HIGH-PERFORMANCE PERIODIC CONTENTION-FREE MULTIPLE-ACCESS PROTOCOL FOR BROADBAND MULTIMEDIA POWERLINE COMMUNICATIONS

By

YU-JU LIN

A DISSERTATION PRESENTED TO THE GRADUATE SCHOOL
OF THE UNIVERSITY OF FLORIDA IN PARTIAL FULFILLMENT
OF THE REQUIREMENTS FOR THE DEGREE OF
DOCTOR OF PHILOSOPHY

UNIVERSITY OF FLORIDA

2004

This is dedicated to
Parents
and to
Yen-Li Sun

ACKNOWLEDGMENTS

I thank the Department of Electrical and Computer Engineering for having the best teachers. I would also like to thank LIST lab members for their support and feedback. It was a pleasure to work with them.

I also thank Intellon Corp. (Ocala, FL) forgiving me opportunities to work with them and learn about powerline communication from the inside. I especially thank Srinivas Karta for contributing useful discussions and knowledge to my research.

I am especially grateful to my parents who raisedme and are longtime supporters. Their full support and love were my stronghold in doing this research.

Finally I express my deepest gratitude to my wife, Yen-Li Sun, who has been a source of strength and inspiration over the years. I especially thank her for her support and understandings during many late nights while researching toward this dissertation.

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By

Yu-Ju Lin

May, 2004

Chair: Haniph Latchman
Major Department: Electrical and Computer Engineering

Applications over PLC networks have drawn much interest in the academic community as well as in the communication industry, not only because of inherent convenience (connecting PLC-capable devices requires no new wires), but also because almost all electrical devices have to connect to a power outlet eventually. This technology makes implementing a digital home entertainment center more realistic than ever. HDTVs in different rooms are now able to share digital content from one set-top box without rewiring, or setting up wireless access points. MP3 players can access music data through PLC networks from different rooms playing different music. PLC networks provide an enabling technology for the smart home.

However, PLC technology is still evolving and many problems remain unsolved. The hostile environment of PLC channels makes reliable data transmission difficult. The PLC channel is known for its hostile nature in transmitting electrical signals. Protocols designed for other media may not be suitable for PLC. In some ways, PLC channels are

similar to wireless channels: both of them face hidden node problems, near-far effects and other channel imperfections. However, PLC network nodes tend not to move. It is unlikely that simply applying protocols designed for another medium would result in good performance in the PLC environment; the overhead may be too high or the assumptions about noise may be too optimistic for PLC networks.

In light of PLC's unique characteristics, we developed a new protocol called: Periodic Contention-Free Multiple Access (PCF/MA). PCF/MA is an explicit R-ALOHA-like protocol specifically designed for the PLC network. We propose an RTS/CTS-like scheme in the reservation stage to mitigate hidden-node problems, and a delayed NACK mechanism to conquer the near-far effect.

Performance of the proposed protocol is evaluated by event-driven computer simulations and by mathematical analysis. Simulation results show that 85 Mbps MAC throughput with a 100 Mbps channel data rate can be obtained, even when there are hidden nodes in the network. To provide smooth video delivery, we propose a mathematical estimation of the required delay in playback time and the amount of playback buffer with tight bandwidth reservation. Our simulation shows that a 100 Mbps channel can deliver up to 9 MPEG-2 video streams simultaneously without dropping any video frames. A visual protocol analyzer was also developed as a tool to study network protocol.

CHAPTER 1
INTRODUCTION

The concept of Information Appliances (IA) became a recent reality. Many next-generation appliances come with communication capabilities with embedded processors built right into the devices. For instance, on April 7, 2001, IBM and Carrier announced that they will produce a new air conditioner with JAVA support that can send e-mails to manufacturers for errors; or the user can send commands to the air conditioner to pre-adjust room temperature. We believe that before long our homes will have many kinds of IA devices communicating among them and with the outside. Many of these IA devices are expected to have multimedia capability. Intelligent homes of the future will need multimedia communication support for these IA devices.

Providing the right infrastructure for connecting these IA devices will be a major need. For home applications, this infrastructure must be easy to setup, inexpensive to install and maintain, and must perform well. Ordinary people are not network experts, and a typical high-performance network is too complicated for casual daily usage. The supporting infrastructure should be easy to set up, and the effort needed to maintain this infrastructure should be minimal. Differences between home networks and ordinary networks can be summarized as follows:

- Home networks are different than other types of networks. The data stream in a home network is much smaller than in ordinary public networks and local area networks, because the coverage of a home network is small. With this small coverage, the expected response time is short too.

- The main purpose of home networks is to share resources. Examples, include sharing a single printer for several computers; using a single Internet connection;

sharing files (such as images, spreadsheets and documents); playing games that allow multiple users at different computers; and sending the output of a device (like a DVD player or Webcam) to other devices.

- Home networks should be easy to setup, maintain, and access. They should be affordable for most families.

Many existing networking technologies compete to support this mission. For example, a comprehensive Ethernet network can be constructed by installing UTP-5 special cabling around the house. Alternatively, wireless networks such as *802.11x*, *Bluetooth*, and *HomeRF* can be constructed by installing multiple interconnected wireless access points (WAP) and base stations within the home. However, the IA devices themselves would need wireless capabilities; and the above three infrastructures all require significant effort and cost to build up the networks externally. Phone-line networks such as *HomePNA* [1] may seem attractive, but the convenience of mobility is limited by the number of phone sockets available in a home. Hughes and Throne [2] did extensive study of other infrastructure options and technologies appropriate for a home network.

In general, for future IA applications, the amount of traffic generated by appliances and computers on a home network is still unknown. Watching digital TV while downloading data from the web should cause no video jitters over home networks. To the best of our knowledge, there is little research to report the performance impact (or at least the likelihood) for supporting multimedia-enabled IA communication over home networks. We are interested in exploring various methods to study the impact of different performance results. We are also interested in analyzing the effect on each individual data streams including continuous media data stream (that is, soft real-time traffic).

In Chapter 2, we studied the performance of multimedia over power line networks using simulation studies and actual measurements on a \emph{Homeplug} 1.0 compliant PLC network. We were particularly interested in measuring the PLC network raw data rate, TCP performance, and the performance impact when QoS support is involved. We were also interested in analyzing network performance with different traffic types, including continuous media data streams (that is, soft real-time traffic).

We first built a network simulator that generates various types of traffic, and then applied the same scenarios to a real-world PLC network and to a simulation model. We compared simulation results and real-world performance. A maximum throughput of 8.08 Mbps for UDP was obtained from our simulation, while a 6.21 Mbps TCP throughput was observed in the real-world PLC network experiment. Results show that PLC networks can successfully deliver real-time traffic concurrently with traditional data traffic.

Our contributions to this research topic also include modelling human behavior in the use of IA devices, modelling the types of traffic generated by IA devices communicating over the power line, and measuring the performance of real applications over PLC networks. We also describe a practical implementation of the PHY and MAC layers for PLC networks associated simulation results.

In Chapter 3, we discuss PLC, 802.11b and 802.11a characteristics, protocol, and theoretical performance. Based on the theoretical analysis, we conducted a series of real-world experiment and performance comparisons between 802.11b and PLC networks. We compared two technologies for implementing a home or small-office networks without new cable installation. For ease of installation, both wireless and power-line networks were well rated. The wireless network uses air and the power-line network uses the existing power cable as a transmission medium. Current field testing results show that under most circumstances, power line has more stable and reliable connections. Even when the line-of-sight distances between two stations are as long as 69 ft., PLC devices can still generate up to 4.52 Mbps throughput. From 128 testing positions, PLC has 72% (from server to mobile station) and 59.38 % (from mobile station to server) better experimental results than 802.11b.

In Chapter 4, we proposed a new Protocol: Periodic-Contention Free Multiple-Access (PCF/MA) Protocol for high-speed Power Line Communications (PLC). PLC networks have impairments similar to wireless networks, but the nodes are largely stationary. There is industrial intent to use PLC networks in the home for delivery of multimedia data, with challenging quality of service (QoS) requirements. Existing protocols for PLC, for wireless networks, or for wired networks cannot meet these challenges efficiently. Our study proposed and analyzed a new protocol designed to provide the high QoS needed for delivering multiple multimedia streams in a PLC environment. The proposed protocol, PCF/MA, addresses hidden nodes and the near-far effect; in addition to addressing the generally noisy medium. PCF/MA performance was analyzed theoretically and was simulated. Simulation showed that 85 Mbps MAC throughput is possible with a channel-data rate of 100 Mbps, even when there are hidden nodes in the network. Through mathematical modeling of buffer space required for tight-bandwidth allocation, such a network can deliver up to 9 MPEG-2 video streams simultaneously without dropping any video frames, compared to 7 video streams using Modified CSMA/CA(MCSMA/CA).

Chapter 5 discusses three common factors that influence the Voice over IP quality and future works. Our experiments show that without dedicated networks and QoS enabled schemes, the factors of distance and degree of congestion were proved to be influential on the achieved end-to-end bit rates. Many research issues still need to be addressed before we can guarantee (at least statistically) the quality of the *Interne Telephony*.

CHAPTER 2
POWER LINE COMMUNICATION NETWORK INFRASTRUCTURE FOR HOME NETWORKS

Low-voltage electrical wiring in homes has largely been dismissed as too noisy and unpredictable to support high-speed communication signals. However, recent advances in communication and modulation methodologies (as well as in adaptive digital signal processing and error detection and correction) have spawned novel media access control (MAC) and physical layer (PHY) protocols capable of supporting power line communication networks at speeds comparable to wired local area networks (LANs). In this chapter we support the use of power-line LAN as a basic infrastructure for building integrated smart homes, where information appliances (IA)—ranging from simple control or monitoring devices to multimedia entertainment systems— are seamlessly interconnected by the very wires that provide their electricity. By simulation and actual measurements using reference design prototype commercial powerline products, we showed that *HomePlug* MAC and PHY layers can guarantee QoS for real-time communications, supporting delay-sensitive data streams for smart home applications.

Home-Network Infrastructures

Many next-generation appliances are being equipped with processors featuring sophisticated communication capabilities. For instance, on April 7, 2001, IBM and Carrier announced plans to produce an air conditioner with JAVA support that can Email manufacturers regarding errors, and will allow users to remotely send commands to the unit to adjust temperatures or shut it down. Smart homes will eventually have many types of information appliances (IAs) communicating among themselves and with the outside world. Soon, many of these IA devices are expected to have multimedia capability. Supporting multimedia communication for these IA devices will be of crucial importance for the smart homes of the future.

Providing the right infrastructure for connecting these IA devices will be a major need. For home applications, this infrastructure must be easy to set up, inexpensive to install and maintain, and must perform well. Ordinary people are not network experts, and a typical high-performance network is too complicated for casual daily usage. The supporting infrastructure should be easy to set up, and the effort to maintain this infrastructure should be minimal. Many existing networking technologies compete to support this mission. For example, a comprehensive Ethernet network can be constructed by installing UTP-5 special cabling around the house. Alternatively, wireless networks such as *802.11x*, *Bluetooth*, and *HomeRF* can be constructed by installing multiple interconnected wireless access points (WAP) and base stations within the home. However, the IA devices themselves would need wireless capabilities, and the above three infrastructures all require a significant effort and cost to build up the networks externally. Phone-line networks such as *Home- PNA* [1] may seem attractive, but the convenience of mobility is limited by the number of phone sockets available in a home. Hughes and Throne [2] did extensive study of other infrastructure options and technologies appropriate for a home network.

In this chapter, we advocate direct use of existing electrical wiring and outlets as the medium for data communication within the home. Using power lines as the network infrastructure has many advantages over other technologies. First, no new wires are needed, since the IA devices will communicate over the very wires that provide their electrical power. Second, there are many access points (power sockets) in a home (4 or more per room). Currently, *Power Line Communication* (PLC) as specified by the *HomePlug* 1.0 standard [3] provides a 14 Mbps raw data rate, which is adequate for daily IA-device communication. It also has a built-in QoS protocol, making it attractive for real-time streaming applications. Finally, the cost to build a power-line network is low when compared with other technologies. For example, it was observed that the *802.11x* wireless network card has approximately the same street price as the *HomePlug* network card (about $120). It is expected that with mass production requiring no expensive RF components, the cost of the PLC cards will be about 50% less than comparable wireless cards. Moreover, the cost of a required *802.11x* base station is high (more than $250). The *100 Base T Ethernet* has the highest performance/cost ratio, but requires new cables and expensive installation. Table 2-1 shows costs and other characteristics of home-network technologies. Installation costs (which are high for 10/100 BT) are not shown.

Table 2-1: Technology comparison

Technology	Media	Data Rate(Mbps)	QoS Support	Cost($)
10 Base T	UTP	10	No	20
100 Base T	UTP	100	No	80
Bluetooth	Wireless	1	Yes	5
HomeRF 2.0	Wireless	10	Yes	110
802.11x	Wireless	11	No	125
HomePNA 2.0	Phone line	10	No	80
HomePlug	Power line	15	Yes	120

From a marketing perspective, less-expensive and easier-to-use PLC home networks are becoming more attractive, and the potential market is huge. The Yankee Group (Boston, MA) estimates that at least 21 million households in the United States are interested in home networking and that 12.4 million would like to implement in-home networks within the next year. According to Parks Associates (Arlington, VA), 30 million households in the United States will have fast internet connections by 2004, and 17 million of them plan to have home networks.

In the past, power lines were considered unacceptable for signal transmission, since the channel contained a lot of noise, interference, and fading. However, the appeal of using the existing power line as a transmission medium for data exchange was too great to be ignored. The advancement of signal-modulation technologies, digital signal processing, and error control coding [4] has minimized the restrictions of channel imperfections; and high-speed signal transmission through power lines is now feasible.

Using the existing power-line infrastructure as the medium for supporting IA communication requires a careful design of the overlaid communication systems in order to provide acceptable communication services. It is desired, for example, that when watching digital TV while downloading data from the web, there will be no delay-jitter in the video quality. Current research shows that the maximum raw data rate of first generation PLC is about 14 Mbps. However, the effective data rate is expected to be around 10 Mbps after compensating for impairments and error corrections. On the other hand, research is currently underway to develop PLC chips that operate at 100 Mbps with average throughput 30-60 Mbps.

In this chapter, we investigate the performance of multimedia over power-line networks using simulation studies and actual measurements on a *Homeplug* 1.0 compliant PLC network. We are particularly interested in measuring the PLC network raw-data rate, TCP performance, and the performance impact when QoS support is involved. We are also interested in analyzing the network performance with different traffic types, including continuous-media data streams (that is, soft real-time traffic).

We first built a network simulator that generates various types of traffic, and then applied the same scenarios to a real-world PLC network and to a simulation model. The performance comparison between the simulation results and real-world performance are given in this chapter. A maximum throughput of 8.08 Mbps for UDP was obtained from our simulation, while a 6.21 Mbps TCP throughput was observed in the real-world PLC network experiment. Results show that PLC networks can successfully deliver real-time traffic concurrently with traditional data traffic.

Our contributions to this research topic also include modelling human behavior in the use of IA devices, modelling the types of traffic generated by IA devices communicating over the power line, and measuring the performance of real applications over PLC networks. We also describe a practical implementation of the PHY and MAC layers for PLC networks as well as associated simulation results.

The next section discusses background for implementing power-line communication networks, typical applications of power line communication networks, and the human behavioral model using these appliances. Section 2.3 presents the physical limitations of power line channels, and describes a practical signal-modulation scheme and MAC protocol as used in the *HomePlug* 1.0 protocol proposed by the *HomePlug Powerline Alliance*. Section 2.4 compares simulation results with real-world PLC network performance. Finally, Section 2.5 offers conclusions and suggestions for future work.

Applications over Power Line Communication

Power Line Communication Application in a Home

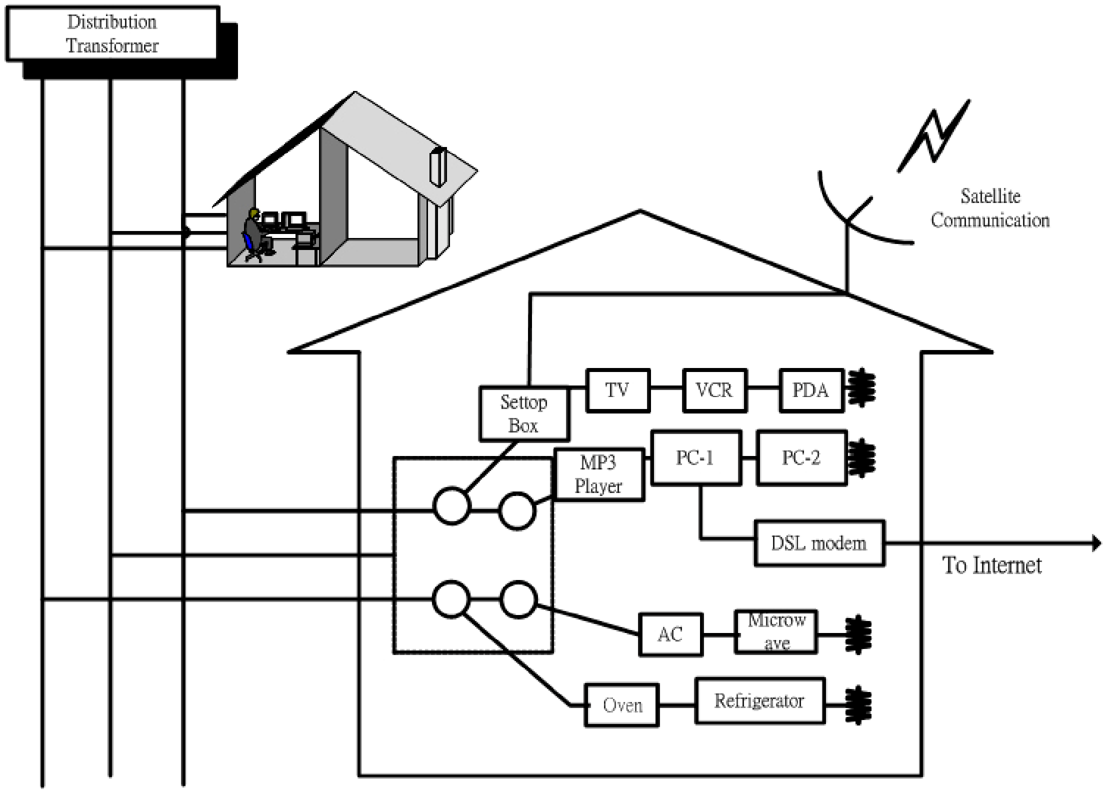

Figure 2-1: Power-line topology in a North American home

Traditionally, power lines are used for conveying electrical power to devices. Power lines were not designed for delivering high frequency signals, and so the electrical and frequency response requirements of a power line are not as critical as those of data network cabling. The poor quality of a power line is not ideal for signal transmission because the channel contains noise and interference. The medium is made of different conductor types; therefore a variety of characteristic impedances will be encountered. Further, the network terminal impedance will tend to vary with frequency and time as the consumer's load pattern and load types vary. Impedance mismatch causes a multi-path effect resulting in deep notches at certain frequencies. These channel imperfections make signal modulation over a power line difficult [5]. However, the advancement of signal modulation and error control coding techniques now make power line communication possible.

The common power line topology of a North American home is shown in Figure 2-1. The figure shows a tree-like power-line topology in a house. Typically, there are two power line trunks: one is 110V and the other is 220V. Each power-line trunk can be divided into several branches. Power-line communication aims to transmit data packets over these branches and trunks. The topology of the power line network and the convenience of its power sockets as potential access points make it a good candidate for smart home IA device networking.

Table 2-2 shows the results of a survey from which we inferred usage and traffic patterns generated by typical IAs. The table also suggests some current and future PLC applications. For instance, when merchandize is advertised on a digital TV service, the product information (such as the barcode or webpage) can be downloaded to your computer through a power line. Afterwards, you can send your order information from the computer to the supplier, or you can use the downloaded URL to browse the product web page and get more details. We also anticipate the ability to record music or videos through a power line. For example, when a song is broadcast on TV or a music channel, you can download the song directly to an MP3 player through the power line. Another application is the opportunity to record digital video directly into a PC or even a digital VCR. Other applications of IA's can be easily accomplished using a PLC network. For example, a refrigerator can order food through the power line network according to its inventory, or it can send cooking instructions to the microwave. A smart oven can send predicted environmental temperature information to the air conditioner through a power line, allowing the air conditioner to pre-adjust the temperature and keep rooms comfortable.

Table 2-2: Application traffic amount in a home

Row No.	From node	To Node	Estimated data size	Frequency	Possible time period
1	Refrigerator	Microwave	160 bytes	2 times a period	7:00-9:00,11:00-1:00, 17:00-19:00,21:00-23:00
2	Microwave	AC	72 bytes	2 times a day	7:00-9:00,11:00-1:00, 17:00-19:00,21:00-23:00
3	TV	Refrigerator	750 bytes	3 times a day	11:00-1:00,17:00-23:00
4	TV	VCR	11KBytes	3 times a day	11:00-1:00,17:00-23:00
5	TV	Computer	360 bytes	3 times a day	11:00-1:00,17:00-23:00
6	TV or Settop box	PDA or MP3 player	15 Mega bytes	3 times a day	11:00-1:00,17:00-23:00
7	Computer	PDA or MP3 player	50 Mega bytes	1 time a day	11:00-1:00,17:00-23:00
8	Computer	Computer	60 MB to 180 MB	1 time a day	6:00-24:00
9	Settop box	Computer	320 MB to 640 MB	1 time a day	11:00-1:00,17:00-23:00
10	Computer	Internet	44 MB to 131 MB	1 time a day	11:00-1:00,17:00-23:00
11	VCR	Computer	320 MB to 640 MB	1 time a day	6:00-24:00
12	Front door camera	Computer	110 MB to 1100 MB	3 times a day	6:00-24:00

The applications over PLC are not only for novel IA devices. PLC as a home network facilitates data exchange between traditional data processing devices such as PCs and computer peripherals. IA devices that talk with PCs are also possible. For example, sending multimedia data from TVs or VCRs to PCs can be easily done by PLC network, but is difficult with other infrastructure technologies. Home security can also be implemented by PLC so that a digital camera installed on the front door can send video to the TV.

Table 2-2 also gives an estimate of the daily traffic volume generated by typical IA applications. These values are based on likely information size. For example, the instruction size that the refrigerator sends to the microwave in Row 1 is estimated by the number of steps required to cook the food (1 byte), the cooking time for each step (4 bytes for each step), the power level for each step (2 bytes), and the packet header size. Added together, the entire instruction size is 160 bytes. Row 7 exemplifies storing digital music from a computer to an MP3 player. The 50 Mbytes traffic volumes is calculated from the number of songs in an album, the length of a song (5 min), the encoded data rate (128 kbps), and the packet header size. The frequency and time period during which each event occurs are also shown. By using this data and typical household dynamics for concurrent events, we can generate a traffic flow for the power-line network for a typical day.

Internet Bridging

Currently, in-home PLC networks rely on other technologies to send data to the Internet and communicate with mobile devices. Most of the homes in the United States will eventually be equipped with broadband connections like DSL or cable modem services. To share the broadband Internet connection with PLC capable devices, we can add a PLC Internet router to the PLC network. One possible setup is shown in Figure 2-2.

In this figure, a desktop computer acts like a data center. Devices that need to communicate with other devices on the Internet will send data to the desktop PC via the power line. The desktop PC decides whether to send it to the Internet. In the future, an IA routing device may be unnecessary. Researchers are developing a solution to make PLC home networks talk directly with other homes, power plants and the Internet using the external distribution power line. Such a network infrastructure for the Internet access would be especially attractive to developing countries, since no additional expenditure is needed for data network infrastructures.

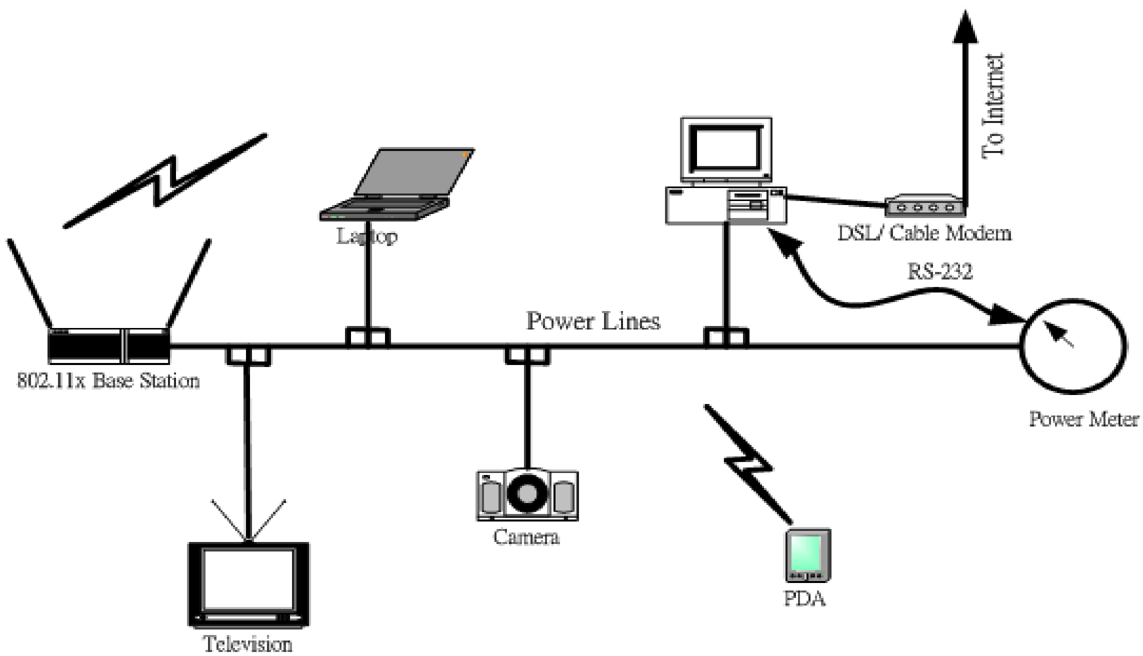

Figure 2-2: An example of using one of the computers as the PLC, DSL or Cable Modem router

To support data exchange with mobile devices, PLC networks will also need to cooperate with wireless networks. The easiest way to achieve this is to make the 802.11x base station PLC compatible. The base station is treated as an ordinary IA device with a PLC chip built in (see Figure 2-2). Mobile devices with wireless capability can then talk to devices attached to the power line. This is especially ideal when communication is desirable but large coverage areas require multiple interconnected wireless access points (the interconnection is then provided with "no new wires" using the existing power line infrastructure which would be needed to power the WAPs in any event).

The above PLC applications require a properly designed protocol. In addition, to make the PLC network real-time traffic friendly, special care is needed to support delay sensitive traffic. In the following sections, we discuss the physical limitations of power line channels and then describe a robust power line protocol.

PLC Design Issues

Physical Limitations

A power line is used for transmitting 50 or 60 Hz signals but was not designed to convey high frequency signals such as the 20MHz communication signal used in the *Homeplug* 1.0 protocol. A power line channel is somewhat like a wireless channel - both of them suffer from noise, fading, multi-path and interference. Power line noise is produced by the operation of electrical devices. Fading, multi-path and interference are caused by the imperfection of power line channels.

Lim C.K., So P.L., Gunawan, E., Chen, S., Lie, T.T.,and Guan, Y.L., [6] describe typical attenuation characteristics in power line channels. The authors report that even when all devices are unplugged, the noise still persists and this drastic variation of attenuation is hostile to power line communication. Furthermore, the Federal Communications Commission (FCC) also limits the available bandwidth for communication purposes. In compliance, the usable bandwidth in the *HomePlug* standard is 25MHz. Liu Weilin, Widmer H.-P., Aldis J.,and Kaltenschnee T., [7] did an extensive study of the power line channel characteristics and design issues.

To conquer these problems, a robust signal modulation and data coding is needed.

Signal Modulation

To modulate digital signals on to the power lines, we can use many of the same techniques that are widely implemented in wireless communication. Basic modulation techniques such as Phase Shift Keying (PSK), Frequency Shift Keying (FSK), Minimum Shift Keying (MSK), and Gaussian Minimum Shift Keying (GMSK) can be used for low data rate communication. Other more advanced techniques such as M-ary Phase Shift Keying (MPSK), M-ary Quadrature Amplitude Modulation (MQAM), M-ary Frequency Shift Keying (MFSK) and Orthogonal Frequency Division Modulation (OFDM) can be used when higher data rates are desired. Karl M., and Dostert K.[8] did a thorough study of signal modulation over power lines.

