PROTOCOL DESIGN FOR REAL TIME MULTIMEDIA COMMUNICATION OVER HIGH-SPEED WIRELESS NETWORKS

BY

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ABSTRACT

The growth of interactive multimedia (IMM) applications is one of the major driving forces behind the swift evolution of next-generation wireless networks where the traffic is expected to be varying and widely diversified. The amalgamation of multimedia applications on high-speed wireless networks is somewhat a natural evolution. Wireless local area network (WLAN) was initially developed to carry non-real time data. Since this type of traffic is bursty in nature, the channel access schemes were based on contention. However real time traffic (e.g. voice, video and other IMM applications) are different from this traditional data traffic as they have stringent constraints on quality of service (QoS) metrics like delay, jitter and throughput. Employing contention free channel access schemes that are implemented on the point coordination function (PCF), as opposed to the numerous works on the contending access schemes, is the plausible and intuitive approach to accommodate these innate requirements. Published researches show that works have been done on improving the distributed coordination function (DCF) to handle IMM traffic. Since the WLAN traffic today is a mix of both, it is only natural to utilize both, DCF and PCF, in a balanced manner to leverage the inherent strengths of each of them. We saw a scope in this technique and develop a scheme that combines both contention and non-contention based phases to handle heterogeneous traffic in WLAN. Standard access scheme, like 802.11e, improves DCF functionality by trying to emulate the functions of PCF. Researchers have made a multitude of improvements on 802.11e to reduce the costs of implementing the scheme on WLAN. We explore improving the PCF, instead, as this is more stable and implementations would be less costly. The initial part of this research investigates the effectiveness of the point coordination function (PCF) for carrying interactive multimedia traffic in WLAN. The performance statistics of IMM traffic were gathered and analyzed. Our results showed that PCF-based setup for IMM traffic is most suitable for high load scenarios. We confirmed that there is a scope in improving IMM transmissions on WLAN by using the PCF. This is supported by published researches on PCF related schemes in carrying IMM traffic on WLAN. Further investigations, via simulations, revealed that partitioning the superframe (SF) duration according to the need of the IMM traffic has considerable impact on the QoS of the WLAN. A theoretical model has been developed to model the two phases, i.e., PCF and DCF, of WLAN medium access control (MAC). With this model an optimum value of the contention free period (CFP) was calculated to meet the QoS requirement of IMM traffic being transmitted. Treating IMM traffic as data traffic or equating both IMM and non-IMM together could compromise a fair treatment that should be given to these QoS sensitive traffic. A self-adaptive scheme, called MAC with Dynamic Superframe Selection (MDSS) scheme, generates an optimum SF configuration according to the QoS requirements of traversing IMM traffic. That particular scheme is shown to provide a more efficient transmission on WLAN. MDSS maximizes the utilization of CFP while providing fairness to contention period (CP). The performance of MDSS is compared to that of 802.11e, which is taken as the benchmark for comparison. Jitter and delay result for MDSS is relatively lower while throughput is higher. This confirms that MDSS is capable of making significant improvement to the standard access scheme.

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APPENDIX A PSEUDOCODE OF THE *wlan_mac* PROCESS MODEL IN OPNET
APPENDIX B SNAP SHOTS OF OPNET CONFIGURATION MENUS FOR SIMULATION OF THE MDSS SCHEME

LIST OF PUBLICATIONS

1. Suhaimi Bin Abd Latif; Rashid, M.A.; Alam, F., "An Analytical Model for MAC Protocol Configuration in WLAN," Advanced Information Networking and Applications, 2009. AINA '09. International Conference on , Bradford, UK, 26-29 May 2009, pp.711-718.
2. Suhaimi Bin Abd Latif, M.A. Rashid, and F. Alam, Interactive Multimedia Communication over WLAN in a Mixed Traffic Environment, The International Journal of Principles and Applications in Information Science and Technology, 2008.
3. Suhaimi Bin Abd Latif, M.A. Rashid, and F. Alam, Profiling Delay and Throughtput Characteristics of Interactive Multimedia Traffic over WLANs Using OPNET. International Conference on Advanced Information Networking and Applications Workshops, Ontario, Canada, 2007. 2(2).
4. Suhaimi Bin Abd Latif, M.A. Rashid, and F. Alam, Effects of Varying Superframe Duration on Jitter in Interactive Multimedia Traffic Transmission over WLANs. 6th IEEE/ACIS International Conference on Computer and Information Science, Melbourne, Australia, 2007. 1(1).
5. Suhaimi Bin Abd Latif, M.A. Rashid, and F.Alam, "Cross-layer protocol design for wireless multimedia network", in Proceedings of the International Conference on Computer and Communication Engineering, ICCCE '06, Volume 2, Malaysia, 9 - 11 May 2006, pp. 755-760.
6. Suhaimi Bin Abd Latif, "Cross-layer protocol design for real time multimeda over WLANs", in Proceedings of the IIMS Post-graduate Conference , New Zealand, 27 October 2005, pp. 1-10.
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LIST OF ABBREVIATIONS

AC – Access Categories
ACK – Acknowledgment
AID – Association Identifiers
AP – Access Point
ARQ – Automatic Repeat Request
BER – Bit Error Rate
BSS – Basic Service Set
CBR – Constant Bit Rate
CFP – Contention-Free Period
CFPri – CFP Repetition Interval
CODEC – Compressor/Decompressor
CP – Contention Period
CSMA – Carrier Sense Multiple Access
CSMA/CA – CSMA with Collision Avoidance
CW – Contention Window
DCF – Distributed Coordination Function
DCF/CD – Distributed Coordination Function with Collision Detection
DES – Discrete-Event Simulation
DIFS – DCF Interframe Space
DRR – Deficit RR
DS – Distribution System
DTIM – Delivery Traffic Indication Message
EBA – Early Backoff Announcement
EDCA – Enhanced Distributed Channel Access
EIFS – Extended Interframe Space
FQ – Fair Queuing
GHz – Gigahertz
HCCA – HCF Controlled Channel Access
HCF – Hybrid Coordination Function
HTML – Hypertext Markup Language
IEEE – The Institute of Electrical and Electronics Engineers
IFS – Interframe Space

IMM – Interactive Multimedia
IP – Internet Protocol
ITU – International Telecommunication Union
Kbps – Kilobits per second
MAC – Medium Access Control
Mbps – Megabits per second
MDSS – MAC with Dynamic Superframe Selection
MPEG – Moving Pictures Expert Group
MPLS – Multiprotocol Label Switching
MSDU – MAC Service Data Unit
Msec – Milliseconds
MWN – Multimedia Wireless Network
NAV – Network Allocation Vector
NIC – Network Interface Card
nRT – non Real Time
OPNET – Optimized Network Engineering Tool
OSI – Open Systems Interconnection
PC – Point Coordinator
PCF – Point Coordination Function
PCM – Pulse-Code Modulation
PER – Packet Error Rate
PHY – Physical (Layer)
PIFS – PCF Interframe Space
QoS – Quality of Service
RF – Radio Frequency
RR – Round-Robin
RT – Real Time
Sec – Seconds
SF – Superframe
SIFS – Short Interframe Space
STA – Workstation
TCP – Transmission Control Protocol
TCP/IP – TCP/Internet Protocol
TS-MP – Two-step Multipolling

TXOP – Transmission Opportunity

UBR – Unspecified Bit Rate

VBR – Variable Bit Rate

WFQ – Weighted FQ

WLAN – Wireless Local Area Network

WRR – Weighted RR

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Chapter 1
Introduction

The growth of wireless applications, for example interactive mobile multimedia (IMM) applications, and interactive gaming, drives the hasty evolution of next-generation wireless networks. This amalgamation of multimedia applications on wireless networks is somewhat a natural evolution. With the proliferation of wireless networks and its integration with multimedia applications and services, work on the improvements of their associated protocols is at a critical point. Numerous researchers such as Cai et. al. [13], Karia et. al. [29], Cano et. al. [37] and Park [104] look into convalescing wireless networks efficiency in transporting this multitude of applications.

1.1. Wireless Local Area Networks

In July 1997, The Institute of Electrical and Electronics Engineers, Inc. (IEEE) published a standard for Wireless Local Area Networks (WLAN) called 802.11 [1]. Using electromagnetic waves, WLANs transmit and receive data over the air without relying on physical connections. This unified standard provided several modes of operation and data rates up to two megabits per second (Mbps). Work began, soon after that, on improving the performance of IEEE 802.11. The concluding results were two incompatible versions of the standard, IEEE 802.11b and IEEE 802.11a. The "b" version operated in the same frequency range as the original 802.11, i.e., the 2.4 GHz Industrial-Scientific-Medical (ISM) band, but the "a" version ventured into the 5 GHz Unlicensed National Information Infrastructure (U-NII) band. Since 802.11b equipment was simpler to develop and build, it was adopted by the industry first. 802.11b technology soon established a grip in the market and proved the viability of WLAN technology in general.

In March 2006, the 802.11e [2] standard was approved. This new standard fortifies the need for establishing quality of service (QoS) for multimedia traffic by modifying the medium access control (MAC) layer. There has been numerous research on WLAN towards the approval of 11e, as seen in Table 1-1. 10% of the research papers in 2008-09 are on multimedia applications. The WLAN platform, is by far, the most widely used as opposed to HomeRF, Bluetooth, High Performance Radio LAN (HIPERLAN) version 1 and 2, and is capable of reaching a data rate of 54Mbps, given by 802.11g. Overall, WLAN is a technology for the future communication field.

Table 1-1: Statistics of research on WLAN from IEEE Explore

Year	2000	2001	2002	2003	2004	2005	2006	2007	2008-2009
Number of research papers	70	127	240	541	776	916	954	1027	903

A WLAN can be set up in two main architectures, ad-hoc (distributed control) and infrastructure LAN (centralized control), as seen in Figure 1-1. The ad-hoc network (also called peer to peer mode), in Figure 1-1a, is simply a set of WLAN wireless stations that communicate directly with one another without using access point (AP) where the point coordinator (PC) resides or any connection to the wired network. For example, an ad-hoc network can be formed by two laptops with a network interface card (NIC). There is no central controller; mobile terminals can communicate using peer-to-peer connections with other terminals independently. The network may still include a gateway node to create an interface with a fixed network. As an example this kind of setup might be very useful in a meeting where employees bring laptop computers together to communicate and share information even when the network is not provided by the company. An ad-hoc network could also be set up in a hotel room or in the airport or where the access to the wired network is barred. The infrastructure LAN network, in Figure 1-1b, consists of an arbitrary number of mobile or fixed terminals in addition to AP. The APs are located between mobile terminals and the fixed network. All data transmission is controlled and conveyed by the AP. The AP, in which the PC resides, is also responsible for sharing resources between terminals. In an infrastructure LAN network the immediate destination will be AP, which then forward the frame to the appropriate destination. Using high-gain external antennas, the AP can also be used in fixed point-to-point arrangements, typically at ranges up to 8 kilometres. This could increase to a range up to 80–120 km (50–75 miles) where line of sight can be established. Typical range of an AP using radio frequencies is roughly 100 to 300 meters. The range varies widely with the geometry and

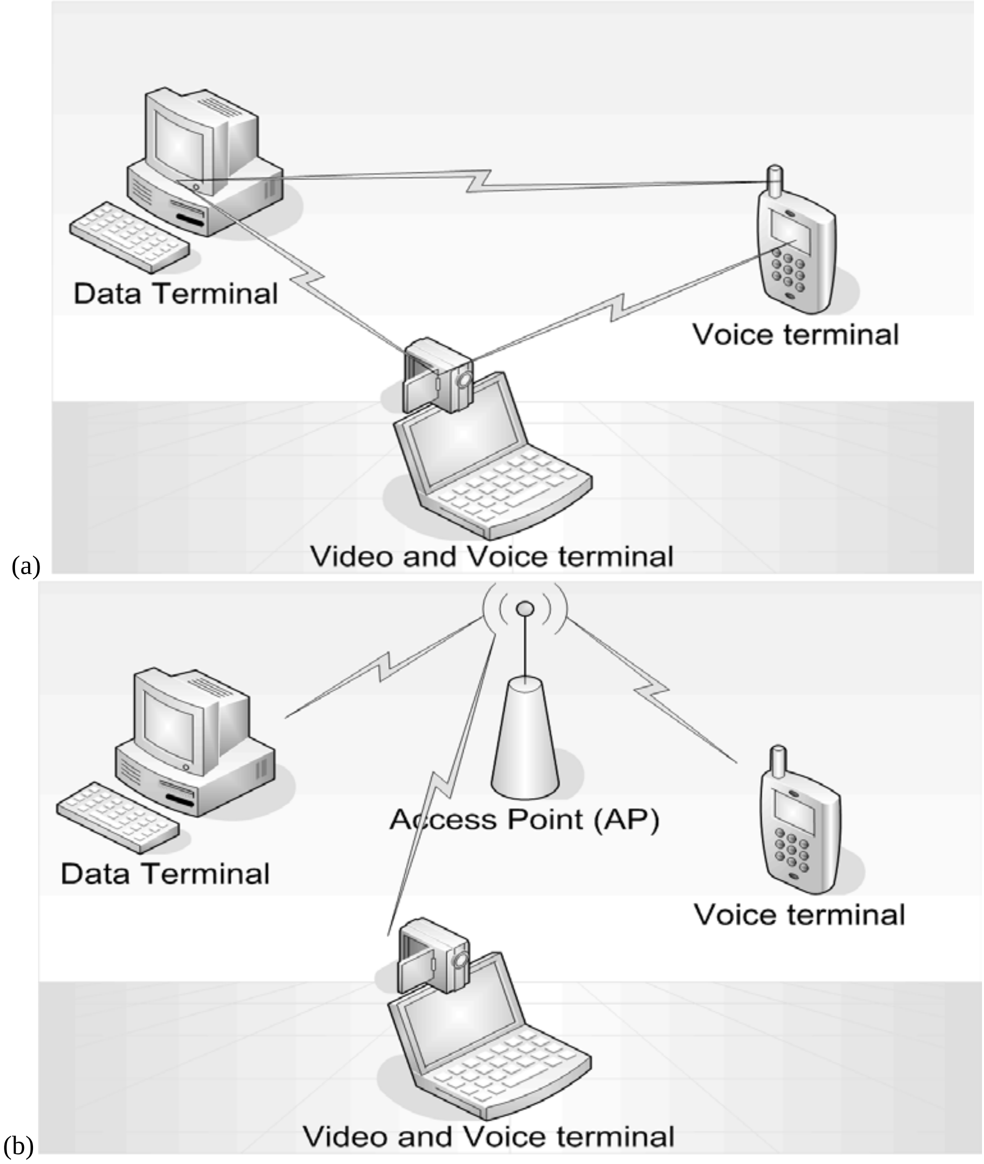


Figure 1-1 (a) ad-hoc and (b) infrastructure networks

other physical properties of the space in which it is used [3]. Throughout this thesis the word AP and PC will be used interchangeably.

The PC is supported by two coordination functions namely the point coordination function and distributed coordination function (PCF and DCF). In a PCF-enabled WLAN, the contention and contention-free (CP and CFP) alternates in a superframe (SF). This concept will be further discussed in section 2.2.

Fulfilling QoS requirements in WLAN is a well known problem. Voice applications, for example, require a stringent QoS requirement in terms of delay. Due to its highly interactive nature, International Telecommunication Union (ITU) [4] recommended that the conversational speech signal have less than 250msec of delay. Interactive-video applications, on the other hand, requires less than 150mesec of delay and no more that 30msec of jitter. Performance requirements of respective applications will be discussed further in the next chapter.

1.2. Scope of the Study

The aim of this study is to design a protocol that will enable transmission of interactive multimedia (IMM) applications over high-speed wireless networks efficiently while providing fairness to non IMM applications. Efficient here means that the QoS requirements of the IMM applications are met. Interactive media can be defined as media that allows for active participation by the recipient, hence interactivity e.g. computer gaming, interactive advertising [5-7]. The WLAN architecture test bed is the infrastructure network, representing high-speed network. A comprehensive study of heterogeneous WLAN with a mix of data, video and voice traffic is made. Performance statistics obtained from numerous simulation scenarios were collected and analyzed. An efficient protocol to provide to transmit IMM traffic is developed. The performance metrics of the proposed protocol is verified against performance of existing standard scheme.

1.3. Problem Statement

The research questions that need to be addressed in this study are:

A. How is the centralized controlled access in the infrastructured WLAN with contention-free coordination function (i.e., PCF) beneficial in implementing time-bounded multimedia or IMM applications?

B. What are the end-to-end delay, jitter and throughput profiles of multimedia traffic in WLANs?

C. What could be a modified access scheme for efficient management of the polling list and reduction of polling overheads for QoS improvements?

D. What performance matrices may be used to support that the scheme works?

E. How can this scheme be implemented?

1.4. Motivation

Optimal distribution of bandwidth in WLAN while providing fairness is a challenge as described by Pong [8]. However, existing literature supported by our preliminary simulation outcomes show a scope of developing a scheduling scheme to support QoS requirements of IMM communications over WLAN.

1.5. Contributions

In summary we have proposed a MAC scheme for IMM communication over WLAN under heterogeneous or mixed traffic scenario. We have developed the protocol by leveraging the inherent advantages of both the contention based and the polling schemes.

This was achieved by optimizing the CP and CFP in the SF. Various critical QoS metrics such as throughput, delay and jitter are taken into consideration during the development of the protocol. We have demonstrated performance improvement over current schemes in terms of these QoS metrics for mixed traffic environment through extensive simulations. The thesis makes four main contributions to the field of Wireless Communication and Protocol Design.

A. The thesis profiles the performance statistics namely delay, jitter and throughput of IMM and non-IMM traffic in a heterogeneous WLAN. This study has found that generally the PCF which provides centralized controlled access is more efficient in carrying IMM traffic in WLAN. Polling of IMM traffic facilitates its transmission even in high load WLAN scenario.

B. The second significant contribution of this thesis is the optimal partitioning of the SF into CP and CFP according to the QoS requirements of the IMM applications. Optimal bandwidth is achieved by maximizing utilization of IMM traffic while providing fairness to the non-IMM traffic.

C. The research contributes a functional self-adaptive medium access scheme that generates an optimum SF configuration that could be implemented in the access point (AP) of a centralized WLAN.

D. This research also introduces a modified analytical model for the PCF and DCF in WLAN.

E. This work provides guidance for further research direction for further improvements in using cross layer optimization.

Empirical results from discrete event simulation validate the scheme in typical WLAN scenarios. The simulation model used in our simulation is verified by comparing the arrival rate, packet length and total volume of data input and output, with that of the standards.

1.6. Organisation of the Thesis

A chapter by chapter summary of this thesis is given below.

Chapter 1 introduces the current scenario of wireless communications. The chapter begins by explaining the available platforms, the applications riding WLANs and the challenges that the scenarios impose on future wireless networking. The scope of the study, its importance and its contribution to the field of knowledge are outlined in the last section of this chapter.

Chapter 2 of this thesis describes the fundamentals in wireless multimedia network. A detailed discussion on the MAC protocol is presented here including the MAC protocol field frames and its uses. Performance requirements of various IMM applications are defined. The cross layer design concept is mentioned as a potential future work.

Chapter 3 gives a comprehensive examination of current state of the art technology for WLAN improvements for mixed traffic environment, in the form of a literature review. Evaluations of various MAC improvement initiatives are made on the methodology, evaluation tools and metrics discussed in the respective literatures. The chapter concludes with justifications in conducting this study, together with the conceptual framework.

Chapter 4 provides explanation for the methodology employed in achieving the outcome of this study i.e., simulation. Tools include the custom C-code development compiled directly into the OPNET, discrete event simulation software. This chapter presents the initial baseline model and prototype, and discussion of preliminary results.

Chapter 5 describes the development of a modified access scheme called the MAC with dynamic superframe selection (MDSS). This chapter starts with prototype experimental design of models used for simulation of the WLAN. This was based on the existing and then, modified Carrier Sense Multiple Access (CSMA) MAC protocol. This simulation models are used as vehicles for further modifications and enhancements as the study progresses. Preliminary findings are analysed and results are presented. The models are validated at the end of this chapter.

