ENSC 427: COMMUNICATION NETWORKS

# Performance Evaluation of Voice Over IP on WiMAX and Wi-Fi Based Networks

Spring 2009

FINAL PROJECT

Group #1
Jeremy Yoo
http://www.sfu.ca/~jty/ensc427/
jty@sfu.ca

# Table of Contents

# List of Figures

# List of Acronyms

| | |
|---|---|
| ETE: | End-to-End |
| MAC: | Medium Access Control |
| POTS: | Plain Old Telephone Service |
| QAM: | Quadrature Amplitude Modulation |
| QoS: | Quality of Service |
| QPSK: | Quadrature Phase Shift Keying |
| VoIP: | Voice Over IP |
| WiMAX: | Worldwide Interoperability for Microwave Access |

# 1. Abstract

In recent years, voice over IP (VoIP) has become an increasingly popular alternative to traditional landline telephone service, allowing users to communicate worldwide with little added cost from their existing Internet connection. When combined with the mobility afforded by long-range wireless networks such as WiMAX, VoIP has the future potential to replace cellular based communication. Unfortunately, network issues such as packet loss and delay affect VoIP call quality, and these issues are more prevalent in wireless networks. This project will investigate the performance of VoIP over WiMAX and Wi-Fi based wireless networks, compared to a baseline wired Ethernet network, to determine how the user calling experience is affected and whether mobile VoIP communication is feasible.

# 2. Introduction

## 2.1 Background and Motivation

Cellular phones have been around for many years, yet the cost of service is still high and voice quality is inferior to plain old telephone service (POTS). The voice over IP (VoIP) protocol enables voice calls of POTS quality to be transmitted over a normal Internet connection.

The availability of wireless Internet is rising with the continued deployment of Worldwide Interoperability for Microwave Access (WiMAX) and Wi-Fi technologies. WiMAX features long range coverage while Wi-Fi technology is well suited for short-distance networks.

This widespread availability means there is potential for portable VoIP devices that operate on these standards to become a popular alternative to cellular phones. However, physical phenomena of the wireless transmission medium can have an effect on how applications running over these networks perform. The viability of the idea lies with whether the technology is able to handle the strict requirements of a real-time transmission such as VoIP.

## 2.2 Project Idea and Goals

The main goal of the project will be to determine whether wireless Internet telephony is a viable replacement for cellular phones.

When dealing with voice traffic, we are strongly concerned about delay and jitter in transmission. A large delay will result in undesired pauses in the conversation, and jitter will degrade the voice quality of the call. In addition, the real-time nature of a telephone call means that dropped packets cannot be retransmitted at a later time. Depending on the encoding scheme, a maximum delay of 150 ms [1] and up to 10% packet loss [2] can be tolerated.

To collect these statistics, I will simulate VoIP calls being made between users connected via WiMAX and Wi-Fi. An Ethernet network will also be created to have a point of comparison for the wireless networks.

If we observe that the amount of increase in delay, jitter, and packet loss in the wireless networks over the Ethernet network is within acceptable tolerances, it will indicate the strong potential of using these technologies as an alternative for cellular networks.

# 3. Simulation Design

OPNET Modeler 14.0 was used to simulate VoIP calls made by users on Ethernet, Wi-Fi and WiMAX networks. Each network has two conversation pairs, where a pair consists of a *caller* node placing a call to a corresponding *callee* node. All nodes are connected via a transmitting base station. To observe the effects of distance from the base station, one of the pairs are placed close to the base station (named Caller A/Callee A), while the other is placed further away (named Caller B/Callee B). A separate scenario was created for each network. All of the nodes are the built-in OPNET models that were configured as needed.

## 3.1 Ethernet Scenario

The network topology of the Ethernet scenario is shown below in Figure 1.