OFDM was adapted by *HomePlug Powerline Alliance* because of its robustness to noise and the fact that it is a parallel data transmission method using a number of parallel frequency division multiplexed subbands. The main problem in using OFDM on wireless networks is frequency offset, caused by the Doppler Effect when the user is moving. The Doppler Effect will cause performance degradation, but in a power line network there are no moving devices, and thus no Doppler effect. The other problem is timing offset, which can be mitigated by offset estimation and compensation.

Spread spectrum signal modulation is different. Since the useful bandwidth in the power line channel is under 25 Mhz, the effect of spread spectrum modulation is considered limited. Using a single carrier modulation on the power line is possible but equalizers could be needed to reduce the delay spread effect, and the associated cost is high.

In order to cope with the wide variation in channel conditions, the physical layer protocol (PHY) for PLC must be adaptive intelligently using more robust modulation and coding schemes, with lower data rates as needed. In addition, critical protocol management information requires high fidelity forward error correction (FEC) coding to ensure that the protocol functions correctly in the worst case situations.

MAC Layer Protocols

In PLC home networks, the power line media can be accessed by multiple devices simultaneously. To decide which device gets the floor to send its data, a medium access control (MAC) protocol is needed. There are many existing protocols that can be implemented on the power line network. CSMA/CD, CSMA/CA, TDMA and hybrid protocols such as TDMA+CSMA are all potential candidates.

The most popular wired MAC protocol, CSMA/CD, could be also applied on a power line network. However, the large variation in noise on the power line makes collision detection very difficult. This characteristic is again very similar to a wireless network, so some have applied the CSMA/CA protocol as suggested in *IEEE802.11* to the power line network. However, the hidden node problem arises when the signal travels through different power lines with highly variable attenuation. To conquer this problem, a RTS/CTS scheme has to be implemented. Though the RTS/CTS scheme solves the hidden node problem, it degrades the network performance.

The benefit of using TDMA is that it provides an upper bound of access delay thus QoS is guaranteed. However, the difficulties in generating a synchronized clock signal in power line networks between devices remains a problem. Other hybrid protocols like TDMA+CSMA provide QoS capabilities in nature, but the network efficiency and beacon generation between TDMA slots and CSMA/CA slots remains unsolved. Romans C., and Tourrilhes J. [9] did a detailed discussion of the hybrid TDMA+CSMA/CA protocol. *Homeplug* 1.0 protocol also provides some level of QoS support in the uses of multiple priority levels that can be used in conjunction with VLAN tagging.

The issue of privacy of power line networks is important to their practicality. Like wireless channels, Power line network channels should be treated as open and as with all open channels; nothing prevents a device from receiving signals. To provide a secure network environment, the *HomePlug Powerline Alliance* defined a 56-bit DES encryption mechanism. Once a signal is encrypted, a device with a different encryption key cannot interpret it and privacy is achieved.

This privacy protection seems adequate but stronger encryption may be needed when power line networks are adopted for office environments or apartment building and hotels. We believe that stronger privacy protection should be implemented in the physical layer, so that hackers can not easily break the code.

Performance Results and Analysis

In this section we report the measurements observed using an event-based C program to simulate a *Homeplug* 1.0 power line network. All scenarios assume QPSK and a 3/4 coding rate on various links and a maximum TCP segment size of 1460 bytes. In this simulation, we use UDP, TCP, and VOIP traffic. UDP traffic is generated with an exponential inter-arrival time with a 100 microsecond average. The UDP packet size is assumed to be a constant 1460 bytes with priority 0. TCP traffic is also generated with exponential inter-arrival time with 100 microsecond average and we assume that TCP traffic sources always have data to send. TCP traffic is treated as priority 0 packets. Every time a node has a chance to send, it is allowed to send the maximum segment size of 1460 bytes without headers. VOIP is isochronous traffic with a 20 msec interval. The packet size of VOIP is 160 bytes and is assigned the highest priority (3).

Simulation Results

In Table 2-3, we provide the of simulation results of a power line network. The UDP traffic simulation scenario 1 shows the best throughput in our simulations since there is no contention at all. Table III also shows channel contention with 2 and 3 UDP nodes causes a modest reduction in channel throughput.

Table 2-3: A power line network simulation results

Throughput of multiple UDP traffic			
	Scenario 1(1 UDP)	Scenario 2(2 UDP)	Scenario 3(3 UDP)
MAC Throughput	8.08 Mbps	7.46 Mbps	7.46 Mbps
Throughput of multiple TCP traffic			
	Scenario 1(1 UDP)	Scenario 2(2 UDP)	Scenario 3(3 UDP)
MAC Throughput	6.16 Mbps	6.15 Mbps	6.12 Mbps
TCP Throughput	5.92 Mbps	5.91 Mbps	5.88 Mbps
Throughput of one VOIP and multiple UDP traffic			
	Scenario 1 (VOIP + 1 UDP)	Scenario 2 (VOIP + 2 UDP)	Scenario 3 (VOIP + 3 UDP)
MAC Throughput	7.89 Mbps	7.33 Mbps	7.29 Mbps
Queueing Delay	0.25 msec	0.25 msec	0.25 msec
Net Delay	2.75 msec	3.00 msec	3.00 msec
Throughput of one VOIP and multiple TCP traffic			
	Scenario 1 (VOIP + 1 TCP)	Scenario 2 (VOIP + 2 TCP)	Scenario 3 (VOIP + 3 TCP)
MAC Throughput	6.04 Mbps	5.85 Mbps	5.77 Mbps
TCP Throughput	5.72 Mbps	5.54 Mbps	5.45 Mbps
Queueing Delay	0.25 msec	0.25 msec	0.25 msec
Net Delay	3.25 msec	3.25 msec	3.25 msec

In the TCP traffic simulation, though scenario 1 has only one traffic source, the bandwidth must be shared with data and response frames (for example, ACK packets) thus it provides lower performance than the UDP traffic simulation. The MAC throughput represents the total number of transmitted bytes divided by the simulation time regardless of successful delivery. The TCP throughput includes only the successfully delivered data and ACKs.

The third metric we provide in Table III is the PLC simulation results of one VOIP and multiple UDP connections. The high priority VOIP always wins the contention and the UDP nodes can send packets only when there is no VOIP traffic. In this simulation, the queuing delay refers to the time a packet waits in a queue before it enters the transmit buffers. The net delay is the total time for which a packet propagates in the networks. Only low priority packets are considered for this delay because the high priority packets will be delivered as soon as they appear in the queue.

The Table 2-3 also shows the simulation results of one VOIP and multiple TCP connections. The throughput of VOIP is only 80 kbps, and hence the total throughput is dominated by the TCP component.

Real-World PLC Network Performance

In addition to simulating the performance of the *HomePlug Powerline Alliance* protocol, we were also able to construct a real PLC network using “reference designs” of actual commercial *HomePlug* devices. Since there are currently no real IA devices with PLC capability, we used traditional network applications (that is, ftp and streaming multimedia content) as the basis for measuring PLC network performance.

In this experiment, there were 4 desktop computers. A 450 MHz Pentium II desktop computer (PC-2 as a file server) is equipped with 128 MBRAM, a 3-COM fast Ethernet card, and a PLC PCI card. Two 700 MHz Pentium III desktop computers (PC-3 and PC-4) are both equipped with 256 MBRAM , and PLC PCI cards. A 266 MHz Pentium MMX desktop computer (PC-1) is equipped with 64 MB RAM, and a 3-COM fast Ethernet card.

The PC-1 computer is connected to an Ethernet-to-power line bridge, which converts packets generated from the Ethernet card into PLC compatible packets, and vice versa. All computers are connected to power lines.

In this experiment, we seek to determine the performance of the PLC network in handling streaming video and large file transfers.

Performance of Delivering Streaming Video

We first examined the ability of the PLC network to deliver real-time traffic. Four video files are involved in this experiment. The first file is encoded in Real media format with a bit rate of 550 kbps; the second is encoded with bit rate of 1396 kbps, the third is encoded at 2 Mbps, and the fourth is an MPEG2 video file with variable bit rate, and the average bit rate is 8 Mbps. In the first experiment, three client computers simultaneously issued requests for low bit rate (550kbps) video service to the file server. In the second experiment, the same procedure was executed, but a medium bit rate (1394 Kbps) video service was requested. In the third experiment, the 3 clients requested a high bit rate (2 Mbps) video service. Finally, the MPEG2 video service request was issued by the PC-3. The experimental results are shown in Table 2-4.

The PLC network successfully delivered both low and medium bit rate streaming videos. We did not observe any packet drops during the experiments. The results met our expectations, since the peak data rate was only 4185 kbps. We did another experiment to further investigate the performance of PLC network in delivering streaming video. A 2 Mbps MPEG-1 file is used in this experiment. As the video begins, a significant video freeze-then-go (halting) phenomenon was observed, causing staccato playback. After several seconds (3-5 seconds) the freeze-then-go phenomenon disappeared.

In the case of MPEG2 video file, the average data rate is 8Mbps. During the experiment, a significant "video staccato" phenomenon was observed. To exclude the possibility that the observed phenomenon was caused by the client computer's hardware capability, the experiment was repeated with same configuration, while connected to a fast Ethernet. During that experiment, no such phenomenon (halting playback) was observed.

Table 2-4: Real-world PLC network performance

Performance of delay sensitive traffic				
	Number of Connections	Aggregated bit rate	Packet drop	delay-jitter
Low Bit Rate	1	550 kbps	No	No
	2	1100 kbps	No	No
	3	1650 kbps	No	No
Medium Bit Rate	1	1395 kbps	No	No
	2	2790 kbps	No	No
	3	4185 kbps	No	No
High Bit Rate	1	2000 kbps	No	No
	2	4000 kbps	No	No
	3	6000 kbps	N/A	Moderate
	Number of Connections	Aggregated bit rate	Environment	
Variable Bit Rate	1	8 Mbps	PLC network	
	1	8 Mbps	Fast Ethernet	
Performance of elastic data traffic				
	Number of Connections		Average bit rate	
Elastic data traffic	1		6.21 Mbps rate	
	2		6.15 Mbps rate	
	3		6.27 Mbps rate	
Performance of combined delay sensitive traffic and elastic data traffic				
	Connections		Aggregated bit rate	
Hybrid data traffic	One ftp connection and One video service		6.26 Mbps rate	
	Two ftp connection and One video service		5.92 Mbps rate	

Performance of Elastic Data Traffic

The occurrence of the momentary video freezing phenomenon during playback of variable bit rate streaming is likely because the aggregated data rate was close to or exceeded the PLC network capacity. To understand the real throughput of a PLC network, we conducted another experiment A 215,502,106 byte file was placed on the server running an FTP daemon (The file size was chosen to minimize hardware uncertainty and human error.) Client computers made FTP requests for the file. We tested different numbers of FTP connections, up to 3, using individual client machines in our PLC network. The experimental results are also given in Table 2-4.

The aggregated traffic in the table is calculated by adding all observed data rates of all connections. The experimental results show that the real PLC network performance is about 6 Mbps. When there is only one FTP connection, the observed throughput is 6.21 Mbps. By our analysis, one TCP connection will not fully utilize the PLC network, because the server has to stop if no ACK packets are received from the client.

Aggregated traffic volume decreased as the number of connections increased from 1 to 2. This phenomenon is because of the ACK packets and the packet overhead increase as the number of connections increased. Although the network utilization improves, the improvement cannot compensate for the loss due to these overheads.

When we increased the number of connections from 2 to 3, the PLC network had the highest throughput of 6.27 Mbps. This is because the network utilization increased as the number of connections increased which compensated for the packet overhead and ACK overhead.

These experimental results explain the phenomenon of momentary DVD video freezing playback. The requested bandwidth for DVD streaming exceeded the maximum bandwidth the present PLC network can provide.

Performance of Combined Delay Sensitive Traffic and Elastic Data Traffic

Although we could not explore the QoS service and packet priority provided by the real PLC network, we were eager to learn the effect of mixed traffic on the PLC network. This experiment was conducted as follows: The file server provided two services: one for streaming video with bit rate 550 kbps, and the other one for file transfer with a file size of 215,502,106 bytes. PC-1 requested streaming video while PC-2 and PC-3 requested the file transfer. Each experiment lasted 285 seconds (that is, the length of the video file), after which both video player and FTP client are forced to stop. Table IV shows our experimental results.

When the number of FTP connections increased, the observed data rate decreased as was the case in the previous experiment. However the overall average data rate was comparable to the case of multiple FTP traffic.

Conclusion

The emergence of Information Appliances (IA) for the smart homes of the future will undoubtedly make our lives much more comfortable than ever. However, the infrastructure that supports multimedia traffic and conventional elastic data traffic for communication among IA devices is a critical component of a smart home.

We advocate power line as the infrastructure for smart homes based on the convenience of the power sockets and the layout of the power line network existing in every home. At present, 6 Mbps of bandwidth was measured through real-world PLC network experiments. Our studies showed that the PLC network can provide 3 low bit rate or 3 medium bit rate multimedia streams concurrently with no packet drops and jitters. It also successfully delivered one low bit rate multimedia data stream and 2 large ftp file transfer concurrently with no packet drops and jitters.

In this chapter, we discussed only the PLC networks for communication *within* the smart home, but the ultimate goal of PLC network could be the ability to connect to the Internet without dialing up to an ISP server, entirely using electrical wiring only. This can be illustrated as in Figure 2-3.

Private home networks are connected to substations, in which a PMTS (Powerline Modem Terminal Service) connects PLC networks within homes to the Internet backbone. The PLC network gateway for a private home network could be installed in the fuse box of that home and then it could be connected to one or more repeaters. Repeaters are for increasing signal strength when the signals level fall below some value.

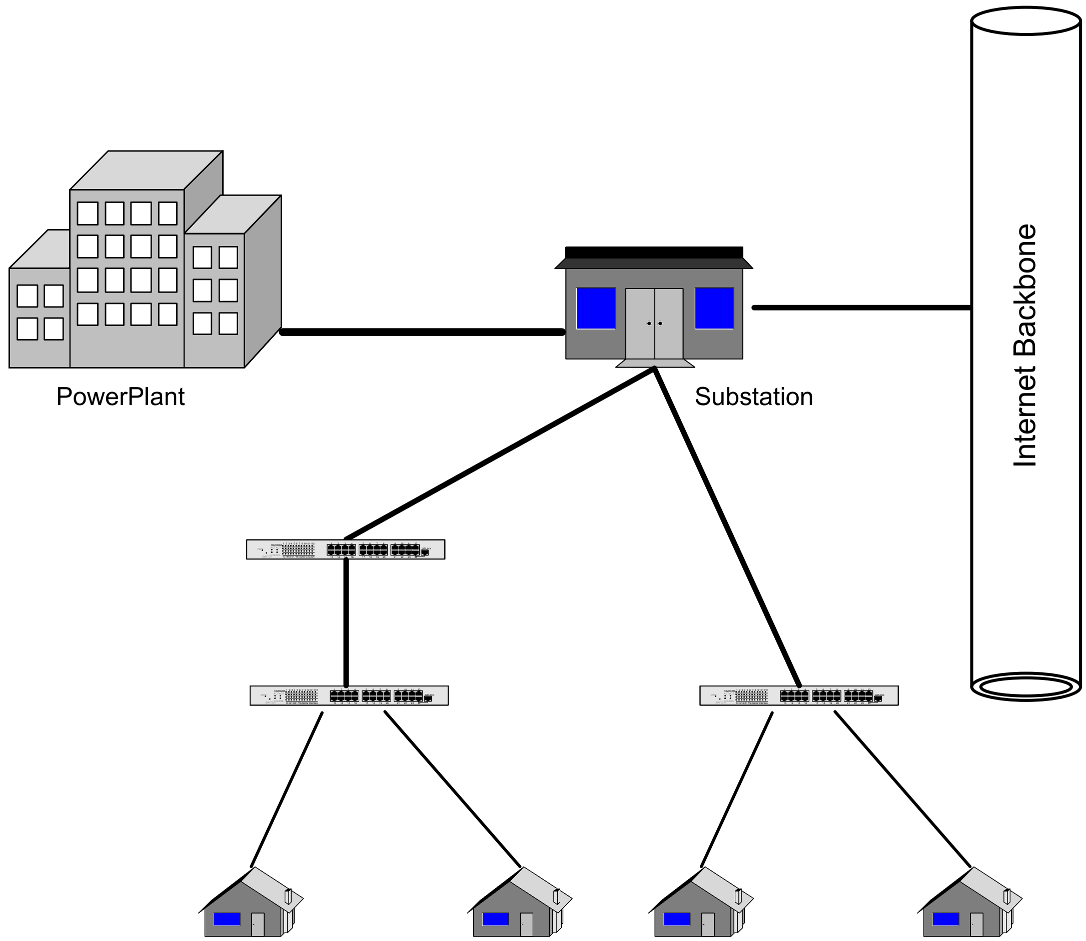

Figure 2-3: Connecting PLC networks to the Internet

We expect to see higher data rates in power line networks in the future as signal modulation technologies improve; however, issues like network security and the network characteristics with a large number of nodes need further development. Further research on these issues is of critical importance when power line networks are applied to offices and large multi-user buildings.

CHAPTER 3
A COMPARATIVE PERFORMANCE STUDY OF WIRELESS AND POWER LINE NETWORKS

Local Area Networks based on the IEEE 802.11a/b wireless networking standards and emerging Power Line Communication (PLC) standards are attractive for establishing networks with "No New Wires" for in-home and business applications. This Chapter presents a theoretical performance comparison of the 802.11 a/b and the Home-Plug 1.0 PLC protocols. We also presents comprehensive comparative field test results addressing such issues as coverage, channel stability and reliability as well as the associated implications on the capability of these technologies to provide QoS support for multimedia traffic in typical residential settings.

Introduction

Candidate networking technologies for providing convenient and widespread residential and SOHO networking services may be categorized as *Wireless Networks*, *Wired Networks* and *No New Wires Networks*. Hughes S, and Thorne D.J. [2] did an extensive study of various infrastructure options and technologies appropriate for home networks. Below, we give a short discussion of networks in the above three categories.

Wireless Networks such as 802.11x, BlueTooth, and HomeRF can be constructed by installing multiple interconnected wireless access points (WAP) and base stations within target areas. The best benefit of using wireless networks is the freedom to move around while maintaining network connectivity. Blue-Tooth technology is targeted at personal communications and the coverage is expected to be limited. On the other hand, though HomeRF has been on the market for a few years, it is not yet widely accepted. Thus the most interesting and widely accepted wireless networking technologies are the 802.11x family. 802.11b operates in the 2.4 GHz band and provides a maximum data rate of 11 Mbps; 802.11a supports speeds of up to 54 Mbps and operates in the 5 GHz band. Standards for the newer IEEE 802.11g, which should provide data rates up to 54 Mbps in the 2.4 GHz band, have not been finalized, and equipment was not available for testing.

For *Wired Networks*, a comprehensive Ethernet network can be constructed by installing special UTP-5 cabling. While the stability and the security of wired networks are guaranteed, installing new wires in existing home or other buildings may be costly, negating the low cost of the network interface cards.

For the *No New Wires Networks* category, there are phone line networks, cable networks, and power line networks. Using the existing phone line as an infrastructure, as in Frank E.H., and Holloway J. illustrated [2], may seem attractive, but it is limited by available phone sockets in a home. Home Cable Network Alliance (Home-CNA) [10], established in June 2001, and proposes a home network infrastructure using existing coaxial TV cable. There is as yet no standard for HomeCNA and it also suffers from the major drawback of limited convenient connection points.

Power Line Communication (PLC) networks such as HomePlug[3] were introduced to the U.S. consumer market in May 2002. European PLC networks have been deployed in recent years. With multiple outlets in almost every room, residential power lines are already the most pervasive network in the home or small office. The HomePlug 1.0 PLC standard supports PHY data rates of 14 Mbps and is thus comparable to the 802.11b declared data rate.

A major objective of this paper is to conduct a real-world performance study of the capabilities of wireless (IEEE 802.11b and 802.11a) networks and PLC networks based on the HomePlug 1.0 standard. Our interest is to determine the relative performance of these technologies.

This chapter presents a comparative analysis of the TCP performance of power line networks and wireless networks using actual measurements on HomePlug 1.0 compliant PLC networks and 802.11a/b compliant wireless networks. The tests were conducted in 20 houses ranging in area from 1500 to 5000 sq. ft. with an average area of 3000 sq. ft. The paper presents qualitative theoretical and measured throughput performance for 802.11a/b and HomePlug 1.0 PLC. Other issues like the relationship between QoS and channel stability as well as overall coverage are also discussed.

The next section briefly describes the HomePlug 1.0 protocol. Section 3-3 presents a theoretical performance analysis of IEEE 802.11a/b and HomePlug 1.0. Section 3-4 describes the experimental setup while Section 3-5 gives our field test results. A summary is given in Section 3-6.

Homeplug 1.0 Protocol

The parameters and details of 802.11x protocols are well documented in the literature and Internet publications [11]. Here, we briefly describe the HomePlug 1.0 standard.

PLC Environment

Power lines were originally devised for distributing electrical power using the frequency range of about 50-60 Hz. The use of this medium for high speed communications presents some technically challenging problems. Electrical noise from appliances and the uncontrolled nature of the wiring result in severe signal distortions.

The PLC channel is made up of different conductor types; therefore a variety of characteristic impedances will be encountered. Further, the network terminal impedance will tend to vary with frequency and time as the consumer's load pattern and load types vary. Impedance mismatch causes multi-path effects resulting in deep notches at certain configuration dependent frequencies. These channel imperfections make signal transmission over a power line very difficult [5].

Reliable data communication over this hostile medium requires powerful Forward Error Correction (FEC) coding, interleaving, error detection and Automatic Repeat Request (ARQ) techniques, along with appropriate modulation schemes as well as a robust Medium Access Control (MAC) protocol. The lack of affordable processing techniques needed to overcome the harsh power line environment resulted in limited success of power line communications in the past. However, both the advances in the ASIC density and speeds, and the advancement of signal modulation, processing and error control coding techniques now make power line communication possible.

HomePlug 1.0 PHY

To overcome the hostile PLC environment, Orthogonal Frequency Division Modulation (OFDM) with a Cyclic Prefix (CP) was adopted by the HomePlug 1.0 PLC standard. Using OFDM has many benefits. For example, it exhibits excellent mitigation of the effects of time-dispersion, provides excellent Inter-Channel Interference (ICI) performance, and is good at minimizing the effect of in-band narrowband interference. OFDM splits available bandwidth into many small frequency bands called sub-carriers, then may mask out unusable subcarriers and apply the best modulation and coding methods to the usable subcarriers. This approach is used by HomePlug 1.0. A more advanced technique called bit-loading allows use of different modulation and coding schemes for each sub-carrier. In either case, OFDM can adapt bandwidth/data rates according to channel conditions.

Unlike 802.11, the bandwidth in HomePlug 1.0 can vary from 1 Mbps to 14 Mbps practically continuously according to the channel conditions[1]. Active HomePlug 1.0 nodes perform channel estimation at least once every 5 seconds. This feature allows the PLC network to maximize its data rate adaptively.

A preamble and frame control form delimiters used for synchronization and for control. The frame control of start of frame, end of frame, and response delimiters all include delimiter type[2], and contention control information. In the start of frame delimiter, the frame control field includes the tone map information needed by the receiver to decode the rest of the frame, and a length field. The end of frame delimiter contains priority information used for contention control. Response delimiters contain information that allows a sender to verify that the response was indeed sent in response to the frame it just transmitted. An end of frame gap (EFG) of 1.5 μs is inserted between the frame's frame check sequence (FCS) and the end delimiter to allow for processing.

HomePlug 1.0 MAC

The HomePlug 1.0 Medium Access (MAC) protocol is a modified CSMA/CA (Carrier Sense Multiple Access / Collision Avoidance) protocol with priority signaling. HomePlug 1.0 devices operate in an ad hoc mode in the sense that devices communicate with each other freely, without any centralized coordination.

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[1] There are 139 distinct data rates in that range according to number of usable carriers, modulation methods, and coding rate.

[2] Response expected is indicated in the delimiter type, and Response delimiter does not have a response expected/not expected
indication.

The frame structure and protocol of HomePlug 1.0 is depicted in Figure 3-1.

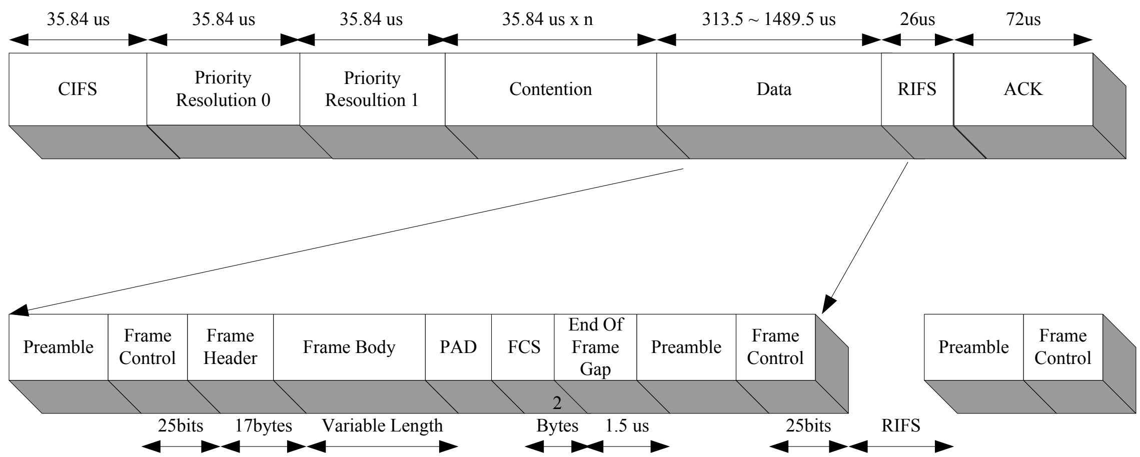


Figure 3-1: HomePlug 1.0 frame structure and protocol

The HomePlug 1.0 standard uses different terms and stages for inter-frame spacing and for the contention windows than 802.11b. The RIFS shown in the figure is “Response Inter-Frame Spacing.” Unlike 802.11, there is no SIFS (Short Inter-Frame Spacing) between continued frames. Rather, a frame control bit is used to indicate the desire of a station to continue to send data, allowing preemption only by higher priority traffic. The spacing between the last frame and the incoming frame is CIFS (Contention Window Inter-Frame Spacing).

HomePlug 1.0 provides four priority classes - CA3, CA2, CA1 and CA0 from highest to lowest. Priority resolution is done by asserting signal of the priority level in the PR0 and PR1 slots. For example, to send a CA2 packet, the PLC device should assert a 1 in PR0, causing any node with CA1 traffic to defer, and not assert 1 in PR1 as it would do otherwise. Nodes with CA3 data assert a 1 in both priority slots and CA0 in neither. This effectively resolves contention between different priority classes. Contention within the same priority class is resolved during the contention period.

The contention period is a contention period. The contention period is a form of CSMA/CA with a priority dependent backoff window size schedule. For the lower two priority classes, it is 8-16-32-64 slots, while it is 8-16-16-32 slots for the two higher priority classes. On collision, the range of contention slots over which a transmission is started is increased according to the schedule. Aside from starting with a smaller range (8 slots compared to 32 slots), a major difference from the IEEE 802.11 standard is that when a HomePlug 1.0 node defers (detects another node's transmission in an earlier slot), it uses this information to back off, but less aggressively than in the case of a collision. This technique serves to reduce costly collisions further. For protocol details, please see the HomePlug 1.0 Specification [3].

Theoretical Performance of 802.11a/b and Homeplug 1.0

To compare the protocol performance of 802.11a/b and HomePlug 1.0, we first analyze the theoretical performance differences between them. In next section the protocol analysis to calculate the theoretical performance of 802.11a/b and HomePlug 1.0 is presented. The analysis assumes that a single station is continuously transmitting frames with 1500 bytes of payload over the medium.

802.11a/b Theoretical Network Performance

In the absence of competition, an 802.11b node picks a contention slot between 0 and 31, and starts transmission then. The average contention period delay for a packet without competition is 31/2 = 15.5 slots or 310 μsec. The transmitting node will start to send data and wait for the receiver's acknowledgment.

