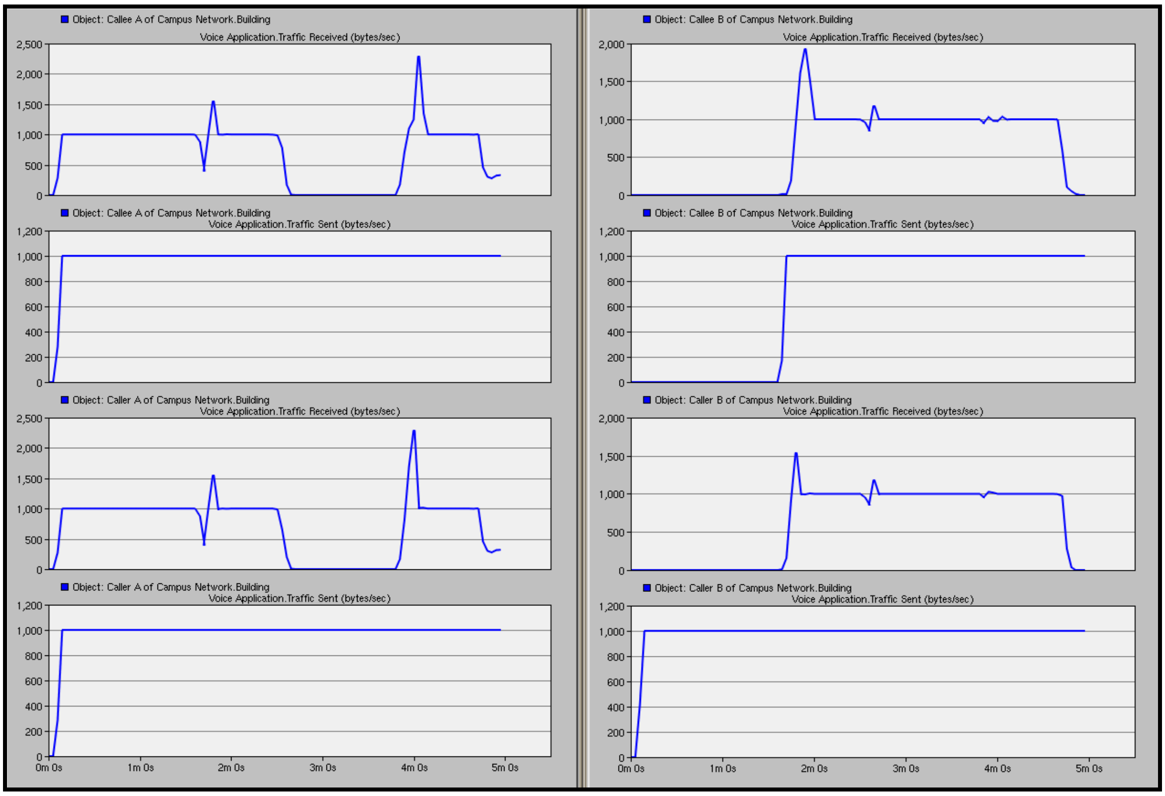

Figure 26: Pair A (left) and Pair B (right) – Data Sent and Received

Finally, the corresponding data loss is shown in Figure 27 below. The loss corresponds to the trajectory of the calling pairs as we would expect: loss is shown when the pairs move out of range.

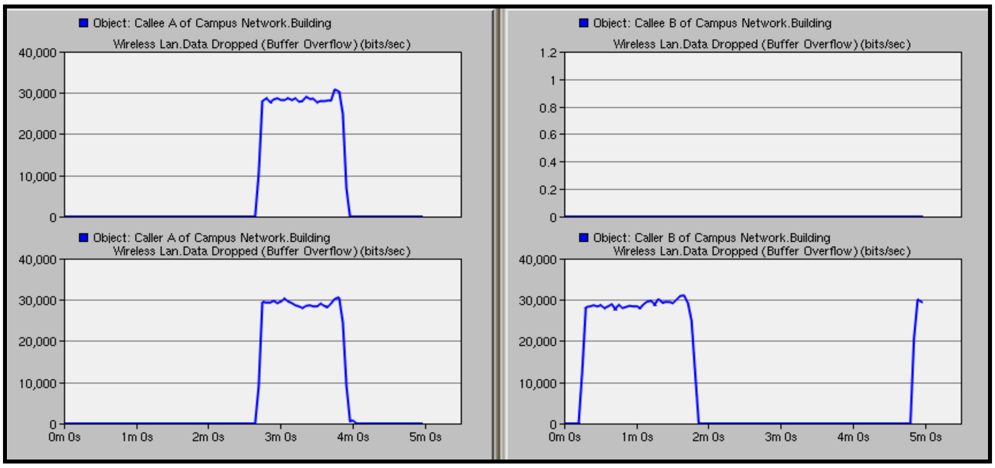

Figure 27: Pair A (left) and Pair B (right) – Data Dropped (bits/sec)

4. CONCLUSION

Our analysis on utilizing VoIP on a wireless network contained many positive aspects when placing such a system into a campus environment. One major aspect behind a network is the costs associated with bringing in a new technology. With VoIP operating on the 802.11g/n standard, it is easy to piggyback on an existing wireless network or wired network. Furthermore, VoIP also offers a variety of functions such as video conferencing which traditional phone lines lack.

In a fixed setting, VoIP on a wireless network performed admirably with very little loss. A few concerns that we did have were that physical walls and the newer 802.11n protocol were not declarable in OPNET. Those problems would make the actual planning of new networks slightly more daunting as real world factors could not be introduced.

In a mobile setting, VoIP on a wireless network signal quality changed depending on the line of sight distance the client was to the access point. Logic dictates that as we moved farther away from the access point, loss would occur causing the QoS to decline steeply. Furthermore, venturing out of the acceptable range of a router and into a new cell would not result in a seamless transition as we see in cellular phones on a GPRS or 3G system.

Despite some shortcomings, factors in the access point can be adjusted to ensure high signal strength such as selecting the protocol used (802.11e/g) as well as raising the actual transmitter power. The handover issue is still a main concern and is currently under research to allow for seamless transitions between networks to networks. As of this moment only "idle" nodes have transitioned successfully making it a hot research topic.

Our findings pointed to the fact that VoIP is a cheaper, more efficient alternative that has a bright future when applied wirelessly. With wireless transmissions hitting up to 300 mbps on the newer 802.11n standard we can see that a high quality audio stream can be carried without worry of oversaturation to the overall network. Lastly, as technology improves in the wireless area we can expect to see issues, such as handover, solved.

5. REFERENCES

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