Chapter 6 discusses a mathematical model suitable for describing the CFP and CP in WLAN. This chapter also describes the improvements made by the author to the existing mathematical model. As the WLAN traffic trends emphasize on IMM than non-IMM traffic hence, maximizing resources for IMM while providing fairness for non-IMM is considered most appropriate. A simpler objective function in turn reduces the complexity of the solution. The chapter concludes with a comparison of the proposed scheme with other existing schemes.

Chapter 7 presents the findings obtained from variation of parameters in the simulation model. The MDSS is implemented in different scenarios closely depicting real-life WLAN environment. Results are compared to existing schemes including 802.11e.

Chapter 8 gives the conclusions of the thesis. It summarizes the main achievements, contributions and results for further research directions.

Chapter 2

QoS in Wireless Multimedia Networks

The scenario of network traffic today is varying and widely diversified. The Wireless Multimedia Forum Technical Working Group, defined the wireless multimedia service as, streaming multimedia (on demand/live/scheduled), downloading multimedia, uploading multimedia, multimedia messaging (e.g. video email), wireless video surveillance (e.g. wireless video camera), real time multimedia communication (e.g. videophone, videoconferencing), and interactive multimedia games and entertainments. These wide ranges of traffic require different guarantee of quality of service (QoS). Streaming multimedia, for example, requires high bandwidth coupled with tight delay constraints as packets need to be delivered in a timely fashion to guarantee continuous media play back. When packets are lost or arrive late, the picture quality suffers. As a result, each of these services imposes varying QoS requirements on the underlying system infrastructure i.e., the network. Wireless Multimedia Network (WMN) sometimes known as Multimedia Wireless Network (MWN) could be termed as any wireless network carrying a mixed composition of traffic/application. The bandwidth limitations of WMN results to trade-offs between presentation-data fidelity and interactive performance [7]. For example, digital video is commonly encoded with lossy compression to reduce bandwidth, and frames may be skipped during playback to maintain synchronization. These trade-offs depend on physical data representations, device and network performance. If the WMN is to support digital video and other IMM applications, QoS provisioning in the medium access control (MAC) layer is a possible candidate for efficient transmissions. Published research on WMN has dated back since 1997 [9-12] to name a few. Physical layer improvements have been the main technique proposed in most of these research but MAC layer improvements were also researched [12-14]. Some of these papers will be discussed in the later sections.

2.1. WLAN Standard and OSI Network Protocol Stack

The nature of 802.11 WLANs, which is a shared resource with limited bandwidth, can be a challenge to support the robust and multiple application types, as WLAN 802.11 is designed for best effort services only. The lack of built-in mechanisms for protocol layers, particularly the MAC layer, to optimally adapt to channel conditions and specific applications requirements of real time services makes it difficult to provide QoS guarantees for multimedia applications.

A network is a wide variety of systems of interconnected components [15]. It follows, that a wireless network is comprised of devices with wireless adapters communicating with each other using radio waves. The layered open systems interconnection (OSI) architecture, for networking, on which the current internet architecture is based on, is the heart of today’s network architecture.

The hierarchy of layers, in the OSI architecture, provides natural abstractions to deal with the hierarchy present in networks. In this networking framework, each layer communicates with its peer using a set of rules and conventions collectively known as layered protocol. Each layer should perform its own defined functions, shielding it from the details of how the services are implemented in the other layers. In implementing protocols in seven layers, control is passed from one layer to the next, starting at the application layer in one station, and proceeding to the bottom layer, over the channel to the next station and back up the hierarchy. The interactions between layers is controlled, each layer has the property that it only uses the functions of the layer below, and only exports functionality to the layer above. This is conducted primarily through the addition of protocol headers added to packets as they pass through the hierarchy.

While tightly interrelated protocol layering is an important abstraction that reduces network design complexity by splitting the network into smaller modules, it is not well suited to wireless networks, since the nature of wireless medium makes it difficult to decouple the layers. An important aspect to note is that wireless channels and networks are dynamic in behaviour, such as temporal and spatial changes in channel quality and user distribution. Furthermore, meeting the end-to-end performance requirements of demanding applications is extremely challenging without interaction between protocol layers. The conventional layered protocol architecture, described in the previous paragraph, is inflexible and unable to adapt to such dynamically changing network behaviours, since the various protocol layers can only communicate with each other in a strict and primitive manner. In such a case, the layers are most often designed to operate under the worst condition, rather than guaranteeing QoS by adapting to changing conditions.

Guaranteed QoS in wireless networks will involve mechanisms, algorithms and schemes at various layers of the OSI architecture; in the physical (PHY) layer, MAC layer, internet protocol (IP) layer and transport layer, interacting dynamically with each other. Hence, enabling more exchange of information between the layers across the protocol stack and better utilization of the resources.

2.2. Medium Access Control (MAC)

Resources in WLAN are managed in the MAC layer. The MAC layer manages and maintains communications between 802.11 stations (radio network cards and access points) by coordinating access to a shared radio channel and utilizing protocols that enhance communications over a wireless medium. References including [16-21] provide excellent references for MAC layer description in 802.11. Often viewed as the "brains" of the network, the 802.11 MAC Layer uses an 802.11 Physical (PHY) Layer, such as 802.11b or 802.11a, to perform the tasks of carrier sensing, transmission, and receiving of 802.11 frames [16]. In 2005, IEEE 802.11e [2] was approved as a standard that defines a set of QoS enhancements for WLAN applications. The standard acknowledged the importance of multimedia in WLAN.

The MAC layer is responsible for moving data packets to and from one device to another across a shared channel. Some of the important components of the MAC layer are the network configuration, channel access, multiple access, user and data privacy, power-management mechanisms, fragmentation, multimedia service, packet forwarding, mobility support, MAC layer management and the MAC frames [19].

The general MAC frame format specifies a set of fields that are present in a fixed order in all MAC frames. This is shown in Figure 2-1. All depicted fields occur in all MAC data frames. The “Address 4” field is only used if the wireless network is being used to implement the distribution system (DS). Other fields, such as Address 2, Address 3, Sequence Control, and Frame Body, may be omitted in certain other frame types [1, 18].

Octets:2	2	6	6	6	2	6	0-2312	4
Frame Control	Duration/ID	Address 1	Address 2	Address 3	Sequence Control	Address 4	Frame Body	FCS

← MAC Header →

Figure 2-1 MAC Frame Format [1]

To ensure a regulated usage of the shared medium, one of the three classes of channel access mechanisms could be employed as mentioned in [21]. The carrier sensing-random access with collision detection (CSMA/CD) gives fairly high throughputs commonly used in the Ethernet but it is not a popular mechanism in WLANS. The controlled access mechanism is often used in manufacturing where terminals could be robotic arms performing specific jobs. Their implementation and maintenance, however, are fairly complex. The combined random and controlled access mechanism is a good candidate for delay-sensitive applications. These mechanisms tell the stations when to transmit and receive, so that packets sent do not collide. The channel access mechanism hence is the core of the MAC protocol. Different protocols are used for different shared networks. This thesis aims to evaluate and improve WLAN combined channel access mechanisms by means of improving the efficiency of polling in WLAN MAC.

The two access mechanisms in the IEEE 802.11 MAC schemes, as shown in Figure 2-2, are namely the mandatory CSMA-based Distributed Coordination Function (DCF) and the optional contention-free Point Coordination Function (PCF). These schemes control how stations gain access to the available bandwidth. PCF is intended to provide better support for real time services due to its contention free trait.

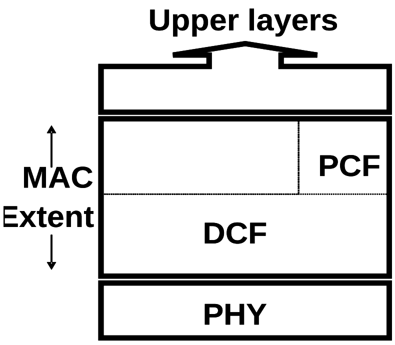


Figure 2-2 DCF and PCF in MAC layer

Figure 2-3 shows the relative lengths of the timing intervals i.e., the short interframe space (SIFS), the PCF interframe space (PIFS), DCF interframe space (DIFS) and the extended interframe space (EIFS). The duration of the basic timing intervals are specified according to the particular physical layer being used.

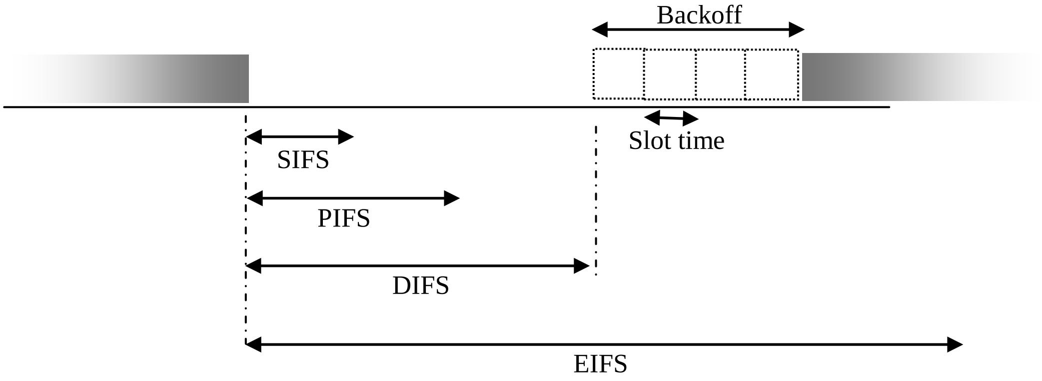


Figure 2-3 Diagrammatic Representation of the Access Mechanisms Showing Interframe Spaces

SIFS is the shortest interval which is the separation of frames within a transmission sequence of the frame exchange protocol. PIFS is slightly longer which is equal to one SIFS plus one slot time, while DIFS is equal to one SIFS plus two slot times. The EIFS is much longer than DIFS and is used to allow stations to regain timing synchronization with the rest of the network when a transmission is received in error.

The DCF is the basic mechanism that controls access to the wireless medium. All 802.11 stations are required to support DCF services. The period during which the DCF operates is referred to as the contention period (CP). After receiving a request for transmission from higher layer protocols, the MAC will check both physical and virtual carrier sense mechanisms. Once the medium is determined to be idle by both sensing mechanisms for an interval of DIFS (or EIFS if the previous transmission contained errors) the MAC may begin transmitting the frame. If the medium is determined to be in use during the DIFS-interval, the MAC will increment the retry counter associated with that frame and defer until an idle DIFS-interval to begin backing off. Transmission of the frame can begin only when backoff timer has expired. The back-off timer significantly reduces the number of collisions and corresponding retransmissions, especially when the number of active users increases. Since WLAN is radio based, a transmitting station cannot listen for collisions while sending data, mainly because the station cannot have its receiver on, while transmitting the frame. As a result, the receiving station needs to send an acknowledgement (ACK) if it detects no errors in the received frame. If the sending station does not receive an ACK after a specified period of time, the sending station will assume that there was a collision (or RF interference) and retransmit the frame. Numerous researches on DCF back-off scheme are being conducted to improve DCF performance, some of which will be discussed in the following chapter.

Due to the combination of contention and backoff employed by DCF, stations may experience extremely long wait-times for access to the medium. This possibly long delays as well as wide variation in delay times may be detrimental to real time traffic, hence PCF could be used.

The PCF is an optional mechanism that uses a poll and response method to eliminate contention for the medium. In this centrally controlled mechanism, the point coordinator (PC) located in the access point (AP), controls access to the wireless medium. The PC gains access to the medium using procedures similar to those used in DCF. However, instead of waiting for a DIFS-interval, it is only required to wait a PIFS-interval before determining the medium is idle and taking control of the medium. Once the PC has acquired the medium, it sends a beacon frame notifying stations of the beginning of the period of PCF operation known as the contention-free period (CFP). The beacon contains the maximum expected duration of the CFP, which stations use to update their network allocation vector (NAV)s. During the CFP, the PC delivers frames to station while also individually polling stations that have previously registered on the polling list requesting contention-free service. Each station can send one data frame for each CF-Poll received. By setting an appropriate CFP repetition interval, this mechanism can guarantee a bounded delay for transmission of packets arriving at stations that have requested this service.

To maintain control of the medium during CFP, the PC ensures that the interval between frames is not longer than PIFS. If the PC does not receive a response to a data transmission or a CF-Poll within a period of SIFS, it will transmit its next frame before the end of a PIFS. The end of the CFP is announced when the PC sends a contention free end (CFP-End) frame. With this frame, stations reset their NAVs and may begin competing for access to the medium under normal DCF.

2.3. Performance Requirements

The trends of the next generation WLAN is ubiquitous, real time, multimedia communications. The only hope for dramatically increased fidelity, akin to communicating in person, is high-speed access and transport for any medium, anytime, anywhere, and in any volume [22]. Given this trend of multimedia applications traversing the WLAN, delivering these applications will require a specific performance requirements [23]. ITU-T G:1010 [24] provides guidance on the key factors that influence Quality of Service (QoS) from the perspective of the end-user. Key parameters that could impact users are delay, delay variation (jitter) and information loss.

Authors in [25] classify communication applications into real time streaming, real time block transfer and non-real time applications. Examples of real time streaming include web pages that have streaming audio or video, internet telephony and new on-demand multimedia playlist. Real time block transfer could be reading mail from a server or application sharing. Non-real time applications include mail delivery to destination e-mail server or file transfer.

Requirements for conversational voice are heavily influenced by one-way delay. In fact, there are two distinct effects of delay. The first is the creation of echo in conjunction with two-wire to 4-wire conversions or even acoustic coupling in a terminal. This begins to cause increasing degradation to voice quality for delays of the order of tens of milliseconds, and echo control measures must be taken at this point (provision of echo cancellers etc [26]. The second effect occurs when the delay increases to a point where it begins to impact conversational dynamics, i.e., the delay in the other party responding becomes noticeable. This occurs for delays of the order of several hundred milliseconds [27]. However, the human ear is highly intolerant of short-term delay variation (jitter). As a practical matter, for all voice services, delay variation due to variability in incoming packet arrival times must be removed with a de-jitterizing buffer. Requirements for information loss are influenced by the fact that the human ear is tolerant to a certain amount of distortion of a speech signal. In IP-based transmission systems a prime source of voice quality degradation is due to the use of low bit-rate speech compression codecs and their performance under conditions of packet loss.

Requirements for information loss are essentially the same as for conversational voice (i.e., dependent on the speech coder), but a key difference here is that there is more tolerance for delay since there is no direct conversation involved. The main issue therefore becomes one of how much delay can be tolerated between the user issuing a command to replay a voice message and the actual start of the audio. There is no precise data on this, but based on studies related to the acceptability of stimulus-response delay for telecommunications services, a delay of the order of a few seconds seems reasonable for this application. In fact, a distinction is possible between recording and playback, in that user reaction to playback is likely to be the more stringent requirement.

Streaming audio is expected to provide better quality than conventional telephony, and requirements for information loss in terms of packet loss will be correspondingly tighter. However, as with voice messaging, there is no conversational element involved and delay requirements for the audio stream itself can be relaxed, even more so than for voice-messaging, although control commands must be dealt with appropriately (see Section 3.4).

A general classification of video into six levels of quality, and a mapping to various services, is given in [25]. Videophone as used here implies a full-duplex system, carrying both video and audio and intended for use in a conversational environment. As such, in principle the same delay requirements as for conversational voice will apply, i.e., no echo and minimal effect on conversational dynamics, with the added requirement that the audio and video must be synchronised within certain limits to provide "lip-synch". Once again, the human eye is tolerant to some loss of information, so that some degree of packet loss is acceptable depending on the specific video coder and amount of error protection used. It is expected that the latest MPEG-4 video codecs will provide acceptable video quality with frame erasure rates up to about 1%.

From a user point of view, a prime requirement for any data transfer application is to guarantee essentially zero loss of information. At the same time, delay variation is not generally noticeable to the user, although there needs to be a limit on synchronisation between media streams in a multimedia session (e.g. audio in conjunction with a white-board presentation). The different applications therefore tend to distinguish themselves on the basis of the delay which can be tolerated by the end-user from the time the source content is requested until it is presented to the user.

Web-browsing could be the retrieving and viewing HTML component of a Web page, other components e.g. images, audio/video clips are dealt with under their separate categories. From the user point of view, the main performance factor is how quickly a page appears after it has been requested. Delays of several seconds are acceptable, but not more than about 10 seconds [25].

Still-image includes a variety of encoding formats, some of which may be tolerant to information loss since they will be viewed by a human eye. However, given that even single bit errors can cause large disturbances in other still image formats, it is argued that this category should in general have zero information loss. However, delay requirements for still image transfer are not stringent and may be comparable to that for bulk data transfer, given that the image tends to be built up as it is being received, which provides an indication that data transfer is proceeding.

Requirements for interactive games are obviously very dependent on the specific game, but it is clear that demanding applications will require very short delays of the order of a fraction of a second, consistent with demanding interactive applications.

Telnet is included here with a requirement for a short delay of a fraction of a second in order to provide essentially instantaneous character echo-back.

E-mail is generally thought to be a store and forward service which, in principle, can tolerate delays of several minutes or even hours. However, it is important to differentiate between communications between the user and the local email server and server-to-server transfers. When the user communicates with the local mail server, there is an expectation that the mail will be transferred within a few seconds.

Instant messaging primarily relates to text, but can also include audio, video and image. In any case, despite the name, it is not a real time communication in the sense of conversational voice, and delays of several seconds are acceptable.

In principle, the only requirement for background applications is that information should be delivered to the user essentially error free. However, there is still a delay constraint, since data is effectively useless if it is received too late for any practical purpose.

A summary of the important QoS requirements are given in table 2-1.

Table 2-1: Performance targets for IMM applications [24].

	Application	**QoS Requirements**		
		Delay	**Jitter**	**Loss**
Audio	**Conversational Voice**	**150 msec**	**1 msec**	**3%**
	Voice Messaging	**1 sec**	**1 msec**	**3%**
	High Quality Streaming Audio	**10 sec**	**1 msec**	**1%**
Video	**Videophone**	**150 msec**		**1%**

2.4. Cross-Layer Design Concept

There is a phenomenal concept that does away with the rigid structure of layered protocol architecture. It is called cross-layer design concept [28] where each layer is not responsible to serve only the higher layer, refer to Figure 2-4. Parameters of the various layers in the protocol stack will be exchanged interactively to cope with the robust characteristics and constraints especially with multimedia traffic and WLANs. While the traditional OSI-layer-based architecture from wired networks has proved to be quite useful for developing smart algorithms and techniques for different communication systems, it seems to be suboptimal for wireless communication systems. This is due to the fact that the wireless medium is available to multiple users who intend to get access and transmit their information, and its inherent variability in both the time and frequency domains.

Potential useful information that could be exchanged between layers includes:

- channel state information (e.g. channel impulse response estimation) both in time and frequency domain
- QoS-related parameters, including delay, throughput, bit error rate (BER), and packet error rate (PER) measurements for each layer involved in cross-layer interaction, especially concerning the end-to-end requirements; and
- traffic pattern offered by each layer to the others, including data traffic information, knowledge of the data rate, data burstiness, data fragmentation, packet sizes, and information about queue sizes.

Cross-layer design comes with cost. Additional signalling to extract relevant parameters from one layer that could be useful to other layers and control plane information that should be exchanged and the corresponding required transmission resources occupied are some of the overheads. Increase in computation complexity of all the protocols involved given realistic computational capabilities of existing hardware or its anticipated evolution should be carefully taken into account when designing cross-layer mechanisms.