Each frame is made up of a PLCP header, a MAC header, a DATA field and a CRC field. If it is an ACK frame then the DATA field is not present.

In practice, the maximum data payload sent via the 802.11a/b is limited to the Ethernet maximum of 1500 bytes. Table 3-1 summarizes the MAC throughput and efficiency for IEEE 802.11x protocols at various data rates. Packet fragmentation and MAC level packet concatenation are not considered.

Table 3-1: 802.11a/b MAC throughput with payload 1500 bytes

Technology	PHY Data Rate	MAC Throughput
802.11b	1 Mbps	0.91 Mbps
	2 Mbps	1.73Mbps
	5.5 Mbps	3.99Mbps
	11 Mbps	6.38Mbps
HomePlug 1.0	1 Mbps	0.70Mbps
	2 Mbps	1.74 Mbps
	5.5 Mbps	3.77 Mbps
	11 Mbps	8.08 Mbps
	14.1 Mbps	8.08 Mbps
802.11a	6 Mbps	5.38 Mbps
	9 Mbps	7.78 Mbps
	12 Mbps	10.02Mbps
	18 Mbps	14.1 2Mbps
	24 Mbps	17.61 Mbps
	36 Mbps	23.74 Mbps
	48 Mbps	28.47 Mbps
	54 Mbps	30.80 Mbps

From Table 2-1, the maximum throughput is 6.38 Mbps for 802.11b and 30.8 Mbps 802.11a, representing efficiencies of 58% and 57% respectively. At 1Mbps the MAC efficiency of 802.11b is as high as 91% assuming there is no packet fragmentation.

Performance Analysis of HomePlug 1.0

From Figure 2-1, the data transmission time ranges from 313.5 to 1489.5 μs, however to transmit a 1500 byte payload at the maximum data rate, 120 symbols are required, taking 1153.5 μs. Excluding physical level control overhead results in 1008 μs for actual data transmission. The maximum data payload size is limited to the smaller of

1500 bytes and 1344*R-OH* bits, where *R* is the physical data rate, and *OH* is the number of overhead bits. Each Ethernet frame incurs an overhead of at least 120 bits for encryption and integrity checking for the corresponding service block. Segment bursting allows a station to send all the segments associated with a service block consecutively, avoiding contention unless it is preempted by a station with higher priority traffic. A service block is broken into physical layer segments, each of which has 19 additional bytes of overhead for addressing and segment control. Additionally, each segment must be a multiple of 20 symbols long, up to 160 symbols maximum, which further complicates throughput analysis.

The initial contention window size is 8 slots, so the average contention delay without competition is 3.5 slots. To successfully deliver a data packet of 120 symbols takes

$$35.84\ \mu s + 35.84\ \mu s + 35.84\ \mu s + 35.84\ \mu s \times 3.5 + 1153.5\ \mu s + 22\mu s + 26\mu s = 1484.86\mu s$$

The maximum physical layer data rate is 14.18 Mbps, thus the maximum throughput is

$$1500 \times 8\ \text{bits}/1484.86\mu s = 8.08\ \text{Mbps}.$$

The efficiency of HomePlug 1.0 at the maximum data rate is 57%. The 70% efficiency of HomePlug 1.0 at PHY rate of 1Mbps is due to the limits of the data transmission time to a maximum of 1484.5µs in order to provide better latency and jitter QoS parameters for higher priority traffic.

Although we can get up to 8 Mbps maximum MAC throughput in theory, the maximum measured TCP throughput in our field testing so far is 6.3 Mbps, matching earlier HomePlug 1.0 simulation results [12].

Experimental Setup

To understand the real world performance of IEEE 802.11a/b and HomePlug 1.0, we conducted field tests in 20 houses located in the Gainesville, Ocala, Orlando, and Belleview areas of Florida. The choice of the houses used in the tests were in the mid-to-large size (1500 sq. ft. to 5000 sq. ft.), since larger houses provide a better range on the performance parameters of interest.

The equipment used in this test included the following.

1. *AP* Server: A Sony notebook with a 700 MHz Pentium III processor and 128k RAM running Windows2000
2. *Mobile* Station: An HP notebook with a 500 MHz Pentium III processor and 128k RAM running Windows2000
3. Linksys HomePlug 1.0-based Powerline-to-Ethernet bridges
4. Netgear[13] IEEE 802.11b Access Point and PCMCIA Card
5. D-Link DWL-5000AP IEEE 802.11a Access Point and D-Link DWL-A650 PCMCIA card.

For PLC testing the two laptops were connected through the power line via Powerline-to-Ethernet bridges. For wireless testing, a Modified Infrastructure Mode (MIM) was used. The *AP* Server was connected to an access point using an Ethernet crossover cable to the built-in Ethernet socket. A PCMCIA slot in the mobile station was used to connect the wireless card. Note that typical wireless networks use an Infrastructure Mode (IM). In this mode, all wireless nodes communicate with each other through the access point, and must share the bandwidth over two hops. MIM can be expected for connection from a node to an access point. Since these tests had the *AP* server connected to the access point via Ethernet, there with no other contention possible, the test results should represent the best case scenarios with respect to this aspect.

Experimental Method

The TCP throughput and distances were measured for various locations of *AP* and *Mobile* stations inside the house. The AP Server was located close to a phone outlet or a cable outlet, the most probable locations for the home network to be connected to the broadband access network. The Mobile Station was located at various places where it would be likely to find other networked devices in the home. The AP Server antenna and Mobile Station antenna were placed randomly to minimize the effect of directional antenna gain. We argue that this is the typical antenna placement since ordinary users probably don't know how to set antenna directions to maximize throughput. Besides, not all locations are susceptible to antenna direction adjustment due to the surrounding environment.

WSTTCP, a popular TTCP implementation ported toWindows sockets, was used. The TCP buffer size was chosen to be 11680 bytes (1460×8). The number of TCP buffers transmitted was chosen such that each test ran for approximately 60 seconds. A single run of WSTTCP involves starting the WSTTCP in receive mode at the receiver on a selected port. WSTTCP was then started at the transmitter with a specific TCP buffer size, number of TCP buffers to be transferred, the receiver IP address and the receiver port number. At the end of transmission, WSTTCP (at both the transmitter and receiver) provided the throughput observed on the link. For several of these tests, real time packet capture was also obtained to observe TCP stability. All the procedures were automated and required minimal human operation.

Results

The amount of data collected is too large to be presented in detail in this paper, so only a summary of the most interesting findings are presented[3]. The location, size and age of each of the houses where testing was conducted is shown in Table 3-2 to 3-4.

Table 3-2: List of the houses tested and connectivity

Location	House Size (Square Feet)	House Age (Years)	802.11a Connectivity	802.11b Connectivity	HomePlug 1.0 Connectivity
Gainesville	1460	4.5	90%	100%	100%
Gainesville	2000	5	90%	100%	100%
Gainesville	2030	3	75%	100%	100%
Gainesville	2100	7.5	100%	100%	100%
Ocala	2300	33	42.9%	78.6%	100%
Gainesville	2700	4.5	61.9%	100%	100%
Gainesville	2700 (two floor)	1	57%	100%	100%
Gainesville	2700	10	20%	100%	100%
Gainesville	3000	1	91%	100%	100%
Gainesville	3000 (two floor)	9	71.4%	100%	100%
Ocala	3000	9	54.2%	100%	100%
Gainesville	3150	13	55.56%	95.8%	100%
Ocala	3500	6	69%	100%	97.6%
Ocala	3600	10	41.7%	100%	100%
Orlando	3600	4	50%	100%	100%
Belleview	3600	4	42%	92%	100%
Gainesville	3900	5	56%	100%	100%
Gainesville	4000	4	9%	84%	81%
Orlando	4200	67	18.75%	50%	100%
Ocala	5000 (two floor)	15	17%	50%	100%

[3] The complete data set from our field tests is available on request.

IEEE 802.11a Indoor Performance

The performance and coverage results of IEEE 802.11a are depicted in Figure 3-2.

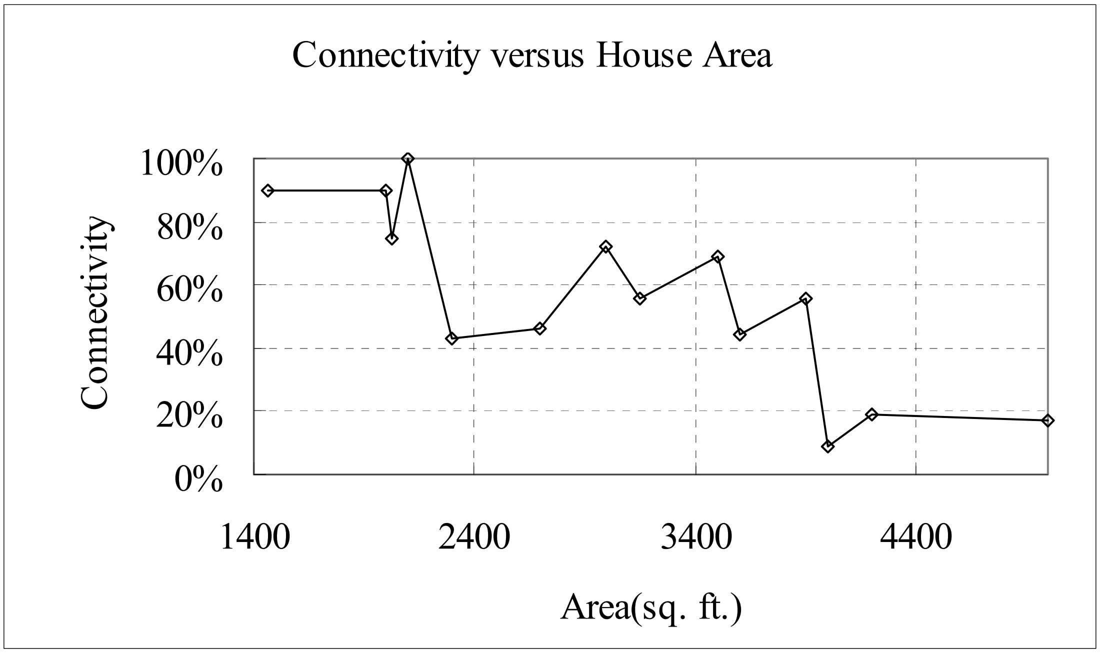


Figure 3-2: IEEE 802.11a indoor connectivity

Figure 3-2 shows the connectivity (that is, percentage of good links) as a function of house area. As expected, the connectivity decreased as the house area increased. Results show that connectivity is poor even in moderate size (2500 sq. ft.) houses. For larger houses (>4000 sq. ft.) the connectivity decreased to 20 %.

Figure 3-3 shows a scatter plot of throughput as a function of distance. It is interesting to note that IEEE 802.11a connectivity is almost zero when the distance is larger than 50 ft.

Figure 3-4 shows the percentage of links that exceed the throughput values indicated on the X-axis. Note that the maximum IEEE 802.11a throughput obtained from the product being tested was larger than those expected from theory. This could be because of manufacturer-specific proprietary enhancements like the use of higher level modulations.

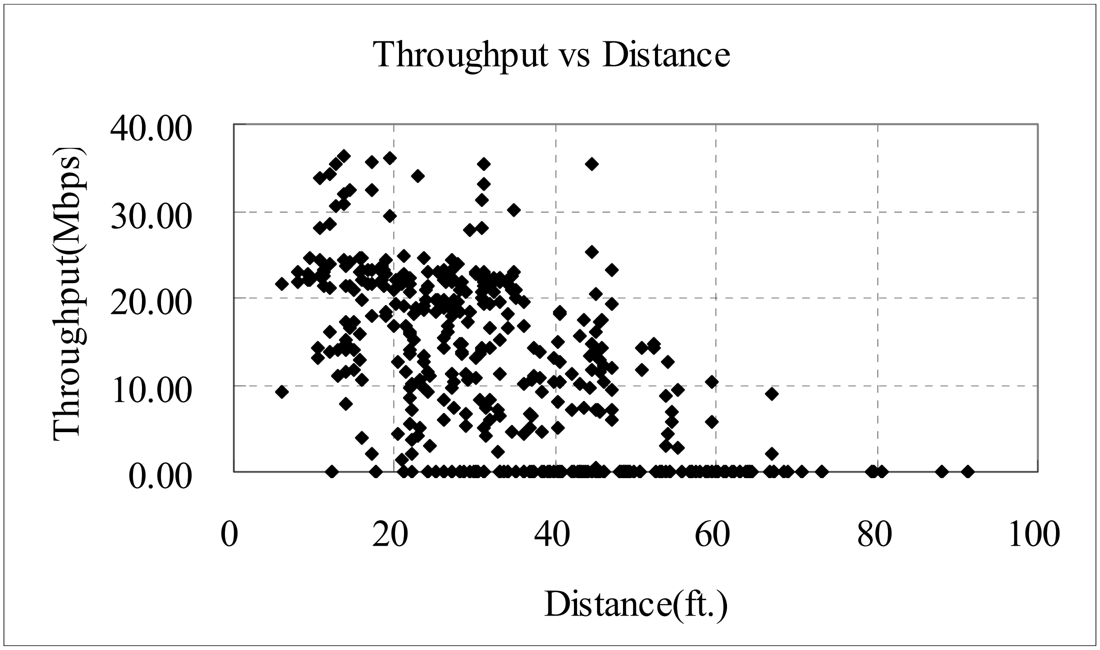


Figure 3-3: IEEE 802.11a indoor throughput as a function of distance

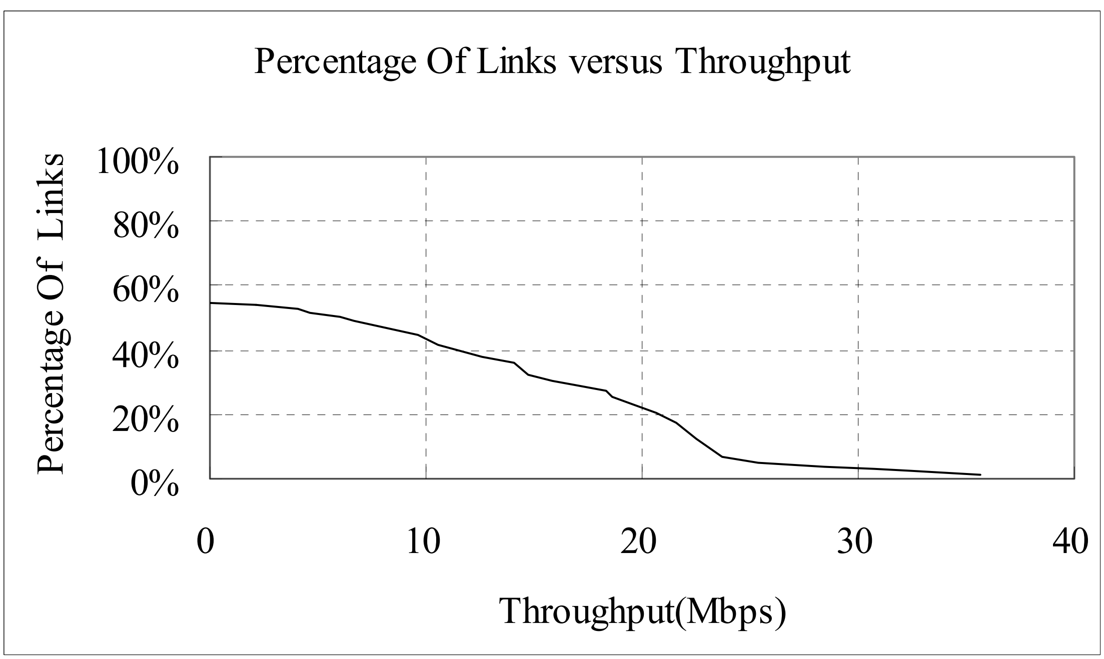


Figure 3-4: IEEE 802.11a indoor percentage of links

In summary, the statistics show that

1. IEEE 802.11a failed on at least one link in 19 of the 20 houses tested,
2. 802.11a failed to connect in 45% of the links that were tested,
3. 802.11a showed close to zero connectivity at distances larger than 50 feet. For shorter distances, 802.11a provided excellent throughput in most cases.

IEEE 802.11b and HomePlug 1.0

To facilitate comparison, the performance and coverage results of IEEE 802.11b and HomePlug 1.0 are shown in Figure 3-5 to 3-8. Figure 3-5 shows the connectivity as a function of house area. Both technologies show high connectivity for houses of size less than 4000 sq. ft. For houses larger than 4000 sq. ft., the connectivity for IEEE 802.11b dropped dramatically to 50% in both of the houses tested, while HomePlug 1.0 continued to show high connectivity.

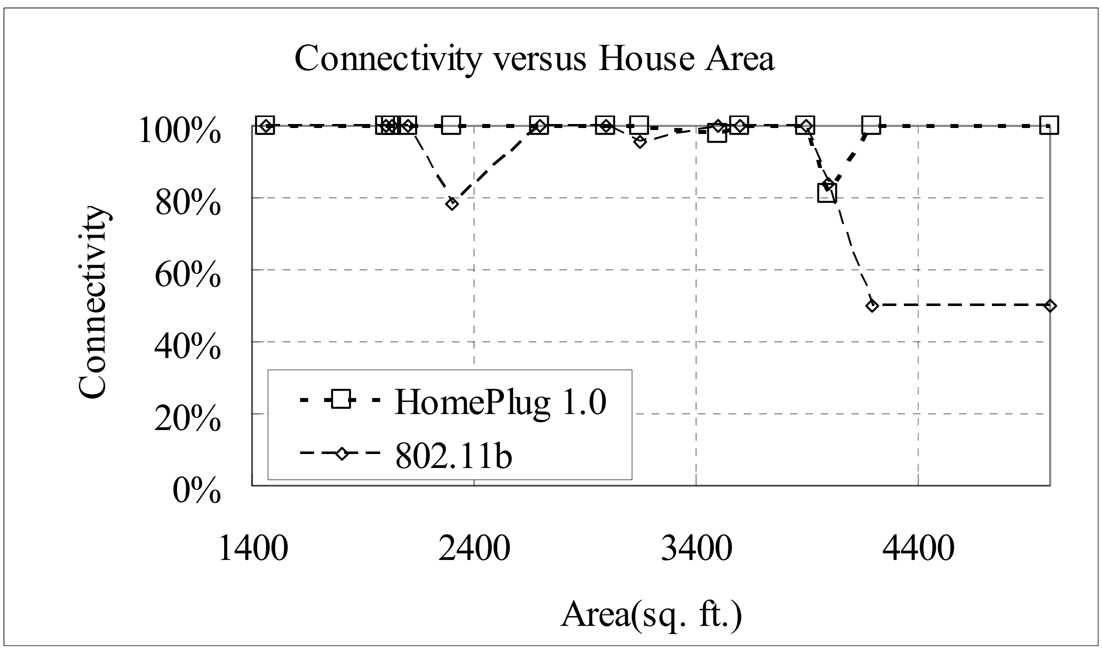


Figure 3-5: IEEE 802.11b and HomePlug 1.0 indoor connectivity comparison

Figure 3-6 shows a scatter plot of throughput as a function of distance. IEEE 802.11b typically provides close to maximum throughput at distances of less than 50 ft.; for distance larger than 50 ft., the performance exhibited large variations. On the other hand, the HomePlug 1.0 system performance is not correlated with the line of sight distances measured in this experiment. This is because HomePlug 1.0 signals have to pass through the convoluted power line cable runs to reach the mobile station.

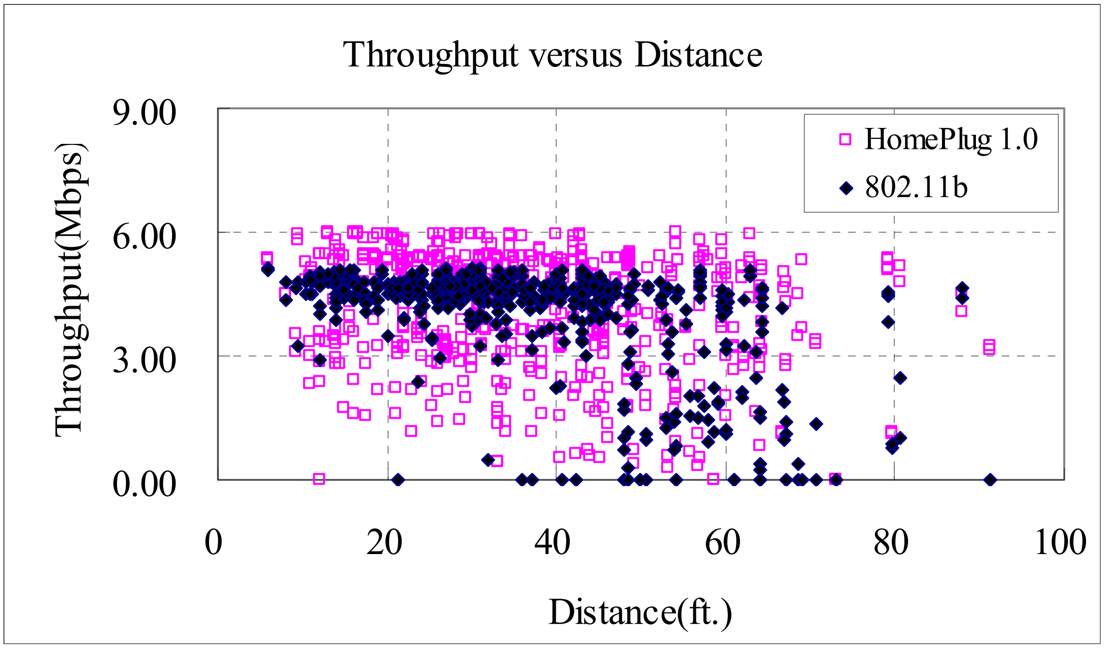


Figure 3-6: IEEE 802.11b and HomePlug 1.0 indoor throughput comparison

Figure 3-7 shows the percentage of links that exceed the throughput value depicted on the X-axis. Our experiments showed that the overall coverage of 802.11b was 92%. The maximum throughput observed in field testing was 5.13 Mbps. Figure 3-7 shows that around 70% of the connections operated at more than 4 Mbps and 10% above 5 Mbps. For HomePlug 1.0, the overall coverage is 98%. The maximum throughput observed in testing was 5.98 Mbps. For HomePlug 1.0, 58% of the connections operated above 4 Mbps and 38% had throughput above 5 Mbps.

The interesting crossover phenomena displayed in the graph reflect three aspects of the systems. First, the paucity of data rates supported by 802.11b hurts its performance when channel conditions are suboptimal. Second, HomePlug 1.0's ability to adapt to the channel conditions with a nearly continuous selection of data rates allows it to perform better under mediocre channel conditions. Finally, the higher maximum data rate of HomePlug 1.0 allows it to outperform 802.11b when channel conditions are favorable.

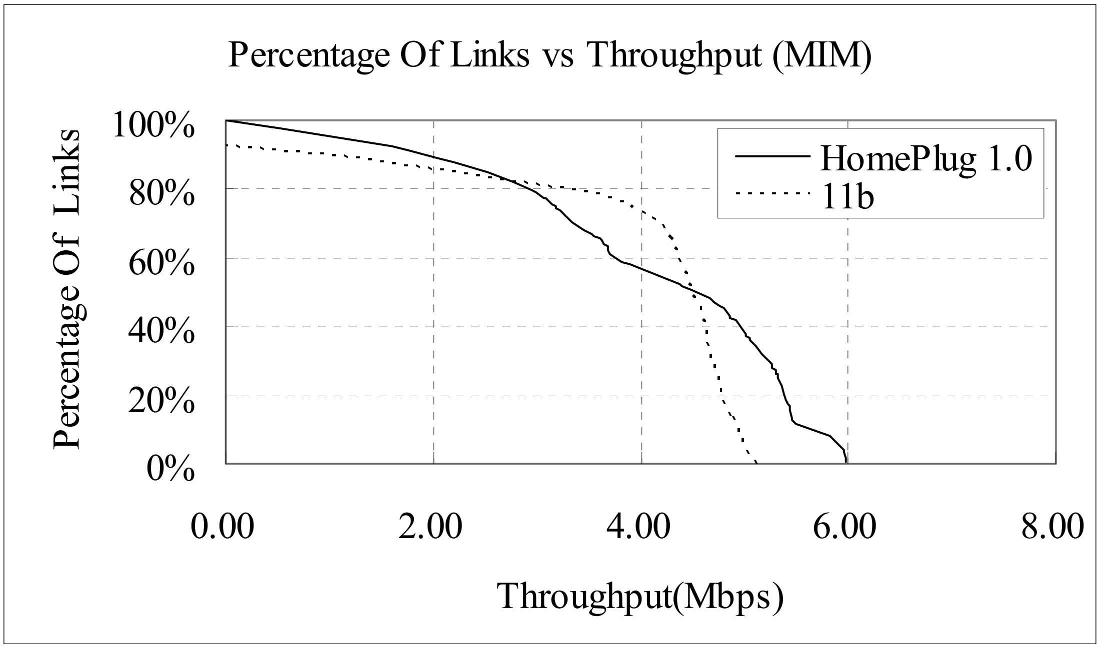


Figure 3-7: IEEE 802.11b and HomePlug 1.0 indoor percentage of link versus throughput (MIM) comparison

It should also be noted that in this experiment, throughput was measured between the access point to the Mobile Station in a modified Infrastructure mode. However, this may not always be the way IEEE 802.11b stations communicate with each other. IEEE 802.11b networks can be configured in either ad hoc mode or Infrastructure mode (IM). In the ad hoc mode, wireless stations communicate with each other directly. However, typical home networks use an IM in which each wireless station communicates with the access point, which in turn forwards the data to the designated receiver. Some of the reasons for using IM include ease of setup, better coverage, and security. Further, most wireless equipment is configured in IM out of the box, and must be reconfigured to ad hoc mode by the customer. Thus, in a typical IEEE 802.11b home network, all station-to-station transmissions, other than those designated to the access point itself (that is, the access point is the final destination of the transmission) or those that originate from the access point, will be retransmitted by the access point. This reduces the effective throughput experienced between such stations.

We use a simple method to extrapolate the Infrastructure Mode (IM) throughput from the MIM link throughput data which was collected in the field tests. A random sample was chosen from the set of collected data and used as the throughput (R_1) from a TCP Source to the Access point. Another random sample was chosen from the sample and was used as the throughput (R_2) from access point to the TCP destination. The aggregate IM throughput then can be obtained by assuming a fixed packet size of x bits is transmitting through two links with speeds R_1 and R_2. The total time to transmit this packet will require $\frac{x}{R_1}+\frac{x}{R_2}$ second. Thus throughput can be calculated by $\frac{R_1 \times R_2}{R_1 + R_2}$. Multiple iterations were used to obtain the distribution of the IM throughput. Although this method is not fully accurate, it is reasonable to expect the actual performance in IM to be close to the values obtained.

Figure 3-8 shows the percentage of links that exceed the calculated IM throughput value depicted on the X-axis. These results show that HomePlug 1.0 stations provide superior coverage and throughput compared to IEEE 802.11b stations in IM.

From the statistics, we make the following key observations

1. HomePlug 1.0 had a larger maximum throughput that 802.11b(about 1 Mbps larger).
2. On 60% of the links HomePlug 1.0 performed better than 802.11b links in MIM,
3. On an average basis, HomePlug 1.0 gave approximately 0.2 Mbps higher TCP throughput than 802.11b in MIM,
4. In 6 of the 20 houses tested, IEEE 802.11b failed on at least one link,
5. In 2 of the 20 houses tested, HomePlug 1.0 failed on at least one link,
6. On an average basis, HomePlug 1.0 gave approximately 2.3 Mbps higher TCP throughput than 802.11b would be expected to give in IM.

In summary, HomePlug 1.0 was found to provide better coverage and slightly better average TCP throughput than IEEE 802.11b in modified infrastructure mode, which would be typical for Internet access. In infrastructure mode, HomePlug 1.0 was estimated to have throughput about 2.3 Mbps greater than 802.11b. For shorter line-of-sight distances, 802.11x performed better than HomePlug 1.0, but for longer distances, the nearly continuous adaptation capability of HomePlug 1.0 allowed it to make better use of mediocre channels.

