ENSC 427 COMMUNICATION NETWORKS

Implementation of VoIP over a Campus Wide Network

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FINAL PROJECT

Xuan Lu

Kevan Thompson

Zhiyu Zhou

http://www.sfu.ca/~zzhou/project.html

xla9@sfu.ca

kjthomps@sfu.ca

zzhou@sfu.ca

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iii. List of Abbreviations

ATA Analog Telephone Adapter

CAN Campus Area Network

Codec: **Co**mpressor-**De-C**ompressor

GSM General System for Mobile Communications

IEEE Institute of Electrical and Electronics Engineers

IP Internet Protocol

LAN Local Area Network

MOS Mean Opinion Score

PCM Pulse Code Modulation

PLC Packet Loss Concealment

QoS Quality of Service

VoIP: Voice over Internet Protocol

WLAN: Wireless Local Area Network

1. Abstract

VoIP is a technology that is greatly rising in popularity[1], and potentially offers many advantages, and cost savings [2] over traditional telephone services [3]. This project will attempt to assess the feasibility of implementing VoIP on a Campus Area Network (CAN). Beginning with a simple Local Area Network (LAN) we'll expand the network, and the background traffic, and compare the voice quality. We'll also compare various encoding schemes. Finally we'll compare a wired LAN to a wireless LAN. We'll being with a simple wireless LAN, and then vary the distance from the wireless access point to wireless workstations, and the number of wireless workstations.

2. Introduction

When making a call using Voice over Internet Protocol (VoIP) the person's voice is sampled using a microphone, and is converted into data and stored on a computer. The data is compressed using a Codec, formed into packets and set across an IP network to the receiver. On the receiver side the packets are buffered since they may not arrive in the correct order. The packets are then decoded, and played using a speaker. [4]

A typical VoIP setup is shown in figure 1.

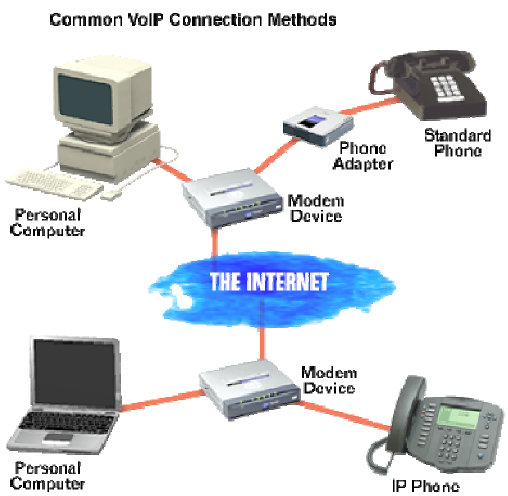

Figure 1 Common VoIP Setup[5]

A VoIP phone may take the form of an analog telephone using an Analog Telephone Adapter (ATA), a dedicated VoIP phone, or software running on a PC.

2.1 Scope of the Project

VoIP is a technology that is greatly rising in popularity [1], and potentially offers many advantages, and cost savings [2] over traditional telephone services [3]. This is particularly true for Universities and companies with large CANs. This project will attempt to assess the feasibility of implementing VoIP on a CAN. Beginning with a simple Local Area Network (LAN) with just 10 nodes we'll expand the increase background traffic and expand the network to 30 nodes, and then we'll compare a wired LAN to a wireless LAN. We'll being with a simple 10 node wireless LAN, and increasing the number of wireless workstations to 30 nodes. Finally, we will compare the voice quality. We'll also compare PCM, and GSM encoding schemes.

In our simulations we will look at Jitter, end to end Packet Delay, and MOS.

3. Performance Measurement

The Quality of Service (QoS) of VoIP depends mostly on: Packet Delay, Jitter and Packet Loss.

3.1 End-to-End Delay

Packet Delay is just the amount of time it takes for a packet to get from the sender to the receiver. A good network has a packet delay of 100ms, but packet delay of up to 400ms is still acceptable [4]. If the delay is too high, then packets maybe dropped because they're arrive too old.

3.2 Jitter

Jitter is the variation in packet delay [6]. Too much Jitter may result in choppy voice quality, or temporary glitches. [4] A Jitter buffer can be used to queue the packets and help to decrease the effects of Jitter.

3.3 Packet Lost Ratio

When packets are lost due to Packet Loss the receiving Codec must fill in the gap, this process is called Packet Loss Concealment (PLC)[4]. To elevate this packets are often sent multiple times in order. This method is called redundancy. A other method to compensate for Packet Loss is to send a part of the previous part in the subsequent packets. This method is called Forward Error Correction (FEC), and allows for the lost packet to be reconstructed [4].

3.4 Quality-of-Service (MOS)

One measure of voice quality is to use the Mean Opinion Score (MOS). MOS is a scale which rates voice quality on a five point scale, as follows:

> **5** - Perfect. Like face-to-face conversation or radio reception.
>
> **4** - Fair. Imperfections can be perceived, but sound still clear. This is (supposedly) the range for cell phones.
>
> **3** - Annoying.
>
> **2** - Very annoying and nearly impossible to communicate.
>
> **1** - Impossible to communicate [7]

4. Network Implementation

In this project, we are implementing a network simulating a campus wide area network.

4.1 Topology of Campus Node

The figure below shows the sub-node model representing the entire campus network. On the geographical map, it is represented by a single sub-node.

Figure 2 Campus Node on a Geographical Map

4.1.1 Campus Node Sub-node Model

The figure below shows the details about the implementation for the campus wide network, where the campus network is simplified into a server sub-node and a residence sub-node.

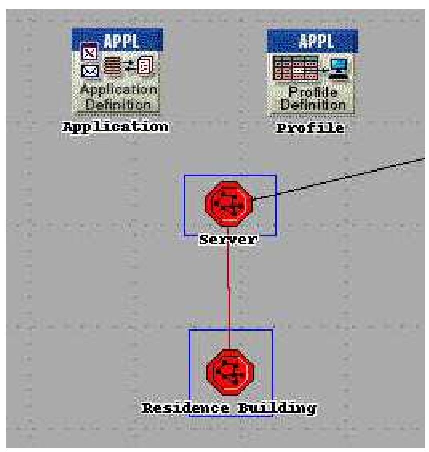

Figure 3 Campus Sub-Node Model

The server sub-node is where the supporting services will be provided. These include HTTP, FTP, Printing and the proposed VoIP service. The details about this particular node will be discussed later. The Residence node represents a particular building (student residence building) where our testing implementation of VoIP will be held. Details about this node will be discussed later. The network connection between the server node and the residence node is where traffic will be routed between the clients from the resident building and the respective service provider by the server node. The transmission lines between these two nodes are set to 10Mbps in bandwidth. Also, because we are trying to model the entire campus network as simple as possible while still maintaining the functionality of such a simplified network model, we'll assumed that the connection between the server node and the resident building node is in fact the same traffic route for all other internet traffics between the campus server and other internet applications for other building on campus, such as library, research center and administration building. To model other internet traffics that already existed elsewhere on the campus, such traffic is modeled as background traffic pattern with constant load as the percentage of total line bandwidth.

