ENSC 427 Communication Networks
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Exploration of VoIP Using Ethernet and Wi-Fi Networks

Under Different Scenarios

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List of Acronyms

Acronyms	Extended Form
LAN	Local Area Network
VoIP	Voice over Internet Protocol
POTS	Plain Old Telephone System
PSTN	Public Switched Telephone Network
MOS	Mean Opinion Score
ETE Delay	End-to-End Delay
ITU	International Telecommunication Union

Abstract

From emails to chat rooms to instant messengers, the impact of the Internet on communication from each other has changed the world. It is foreseeable that the next step in communication with Internet with be transmission of voice. This once thought to be an innovative concept has been brought into reality. Voice over Internet Protocol (VoIP) is a technology which converts the analog audio signal from human voice into digital data and relay it through the Internet to destination. In this project, we will examine and compare the performances of VoIP-to-VoIP under WLAN 802.11g and Ethernet connections within a company and between 2 companies. Evaluation of jitter, delay, Mean Opinion Score (MOS) and packet loss will be used to dictate whether company should abandon the use of the traditional copper wire telephone and adopt the lower cost VoIP systems. Moreover, situational parameters such as heavy traffic and interference will also be discussed and considered when making the decision to determine whether an upgrade will be worthwhile.

1.0 Introduction

1.1 Public Switched Telephone Network

Our telephone system or what we technically called the Public Switched Telephone Network (PSTN) has been allowing people of different location to communicate with each other for over a century. The plain old telephone system (POTS) employs the highly inefficient circuit switching as the method of transferring data across the network through a dedicated path which is released upon termination of a call. Thus, it is costly to deploy. Moreover, sound quality of POTS is 10 kHz with an 8-bit resolution at its very best, which is much similar to the quality of an AM radio station [1]. However, it does have its benefits as POTS is highly reliable as the average drop call rate is 1 out of 1000 [2].

1.2 Voice over Internet Protocol

The use of Voice over IP (VoIP) has increased in its popularity tremendously within the last few years. It is the modern mean of transmitting digital voice data through the Internet. VoIP utilizes the packet switching network which is highly efficient as data are distributed into packets before transmission to their destination address via different paths that are shared among different users. Thus, it better utilizes the available bandwidth of the network. In addition, call quality is much higher with a 16-bit resolution operating from 22 kHz to 44.1 kHz to ensure crystal clear communication [1]. Moreover, it is free to make a VoIP-to-VoIP call to anywhere in the world without having to face long-distance charges. VoIP-to-PSTN services are available as well with a usual monthly charge but this is beyond the scope of our discussion for this project. However, VoIP do have its share of short-comings as it suffers up to a 5% drop calls [2]. Furthermore, it will not function in case of electricity outage and there are currently no direct VoIP-to-VoIP emergency call lines such as 911 available.

1.3 Purpose and Goals

The purpose of the project is to determine whether VoIP-to-VoIP should be employed in organization to replace the leasing of dedicated phone lines between companies located at various locations. Successful implementation of this technology will result in a saving of millions of dollars.

Currently the standard for acceptable Mean Opinion Score (MOS) for the landline telephone is 4.0 while MOS for VoIP is set to 4.0 – 4.5 set up the International Telecommunication Union (ITU). The recommended target for end-to-end delay is at 150ms as shown in **Figure 1.3.1.** However, in reality, many VoIP conversations occur over satellite connection as well so the acceptable delay is around 250ms [3]. Finally, the unpredictable delay of packet delivery known as “jitter” will be sampled. The ITU recommended tolerable range is between 20 to 30 ms [6].

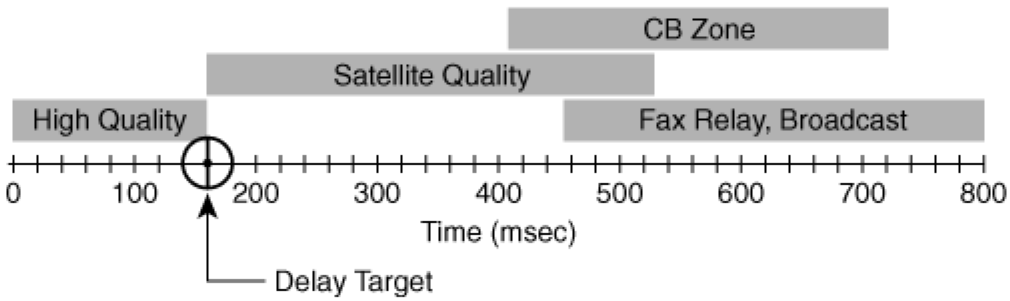


Figure 1.3.1 Recommended End-to-End Delay for High Quality Voice Transmission by ITU

Analysis of MOS, end-to-end delay, and jitter under WLAN802.11g and Ethernet networks will be evaluated from the statistics collected in the scenarios stated in **Figure 1.3.2** to ensure they meet within the international standards suggested by the ITU above. If the observed statistics are shown to be within tolerable standards then it will better indicate that VoIP has the potential to be employed as an alternative to the traditional lease lines used by organizations.

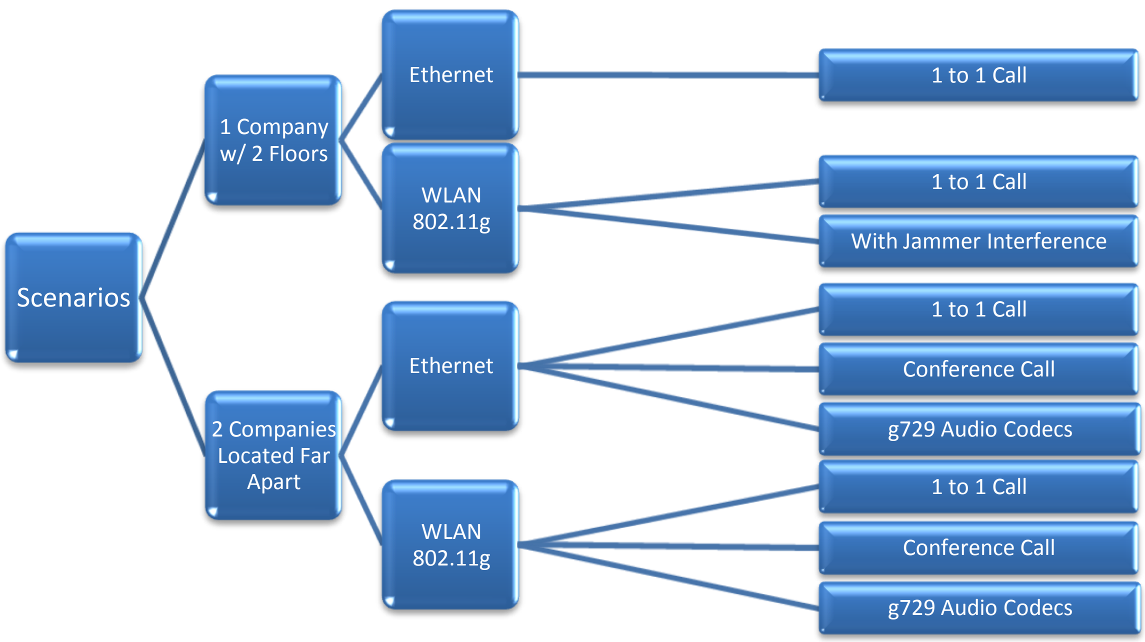


Figure 1.3.2 Scenarios under Investigation

2.0 Configurations and Designs

In our project we have created nine different scenarios in total as shown in **Figure 1.3.2**. We will use these scenarios to compare the performance of VoIP under the following five circumstances:

1. WLAN 802.11g versus Ethernet Network
2. Local versus Long Distance Calls
3. Audio Codec: G.711 versus G.729
4. Introduction of 2.4GHz Interference to WLAN 802.11g Network
5. Conference Calls (Multiple Call Conversations)

All simulations are done via direct VoIP to VoIP connections using G.711 codec unless otherwise specified. 1 voice frame has been configured to be sent per packet as shown in **Figure 2.0.1** below.