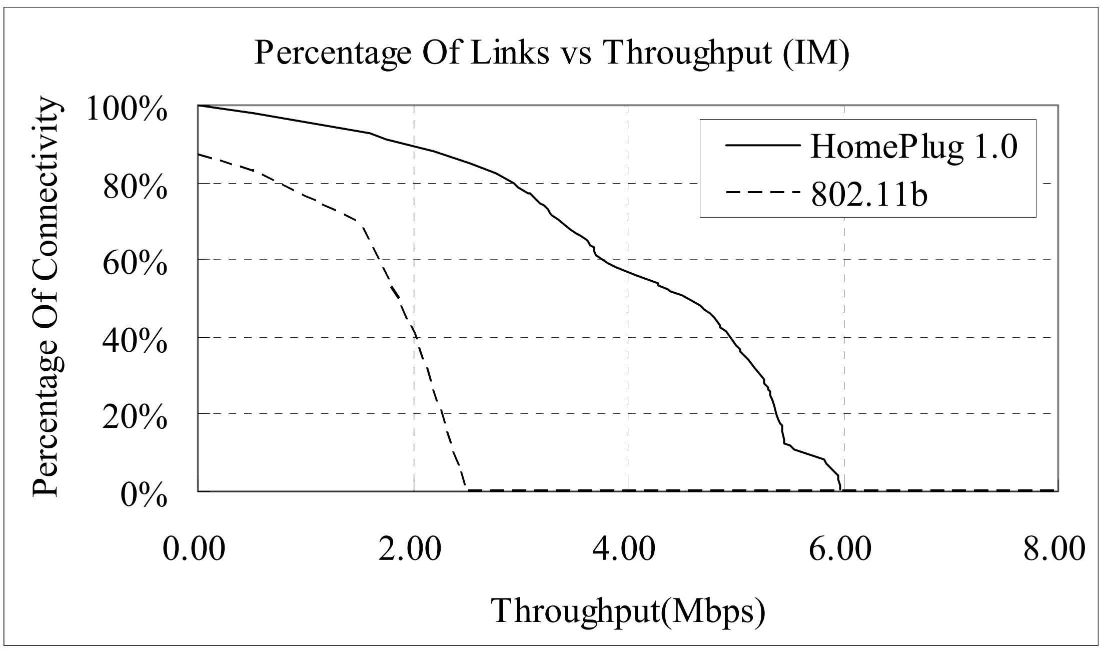


Figure 3-8: IEEE 802.11b and HomePlug 1.0 indoor percentage of link versus throughput (IM) comparison

<u>TCP Link Stability</u>

QoS algorithms usually deal with admission control and resource allocation. Admission control is concerned with the acceptance of new connections, while resource allocation deals with packet-level throughput, delay, and fairness. In either case, predictability is desirable.

Previous studies [14] showed that high channel error rate will reduce the effective bandwidth available for applications, thus negatively affecting application performance. This problem is even more severe for multimedia applications, which typically have bandwidth, delay, and jitter requirements for effective operations; it is important for them that the link remains stationary. However, the 802.11a/b displayed link instability when the PHY data rate was low. This section studies TCP link stability from the realtime capture of 802.11a, 802.11b, and HomePlug 1.0 packets.

During testing, when connection speeds were less than 3 Mbps, the wireless network became unstable. This might have been due to problems in rate adaptation. The 802.11b standard indicates around a 4 dB difference in signal strength between 11 Mbps and 2 Mbps mode. Since the signal strength changes continuously with time (for example, due to movement of people), PHY rate adaptation may cause packet drops that make the wireless network unstable when using TCP. During our tests, few links were found with throughputs in the 1 to 3 Mbps range. To find out the cause of this phenomenon, we used real time packet capture to monitor the TCP link stability. TCP throughput was measured at 100 msec intervals.

Figure 3-9 and 3-10 shows real time captures of typical high speed and low speed links of 802.11a. High speed links have a mean data rate of 20 Mbps, while it is 5 Mbps for low speed links. Note that the link performance from Mobile station to AP server and from AP server to Mobile station (separated by a 3 second delay) both are shown in the figures.

For high speed links, Figure 3-9 shows irregular throughput dropouts during transmission. For AP to mobile link, the throughput differences from one sample to the next can be as high as 22 Mbps. This kind of behavior is exhibited by TCP when packets are dropped. These instabilities would make it very difficult to guarantee QoS. Similar instability was observed for the link from Mobile to AP. A maximum throughput difference of 23 Mbps was observed in this case.

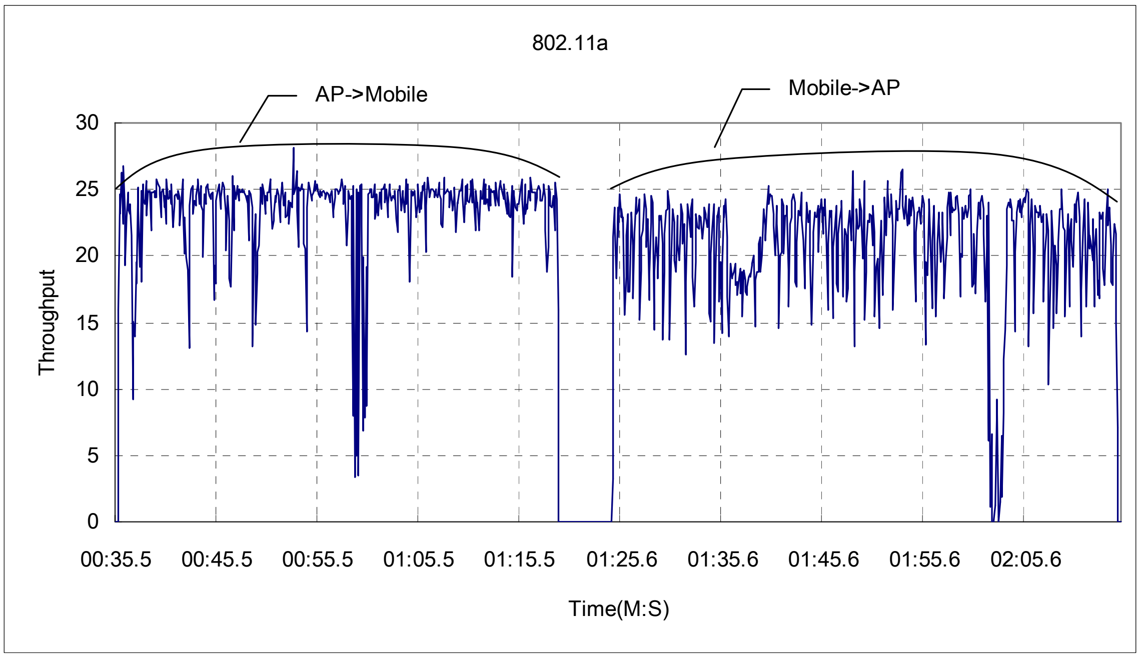


Figure 3-9: IEEE 802.11a high speed real-time capture

When the data rate of the link was low (Figure 3-10), the throughput also displayed large variations. A maximum throughput difference of 13 Mbps was observed. For Mobile to AP, a maximum throughput difference of 10 Mbps was observed.

Figure 3-11 and 3-12 shows the real time capture for a typical high speed and low speed links using IEEE 802.11b. High speed links for 802.11b are links with a data rate over 4 Mbps; low speed links for 802.11b are links with data rate lower than 2 Mbps. The same criteria were applied to the HomePlug 1.0 networks.

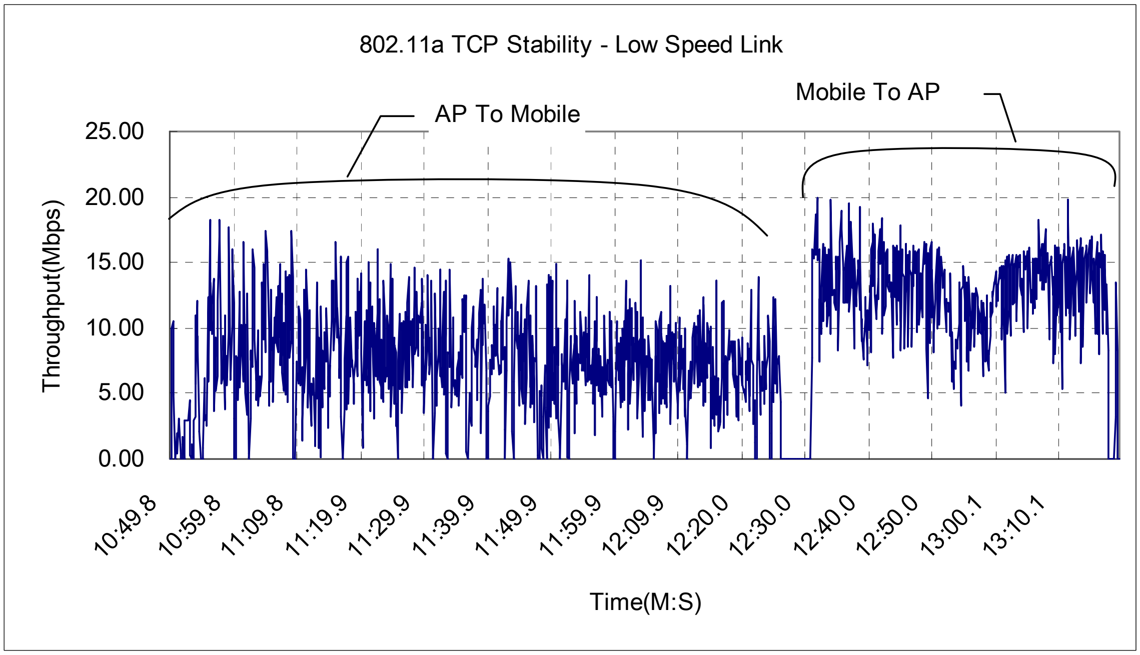


Figure 3-10: IEEE 802.11a low speed real-time capture

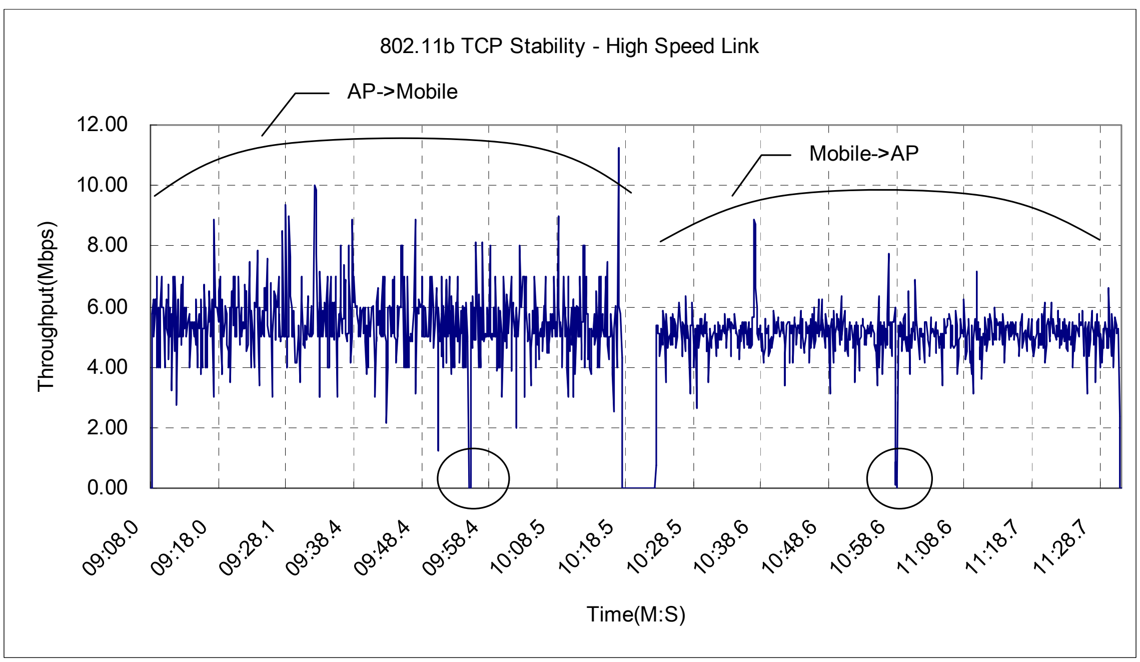


Figure 3-11: IEEE 802.11b high speed real-time capture

Figure 3-11 shows that the 802.11b links were typically more stable than the 802.11a links. However, there were two dropouts - one in the link of AP to Mobile, and the other one is in the link of Mobile to AP as indicated by circles in the figure. This type of dropout often appeared in other captures. In this figure, the maximum throughput differences observed was about 6 Mbps.

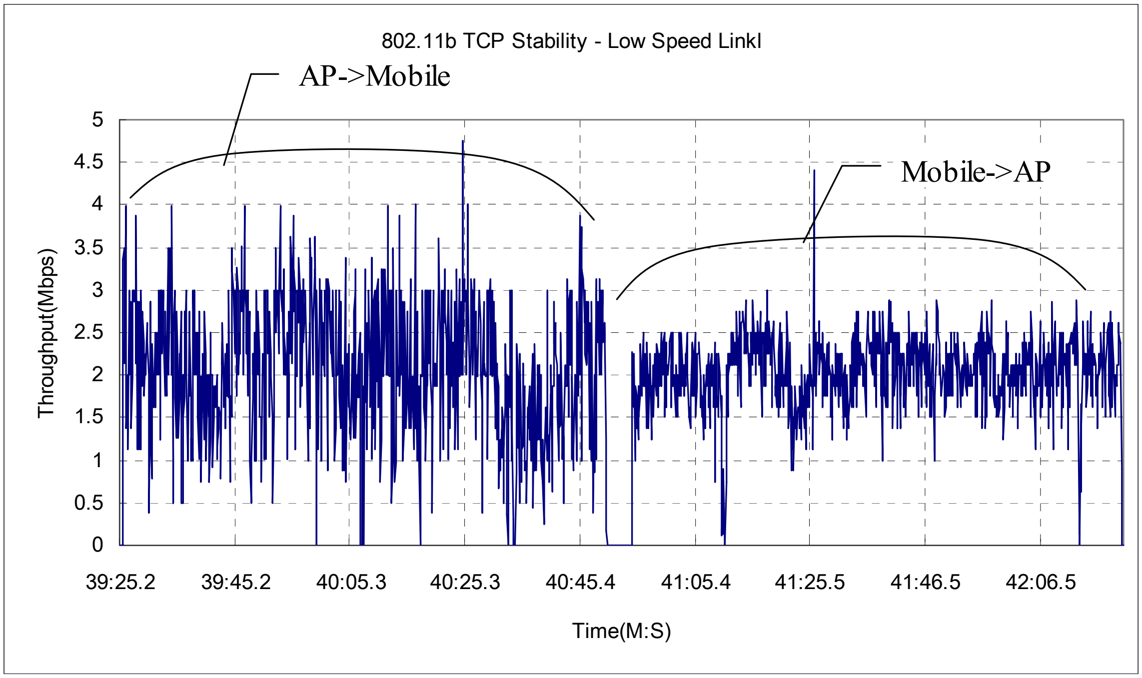


Figure 3-12: IEEE 802.11b low speed real-time capture

Figure 3-12 shows the real time capture for a typical low speed of 802.11b link. For AP to mobile link, the throughput differences can be as high as 3 Mbps. For Mobile to AP link, A maximum throughput difference of 3 Mbps was observed in this case.

Note that the throughput variation is critical at low data rates. User experience will be poor for links with such throughput variations. For example, under these marginal conditions, a file transfer might halt due to excessive packet drops. Applications can crash or show strange behaviors - an extremely unpleasant situation for the user. These links can be considered equivalent to no-connects, in the sense that users will not use these wireless links.

The HomePlug 1.0 TCP link stability is depicted in Figure 3-13 and 3-14. The figure shows that HomePlug 1.0 provides a fairly stable TCP link on high speed connections. There are no dramatic dropouts during the test. The maximum variation we observed in the figure was 3.6 Mbps for high speed links.

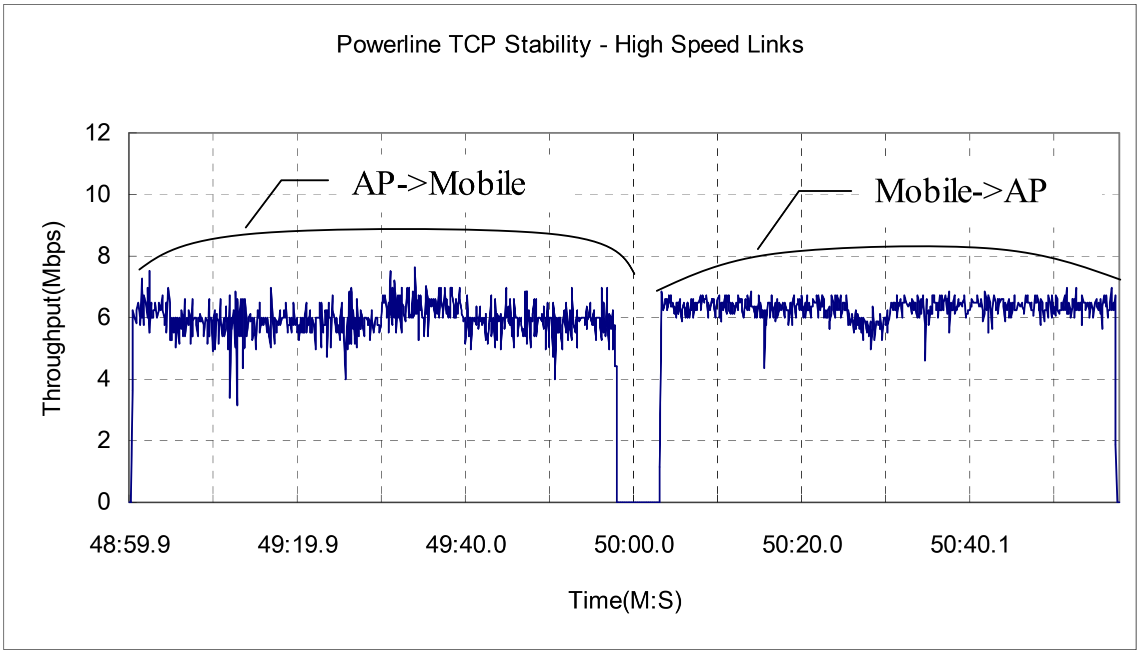


Figure 3-13: Homeplug 1.0 high speed real-time capture

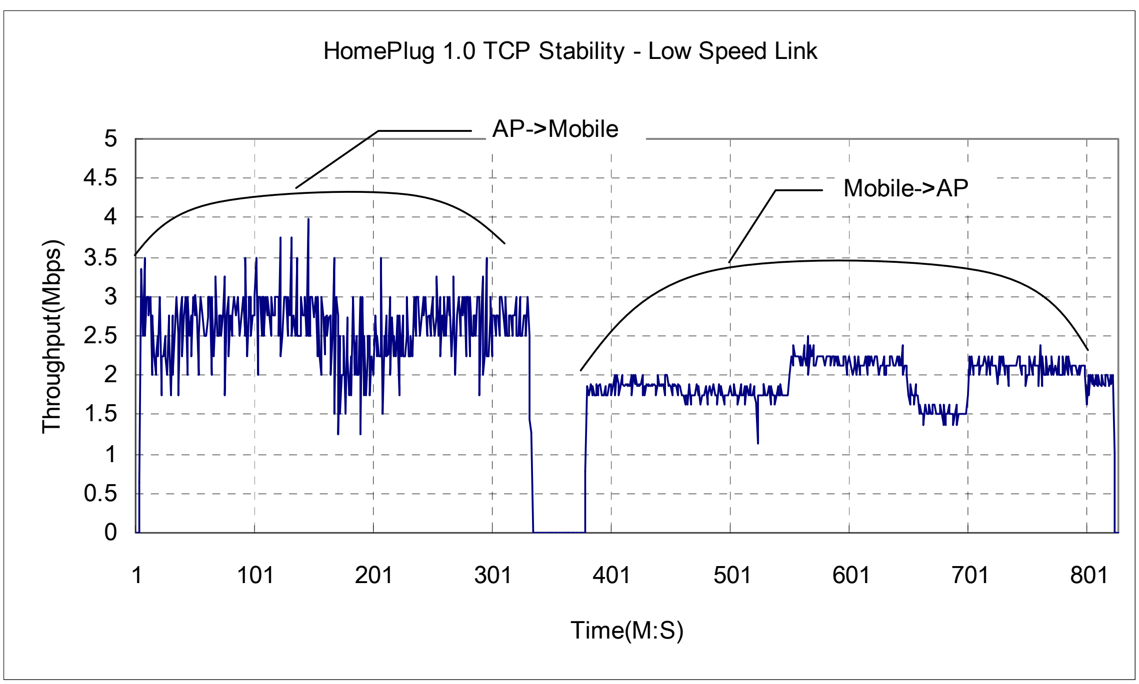


Figure 3-14: Homeplug 1.0 low speed real-time capture

Figure 3-14 shows an interesting stair-wise rate adaptation on the transmission from Mobile to AP. This effect is a manifestation of the channel estimation mechanism. Noise over power lines tends to vary with the line cycle. In HomePlug 1.0 stations, the channel estimation is done asynchronously using a small channel estimation packet. Thus the estimated data rate varies depending on when the packet arrives with respect to the line cycle. Thus different channel estimations (typically done every 5 seconds) will result in a different throughput, and hence this stepped behavior. However, the TCP links can be assumed to be stable between channel estimation cycles.

During our testing, we observed that IEEE 802.11b links that operated below 1 Mbps are highly unstable and are frequently marked by disconnects.

Discussion and Conclusion

The main goal of this chapter is to conduct a practical and theoretical comparison of the IEEE 802.11x and HomePlug 1.0 protocols and their capabilities in providing networking functionalities. This was done through theoretical analysis and by a thorough field test conducted in 20 houses to obtain the performance of IEEE 802.11b/a and HomePlug 1.0 products.

From the theoretical results shown in Table I, it can be derived that HomePlug 1.0 and 802.11x have similar maximum efficiency. The significantly higher maximum PHY data rate of 802.11a would indicate that it should perform better than the other two standards, but in field tests its coverage was not as good. In the field tests, the 802.11x products were configured as they came out of the box, and in some cases the wireless links may have used the RTS/CTS (Request To Send/Clear To Send) mechanism needed by 802.11x to handle hidden nodes. Use of RTS/CTS can have significant effects on the performance of the 802.11x protocols. RTS/CTS overhead degrades throughput and is most significant at high data rates, theoretically costing up to 10% in throughput performance. However, use of RTS/CTS can improve performance when there are collisions, as one can expect even on a single link when there are asynchronous, bidirectional exchanges (as TCP does by acknowledging received segments). 802.11x infrastructure mode (IM) also degrades performance, as opposed to modified IM (MIM) or ad hoc mode. In field tests, ad hoc mode was found to be nearly unusable over any but short distances. Fortunately, MIM would be typical for connection to an internet access point, which one might expect to bear the greatest amount of traffic. The wireless protocols have an advantage that these tests cannot show, is their use of multiple channels. With three channels available, one could expect the wireless protocols to perform better under congested conditions, whereas the PLC protocols use all of the available bandwidth for a single channel.

Coverage is much harder to predict theoretically, but field tests showed that HomePlug 1.0 had the best coverage, followed by 802.11b, with both trailed significantly by 802.11a. The latter only functioned over line-of-sight distances under 50 ft., and had complete coverage in only one of the 20 houses tested. IEEE 802.11b had severe coverage problems in houses over 4000 sq. ft., and had non-connects or marginal links in more than half of the houses tested. HomePlug 1.0 provided 100% coverage in all but two of the houses tested. This showed convincingly that the absence of an RTS/CTS mechanism in HomePlug 1.0 is not likely to be a problem for single home deployments. Whether or not it is an issue for multiple residences serviced by the same transformer remains to be seen.

Throughput showed more interesting behavior. The throughput of IEEE 802.11a was nearly always very high on links less than 50 ft. line of sight, but dropped to zero for links longer than this. 802.11b and HomePlug 1.0 showed crossovers in the percentage of links with data rates meeting some target rate. With its better coverage due to a more robust PHY and MAC, HomePlug 1.0 did better at meeting minimal data rates, but was surpassed by 802.11b at rates around 4 Mbps. For data rates of 5 Mbps and higher, HomePlug 1.0 retook the lead due to its slightly higher maximum data rate. HomePlug 1.0 also showed a much more gradual curve, due to its greater range of PHY data rate selections for adapting to the channel.

The continuum of link speeds also allowed HomePlug 1.0 to exhibit greater link stability (as measured by short term variability in TCP throughput). Effects of channel estimation at 5 second intervals and variability of the channel due to the 60 Hz line cycle were evident in these tests. IEEE 802.11b showed greater variability in both low- and high-speed links, most likely due to its sparser choices in adaptation to channel conditions. 802.11a had tremendous and frequent speed fluctuations, which brings into question its ability to offer QoS guarantees for multimedia applications.

Both PLC and wireless technology have scope for improvements over the exist standards as evaluated in this paper. For PLC networks, larger bandwidth, the use of higher order modulation, more powerful forward error correction technique, and improved channel estimation can substantially improve performance. Wireless technologies can also invoke similar enhancements along with larger transmit power and antenna diversity to achieve significant improvement. However, increasing crowding in the 2.4 GHz ISM bands used by 802.11b/g are likely to degrade their performance in locations where they must coexist with competing transmitters (including BlueTooth, HomeRF, 2.4 GHz phones, microwave ovens, etc.).

Despite its problems, there are still situations for which wireless technologies are needed. Mobile users with handheld devices and nomadic users without access to power outlets will require wireless connectivity. For nomadic users who are able to plug into the home power distribution system, however, PLC offers a robust, stable, and speedy alternative. PLC solutions will be even more desirable for providing QoS support for multimedia applications with the future and emerging PLC technologies offering data rates in excess of 50 Mbps.

CHAPTER 4
PERIODIC CONTENTION FREE MULTIPLE ACCESS FOR POWER LINE COMMUNICATION NETWORKS

Introduction

Recently, applications over Power Line Communication (PLC) networks have drawn much interest in academe and industry, not only because of their convenience (connecting PLC capable devices requires no new wires), but also because all electrical devices have to connect to a power outlet eventually. This technology makes implementing a digital home entertainment center more realistic than ever – HDTVs in different rooms are now able to share digital content from one set-top box without rewiring or setting up wireless access points. MP3 players can access music data through PLC networks from different rooms playing different music. PLC networks make the smart home possible [12].

However, PLC technology is still evolving and many issues remain unsolved. The hostile environment of PLC channels makes reliable data transmission difficult. Much effort is required to ensure data transmission is correct and efficient. The *HomePlug Alliance* set standards for 14Mbps class data transmission. This resulted in a variety of PLC devices for computer-oriented network communications. Their performance and reliability is comparable to wireless networks [15]. However, the *HomePlug 1.0* protocol is not suitable for video playback because of its limited network throughput.

To support multimedia streaming for homes, the *HomePlug Alliance* is now developing the standard for a second generation of PLC devices capable of delivering multiple HD-Videos through newly designed chipsets called *HomePlug AV*, which supports up to 200Mbps raw data rate. The goal of the *HomePlug AV* is to make PLC devices capable of delivering two hours of HD-Video without video frame drops, while simultaneously delivering one or more data streams of various data rates and traffic types. To make above mentioned goals possible, the efficient cooperation of high speed PHY and MAC protocols becomes important. However, current existing MAC protocols do not provide such functionalities and they are not suitable for *HomePlug AV*. The need for a new protocol is urgent.

The PLC channel is known for its hostile nature in transmitting electrical signals. Protocols designed for other media may not be suitable for PLC. PLC channels are in some ways similar to wireless channels - both of them face hidden node problems, near-far effects and other channel imperfections. However, PLC network nodes tend not to move. It is unlikely that simply applying protocols designed for another medium would result in good performance in the PLC environment; the overhead may be too high or the assumptions about noise may be too optimistic for PLC networks.

In light of PLC's unique characteristics, we developed a new protocol – Periodic Contention-Free Multiple Access (PCF/MA). PCF/MA is an explicit R-ALOHA-like protocol specifically designed for the PLC network - we propose an RTS/CTS-like scheme in the reservation stage to prevent hidden node problems, and a delayed NACK mechanism to conquer near-far effect. Performance of the proposed protocol is evaluated by event driven computer simulation and by mathematical analysis. The simulation results show that 85Mbps MAC throughput under 100Mbps channel data rate can be obtained, even when there are hidden nodes in the network. To provide smooth video delivery, we propose a mathematical estimation of the required delay in playback time and the amount of playback buffer with tight bandwidth reservation. Our simulation shows that an 100Mbps channel can deliver up to 9 MPEG-2 video streams simultaneously without dropping any video frames, however, using Modified CSMA/CA (MCSMA/CA) on the same environment supports 7 video streams only because of its unfairness, unpredictable behaviors and high overhead.