4.2 Topology of Residence Node

The residence node is used to represent the cluster of users at a resident building, where general internet usage already exists, and where new VoIP will be introduced and tested.

In this residence node, the network is modeled as two classes of users, the general internet user group and the VoIP user group.

4.2.1 General Internet Users

The general internet user group is modeled as a 100BaseT workstation group, where the total number of user in this group can be configured to anywhere from 1 to 100. This network group is used to represent the already existing internet users running other internet application such as HTTP, FTP, Printing and other servers defined by corresponding profile setting.

4.2.2 VoIP User

For the proposed VoIP users, there are two topologies: VoIP over the LAN network and over the wireless LAN network. They are shown in figures 4, and 5 respectively.

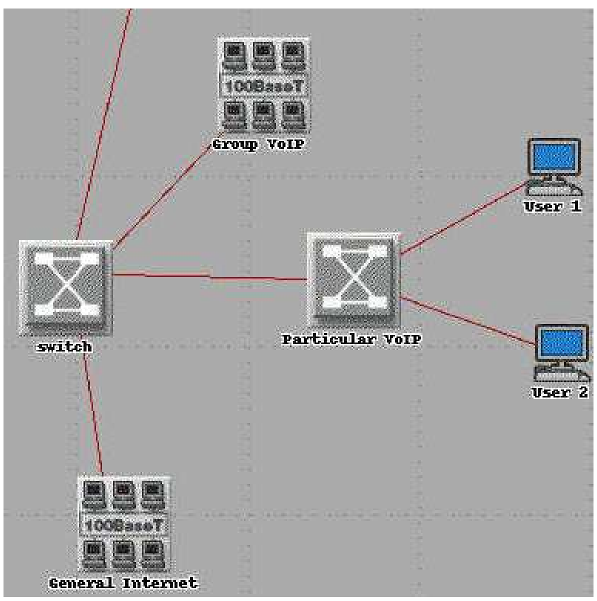

Figure 4 Resident Building Sub-Node with VoIP by LAN

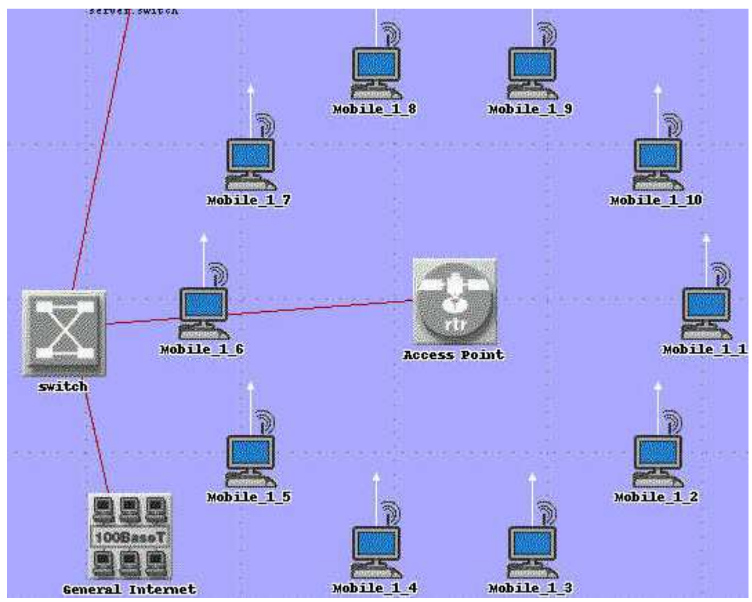

Figure 5 Resident Building Sub-Node with VoIP by Wireless LAN

4.2.2.1 VoIP over LAN Network

The proposed VoIP user terminals are modeled as two separate entities; one of them includes two identical workstation running VoIP profile only and sharing a common switch, and a 100BaseT workstation group representing the rest of the VoIP users as a group. It is modeled in such a way that we can easy trace the call between two local terminals (from User 1 to User 2).

4.2.2.2 VoIP over Wireless LAN Network

Another topology for the VoIP terminals implementation is by wireless LAN workstations, where VoIP users are modeled as wireless terminals running VoIP profiles identical to LAN users, but are communicating to the main switch and server through wireless networks access point. The wireless access point is modeled as a single station with 802.11b speed or 11Mbps bandwidth [8]. Each wireless node is located equally from each other and is 100 meters away from the access point. Also, the access point is configured to have 0.85W of transmitting power.

All workstations and switches are inter-connected by 10Mbps Ethernet cables.

4.3 Topology of Server Node

The server node is the backbone of the entire campus network, where internet request and application services are processed and managed, and where the proposed VoIP server will be implemented. Also, within the server node is the main campus router used to provide internet protocols traffic (IP traffic) between the internet campus wide network and the external network.

Figure 6 shows the sub-node model of the server node.

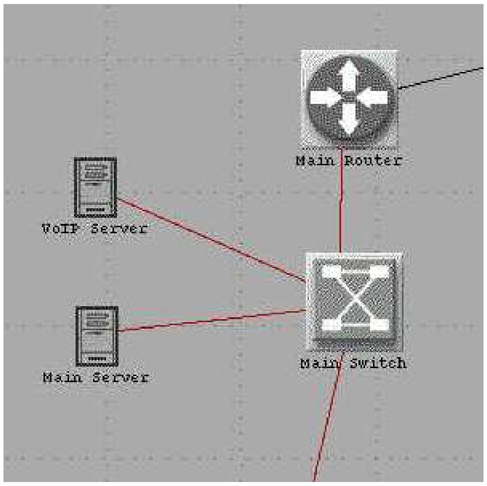

Figure 6 Server Sub-Node Model

In this node, all existing general internet services are provided by the server named “other service”, and its supported applications are configured to support all other services including HTTP, FTP, Printing and others.

The proposed VoIP services are supported by “VoIP server”, and it is configured to support only the VoIP applications.

“Main Switch” is used to router the internal internet traffic between servers and workstations across the campus network.

“Main Router” is used to manage the internet traffic exchange between the internal traffic within the campus and the external network outside the campus.

All network connections between switch, server and router are 10Mbps.

4.4 Application Attributes Configuration

Application attributes configuration is used to setup application setting for different supported applications and services running over the campus network terminals and servers. In this case, there are 5 kinds of applications supported by the campus network, and they are HTTP, FTP, Printing, Email and proposed VoIP services.

The figure 8 illustrates the setup for the application attributes.