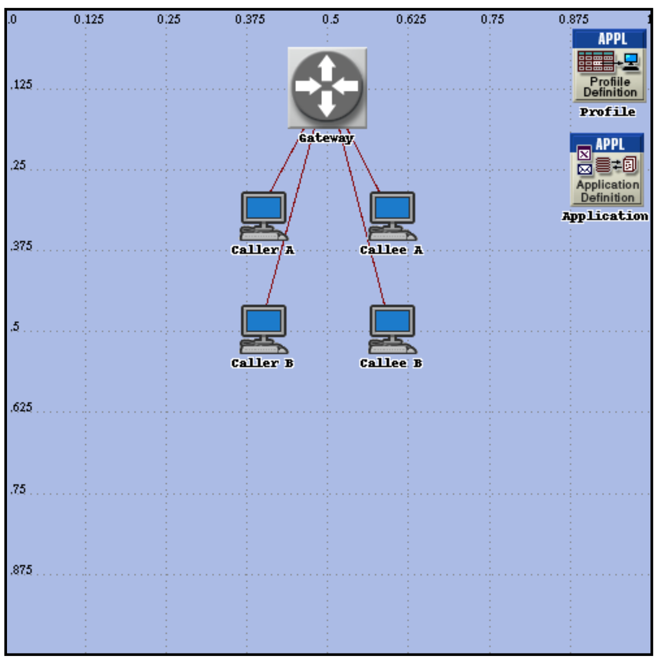


**Figure 1: Ethernet Scenario Topology**

The Ethernet network was created to collect baseline statistics for comparing the Wi-Fi and WiMAX networks. As Ethernet uses wired connections, the base station is instead analogously a wired Ethernet gateway.

The Ethernet workstation model was used for the four users. Caller A and Callee A were placed 223.6 m (x = 100 m, y = 200 m) from the base station, and Caller B and Callee B were placed 388.1 m (x = 100 m, y = 375 m) from the base station. These distances were chosen based on approximating a reasonable upper end for the range of Wi-Fi technology from the simulation results in [3].

## 3.2 Wi-Fi Scenario

As shown in Figure 2, the Wi-Fi network replaces the Ethernet workstations with Wi-Fi workstations and the gateway with a Wi-Fi router. None of the placement locations are changed.

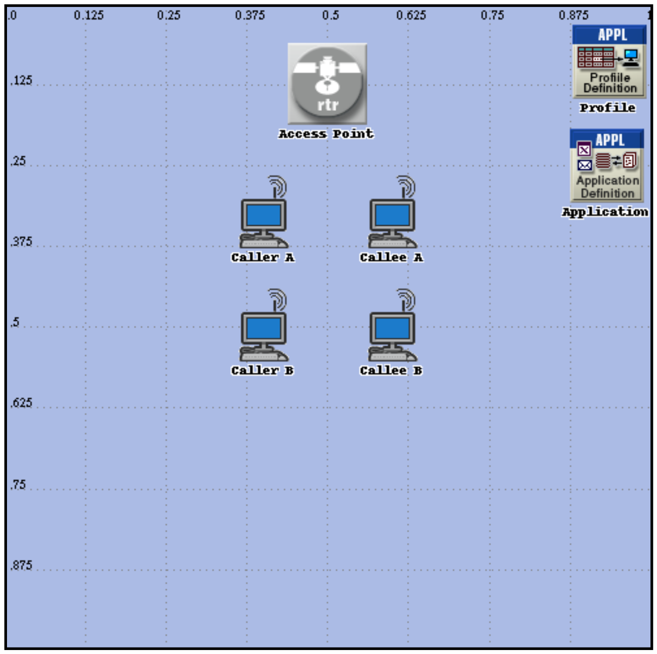


**Figure 2: Wi-Fi Scenario Topology**

By default, the Wi-Fi models use the 802.11b standard at 11 Mbps. The nodes and base station were instead configured to support the newer high-speed 802.11g standard at 54 Mbps, as shown in Figure 3.

| | |
|---|---|
| ⊟ Wireless LAN | |
| Wireless LAN MAC Address | Auto Assigned |
| ⊟ Wireless LAN Parameters | (...) |
| BSS Identifier | Auto Assigned |
| Access Point Functionality | Disabled |
| Physical Characteristics | Extended Rate PHY (802.11g) |
| Data Rate (bps) | 54 Mbps |
| ⊞ Channel Settings | Auto Assigned |
| Transmit Power (W) | 0.005 |

**Figure 3: Wi-Fi Parameters**

All other parameters were kept with default values, including the transmit power of 5 mW.