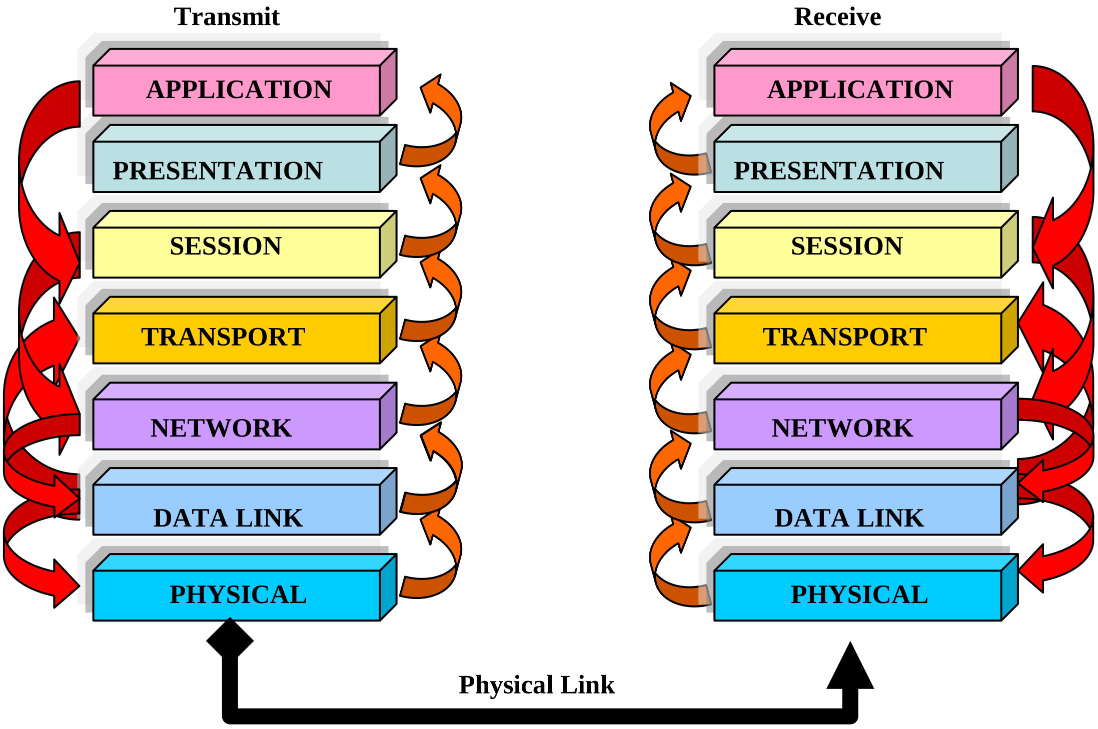


Figure 2-4 Cross-layer Protocol Design Concept Showing Dynamic Exchange of Information between Layers.

Implementation of cross-layer concept in WLAN is indeed a challenging task and would be a good candidate for further research.

Chapter 3

Review of Related Work

This chapter provides a comprehensive review of published literature related to wireless local area network (WLAN) medium access control (MAC) improvements. A particular emphasis is given to 802.11e acknowledging its role in providing a standard for quality of service (QoS) in WLANs.

The growth of real time (RT) and non real time (nRT) applications is one of the major driving forces behind the rapid evolution of next-generation wireless networks where the traffic is expected to be varying and widely diversified [29]. Consequently, the incorporation of interactive multimedia (IMM) applications on wireless networks is somewhat a natural evolution. WLAN was initially developed to carry nRT data (e.g. file transfer, file printing etc.). Since this type of nRT traffic is bursty in nature, the channel access schemes were based on contention. However RT traffic (e.g. voice, video and other IMM applications) are different from this traditional nRT traffic as they have stringent constraints on QoS metrics like delay and jitter.

Performance improvements of a WLAN carrying IMM traffic means improving the critical QoS metrics, namely end-to-end delay, jitter, percentage of packet loss and throughput. A survey of QoS enhancements by Ni [30] illustrates the natural limitations of 802.11 in handling IMM traffic. Employing contention free channel access schemes are the plausible and intuitive approach to accommodate these innate requirements. Thus it is only natural to have a combination of contention free and contention based schemes for the channel access protocol of WLANs carrying both bursty and RT traffic. IEEE 802.11 MAC had both of these phases but the standard did not specify how to balance them. The contention based channel became the de facto scheme because of the paucity of IMM traffic during the early years. The contention free scheme was left as an option and was not usually implemented. Another open issue with contention based scheme is fairness. The authors in [31] proposed a CSMA with copying collision avoidance

(CSMA/CCA) to overcome this fairness issue. However, their simulations were done only for fixed data rate, consequences of mixed data rate has not been explored.

3.1. WLAN MAC Improvements

Approval of IEEE 802.11e in the recent year has proven the popularity of IMM applications in carrying real time traffic over WLAN. Various studies on modifying the contention based channel schemes to enable them to meet the QoS requirements of real time IMM traffic has been conducted. Majority of researches revolve around the measures taken on MAC protocol enhancements to satisfy the need of IMM applications with respect to their QoS requirements [32-35]. Relevant articles will be reviewed in the next section. This trend of research is mainly due to the fact that contention based channel access protocols were the norm for practical WLANs. Comparatively less significant number of work was performed on the contention free centralized channel access mechanism for the WLAN. We believe that integrating the contention free channel access to MAC protocol is imperative to enable the WLAN to carry a mixed traffic of both nRT data and RT IMM traffic. A balanced utilization of the contention based and contention free schemes will let us leverage the inherent advantages associated with the two schemes to handle non real time data and real time IMM traffic respectively.

Basically, the function blocks of wireless MAC protocol technology enhancements could be illustrated as shown in Figure 3-1. Available literature can be categorized into five major blocks. Policing, call admission, and congestion avoidance are edge-oriented technology where schemes are applied on the WLAN edges. While resource reservation and scheduling could be collectively called (intra) media access technology. Sole or combination of these building blocks has produced numerous novel solutions to WLAN MAC protocol improvements.

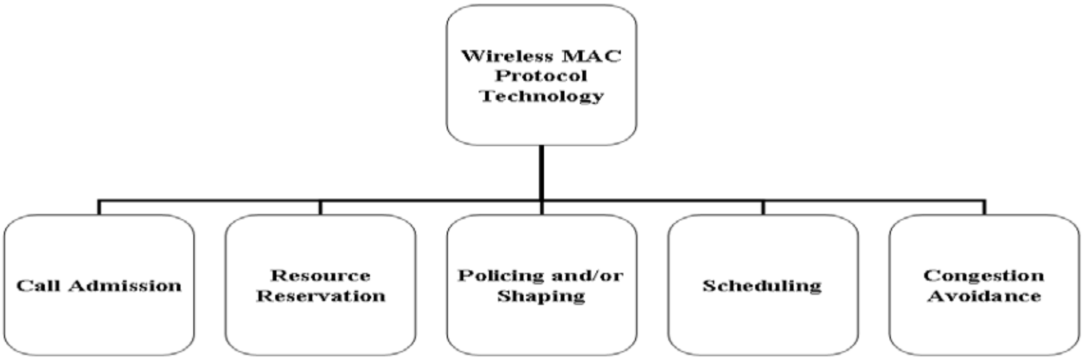


Figure 3-1 Wireless MAC Protocol Technology Basic Functional Blocks

Admission control mechanism decides whether or not a new connection should be admitted by the system [8, 36, 37]. In the call admission and resource allocation strategy, the application will signal the network on type of connection and QoS that it would require. The network in turn will perform a pro-active bandwidth management strategy to accommodate the application requests. This will include, establishing the route of the connection, reserving enough resources to meet the QoS requirements and rejecting a call if the network does not have enough resources to meet the call. The flow classification method uses a certain classification criteria to identify packets/cells in order to provide differential treatment. Classification criteria would include type of service (ToS) or multiprotocol label switching (MPLS). In traffic shaping, the rate at which a flow is allowed into the network is regulated. The system may use typically leaky or token bucket mechanism. The congestion avoidance mechanism typically works best with TCP. A common scheme used in congestion avoidance mechanism is random early drop (RED) which stochastically drops packets as congestion occurs [38]. We will be focusing on scheduling mechanisms in this review. Typical scheduling disciplines are first-in first-out (FIFO), priority queuing (PQ) and weighted round-robin (WRR). WRR has other variants for example fair queuing (FQ), weighted fair queueing (WFQ) and deficit round-robin (DRR). These are queuing mechanisms to provide differing levels of services. Some of these scheduling disciplines will be explained in the later part of this section.

The trends assume homogeneous traffic whereas practical WLANs of today are intended to carry mixed traffics. On top of this, most of the research works focus on contention-based channel access performance improvement mechanisms. Given the growing trend of heterogeneous traffic mix over WLAN we perform a comprehensive literature review on WLAN protocols improvements with particular emphasis on mixed traffic to establish the scope of improving the WLAN access scheme to carry IMM applications in a mixed traffic environment. This chapter will provide a comprehensive overview of the state of the art for WLAN MAC improvements with a particular emphasis for IMM traffic in a heterogeneous environment. Relevant researches on media access technology will be discussed in the following section.

3.2. IEEE 802.11e

IEEE 802.11 [1] and its variants have been the most popular WLAN standards. It is only relevant that review of the most recent research work related to improving IMM transmission by means of access schemes be discussed here.

In the legacy 802.11 MAC, the channel access is generally defined in two schemes, as shown in Figure 3-2, the mandatory DCF and the optional PCF. DCF employs CSMA/CA which is appropriate when the underlying traffic scenario is predominantly nRT data. The PCF, as opposed to the DCF, was designed to support time-bounded (i.e., real time multimedia) traffic. The access point (AP) would poll stations with real time traffic according to a polling list, and then switch to the contention period when stations use DCF. PCF was rarely implemented as time bounded applications over WLAN were not very common.

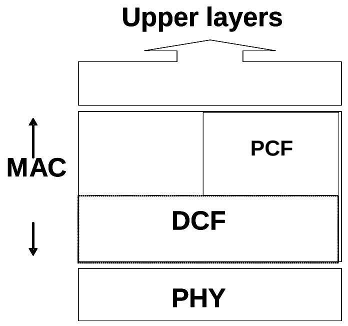


Figure 3-2 Channel Access in IEEE 802.11 MAC Layer

IEEE 802.11e amendment has been incorporated into the published IEEE 802.11-2007 standard. The enhanced distributed channel access (EDCA) mechanism extends the legacy DCF mechanism to include priorities, while hybrid coordination function (HCF)-controlled channel access (HCCA) mechanism [39] replaces PCF by centrally managing medium access. EDCA intends to address the QoS issue by including priority by grouping traffic into eight categories, ranging from lowest-priority best-effort traffic through highest-priority RT traffic that is extremely latency-sensitive. These traffic groups are mapped into four access categories (ACs). The AC specific traffic prioritization is implemented by using a combination of three parameters: Arbitration Inter-frame Space (AIFS), Contention Window (CW) and Transmission Opportunity (TXOP) limit. Each category is associated with AIFS, which it must wait for after detecting that the broadcast medium is idle before transmitting the frame. Higher-priority traffic has a smaller AIFS, which gives it freer access to the broadcast medium than lower-priority traffic. In addition to the AIFS, each station waits an additional, random period of time before transmitting, in order to avoid collisions with other traffic in the same EDCA category. The choice of AIFS, CW and TXOP limit has to be made in providing optimum QoS. Making this correct choice requires extensive research and testing[40-42].

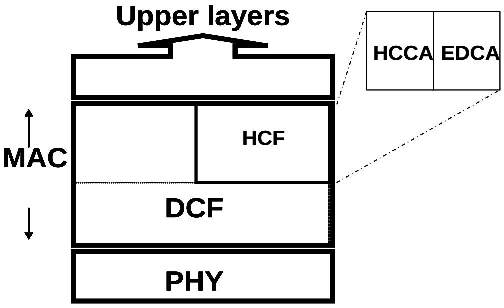


Figure 3-3 Channel access in IEEE 802.11e MAC layer

This ratification of IEEE 802.11e standard, as shown in Figure 3-3, has acknowledged the usefulness of central contention-free mechanism in handling real time multimedia traffic by introducing HCCA. Coexistence of the mechanisms of the legacy and new standards is one of the open issues. Improving the legacy PCF scheme could provide relatively less costly solution via centralized contention-free control of IMM traffic. A simpler scheme that gives a relatively comparable if not better outcome to 802.11e would be complement. Performance improvements by varying the PCF parameters like the PCF interframe space (PIFS), in the AP could be employed. Exclusive setting of the AP only implementing PCF could be worth exploring.

The design of WLAN MAC took into consideration diverse issues such as time-varying channel, bursty errors, and localized carrier sensing. Researchers have generally focused their work on rectifying problems, enhancing performance, and defining configuration parameters and tools for 802.11e and its predecessor. Recent research works have progressed from pre to post 11e ratification issues, which concerns coexistence of different WLAN standards, especially the MAC layer issues, that concerns development of new schemes to achieve fairness for WLANs supporting multimedia services. Cheng et. al. [41] and Wei et. al., [42], for example, have proposed schemes which make use of differentiations of both inter frame space (IFS) and contention window (CW) to achieve weighted fairness for two classes of services under enhanced distributed channel access (EDCA) mode in an 802.11e WLAN. Given the arbitration IFSs (AIFSs), their proposed scheme can properly set the corresponding CWs such that the ratio of the two classes' successful transmission probabilities can attain a pre-defined weighted-fairness goal.

Additionally, the throughput and delay of the two classes of services are derived. Their simulation results compare well with the analytical results. They referred to A. Banchs and X. Perez (2002), as bench marks [43]. The optimal configuration of EDCA is further addressed by Wei et. al. [44] where the author gathered that most of the previous studies assumed that all access categories (ACs) have the same packet length which untrue in most realistic cases. Generally, different applications have different traffic characteristics including packet length. They also studied how to achieve the maximal protocol capacity of EDCA by properly choosing the protocol parameters. They have proven that the maximal protocol capacity can be approximately achieved by setting the network at an optimal operating point. A parameter configuration scheme was proposed to achieve efficient channel utilization and proportional fairness among different ACs. Traffic is treated as best effort traffic not as IMM traffic though they are carrying IMM applications.

Research works traversing edge-oriented solutions like traffic shaping and admission control to schemes which exploits differentiations of both inter frame space (IFS) and contention window (CW) to achieve weighted fairness for different classes of services. The delayed approval of 802.11e motivated researchers [36, 45-48] in contributing more ideas and knowledge into defining the specifications of the standard.

Pre 11e scenarios saw work on improvements on WLAN by improving the efficacy of MAC by means of edge-oriented techniques [37, 49-52], like admission control and traffic engineering. One obvious disadvantage about this technique is the additional overhead (e.g. memory and frame buffer size) and delay caused by the reservation process, it can degrade overall performance.

On handling real time multimedia traffic, with the MAC protocols have to deal and interact with complex physical layer characteristics in a flexible dynamic manner, where many transmission options could be made available. Categorization and prioritization of traffic classes and treating them differently in the MAC layer, favours polling over contention [34, 35]. This would be justified from our preliminary simulation studies which direct towards a scope of improving the WLAN QoS in IMM mixed traffic environment via improvement of the legacy PCF.

MAC protocols need to deal with complex physical layer characteristics in a flexible dynamic manner, where many transmission options are available. This issue was also the main theme of research in [12], and references therein, where MAC protocols for multimedia traffic in wireless networks for various classes of traffic were inspected. These classes are namely, constant bit rate (CBR) traffic for digital streaming voice and video applications, real time variable bit rate (RT-VBR) traffic for compressed voice and video, non-real time variable bit rate (nRT-VBR) traffic for data application, available bit rate (ABR) traffic for non-time-critical data application and unspecified bit rate (UBR) traffic for file transfer, system backup, email activities. Their article [12] effectively presents a survey on MAC protocols for IMM in wireless networks by giving basic overview of MAC protocol concepts and a framework on which a base qualitative comparisons on a multimedia traffic requirement is made. We adopt this technique in Chapter 4.

Subsequent published researches on MAC protocols for resource sharing include dedicated assignment, random access and demand-based assignment. It would be evident that demand-based channel assignments are useful for VBR and the hybrid condition of multimedia traffic. However, most of the rectifications are costly in terms of processing time or implementation. The survey of research publications shows that vast research efforts are focused on improving the DCF and contention based mechanisms in providing sufficient QoS to time-sensitive traffic. The design of high-speed DCF-based wireless protocols is an area that attracted significant research [32] with additional challenges to provide QoS in these networks. Treating IMM traffic as non-IMM and making the traffic contend for resources will not be the best way of fulfilling IMM QoS requirements.

Suzuki et. al and Zhu et. al, in [53, 54] reported that the weakness of the legacy DCF protocol namely the lack of QoS support was due to the fact that different traffic classes have different frame sizes. This is in contrast to data traffic which could be treated in a first come first serve, best-effort manner. In addition, the authors found that, high variation of throughput and delay is inherited from the exponential back off mechanism, particularly in overloaded situations i.e., in network saturation. We will vary our scheme to saturation if the effects discussed by Suzuki occur.

Typical demand-based assignment attempts to implement some form of contention reservation period to support multimedia applications. On support of multimedia applications, authors in [55] focused on two methodologies, namely, random access for asynchronous traffic and polling for real time asynchronous traffic. The authors have analysed several existent protocols' ability to support user QoS requirements. They found that all the protocols analyzed except HIPERLAN/1 favoured synchronous traffic over asynchronous traffic, which is transmitted only if there is sufficient bandwidth left over by synchronous traffic. Asynchronous traffic from RT-VBR applications supports service differentiation and traffic classes however were not provided by the underlying wireless network. In order to achieve that, the authors proposed priority-based medium access (with reservation) and efficient management of the waiting queues (in polling).

Priority-based access solutions were also proposed in [56] to overcome the weaknesses of WLANs in supporting multimedia traffic. The proposed priority DCF scheme modifies the backoff scheme so that higher priority stations have a shorter backoff time. This approach, however, may encounter a priority reversal phenomenon, as mentioned in [57, 58] . Since the contention window is exponentially proportional to the number of retransmission attempts, a high-priority backlogged station may experience a longer backoff time than a low-priority "unbacklogged" station. Hence a lower-priority station may take priority over a higher-priority station. Such a priority reversal phenomenon may discourage customers from buying high-priority WLAN services. Clearly, issues in DCF IMM transmission require further research and improvements before it could be implemented. The authors in [59] explored and analyzed the feasibility of IEEE 802.11b to sustain video applications. The bandwidth efficiency to transport packets of various sizes over 802.11b WLAN in distributed coordination function (DCF) mode was evaluated and its corresponding theoretical maximum data rates were proposed. Due to the overhead associated with the different protocol layers, the achievable efficiency of transport was found to vary widely with different packet sizes. Through controlled WLAN experimentation the effect of this phenomenon on video traffic is studied. Application to MPEG4 streams show that the packet overhead in 802.11 DCF modes causes a significant bottleneck for simultaneous multicast streams. To alleviate the problems the authors suggested operation in 802.11 PCF modes, using a statistical multiplexer and improving design of queues in the wireless access point.

Priority techniques could increase the complexity of a scheme, but are necessary for the realization of QoS constraints of varying types of IMM traffic. The authors in [60], have performed enhancements of DCF in large number of competing nodes. Unfortunately, this Early Backoff Announcement (EBA) which is a distributed reservation-based MAC protocol faces the problem of co-existing with legacy DCF stations.

EDCA of the amended IEEE 802.11e [2] extends the legacy DCF mechanisms to include priorities, while HCCA replaces PCF by centrally managing medium access built on top of EDCA, thus no issues of coexistence. The new standard however, left further open issues [45, 61, 62]. The choice of AIFS, CW and TXOP limit has to be made in providing optimum QoS. Making this correct choice requires extensive research and testing. This new standard simply acknowledged the need to apprehend issues of QoS in WLANs through the manipulation of MAC layer parameters when carrying time critical applications. EDCA simply provides a very high statistical likelihood [63] that it allocates higher levels of bandwidth to higher-priority traffic over the shared wireless medium, instead of explicitly guaranteeing specific QoS levels for distinct quality types. This, however, opens more doors for further improvements.