Figure 2.0.1 Configuration of VoIP Settings under VoIP Application Definition

2.1 Physical network model of a company with 2 floors

One of our testing cases is to determine whether to use a wireless or Ethernet local area network within a company of 2 floors. Thus, we have set up the following topologies. In both cases (wireless and Ethernet), the work stations are set 4 meters apart from each other in altitude to represent the difference of 2 floors.

The network topology for a 1 to 1 Ethernet VoIP call between 2 floors of a company is as shown in **Figure 2.1.1**. The workstation from each floor is connected via an Ethernet switch, one for its respected floor. These switches are connected to a Cisco 4000 router to set up a LAN environment within the company to enable VoIP connections.

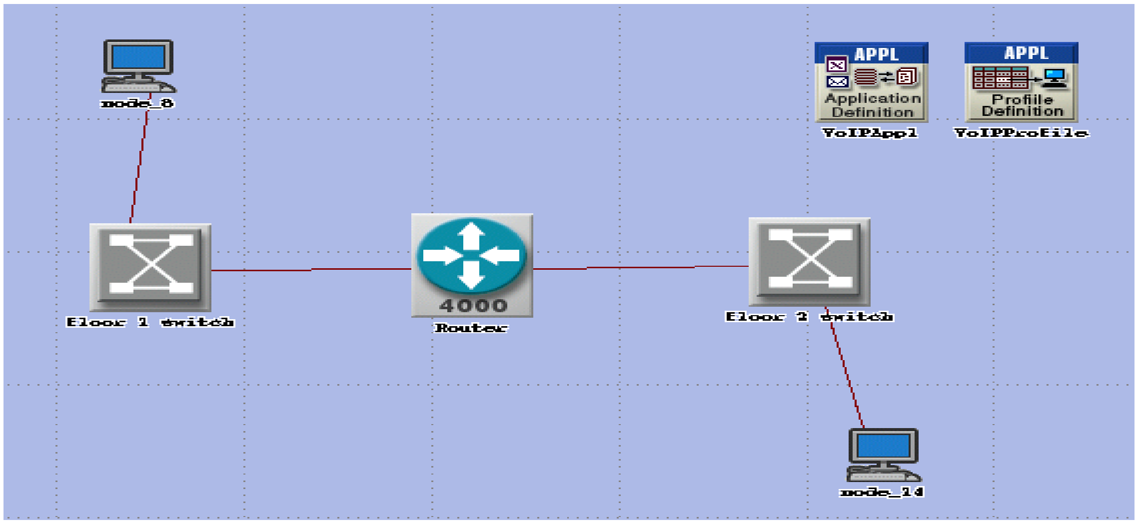


Figure 2.1.1 Ethernet Topology between 2 Floors of a Company

The topology for a 1 to 1 connection between 2 floors using WLAN 802.11g is as shown in **Figure 2.1.2**. We have placed the wireless router on the ground floor and configured its data transfer rate to 56Mbps to ensure it is operating under WLAN 802.11g condition in **Figure 2.1.2**.

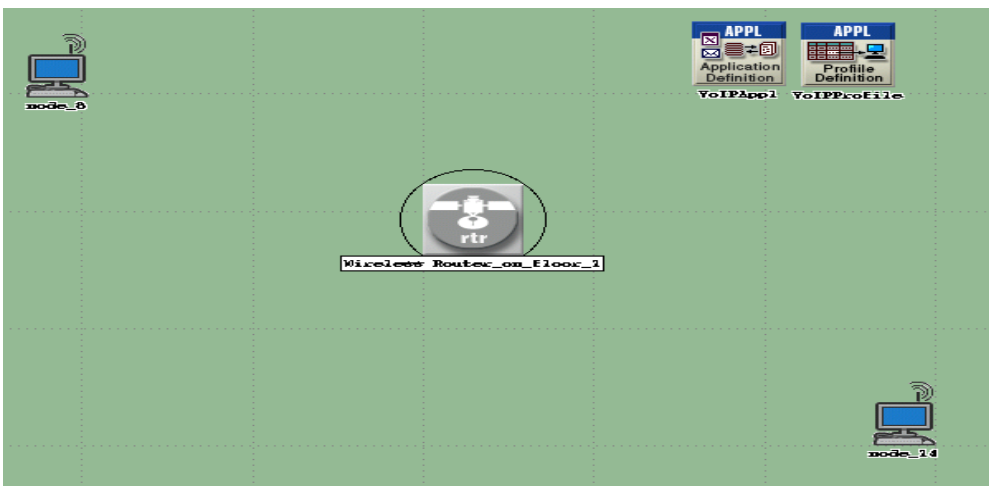
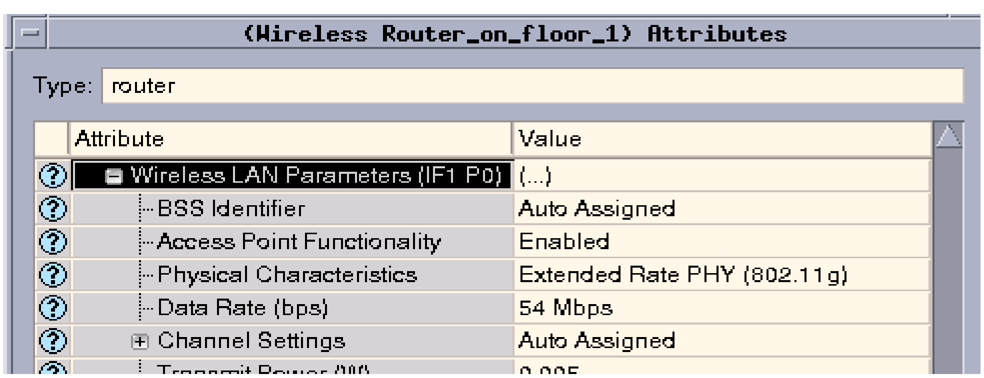


Figure 2.1.2 WLAN 802.11g Topology between 2 Floors of a Company


(Wireless Router_on_floor_1) Attributes

Type: router

Attribute	Value
Wireless LAN Parameters (IF1 P0)	(...)
BSS Identifier	Auto Assigned
Access Point Functionality	Enabled
Physical Characteristics	Extended Rate PHY (802.11g)
Data Rate (bps)	54 Mbps
Channel Settings	Auto Assigned

Figure 2.1.3 Wireless Router Attributes Table

2.2 Physical network model of company located at a distance

Two companies or subnets located in Vancouver and Montreal respectively are set up to demonstrate the performance of long distance VoIP calls using wireless or Ethernet connections as their LAN. However, due to the limited signaling range of wireless networks. A point-to-point duplex linkage (PPP DS3) is used to connect the CISCO 4000 routers of the 2 subnets in **Figure 2.2.1**. The topologies of both companies are identical to that of the other for consistency.

