Abstract

VoIP (Voice over Internet Protocol) phone systems have been gaining popularity in businesses as replacements of existing PBX (Private Branch Exchanges). These VoIP systems usually rely on the Ethernet structure readily available in a business facility. The emergence of wireless LAN (local area network) is also gaining immense popularity due to the additional mobility given to the users.

In this project, a PBX-style VoIP system employed over an 802.11b network is simulated. Three performance factors were analyzed. They are end-to-end delay, delay jitter, and packet loss. The variable parameters of the system included the speed of the wireless network, the type of voice encoding (G.711, G.723, G.729, etc.), and the number of stations in the network.

1. Table of Contents

2. Introduction

VoIP (Voice over Internet Protocol) phone systems have been steadily gaining popularity in the office space. Businesses are swapping their old PBX (Private Branch Exchange) systems with new IP (Internet Protocol) based systems. The main advantage of IP based systems is the ability to utilize bandwidth more efficiently by encoding voice data into small packets and transmitting the packets in a high-speed data network.

As VoIP technology matures, customers want more functionality. One of the most popular features is mobility. To add mobility to a VoIP system, 802.11b is the natural choice of technology because it easily extends the connectivity of existing Ethernet data networks. 802.11b provides wireless connectivity with speeds of up to 11Mbps. A simple overview of 802.11b wireless networks is included in Section 3.

The main goal of this project is to analyze the 3 main QoS (Quality of Service) measurements in a Voice over 802.11b system. They are end-to-end delay, delay-jitter, and packet loss. These measurements are explained in Section 6. The parameters in a Voice over 802.11b network which affect these QoS values include the connection speed of the network, the type of voice encoding chosen, and the number of stations in the network. These parameters are explained in Section 7.

The project was implemented in ns-2. Details of the implementation are included in Section 8. Section 9 of this report provides the simulation results and analysis. Lastly, the results are summarized in the conclusion in section 10.

2.1. Acronyms

AP	- Access Point
CSMA/CA	- Carrier Sensing Multiple Access / Collision Avoidance
IEEE	- Institute of Electrical and Electronics Engineers
ITU	- International Telecommunication Union
MAC	- Medium Access Control

Mbps	- Mega-bits per second
PBX	- Private Branch Exchange
PHY	- Physical Layer
QoS	- Quality of Service
VoIP	- Voice over Internet Protocol

3. 802.11b Wireless Networks

802.11b is a standard published by the IEEE (Institute of Electrical and Electronics Engineers) specifying the MAC (Medium Access Control) and PHY (Physical Layer) procedures of a wireless local area network. It is an extension to IEEE's 802.11 standard. The basic principle behind 802.11b is using the 2.4GHz frequency range as a shared medium between wireless stations. Each wireless station employs a CSMA/CA (Carrier Sensing Multiple Access with Collision Avoidance) scheme that allows the stations to communicate with each other on the shared medium. For details of the CSMA/CA mechanism, please refer to [1].

802.11b networks offer two modes of operation: Managed and Ad-hoc. In a managed network, an AP (Access Point) serves as an administrative node controlling admission of stations onto the network. In ad-hoc mode, all stations are free to communicate with each other as peers.

Two main advances of 802.11b over 802.11 are the availability of faster data rates (up to 11Mbps) and the optional use of short preambles. The four available data rates are 1Mbps, 2Mbps, 5.5Mbps, and 11Mbps. The use of short preambles reduces the transmission time of each packet by 96μs by reducing the time required by the sender and receiver to synchronize with each other.

4. Voice Packetization

To transmit voice data over a data network, the voice data must first be digitized and then compressed into small units known as packets. The compression algorithm (usually referred to as codec) determines the size and the transmission interval of these packets. In order to transport these voice packets over the network, they are encapsulated as RTP (Real-time Transport Protocol) packets using UDP (User Datagram Protocol) as the transport protocol. These protocols incur an overhead of 40bytes for each packet sent. The following figure illustrates the packet format.

IP Header	UDP Header	RTP Header	Voice Data
20 bytes	8 bytes	12 bytes	size depends on codec

Figure 1 - Format of a voice packet

Some commonly used codecs are G.711, G.723, and G.729a. These codecs are defined by the ITU (International Telecommunication Union). The following table summarizes their characteristics.