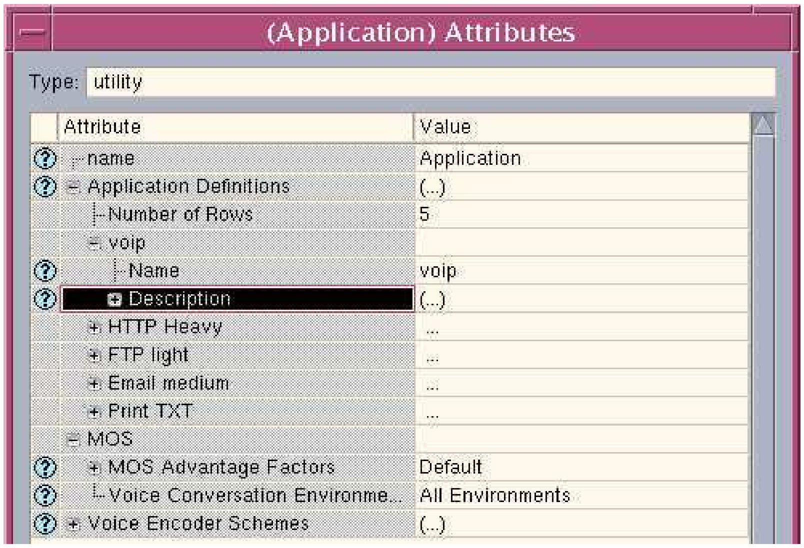

Figure 7 Application Configuration

4.5 Profile Attributes Configuration

The profile attributes configuration is used to set different user profile for different terminals or servers running different services. In this case, there are two profiles being used: General services profile which represents HTTP, FTP, Email and Printing application services, and VoIP profile which represents VoIP application only. Each network terminals such as LAN users, Wireless LAN users and servers will be configured to either run or support one of the profile settings according to their designated simulation purpose.

The figure 9 illustrates the setup for the profile attributes.

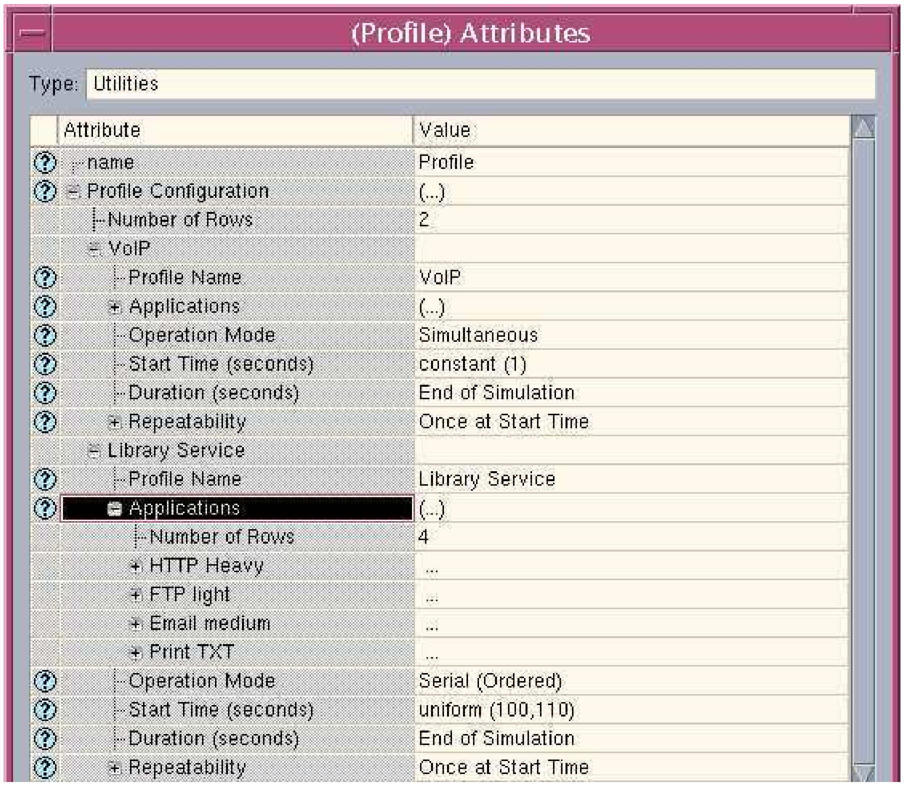

Figure 8 Profile Configuration

5. Scenarios Simulations and Results

To test out our proposal of implementing VoIP network on the resident building within the existing campus network, and to evaluate the performance of such a network and the possible impact on the already existing internet services, scenarios will be created and compared to gain further insight. In those scenarios, parameter(s) will be manipulated to simulate different network condition, configuration and other parameters, and results will be compared between performance identifiers including End-to-End Delay, Jitter, MOS and Packet Loss Ratio.

5.1 10 Different Background Traffic Load

In this scenario, various background traffic pattern loads are introduced into the network connection between the resident building node and the server node, where such traffic simulates the different internet activities running between other areas within the CAN. Because the network link between the server node and the resident building node are at 10Mbps, four background traffic patterns are created to mimic no traffic load (0Mbps), light traffic load (5Mbps, or 50% load), heavy traffic load (8Mbps, or 80% load), full traffic load (9.8Mbps, or 98% load).

The following figures show the End-to-End Delay, Jitter and MOS. Note that there is no packet loss during all four scenarios. In those figures, blue line represents the full load, red line represents the heavy load, green line represents the light load and cyan represents the no load.

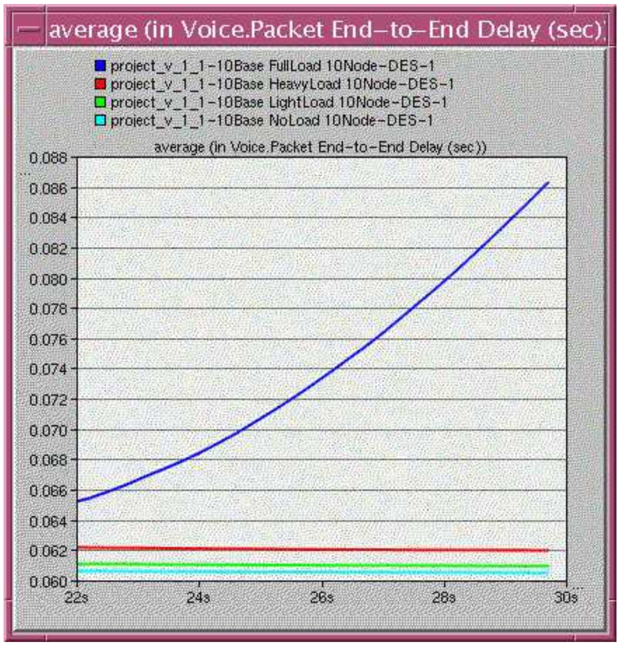

Figure 9 End-to-End Delay of Different Background Traffic Load Setting

As we can see from the figure above, end to end delay increase as the background traffic increases. When the background traffic increases from 0% to 50%, we see a very small increase of delay time, because network still has plenty of bandwidth to be used for VoIP. However, as the background traffic increases from 80 to 95, we see a very large increase in end-to-end delay and the LAN is becoming unstable, because network is close to its full capacity, and are not capable of handling much more traffic. So packets have to be queued for delivery, thus a much larger delay overall.