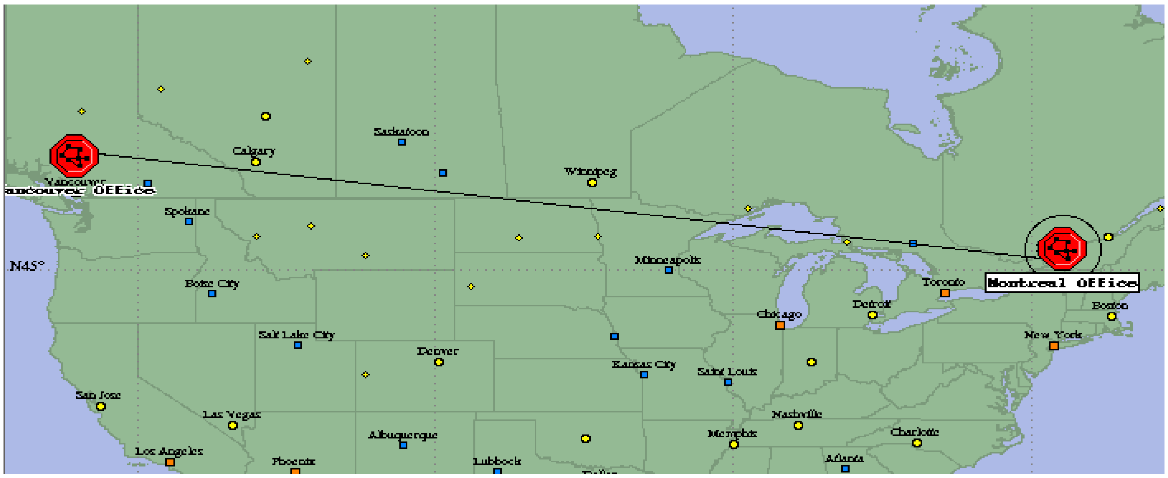


Figure 2.2.1 Network Model for Vancouver and Montreal Companies

Direct 1 to 1 VoIP calls under Ethernet and WLAN 802.11g employs the identical topology as **Figure 2.1.1** and **Figure 2.1.2** respectively. However, conference calls between the two locations is established to allow 32 simultaneous VoIP connections in order to test its performance under heavy traffic as shown in **Figure 2.2.2** and **Figure 2.2.3**.

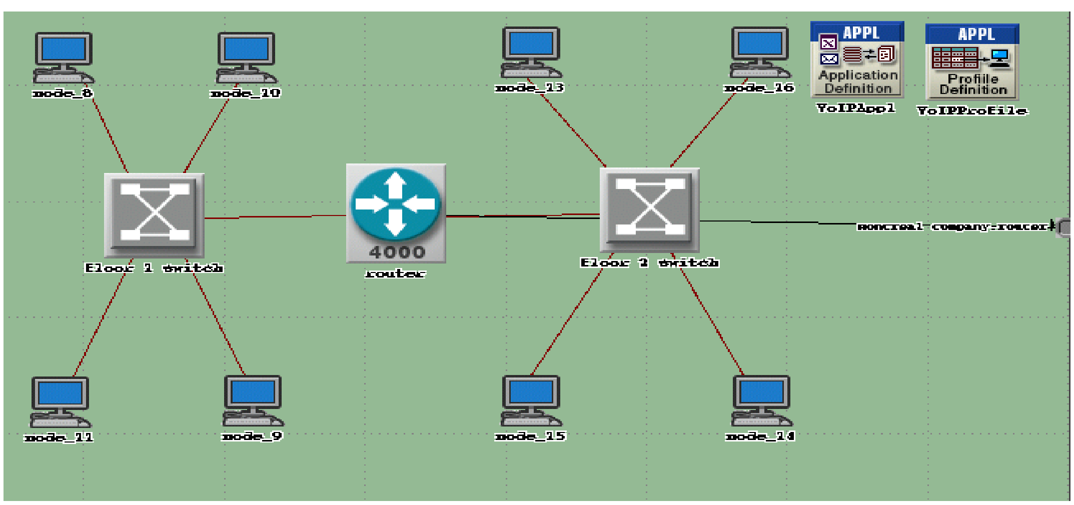


Figure 2.2.2 Conference Call Topology under Ethernet Network

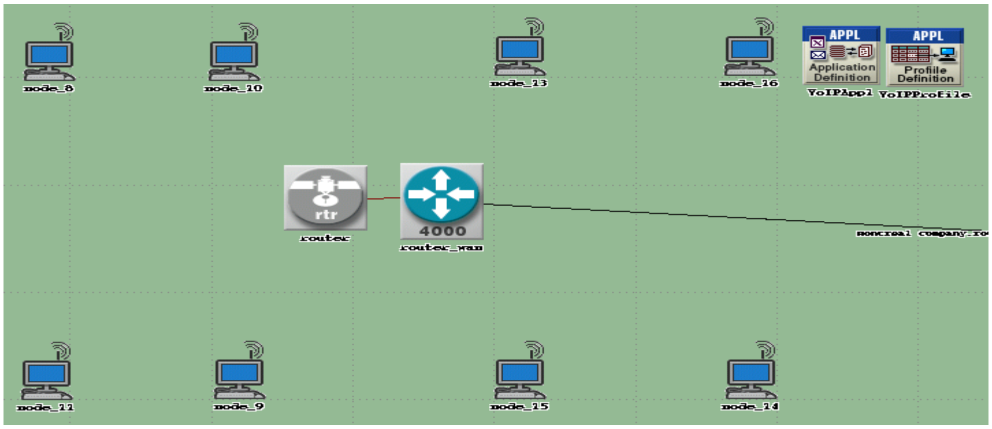


Figure 2.2.3 Conference Call Topology under WLAN 802.11g Network

2.3 Physical network model with wireless interference

Whether it is WLAN 802.11b or WLAN 802.11g, wireless network do sometimes suffer interference from other electronics. Due to the fact that WLAN 802.11g operates in the 2.4GHz range [7], there are many electronics devices that also operate within that region such as cordless telephones, microwave oven and Bluetooth devices. In this network, we introduce a jammer to operate in the 2.40 GHz frequency to interfere with the wireless router as shown in **Figure 2.3.1**.

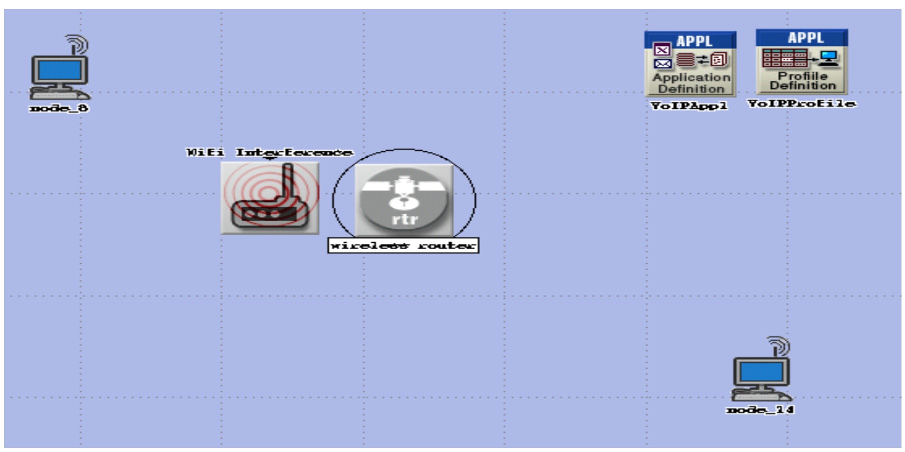
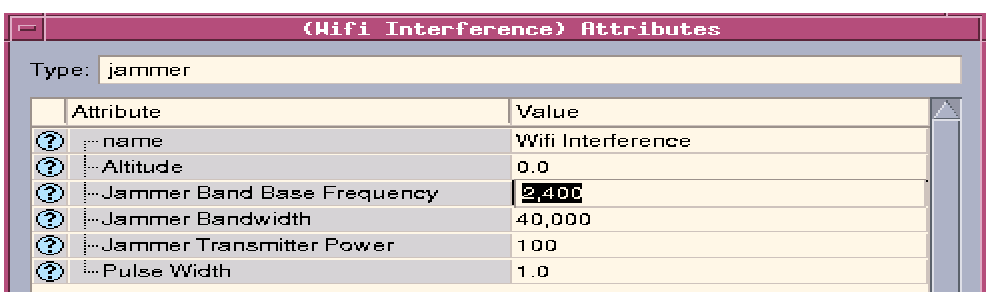