Table 1 - Packetization size and rate for commonly used codecs

	Voice Data Size	Header Size	IP Packet Size	Packet Interval
G.711	240 bytes	40 bytes	280 bytes	30 ms
G.723	24 bytes	40 bytes	64 bytes	30 ms
G.729a	20 bytes	40 bytes	60 bytes	20 ms

5. Voice over 802.11b Network

Combining the 802.11b wireless network with packetized voice, the following network results:

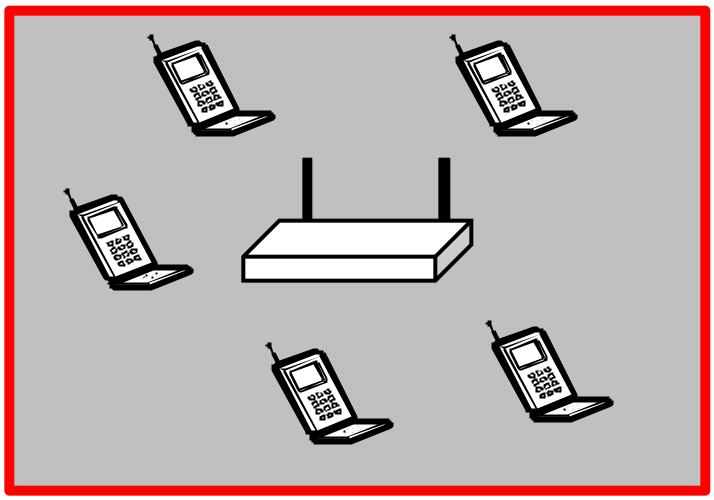

Figure 2 - Voice over 802.11b Network

This network setup is the basis of this project. Within this network, each wireless station is allowed to communicate directly with each other in voice conversations. In addition, 1 station may communicate with the AP to allow a call to the "outside world".

6. Quality Measurement

Speech quality in general is a very subjective measurement. Fortunately, in a VoIP system, there are 3 main objective measurements that can be used. They are end-to-end delay, delay-jitter, and packet loss rate.

End-to-end delay is a measurement of the average time needed to transport the voice data from the sender to the receiver. In a conversation, one-way end-to-end delay of less than 100ms is unnoticeable. When the delay becomes more than 150ms, users will perceive the quality as poor. As a guideline, the ideal requirement for end-to-end delay is 50ms.

Delay-jitter is a measurement of the variation in the arrival times of the voice packets. Such variations are caused by the queuing involved in the data path. For ideal quality, the delay-jitter should be less than 20ms. Delay-jitter of above 60ms starts to cause problems at the receiver and hence degrades the quality of service.

Packet loss rate is a measurement of how many voice packets fail to reach the receiver. In general, the voice quality depends largely on the ability of the encoding algorithm to recover; however, a general guideline suggests 1% loss as a maximum for toll quality conversation. Packet loss rate of higher than 5% is unacceptable.

These requirements of the QoS measurements are summarized in table below.

Table 2 - Requirements for Quality Measurement

	Average Quality	Ideal Quality
End-to-end Delay	< 150 ms	< 50 ms
Delay-Jitter	< 60 ms	< 20 ms
Packet Loss Rate	< 5%	< 1%

7. Parameters and Expected Results

The main focus of this project is to simulate the effects of various network parameters on the main QoS indicators (delay, jitter, and loss). The parameters tested are the speed of the network, the codec used for voice compression, the number of stations in the network, and the use of short preambles.

7.1. Transmission Speed

By testing each of the 4 available rates of the 802.11b standard (1Mbps, 2Mbps, 5.5Mbps, and 11Mbps), we expect to see all of the QoS indicators (delay, jitter, and loss) improve as the rate increases. By improvement, it means end-to-end delay would be reduced, delay-jitter would be reduced, and packet loss rate would be reduced.

7.2. Voice Codec

The choice of voice codec determines the rate and size of the transmitted voice packets. Both packet rate and packet size would affect the QoS indicators. It is expected that with a higher packet rate, a higher loss rate would be observed. As for packet size, it is expected that a larger size would yield longer delay and higher loss rate.

7.3. Number of Stations

Since the 802.11b network medium is shared amongst all stations, the number of stations in the network adversely affects the performance. It is expected that with more stations in the network, the lower the performance will be, i.e. end-to-end delay would be higher, delay-jitter would be higher, and packet loss rate would also be higher.

7.4. Short Preambles

The use of short preambles reduces the transmission time of each packet by 96μs. With this reduction in transmission time, the end-to-end delay should also be reduced.

8. Implementation

The project was implemented using the ns-2 simulator (version 2.26). The implementation is split up into three portions: protocol implementation, simulation scripting, and post-processing. The work involved in each portion is approximately 15% for protocol implementation, 45% for simulation scripting, and 40% for post-processing.