The issue of fairness in EDCA for example, was addressed in [64] where an algorithm that improves EDCA fairness is proposed. EDCA failed to protect the low bit rate flow from the same class of traffic and the authors claimed to be able to reconcile this. However, this algorithm was developed assuming only ideal channel conditions. Real network conditions like mixed traffic environment were not considered. Their effort was extended in [65], but ideal conditions was still assumed. Authors in [66] proved that EDCA parameters cannot adapt to changes in network conditions. Their performance evaluation revealed that CW provided in EDCA are too small for a large number of users and suggests adaptation of backoff parameters. This is consistent with what mentioned previously by [57, 58] . EDCA counterpart, the IEEE 802.11e HCCA, which was supposed to be designed for use with periodic traffic, unfortunately has mechanisms best suited for CBR traffic characteristics only. This will be explained later. Similarly, authors in [67] attempted to enhance the performance of EDCA by contention adaptation. These are some of the open issues related to EDCA still showing opportunities for improvements and rectifications. Thus the latest standard can be seen as far from being stable. However, the focus on polling-based scheduling as means of providing a more efficient transmission of IMM applications is central to our research.

Despite works on improving DCF and EDCA, most have dealt with traffic of constant bit-rate (CBR). Improvements for real time variable bit-rate (RT-VBR) traffic like video conferencing traffic were still open issues. DCF and EDCA are not efficient in handling RT-VBR traffic even through rectifications as contentions will impose further delay to these time-critical applications.

One of the well-known implementers in WLAN technologies [39] has commented that PCF successor HCCA, simply has too many variables to maintain. In its provision to support parameterized QoS, 802.11e HCCA inherits some of the rules of legacy PCF, and introduces many extensions. Like PCF, HCCA uses a polled-based mechanism to access the medium, thereby reducing contention on the wireless medium. The key differences between HCCA and PCF are that HCCA can poll the stations during CP and that it supports scheduling of packets based on the node’s specific traffic-flow requirements. Despite enhancements on polling schemes HCCA has still not addressed the problems of low throughput due to overhead induced by the polling frames, inefficient round-robin scheduling algorithm and consequences of varying channel conditions.

Ni et. al. in [66], illustrated that IEEE 802.11e HCCA was designed to be used to handle periodic traffic but unfortunately its mechanisms are best suited for CBR traffic characteristics only. It was reported that HCCA can guarantee the delay requirements for CBR traffic. However, the delays of VBR video flows were completely unrestrained. The HCCA polled-based scheduling scheme did not allow itself for supporting VBR traffic. It was not an adaptive mechanism which allowed for change of the negotiated parameters of the stream due to changing channel conditions. It was showed that HCCA delivered 99 percent of voice packets within a delay of 50 msec and 97 percent of CBR video packets within 50 msec, which was equal to the selected duration. Another type of scheduling mechanisms employed by Yang in [69] was based on queue length information updated in the QoS AP (QAP) each time a QoS data packet is received. However this scheme seemed inflexible as queue length of already polled QoS stations (QSTA)s and non-polled QSTAs may not be updated any more since they do not have the opportunity to transmit any packet to inform the QAP of its current queue length. The author in [27], proposed the extended HCCA (E-HCCA) to support VBR by evaluating the mean application data rate in the next round TXOP. It was reported in [70] that performance is worsened by the introduction of HCCA in their simulation. There are still considerable tuning needed for HCCA before it would perform as it is required.

Since the mechanisms proposed in 802.11e [2] to provide different levels of QoS, are far from being settled, the task of providing a QoS solution is left open in the standard and efforts in improving and fine tuning these mechanisms are ongoing. Several researchers have also reported on the significant increase in the complexity of the 802.11e MAC architecture. Most of the changes are direct consequences of introducing HCF with two new channel access functions: EDCA and HCCA. Upgrading MAC in the new standard would require extensive changes to existing functional blocks as well as adding new ones. Its implementation could significantly increase memory. The amount of additional memory is a function of the increase in the number of transmission queues. In the original standard there are two queues proposed: broadcast and multicast, and unicast. In the new standard, there are at least five queues proposed: broadcast and multicast, and four access categories (AC). If HCCA is also implemented, the numbers of additional queues for traffic streams vary between 1 to 8 for QSTA and 1 to any number for QAP limited by available memory. This will impact on both software architecture of the MAC and the operating system.

It is a general perception that HCCA has too many variables to maintain. In its provision to support QoS, the 802.11e HCCA inherits some of the rules of legacy PCF, and introduces many extensions. Like PCF, HCCA uses a polled-based mechanism to access the medium, thereby reducing contention on the wireless medium. The key differences between HCCA and PCF are that HCCA can poll the stations during CP and that it supports scheduling of packets based on the node’s specific traffic-flow requirements. Despite enhancements on polling schemes HCCA has still not addressed the problems of low throughput due to overhead induced by the polling frames, inefficient round-robin scheduling algorithm and consequences of varying channel conditions. Authors in [20, 21] has showed that HCCA still requires considerable tuning before it would perform satisfactorily and enhancements of the scheme are needed to support a mix of heterogeneous traffic.

The numerous works on 802.11e to date simply shows that this scheme still requires several adjustments to get settled. On that note, improvements of the legacy (more settled) protocol particularly the PCF can be made as a good compliment to the 802.11e technology.

3.3. Centralized Contention Free Function with PCF

The IEEE 802.11 standard has acknowledged the usefulness of central contention-free mechanism in handling real time multimedia traffic by introducing HCCA. Coexistence of the mechanisms of the legacy and new standards is one of the numerous open issues discussed in the previous sub-section. This study, however, will not be focussing on coexistence issue. PCF could provide relatively less costly in terms of solution via centralized contention-free control of IMM traffic, as described previously. Performance improvements by varying the PIFS, for example, in the AP have not been employed. Exclusive setting of the AP to only implementing PCF, could be explored. There is also scope to extend the PCF mechanisms by means of manipulating the superframe segmentation in a mixed traffic environment.

PCF, as opposed to DCF, was designed to support time-bounded IMM traffic. It uses a centralized polling-based channel access method to support these time-bounded services. Rasheed et. al. in [71] has shown that PCF-based nodes perform relatively more efficiently than DCF-based node. Some of the issues discussed in literatures on PCF are namely issues on polling sequence (scheduling), unpredictable beacon delays, polling methods, reduction of overhead caused by polling frames, superframe segmentation and configuration, coexistence issues of DCF and PCF and its switching and problems of contention. Authors in [72] show that contention-based medium access causes non-deterministic delays, therefore such schemes are not suited to voice traffic which require strict delay bound guarantees. Their research focuses on the schemes which do not use contention based approaches for voice traffic. Analytical performance evaluation and comparison of such schemes is carried out. They have acknowledged the contributions of adaptive polling scheme in reducing delay in voice transmission over WLAN. Their research, however, did not take into consideration a more realistic mix of WLAN traffic e.g. video.

Published literature in improving polling implementations include unpredictable beacon delays resulting in significantly shorter contention-free period (CFP) and unknown transmission duration of polled stations. These make it very difficult for the point coordinator (PC) to predict and control the polling schedule for the remainder of the CFP. Ksentini et. al. in [73] reported that PCF lacks mechanisms to differentiate traffic types, provided no mechanisms for the stations to communicate their QoS requirements to the AP, had no management interface to control and setup CFP and the polling schedule was not tightly controlled. This was proven via simulation reported in [74].

Despite proposals on the improvements of PCF, the industry implementers have not picked up these ideas. Now, its successor in HCF (i.e., HCCA) provides a polling mechanism similar to PCF that polls stations during a contention-free period and assigns a defined-duration period during which each station can transmit. In this way, HCF hopes to overcome PCF's limitation in terms of unpredictable transmission periods.

PCF limitations in performing its designated task in handling real time multimedia traffic have been a popular area of research. Figure 3-4 shows that the number of publications in this area is increasing every year. More are focused on the scheduling than polling issues, trying to deal with the transmission times of the information packets and overhead due to polling of stations that has no frames to transmit. Another issue that draw much attention is related to simulation modelling of wireless MAC protocols, including physical layer models, antenna and radio models, propagation models, modelling of smart antennas, impact of detailed physical models on protocol performance, and simulation studies of MAC protocol performance using OPNET simulator.

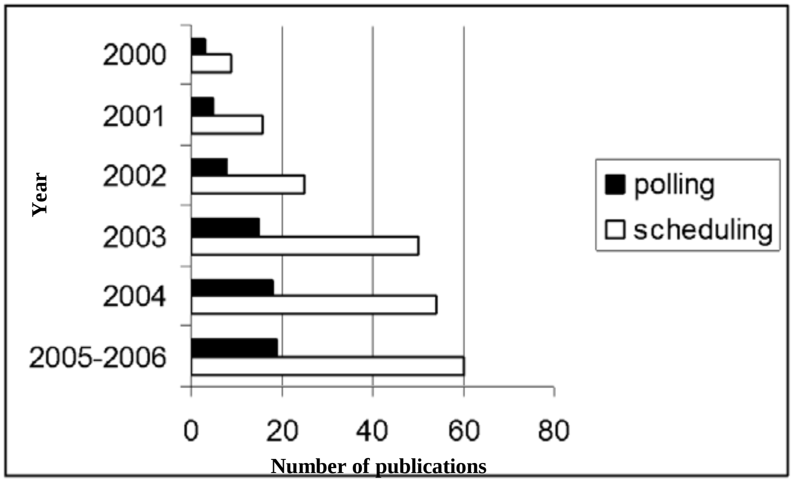


Figure 3-4 Graph Indicating Number of Publications for Polling and Scheduling in WLANs

Al-Karaki et. al. [63] supported by Cheng et. al. [75] made two comments that became the basis of our work. Firstly, during the controlled contention period, each host can update the channel requirement information only when they get the opportunity during the contention phase. However, communications can be made more active by allowing stations to indicate the change in their requirements in a prompt manner. Secondly, HCF entitles complexity in its operation due to the large number of variables that it needs to maintain. For example, AP will consider many variables to decide whether or not to admit new stations that are requesting association. Managing a high number of variables and communicating them between the stations and the AP, as the PC, on a regular basis may impose a large burden on the network. This is an issue that is popular among researchers.

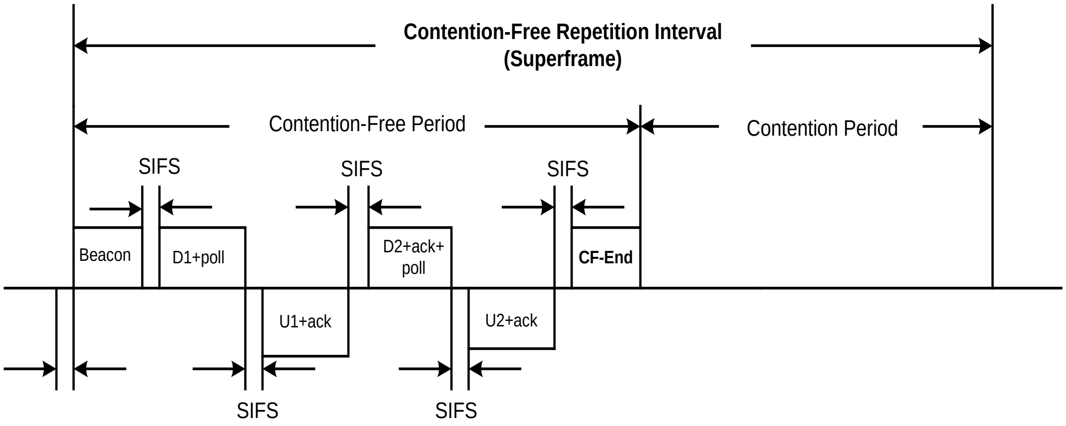


Figure 3-5 Segments of Superframe [1]

As the PC performs polling, to enable polled stations to transmit without contending for the channel, the CF-pollable station may transmit only the MSDU, that can be sent to any destination, which responds with (+)ACK in case of successful reception. Within a repetition interval (superframe) a portion of the time is allotted to contention free traffic, and the remainder is provided for contention based traffic, as shown in Figure 3-5. The contention free period (CFP) repetition interval is initiated by a beacon frame which is transmitted by the AP. It is up to the AP to determine how long to operate the CFP during any given repetition interval. In such case the AP has some capabilities to determine the length of the CFP with respect to the contention period according to the weight the traffic. The process of polling is executed by the PC according to a predetermined schedule. It is common that when the PC sends a CF-Poll frame, which has the duration of SIFS to the polled station, if the polled station has no data to send, it will respond by a null function (no data) frame back to the PC. On the other hand, if the polled station has data ready for transmission, it will respond with a MSDUKF-ACK back to the PC. The

PC will send the MSDU to the destination which will respond with (+)ACK in case of correct receipt. The process of polling continues with the next station and so on.

3.4. Polling and Scheduling

As discussed, earlier, PCF can be used for multimedia transmission. Several scheduling schemes have been proposed for providing this functionality. According to the round robin scheme, as seen in Figure 3-6, which is the most popular amongst the reported schemes, the AP polls the stations sequentially in the order they are placed in the polling list. When the current polling round is over, the AP memorizes the place and in the next round starts polling from where it stopped.

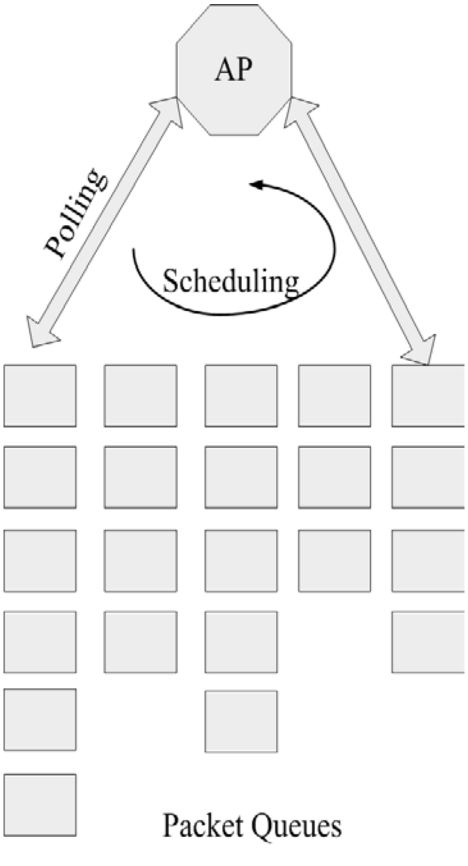


Figure 3-6 Round-Robin Scheduling of a Polling List

Scheduling can be termed as a process that allocates resources at a specified interval. The types of scheduling include opportunistic scheduling, weighted round-robin, and distributed scheduling, fair scheduling. Service differentiation may represent the general idea of the QoS provision in network resource management. The tuning of MAC parameters e.g. IFS and CW makes it a MAC specific solution to QoS provision. Applicable to both centralized and distributed mode of control, service differentiation has to be the fundamental tool for QoS provision. Scheduling is well accustomed method in centralised architecture and its algorithms will consider fairness, so as not to starve less priority traffics. Admission control, gives a preventive measure of QoS provision, where discrimination of traffic is done at the network edges. In terms of importance, service differentiation topped the three. Being a relatively widely applicable MAC-based manipulation technique, service differentiation could be achieved by the tuning of MAC parameters with distributed and centralised control. However, there is a scope in identifying the gap in providing QoS for real time multimedia traffic with fairness.

Typical scheduling algorithms in literatures include:

Round-robin – This is the simplest algorithm, implemented in most hardware and simulators (including OPNET). Starting with the first queue to have a packet in, this algorithm simply iterates through the packet queues one by one. This means that all queues will get even access to the physical layer.

Priority queuing – Each queue is assigned a priority. The algorithm simply chooses the queue that is non-empty and has the highest priority. This is similar to the 802.11e QoS differentiation in that, certain queues (as referred to therein; Access Categories) will always get access to the physical layer if it is needed.

Weighted fair queuing –This queuing discipline belongs to the family of Fair Queuing (FQ) algorithms. This algorithm is a combination between Priority queuing and Round-robin. Every queue gets a turn as in Round-robin, but each queue’s turn is weighted in time, meaning the higher priority queues get more time to transmit in their turns, and the lower priority queues get smaller amounts of time. FQ was proposed by John Nagle in

1987. This queuing discipline was designed to ensure that each flow has fair access to network resources and to prevent bursty flows from consuming more than its share output bandwidth. WFQ queuing discipline was developed independently in 1989 by Lixia Zhang et. al. [76] and by Demers et. al. [77].

Deficit round robin - Deficit round robin (DRR), also deficit weighted round robin (DWRR), is a modified weighted round robin scheduling discipline. DRR was proposed by Shreedhar et. al. [78] in 1995. It can handle packets of variable size without knowing their mean size. A maximum packet size number is subtracted from the packet length, and packets that exceed that number are held back until the next visit of the scheduler.

Deficiencies in the implementations of PCF include unpredictable beacon delays resulting in significantly shorter contention-free period (CFP) and unknown transmission duration of polled stations, because of these, it is difficult for the point coordinator (PC) to predict and control the polling schedule for the remainder of the CFP. Ksentini et. al. in [22] reported that PCF lacks mechanisms to differentiate traffic types, provided no mechanisms for the stations to communicate their QoS requirements to the AP, had no management interface to control and setup CFP and the polling schedule was not tightly controlled. Despite proposals on the improvements of PCF, the industry implementers have not picked up these ideas. Interestingly, IEEE 802.11e HCCA provides a polling mechanism similar to PCF that polls stations during a contention-free period and assigns a defined-duration period during which each station can transmit. In this way, HCF hopes to overcome PCF's limitation in terms of unpredictable transmission periods.

Work in improving PCF and centralized channel access schemes are significantly less in number. PCF-based nodes performs relatively more efficiently than DCF-based node by varying CFP duration in the SF, to handle time sensitive traffic. This would provide a scope in the transmission improvement. Simulation results in [79], which will be presented in the next section, indicate that not only the size of the CFP, the size of the SF also influences WLAN performance.

Improving WLAN performance by means of improving the polling mechanisms through the reduction of polling overheads were discussed in [33, 80-82]. Choosing SF sizes in balancing the demands of the different types of traffic has been researched by Haines et. al. in [83]. In their paper a non-linear optimization theory-based approach, for deriving optimum superframe configurations with centralized control functions were considered. Another mathematical model of delay statistics for multi user scenario was presented in [84].

Theoretical representation alone, however, is insufficient in representing the vast characteristics of interactive multimedia traffic. With the aid of simulation, the authors in [83], has found that optimal SF segmentation directly determines how well different types of traffic are supported. The authors found that the total bandwidth of the polled traffic (i.e., a function of number of nodes and the bandwidth requirements of each) appeared to be the primary determining factor in selecting maximum CFP duration regardless of the duration of the superframe. Results from [83] will be replicated and further exploration will be made to improve the outcome of the scheme proposed. We see a scope for improvements by the implementation of the PCF in a mixed traffic environment. However the configuration of the SF will directly affect the ability of the system to handle mixed traffic and achieve QoS targets. There needs to be an optimum ratio between the CFP and CP. We will develop a method where the PC can deliver the QoS requirements to different types of traffic by choosing the appropriate division of the CP and CFP phases.

The model proposed by Le et. al. in [84] with exact delay statistics for multi user scenario provides an analytical framework useful for cross-layer analysis, design, and optimization of wireless systems. Exact queue length and delay distributions are derived. Admission control policy was setup in their model under statistical delay requirement. Service differentiation was obtained using weighted round-robin scheduling in the MAC and ARQ-based error control in PHY layer. The system model was assumed to have two classes of users: high priority and low priority and an error-free and instantaneous feedback channel. In the physical layer, multirate transmission by using adaptive modulation and coding (AMC) was employed where the number of transmitted packets in one time slot varies depending on the channel condition. Queuing performances for users of each class were assumed to be statistically the same; only one user from each class is engaged. A more efficient scheduling mechanism, particularly polling, will complement work of Le or any other scheme in general.