Figure 2.3.1 WLAN 802.11g topology with 2.4GHz interference



(Wifi Interference) Attributes

Type: jammer

Attribute	Value
name	Wifi Interference
Altitude	0.0
Jammer Band Base Frequency	2,400
Jammer Bandwidth	40,000
Jammer Transmitter Power	100
Pulse Width	1.0

Figure 2.3.2 Jammer configured to emit 2.4GHz interference

3.0 Simulation Results and Comparisons

Simulation parameters such as jitter, mean opinion score (MOS) value, delay variation and end-to-end delay (ETE Delay) results will be gathered and compared to dictate whether VoIP is a potential candidate to replace the traditional telephone system in the near future.

Jitter: The delay in packet transmission that leads to pulse displacement. Jitter can be thought as "shaky pulse".

Mean Opinion Score Value (MOS Value): The numerical measurement of voice quality. MOS is expressed in a scale from 1(worst) to 5(best).

Delay Variation: The difference measurement in end to end delay between packets.

End to End Delay (ETE Delay): the time required for a packet to travel from source through network to destination.

Before we simulate, VoIP traffic flows must be created in each of the scenarios as shown in **Figure 3.0.1**. For each of the simulation lasting 6 minutes, we will generate 5 calls worth 1 minute long each. The first packet will be delivered at the 10th second into the simulation.

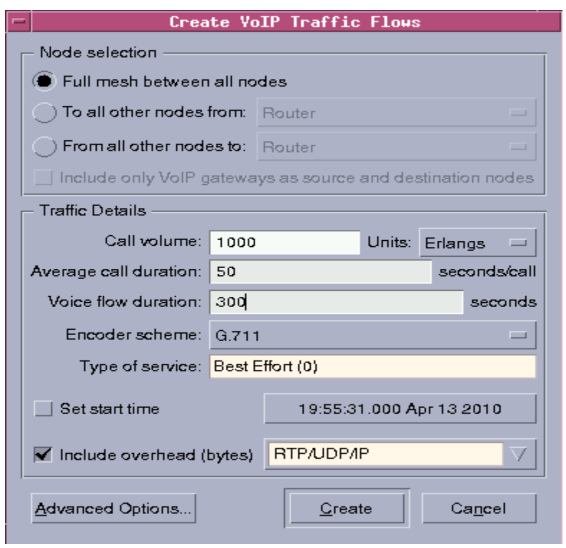


Figure 3.0.1 Creating VoIP Traffic Flow to Simulate

3.1 Ethernet vs. WLAN 802.11g networks under Local/Long Distance Call

In this section, we want to explore the effects that WLAN 802.11g and Ethernet networks has on both local and long distance VoIP calls. This test will be performed using a direct VoIP to VoIP from 1 workstation to another in an ideal environment where traffic congestion and wireless interference is not present.

As shown in **Figure 3.1.1**, the average amount of voice jitter is next to nothing. However, jitter is noticeable in the wireless scenarios. In **Figure 3.1.2**, we noticed that the MOS value is at its highest when it is a local call using Ethernet networks. With long distance calls Ethernet still provides a better score than wireless which is what we intuitively thought. Moreover, from our intuition the further the packets have to travel, the longer time it will take. This is exactly what we observed in the average end-to-end delay of the packets in **Figure 3.1.3**.

Implications:

From our observation, we noticed that the MOS value, thus the quality, decreases with distance between the two nodes in the conversation. The type of network played no significant difference in affecting the MOS value. Moreover, delay increases with distance needed to be travelled by the packets. From our previous research, we found the fact that the acceptable delay for POTS is within 150 ms [3]. The maximum delay we received with a packet travelling from Vancouver to Montreal was approximately 72 ms, which greatly satisfy the acceptable standards set by traditional telephone system. Thus, it is safe to say in this scenario both local and long distance calls is acceptable under either Ethernet or WLAN 802.11g network.

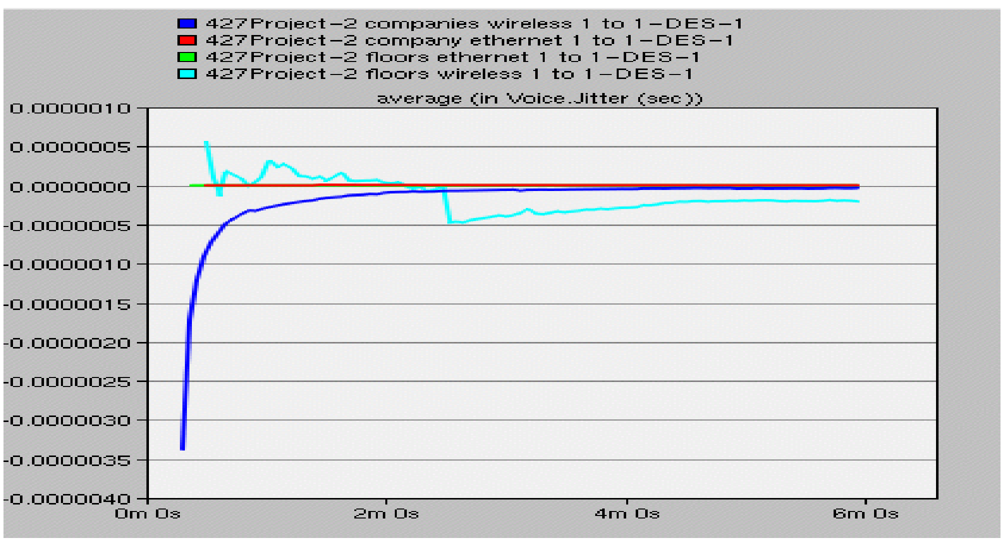


Figure 3.1.1 Average Voice Jitter (Local vs. Long Distance Call)