8.1. Protocol Implementation

Protocol implementation involves adding the 802.11b extensions to the already existing 802.11 implementation in ns-2, as well as verifying and fixing some of the 802.11 code in ns-2. The MAC/802_11 object in ns-2 was extended to allow the setting of the 802.11b data rates (1Mbps, 2Mbps, 5.5Mbps, and 11Mbps). It was also extended to allow the use of short preambles during transmission. Affected ns-2 files are:

/mac/mac-802_11.cc

/mac/mac-802_11.h

/tcl/lan/ns-mac.tcl

/tcl/lib/ns-mobilenode.tcl

See Appendix A for details.

8.2. Simulation Scripting

The simulation portion includes writing a main test script which sets up the wireless environment to be tested. The test script allows the specification of the following parameters:

- The network transmission rate (1Mbps, 2Mbps, 5.5Mbps, or 11Mbps)
- The number of wireless stations

- The choice of codec (G.711, G.723, or G.729a)
- The use of short preamble (enabled or disabled)

To simulate the wireless stations and the AP in the network, the script uses the following configuration for each node:

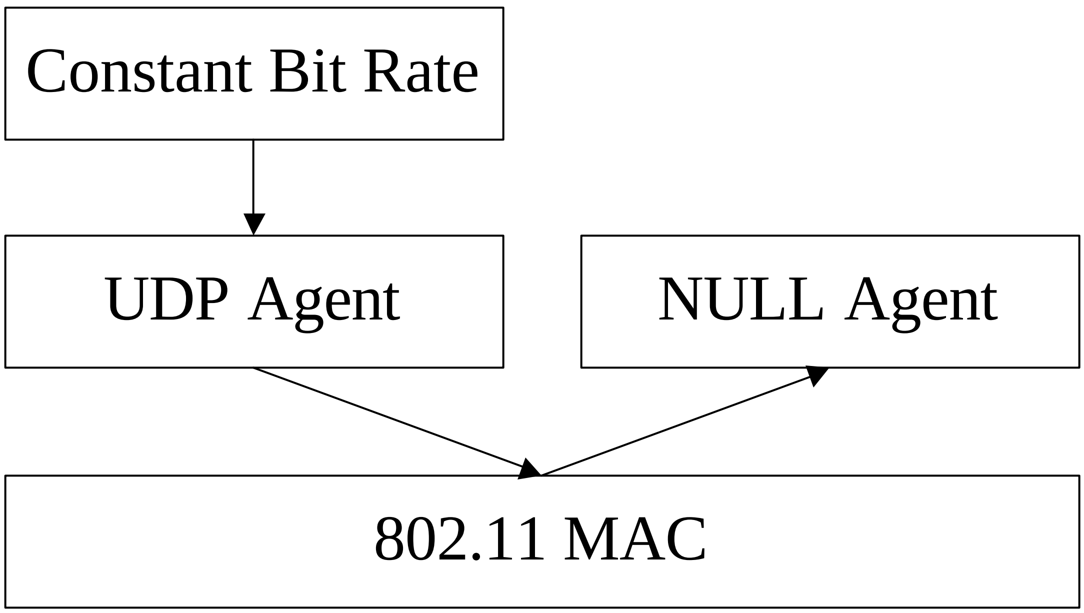

Figure 3 - Wireless Node Configuration

The CBR (Constant Bit Rate) sources are configured to behave with the desired packet size and rate of the chosen codec. The script was written in TCL and produces traffic trace files for each wireless station and the AP in the network.

8.3. Post-Processing

The post-processing portion of the project is the calculation of the QoS measurements from the trace files acquired during simulation. The three indicators are measured as follows:

End-to-end Delay:

reception time of packet #n – sent time of packet #n

Delay-Jitter:

if packet #(n-1) was received successfully

reception time of packet #n – reception time of packet #(n-1)

Packet Loss Rate:

100% * (1 - total number of packets received / total number of packets sent)

The post-processing tool was written in C and can analyze all of the different connections captured in a trace file.

9. Simulation and Results

Different scenarios were simulated to test the effects of 4 different parameters: transmission speed, voice codec, number of stations, and the use of short preambles. All of these scenarios are compared to the following base scenario:

Rate:	11Mbps
Voice Codec:	G.729a
Short Preamble:	Disabled
Number of Stations:	5

For this base scenario, the average QoS indications are observed as:

End-to-end Delay:	2.458ms
Delay-Jitter:	5.059ms
Packet Loss Rate:	0.67%

As observed in the simulation, the base scenario passes all the requirements for an ideal VoIP communication. The following figures illustrate the delay and jitter of a connection in the base scenario.