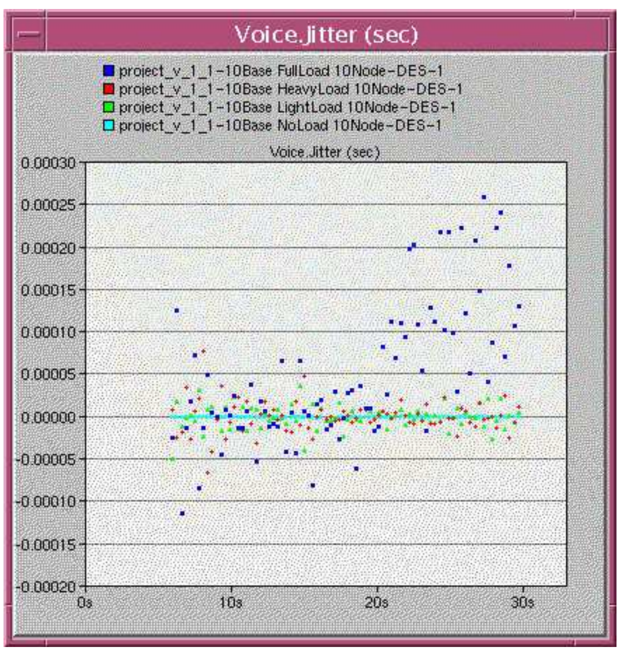

Figure 10 Jitter of Different Background Traffic Load Setting

From the figure above, we can clearly see that as the background traffic load increases, jitter increases as well.

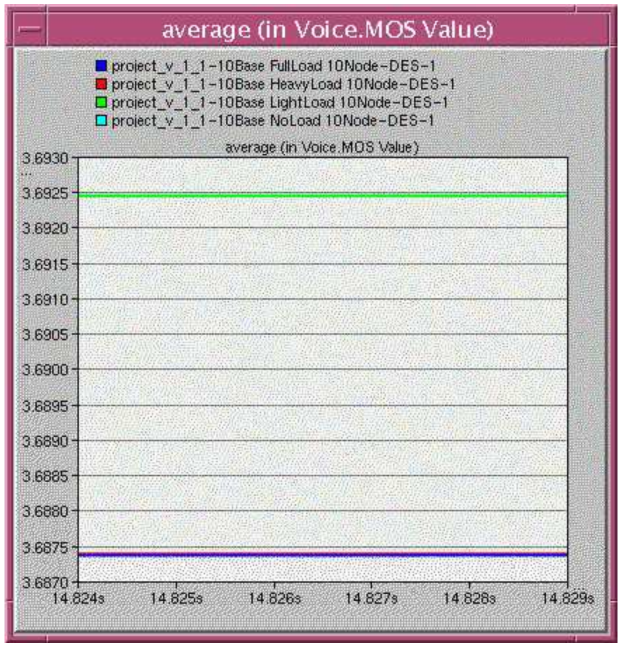

Figure 11 MOS of Different Background Traffic Load Setting

Figure 12 shows that MOS decreases as the background traffic increases.

From all three figures above for this scenario, it is conclusive that as background traffic increases, both end-to-end delay and jitter increases, and this leads to a decreased MOS. It means that VoIP quality of service is affected by the network conditions within the local network, such that background traffics generated by other applications and clients will greatly affect how VoIP perform.

5.2 10 vs. 30 VoIP Users under Heavy Traffic Load

In the previous scenario, we implemented a LAN network where only 10 VoIP clients are connected. However, more realistically, there will be more students using VoIP service at the resident building, so this scenario will compare the effect of having more VoIP clients (30 clients vs. 10 clients). Also, we find that background traffic will cause a longer end-to-end delay and larger jitter, so to simulate the background traffic more closely to reality, in this scenario, we will assume that the background traffic is at its busy state, where 80% of the total bandwidth is used elsewhere within the campus network serving other internet applications.

Following figures are packet end-to-end delay, jitter and MOS for the comparison of 10 vs. 30 VoIP clients over LAN under heavy traffics.

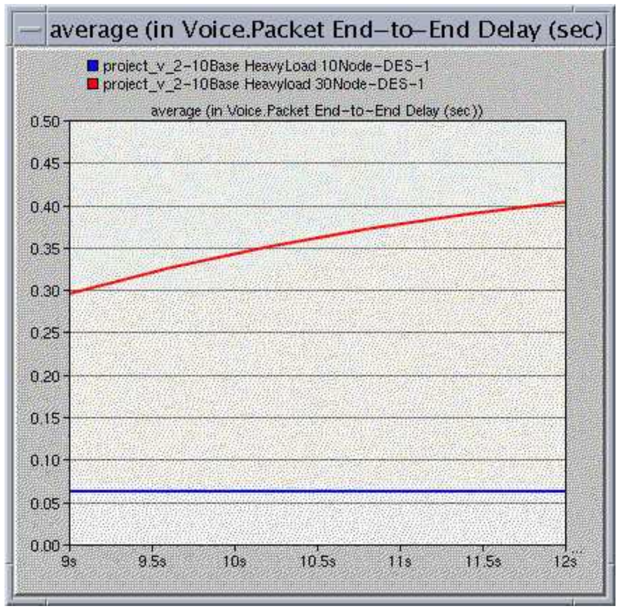

Figure 12 End-to-End Delay of 10 vs. 30 VoIP Users over LAN under Heavy Traffic Load

As we can see the figure above, when there are 30 VoIP enabled clients connected to the network, the end-to-end delay is much larger. The delay time is can be greater than 0.4 seconds. This is not an acceptable end-to-end delay time and will make VoIP conversation intolerable.

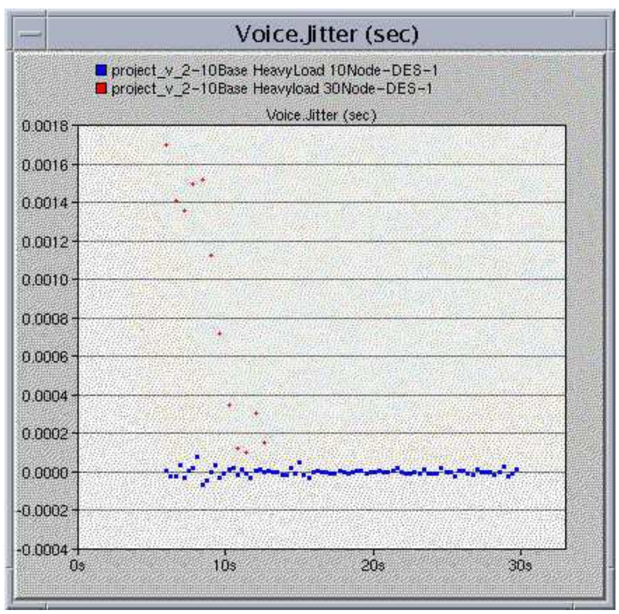

Figure 13 Jitter of 10 vs. 30 VoIP Users over LAN under Heavy Traffic Load

From figures above, it is clear that if 30 VoIP clients are using the network instead of 10 clients, the jitter becomes very large due to the fact that at assumed busy background traffic, bandwidth is limited, and all VoIP clients' packets need to be queued and delay time is greatly varied. This means the sound quality is very jumpy and uneven, thus an unacceptable voice quality.