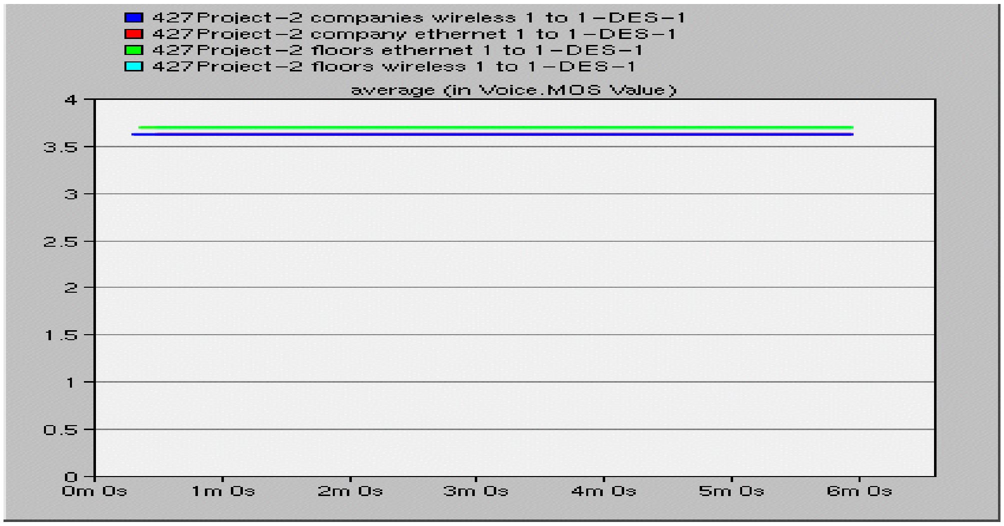


Figure 3.1.2 Average MOS Value (Local vs. Long Distance Call)

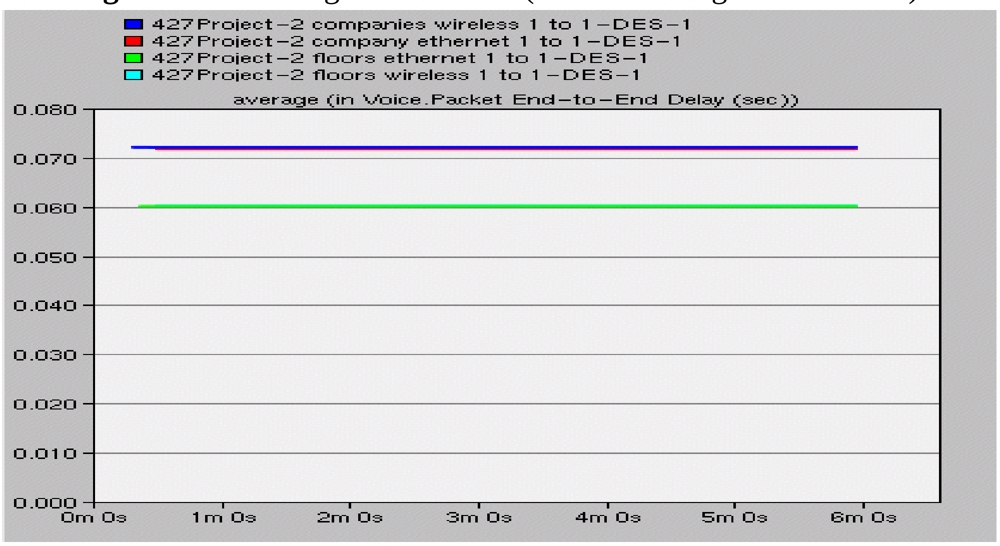


Figure 3.1.3 Average Packet End-to-End Delay (Local vs. Long Distance Call)

3.2 Audio Codec: G.711 vs. G.729

Both G.711 and G.729 audio codec are a popular choice when it comes to VoIP. G.711 was first standardized by the International Telecommunication Union in 1988 [8]. Due to the fact that it uses uncompressed audio stream at 64 Kbps and the public was still using 56k dial-up internet connections, conversations were often very choppy. Therefore, in 1995, the G.729 codec was adopted as it uses compressed audio streams, thus requiring less bandwidth [9]. The tradeoff between the two is basically superior quality (G.711) vs. low usage of bandwidth (G.729).

In this section, we want to compare the performance of the two audio codec during a 1 to 1 long distance call from Vancouver to Montreal under wireless network setting. **Figure 3.2.1** shows the average voice jitter amount to flavor the G.729 codec as it shows less fluctuation. However, in **Figure 3.2.2**, the MOS value for G.729 is at approximately 3.1 while G.711 codec's MOS value is at 3.7, which does not change with an Ethernet network setup. Moreover, in **Figure 3.2.3**, the G.729 starts with a higher delay variation towards the beginning of the call then stabilizes as the call continues. This may be due to the fact that it is much more difficult to predict the next pulse in the beginning of a call as it uses a linear prediction compression algorithm [10]. In **Figure 3.2.4**, it is shown that the average ETE delay remains constant between the two scenarios.

Implications:

Saving bandwidth is certainly important, but with the emergence of broadband internet connections to almost every home in North America in recent years, we can afford to use up a little bit more bandwidth for better quality. The MOS value of 3.1 is unacceptable by many users. Through our research and comparison with our OPNET results, we noticed that OPNET tends to generate a MOS value 0.4-0.6 lower than their theoretical values. However, based on our simulation results we believe it is better to continue the use of the G.711 codec to ensure quality when bandwidth is not an issue.

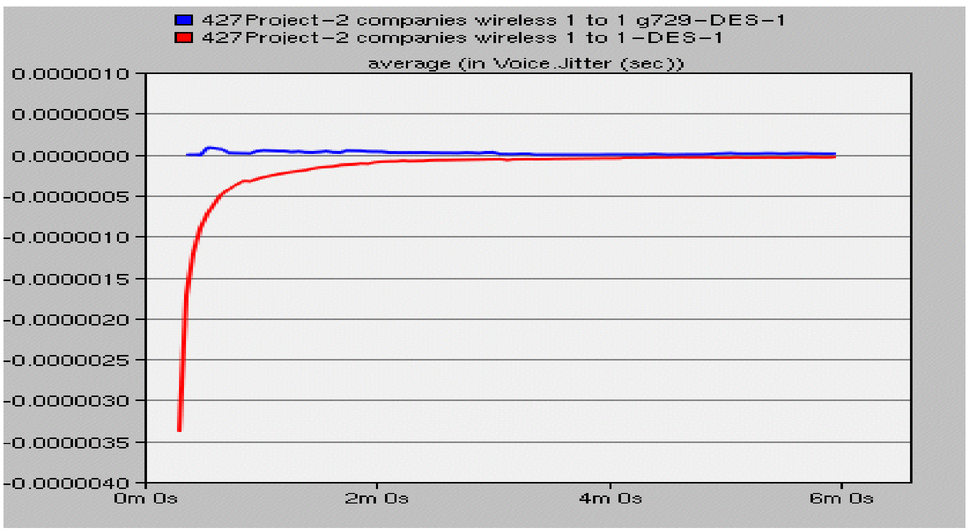


Figure 3.2.1 Average Voice Jitter (G.729 vs. G.711)

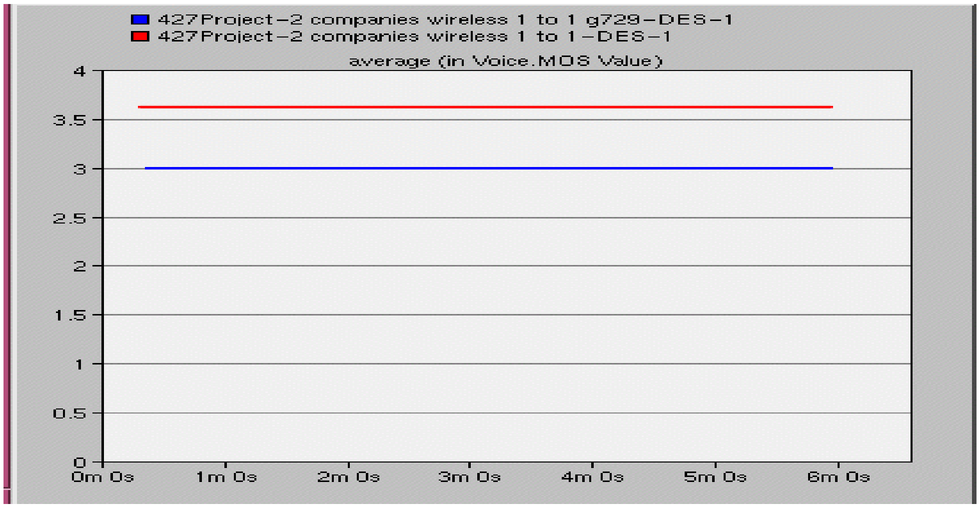


Figure 3.2.2 Average MOS Value (G.729 vs. G.711)