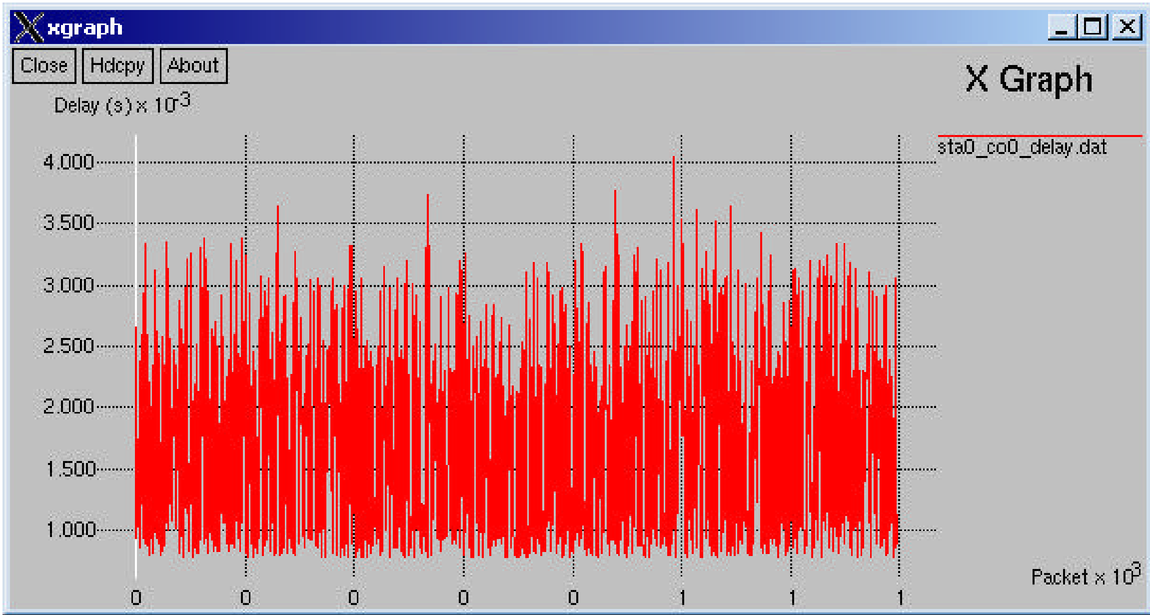

Figure 4 - Delay in base scenario (11Mbps, G.729a, Long Preamble, 5 Wireless Stations)

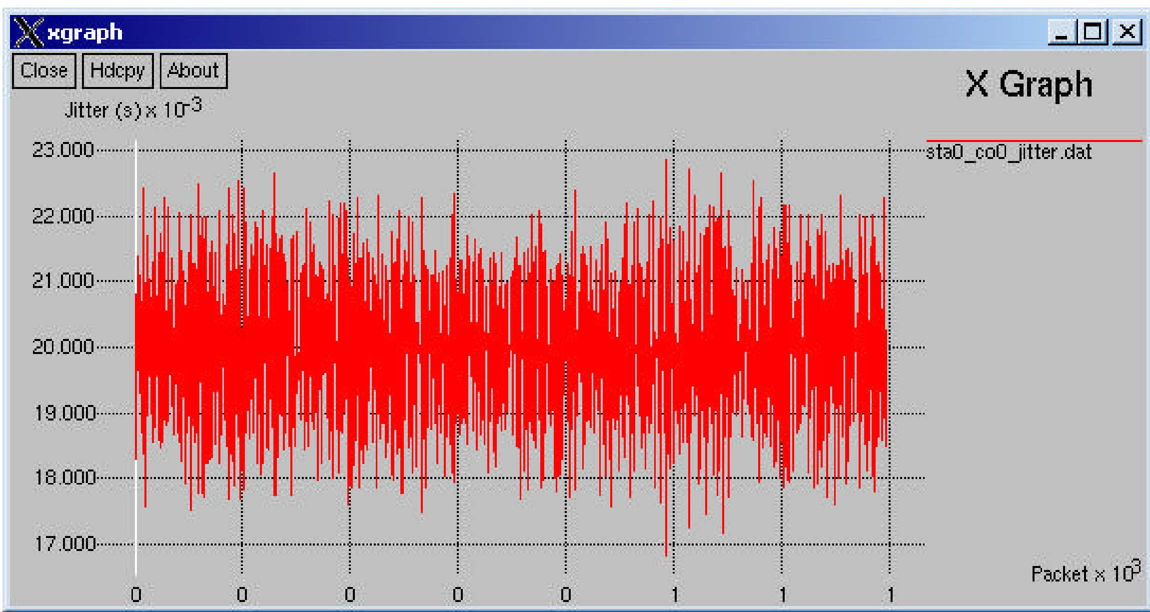

Figure 5 - Jitter in base scenario (11Mbps, G.729a, Long Preamble, 5 Wireless Stations)

The following sections illustrate the results of varying the different parameters.