## 3.3 WiMAX Scenario

The WiMAX network is also based off the Ethernet network. As shown in Figure 4, the Ethernet workstations have been replaced with WiMAX workstations and the gateway is now a WiMAX base station router.

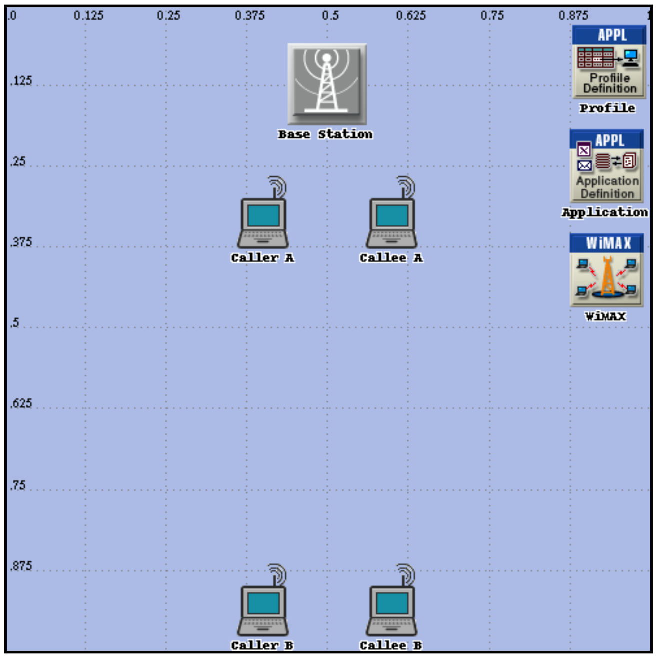


**Figure 4: WiMAX Scenario Topology**

A notable change to the topology is that the Caller B/Callee B pair is brought further away from the base station in order to observe performance with the increased range offered by the WiMAX technology. The nodes are now 806.2 m (x = 100 m, y = 800 m) away from the base station.

WiMAX models in OPNET require an additional configuration node to specify WiMAX parameters. The key parameters are the Efficiency Mode and MAC Service Class Definitions.

It is important that the Efficiency Mode is set to Physical Layer Enabled in order to observe any effects of distance. This was a source of confusion during initial simulations as it is set to Efficiency Enabled by default, resulting in zero packet loss.

The MAC Service Class Definitions allow for configuring the WiMAX quality of service (QoS) control, which allows priority of packets based on the type of packet being transmitted. While QoS is an important part of VoIP due to its real time nature and has been shown to be helpful in WiMAX [4], we are only modeling one type of traffic so priority will not have a role in the performance of the system. The pre-defined Gold/Silver/Bronze model was used and the voice traffic was set as Silver (real time polling service). The configuration is shown in Figure 5.

| | |
|---|---|
| ⊞ Contention Parameters | (...) |
| Efficiency Mode | Physical Layer Enabled |
| ⊞ MAC Service Class Definitions | Gold/Silver/Bronze |
| ⊞ OFDM PHY Profiles | (...) |
| ⊞ SC PHY Profiles | (...) |

**Figure 5: WiMAX Configuration**

The transmit power, antenna gain, and modulation scheme are the key parameters that were manually changed for the base station and workstations. Typical values for transmission power are 10 W for base stations [5] and 0.5 W for fixed users [6]. To keep the simulation realistic, we set our transmission power to these values. The antenna gain of the base station and workstations are set at 15 dBi and 14 dBi, respectively. Figure 6 below shows the WiMAX configuration for the base station and workstations.