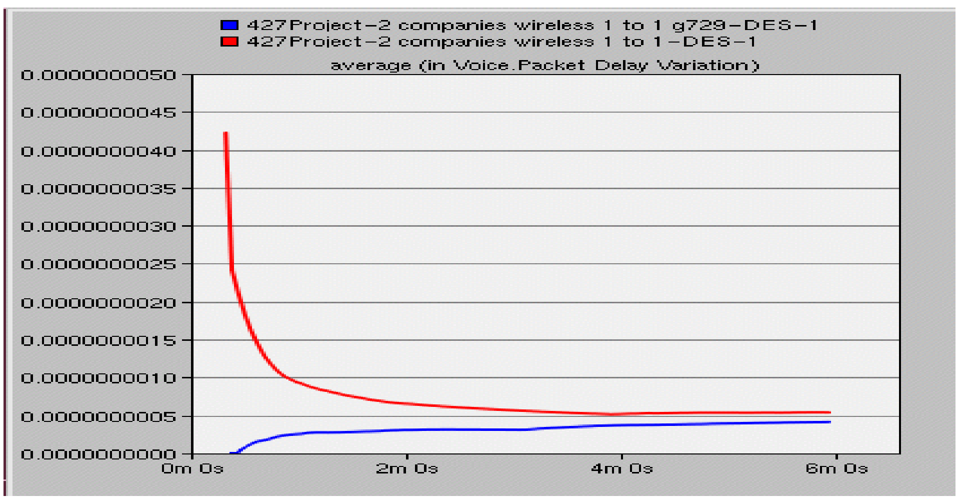


Figure 3.2.3 Average Packet Variation Delay (G.729 vs. G.711)

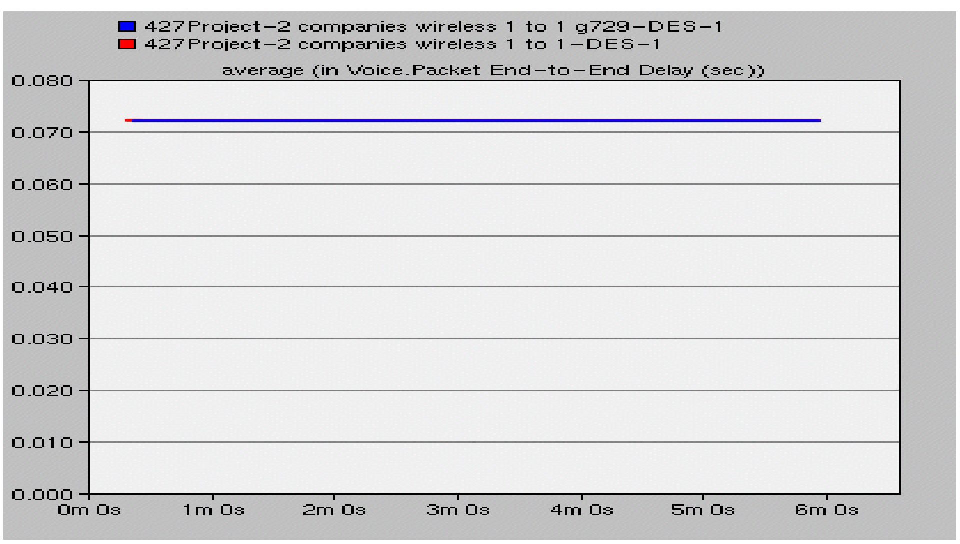


Figure 3.2.4 Average End-to-End Delay (G.729 vs. G.711)

3.3 WLAN 802.11g under wireless Interference

In this section, we wish to study the effect of wireless interference on VoIP transmission. The scenario in **Figure 2.1.2** had been duplicated and added a jammer. **Figure 3.3.1** to **Figure 3.3.4** below shows the simulation result of this scenario. In **Figure 3.3.1**, there is so much higher jitter and jitter variation when jammer is introduced compare to the no jammer scenario. Maximum jitter in jammer case goes up to 0.0022 second where as on the no jammer case, highest jitter only go up to 0.0000005 second. When comparing MOS values between two cases, we found the scenario with jammer has lower MOS score than no jammer scenario by roughly 0.062. We believe this is caused by to all jitter mention above. **Figure 3.3.3** shows our result for delay variation for both scenarios. Scenario with jammer has very high delay variation of 0.09 second compare to 0.05 microsecond of no jammer scenario. On **Figure 3.3.4**, we see the difference in end to end delay between two scenarios. Highest packet end to end delay of jammer case is 0.265 second compares to 0.06 of no jammer case. Moreover, the delay in jammer case is much less stable than delay of no jammer case. The effect of delay variations on **Figure 3.3.3** can also be seen on this **Figure 3.3.4**.

Implications:

From our simulation results, we observed that introducing jammer into the scenario can significantly change jitter, MOS value, delay variation and end to end delay. As the power of interference increase, value of jitters and delays increase, MOS value decrease. However, wireless interference had been a known issue for a while. There had been many studies, researches done to suppress interference effect. From researches, we found that WLAN is divided into channels of different frequencies [7]. When interference occurs, user can switch to different channel avoid interference effect. In Bluetooth and cordless phone, frequency hopping is a method of rapidly changing transmission frequency. Hence, these electronic devices will less likely to interfere with WLAN router. [12] Therefore, interference is not a critical issue of VoIP on WLAN.

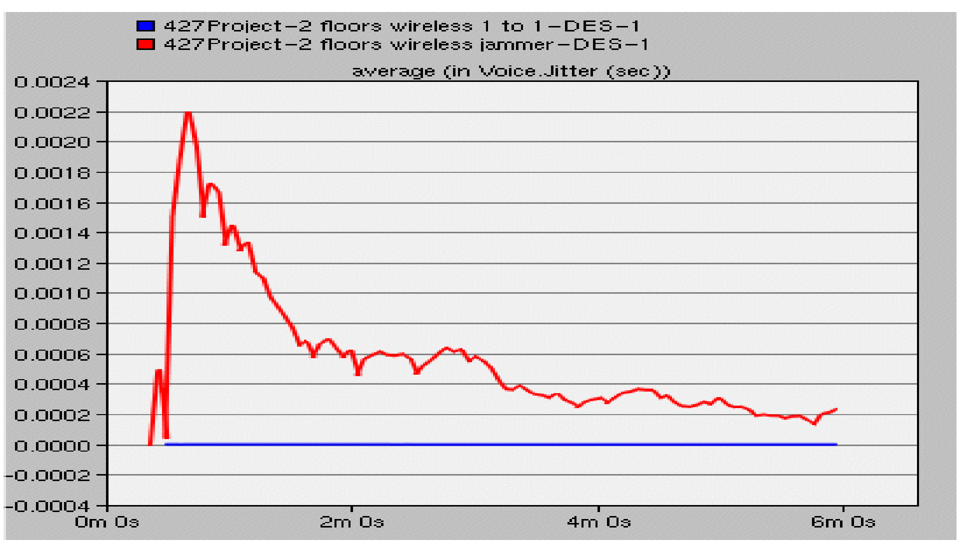


Figure 3.3.1 Average Voice Jitter (no jammer vs. jammer)