The basic concept of a polling system is to have a server poll a set of queues in a cyclic order. While prior work [85-90] on the performance modelling of polling systems yielded significant results, they were primarily targeted at computer data applications and assumed that customers arrive at queues according to a Poisson process. Increasingly, interest in polling systems is shifting from computer data applications to multimedia applications, e.g., in IEEE 802.11 wireless LANs and Bluetooth.

WLAN polling mechanisms were central issues discussed by authors in [82, 91] i.e., adaptive polling list arrangement and scheduling scheme, respectively. Both are implemented to enable IMM packet transmission. It can be gathered that, most studies on the PCF in WLAN have been focused on the scheduling scheme and the overheads. Scheduling schemes are proposed to support multimedia services. In these schemes, all traffic types are differentiated by priorities and the polling sequence is scheduled according to the priorities of the traffics. To reduce the overhead caused by the polling frames, multipolling schemes are proposed in [80]. The idea is to poll all stations in one attempt by one polling frame instead of polling one station at a time. This way, the overheads due to the polling frames can be reduced. The protocols in [92] aimed at reducing the unnecessary polling frames used for stations with no pending frames to transmit based on statistical estimation of the traffic characteristic or information reported by a station during the contention period (CP). Obviously estimating traffic characteristic would be tedious. In [93] , further performance improvement in PCF is attempted by simply removing an acknowledgment (ACK) frame. These are examples of literatures that proposed polling algorithms considering constant physical transmission rate. Nonetheless, a typical wireless channel is time-varying and most wireless networks support several different data rates in the physical layer, an efficient communication system should select the data rate according to the channel conditions. A rate-adaptive polling-based MAC protocol for WLANs was then proposed in [94]. Possible issue in multipolling including RT traffic and frame multiplexing was discussed. A contention-free multipoll frame is used to poll a group of stations in the order specified by a list of corresponding association identifiers (AID). To each of the stations, a time period is allocated for packet transmission such that two consecutive periods are separated by an SIFS period. By polling stations in a group, the polling overhead can be greatly reduced. However, one problem is that if a station cannot receive the poll frame correctly, the corresponding allocated time period will be wasted.

In reducing the polling overhead, particularly in finding ways to poll multiple stations, it was shown in [81] that for the IEEE 802.11b wireless LAN parameters and codec rate of 8.5Kbps, statistical multiplexing gain by exploiting the ON-OFF characteristics of telephony traffic were unable to be achieved much because of this large polling overhead. By decreasing the polling and protocol overheads by half (which is feasible with an optional short preamble and a header compression technique), their system can exploit the silences in telephony traffic and accommodate a greater number of voice calls even with a higher-rate codec (64Kbps). Statistics identified appear to be acceptable because the range of an 802.11b access point is small (on the order of 100m).

Some multimedia applications have problems implementing this multipolling scheme. As explained in [33], the QoS implementation in the 802.11e standard for voice applications has limitations in implementing multipolling. A mechanism is proposed to switch to DCF mode when there are no voice packets to transmit in during PCF mode.

The polling-based MAC protocol in [94], was a two-step multipolling (TS-MP), to support real time applications in a centralized WLAN. Two multipolling frames with different purposes were used. The first multipolling frame was sent to collect information such as the number of pending frames at each station and the physical transmission rate of each communication link. Based on such information, the PC schedules a polling sequence for data transmission and the sequence was then broadcasted in the second multipolling frame. The protocol implemented rate adaptation for a polling-based MAC protocol.

Lost polls can occur due to stations not receiving polling frames correctly. To address issue of lost poll frames, the protocols proposed in [95] introduces a chaining concept; that is, to each uplink packet, a copy of the multipolling frame for the remaining stations is attached. Although the replication of the poll frame can certainly increase the chance of successful reception of the poll frame, this is done at the expense of increased overhead in terms of bandwidth usage.

On top of these polling schemes discussed, configuration of the superframe in balancing the demands of the different types traffic has been researched by Haines et. al. in [96] . In their research a non-linear optimization theory-based approach for deriving optimum configurations with centralized control functions were considered. Optimization algorithm, using the interior point optimization method to form the utility function and its constraints, were explored in detail. We will adapt this optimization method in our work to obtain a dynamic optimum reference for the AP to refer in finding a suitable CP and CFP segment.

Haines scheme in [83] found that the time-based organizational SF structure on the medium, allocating part of the SF to polling traffic and part to contending traffic were useful in improving IMM transmission. The authors found that this allocation directly determines how well different types of traffic are supported. Given the vital role of this allocation in the success of a system, authors believe that researchers must have confidence in the configuration used, beyond that provided by empirical simulation results. Their model configured and assessed the performance of different super-frame configurations and the effects of different traffic patterns. Their results will be simulated for comparison in our preliminary findings.

In order to support varying application requirements over dynamic nature of WLANs, numerous researches are not only done on the MAC layer but other layers of the open system interconnection (OSI). For example link adaptation in the PHY layer [97, 98] . These approaches focus on different network layers separately. However, they are in fact interrelated, as providing QoS guarantees only by differentiating flows and coordinating the order of channel access in the MAC layer cannot be effective under high traffic loads without admission control and resource allocation in the PHY layer. Cross-layer is the new design paradigm in wireless communications research. The redefinition of the overall design strategies breaks the classical OSI model. Though still scarce, literature on cross-layer-related issues has recently shown potential obtainable gains that deserve the increasing attention. Most literatures appear in magazines and conferences, as opposed to transactions and journals. This is because, the concept is relatively new and most ideas are still undergoing tests and modelling.

The authors in [99, 100] describes a framework for further enhancements of the traditional IP-based protocol stack to meet current and future requirements. The problems associated with the strictly layered protocol architectures were summarized and classified. One possible model for inter-layer coordination consists in a set of modules (protocols) connected to a central inter-layer coordination manager. The modules expose events and state variable to the manager. Events are notifications sent to the manager, such as "handover begins" or "link lost". A simple framework for studying and solving problems faced by all-IP wireless mobile terminals, with cross-layer design was presented. This framework first classifies known problems in four coordination planes: security, QoS, mobility, and wireless link. The author pointed that QoS problems affect flows with QoS requirements, and are caused by lack of information from transport layer congestion control and link layer ARQ. Wireless problems are caused by packet corruption and losses that are perceived by TCP as congestion indications, causing it to have poor performance.

Justification to introduce the cross-layer design methodology into the IP-based next generation wireless systems was presented in [101]. This research is the author's initial effort targeted to the problems of cross-layer signalling - a key enabler to achieve cross-layer design; comprehensive cross-layer signalling scheme - still missing; and standardisation and evaluation criteria. The design framework of a new method was proposed. Survey of existing cross-layer signalling methods was conducted, namely, Interlayer signalling pipe, "punch holes in the protocol stack", wireless channel information server and Lower-layer / higher-layer profile. The first two store cross-layer information inside the memory while the last two store cross-layer information outside of the memory. Drawbacks of the existing methods are signalling propagation paths across the protocol stack are not efficient and signalling message formats are either not flexible enough for active signalling in both upward and downward directions, or not optimised for different signalling inside and outside the mobile host respectively. Proposed framework include identifying the layer-specific contributions to this task from each layer; working out the cross-layer contribution from each layer; and identifying how the layers interact with each other to accomplish the task. The proposed scheme, however, has high complexity. Authors in [102] has developed a cross-layer scheduling algorithm at the MAC layer for multiple connections with diverse QoS requirements, which can be used in cellular networks, mobile ad hoc networks, and wireless sensor networks. Methodology used was via admission control, where traffic was assigned a priority, which is updated dynamically depending on its channel quality, QoS satisfaction, and service priority; thus, the connection with the highest priority is scheduled first each time. Their proposed scheduler offers prescribed delay, and rate guarantees for real time and non real time traffic; at the same time, it uses the wireless bandwidth efficiently by exploiting multi-user diversity among connections with different kinds of services. The fairness issue, however, was ignored by the authors.

Most protocol design methodologies currently in use are inadequate [103], either because they do not rely upon formal techniques and therefore do not guarantee correctness, or because they do not provide sufficient support for performance analysis and design exploration and therefore often lead to sub-optimal implementations. The authors [103] used a refinement-based formal methodology that relies upon the orthogonalization of function and architecture design, and; emphasizes the use of formal models to ensure correctness and reduce design time. OPNET is chosen as our simulation tool for its correctness and good standing in modelling network components.

3.5. Chapter summary

In summary gives a comprehensive examination of current state of the art technology for WLAN improvements for mixed traffic environment, in the form of a literature review. Evaluations of various MAC improvement in WLAN are an ongoing challenge. The task is even more crucial with the presence of IMM traffic. The mechanisms proposed in 802.11e acknowledge the need to cope with QoS sensitive IMM by means of categorizing traffic types. Treating IMM like non-IMM traffic (i.e., in DCF) is not the best way to go. Researchers have looked into improving polling and scheduling to handle IMM traffic. HCCA is one means towards this. Issues in HCCA require further work before it can be adopted by the industry. Another lucrative method that researchers are currently working is improving the legacy 802.11 PCF. Published literature reviewed in this chapter show a scope by optimally segmenting the SF into CFP and CP. More efficient polling and cross-layer strategies are also candidate methodology to achieve our objectives. Preliminary simulation results are presented in the following chapter to justify this scope.

Chapter 4
Modelling and Simulation of the PCF

4.1. Fundamentals of PCF and Polling

The superframe (SF) represents an access to a wireless medium in a time axis [104]. The SF is divided between a fixed time phase called contention period (CP) and contention-free period (CFP). During CP and CFP, the distributed coordination function (DCF) and point coordination function (PCF) will respectively manage available bandwidth for application transmission. DCF uses a contention based Carrier Sense Multiple Access/Collision Avoidance (CDMA/CA) protocol while PCF uses a polling list to transmit data. A typical SF duration is predetermined by the access point (AP), normally half for contending and half for polled traffic. One SF contains, as in Figure 4-1, beacon (B), CFP for polled interactive multimedia (IMM) traffic and CP for contending non-IMM traffic. SF repeats itself throughout the transmission. The CFP will alternate with a CP. The SF length is determined according to the number of beacon frames, when workstations (STAs) are initialized and when a basic service set (BSS) is produced. Thus the SF length could not be varied until the BSS is terminated. Since CFP is a fraction of the SF, its length is limited to the length of the SF throughout the BSS association.

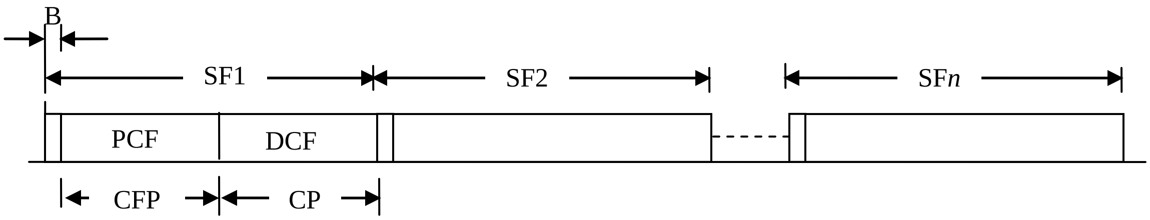


Figure 4-1 Typical Superframe Configuration

PCF controls transmission during CFP. The CFPs will occur at a defined repetition rate, which synchronises with the beacon interval. This rate is often called CFPRate. The maximum CFP duration is defined by CFPMax while minimum duration of CFP by CFPMin, as defined in the CF Parameter set. The AP could terminate any CFP at or before the CFPMax, based on available traffic and size of the polling list.

Transmissions under PCF are typically frames alternately sent from and to the AP. During the CFP, the ordering of these transmissions, and the STA allowed to transmit to the AP at any given point in time is controlled by the AP. The CFP will end when the CFPDurRemaining time has elapsed since the beacon frame originating the CFP or when the AP has neither further frames to transmit nor STAs to poll.

The value of CFPMax is limited to allow coexistence between contention and contention-free traffic. The minimum value for CFPMax will allow sufficient time for the AP to send one data frame to a STA, while polling that STA, and for the polled STA to respond with one data frame. The maximum value for CFPMax will allow sufficient time to send at least one data frame during the CP.

The AP maintains a polling list for use in selecting STAs that are eligible to receive CF-Polls during CFPs. The polling list is used to force the polling of CF-pollable STAs, whether or not the PC has pending traffic to transit to those STAs. The AP (in which PC resides) will send a CF-Poll to at least one STA during each CFP when there are entries in the polling list. During each CFP, the AP shall issue polls to a subset of the STAs on the polling list. While time remains in the CFP, the delivery of all of CF frames has been completed and all STAs on the polling list have been polled, the PC may generate one or more CF-Polls to any STAs on the polling list. While time remains in the CFP, the delivery of all CF frames has been completed and all STAs on the polling list have been polled, the AP may send data or management frames to any STAs.

Round-robin scheme is the most typical polling mechanism adopted by most APs. Stations are polled sequentially according to their placement in the polling list. When the CFP ends, the AP memorizes the location in the polling list where it stopped to resume again in the next CFP start. An example of a typical polling in round robin was illustrated previously in Figure 3-6.

The standard of IEEE 802.11 PCF does not define the scheduling scheme to poll the stations in the polling list. Most of PCF scheduling protocols implement the traditional Round Robin (RR)[105]. That is, the polling list is constructed after the stations join the WLAN networks. Then, the AP starts to poll the stations one by one.

4.2. Discrete-event Simulation

Discrete-event simulation (DES) is chosen for its suitability to model communication networks. DES models are dynamic as compare to mathematical and statistic models [106]. The representation of the components of a communication system and their interactions, occur as a consequence of activity times and delays. Entities may compete for system resources, like in this case the bandwidth of WLANs.

The advantages and disadvantages of simulation as described by [106-111] are the followings:

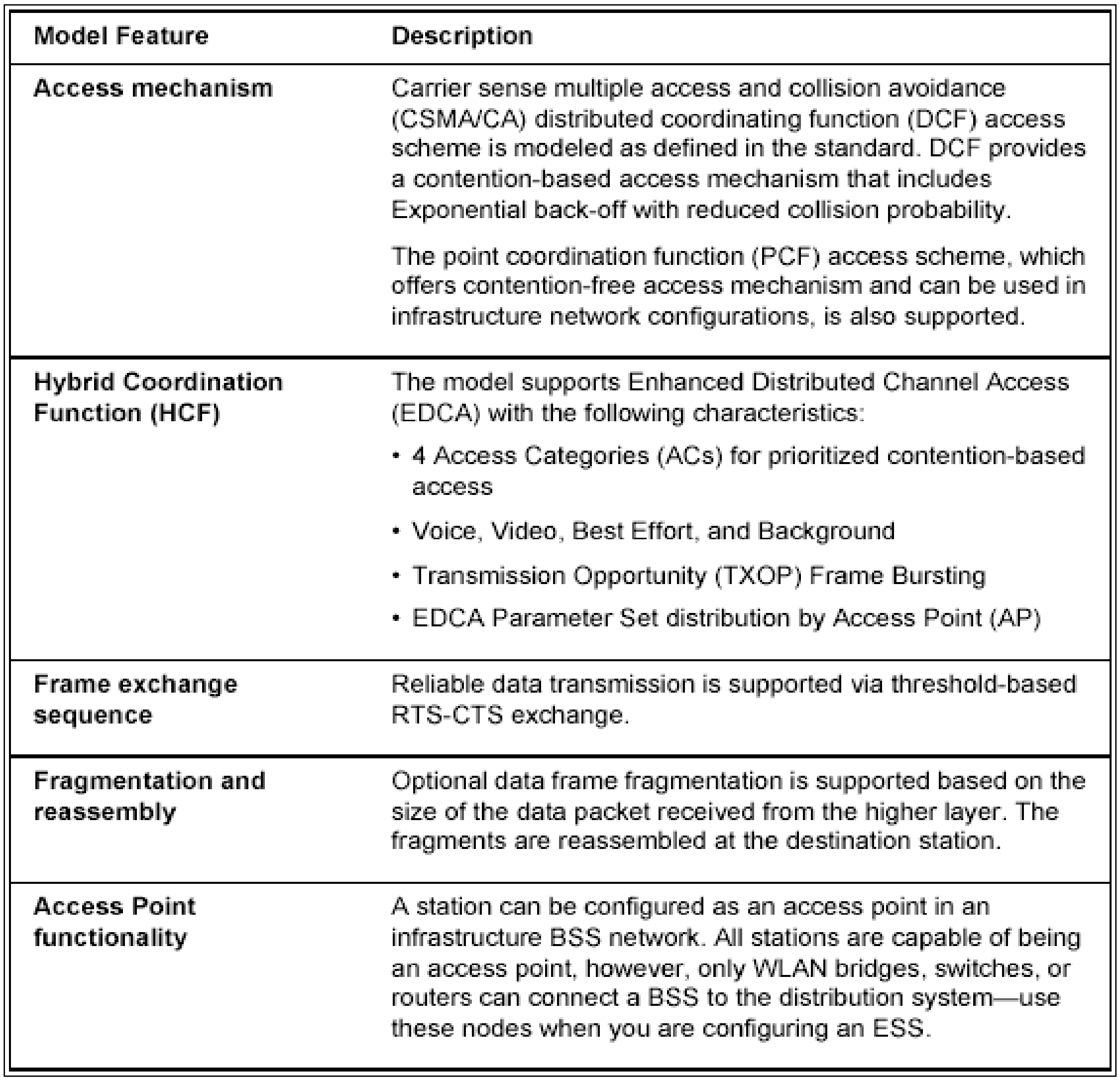
Firstly, simulations will allow the examination of a lengthy event in a matter of minutes. A 15 hours event can be simulated, via time compression, in 5 minutes. Time saves critical resources on the researcher part. Secondly, it allows the test of proposed change without committing physical resources to their attainment. If a real life system has been installed, changes to it will be extremely expensive. One of the main advantages of simulation is, once a valid simulation model is developed, exploration of new policies and protocols can be done without disrupting the real system. Diagnosis of a complex system can be done relatively easily. Constraints can be more easily identified and dealt with. Improvements could be set in and further tests could be made as and where needed. Major disadvantages of simulation include specialized training to work the software and interpretation of simulation results may be tedious due to system interrelationships or randomness.

Model formulation is the process by which a conceptual model is envisioned to represent the system under study. The conceptual model is the model that is formulated in the mind of the modeller. Authors in [112] and the references therein states that model formulation and model representation constitute the process of model design. Model representation translates the conceptual model into communicative model, which is more graphical, either in structured English or flowcharts. Translation of the communicative model into programmed model is termed as the process of programming. A programmed model is an executable simulation model representation which does not incorporate an experiment design. The process of programming can be performed by the modeller using a simulation software [113, 114]. In our case, the simulation software is OPNET.

4.3. The OPNET Simulation Software

OPNET is expansively used in developing the simulation models for performance studies of the developed MAC protocols. The idea and concept for improvements of the MAC performance has been integrated into OPNET with the required modifications and fine tuning. Much time is saved from having to design scenarios, realistic WLAN nodes, traffic conditions, traffic types etc., as these are provided for and built-in into OPNET. OPNET models communications devices, protocols, technologies, and architectures, and simulate their performance in a dynamic virtual network environment using event-driven technique. Integrated code debugging (in C/C++) and data analysis features have been very helpful in facilitating the design process.

The descriptions, tables and figures in this and the immediate next two Chapters are obtained from OPNET WLAN model suite [115] which includes models to analyze and implement discrete event simulation for the IEEE 802.11 standards including 802.11e standards. The 802.11e will be one of the benchmarks for our performance comparison. The protocol feature is shown in Figure 4-2.