| | |
|---|---|
| ⊟ WiMAX Parameters | |
| Antenna Gain (dBi) | 15 dBi |
| ⊞ BS Parameters | Default |
| ⊟ Classifier Definitions | (...) |
| Number of Rows | 1 |
| ⊟ Row 0 | |
| Type of SAP | IP |
| ⊟ Traffic Characteristics | (...) |
| Match Property | IP ToS |
| Match Condition | Equals |
| Match Value | Any |
| Service Class Name | Silver |
| MAC Address | Auto Assigned |
| Maximum Transmission Power (W) | 10 |
| PHY Profile | WirelessOFDMA 20 MHz |
| PHY Profile Type | OFDM |
| PermBase | 0 |

| | |
|---|---|
| ⊟ WiMAX Parameters | |
| Antenna Gain (dBi) | 14 dBi |
| ⊟ Classifier Definitions | (...) |
| Number of Rows | 1 |
| ⊟ Row 0 | |
| Type of SAP | IP |
| ⊟ Traffic Characteristics | (...) |
| Match Property | IP ToS |
| Match Condition | Equals |
| Match Value | Any |
| Service Class Name | Silver |
| MAC Address | Auto Assigned |
| Maximum Transmission Power (W) | 0.5 |
| PHY Profile | WirelessOFDMA 20 MHz |
| PHY Profile Type | OFDM |
| ⊞ SS Parameters | (...) |

**Figure 6: WiMAX Base Station and Workstation Parameters**

WiMAX allows different modulation schemes to be used for the downlink (base station to user) and uplink (user to base station) signals. As the transmit power of the base station is high, the default 64-QAM (with 3/4 coding rate) modulation was kept for the downlink. However, based off initial test runs, it was observed that too many uplink packets were dropped when users were set at 0.5 W transmit power. The uplink signal modulation was changed from the default 16-QAM (with 3/4 coding rate) to QPSK (with 3/4 coding rate), which is a more robust scheme for low SNR transmission [7]. These settings are shown below in Figure 7.

(Downlink Service Flows) Table

| | Service Class Name | Initial Modulation | Initial Coding Rate | Average SDU Size (bytes) | Activity Idle Timer (seconds) | Buffer Size (bytes) | ARQ Parameters | PDU Dropping Probability | CRC Overhead |
|---|---|---|---|---|---|---|---|---|---|
| 0 | Silver | 64-QAM | 3/4 | 1500 | 60 | 64 KB | Disabled | Disabled | Disabled |

1 Rows Delete Insert Duplicate Move Up Move Down
Details Promote ☑ Show row labels OK Cancel

(Uplink Service Flows) Table

| | Service Class Name | Initial Modulation | Initial Coding Rate | Average SDU Size (bytes) | Activity Idle Timer (seconds) | Buffer Size (bytes) | ARQ Parameters | PDU Dropping Probability | CRC Overhead |
|---|---|---|---|---|---|---|---|---|---|
| 0 | Silver | QPSK | 3/4 | 1500 | 60 | 64 KB | Disabled | Disabled | Disabled |

1 Rows Delete Insert Duplicate Move Up Move Down
Details Promote ☑ Show row labels OK Cancel

**Figure 7: WiMAX Modulation Settings**

## 3.4 Traffic Model

For all three scenarios, the Voice Application set as IP Telephony was used to generate the traffic data. The IP Telephony configuration uses the G.729a encoder scheme, which transmits a 10 byte packet every 10 ms (i.e. 100 packets/sec). Traffic begins immediately at the start of the simulation and continues until the end of the simulation.

As shown below in Figure 8, the caller nodes are set to support the profile configured with this traffic model, and the callee nodes are set to support receiving voice traffic. The callee node responds to the caller, simulating a two way voice call (i.e. the caller node will both send and receive 100 voice packets/sec).