This chapter is organized as follows. A brief survey of existing protocols is given in section 2. Section 3 provides the *PCF/MA* methodology, analysis, and performance evaluation and simulation results. The discussion and conclusion is given in section 4.

Previous Works

Packet contention techniques such as Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) and ALOHA find widespread use in data communications, including the first generation PLC networks. Both have the ability to serve a large number of terminals. While they function with little to no coordination, packet contention results in unpredictable behaviors such as unfairness and possibly long delays, which make them unsuitable for delay-sensitive packet delivery. To solve these problems, one can use packet scheduling or reservation-based methods.

Reservation-based methods were designed to remove delay-sensitive, high-rate connections from the random-access competition for channel time. The time domain is partitioned into a reservation and data session; each session is then again partitioned into slots. In these methods some slots are reserved for specific stations. Other stations are restrained from using a reserved slot. Generally, these MAC schemes can be categorized according to whether the reservation is done implicitly or explicitly.

Packet Reservation Multiple Access (PRMA) is a centralized and slotted multiple access protocol that allows voice and data sources to share the same access channel at the talkspurt level, targeted for wireless local area networks, and was originally proposed by Goodman D. J., Valenzuela R. A., Gayliard K. T., and Ramamurthi B. [16]. PRMA utilizes the speech on-off activity to improve bandwidth efficiency and system capacity. Though PRMA is an implicit reservation-based algorithm, between talkspurts stations must contend for reservation, leading to packet loss and degraded speech quality. It also relies on central control to broadcast an ACK at the end of each slot, hence it may not be appropriate for a home ad-hoc environment. A few modified PRMA protocols such as D-PRMA were proposed to support mobile ad-hoc environments. These emphasize talkspurt-level packet reservation without relying on a central entity. They also try to deal with hidden/exposed node problems by asserting an RTS/CTS-like dialog between sender and receiver.

Hidden node and near-far effects make implicit reservation difficult in PLC networks. Impaired receivers may not agree on the current network state. Further, as the PLC network is targeted to be an in-home networking infrastructure, it is not desirable to have a central control device just for medium access, especially if the central control device may become a performance bottleneck and a single failure point of a network. Lack of a central control device makes collision detection even more difficult since PLC devices are not able to detect signal collisions during transmission.

Explicit reservations like “Five-Phase Reservation Protocol” [17] try to implement a dynamic parallel reservation with arbitrary scalable network size. The five phases mentioned in the literature are: Reservation Request (RR); Collision Report (CR);

Reservation Confirmation (RC); Reservation Acknowledgement (RA); Packing and Elimination (P/E) phase. However, this protocol makes assumptions inapplicable to the PLC environment. Firstly, the protocol works on a 2 hops radius network, with perfect timing. Secondly, near-far effects do exist in the PLC network especially when a robust signal modulation is applied.

The above discussion leads to the new protocol design based on an explicit reservation scheme. Robert's reservation scheme and R-TDMA are good candidates; their merit is that a few slots are dedicated for reservation purpose and the rest of them are for data transmission. However, these schemes lack distributed control, hidden-node prevention and ignore near-far effects, which makes them inappropriate for PLC networks. Careful re-design of Robert's and the R-TDMA protocols results in the new PCF/MA protocol as stated in the next section.

Proposed PCF/MA Protocol

Though the power line channel is similar to a wireless channel, there are some unique characteristics specific to the power line channel. Firstly, the characteristics of the PLC channel in a home are in general steady over time rather than dramatically changing as in the wireless channel, that is, while the attenuation of the signal may be affected by near-by electronics in the short term, from a long term point of view the attenuation is almost identical. This makes long-term bandwidth scheduling possible. Secondly, the devices in PLC networks are quasi-stationary, that is, the stations in the PLC network are not moving as constantly as devices in the wireless network, thus the bandwidth fluctuations are much less erratic [15]. These two characteristics place PLC networks between Ethernet networks and wireless networks in that they have the properties of a fixed network topology but with more noise and attenuation.

The Concept of PCF/MA

The PCF/MA protocol is a distributed, contention-free protocol which uses a two-way handshake reservation process to establish TDMA slot assignments. The reservation process for a given node only involves nodes within a one hop radius.

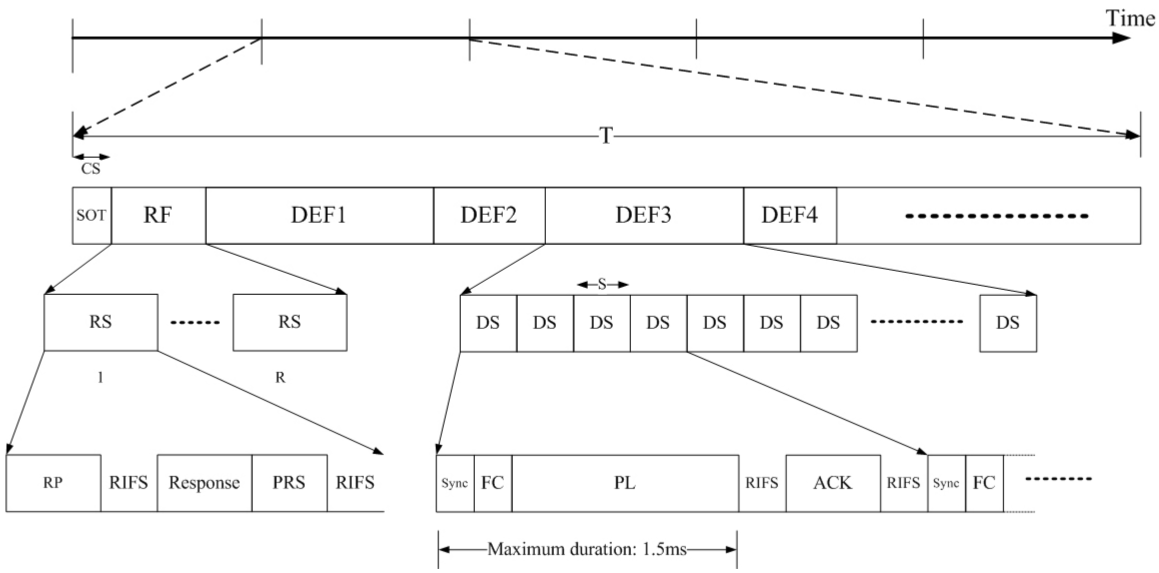


Figure 4-1: PCF/MA frame structure

Fig. 4-1 shows the protocol's frame structure. Time is divided into several TDMA sessions with duration of $T\,\mu s$. Each session is partitioned into a Reservation Frame (RF) and one or more Data Exchange Frames (DEFs). Before the RF is a Starting of TDMA (SOT) frame issued by the leader station on the network that lasts $38.4\,\mu s$[4]. Table 4-1 summarizes the parameters used in this research.

Distributed Admission Control

The RF is divided into *R* Reservation Slots (RS). An RS does not correspond to a data slot but rather serves as a period of time for the making a reservation. This process involves exchange of Reservation Packets (RP) and ACKs. An RP contains the Source

[4] The parameters we used in this paper follows those of *HomePlug 1.0* as published in [19]

Address (SA), Destination Address (DA), Starting reserved Slot number (SS) and Total of reserved Slots (TS) as depicted in Fig.4-2. To illustrate the reservation process more clearly, we also depict the reservation flowchart in Fig.4-3.

Table 4-1: HomePlug 1.0 and PCF/MA parameters

	HomePlug 1.0	PCF/MA	MCSMA/CA
SYNC	38.4 μs	38.4 μs	38.4 μs
FC	33.6 μs	33.6 μs	33.6 μs
RIFS	26 μs	26 μs	26 μs(SIFS)
DEL	72 μs	72 μs	72 μs
Max MPDU Duration	1.5*ms*	1.5*ms*	1.5*ms*
CRS	35.84 μs	200 μs(RS)	35.84 μs
DIFS	35.84 μs	-	35.84 μs
TDMA Session Length	-	$T \mu s$	-
Reservation Slots	-	R	-
SOT	38.4 μs(SYNC)	38.4 μs	-
Priority Resolution	35.8 μs	35.84 μs	-

In Fig. 4-3, the dashed box represents a state, the rectangle and lattice boxes represent processes and decisions. The oval boxes represent terminals, which lead to other states.

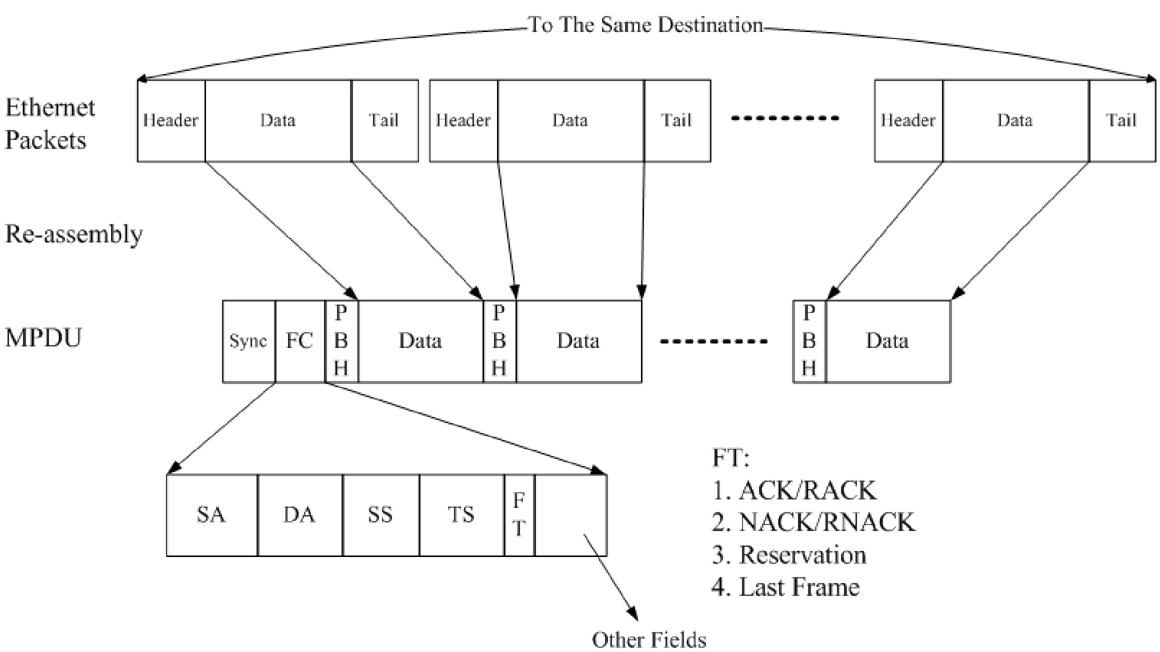


Figure 4-2: MPDU process and format

If a node wants to reserve a DEF, it first listens to the network for at least one TDMA session. During this period, it monitors the network activities and learns the reservation of each station as depicted in Fig.4-3 CONDIRIONING state. When the RF begins, it chooses a random RS and broadcasts an RP to the networks.

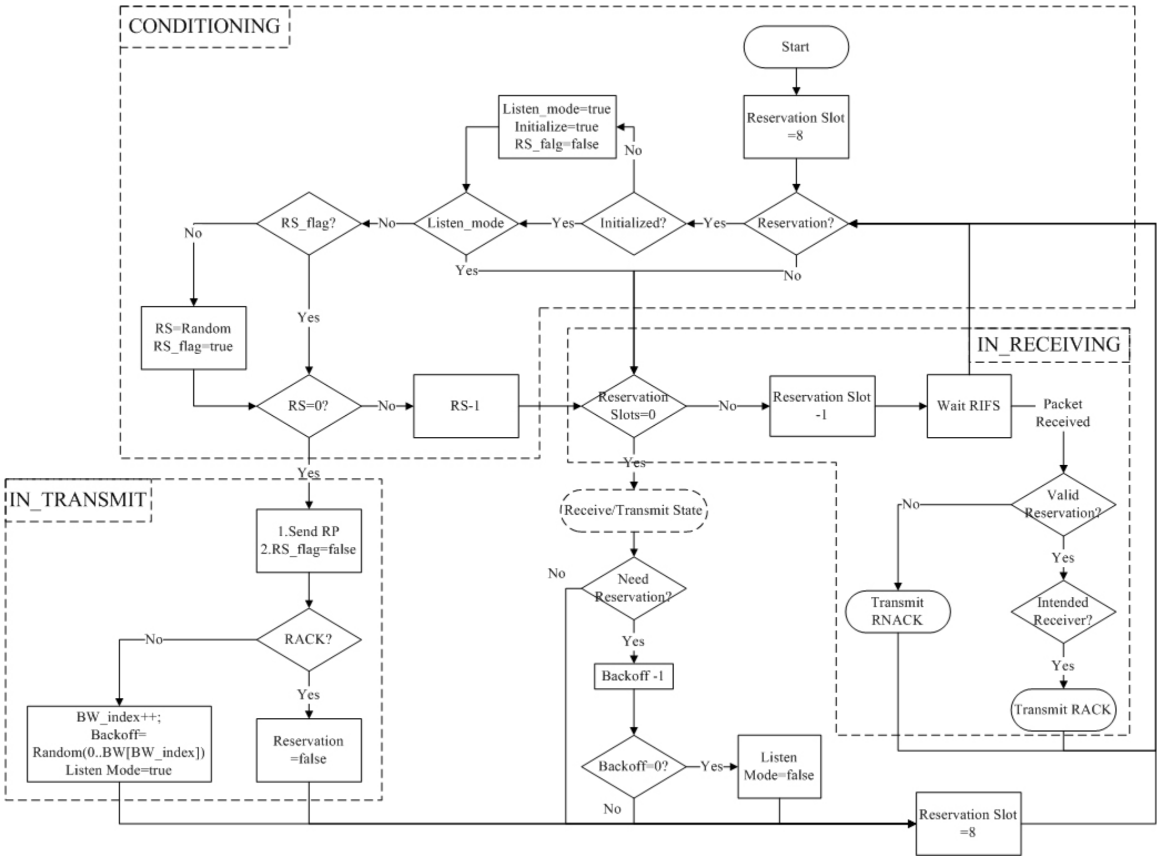


Figure 4-3: Reservation process flow chart

The destination receives and compares the request with its internal table. If the reservation does not conflict with previous reservations, the node sends a Reservation Acknowledgement (RACK) to the transmitter and completes the reservation process. This process is illustrated in Fig.6.3 IN RECEIVING state. Other stations in the network also listen and broadcast a Reservation Negative ACKnowledgements (RNACK) packet to jam the possible RACK signal if they detect a conflict. To ensure the requester receives the RNACK packet successfully rather than interpret the signal as noise, we use a delayed NACK methodology. We will illustrate this method in hidden node resolution section.

If a collision/RNACK happens, the requester will move to backoff stage with random backoff time. The backoff algorithm is the same as in the CSMA/CA, that is, after each unsuccessful reservation, the backoff window size is doubled, up to a maximum value $2^m W$, where W equals to (CW_{min} + 1) and $2^m W$ equals to (CW_{max} + 1) and the backoff time is uniformly chosen in the range (0*;CW*-1). Since PCFMA is a persistent reservation scheme, the competition between stations suppose to be small, thus the maximum window size is set to 16. Once the backoff window size reaches CW_{max}, it will stay at the value of CW_{max} until it rests.

Data Exchange

The data exchange period is divided into N data slots(DSs), each with a duration of $S\,\mu s$. A DEF is composed of several DSs. The length of a DEF is decided by the TS field in owner's RP. A DEF always begins at the start of a DS.

If a node successfully reserved a period of time for transmission, it listens and waits until its DEF time, then starts to transmit MAC Protocol Data Units (MPDUs). If the receiver receives a successful MPDU, it will send an ACK. The sender can continue until the end of its reserved DEF.

At the end of a transmission, the sender broadcasts a Last Frame MPDU to the network. The receiver also broadcasts a Last Frame MPDU to eliminate possible hidden node problems and allow all nodes to update their internal tables. When a station ungracefully terminates its connection, affected stations (usually it is the receiver) also broadcast a Last Frame MPDU during the reserved slots to synchronize internal tables with other stations.

MAC Protocol Data Unit

MPDUs combine several Ethernet packets belonging to the same path into a jumbo packet to increase overall efficiency. The original Ethernet packet header and trailer is removed and a new small header called PHY Block Header (PBH) containing the sequence number of the original packet is added to indicate the order of the packet. After this process, the original Ethernet packet becomes a new block called a PHY Block (PB). The process is depicted in Fig. 4-2.

Several PBs are then combined with the original Ethernet packet sequence into a jumbo packet with a common header to become an MPDU ready for transmission. Based on the current transmission speed, as many Ethernet packets as possible are combined until the transmission duration reaches 1.5*ms*.

Solutions to the Hidden Node Problems and Near Far Effect

A robust signal modulation is required due to the attenuation and noise on PLC channels. However, this makes detecting packet collision difficult - the signal modulation process may interpret contending packets as noise and remove the noise to restore the intended signal! This phenomenon causes near-far effects and hidden-node problems.

A common solution to the hidden-node problem is the use of a RTS/CTS handshake before data transmission. The purpose of RTS/CTS is to notify nearby stations of the incipient data transmission period so that those who are not involved in the data exchange will avoid the channel during that period of time. Signal reception of mobile devices in wireless networks usually suffered from nearby activities, reception is expected to be constantly changing and thus exchanging RTS/CTS packets before every data transmission is required.

In PLC networks, we adopt the similar RTS/CTS scheme but only in the reservation stage. In the RF, nodes want to make reservations broadcast RPs to the network. The RP also act as “Request to Send” signals as in the RTS/CTS scheme. Nodes that receive this packet avoid the reserved period. Nodes outside the broadcasting range may not aware of the reservation resulting in inconsistent databases. If they do not want to make new reservations, the inconsistency becomes irrelevant. If one of the nodes wants to make a new reservation those conflicts with the scheduled reservations, all nodes that hear this RP broadcast RNACK. To deal with the possibility that the RNACK is not heard by the transmitter because of near-far effect, a delayed RNACK scheme is applied as shown in Fig 6.1.

The delayed RNACK scheme use Priority Resolution Signal (PRS) to inform the requester the received RACK is incorrect. All stations object to the reservation should wait 98ʹ*s* and broadcast a PRS after the PR packet. The requester should listen to the network after it broadcasts PR packets. If a RACK is received, it should keeps listening to the network for a PRS packet duration (35.84 μs). If it senses the PRS packet, it determines the reservation is invalid; otherwise it is a successful reservation.

If the intended receiver also recognizes the reservation is invalid, it should reply the requester a RNACK with corrected SS and TS fields to make requester database more consistent to the network status.

Approximate Performance Analysis and Simulation Results

To calculate the maximum throughput, we assume there are always data to send for each node. We define efficiency as the ratio of time spent on transmitting payload to the total time spent on the whole data exchange process. The parameters we used in the calculation is summarized in Table 6.1.

The minimum reservation slot time can be obtained by adding a RP duration, two RIFS, a Response, and a PRS which leads to (72 + 26 + 72 + 35.84 + 26) μs = 231.84 μs. A successfully packet transmission process requires a MPDU, an ACK and two RIFSs. The total time required for this process is 1624 μs when sender sends a maximum length MPDU. The total DAE slots in a TDMA session can be calculated by $\left\lfloor \frac{\text{T - 231.84R - 29}}{s} \right\rfloor$ assume R reservation slots. If there are m transmitters, and the bandwidth is evenly distributed to all transmitters, then a transmitter can have b slots, where

$$b = \frac{\left\lfloor \frac{T - 231.84R - 29}{s} \right\rfloor}{m} \text{ slots}$$

The total allowed transmitting time for each data stream can be calculated by $b \times s$. A sender can transmit p maximum size of MPDUs in a TDMA session, where

$$p = \frac{\left\lfloor \frac{T - 231.84R - 29}{s} \right\rfloor s}{1624m}$$

Since the maximum MPDU has duration of 1*:5ms* and the overhead of a MPDU is 72 μs, thus the total time spent on data transmission for each node t can be calculated by

$$t = \left\lfloor \frac{\left\lfloor \frac{T - 231.84R - 29}{s} \right\rfloor s}{1624m} \right\rfloor 1428\ \mu s$$

Since there are m nodes, the protocol efficiency E can be calculated by

$$E = \frac{\left\lfloor \frac{\left\lfloor \frac{T - 231.84R - 29}{s} \right\rfloor s}{1624m} \right\rfloor 1428m}{T} \tag{4-1}$$

If we ignore the *floor* ($\lfloor . \rfloor$) operation in Eq.4-1, we can calculate the maximum efficiency E_{max} as follows:

$$\begin{aligned} E_{\max} &= \frac{\left(\frac{\left(\frac{T - 231.84R - 29}{s} \right) s}{1624m} \right) 1428m}{T} \\ &= \frac{0.88(T - 231.84R - 29)}{T} \end{aligned} \tag{4-2}$$

$$\text{when } T \to \infty$$

$$E_{\max} \approx 88\%$$

One must note that each DEF starts from the beginning of a DS, if a node does not fully utilize the reserved DS, the resulted efficiency will decrease.

From Eq.6.1 we learned that the efficiency of PCF/MA protocol is determined by *R*, *m* and *T*, where *m* is dynamic. To obtain the optimal performance, we derived a few mathematical forms as follows.

<u>Determine Parameter R</u>

To minimize fixed overhead, a small *R* would be desirable considering the number of contenders is relatively small since all reservations are persistent. However, a small *R* could make the system unstable when the number of contenders increases.

It is obvious that when the number of new data streams n is larger than the available reservation slots r, at least two stations will have a reservation packet collision. We call this over-saturation. However, since the reservation session comes every $T\,\mu s$, as long as the number of new data streams is less than 1 every TDMA session and at least 1 data stream successfully makes a reservation, the waiting queue should converge. When there are n new data streams, this problem can de described as the probability that at least one station does not conflict with other stations. To determine this probability, we derive the following calculations.

The permutations P of choosing n numbers from r numbers is

$$P = r^n \tag{4-3}$$

The generation function $G(x)$ corresponding to the random choice of n numbers from r numbers with unlimited repetition can be written as

$$\begin{aligned} G(x) &= (1 + x + \frac{x^2}{2!} + \frac{x^3}{3!} + \cdots) \\ &= e^{rx} \\ &= \sum_{n=0}^{\infty} \frac{r^n}{n!} x^n \text{ , where } n \geq r \end{aligned}$$

We want to know the probability that at least one RP does not conflict with others. That is, at least one node chooses a random number that is different than the others. We can first calculate the permutations p of choosing n numbers from r numbers such that a number is chosen 0 times, 2 times, 3 times...., then calculate *(1- p)/P* to get the desired probability. The generation function of p with unlimited repetition can be written as

$$\begin{aligned} g(x) &= (1 + \frac{x^2}{2!} + \frac{x^3}{3!} + \cdots)^r \\ &= (e^x - x)^r \end{aligned}$$

$$
\begin{aligned}
&= \sum_{i=0}^{r}\tbinom{r}{i}(-1)^i x^i e^{(r-i)x} \\
&= \sum_{i=0}^{r}\tbinom{r}{i}(-1)^i x^i \sum_{j=0}^{\infty}\tfrac{1}{j!}(r-i)^j x^j \\
&= \sum_{j=0}^{\infty}\tfrac{1}{j!}\sum_{i=0}^{r}(-1)^i\tbinom{r}{i}(r-i)^j x^{(i+j)} \\
&= \sum_{j=0}^{\infty}\tfrac{1}{j!}((-1)^0\tbinom{r}{0}(r-0)^j x^j) + \\
&\quad (-1)^1\tbinom{r}{1}(r-1)^j x^{(j+1)} + \\
&\quad (-1)^2\tbinom{r}{2}(r-2)^j x^{(j+2)} + \\
&\quad (-1)^3\tbinom{r}{3}(r-3)^j x^{(j+3)} + \\
&\quad \cdots + (-1)^r\tbinom{r}{r}(r-r)^j x^{(j+r)}) \\
&= \tfrac{1}{0!}[(-1)^0\tbinom{r}{0}(r-0)^0 x^0 + \\
&\quad (-1)^1\tbinom{r}{1}(r-1)^0 x^{(0+1)} + \\
&\quad (-1)^2\tbinom{r}{2}(r-2)^0 x^{(0+2)} + \cdots \\
&\quad (-1)^r\tbinom{r}{r}(r-r)^0 x^{(0+r)}] + \\
&\quad \tfrac{1}{1!}[(-1)^0\tbinom{r}{0}(r-0)^1 x^1 + \\
&\quad (-1)^1\tbinom{r}{1}(r-1)^1 x^{(1+1)} + \\
&\quad (-1)^2\tbinom{r}{2}(r-2)^1 x^{(1+2)} + \cdots \\
&\quad (-1)^r\tbinom{r}{r}(r-r)^1 x^{(1+r)}] + \\
&\quad \vdots \\
&\quad \tfrac{1}{k!}[(-1)^0\tbinom{r}{0}(r-0)^k x^{(0+k)} + \\
&\quad (-1)^1\tbinom{r}{1}(r-1)^k x^{(1+k)} + \\
&\quad (-1)^2\tbinom{r}{2}(r-2)^k x^{(2+k)} + \cdots \\
&\quad (-1)^r\tbinom{r}{r}(r-r)^k x^{(r+k)} + \cdots] + \\
&\quad \vdots
\end{aligned} \tag{4-4}
$$

Since we are looking for the coefficient of $x^n/n!$, rearrange Eq.4-4 yields

$$
\begin{aligned}
g(x) = \quad & \tfrac{1}{0!}(-1)^0\tbinom{r}{0}(r-0)^0x^0 + \\
& [\tfrac{1}{0!}(-1)^1\tbinom{r}{1}(r-1)^0 + \\
& \tfrac{1}{1!}(-1)^0\tbinom{r}{0}(r-0)^1]x^1 + \\
& [\tfrac{1}{0!}(-1)^2\tbinom{r}{2}(r-2)^0 + \\
& \tfrac{1}{1!}(-1)^1\tbinom{r}{1}(r-1)^1 + \\
& \tfrac{1}{2!}(-1)^0\tbinom{r}{0}(r-0)^2]x^2 + \\
& \vdots \\
& [\tfrac{1}{0!}(-1)^k\tbinom{r}{k}(r-k)^0 + \\
& \tfrac{1}{1!}(-1)^{(k-1)}\tbinom{r}{k-1}(r-(k-1))^1 + \\
& \vdots \\
& \tfrac{1}{k!}(-1)^{(k-k)}\tbinom{r}{k-k}(r-(k-k))^k]x^k + \\
& \vdots \\
= \quad & \textstyle\sum_{n=0}^{\infty}\sum_{l=0}^{n}\frac{1}{l!}(-1)^{(n-l)}\binom{r}{(n-l)}(r-(n-l))^l x^n \\
= \quad & \textstyle\sum_{n=0}^{\infty}\frac{1}{n!}\sum_{l=0}^{n}\frac{n!}{l!}(-1)^{(n-l)}\binom{r}{(n-l)}(r-(n-l))^l x^n
\end{aligned}
\tag{4-5}
$$

The coefficients of $x^n/n!$ becomes

$$
p = \sum_{l=0}^{n}\frac{n!}{l!}(-1)^{(n-1)}\binom{r}{n-l}(r-(n-l))^l \tag{4-6}
$$

From Eq.4-6 and 4-3, we can get the desired probability as

$$
1-\frac{p}{P} = 1-\frac{\sum_{l=0}^{n}\frac{n!}{l!}(-1)^{(n-1)}\binom{r}{n-l}(r-(n-l))^l}{r^n} \tag{4-7}
$$

To verify formula 6.7, we simulated a constant number of contenders. The simulator simulates one million reservation sessions. The number of contenders is the same from session to session. The simulator counts the event that at least one contender successfully makes its reservation. The number of reservation slots is increased from 2 slots to 9 slots. Simulation results is shown in Fig.4-4

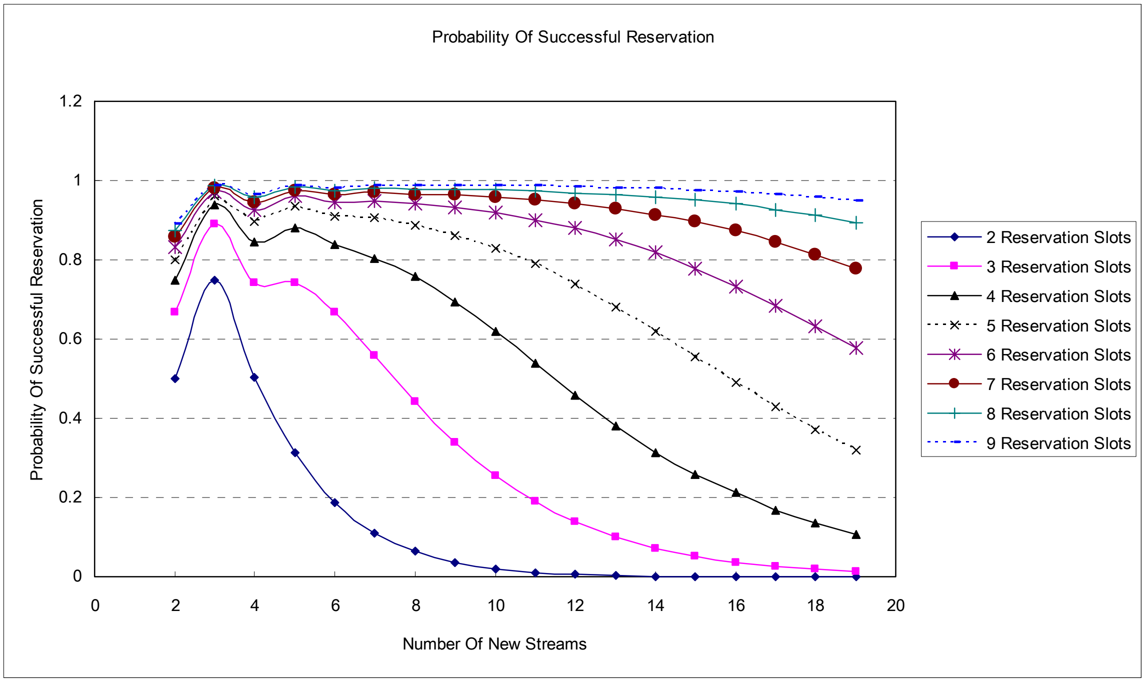


Figure 4-4: Probability of successful reservation

Fig.4-4 shows that even $r = 2,\ n = 4$, the probability that at least one station successfully makes its reservation is about 75%. When $r = 9$, even if $n = 20$, the probability is as high as 90%. To verify our assumption, we simulate a Poisson arrival of λ=1, r=8, PCF/MA remains stable for one million reservation sessions. When we increase the inter-arrival rate λ to 3, r have to increase to 16 to make PCF/MA stable. However, 1 new streams for each reservation session is equal to 40 new streams per second if $T = 25$. We believe 40 new streams per second are more than enough for a home network.