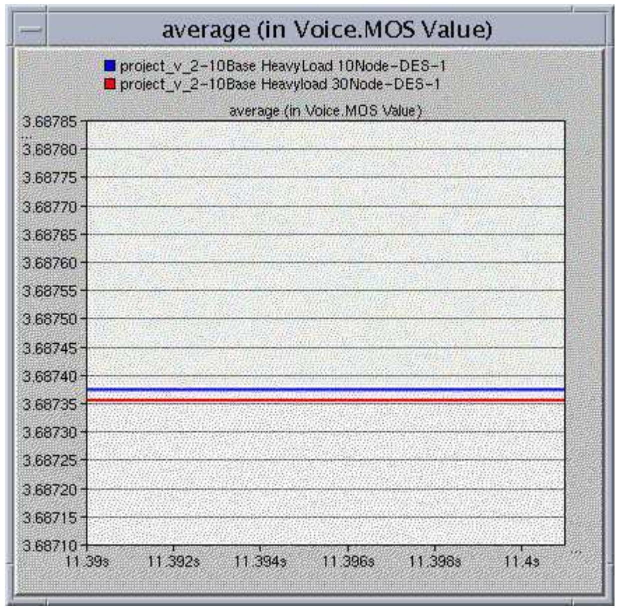

Figure 14 MOS of 10 vs. 30 VoIP Users over LAN under Heavy Traffic Load

As we can see, when more VoIP clients are using the LAN network, there is a lower MOS due to the over congestions, especially when the background traffic is heavy and busy.

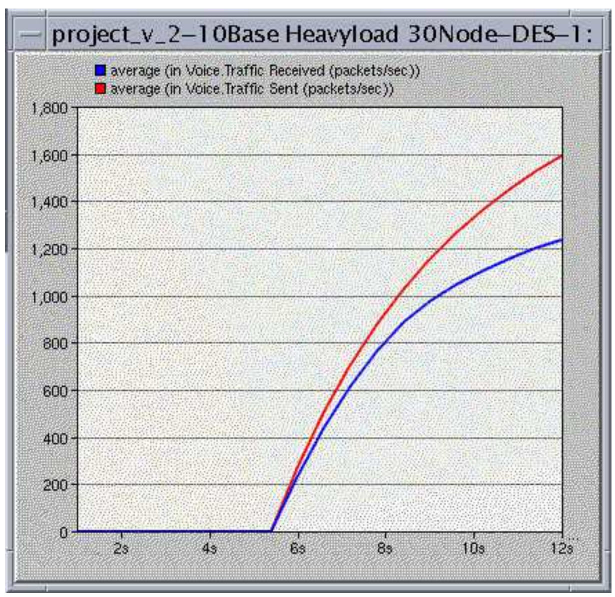

Figure 15 Packet Loss of 30 VoIP Users over LAN under Heavy Traffic Load

Seen from figure above, when there are 30 VoIP clients connected to the LAN network, there is a large packet loss due to traffic congestion. For instance, at some point, over 1600 packets/sec are send but only over 1250 packets/sec are received, this mean at many VoIP packets are dropped or re-transmitted due to long delay and ultimate time-out.

Clearly, as the number of VoIP clients increases, the total end-to-end delay increases as well because more packets are transmitting at the same time to compete for any available bandwidth. The jitter increases as well because many packets have to wait to be transmitted or possibly re-transmitted due to time-out, thus high packet loss. So the variation in time delivery is large. Overall, larger VoIP clients will drop the service quality and requires a better connection (larger bandwidth) between the nodes such as the switches and the server.

5.3 10 VoIP over LAN vs. by WLAN (11Mbps) under Light vs. Heavy Traffic

In the previous cases, we are implementing VoIP over LAN, where terminals are fixed and immobile. However, WiFi, or wireless LAN is more convenient, and students are more likely to use their laptops to wirelessly connect to the access point for internet service. VoIP on WiFi also has the potential to provided better coverage, and signal quality then cell phones indoors. [9] Therefore, in this scenario, we will implement VoIP application over wireless network, and compare the performance against that of over LAN network. For both VoIP over LAN and wireless LAN, we will also consider the background traffic load as a simulating parameter.

Figures below show the packet end-to-end delay, jitter and MOS, where red line represents VoIP over wireless LAN with heavy background traffic, cyan represents the VoIP over wireless LAN with no background traffic, blue represents VoIP over LAN with heavy background traffic, and green represents VoIP over LAN with no background traffic.

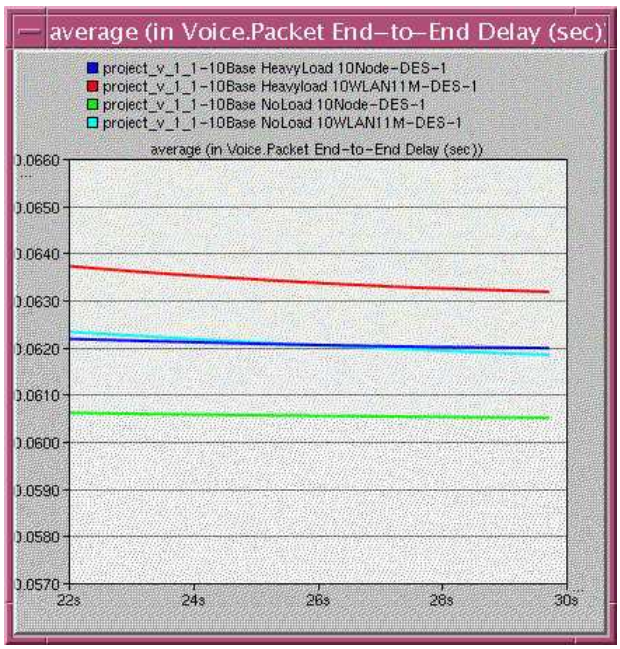

Figure 16 End-to-End Delay of VoIP over LAN vs. WLAN under No vs. Heavy Traffic

From this figure, we can clearly see that VoIP over wireless network generally has a much higher end-to-end delay. We can also see that the effect of having a heavy load background traffic increase the end-to-end delay is identical for both LAN and wireless LAN network.