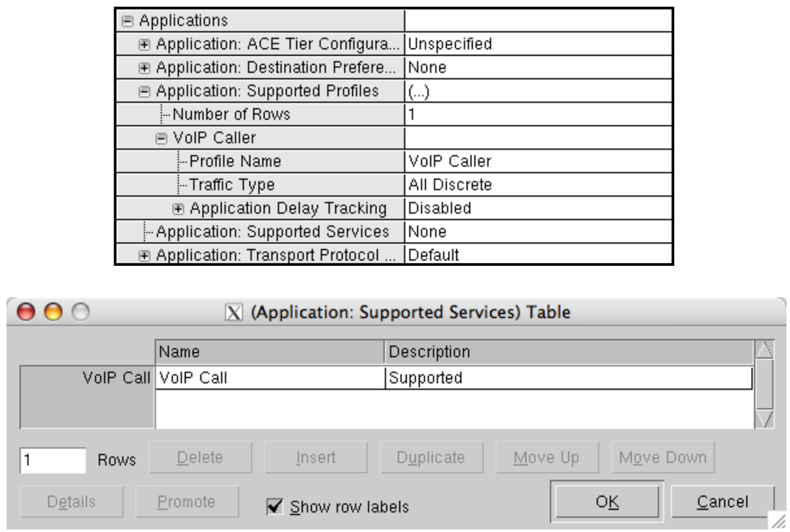


**Figure 8: Caller/Callee Traffic Settings**

Destination Preferences is one of the parameters available to the caller nodes, so that the traffic can be directed to a particular callee node. I initially configured the preferences such that Caller A had a preference for Callee B and Caller B had a preference for Callee B. However, it was found during testing that these preferences were not followed and OPNET appeared to pick the destination randomly.

As I observed that the same destination was always chosen for subsequent simulations, the workaround to ensuring proper pairing was to swap and rename the callee nodes if necessary after running the first simulation on a scenario.

# 3. Simulation Results

The simulation duration for all three scenarios was 30 seconds. We focus on three key statistics: packet loss, end-to-end delay, and jitter.

## 3.1 Packet Loss

Figures 9 and 10, respectively, show the packets received by caller A and caller B for the three scenarios. If no packets are dropped, we expect to see 100 packets received per second.

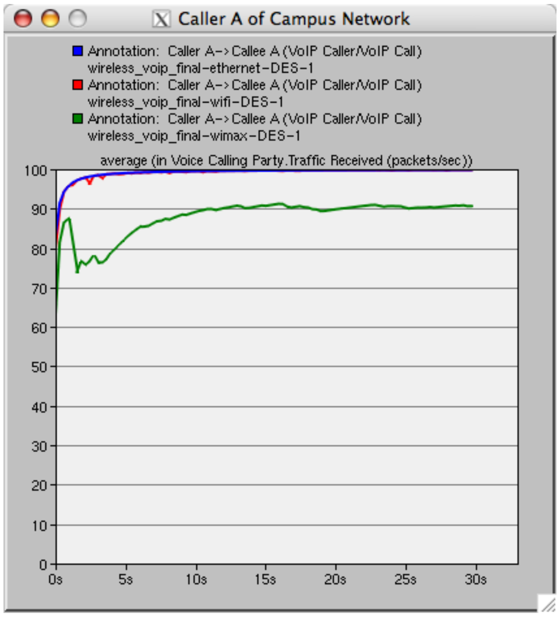


**Figure 9: Comparing Packet Loss of Caller A**

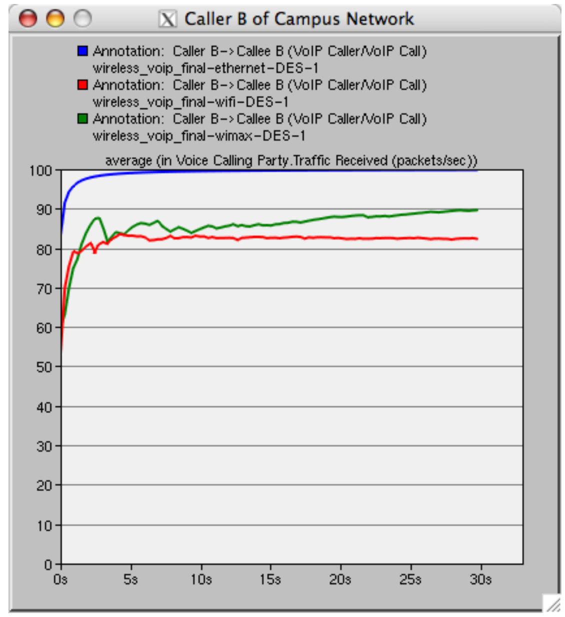


**Figure 10: Comparing Packet Loss of Caller B**

As expected, there is no loss for both callers from the Ethernet network regardless of distance.

The Wi-Fi network performs very well when the callers are close to the base station, matching Ethernet performance with only a slight initial packet loss observed. However, a significant packet loss occurs when the callers are further away. The almost 20% loss is below the acceptable threshold for VoIP. The amount of data being dropped by Caller A in Figure 11 (note that OPNET provides this statistic in bits/sec) shows that the distance is too far for a successful continuous transmission.