9.1. Effects of Transmission Speed

In this section, different transmission speeds are evaluated. The scenarios are:

Rate: 1Mbps, 2Mbps, 5.5Mbps, and 11Mbps

Voice Codec: G.729a

Short Preamble: Disabled

Number of Stations: 5

The average QoS indications are observed as:

Table 3 – Average QoS for different transmission speed.

	1 Mbps	2 Mbps	5.5 Mbps	11 Mbps (base)
End-to-end Delay	4.537 ms	3.389 ms	2.660 ms	2.458 ms
Delay-Jitter	8.649 ms	6.348 ms	4.913 ms	5.059 ms
Packet Loss Rate	0.61%	0.65%	0.53%	0.67%

For end-to-end delay, the simulation results are consistent with the expected behaviour. The faster the speed, the shorter the delay is observed. The following figures illustrate the delay and jitter for the 1 Mbps rate.

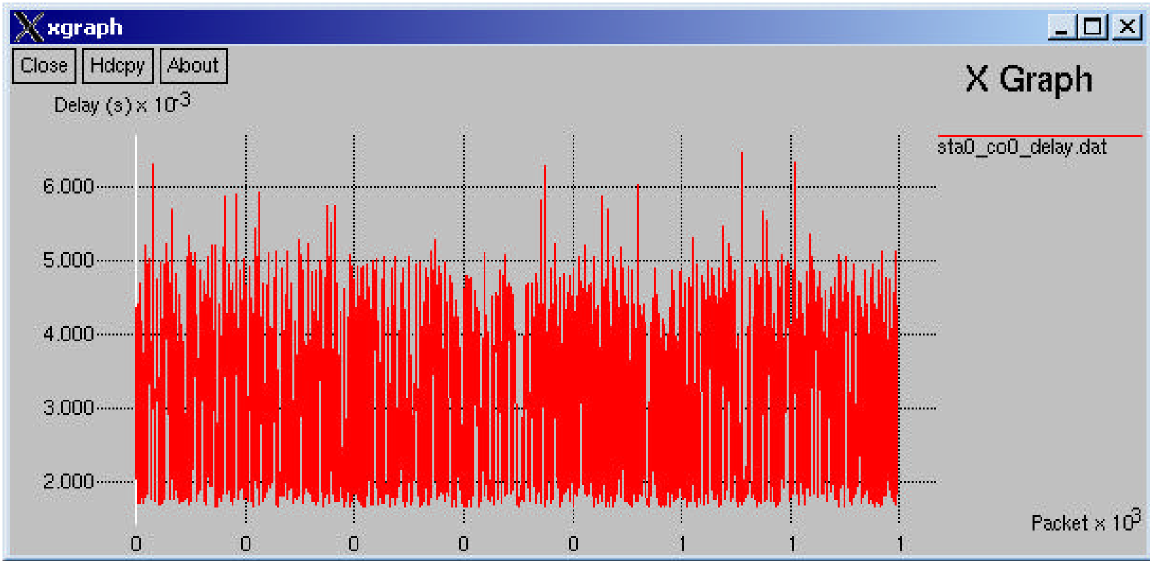

Figure 6 - Delay in base scenario (1Mbps, G.729a, Long Preamble, 5 Wireless Stations)