Model Feature	Description
Access mechanism	Carrier sense multiple access and collision avoidance (CSMA/CA) distributed coordinating function (DCF) access scheme is modeled as defined in the standard. DCF provides a contention-based access mechanism that includes Exponential back-off with reduced collision probability. The point coordination function (PCF) access scheme, which offers contention-free access mechanism and can be used in infrastructure network configurations, is also supported.
Hybrid Coordination Function (HCF)	The model supports Enhanced Distributed Channel Access (EDCA) with the following characteristics: • 4 Access Categories (ACs) for prioritized contention-based access • Voice, Video, Best Effort, and Background • Transmission Opportunity (TXOP) Frame Bursting • EDCA Parameter Set distribution by Access Point (AP)
Frame exchange sequence	Reliable data transmission is supported via threshold-based RTS-CTS exchange.
Fragmentation and reassembly	Optional data frame fragmentation is supported based on the size of the data packet received from the higher layer. The fragments are reassembled at the destination station.
Access Point functionality	A station can be configured as an access point in an infrastructure BSS network. All stations are capable of being an access point, however, only WLAN bridges, switches, or routers can connect a BSS to the distribution system—use these nodes when you are configuring an ESS.

Figure 4-2 OPNET WLAN Protocol Features [116]

We will be utilizing the *wireless terminal station*, which has the WLAN MAC—this node is the most suitable for studies that focus only on MAC and physical layers. The WLAN modelling of the MAC and the physical layer is comprised of the *wireless_lan_mac* process as shown in Figure 4-3. The ARP (address resolution protocol) is an interface between the MAC and the higher layers. To study the WLAN MAC without a higher-layer stack (such as TCP/IP and applications), use the station node model that uses source and sink models to simulate higher layers. This node model will be used to generate controlled traffic in the WLAN network and evaluate the performance of the MAC, to simulate the effect of WLAN attributes independent of the higher layer and to obtain shorter simulation time for large networks.

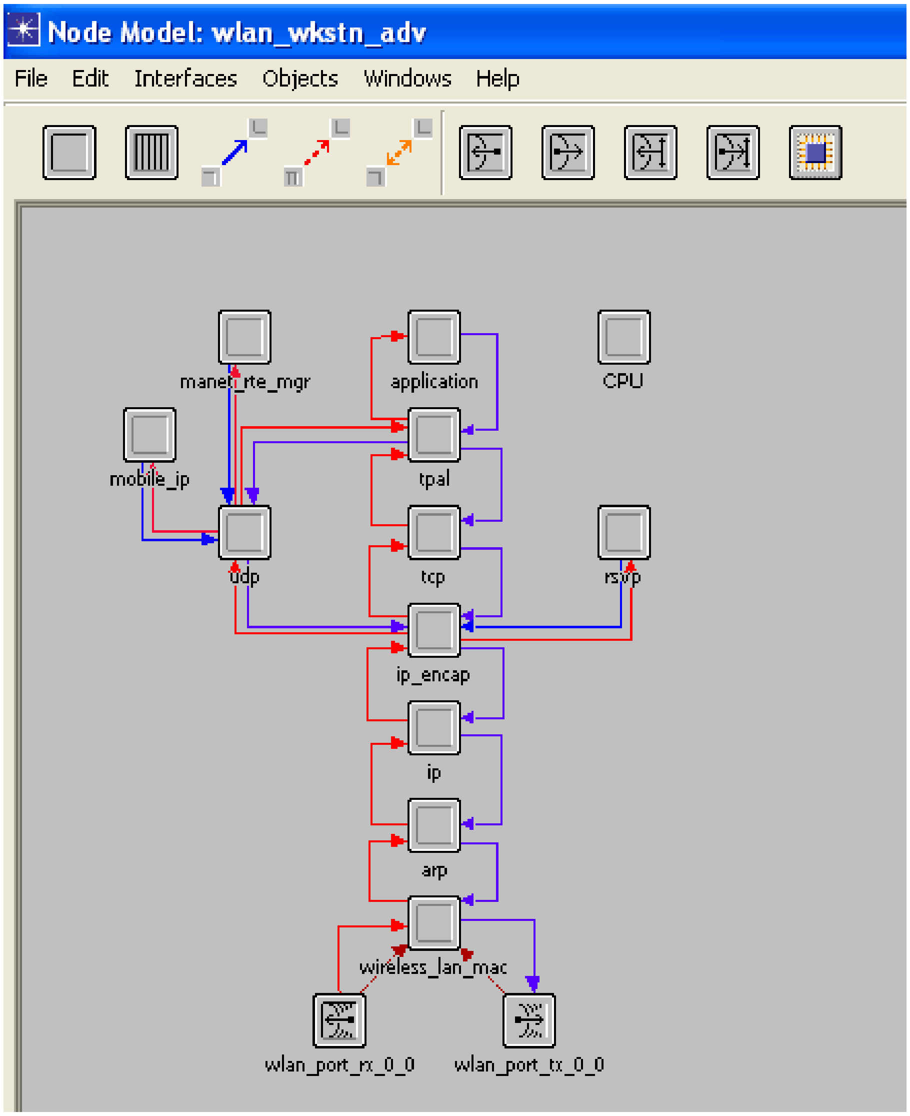


Figure 4-3 OPNET WLAN Node Module

This protocol feature supports the infrastructure BSS network configuration where nodes communicate with each other through the AP. An example is shown in Figure 4-4.

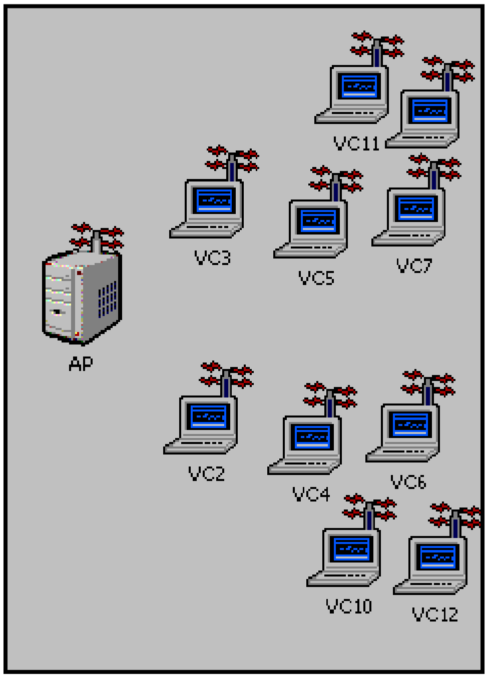
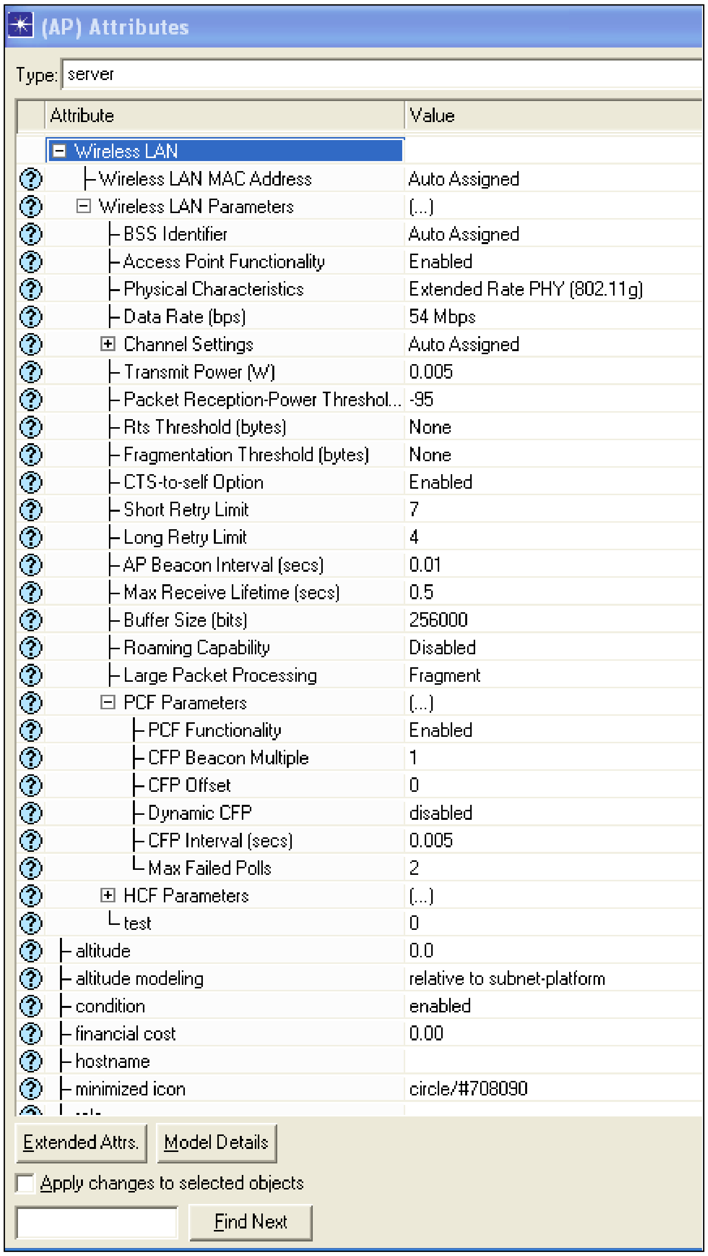


Figure 4-4 OPNET WLAN Model for Infrastructured Network

4.4. PCF and Polling Mechanisms in OPNET

In our experimental set-up where both schemes are used, CFP and CP would alternate. Within a CFP repetition interval or CFPri also known as SF, a portion of the time is allocated as CFP during which the PCF is active and the remaining time is allocated as CP. CFPs are generated at a defined rate and each CFP begins with a beacon frame. The PC, in our case the AP, will determine this rate. This rate, also known as the CFPRate, is defined as Delivery Traffic Indication Message (DTIM) intervals. These parameters are set in each of the node attributes including the AP. An example of the parameters setting is shown in Figure 4-5. The parameters that would concern us most would be *AP Beacon Interval* in seconds (secs) which is the SF duration, *PCF Parameters* particularly the *CFP Interval (secs)* which is the CFP duration. By default SF and CFP durations are static i.e., unchanged throughout the entire simulation period. However, we have added another parameter in this *wlan_mac* process model i.e., the *Dynamic_CFP parameter* we can compare the effects of static and dynamic superframe configuration. Appendix B shows relevant process models used to implement dynamic SF configuration in OPNET.



Type: server

Attribute	Value
⊟ Wireless LAN	
Wireless LAN MAC Address	Auto Assigned
⊟ Wireless LAN Parameters	(...)
BSS Identifier	Auto Assigned
Access Point Functionality	Enabled
Physical Characteristics	Extended Rate PHY (802.11g)
Data Rate (bps)	54 Mbps
⊞ Channel Settings	Auto Assigned
Transmit Power (W)	0.005
Packet Reception-Power Threshol...	-95
Rts Threshold (bytes)	None
Fragmentation Threshold (bytes)	None
CTS-to-self Option	Enabled
Short Retry Limit	7
Long Retry Limit	4
AP Beacon Interval (secs)	0.01
Max Receive Lifetime (secs)	0.5
Buffer Size (bits)	256000
Roaming Capability	Disabled
Large Packet Processing	Fragment
⊟ PCF Parameters	(...)
PCF Functionality	Enabled
CFP Beacon Multiple	1
CFP Offset	0
Dynamic CFP	disabled
CFP Interval (secs)	0.005
Max Failed Polls	2
⊞ HCF Parameters	(...)
test	0
altitude	0.0
altitude modeling	relative to subnet-platform
condition	enabled
financial cost	0.00
hostname	
minimized icon	circle/#708090

Figure 4-5 OPNET Access Point (AP) Attributes

The *wlan_mac* process model, as shown in Figure 4-6, in OPNET will be modified to simulate our proposed scheme. Appendix A shows the relevant functions in *wlan_mac*.

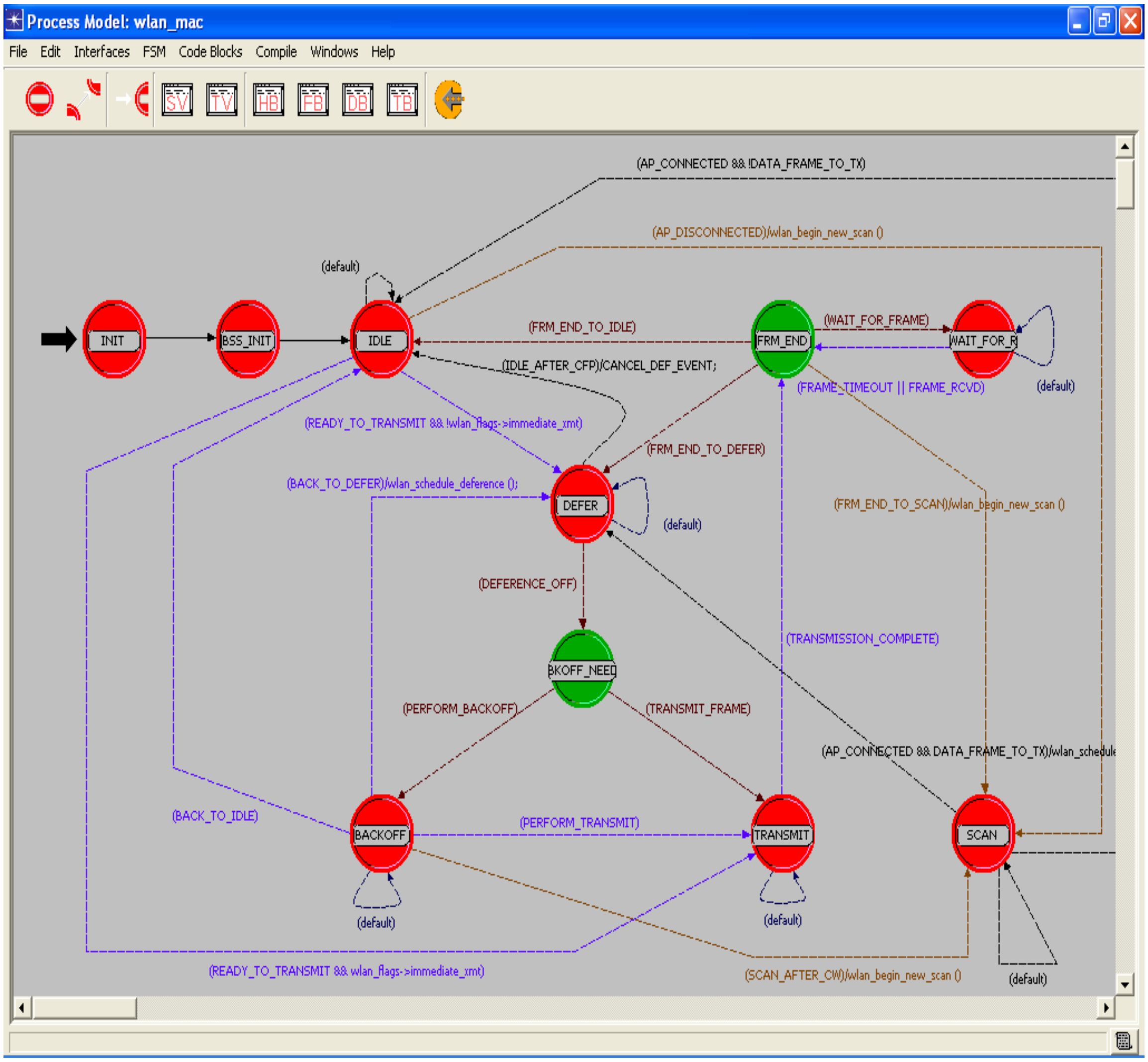


Figure 4-6 OPNET *wlan_mac* Process Model

As depicted in Figure 4-7, the station sends an associate request to join the polling list of AP. Once the AP forms the polling list, then it sends a beacon frame with CFP period and beacon interval to all stations in the base station (BSS). The AP sends CF-Poll frame to the first station on the list. If there is no response from that station within PIFS timeframe, then the AP switches to the next station, and continues until it reaches the end of the list. If there is traffic after the PIFS time frame, then the AP polls to next station after the completion of data transfer. A station receives a poll from the AP, then the station transfers the data to the destination if there is one or idle. If a station receives CF-

End from AP, then it sets the NAV = 0 and waits for a poll from AP. Selection of AP, beacon interval, CFP period, and frame size are configured in *attribute file* as previously shown in Figure 4-5.

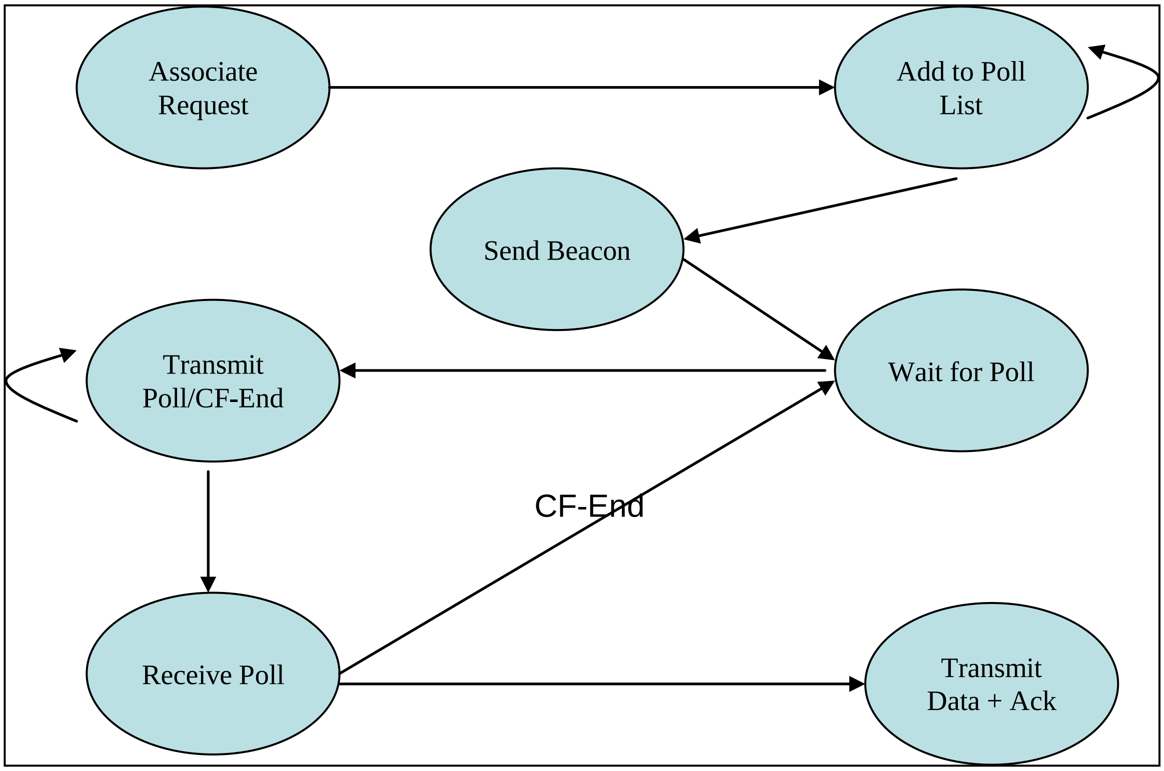
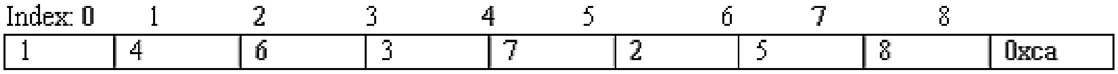


Figure 4-7 OPNET Polling Process Diagram

In OPNET the AP uses the round robin scheduling scheme during polling. AP maintains an array of number of nodes in the BSS. If a node associates with the AP, then the AP finds an empty space in its list and fills the station id on the index. The end of the list is identified by “0xca”, as shown in Figure 4-8. AP polls a station in CFP period from the beginning of the list. If the AP reaches the end of the polling list, then it starts from the beginning of the list to poll, and continues this operation until it reaches the end of the CFP period. The polling list and its content is depict in Figure 4-8.



Index:	0	1	2	3	4	5	6	7	8
	1	4	6	3	7	2	5	8	0xca

Figure 4-8 An Example of OPNET Polling List

The performance of the round-robin algorithm depends heavily on the size of the time-slice. If this quantum is very large, the round-robin algorithm is similar to the first-come, first-served algorithm. If the quantum is very small, the round-robin approach is called *processor sharing*. Both extremes give adverse impact on the QoS of IMM traffic. An optimum time allocation is essential for efficient IMM transmission.