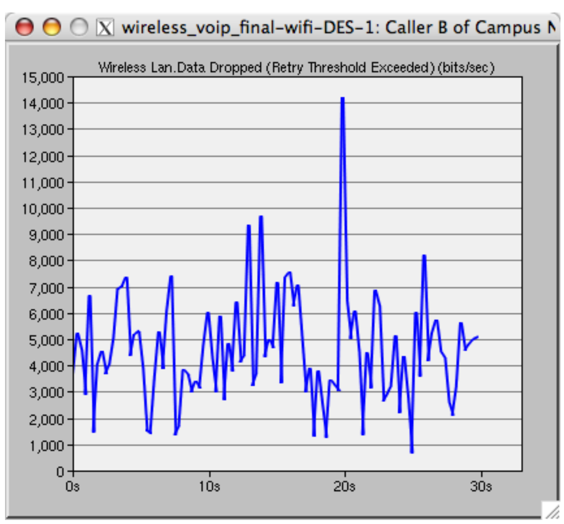


**Figure 11: Dropped Wi-Fi Data of Caller B**

Looking at WiMAX, the packet loss of both callers average out to within acceptable levels, at about 10% loss. Caller B in the WiMAX network fared very well at more than double the distance away from the base station of the Wi-Fi caller. However, it was surprising to observe Caller A performing more poorly than its Wi-Fi counterpart, especially with the larger transmitting power used in WiMAX. From the physical layer statistics, Figure 12 shows more packet loss occurring in the uplink direction for Caller A, even with the robust modulation scheme.

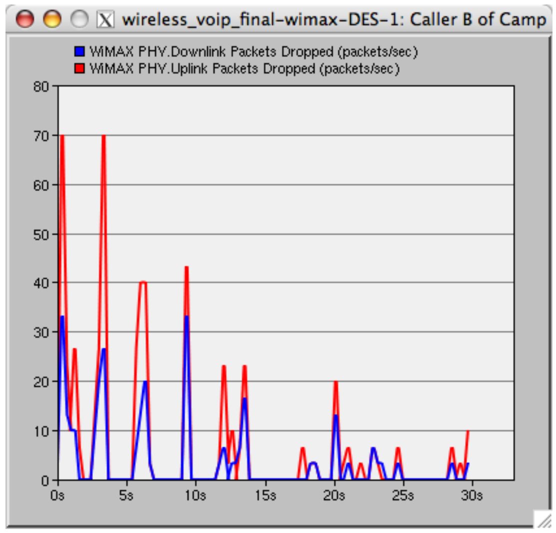


**Figure 12: Observing Packet Loss in WiMAX Uplink and Downlink of Caller A**

## 3.2 End-To-End Delay

The end-to-end (ETE) delay for Callers A and B are shown in Figures 13 and 14, respectively.