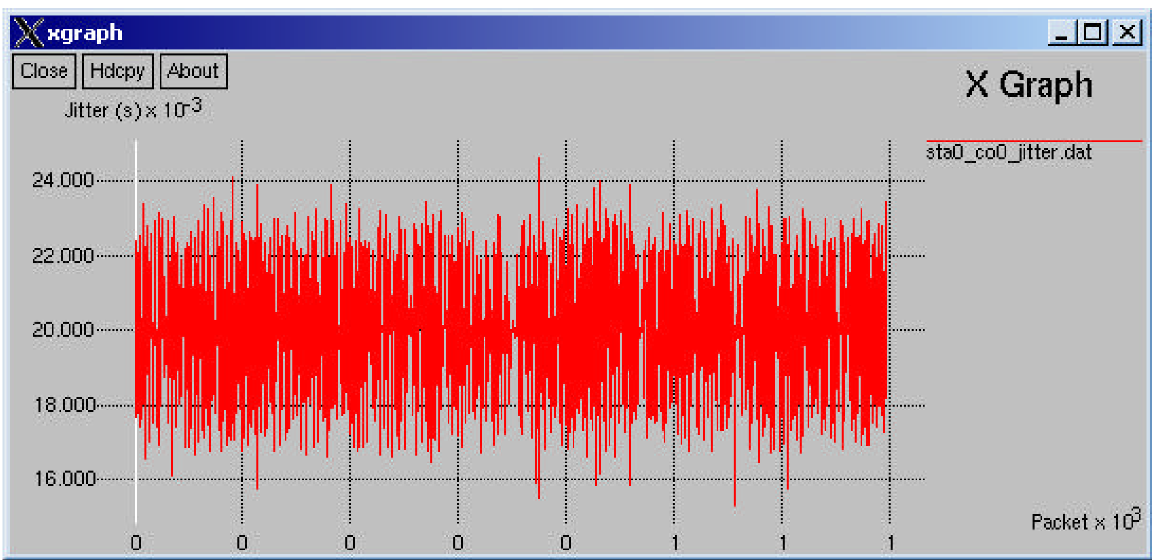

Figure 7 - Jitter in base scenario (1Mbps, G.729a, Long Preamble, 5 Wireless Stations)

9.2. Effects of Packetization Rate and Size

In this section, different voice codecs are evaluated. The scenarios are:

Rate: 11Mbps

Voice Codec: G.711, G.723, and G.729a

Short Preamble: Disabled

Number of Stations: 5

The packetization size and rate for these codecs are listed below.

Table 4 - Packetization size and rate for commonly used codecs

	Voice Data Size	Header Size	IP Packet Size	Packet Interval
G.711	240 bytes	40 bytes	280 bytes	30 ms
G.723	24 bytes	40 bytes	64 bytes	30 ms
G.729a	20 bytes	40 bytes	60 bytes	20 ms

The average QoS indications are observed as:

Table 5 – Average QoS for different voice codecs.

	G.729a (base)	G.723	G.711
End-to-end Delay	2.458 ms	2.464 ms	2.803 ms
Delay-Jitter	5.059 ms	4.865 ms	5.193 ms
Packet Loss Rate	0.67%	0.63%	0.63%

For end-to-end delay, the simulation results are consistent with the expected behaviour. The larger packet size of G.711 yields a higher delay. The lower packet rate of G.723 results in a slightly lower jitter. The following figures illustrate the delay and jitter for G.723.

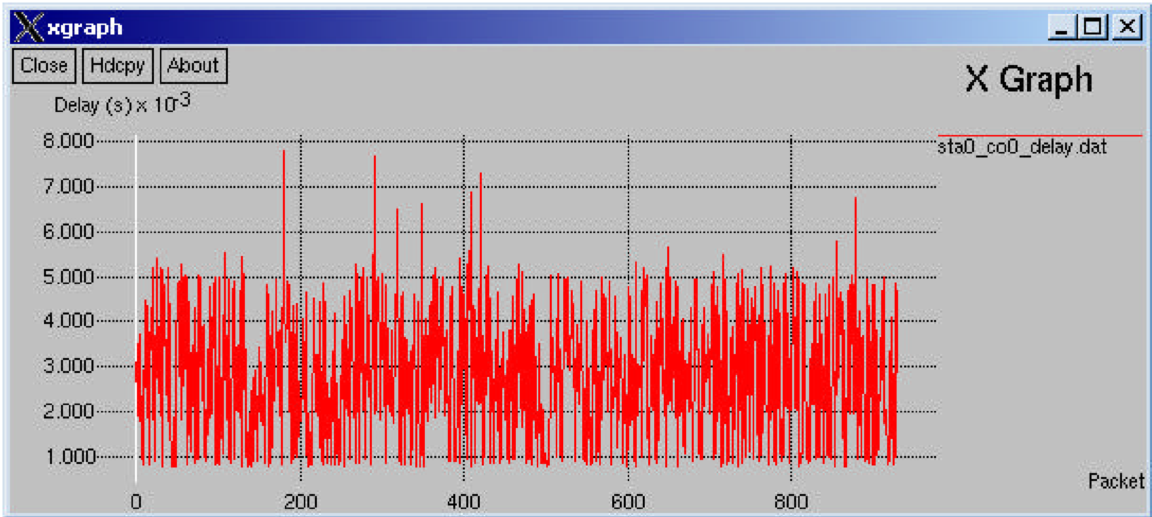

Figure 8 - Delay in base scenario (11Mbps, G.723, Long Preamble, 5 Wireless Stations)