From the simulation results, we chose $r = 8$ for the optimal parameter for PCF/MA.

Near-Far Effect Modeling

Hadzi-Velkov, Z. and Spasenovski, B. [18] successfully modeled the 802.11b near-far effect under a Rayleigh Fading Channel. It is known that different modulation and transmitting power under different channels will have different near-far effects. To investigate the performance impact with and without the near-far effect, we choose a near-far effect model described as the probability a receiver captures the desired packet (P_{cap}) in the presence of interfering packets (R_i)

$$
\begin{aligned}
&P_{cap} = \text{Probability}\left(\frac{R_d - \sum_{i=0}^{m} R_i}{R_d} \right) \\
&\text{when } \frac{\sum_{i=0}^{m} R_i}{R_d} \geq C_{TH}
\end{aligned}
\qquad (4\text{-}8)
$$

where R_d is the received power level of the desired packet and the C_{TH} is the capture threshold.

Extra Allocation

Since the packet loss rate is high (10^{-3} packets per second or higher), a silent retransmission is required when the receiver receives corrupt packets. This requires data streams to allocate extra bandwidth despite the estimated bandwidth.

Assume packet loss rate is P and the number of transmission attempts is R. Suppose at one instance, the sender sends N PBs in a MPDU, the amount of over allocation for next transmission is NP. Thus for quasi-error free environment with unlimited retries, we need to over allocate (O) resources by

$$
\begin{aligned}
O &= \left| NP + NP^2 + NP^3 + \cdots NP^R + \cdots \right| \\
&= \left\lceil NP(1 + P + P^2 + \cdots + P^{R-1} + \cdots \right\rceil \\
&= \left\lceil NP\left(\frac{1}{1-P} \right) \right\rceil
\end{aligned}
\qquad (4\text{-}9)
$$

When $P << 1$, Eq.4-9 becomes $\lceil NP \rceil$

We depict the computer simulation and theoretical calculations in Fig.6.5. We simulated 10^6 TDMA sessions to collect meaningful statistics. The simulation results closely match to the mathematical calculation.

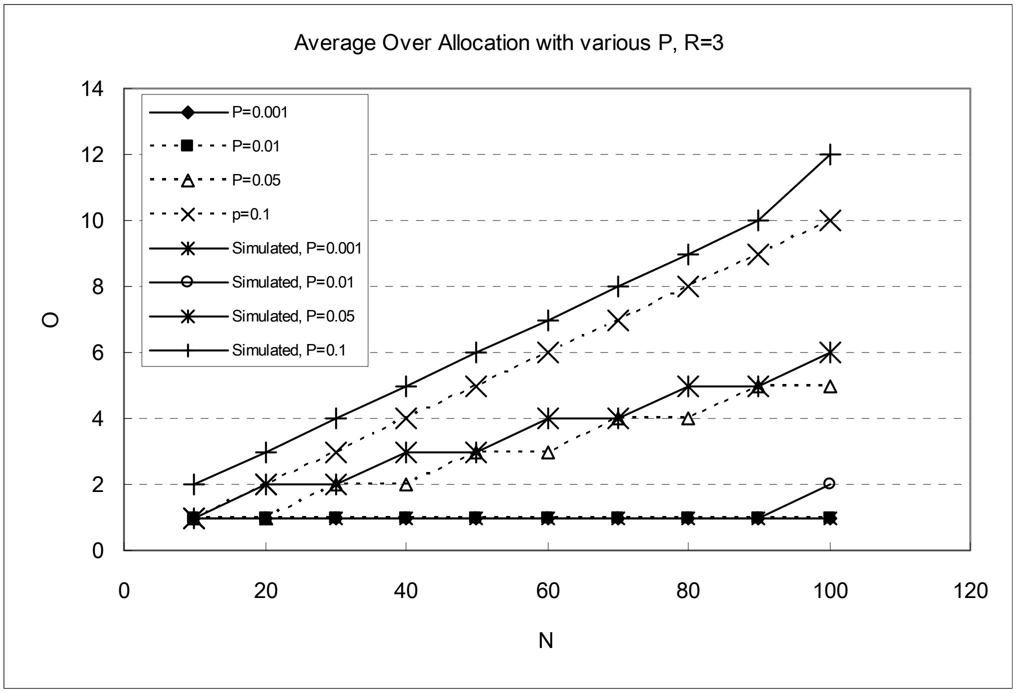


Figure 4-5: Maximum over allocation PBs various PB loss rate comparisons

<u>Data Stream Model</u>

Event driven simulation was used to investigate PCF/MA performance and to compare PCF/MA with a CSMA/CA adapted to PLC. The parameters of a PLC network are assigned path by path to simulate real world PLC environment.

To estimate the maximum throughput, we applied *Always On* data stream to both PCF/MA and MCSMA/CA. By Always On data stream we mean the sender has unlimited input buffer and always obtains maximum duration of MPDUs once it has chance to transmit. We assume all channel data rates are 100Mbps and the received power levels from a node to other nodes are randomly generated with Poisson distribution with λ= 50. There are no hidden nodes in this simulation, the packet loss rate is 0.001 and C_{TH} = 0.3, the simulated results is depicted in Fig.4-6.

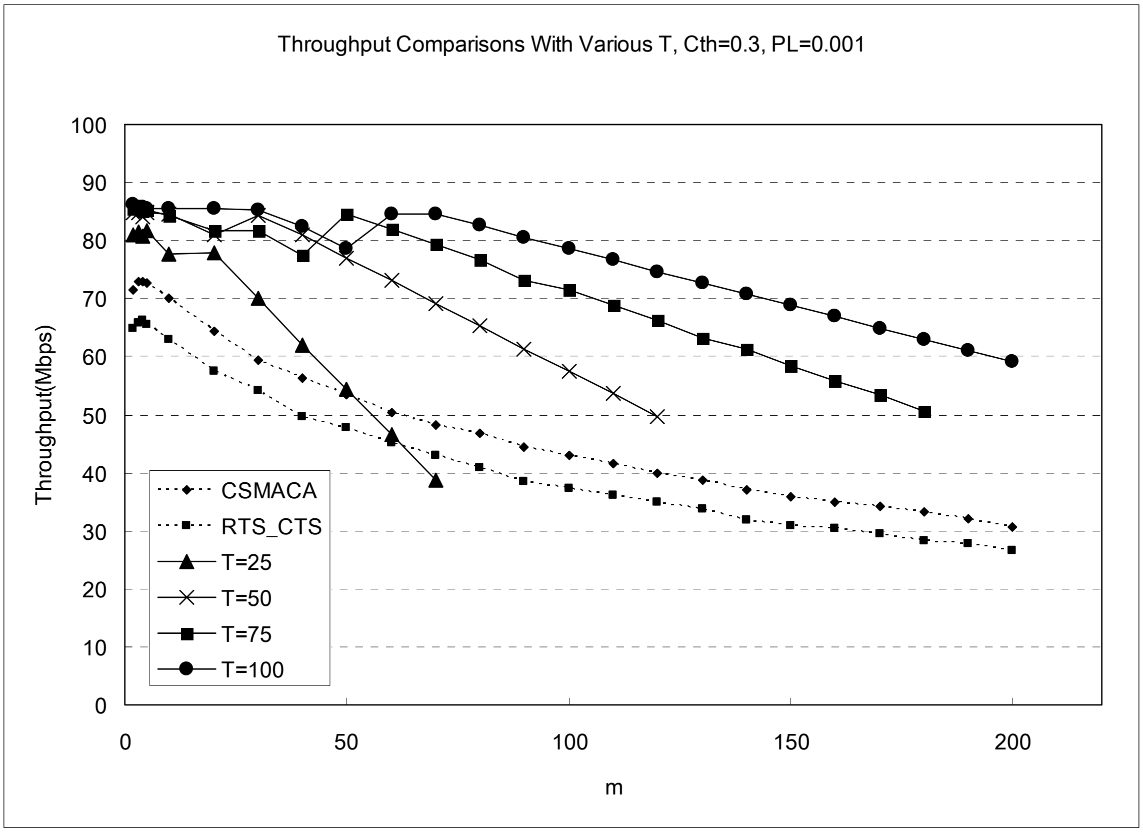


Figure 4-6: PCF/MA various T versus MCSMA/CA throughput comparisons

Fig. 4-6 shows PCF/MA has maximum performance gain over MCSMA/CA about 100% when the number of contenders is large ($m \geq 60$, T = 100). It also shows that PCF/MA with large T not only increases throughput, but also increases the number of possible contenders (m). The reason that T decides m can be inferred from Eq. 4-1 which shows that for a given T, the floor operations cause the useful time of each MPDU to become less as m increases. There are also some dropouts in the PCF/MA throughput chart which are caused by the floor operations.

PCF/MA benefits greatly from large T. However, with large T the packet delay also increases.

Delay Model

To find out the timed average delay of PCF/MA we propose the following model depicted in Fig. 4-7.

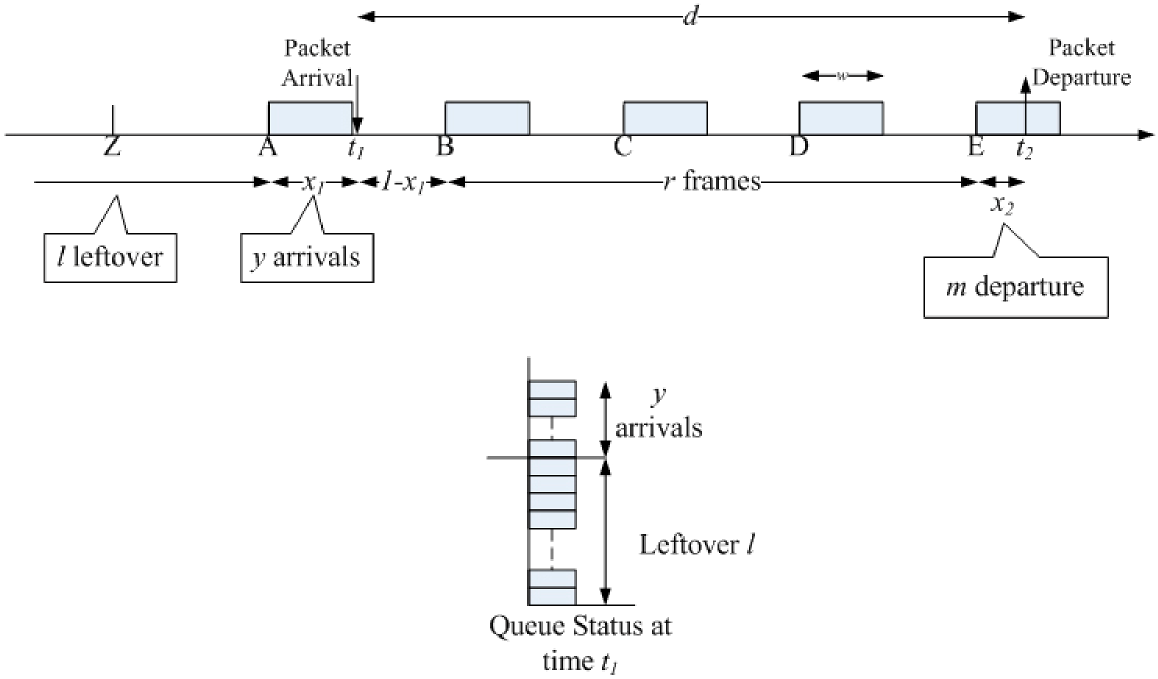


Figure 4-7: PCF/MA delay model

Since PCF/MA divides time into periods each data stream transmits at its reserved period. In Fig. 6.7 we show a stream that has opportunities to transmit at times *Z, A, B, C, D,* and *E*. Each transmission lasts for a portion of a frame. The duration is represented as symbol w. Assume a MPDU arrived at time t_1 and departed at time t_2. Assume the system is a steady state system. Thus, the number of MPDUs in the queue at time A is the number of leftover MPDUs, l. Assume y MPDUs arrived during the time from A to t_1. This period of time lasts a portion of a frame represented as symbol x_1.

From the above assumption we found that at time t_1 the queue length is the sum of l leftover MPDUs and y arrivals. Assume PCF/MA can deliver μ MPDUs per frame for current stream. Thus, delivering $l+y$ MPDUs takes $r\mu+x_2$ frames where x_2 is a portion of w. The delay of the new arrival MPDU thus can be calculated by

$$d = (1 - E(x_1)) + r + E(x_2) \text{ frames} \tag{4-10}$$

where $E(.)$ represents the expectation of a variable.

During this time all MPDUs in the queue must be processed. This amounts to

$$E(l) + E(y) = (1 - E(x_1)) \times \mu + r \times \mu + E(x_2) \times \mu \text{ MPDUs} \tag{4-11}$$

Assume the arrival rate is λ arrivals in a frame. Since the system is a steady state system the inter-arrival time of MPDUs can be treated as a uniform distribution. The expected value of x_1 is

$$\begin{aligned} E(x_1) &= \frac{\frac{1}{\lambda} + \frac{2}{\lambda} + \frac{3}{\lambda} + \cdots + \frac{\lambda}{\lambda}}{\lambda} \\ &= \frac{(1+\lambda)\lambda}{2\lambda} \\ &\approx \frac{1}{2} \end{aligned} \tag{4-12}$$

In a similar way we can calculate $E(x_2)$ as

$$E(x_2) \approx \frac{1}{2} w \tag{4-13}$$

Using queuing theory we can calculate $E(y)$ as

$$E(y) = E(x_1) \times \lambda \approx \frac{1}{2} \lambda \tag{4-14}$$

From Eq. 4.11 ~ Eq. 4-14 we can obtain r as

$$\begin{aligned} r &= \frac{E(l) + E(y) - \mu + E(x_1)\mu - E(x_2)\mu}{\mu} \\ &= \frac{E(l) + \frac{1}{2}\lambda - \frac{1}{2}\mu - \frac{1}{2} w\mu}{\mu} \\ &= \frac{1}{\mu}\left(E(l) + \frac{1}{2}(\lambda - \mu - w\mu \right) \end{aligned} \tag{4-15}$$

Substituting Eq. 4-15 to Eq. 4-10 we get the average delay

$$\begin{aligned} d &= \frac{1}{2} + r + \frac{1}{2} w \\ &= \frac{1}{\mu}\left(E(l) + \frac{1}{2}\lambda \right) \end{aligned} \tag{4-16}$$

Since the service time for each maximum MPDU is the same the system can be modeled as a M/D/1 system, where *D* means deterministic, we have $\overline{X^2} = \frac{1}{\mu^2}$. From the Pollaczek-Khinchin (P-K) formula we denote the residual queue length *l* as

$$\begin{aligned} E(l) &= \frac{\lambda^2 \overline{X^2}}{2(1-\rho)} \\ &= \frac{\lambda^2 \frac{1}{\mu^2}}{2(1-\rho)} \\ &\text{where } \rho = \frac{\lambda}{\mu} \end{aligned} \tag{4-17}$$

Thus the average delay *d* becomes

$$\begin{aligned} d &= \frac{1}{\mu}\left(\frac{\lambda^2 \frac{1}{\mu^2}}{2(1-\rho)} + \frac{1}{2}\lambda \right) \\ &= \frac{1}{2}\rho + \frac{\rho^2}{2\mu(1-\rho)} \end{aligned} \tag{4-18}$$

The simulation and analytical results are shown in Fig. 6.8 ~ 6.11. When the reserved duration approaching to the full frame (in this case, *w*=*T*), the simulated results closely match to the estimation from M/D/1. However, when the reserved duration is short, the simulated results becomes approaching to the estimation from our analytical model. For long reserved duration, most of the MPDUs arrived during the transmission duration yielding shorter waiting delay. For short reserved duration, however, most of the MPDUs arrived outside the reserved period resulting in a long waiting time.

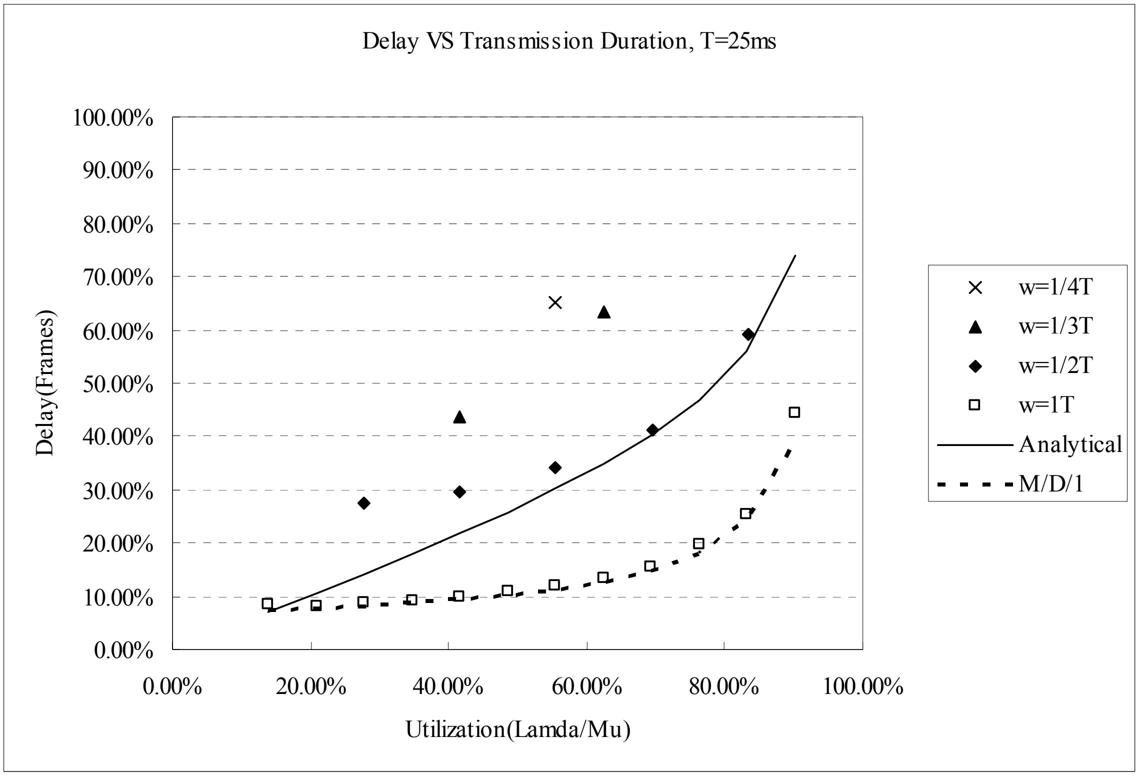


Figure 4-8: Analytical and simulated average delay with various w, T=25ms

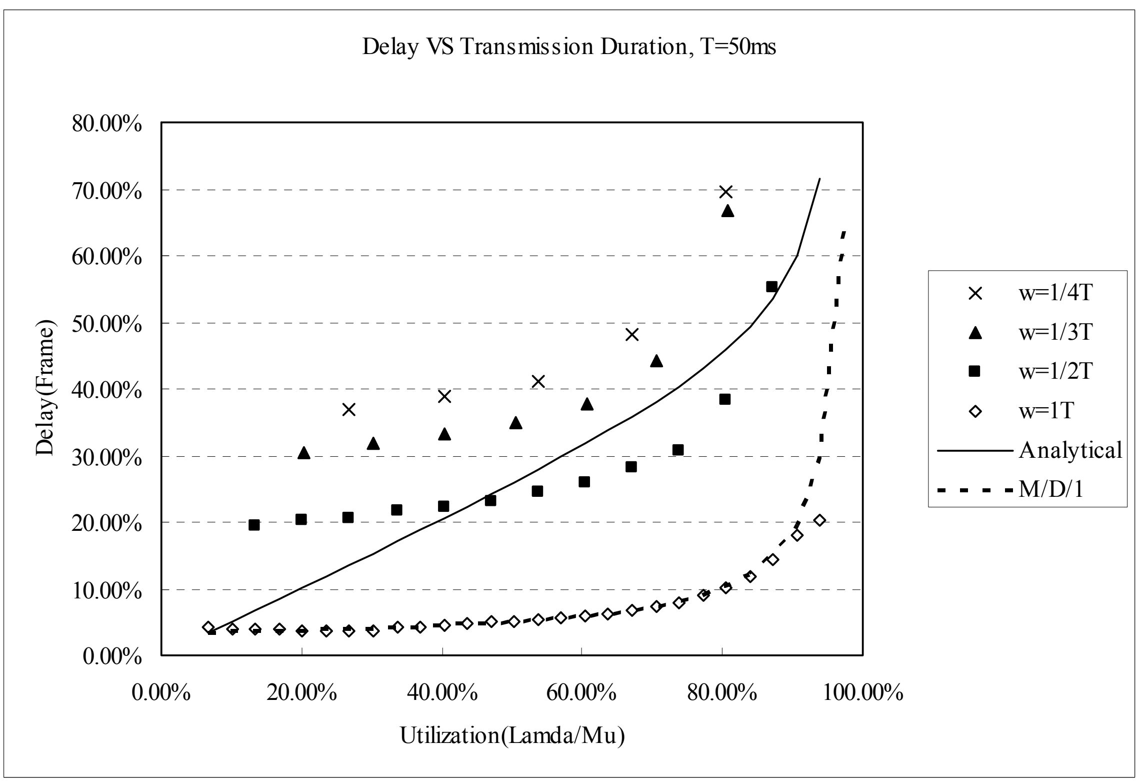


Figure 4-9: Analytical and simulated average delay with various w, T=50ms

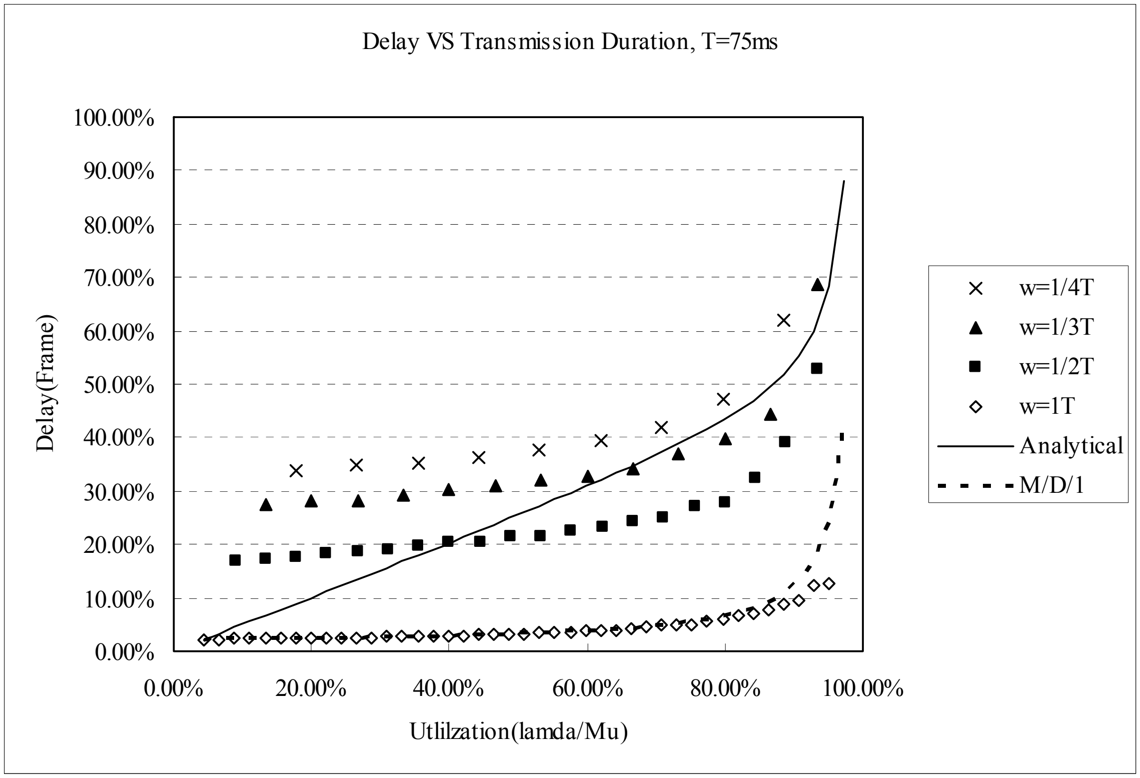


Figure 4-10: Analytical and simulated average delay with various w, T=75ms

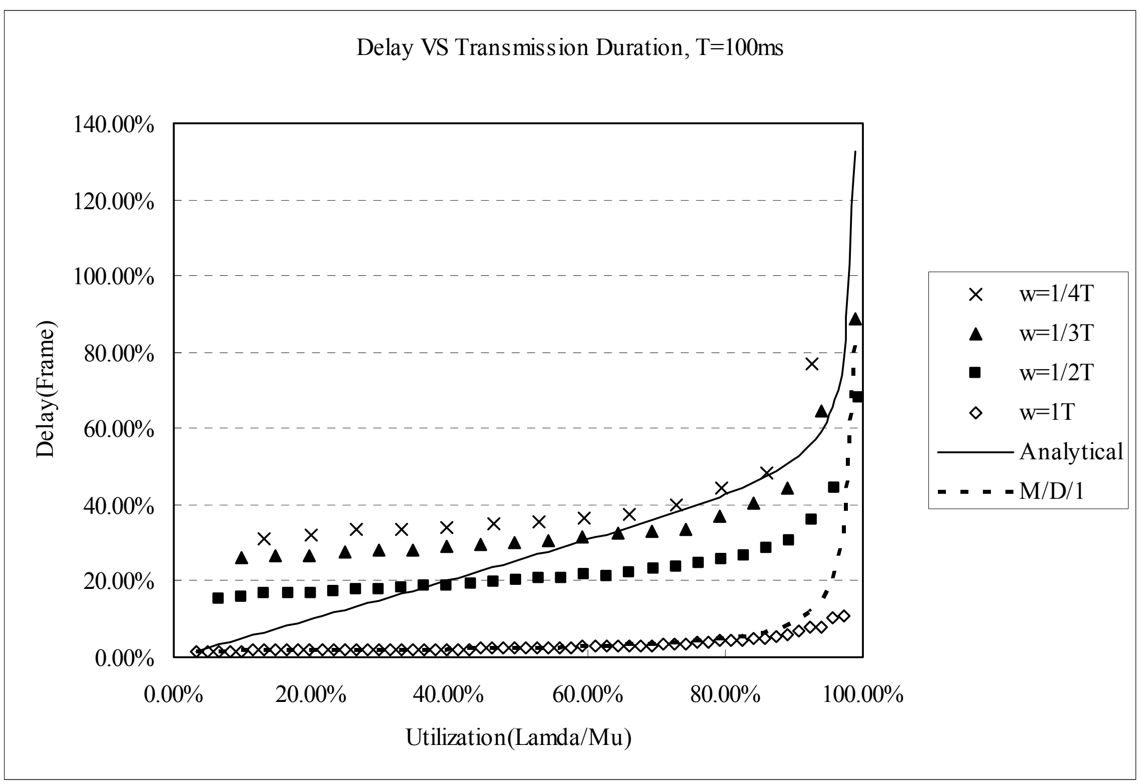


Figure 4-11: Analytical and simulated average delay with various w, T=100ms

Video Traffic

The main focus of this section is to discover the performance of PCF/MA in delivering DVD video streams. To do that we build a video traffic model as depicted in Fig.4-12.