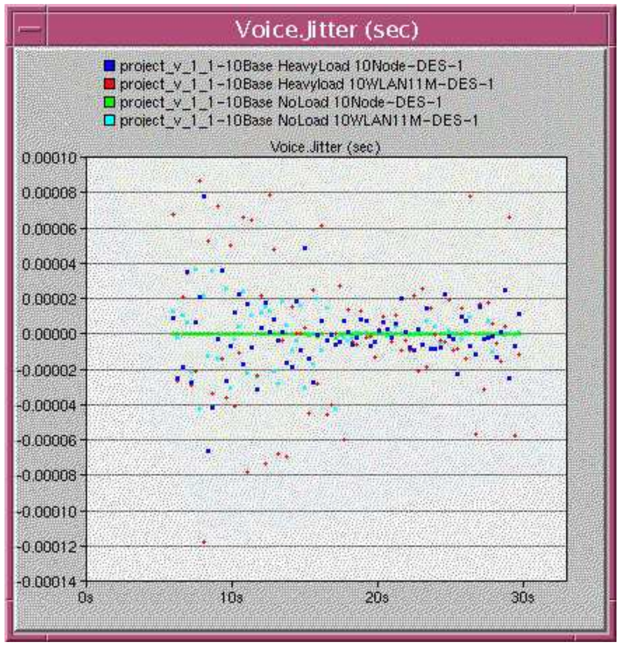

Figure 17 Jitter of VoIP over LAN vs. WLAN under No vs. Heavy Traffic

Comparing between VoIP over LAN and wireless LAN, it is clear that VoIP over wireless network has a much larger jitter and thus worse quality. Comparing between VoIP over wireless LAN with no background load and heavy background traffic load, VoIP with heavy traffic load has a much larger jitter than that of without background traffic load. Such finding coincides with that of VoIP over LAN.

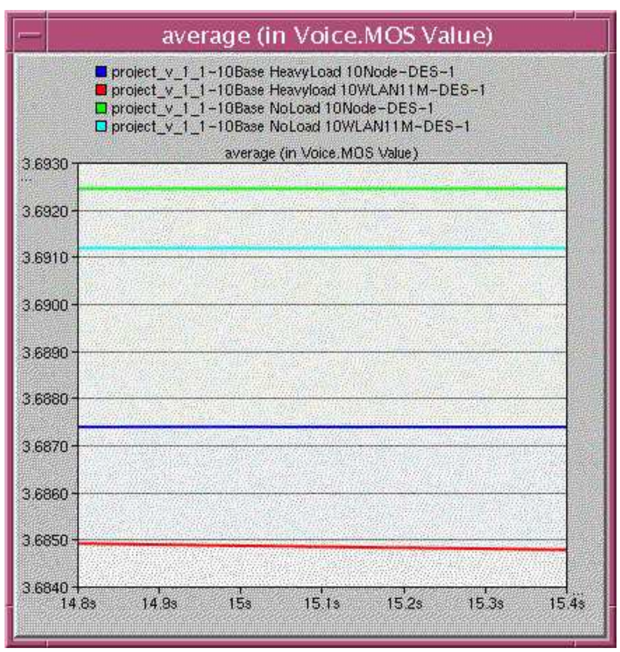

Figure 18 MOS of VoIP over LAN vs. WLAN under No vs. Heavy Traffic

As we can see, MOS is not affected much by VoIP over LAN or over wireless LAN, with no background traffic load or with heavy background traffic load.

Clearly, VoIP over wireless LAN has a higher end-to-end delay and jitter, and lower MOS than that of VoIP over LAN. It is also affected by the background traffic conditions, such that heavy background traffics will increase the end-to-end delay and jitter even more and decrease the MOS as well.

5.4 10 vs. 30 VoIP Users by Wireless LAN under No Traffic

In this scenario, we will investigate how an increased number of VoIP clients over wireless network will change the performance of the system. We will compare between a single wireless access point with 10 wireless clients and with 30 wireless clients, and in all cases, both access points and clients will have bandwidth of 11Mbps.

Figures below show the packet end-to-end delay, jitter and MOS, where red line represents the VoIP over WLAN with 30 clients, and blue line represents the VoIP over WLAN with 10 clients.

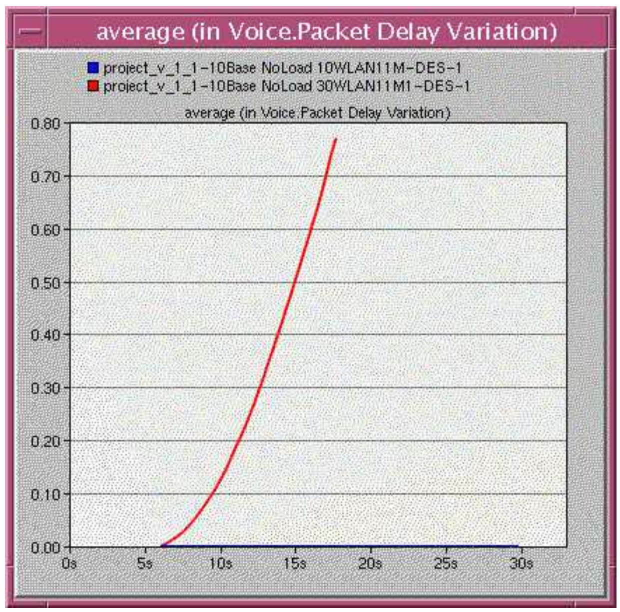

Figure 19 End-to-End Delay of 10 vs. 30 VoIP Clients over WLAN under No Traffic

As we can see, VoIP over WLAN with 10 users has a smooth and stable packet delay, because the wireless network is capable of handling all the VoIP traffic. However, in case of 30 VoIP clients connecting through WLAN, network became unstable and end-to-end delay time because infinitely large because a single access point cannot provide enough bandwidth for all 30 VoIP clients. Because in both cases, there is no background traffic, it means that 30 VoIP clients over WLAN used up all the bandwidth of the internet resources available.

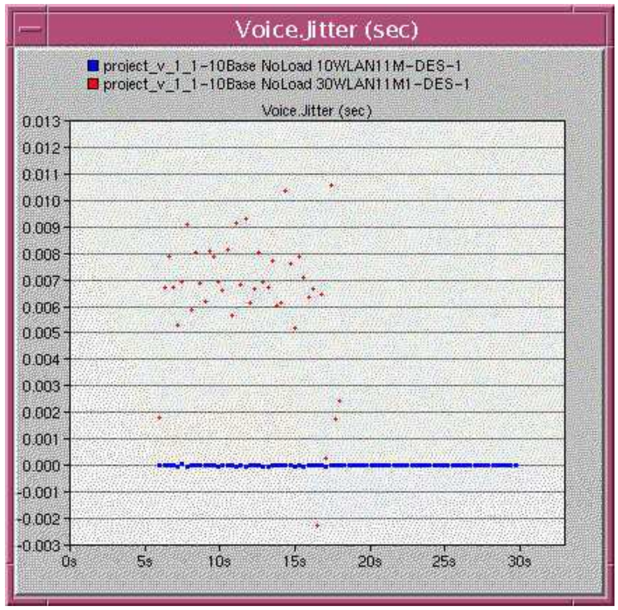

Figure 20 Jitter of 10 vs. 30 VoIP Clients over WLAN under No Traffic

From this, we see that jitter of 30 VoIP clients over WLAN is so large that sound quality may become intolerably bad. On the other hand, 10 VoIP clients over WLAN yield a much smaller jitter.