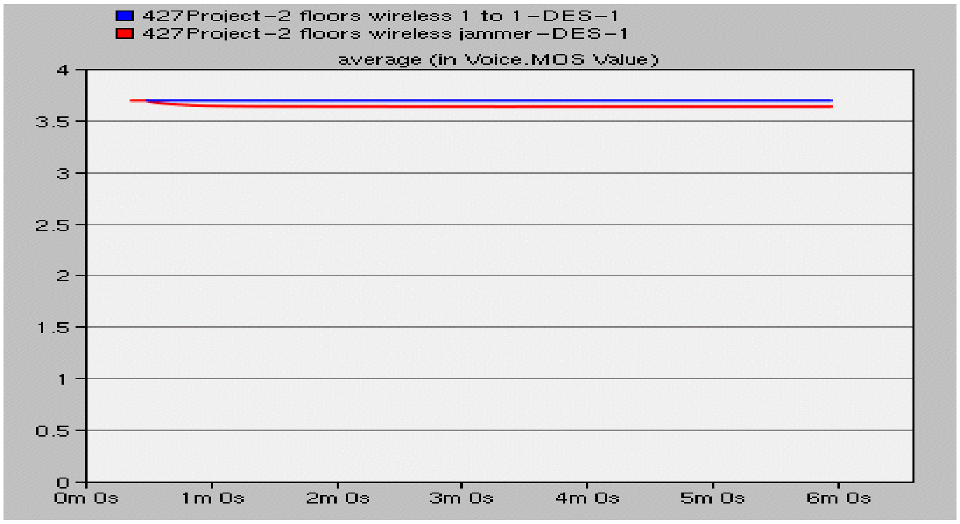


Figure 3.3.2 Average Voice MOS Value (no jammer vs. jammer)

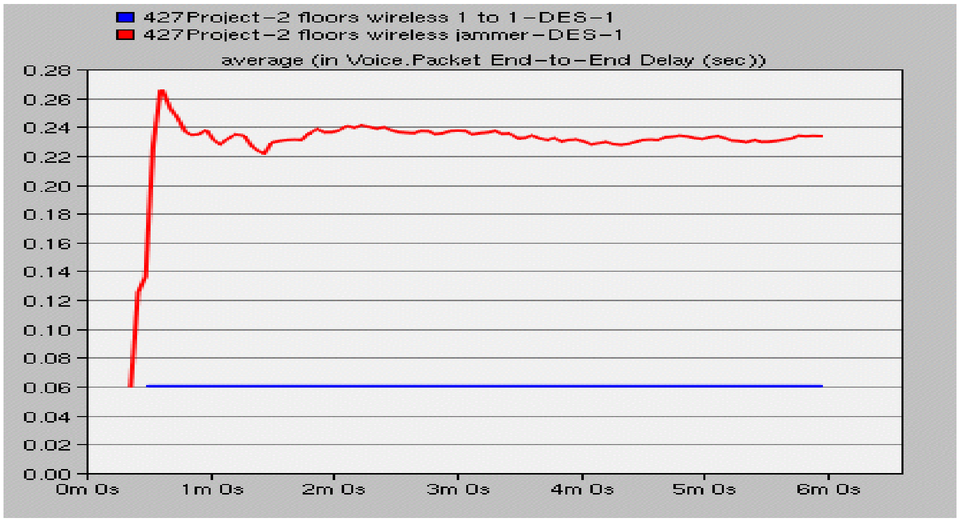


Figure 3.3.3 Average End to End Delay (no jammer vs. jammer)

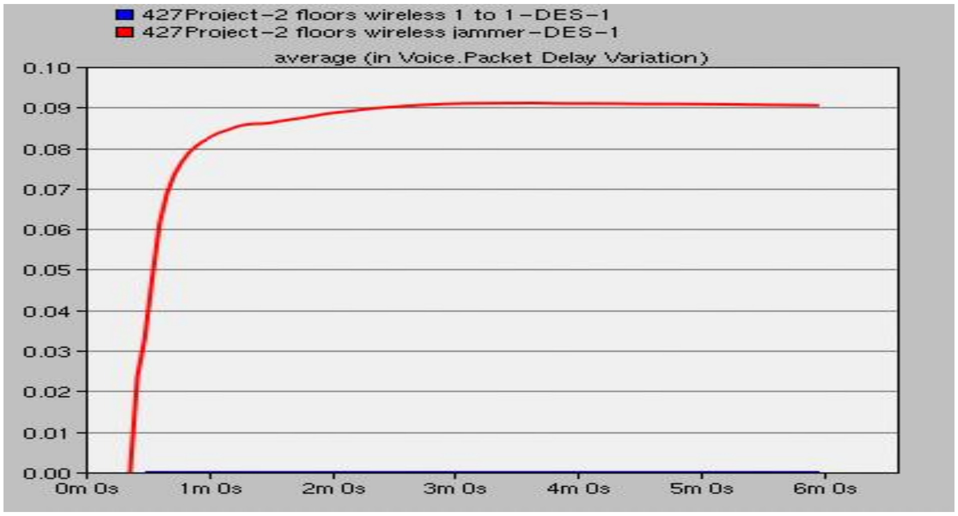


Figure 3.3.4 Average Voice Delay Variation (no jammer vs. jammer)

3.4 VoIP Conference Calls (Heavy Traffic)

This section provides simulation results and analysis of VoIP conference call. Based on scenario of long distance call as in **Figure 2.2.1**, we add in more work station and generate mesh calls to increase the traffic. **Figure 3.4.1** to **Figure 3.4.4**, show our average jitter, MOS value, delay variation and end to end delay comparison between conference call using Ethernet local connection and wireless local connection.

Between Ethernet conference call and wireless conference call, wireless case has more jitter variations. Although this does not seem to affect MOS values since both are a still at about 3.7 score. Again, wireless shows more delay variation on **Figure 3.4.3** compare to Ethernet. End to end delay of Ethernet scenario appears to be slightly higher than of wireless scenario.

Implications:

For both Ethernet and wireless, we learned that there are small jitter and delay variation. However, MOS score value for voice is still quite high at 3.7. This means voice quality for long distance conference call is still very acceptable. Compare to long distance 1 on 1 scenario, delays increase as the number of work station increase. This delay, however is still under ITU generally accepted 150ms one-way delay limit. [13]