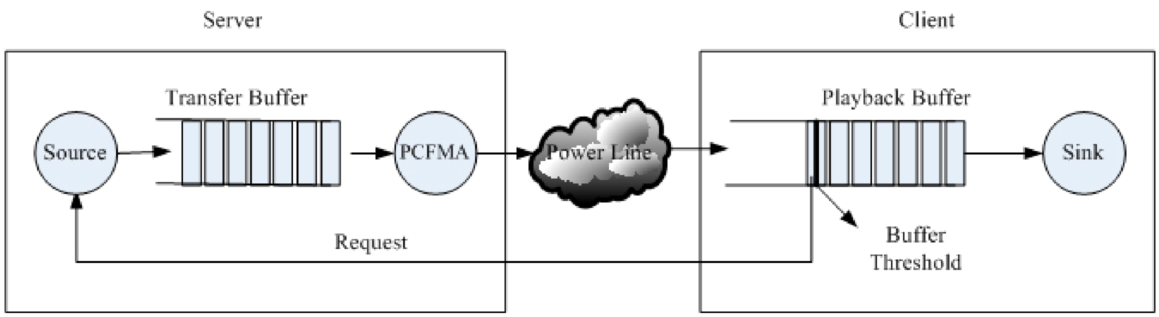


Figure 4-12: Video traffic model

A PLC station acts as a video server. It supports several DVD streams to different destinations. To simplify our discussion we assume that the channel data rate remains stable throughout the whole video display length. We assume the video source is able to supply one video frame immediately after the request is received and that the server will start to transmit MPDUs once it gets permission to transmit. The client moves the received MDPUs to the playback buffer and re-assembles them to a normal video frame before playback. If a video frame is still in transmitting when it is time to display, the video frame is counted as a miss deadline video frame. The client will try to display the video frame 33.3*ms* later. The miss deadline count keeps incrementing until the video frame is successfully displayed.

To smooth video streams, a certain amount of playback buffer (B) is required. When the buffer level is lower than the predefined buffer threshold (B_{TH}), the client sends requests to the server to ask for more video frames. If at time T, we have $B_T > B_{TH}$, the server stops transmitting MPDUs.

To make sure the missing video frames are not caused by the protocol in use but by another factor, we investigate the relationships between the playback buffer size at a specific time T (B_T), playback delay (D), average frame size (V_a), maximum video frame size (V_{max}) channel bandwidth (B_c) and the effective bandwidth (B_w) reserved for this video stream.

We classify the way the playback buffer accumulates data into 3 categorizes: "Depleting", "Accumulating" and "Balanced". Usually it is the "Depleting" situation that clients drop video frames. Consider the captured time frame of a DVD movie depicted in Fig. 4-13. At time T, the buffer level at the client side can be calculated by

$$B_T = B_w D + B_w T - \sum_{i=1}^{30T} V_i \qquad (4\text{-}19)$$

where V_i is the data size of frame i. If we assume T is relatively large, such that $V_i = V_a$ and Eq.4-19 becomes

$$B_T = B_w D + B_w T - 30TV_a \qquad (4\text{-}20)$$

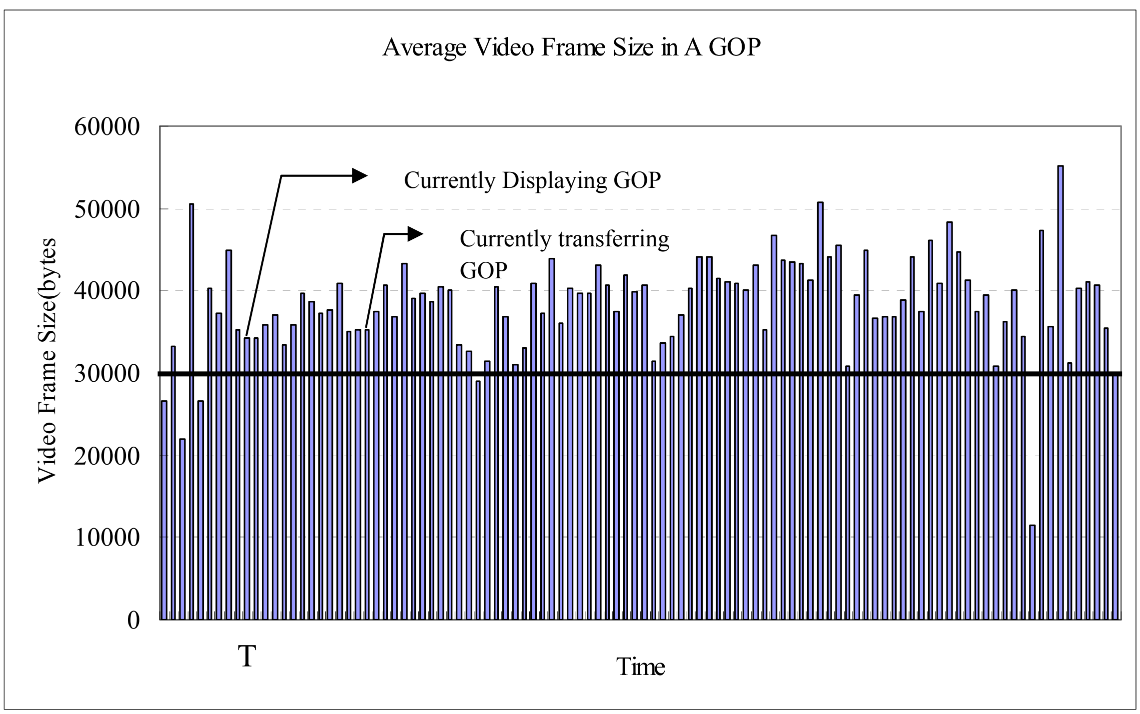


Figure 4-13: A large video frame in transmit

At time T, the server encounters a Group of Picture (GOP) with large video frames of size V_{max} bytes as shown in Fig.4-13. Assume the number of video frames in a GOP is 15, then it takes $t = \frac{15V_{\max}}{B_w}$ second to deliver this GOP. During this t time frame, the client consumes $N = 30 \times \frac{15V_{\max}}{B_w}$ video frames, that is, the client consumes $N \times V_{\max} = \frac{450V_{\max}^2}{B_w}$ bytes. Meanwhile it accumulates $15 \times V_{\max}$ bytes. Suppose there are G such GOPs, the Eq.4-20 becomes

$$B_T = B_w D + B_w T - 30TV_a + G\left(15V_{\max} - \frac{450V_{\max}^2}{B_w}\right) \geq 0 \qquad (4\text{-}21)$$

In order to prevent video frames from missing their deadlines, we must make Eq.4-21 always larger than zero. To help us analyzing Eq.4-21, we re-arranged it as follows:

$$B_T = B_w D - G\left(\frac{450V_{\max}^2}{B_w} - 15V_{\max}\right) + \left(B_w - 30V_a\right)T \geq 0 \qquad (4\text{-}22)$$

The term $\left(B_w D - G\left(\frac{450V_{\max}^2}{B_w} - 15V_{\max}\right)\right)$ in Eq.4-22 is the accumulated buffer size during the playback delay time plus buffer loss during transmitting large video frame GOPs. The term $\left(B_w - 30V_a\right)$ is the difference of the reserved bandwidth and the average DVD title bit rate. In real world applications, B_w, V_{max} and V_a are DVD statistics thus can be easily defined.

To make B_T always larger than zero, we conclude the following sufficient, but not required, conditions:

$$B_w D - G\left(\frac{450V_{\max}^2}{B_w} - 15V_{\max}\right) \geq 0 \qquad (4\text{-}23)$$

and

$$\left(B_w - 30V_a\right) \geq 0 \tag{4-24}$$

If we assume the reserved bandwidth is the video average bit rate, the D can be calculated by

$$\begin{aligned} D &\geq \frac{G(450V_{\max}^2 - 15B_w V_{\max})}{B_w^2} \\ &= \frac{G(450V_{\max}^2 - 450V_a V_{\max})}{\left(30V_a\right)^2} \\ &= \frac{G(V_{\max}^2 - V_a)}{2V_a^2} \end{aligned} \tag{4-25}$$

From Eq.4-23 ~ 4-25 we observed that when reserved bandwidth is equals the average DVD bit rate, the startup delay D should be large enough to absorb the buffer loss due to large video frames. However, when we reserve a large bandwidth and a small playback buffer, the buffer will build up quickly. Once B_{TH} is reached, the server has to stop sending further video frames to prevent buffer overflow. In this case, the term B_w in Eq.4-24 becomes over-estimated. To prevent this problem, a large playback buffer is desirable. Since the minimum playback buffer size should satisfy Eq.4-23 and Eq. 4-24.

$$B \geq G\left(\frac{30V_{\max}^2}{B_w} - V_{\max}\right) \text{and } B \geq 30V_a \tag{4-26}$$

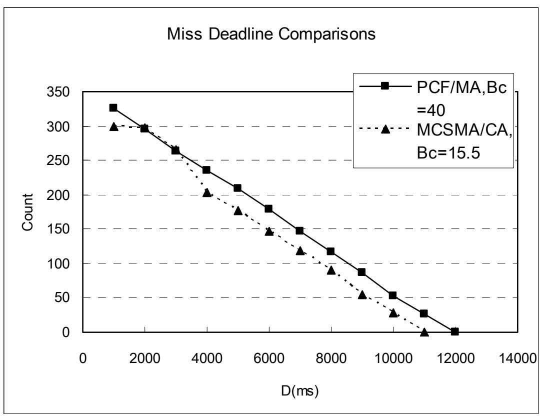
Since the playback delay D and playback buffer is for the worst scenario in the DVD playback, the G should be large enough to satisfy this situation; thus we set G = 60.

To verify our formula, we simulated a video server/client assuming no other data streams on the network where the client has an unlimited buffer, the video parameters is listed in Table 4-2. The efficiency of PCF/MA is about 80% when T = 25*ms*. The reserved bandwidth for PCF/MA is 9.44Mbps, the observed throughput is 7.45Mbps which is close to average bit rate. We slowly increase delay playback time D, and observe if the simulation drops video frames. If it drops video frames, we record the maximum buffer it accumulates before dropping. The total simulated time is 20 minutes which is about one sixth of the original movie length.

Table 4-2: Video traffic parameters

Video	
Average bit rate	7.1Mbps
Vmax	36Kbytes
Va	30Kbytes

In simulating MCSMA/CA protocol, we follow the same procedure except the channel data rate is assigned as 15.5Mbps and the observed throughput is around 7.58Mbps. Simulation results are shown in Fig. 6-14.



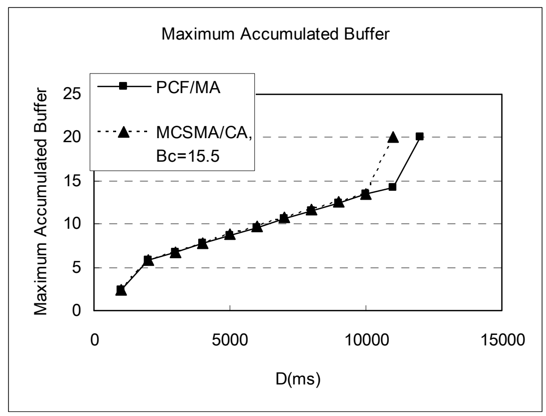


Figure 4-14: Video playback performance comparisons

The simulation results show that without competition, the MCSMA/CA performs slightly better than the PCF/MA (though at worse efficiency). In Fig.6-14, when $D = 1$, the client drops about 320 video frames and the maximum buffer accumulated is about 2Mbytes. The result shows that the delay time is too short to build up a safe buffer for future video bit rate fluctuation even with unlimited buffer. The figure also shows that the video stops dropping after we increase D to 12 for PCF/MA and 11 for MCSMA/CA. In both cases, the maximum accumulated buffer size is more than 20Mbytes. This result suggests that when the delay playback time and buffer is large enough, both protocols have ample time to deliver large video frames thus no video frames were dropped.

From Eq.4-25 and 4-26, we obtain B = 6.48 Mbytes and D = 11 second. The results though different from the simulated results, are acceptable closely. The difference between the simulated and analytical results is because the channel efficiency did not maintain at 80% in the simulation. The problem can be illustrated by the Fig. 4-15.

Figure 4-15: Capture of a period of the DVD simulation

Figure 4-15 shows the DVD video frames in transmit and the time takes to deliver a video frame. Real world DVDs usually have lots of small video frames especially B frames. These small video frames can not fill-up a MPDU result in low channel efficiency. One has to note that Eq.4-25 and 4-26 do not estimate the exact B and D since channel efficiency used is the maximum channel efficiency. Besides, there are too many variations in real world DVD video streams. But Eq. 4-25 and 4-26 gives us an idea of how much bandwidth is required with a given set of video parameters.

To demonstrate PCF/MA in delivering multiple video frames at the same time, we conducted multiple DVD server-client simulations. The simulate time is 20 minutes with $D = 1$ and $B = 10000$Bytes. The simulation results are depicted in Fig.4-16.

In Fig.4-16, the simulation result showed that PCF/MA supports more streams when the channel data rate is high. However, under the same channel data rate, the MCSMA/CA supports less DVD streams. This shows MCSMA/CA suffered from competitions between MPDUs and protocol unfairness. Though PCF/MA is capable of delivering up to 11 DVD streams in theory, the real work DVD streams contains too many small video frames makes channel efficiency much lower than expected. From our simulation, we found the efficiency drops to about 64%! To increase the efficiency, a new streaming protocol that supports striping several small video frames into a MPDU is required. Because of the time constrain, we did not further investigate this issue.

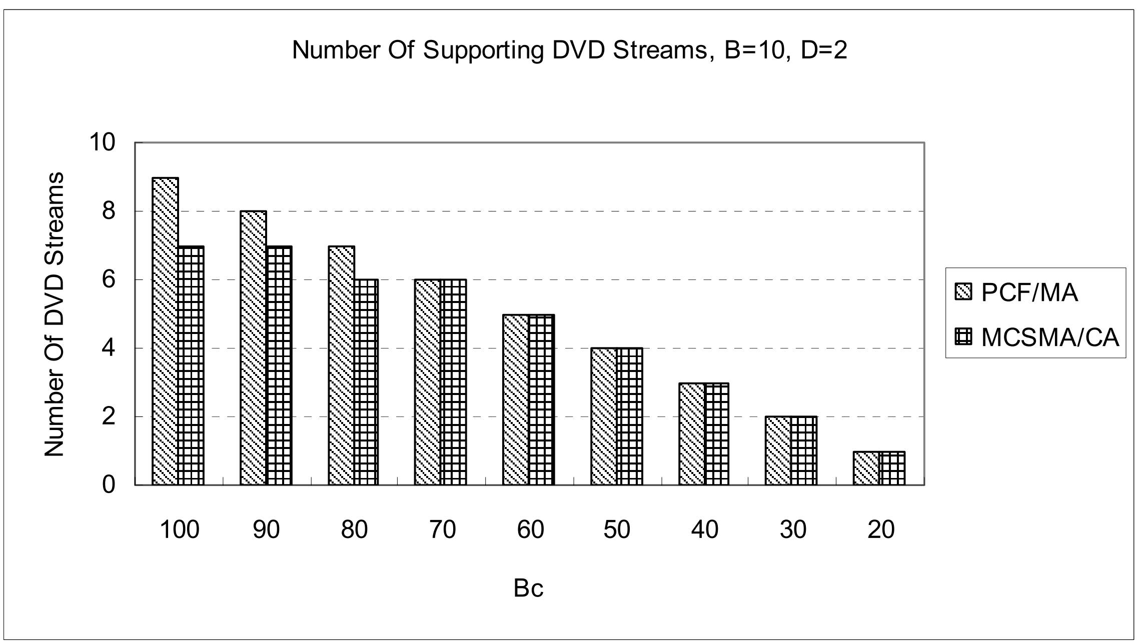


Figure 4-16: Multiple video streams comparisons

Visualization of Protocol Simulation Data

When developing or debugging network protocols, it is necessary to understand both behavior in the aggregate, such as throughput or delay, and specific, pathological scenarios. For the former, graphs are useful ways to present data, but for the latter, it has been more challenging to process the data into a form convenient for human consumption. This section describes a system that represents local area network (LAN) simulation data in the form of time-space diagrams, with station states shown in color. The interface is rich in that it not only allows time scale expansion and compression, but also allows the user to click on specific times and events to obtain detailed information about interesting events. The authors have found the program most useful for understanding and improving various protocols types of LAN protocols.

Introduction

Traditional approaches to visualization of local area network (LAN) simulation data mainly involve graphing aggregate behavior as a system parameter is varied. This is very useful and has been employed to optimize system settings as well as to select between alternative approaches. Histograms and waterfall charts showing the distribution of behavior classes has also been employed, particularly for determining behaviors relevant to Quality of Service (QoS) metrics. However, these forms of data representation have limitations, especially when attempting to understand detailed behaviors that give rise to the observed performance.

When simulating complex protocols such as PCF/MA or MCSMA/CA protocols, it becomes difficult to judge if the simulator works as the desired way using the text based log files. The protocol designers have to look into each text generated by the simulator and analyze the state transition imaginarily. This method works fine when simulating small scale networks or simple scenarios, whereas when simulating large scale networks or complex scenarios, the huge text logs becomes un-manageable.

To improve the quality of interpretations of simulation results, Visual Protocol Analyzer (VPA) is designed for efficient analysis of the results of simulations.

Goal of User Interface

Humans are more sensitive to graphs rather than text. Especially when analyzing complex situations, people usually use drawings to help themselves to reason. For example, Fig. 4-17 shows two modes of the same event transition of a simulation results. People can easily understand the state transition displayed as a diagram on the figure in the right hand side rather than the texts shown on the figure in left hand side. This comparison gives us a clear view of the need of a VPA. To effectively analyze the results of simulations, the desirable VPA should contain the following features

Figure 4-17: Text versus graphical event analyzer

- Easy and efficient to communicate with the analyzer.

The way to communicate with the analyzer decides the efficiency of analyzing a simulation result. We choose to use mouse cursor as the major User Interface(UI) because: First, using mouse cursor is an intuitive way of operating modern operating systems Thus people can become more acquainted to the mouse operation. Secondly, using mouse is more effective if the User Interface (UI) is reasonably designed. For example, people do not have to memorize various instructions in order to execute a “Save” operation.

However, we did not completely abandon the usage of a keyboard. It is undeniable that under some circumstances the keyboard is more efficient than mouse operations. For example, to jump to a specific time stamp (if the value of the time stamp is known) simply type the value into an edit box and that would be faster than using mouse to browse to the desired time mark.

- Easy to zoom in and out.

Some events rarely happen except when simulated for a certain amount of time. Thus usually protocol designers will usually simulate a scenario for a long time in order to find out the possible of design flaws. In this case, the time line in the analyzer would become lengthy. Thus an analyzer should be able to zoom into a specific time frame to reveal detailed protocol state transition. It also should have the ability to zoom out to display a global view of the state transitions to find out the correlations between events.

To reach this goal, we designed mouse gestures to represent zoom in and out. Designers can use mouse to mark a range of time which represents the time frame for operation. If designers click within the range, then the analyzer assumes it is a command of “Zoom In”. Otherwise it will zoom out so that the marked time frame occupies half of the window. Designers can easily mark various lengths of time frames to achieve various zoom factors.

If designers want a specific zoom factor, they can use the keyboard and type the value into the zoom factor edit box. To prevent ambiguity, the marked time frame is also shown as a text display on the display window.

- User should be able to mark a specific time.

When examining simulation results, designers usually have to look at various time stamps to understand the causes of aberrant events. Without the mark function, it would be difficult to move back and forth between time frames.

In VPA system, designers can achieve this effect by simply clicking on the display window. To help designers understand the marked time, VPA also shows the marked time value in the display window.

- Easy to navigate throughout whole time line.

Since the simulation results are usually large, a display window width is not able to display all events at a time. To improve the experiences of browsing events, we use scroll bars to travel between time frames. Since scroll bars are elastic to the mouse movements, designers can drastically move the mouse to achieve large step jumps or click on the end arrow box to achieve small scale time line either forward or backward.

- The analyzer should be flexible in changing event display colors or event names.

Since humans are sensitive to colors, different events should use different colors to represent them. Meanwhile, different protocol designs have different event naming. The VPA should be elastic enough to adapt these minor changes without recompiling the source code.

To achieve the above goals, the VPA will read a configuration file before it starts to parse log files. Designers can modify the configuration file to suit their needs. For example, if a protocol has a new state called “Preparation”, designers can open the configuration file using a text editor, and add a new line of the event name and the associated representing color. The VPA will automatically add the new event into its control panel and in the representing diagram.

VPA is not only elastic to the offline event representing modifications, but also elastic to the real time changes. For example, if designers are looking for a specific event, they can turn off other events to make desired event stands out in the final diagram as shown in the Fig.4-18.

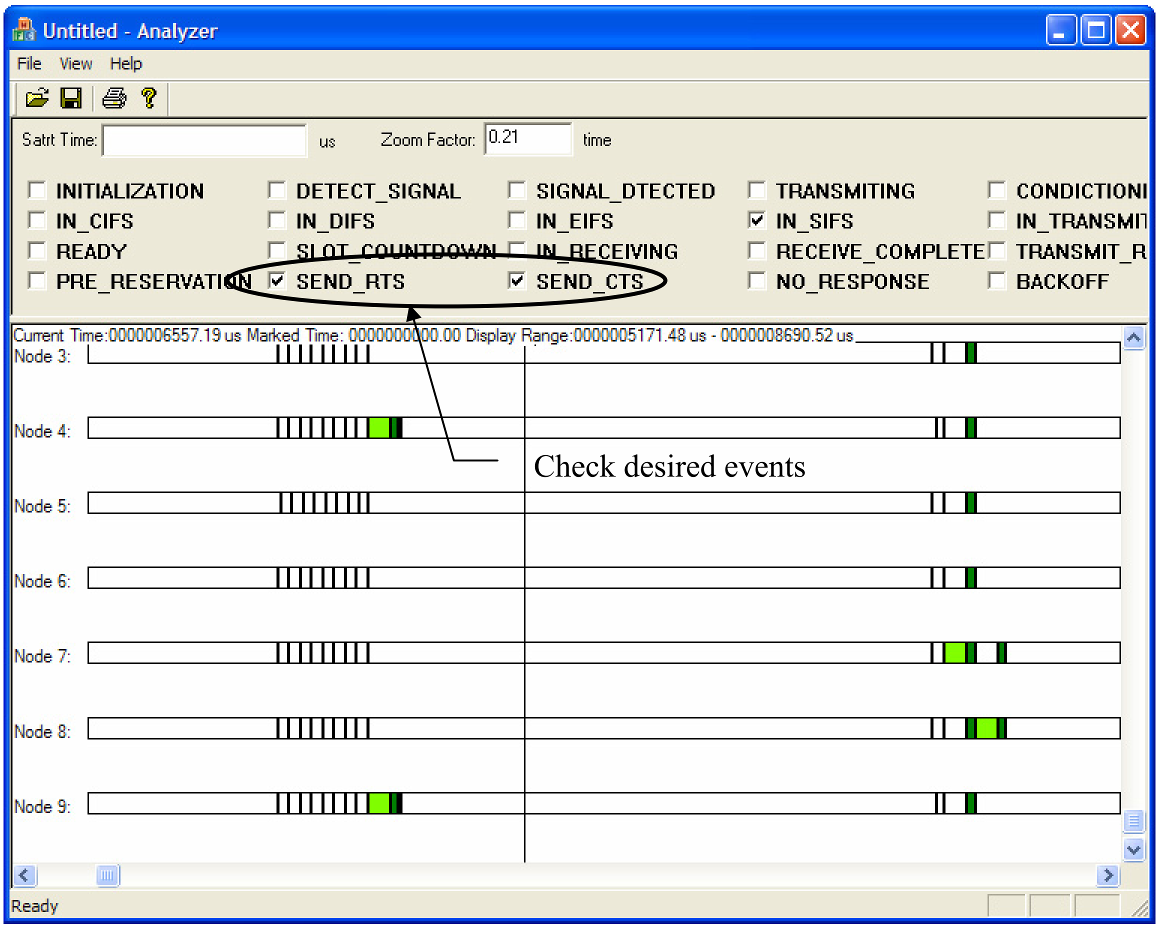
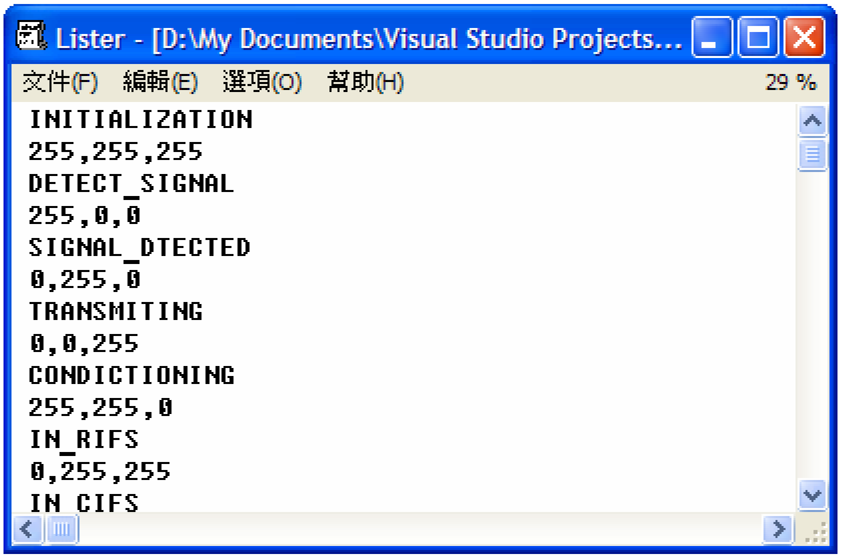


Figure 4-18: Turning off unused events can make desired events stands out

The above mentioned VPA features makes using VPA for analyzing protocol events efficient and easy. It can largely reduce the time required in debugging a simulator.

Data Presentation Mechanism and System Design

As we mentioned earlier, the VPA requires a configuration file for state presentations. The file format is shown in Figure 4-19.