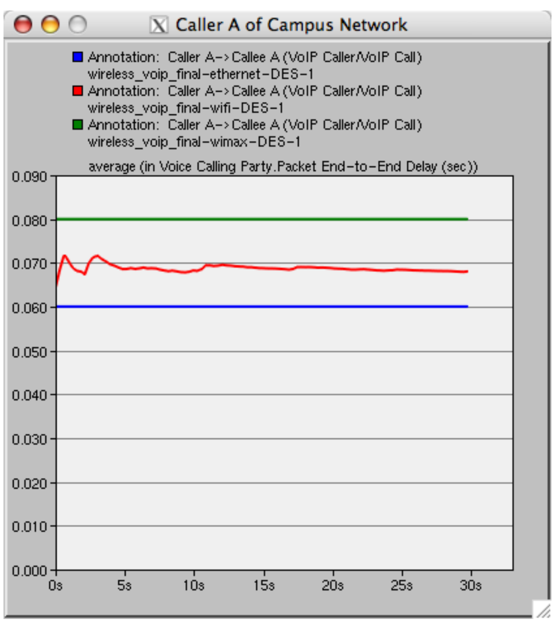


**Figure 13: Comparing ETE Delay of Caller A**

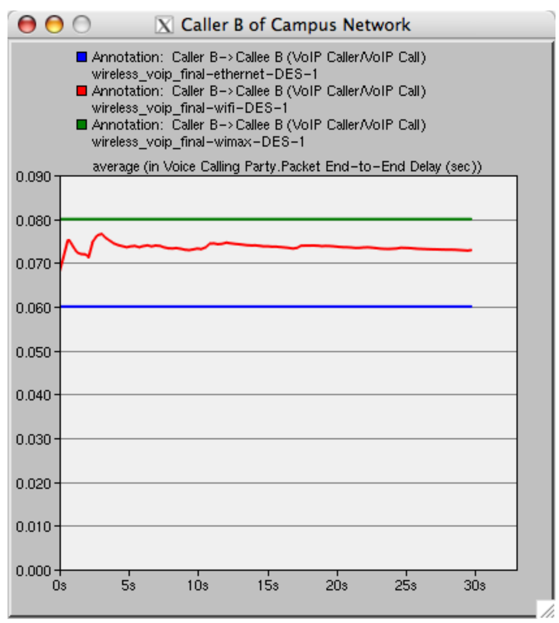


**Figure 14: Comparing ETE Delay of Caller B**

All three technologies show results well under the maximum acceptable delay with Ethernet at 60 ms, WiMAX at 80 ms, and Wi-Fi around 70 ms. It is interesting to observe that distance plays a role in the

ETE delay for the Wi-Fi network, increasing slightly for the further node, but has no effect with WiMAX. The varying ETE delay seen for the Wi-Fi network is indicative of jitter.

### 3.3 Jitter

Figures 15 and 16 show the jitter for Caller A and B, respectively.

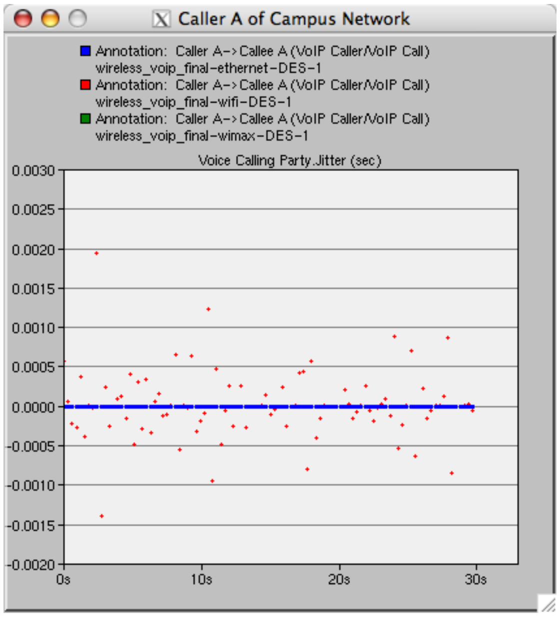


**Figure 15: Comparing Jitter of Caller A**

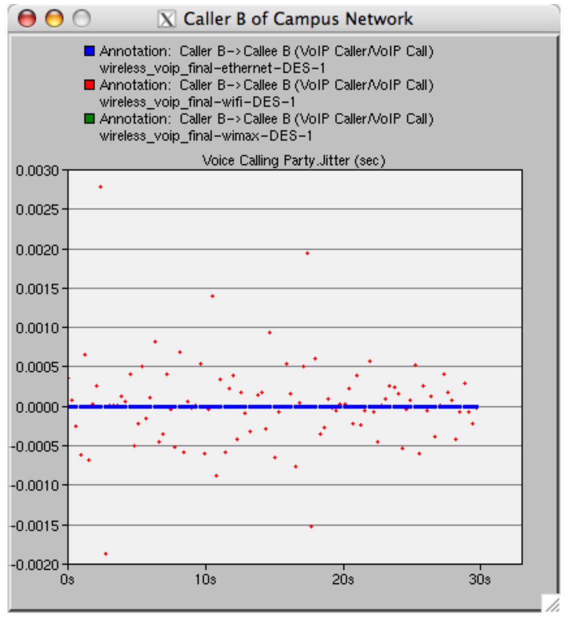


**Figure 16: Comparing Jitter of Caller B**

We observe that jitter is indeed present for the Wi-Fi callers, increasing with distance. However, as the graph shows a maximum jitter of less than 3 ms, it is unlikely that it will have any effect on call quality. There is no noticeable jitter for the Ethernet and WiMAX networks.