The queuing of the packets in the AP is based on the node type (DCF/PCF) of the destination. If the destination node is a PCF node, then the packets are queued into the *cfpd_list_ptr* and transmitted only during the PCF period. Likewise packets for a DCF destination node will be inserted into the *hld_list_ptr* and transmitted only during DCF. Packets with broadcast address are transmitted during DCF. This is shown in a snippet of the polling association in Figure 4-9. The packets are inserted into the queue and sorted in the order of the MAC addresses.

```
/* Check for PCF terminal and also if this station has been polled.   */
if (polling == OPC_TRUE)
        {
        /* Insert the packet sorted in the order of the MAC addresses.        */
        op_prg_list_insert_sorted (cfpd_list_ptr, hld_ptr, wlan_hld_list_elem_add_comp);
        }
else
        {
        /* Insert a packet to the list.*/
        op_prg_list_insert (hld_list_ptr, hld_ptr, OPC_LISTPOS_TAIL);

        /* Enable the flag indicating that there is a data frame to           */
        /* transmit.                                                          */
        wlan_flags->data_frame_to_send = OPC_TRUE;
        }
```

Figure 4-9 An Example of OPNET Polling Code

If a poll fail count reached the max poll fails count or the previous poll was successful and no more data from this station and last data transmission was successful and no more fragments exist and no more data exist in the *hlk queue* for this station then next station will start transmission. If we finished polling all the pollable stations in the list but still have some contention free frames to send, then restart polling the pollable stations since we still have some CFP time to go. We will end the CFP prematurely since we have no stations to poll and no CF frames to send. We will also send an ACK if necessary.

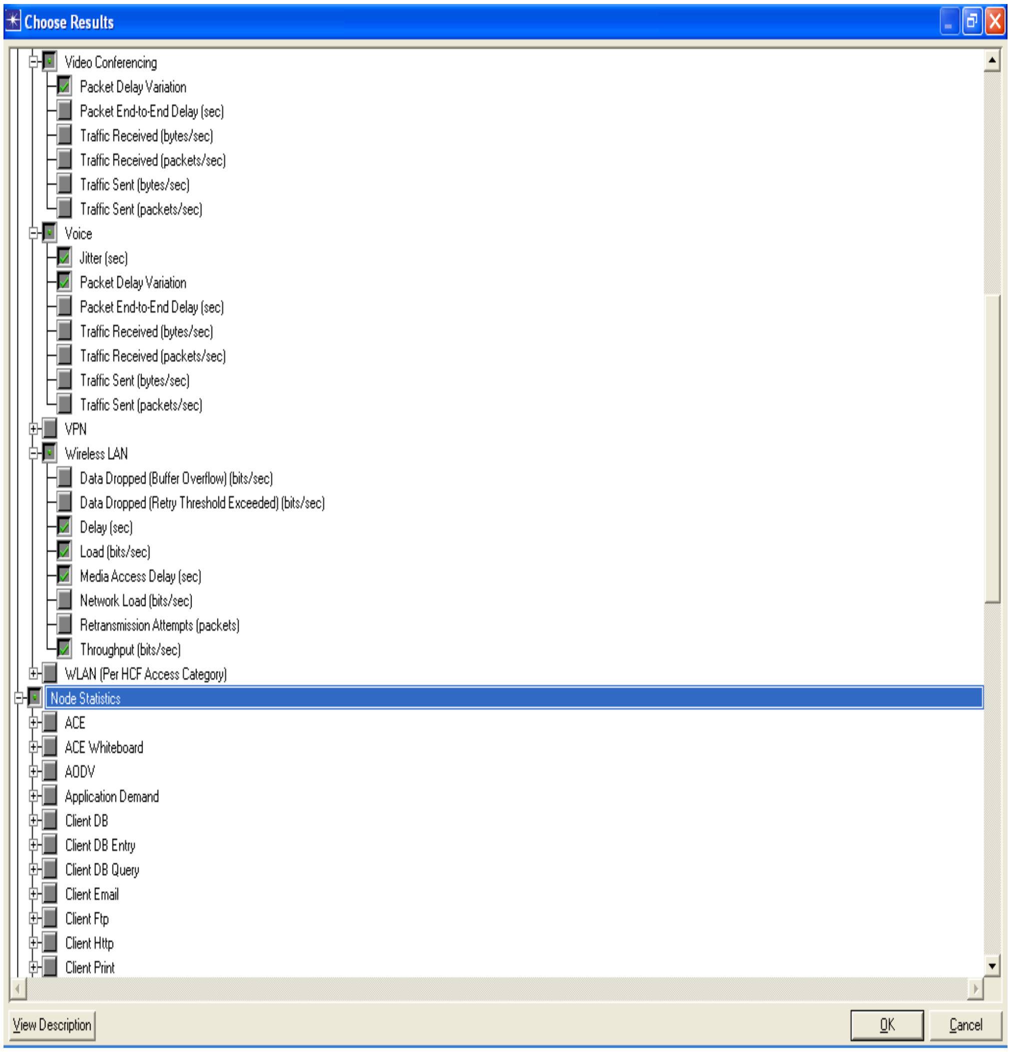
If we have finished polling all the pollable stations in the list but still have some contention free frames to send, then restart polling the pollable stations since we still have some CFP time to go.

4.5. Performance Measurement Metrics

The prevailing components of network design for supporting interactive multimedia could be classified into topological, capacity, signalling and QoS design.

In topological design, the core network topology, soft-switch and router placement, and backhaul are mostly considered. Topologies are designed for reliability and scalability. The capability of network elements like soft-switches, routers, facilities and servers to support voice traffic and signalling are taken into consideration in capacity design. Signalling network design looks into interconnecting voice end points independent of their access arrangements and corresponding signalling protocols. Nonetheless, QoS design technique will be used in this research, where end-to-end quality of service objectives for latency, jitter, and throughput. A particular emphasis will be made on queuing/scheduling strategy in the MAC scheme.

The WLAN performance metrics that we are most concerned of are namely end-to-end delay, jitter and throughput. We are able to collect the statistics for these metrics from the DES Statistics menu as shown in Figure 4-10.



- ☐ Video Conferencing
  - ☑ Packet Delay Variation
  - ☐ Packet End-to-End Delay (sec)
  - ☐ Traffic Received (bytes/sec)
  - ☐ Traffic Received (packets/sec)
  - ☐ Traffic Sent (bytes/sec)
  - ☐ Traffic Sent (packets/sec)
- ☐ Voice
  - ☑ Jitter (sec)
  - ☑ Packet Delay Variation
  - ☐ Packet End-to-End Delay (sec)
  - ☐ Traffic Received (bytes/sec)
  - ☐ Traffic Received (packets/sec)
  - ☐ Traffic Sent (bytes/sec)
  - ☐ Traffic Sent (packets/sec)
- ☐ VPN
- ☐ Wireless LAN
  - ☐ Data Dropped (Buffer Overflow) (bits/sec)
  - ☐ Data Dropped (Retry Threshold Exceeded) (bits/sec)
  - ☑ Delay (sec)
  - ☑ Load (bits/sec)
  - ☑ Media Access Delay (sec)
  - ☐ Network Load (bits/sec)
  - ☐ Retransmission Attempts (packets)
  - ☑ Throughput (bits/sec)
- ☐ WLAN (Per HCF Access Category)

- ☐ Node Statistics
  - ☐ ACE
  - ☐ ACE Whiteboard
  - ☐ AODV
  - ☐ Application Demand
  - ☐ Client DB
  - ☐ Client DB Entry
  - ☐ Client DB Query
  - ☐ Client Email
  - ☐ Client Ftp
  - ☐ Client Http
  - ☐ Client Print

Figure 4-10 OPNET DES Statistics

End-to-end delay is defined as the time (in milliseconds) taken to transmit a packet from the time it is passed to the TCP layer until it is successfully received at the destination (TCP layer). This is the delay observed from the application's perspective. In other words, it represents the end to end delay of all the packets received by the wireless LAN MACs of all WLAN nodes in the network and forwarded to the higher layer. This delay includes medium access delay at the source MAC, reception of all the fragments individually, and transfers of the frames via AP, when access point functionality is enabled.

Jitter defined by OPNET, in [115], is calculated as the following: If two consecutive packets leave the source node with time stamps t1 & t2 and are played back at the destination node at time t3 & t4, then:

$$\text{jitter} = (t4 - t3) - (t2 - t1) \qquad \text{Equation (3-1)}$$

Negative jitter indicates that the time difference between the packets at the destination node was less than that at the source node.

Throughput is the number of bits per second delivered over the medium. The measure includes both packet payload and headers. Unit of measurement is in mega bits per seconds. In OPNET, it represents the total number of bits (in bits per sec) forwarded from wireless LAN layers to higher layers in all WLAN nodes of the network.

4.6. Summary of Simulation Models

The mixed traffic scenario used in our simulations is shown in Table 4-1. An AP buffer of 256 Kbits is also simulated. In summary, this chapter explains the basic building blocks of PCF and polling as implemented in OPNET. DES facilitates the simulation of communication networks which have interactive network components. Delay, jitter and throughput are performance metrics that will be measured and analysed in our simulation results.

Table 4-1: IMM and Non-IMM nodes specifications [24].

IMM nodes		Non-IMM nodes
Video Conferencing	128 x 120 pixel frames	Best Effort
	10 frames per second	
Voice (VOIP)	G.711 (silence)	
	PCM Quality	

Chapter 5

The MAC with Dynamic Superframe Selection Scheme

5.1. Introduction

The trend of research, as seen through the review of related work and results obtained from our preliminary studies directs towards a scope of improving quality of service (QoS) in interactive multimedia (IMM) transmission in wireless local area network (WLAN) via the point coordination function (PCF).

Current and future WLAN traffic are and will be dominated by a wide range of traffic requiring different guarantees of QoS. Streaming multimedia, for example, requires high bandwidth coupled with tight delay constraints as packets need to be delivered in a timely fashion to guarantee continuous media play back. When packets are lost or arrive late, the picture quality suffers. Voice application, however, is delay-sensitive but loss-tolerant. It has a stringent end-to-end delay requirement of 150 msec for excellent-quality voice and 400 msec for acceptable-quality voice, both with echo cancellers (Table 2-1). A packet loss ratio up to 5% is tolerable for some voice encoding schemes. As a result, each of these services imposes widely varying QoS requirements on the underlying system infrastructure i.e., the WLAN.

This chapter demonstrates the impact of dynamic superframe (SF) duration selection in improving QoS provisioning for IMM traffic by OPNET simulation results and a proposed solution. The proposed self-adaptive scheme, called medium access control (MAC) with Dynamic SF Selection (MDSS) scheme, generates an optimum SF configuration according to the QoS requirements of traversing IMM traffic. The scheme will provide optimum bandwidth for IMM traffic while giving fairness to non-IMM traffic. MDSS will be implemented in the AP as a dynamic lookup table indexed by number of IMM traffic currently in the polling list.

5.2. CFP and SF Duration Variation

The goal of the simulation is to show the effectiveness of the PCF based channel access scheme to deal with heterogeneous traffic. This is achieved by choosing the right balance between the contention period (CP) and contention-free period (CFP) of a super frame. The QoS requirements of IMM traffic is satisfied whilst ensuring that the maximum possible amount of medium time remains for non-IMM traffic. The following sections will demonstrate the performance of PCF in a mixed traffic environment and the impact of varying CF and SF duration. The system parameters for the AP used for the OPNET simulations are previously shown in Figure 4-5.

5.3. PCF vs. DCF in mixed traffic

An infrastructure WLAN, as shown in Figure 5-1, carrying mixed traffic was set up in OPNET with twenty nodes. Ten nodes were transmitting best effort data traffic and the other ten were transmitting IMM traffic.

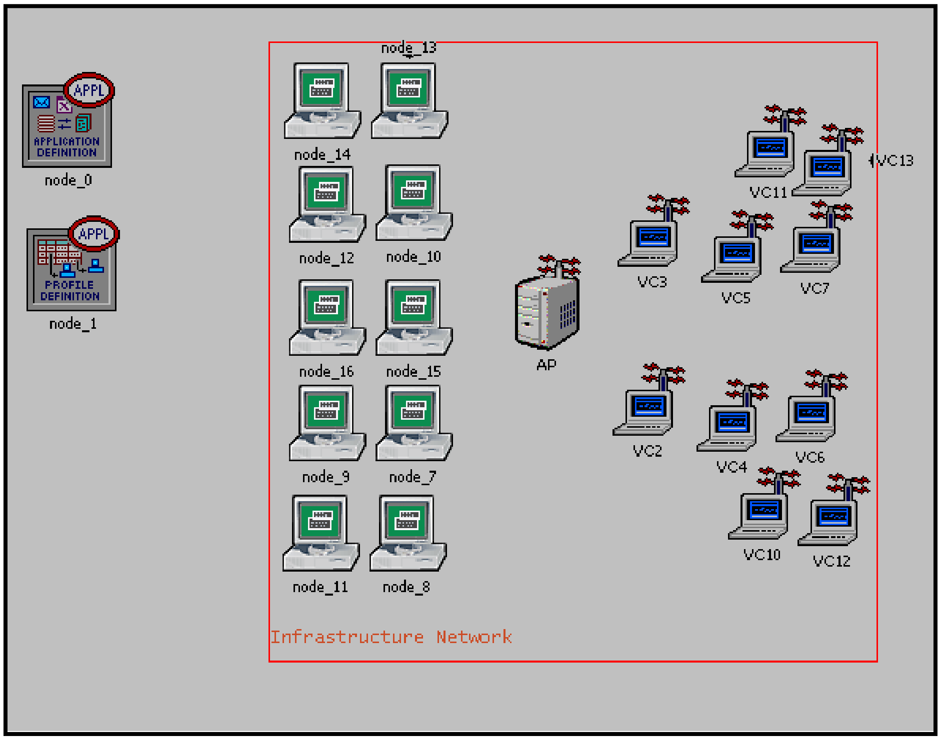


Figure 5-1 Experimental Setup of Infrastructured WLAN

In the first scenario all 20 nodes are modelled to operate under the DCF mode (i.e., PCF is not activated in the AP) and contend for resource allocation. The duration of the SF is 20msec. In the second scenario the PCF is activated in the AP. So the IMM nodes are polled during the CFP and the other 10 nodes contend for access during the CP. The polling mechanism supported by OPNET is the round robin scheme as discussed in the previous chapter. Default duration of 20msec was defined for SF and 10msec CFP duration was set in PCF activated nodes. The QoS metrics for both the scenarios, gathered from simulation runs, are shown in Figure 5-2, 5-3 and 5-4 respectively for delay, jitter and throughput.