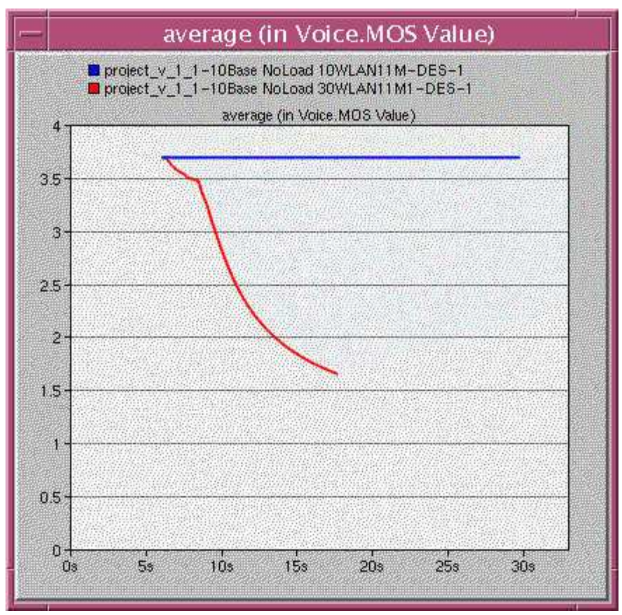

Figure 21 MOS of 10 vs. 30 VoIP Clients over WLAN under No Traffic

As we can see, when 30 VoIP clients are connecting through a single access point, MOS becomes so low, meaning that it is impossible to communicate between users. This is due to the fact that wireless access point is full and VoIP packets are queued and timed out so they cannot be delivered at a timely manner.

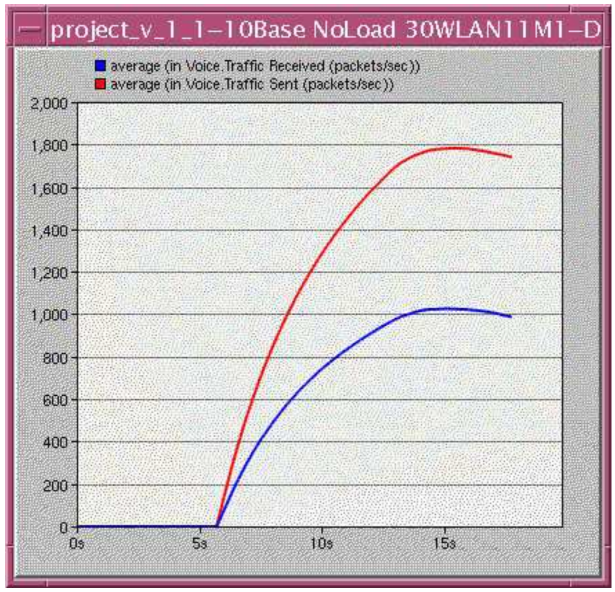

Figure 22 Packet Loss 30 VoIP Clients over WLAN under No Traffic

From the figure, clearly shown is that packet lost is evident. As we can see, at a point, about 1800 packets/sec are being send, and only 1000 packets/sec are received, and this represents a 44% packet lost, and this is why when 30 VoIP clients connects through a single access point, traffic is so congested that virtually no traffic can pass through to be useful, and sound quality degrades to worst possible because packets are being delivered with large variation of time differences (large jitter), and overall packets delivery time is very large (large end-to-end delay). Therefore, MOS value is very low, and such VoIP network is unacceptable or useful. It means that to ensure a usable VoIP experience, number of wireless clients must be limited according to the total bandwidth allowed for the particular access point. Either multiple wireless access points or higher connecting speed for the access point needs to be provided to provide the same quality of services as by LAN.

5.5 Audio Sampling and Compression Standard

5.5 Audio Sampling and Compression Standard

In all previous scenarios, different network topologies are compared such as number of clients and internet access method, however, there is also another factor determining the overall quality of sound for the VoIP service, which is the codec used to sample and compress voice. In this scenario, 10 VoIP clients will be using both PCM and GSM quality of VoIP audio quality for comparison over a LAN connection.

The following figures shows the packet end-to-end delay, jitter and MOS for both PCM and GSM standard VoIP codec over a 10 clients enabled LAN network with no background traffic load.

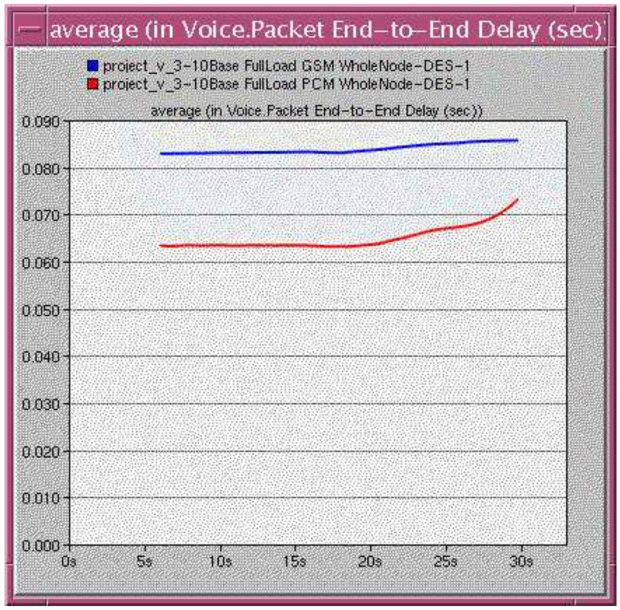

Figure 23 10 End-to-End Delay of 10 VoIP Users over LAN with Heavy Background Traffic Load Using PCM vs. GSM Codec

As we can see, generally speaking, GSM has a higher packet end-to-end delay than that of PCM codec.

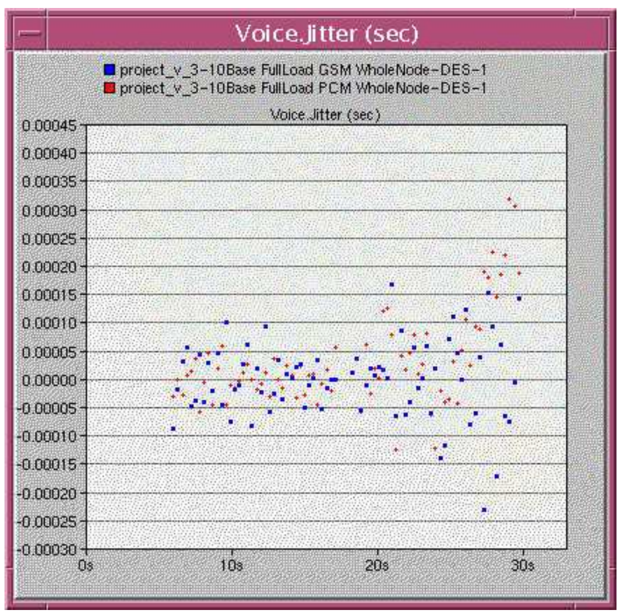

Figure 24 Jitter of 10 VoIP Users over LAN with Heavy Background Traffic Load Using PCM vs. GSM Codec

From figure above, it is clear that PCM codec has much higher jitter than that of GSM.