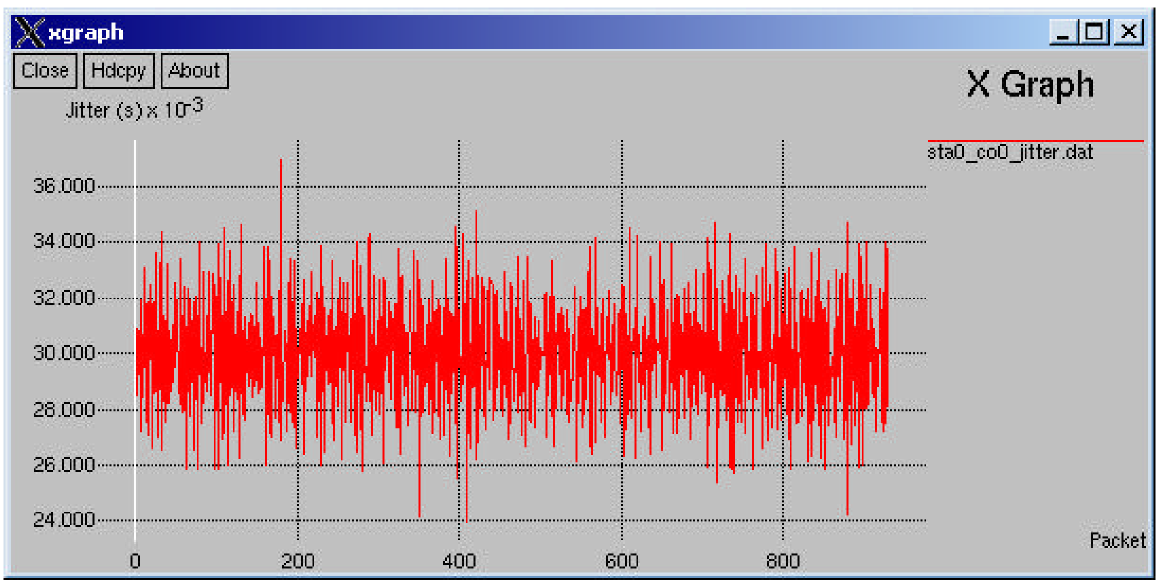

Figure 9 - Jitter in base scenario (11Mbps, G.723, Long Preamble, 5 Wireless Stations)

9.3. Effects of Number of Stations

In this section, the effects of the number of wireless stations are evaluated. The scenarios are:

Rate: 11Mbps

Voice Codec: G.729a

Short Preamble: Disabled

Number of Stations: 3, 5, 7, and 9

The average QoS indications are observed as:

Table 6 - Average QoS for different voice number of stations.

	3 stations	5 stations (base)	7 stations	9 stations
End-to-end Delay	1.726 ms	2.458 ms	3.323 ms	4.286 ms
Delay-Jitter	3.999 ms	5.059 ms	6.974 ms	11.087 ms
Packet Loss Rate	0.50%	0.67%	0.97%	1.47%

The simulation results are consistent with the expected behaviour. The higher the channel count, the worse performance becomes in all 3 QoS indicators.

9.4. Effects of Short Preambles

In this section, the effects of using short preambles are evaluated. The scenarios are:

Rate: 11Mbps

Voice Codec: G.729a

Short Preamble: Disabled, Enabled

Number of Stations: 5

The average QoS indications are observed as:

Table 7 - Average QoS for the use of short preambles.

	Long Preamble (base)	Short Preamble

End-to-end Delay	2.458 ms	1.683 ms
Delay-Jitter	5.059 ms	3.945 ms
Packet Loss Rate	0.67%	0.68%

The simulation results are consistent with the expected behaviour. The use of the short preamble results in much shorter delay and smaller delay-jitter. The following figures illustrate the delay and jitter for using short preambles.

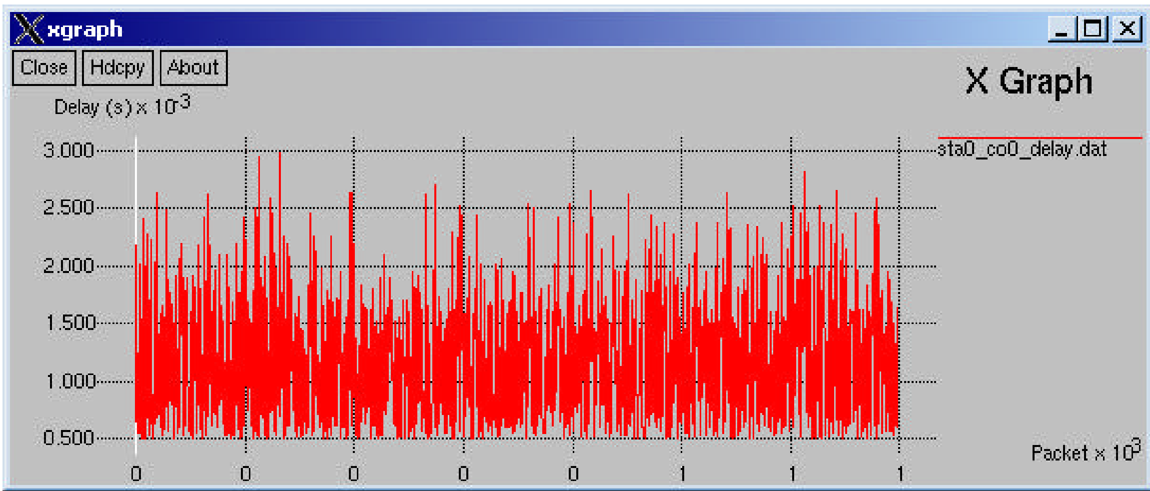

Figure 10 - Delay in base scenario (11Mbps, G.729, Short Preamble, 5 Wireless Stations)