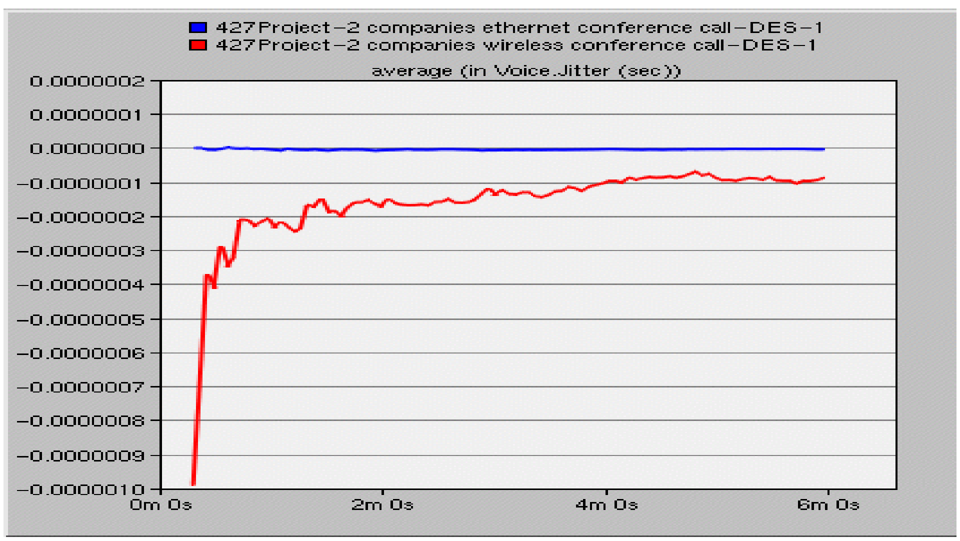


Figure 3.4.1 Average Voice Jitter (conference ethernet vs. wireless)

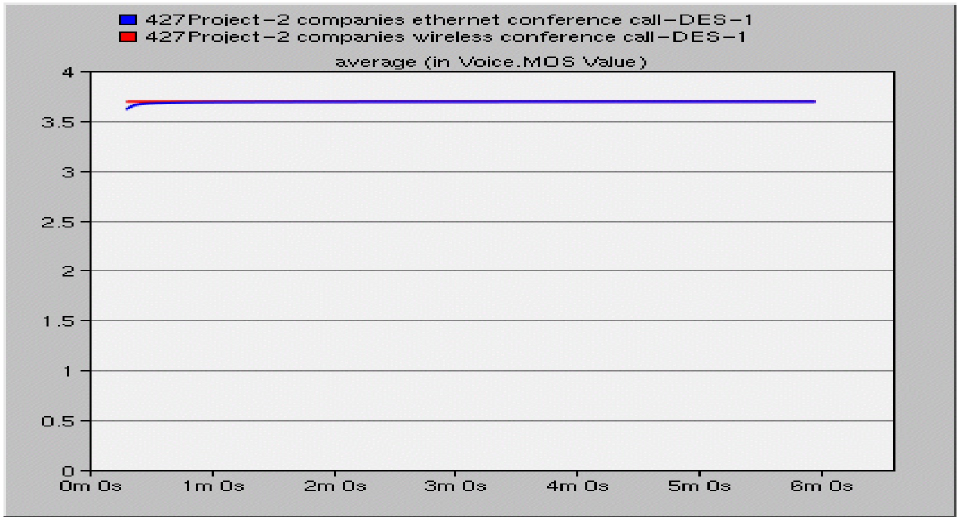


Figure 3.4.2 Average Voice MOS Value (conference ethernet vs. wireless)

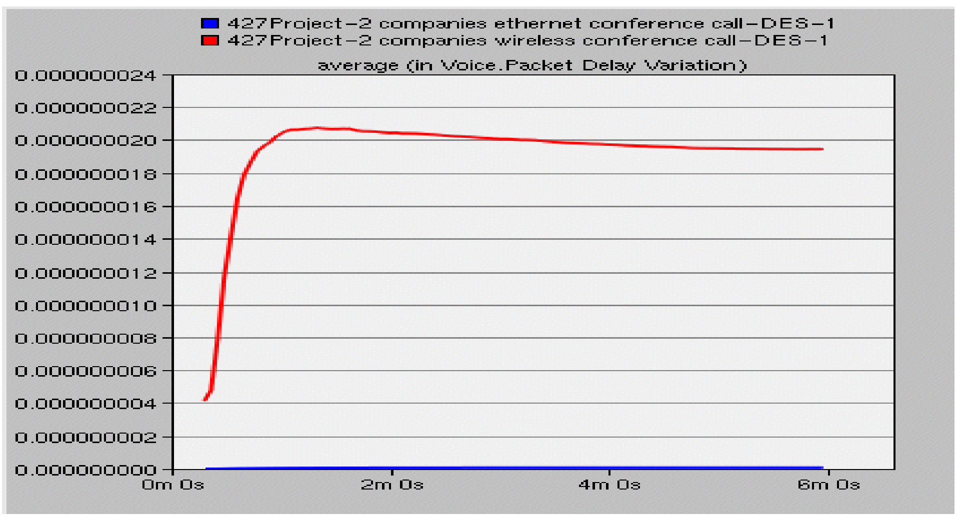


Figure 3.4.3 Average Delay Variation (conference Ethernet vs. wireless)

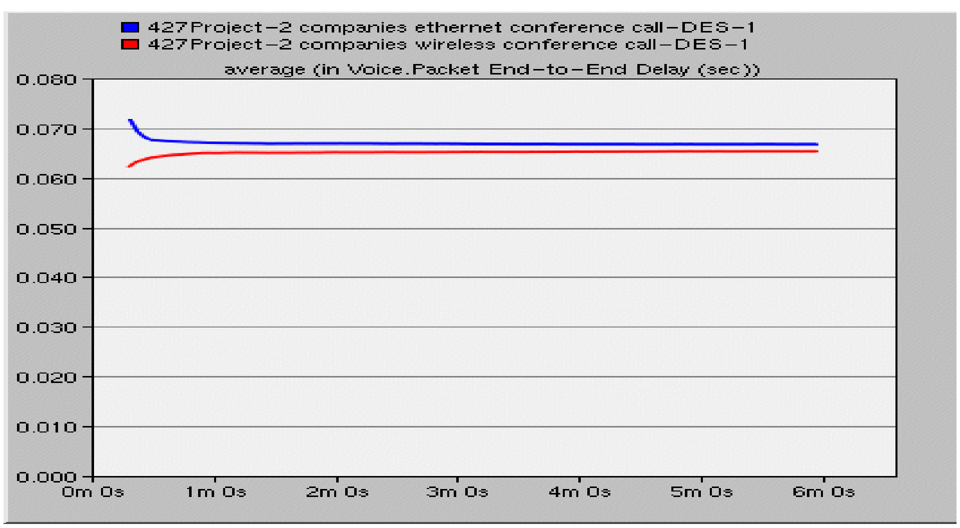


Figure 3.4.4 Average End to End Delay (conference ethernet vs. wireless)

3.5 Result Summary Analysis

Assuming the base model is a 1 to 1 local call under Ethernet network. The following table shows the respected change to parameters such as jitter, MOS Value, packet delay variation and ETE delay by applying the factors suggested on the left-most column. These generalizations are analyzed based on the above simulation results.

Factors	Jitter	MOS Value	Delay Variation	End-to-End Delay
WLAN 802.11g	Increase	No Change	Increase	No Change
Increase Distance between Callers	Increase	Decrease	Increase	Increase
Added Wireless Interference	Increase	No Change	Increase	No Change
Increase Workstations	Increase	No Change	Increase	Increase
Under G.729 Audio Codec	Decrease	Decrease	Decrease	No Change

Table 3.5.1 Summary of VoIP Performance Affected by Different Situations

4.0 Conclusion

In this project, we studied simulation results of many scenarios. We compared differences wireless versus Ethernet, G711 versus G729, interference versus no interference, 1 to 1 call versus conference all. All simulation results summary had been compiled into **Table 3.5.1** above. For many difference scenarios, VoIP appears to be a very good candidate for voice transmission. Although, long distance, heavy traffic and interference can decrease voice quality, increase delay, there are many approaches to deal with this issue. From all the researches and studies we had done, we believe that VoIP will be an excellent substitution for traditional telephone landline in the new future.

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