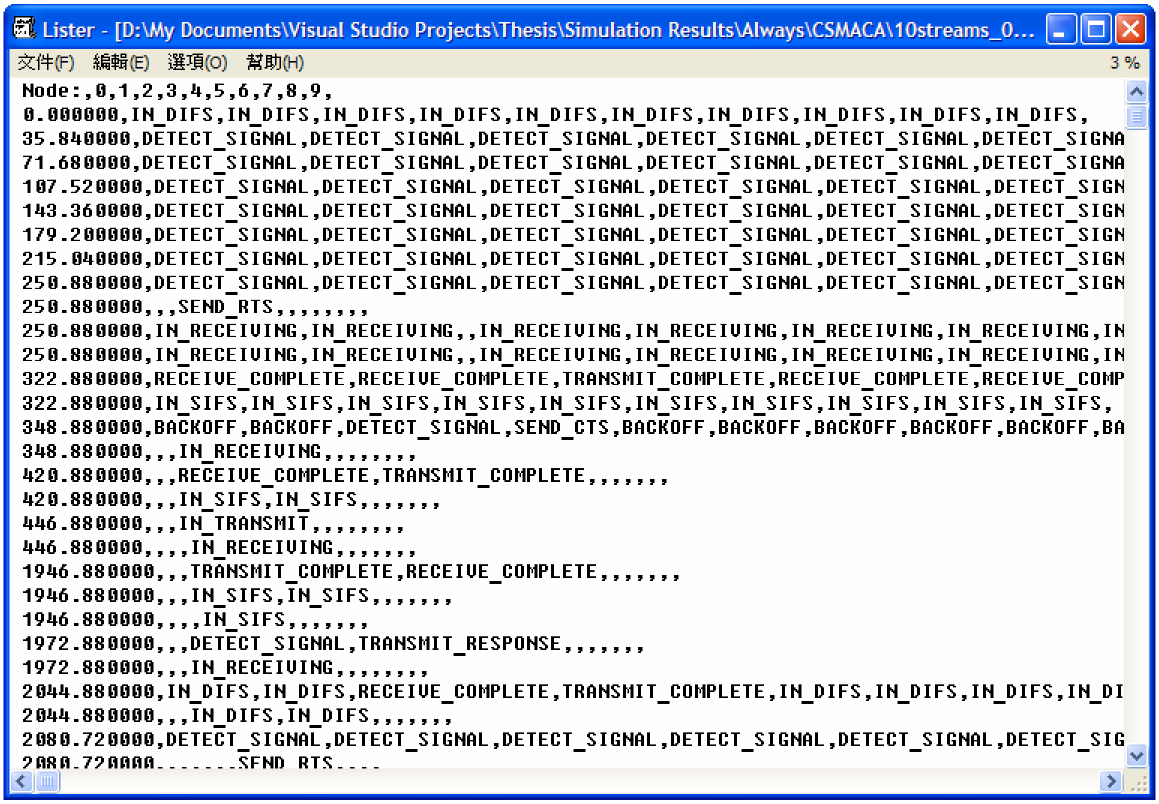

```
INITIALIZATION
255,255,255
DETECT_SIGNAL
255,0,0
SIGNAL_DTECTED
0,255,0
TRANSMITING
0,0,255
CONDICTIONING
255,255,0
IN_RIFS
0,255,255
IN_CIFS
```

Figure 4-19: VPA configuration format

The first line in Figure 4-19 shows a state name: INITIALIZATION, the second line shows the color that representing this event. Each statement occupies one line. VPA supports 100 different states and 255 nodes. Users do not have to specify the total number of states in the configuration file. VPA will judge the effective state names and number of states.

To translate text log files into diagram forms, VPA has to parse the recorded information. The sample log file format is shown in Fig.4-20.

The first line of the log file in Fig. 4-20 defines the total number of nodes in the simulation. Starting from the second line, it contains the state transitions of each node of each time stamp. Whenever a node changes its states, the protocol simulator will record the transition. Other nodes that are not involved in the state transition is not recorded in order to reduce the file size. The protocol simulator first records the event time stamps, then record the states of each node. If the state of a node remains unchanged, the protocol simulator skips that node.




```
Node:,0,1,2,3,4,5,6,7,8,9,
0.000000,IN_DIFS,IN_DIFS,IN_DIFS,IN_DIFS,IN_DIFS,IN_DIFS,IN_DIFS,IN_DIFS,IN_DIFS,IN_DIFS,
35.840000,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNA
71.680000,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNA
107.520000,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGN
143.360000,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGN
179.200000,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGN
215.040000,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGN
250.880000,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGN
250.880000,,,SEND_RTS,,,,,,,,
250.880000,IN_RECEIVING,IN_RECEIVING,,IN_RECEIVING,IN_RECEIVING,IN_RECEIVING,IN_RECEIVING,IN
250.880000,IN_RECEIVING,IN_RECEIVING,,IN_RECEIVING,IN_RECEIVING,IN_RECEIVING,IN_RECEIVING,IN
322.880000,RECEIVE_COMPLETE,RECEIVE_COMPLETE,TRANSMIT_COMPLETE,RECEIVE_COMPLETE,RECEIVE_COMP
322.880000,IN_SIFS,IN_SIFS,IN_SIFS,IN_SIFS,IN_SIFS,IN_SIFS,IN_SIFS,IN_SIFS,IN_SIFS,IN_SIFS,
348.880000,BACKOFF,BACKOFF,DETECT_SIGNAL,SEND_CTS,BACKOFF,BACKOFF,BACKOFF,BACKOFF,BACKOFF,BA
348.880000,,,IN_RECEIVING,,,,,,,,
420.880000,,,RECEIVE_COMPLETE,TRANSMIT_COMPLETE,,,,,,,
420.880000,,,IN_SIFS,IN_SIFS,,,,,,,
446.880000,,,IN_TRANSMIT,,,,,,,,
446.880000,,,,IN_RECEIVING,,,,,,,
1946.880000,,,TRANSMIT_COMPLETE,RECEIVE_COMPLETE,,,,,,,
1946.880000,,,IN_SIFS,IN_SIFS,,,,,,,
1946.880000,,,,IN_SIFS,,,,,,,
1972.880000,,,DETECT_SIGNAL,TRANSMIT_RESPONSE,,,,,,,
1972.880000,,,IN_RECEIVING,,,,,,,,
2044.880000,IN_DIFS,IN_DIFS,RECEIVE_COMPLETE,TRANSMIT_COMPLETE,IN_DIFS,IN_DIFS,IN_DIFS,IN_DI
2044.880000,,,IN_DIFS,IN_DIFS,,,,,,,
2080.720000,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIGNAL,DETECT_SIG
2080.720000,,,,,,,SEND_RTS,,,,
```

Figure 4-20: Simulator log format

The tokens are defined in the configuration file. When VPA sees a text that is equal to a token, the index of the token is stored into an internal state queue associated with the starting time stamps. If a text is not recognizable during parsing, the VPA will skip that text. The VPA will keep parsing every text in the log file until the end of the file. When a token is recognized, the ending time stamp of the previous state is also decided.

After a whole file is parsed, the VPA will start to display each states with the colors defined in the configuration file. Since the window height and width is limited and is usually smaller than the simulation duration, the real data displayed is a small portion of the whole data as shown in the Fig.4-21. To help designers browse, a “Zoom In” and “Zoom Out” command is required.

To implement “Zoom In” and “Zoom Out”, we need to calculate the Zoom Factor. Initially the Zoom Factor is defined as one. Before calculating the zoom factor, the VPA first translates the marked time frame to coordinates of the display. The translation can be done by Eq. 6.29.

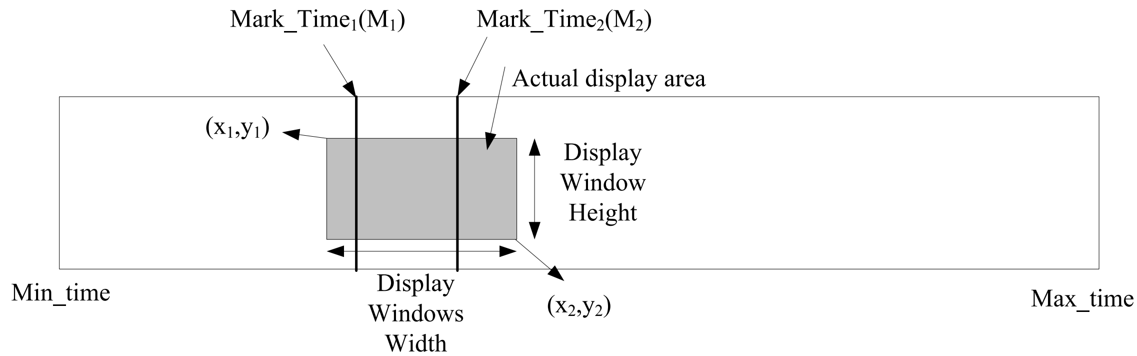


Figure 4-21: The actual display window is a portion of the whole data

$$\begin{aligned} M_1 &= (Mark\_Time_1 - Min\_time) \times Zoom\_Factor \\ M_2 &= (Mark\_Time_2 - Min\_time) \times Zoom\_Factor \end{aligned} \qquad (4\text{-}27)$$

If the mouse gesture is a "Zoom In" command, that is, user makes a mouse click within the marked time frame, the zoom factor can be calculated by dividing the current display time frame with the marked time frame as shown in Eq. 4-28.

$$Zoom\_Factor = \frac{x_2 - x_1}{M_2 - M_1} \qquad (4\text{-}28)$$

If the command is a "Zoom Out", the zoom factor is calculated by dividing the marked time frame with the current display time frame as shown in Eq. 4-29.

$$Zoom\_Factor = \frac{M_2 - M_1}{x_2 - x_1} \qquad (4\text{-}29)$$

Since the simulation log file is huge, we need a proper data structure to represent state transitions. An event queue is used in VPA. The queue is dynamically allocated and de-allocated in order to minimize the system memory requirement.

To reduce window flickering due to the updating of display contents, we record the areas that are affected by the current operation and update those areas only. This way, only a small portion of the display areas needs to be updated at a time.

Experience Using the Visualization System

Because of the help of the VPA, we largely reduced the protocol design process and the time spent on debugging the simulation result. For example, in the process of designing the MCSMA/CA protocol, we found that searching for the causes of nodes moving to the backoff state using the text based log file is painful since there are so many different state transitions occurring at the same time. However, using VPA we can easily find the cause using the displayed diagram. In Fig. 4-18, we found the cause of the backoff is because of the RTS collisions.

The experience of using VPA is positive; it is efficient and intuitive to operate.

CHAPTER 5
FUTURE WORK AND CONCLUSION

Future Work

Though the PCF/MA protocol proved to be high performance protocol, there is much work to be done. Currently, we do not implement priority classes and treat the incoming queue length of the PCF/MA protocol as unlimited. We will address these issues in the future publications.

We expect to see higher data rates in power line networks in the future as signal modulation technologies improve; however, issues like network security and the network characteristics with a large number of nodes need further development. While we expect the IA devices communicating with other devices on the Internet, the behaviors of these data flows remains unknown. Further research on these issues is of critical importance when power line networks are applied to offices and large multi-user buildings.

Currently we are in the process of investigating the above mentioned issues. Next section, we investigated the factors affecting the Voice over IP quality as the pioneer study of our future work.

Voice over IP

The ultimate goal of VOIP is to provide the communication quality closed to one provided by (circuit-switching) conventional phone systems [20, 21, 22, 23, 24]. However, since the voice conversations are digitized, compressed and packetized into IP packets, to guarantee the same level of QoS as conventional phone networks is challenging. There are many factors that affect the overall end-to-end *Internet Telephony* delivery. For instances, many background traffic (not necessarily voice packets) can arrive anytime from different places. Since IP-based Internet will accept any packets with the *best effort* service, these background traffic packets will affect the voice packets intended for *Internet Telephony*.

We are interested in exploring these factors that influence the end-to-end *Internet Telephony* QoS. Note that these factors certainly will be affected by the compression/decompression algorithms that the system has adopted. Therefore, we provide two tracks of experimental study with either PCM or GSM schemes. We intend to investigate these factors in a simple (while realistic) environment. In order to present the average performance trend, we also decide to investigate the performance results without a dedicated network. Thus, we hope to answer the ultimate question on *what QoS should be accomplished in order to satisfy the quality requirement in an experimental setting*.

This chapter presents the preliminary results from our first set of ongoing investigation. The study was wholly performed by real experiments, thus presenting valuable data/information to be shared with research community. The rest of this section introduces these factors in short while the next section demonstrates the experimental results in greater details.

Distance Factor

Many reasons with the distance can affect the quality. Without a dedicated network, distance influences voice quality (at least) in the following aspects:

- Because of the long distance, usually conventional 56-Kbps (or less) modems are the usual devices used by the average users. Therefore physical bandwidth, though sufficient, is always limited.

- Usually modem connections need to communicate with ISP's modem server. In our case, students access the general modem's server provided by the university. Since this line has been open to all students, the load on the modem server can be varied, thus further affecting the quality.

- The total number of routers can affect QoS in a significant degree. The probability of congestion (thus extra delay) is proportional to the number of routers packet traveled.

- When voice packets travel through the Internet, the packets may be randomly dropped. Therefore, the probability that a voice packet has been affected can be directly related to the distance it travels.

Our preliminary results indicate a consistent performance trend. By designing the experiments carefully, we found that the longer the distance the worse the service quality. Depending on the compression codec used, the achieved-bit-rate[5] measured on end system applications (that is VAT) affected by the long-distance communication can be in the range between 40% and 87%, even only within 10 miles. Some proprietary products (for example, Net2Phone) did optimize the VOIP software in some degree. However, the know-how still remains to be confidential. Our goal, as academic researchers, is to use the public-domain source code as the research vehicle for investigate different factors (in this chapter) and propose solutions (in the future publications). We intend to publish the optimization methods and open-source solutions once they become available.

Congestion Degree

In the classical definition, network congestion occurs when the combined bandwidth of all flows destined for a given output link exceeds the link's capacity. When congestion occurs, the network (mostly switches and routers) discards packets randomly to reduce the congestion. Our recent study [25] for ATM networks indicates that

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[5] We have added the time probes into the open-source VAT software. The achieved-bit-rate is thus collected as the sustained rate in the application level.

congestion is indeed a great factor to influence the end-to-end multimedia delivery over TCP and UDP protocol stack. Performing these kind of experiments over ATM switches and high-speed routers requires a total control over these equipments. Therefore, we are in the process of working with university authorities for performing these similar experiments in the near future.

Meanwhile, we will define the congestion in a slightly different manner. Our voice streams typically go through BellSouth's PSTN, University's shared modem server, and all-traffic-combined shared campus backbone as part of the Internet infrastructure. Since we did not have a total control over this Internet infrastructure, we defined the congestion degree as the experienced background traffic during either day or night time. In order to rule out the possible congestion on a single modem pool server, separated modem servers were used by the caller and colleen.

It is reasonable to assume that the background traffic in the night (for example, 4am) will be significantly less than the daytime (for example, 12pm at noon) for the university setting. The typical total traffic at University of Florida around 4am can be as low as 5 Mbps, while as high as 34 Mbps around noon. Certainly we expect the achieved quality will be better when the background traffic is light. Our experimental results revealed that a 39% to 45% bit rate improvement can be achieved with light load at night compared to heavy load in daytime.

Priority Factor

It has been recognized [26, 27] that priority support and specification are essential for multimedia communication and control systems. Priority schemes can be designed and enforced at both the end systems (that is, operation systems or middleware), and switch/routers in the middle of delivery paths.

Priority specification and enforcement at the end systems seem to be a solution that can be designed and implemented without much difficulty. Since our goal is to provide the voice communication with a quality close to the PSTN's dedicated circuit, higher priority should be assigned to (digitized) voice packets. Other IP data packets from conventional applications (for example, E-mail and FTP) should be assigned as low priority.

Certainly the other side of this issue is *how to enforce* the priority schemes. The software-VAT we investigated does implement RTP (Real-time Transport Protocol) in it. One of the most important features of RTP is that it can provide flexible de-multiplexing. It is defined as application-level framing [28, 29]. Basically RTP uses a range of ports to represent different priorities. The VAT (on top of RTP) provides three priorities: high, medium and low priority at the end stations for the voice streams. VAT/RTP then uses different ports for transmitting and receiving the packets in different priorities. High-priority packets will be served first before low-priority packets (by scanning the ports).

Through this chapter, we intend to provide an integrated observation which offers design guide-lines for system designers. The overall recommendations from our performance study seem to indicate that a range of 15-Kbps and 50-Kbps produces acceptable or satisfactory quality using PCM encoding scheme. By using the GSM encoding scheme, the range can be between 6-Kbps and 15-Kbps for acceptable or satisfactory quality perception.

<u>Related Study</u>

Though the great significance of *Internet Telephony* has been clearly identified in recent years, very few experimental results have been published in the existing literature. Many draft standard recommendations are available. Toga and Ottv [30] did a good overview of these inter-related standards. The VAT/RTP is one of the leading software solutions that is compliant with these emerging standards.

To the best of our knowledge, this chapter perhaps presents one of the first few research results that investigate the impact of different factors in an experimental setting. For instance, Kostas [21] discussed mean delay from different locations in USA and the relationship between delay and hop count. Goyal [31] discussed discussed calling signaling and resource management. Schulzrinne [32] mainly focused on the signaling protocol to initiate a voice session. Rosenberg [33] introduced a unique aspect for programming languages to support *Internet Telephony*.

None of these studies addressed the experimental results by jointly considering distance, congestion and priority factors. Therefore, it is very difficult for us to compare our work with others. The most related work that we could find is perhaps the recent work from [34], but only simulation results were reported instead of real experiments. We not only measured the achieved end-to-end bit rates, but also measure the quality in term of human perception.

Experimental Performance Results

We have adopted the VAT [35] from LBNL (Network Research Group of Lawrence Berkeley National Laboratory) in order to collect experimental performance in a pure software *Internet Telephony* setting. VAT is based on IETF RTP and is for real-time, multi-party audio conferencing over the Internet. Since our university environment does not encourage multicast and conferencing, therefore support for IP Multicast[6] and IP

[6] [6]Many departments choose to disable the MBone routers to avoid the large amount of multicast traffic flowing into their departmental networks.

Multicast Backbone (MBone) is not possible. Thus we use standard unicast IP addresses for point-to-point audio chat.

Source code was down-loaded, and recompiled over Linux for PC and Solaris 2.6 for Sun's Ultra workstations. Sun workstations were connected to the campus Ethernet networks, and PCs were connected with 56-Kbps modem at students' home apartments. All other applications (for example Netscape) were terminated to keep a clean environment and no background network services were provided on each end system. We then performed the experiments in a systematic manner. Every experiment was performed a few times to ensure that a steady performance trend can be observed. In addition to the objective measurement of actual bit rate at the host, we also had a few participants to subjectively measure the voice quality.

The subjective measurements were recorded down as one of the following results: *(Satisfactory, Acceptable, and Unsatisfactory)*. *Satisfactory* was the judgment that the achieved quality was excellent for comprehension. *Acceptable* was the indicator that, though the achieved quality was not excellent, comprehension was still reasonable. *Unsatisfactory* stands for the situation that the archived voice quality was too bad to comprehend the conversation. Note this subjective measurement technique is consistent to emerging industrial subjective performance measurement. The only difference is that we do not convert them into a number system to calculate the performance average for acceptance. Our measurement requires a majority of reviewers to agree on the resulted recommendation, thus presenting the same degree of consistent judgment.

We first presented the effect of distance on the QoS of IP telephone. It was performed in the midnight and packets were assigned with high priority. In this experiment, interesting results were observed. We then fixed distance factor for investigating performance results under congested situation. We then fixed the traffic factor for examining the effect or priority specification.

Results Based on the Distance Factor

In order to isolate the effect of distance from other factors, the experiments were performed at midnight. We observed that the amount of background traffic was significantly less at midnight compared to the daytime. We also fixed the codec scheme using PCM mechanism first, and then extended to GSM. Short-distance experiments were performed at the campus within a one-mile distance. Longer-distance experiments were performed between two students' home apartments, which usually ranged from 3 to 7 miles.

Table 5-1: Performance comparison for distance factor

Codec Used	Distance Covered	Subjective Evaluation	Achieved Bit-Rate(Kbps)		
			Min	Mean	Max
PCM	short	*Satisfactory*	33.5	48.2	59.2
PCM	long	*Satisfactory*	24.5	34.4	45.2
GSM	short	*Satisfactory*	13.7	14.8	15.8
GSM	long	Acceptable	7.5	7.9	8.2

By using PCM codec, it can be observed in Table 5.1 that a higher throughput can be achieved by using short-distance communication. The average sustained bit rate was about 40% higher (that is, 13.8 Kbps) than the longer-distance Internet telephony. Note that PCM only samples up to 64-Kbps, thus higher rates are not possible (though LAN connections were used). In addition, the performance difference was also jointly caused by the hardware capability. Long-distance communication was mainly supported by 56-Kbps modems. Nevertheless, experiments using PCM codec demonstrated a satisfactory performance from human judgment independent of the distances.

On the other hand, GSM specification usually adopted AMR (Adapted Multi-Rate) encoding, which generates up to 22-Kbps compressed voice streams. We did not use any silence suppression. Our experimental results showed that an average 14.8 Kbps voice stream was measured in LAN environment, while only an average of 7.9 Kbps in longer distance. Therefore, the experimental gain from short-distance communication over long-distance was about 87% (that is, 6.9 Kbps).

Note that the potential impact by the distance factor seemed to be significant with GSM coding scheme. For instance, with a longer distance, the subjective quality has been reduced from from *Satisfactory* to *Acceptable*. We are in the process to investigate the reason(s) why overall quality of GSM codec suffered in such a degree when distance was long. Baldi M., Risso F., and Torino P., [34] observed a similar phenomenon (though from simulation). The relationship between packetization delay and the achieved bandwidth has been addressed. It has been found that when packetization delay was more than 18ms, with more bandwidth allocation was required in order to keep end-to-end delay below QoS requirement. Our early evidences indicate that the GSM codec in VAT/RTP has a packetization delay with the range of 80ms. We will verify these possible reasons through experiments in the future.

It also can be observed in Table 5.1 that the high-bandwidth network within the campus was not able to further increase the sustained bandwidth beyond 60 Kbps. We believe that it was caused by the PCM encoding since it only can support maximally 64 Kbps. Since we use software solution, the system overhead increased a certain percentage (for example, about 4 Kbps in this case) because of the interaction between encoding and network transmission is not good enough. In summary, it is shown by our experimental results that a range of 15-Kbps and 50-Kbps is considered to be satisfactory for voice communications (that is, similar to carrier-level quality with 64-Kbps circuit-switching networks).

Results Based on Congestion Degree

Congestion degree from the background traffic in the Internet will certainly affect the overall end-to-end performance. The more background traffic, the less achieved bit rate will be. It is because the fixed network bandwidth is shared by all the data flows. In order to perform the comparisons, we ran the experiments separately at midnight and daytime (for example, noon). We expected that the background traffic within the University of Florida in the day time will be significantly higher than night time.

Table 5-2: Performance comparison based on congestion factor

Codec Used	Degree of Congestion	Subjective Evaluation	Achieved Bit-Rate(Kbps)		
			Min	Mean	Max
PCM	small	Satisfactory	24.5	34.4	45.2
	large	Acceptable	19.8	23.7	31.2
GSM	small	Acceptable	7.5	7.9	8.2
	large	Acceptable	5.1	5.7	6.2

Table 5-2 lists the performance comparisons between these two experiments. In here, we included the previous results with long-distance voice communication for comparison. We believe, with a longer distance, the background traffic within the campus and all channels connected to it will affect the voice quality in a significant degree. Therefore, all the results were collected over long-distance 56-Kbps modems.

The PCM results clearly indicate that the quality of sound has been degraded from *Satisfactory* to *Acceptable* because of the background traffic. While an average 34.4-Kbps stream was achieved in the night time (that is, much less background traffic), only 23.7-Kbps was achieved in the day time. The mean bit rate achieved in the night time was as high as 45% compared to the mean bit rate in the daytime. Thus, for domestic Internet phone conversations, the results suggested users should take advantage of the night times if no dedicated circuits were used.

The GSM results showed a similar effect from the congestion. An average 5.7 Kbps was achieved in the daytime. Therefore with GSM codec, 38.6% more bit rate was achieved in the night time. Compared with PCM, the potential effect of background traffic seemed to be less influenced with GSM. This was an interesting finding, and we were in the process to further identify the exact reason(s) for this observation. Nevertheless, the subjective measurement has been rated as *Acceptable* though a lower bit rate was achieved in the daytime.

Results Based on Priority Factor

Since we do not have the RSVP or WFQ-enable routers within the campus, the specification will only affect the sending and receiving ordering within the hosts. High-priority specification indicates that this voice stream should be served as urgently as possible. On the other hand, low-priority specification on a voice stream will potentially delay or drop the stream within the hosts (either at the sending or the receiving hosts).

The results that we presented in the previous subsections were specified as high-priority voice streams. In order to demonstrate the counter effects for using low priority specification, we also performed an extra set of experiments in the daytime with long-distance communication. The experiment results are shown in Table 5-3.

The VAT/RTP priority scheme proved to be useful when the degree of network congestion degree was high. Given the usual amount of background traffic in the day time, the voice stream decreased to average 0.6 Kbps with low-priority setting for PCM codec. The achieved subjective quality was *Unsatisfactory* for human judgment.

Therefore, it appeared that we should use high-priority specification for VOIP applications all the time.

Table 5.3: Performance comparison based on priority factor in day time

Codec Used	Priority	Subjective Evaluation	Achieved Bit-Rate(Kbps)		
			Min	Mean	Max
PCM	High	Acceptable	19.8	23.7	31.2
	Low	Unsatisfactory	0.4	0.6	1.0
GSM	High	Acceptable	5.1	5.7	6.2
	Low	Unsatisfactory	1.4	1.7	1.9

The voice stream with GSM codec again indicated a similar trend by only achieving 1.7 Kbps experimentally. The subjective measurement was evaluated as *Unsatisfactory*. Note that GSM was particularly designed for low-bandwidth communication (for example, cellular phone communication). The results thus suggest that perhaps an even more sophisticated encoding scheme beyond GSM is needed for the bit rates less than 2 Kbps. Though interesting, this research topic is beyond the scope of the chapter.

Conclusion

It took many decades for scientists and engineers to optimize the PSTN until it was widely used by people as a life commodity. Using Internet's packet switching for supporting concurrent voice streams with guaranteed quality is still in the early stage. Many environmental factors still affect the end-to-end quality. Only when IP telephones provides as good QoS as PSTN does (that is, much less influenced by the environmental factors), then we can expect it to be widely used as another commodity to the general public.

Last section presented our preliminary results on investigating the performance impact from three common factors. Without dedicated networks and QoS enable schemes, the factors of distance and congestion degree were proved to be influential on the achieved end-to-end bit rates. There are still many research issues need to be solved in order to guarantee (at least statistically) the quality of *Internet Telephony*.

This study proposes a new protocol - PCF/MA for high speed PLC networks. To lower overhead caused by contention between MPDUs, we choose a contention-free method. Through simulation, we observed the network efficiency as high as 85% at T = 100. Theoretical analysis found that with eight reservation slots, it can provide a good contention/overhead balance. We also conducted a simulation with constant contenders to verify our analysis. For comparison purposes, we modified the widely used CSMA/CA protocol into a PLC version MCSMA/CA protocol. The simulation results show that PCF/MA protocol has the maximum performance gain of 100% over MCSMA/CA when the T = 100. Our protocol also proved to be able to support more than 200 streams at the same time.

The high speed PLC is targeted for Audio/Video applications over local area networks, however, the parameters for clients to support variable bit rate data streams is remain uncertain at this stage. We analyzed the behaviors of PCF/MA protocol in delivering commercial DVD titles, and derived relationships between required buffer, reserved bandwidth, delay playback time and video statistics. We then used the estimated parameters to design a video server/client simulator in which PCF/MA and MCSMA/CA as MAC protocols were used for comparisons. The simulation results showed that PCF/MA performed much better than MCSMA/CA protocol because of its predictable behavior and low contention/overhead.

The emergence of Information Appliances (IA) for the smart homes of the future will undoubtedly make our lives much more comfortable than ever. However, the infrastructure that supports multimedia traffic and conventional elastic data traffic for communication among IA devices is a critical component of a smart home.

We advocate power line as the infrastructure for smart homes based on the convenience of the power sockets and the layout of the power line network existing in every home. The ultimate goal of PLC network could be the ability to connect to the Internet without dialing up to an ISP server, entirely using electrical wiring only.

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BIOGRAPHICAL SKETCH

Yu-Ju Lin was born in Taiwan. He received the Bachelor of Engineering degree from National Central University, Taiwan in 1990;and the MS degree in computer and information engineering from Chung-Yuan Christian University, Taiwan in 1995. He enrolled in the Ph.D. program at the University of Florida in 1998, doing research on power line communications and multimedia under the direction of Dr. Haniph A. Latchman. His research interests include multimedia communication and computing, power line communication, and high-speed networks. He graduated in May 2004.