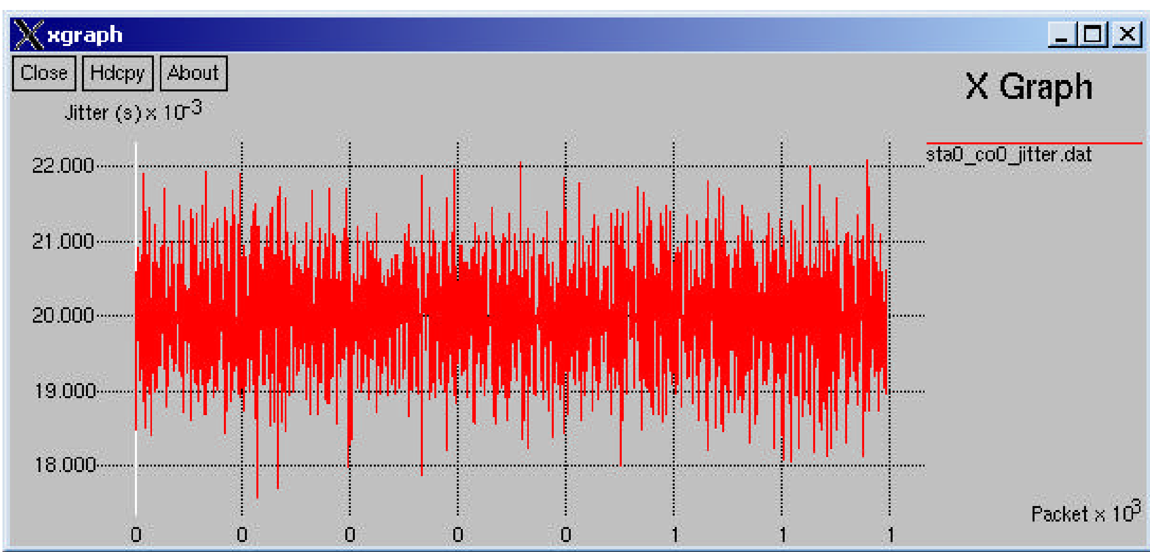

Figure 11 - Jitter in base scenario (11Mbps, G.729, Short Preamble, 5 Wireless Stations)

10. Conclusion

From all the simulation results, it was successfully shown that the expected effects of the parameters are correct.

To summarize:

- The end-to-end delay is shorter the faster the transmission speed is.
- The larger packet size of G.711 yields a higher delay.
- The lower packet rate of G.723 results in a slightly lower jitter.
- The higher the channel count, the worse performance becomes in all 3 QoS indicators.
- The use of the short preamble results in much shorter delay and smaller delay-jitter.

In conclusion, the project was a success in evaluating all of the effects of the different parameters of the network.

11. Future Work

To further extend this project, newer protocols such as 802.11g and 802.11e can be evaluated. Also, the Point Coordination Function of the 802.11 specifications should be fully implemented in ns-2 to properly evaluate its effects on QoS performance.

12. References

[1] IEEE Std 802.11-1999, IEEE Standard for Local and Metropolitan Area Networks: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications: http://standards.ieee.org/getieee802/download/802.11-1999.pdf

[2] IEEE Std 802.11b-1999, Supplement to IEEE Standard 802.11, 1999 Edition: Higher-Speed Physical Layer Extension in the 2.4 GHz Band: http://standards.ieee.org/getieee802/download/802.11b-1999.pdf

[3] H. Schulzrinne et al., "RTP: A Transport Protocol for Real-Time Applications," RFC 3550, IETF, July 2003: http://www.ietf.org/rfc/rfc3550.txt

[4] H. Schulzrinne et al., "RTP Profile for Audio and Video Conferences with Minimal Control," RFC 3551, IETF, July 2003: http://www.ietf.org/rfc/rfc3551.txt

[5] "IP Telephony Design Guide - An Alcatel White Paper," Alcatel, 2003.

Appendix A

Code listing for mac/802_11.cc, mac/802_11.h, ns-mac.tcl, ns-mobilenode.tcl, test.tcl, postproc.c.