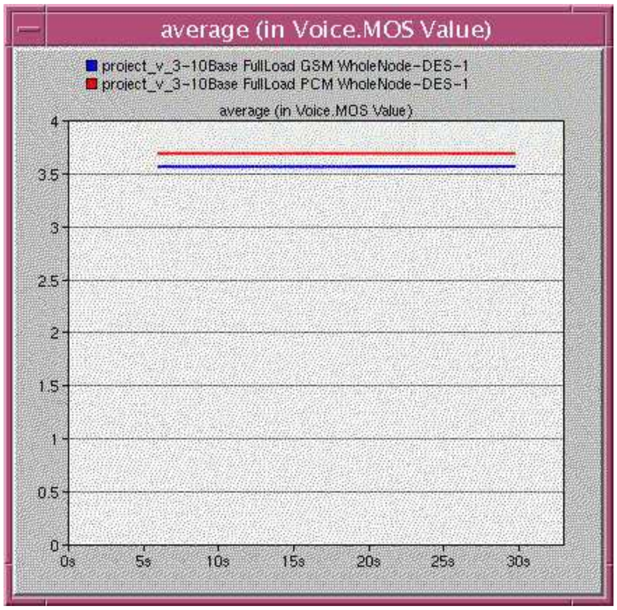

Figure 25 MOS of 10 VoIP Users over LAN with Heavy Background Traffic Load Using PCM vs. GSM Codec

From the figure above, we can see that overall speaking, PCM codec has a higher MOS than that of GSM codec, and it means that VoIP users will find PCM codec with more pleasant audio conversation.

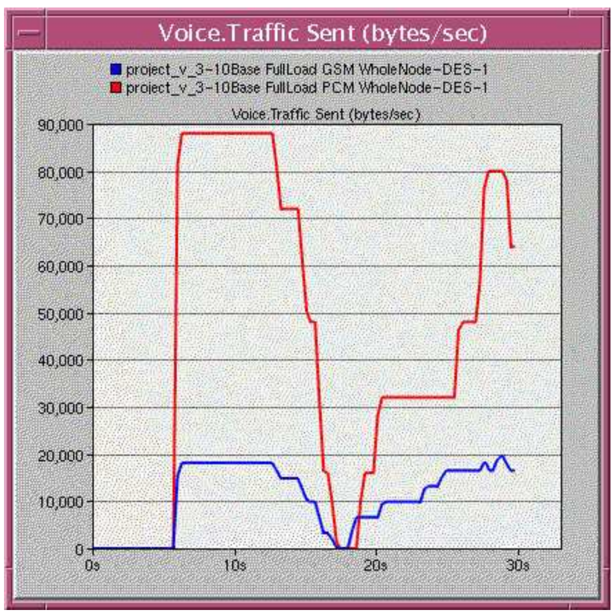

Figure 26 Traffic Sent of 10 VoIP Users over LAN with Heavy Background Traffic Load Using PCM vs. GSM Codec

PCM (G.711) has an average bit rate of 64kpbs[15], and GSM full rate (GSM FR) has an average bit rate of 13kbps[16], and this means that PCM has a bit rate about 5 times of that of GSM. And our figure above confirms that.

Clearly, Although PCM has slightly better MOS and lower delay, it uses about 5 times more bandwidth than that of GSM, and it means that GSM audio codec is a better codec.

6. Conclusion

From this lab, we implemented a campus width network simulating the proposed implementation of VoIP service for the resident building, and gradually expanding it into full campus width implementation. Many scenarios are created for various comparisons, and table below summarizes the important statistics associated with each scenarios:

Scenario Number	Scenario Description				Packet End-to-End Delay (s)	Jitter	Packet Loss Ratio	MOS
1	LAN Network	10 Nodes	No Background Traffic		0.0608	< 0.00001	Minimal	3.6925
2			Light Background Traffic		0.0612	< 0.00003	Minimal	3.6925
3			Heavy Background Traffic		0.0621	< 0.00008	Minimal	3.6874
4			Full Background Traffic		> 0.0871	< 0.00027	Minimal	3.6874
5		30 Nodes	Heavy Background Traffic		> 0.4100	< 0.0017	> 25%	3.6873
6	Wireless LAN Network	10 Nodes	No Background Traffic		0.0605	< 0.00004	Minimal	3.6912
7			Heavy Background Traffic		0.0633	< 0.00009	Minimal	3.6848
8		30 Nodes	No Background Traffic		> 0.7800	< 0.1100	➢ 44%	< 1.700
9	LAN Network	10 Nodes	Heavy Background Traffic	PCM Codec	>0.0871	<0.00027	Minimal	3.6874
10				GSM Codec	>0.0750	<0.00035	Minimal	3.7300

Comparing Scenario 1, 2, 3 and 4, we can see that end-to-end delay and jitter increases while MOS drops. Comparing scenario 3 and 5, we can see that having more nodes in a LAN network increase end-to-end delay and jitter and decreases MOS. Comparing scenario 1 and 5, and 3 and 7, we can see that while all others held the same, wireless network has a higher end-to-end delay and jitter, and lower MOS. Comparing scenario 5 and 8, we can see that at both 30 nodes inside the network, wireless network gives a much worse QoS over LAN network. Comparing scenario 5, 8 and the rest of other scenarios, we can see that packet loss rarely occur when only a few nodes are implemented, even when the network condition is not ideal (heavy traffic); however, as soon as over 30 nodes are implemented inside the network, packet loss becomes dominant and evident that end-to-end delay and jitter is so large that MOS drops to minimal possible level. Also, wireless network is much more susceptible to background traffic load and large number of users. From scenario 9 and 10, it is evident that the choice of audio codec is also a significant factor.

7. Future Work

7.1 Project Limitations and Difficulties

In this project, many assumptions and simplifications are made to reduce the complexity of the project. Because simulation time is very slow for larger number of nodes, we are limited to have 30 nodes for both LAN and WLAN topology. Also because we assume a campus wide network, we did not take into account of external network traffic conditions that provide internet service to the campus. Another pitfall in the project is that we assumed all traffic other than VoIP traffic is described as traffic load on the connection between server node and resident building node, and this is a very accurate model.

7.2 Project Improvement in the Future

If time permits, we would like to implement the following:

1. Simulate the network model so that all traffics including the background traffics are realistically generated by profile setting and according to some traffic flow statistics.

2. Implement network stations with more number of nodes, and are being connected by multiple level of switches instead of single main switch.

3. In the case of wireless network, having multiple access point across the c campus network and with mobile workstations being mobile such that each node is moving according to some defined trajectory.

4. Add an external VoIP node outside the campus network, and simulating some background traffic load on the connation between the campus network and out-of-campus networks.

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