### 3.4 Results Summary

The performance of the closer Caller A/Callee A pair is within acceptable limits for both the Wi-Fi and WiMAX networks. However, only the WiMAX network is acceptable for the Caller B/Callee B pair; the Wi-Fi network resulted in too large of a packet loss to maintain a usable call.

## 4. Conclusions and Future Work

This project explored the feasibility of operating VoIP on Wi-Fi and WiMAX wireless networks as a replacement for cellular communications. Using OPNET, I simulated two pairs of users making a call to each other at different distances on these networks to collect statistics for packet loss, delay, and jitter.

The simulation results have shown that while the end-to-end delay and jitter are low enough to not have any significant effect on the call quality, packet loss becomes an issue. Even so, except for Caller B of the Wi-Fi network, all of the simulations showed the packet loss to be within a tolerable range to maintain a conversation.

Based on these findings, we can conclude that both Wi-Fi and WiMAX are able to handle VoIP traffic when within bounds of their wireless range.

However, we can not necessarily say that one technology is better than the other when it comes to potential as a substitute for cellular phones. The results have shown that each has its own place in the market.

When at close distances from the base station, the Wi-Fi network proves to be more ideal, with lower power usage and better performance. This makes Wi-Fi based VoIP calls a good choice for a contained setting where Wi-Fi could be readily available, such as a downtown office district or university campus. At further distances, Wi-Fi became unusable while WiMAX maintained most of its performance. If we want to enable cellular-like communications for users in rural areas, WiMAX becomes the only option.

We also have to consider cost and availability when choosing a technology. WiMAX is still a newer standard and deployment will be expensive, but many Wi-Fi access points will be required to cover the same area as a single WiMAX base station.

The project and the results raise several new questions that could be answered with future studies. If we are to deploy citywide Wi-Fi and WiMAX networks, how do we build a larger network with multiple base stations to provide the best coverage and signal strength? How many users could be supported on a single base station before performance is degraded? What is the effect of having users move about town while making a call?

The biggest challenge with the project was the lacking WiMAX model documentation and slow connection to the remote OPNET server. Combined, this made it difficult to initially prepare the WiMAX scenario, which required many trials to determine all of the required settings for a successful simulation. In addition, there was a steep learning curve to reach the point of reasonable competency with OPNET. Overall, the project gave a good insight into the technical details of Wi-Fi and WiMAX while learning the intricacies of the OPNET simulator.

# 5. References

[1] A. Safak and B. Preveze, "Analysis of delay factors for voice over WiMAX," *Computer and Information Sciences, 2008. ISCIS '08. 23rd International Symposium on*, Oct. 2008, pp. 1-6.

[2] Kostas, T.J, et al., "Real-time voice over packet-switched networks," *Network, IEEE*, vol. 12, no. 1, pp. 18-27, Jan/Feb 1998.

[3] S. Chiu, "Evaluation of Interactive Gaming Traffic over 802.11 Network," ENSC 835: High Performance Networks Final Project, Simon Fraser University, Apr. 2006.

[4] D. Zhao and X. Shen, "Performance of packet voice transmission using IEEE 802.16 protocol," *Wireless Communications, IEEE*, Feb. 2007, pp 44-51.

[5] Z. Yang, et al., "Downlink Performance of WiMAX Broadband from High Altitude Platform and Terrestrial Deployments sharing a common 3.5GHz band," presented at 2005 IST Mobile and Wireless Communications Summit, Dresden, Germany, 2005.

[6] S. Lloyd, “Challenges of Mobile WiMAX RF Transceivers,” *Solid-State and Integrated Circuit Technology, 2006. ICSICT '06. 8th International Conference on*, Oct. 2006, pp. 1821-1824.

[7] W. Hrudey and Lj. Trajkovic, "Streaming video content over IEEE 802.16/WiMAX broadband access," *OPNETWORK 2008*, Washington, DC, Aug. 2008.