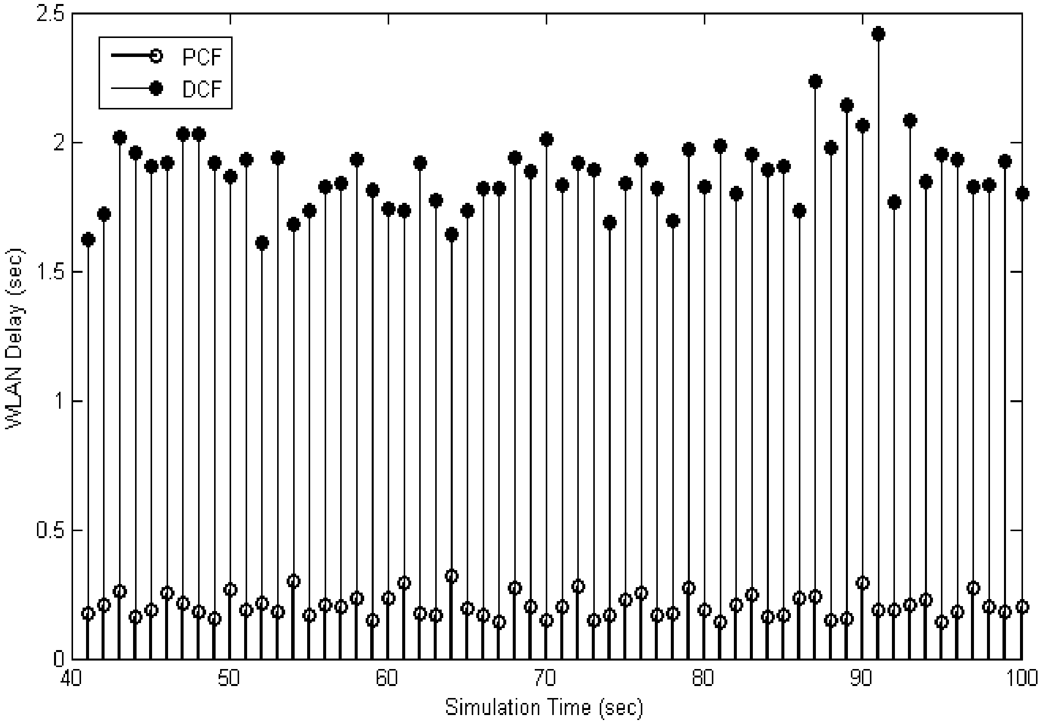


Figure 5-2 Average Packet Delay for 20 Nodes Operating Under PCF and DCF

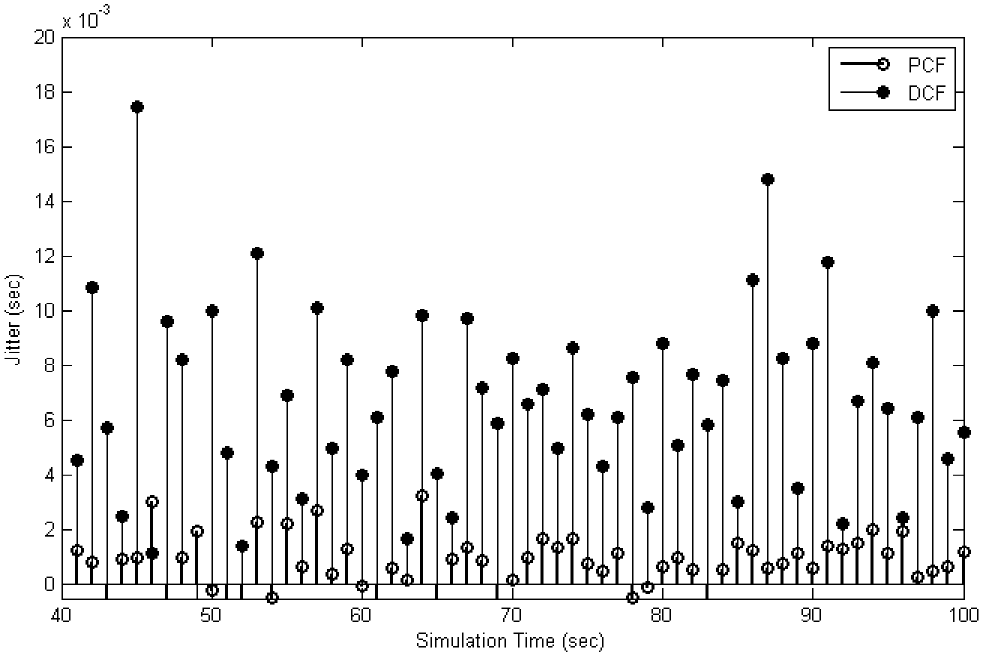


Figure 5-3 Average Jitter for 20 Nodes Operating Under PCF and DCF

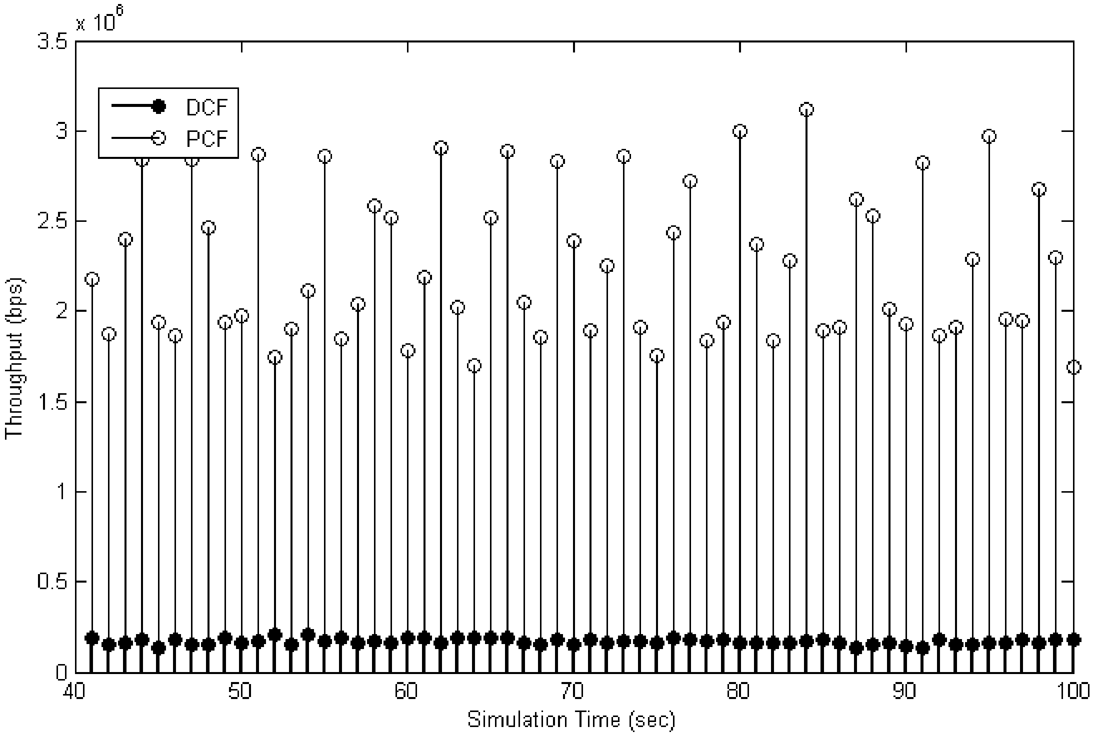


Figure 5-4 Average Throughput for 20 nodes Operating Under PCF and DCF

It may be observed from Figures 5-2 and 5-3 respectively that the end to end delay and the jitter experienced in the PCF-activated scenario is notably lower than the delay experienced by DCF-activated scenario. This is expected since polling itself results in less number of retransmissions as compared to contention. Retransmission attempts influence the delay and variation of delay (i.e., jitter). IMM traffics are very sensitive to retransmission attempts, with video conferencing requiring the least amount of it. For the PCF based scheme, the real time IMM traffic was polled during contention free period (CFP) and did not have to contend for resources. The large end to end delay and the large variation of delay observed for the DCF based scheme could be a key quality setback for most IMM applications. We can also observe from Figure 5-3 that the throughput measured in the PCF-activated scenario is higher compared DCF-activated scenario. This is due to the fact that in the latter scenario, most packets from the IMM applications were dropped due to the failure in contention. In conclusion the IMM nodes perform better in the PCF based access scheme.

5.4. Impact of the size of CFP

Since it has been established, in the previous section, that IMM traffic preferred polling to contention; this section will demonstrate further, the relationship between polling mechanism and QoS provisioning. OPNET uses the default Round Robin scheme in allocating resources to polled stations. During SF duration, the AP will have to manage a list of stations to be polled. Polled stations will be given time to utilise the resource during the CFP. After the time is up, the polled station will have to wait for its turn again either in the current SF or the next SF. Here, it can be inferred that, a CFP with a small number of polled stations will incur smaller delay compared to a CFP servicing a large number of polled stations. A large CFP however, will result to a longer waiting time for a polled station as each time slice will be longer. Hence a wrong selection of the CFP will result in longer delay and larger jitter, which will not be suitable for IMM traffic. To demonstrate this, we vary CFP with respect to a fixed SF duration and with a varying SF duration. The effects of varying IMM nodes are also simulated.

Three scenarios are setup in OPNET and their statistics analysed. These setups were representative of the possible scenarios of varying CFP, SF and number of IMM nodes. This section demonstrates the effects of dynamically varying the CFP, SF and number of IMM stations in the QoS metrics. An OPNET simulation setup with 20 nodes similar to the one employed in Figure 5.1, was modelled for this purpose.

In the first scenario, SF was fixed at the default 20 msec duration while the CFP was allowed to vary from 2 msec to 20 msec to demonstrate the effects on jitter, throughput and delay. In the second scenario, SF was allowed to vary from 4 msec to 40 msec with CFP taking up 50% of the SF duration. The final scenario simulates increasing number of IMM nodes with fixed SF and CFP at the default values of 20 msec and 10 msec respectively. Statistics with respect to simulation time is taken. The average value is also presented to analyse the trend of the QoS metrics.

5.5. Simulation Results

Results obtained from these simulations are depicted in Figures 5-5 to 5-10. Descriptions, analysis and comments are presented in each of the following subsections.

5.5.1. Varying CFP (20msec SF)

In this scenario, the SF is fixed at the default value of 20msec. CFP duration is allowed to vary from 2 msec to 20 msec in order to see the responsiveness of the performance metrics to varying CFP sizes. Figures from 5-5 to 5-7 show the resulting delay, jitter and throughput respectively from one of the simulation instances. While Figures 5-8 to 5-10 show the corresponding average delay, jitter and throughput with respect to CFP duration.

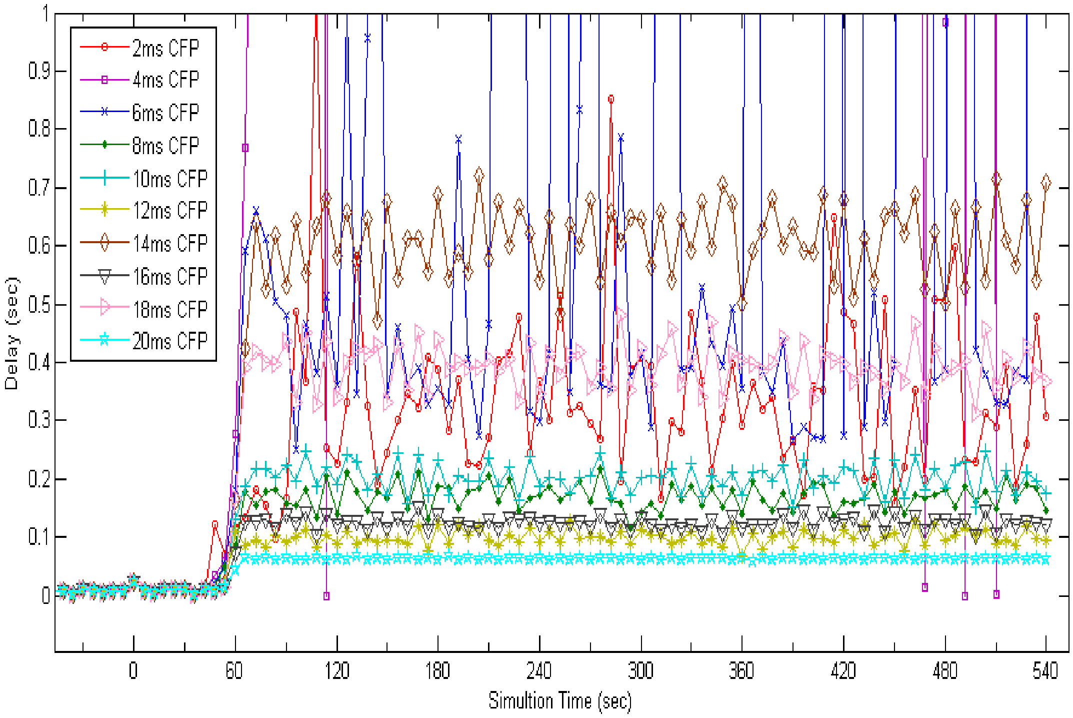


Figure 5-5 Delay as a Result of Varying CFP

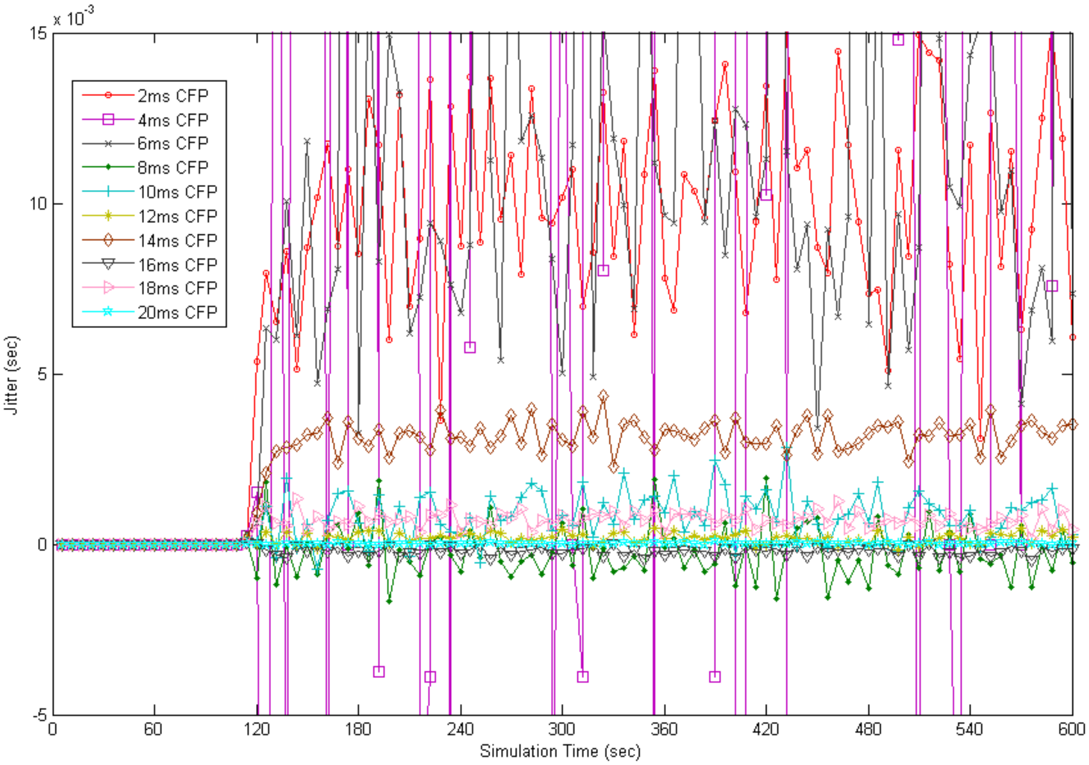


Figure 5-6 Jitter as a Result of Varying CFP

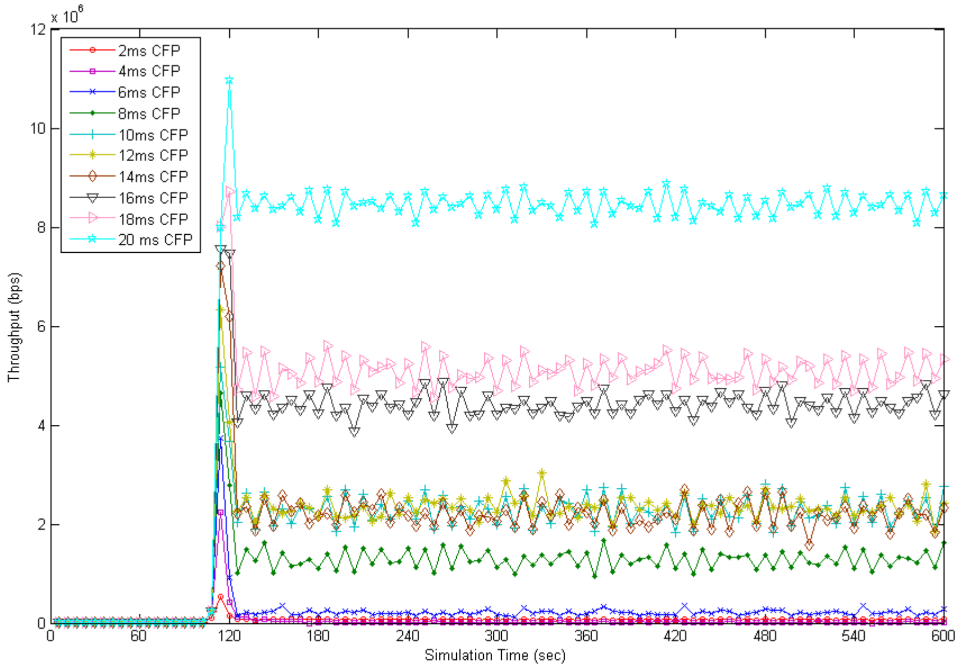


Figure 5-7 Throughput as a Result of Varying CFP

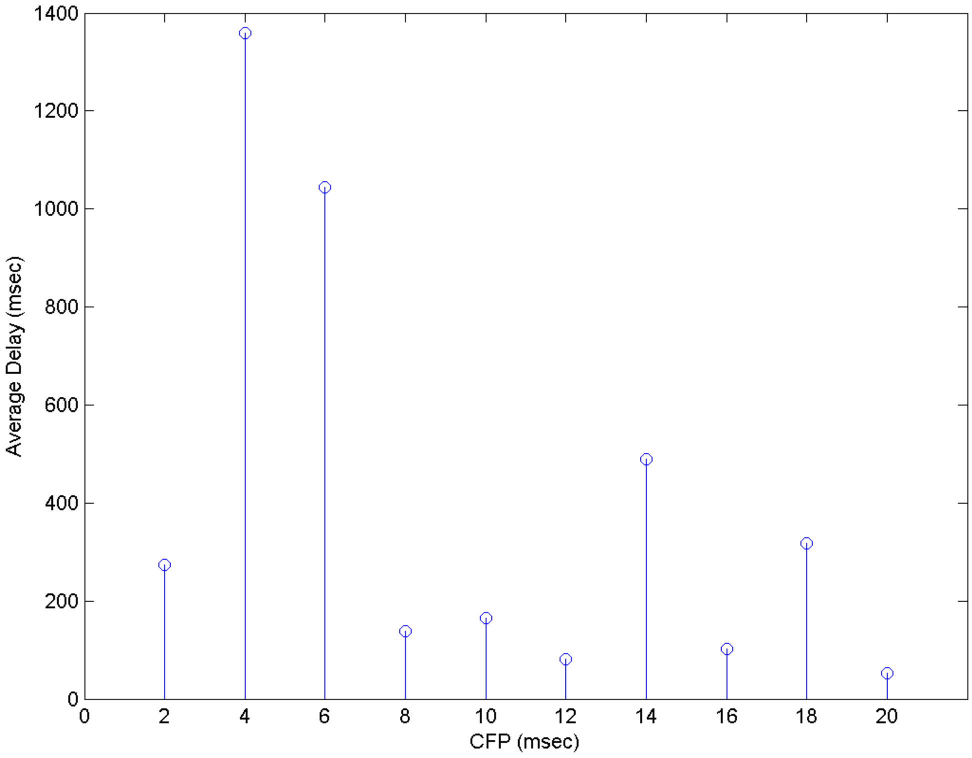


Figure 5-8 Average Delay as a Result of Varying CFP

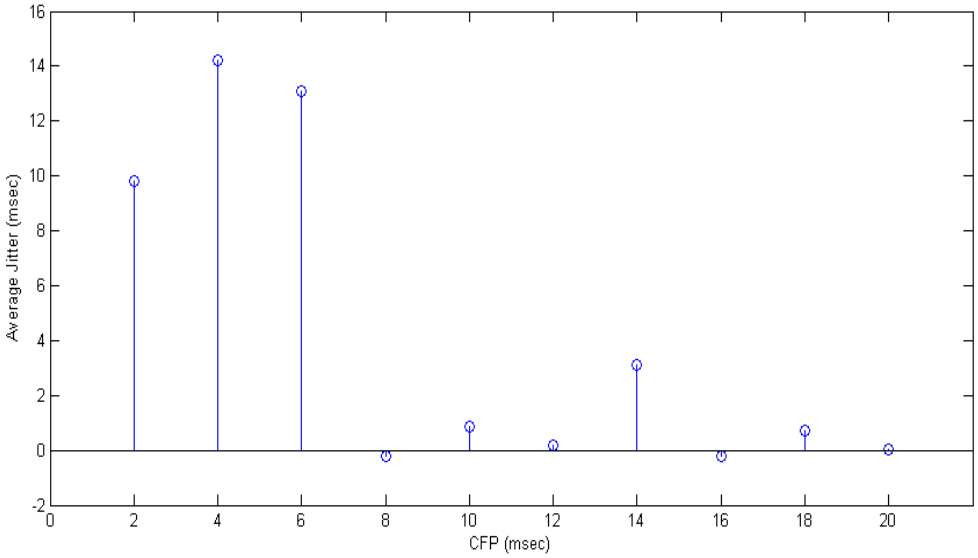


Figure 5-9 Average Jitter as a Result of Varying CFP

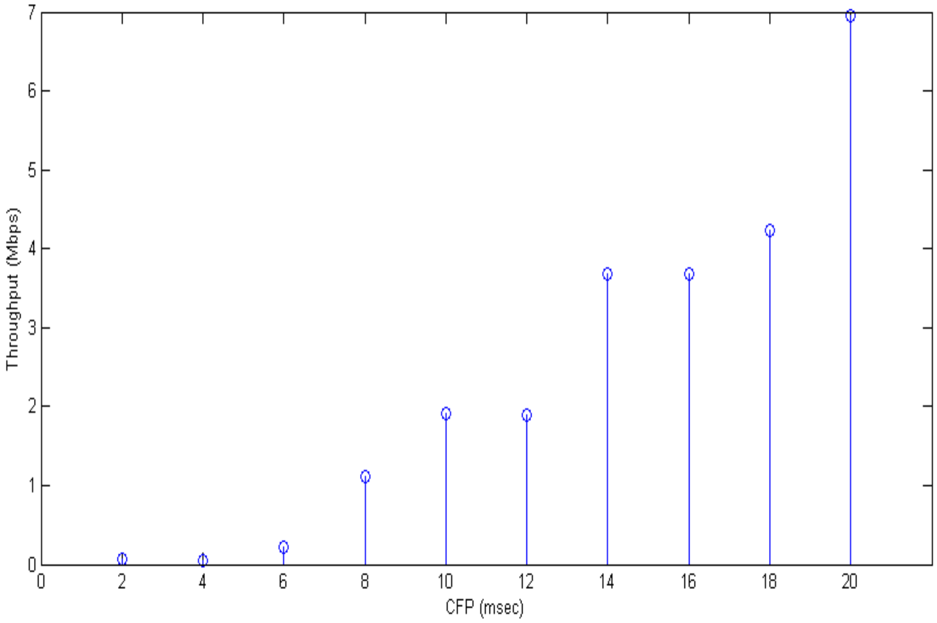


Figure 5-10 Average Throughput as Result of Varying CFP

Figures 5-5 to 5-7, show the delay, jitter and throughput statistics taken directly from the OPNET model simulated for 600 seconds. A probe is sent every 6 seconds to retrieve the delay, jitter and throughput levels. While the stem graphs in Figures 5-8 to 5-10, show the average of the delay, average jitter and average throughput statistics for increasing CFP duration.

The trend in throughput seen in Figure 5-10 is anticipated as increasing CFP will allow more polled applications to be transmitted on the WLAN, hence increasing throughput. Delay and jitter has decreasing trend as CFP increases. An overall conclusion will be made at the end of this section.

5.5.2. Varying SF with CFP Fixed at 50% of SF

In this scenario the SF duration is varied while CFP is fixed at 50% of SF. For example, for 40 msec SF, CFP is 20 msec and for 16 msec SF, CFP is 8 msec, etc. The following Figures 5-11 to 5-13 shows the delay, jitter and throughput resulting from the varying of SF duration.

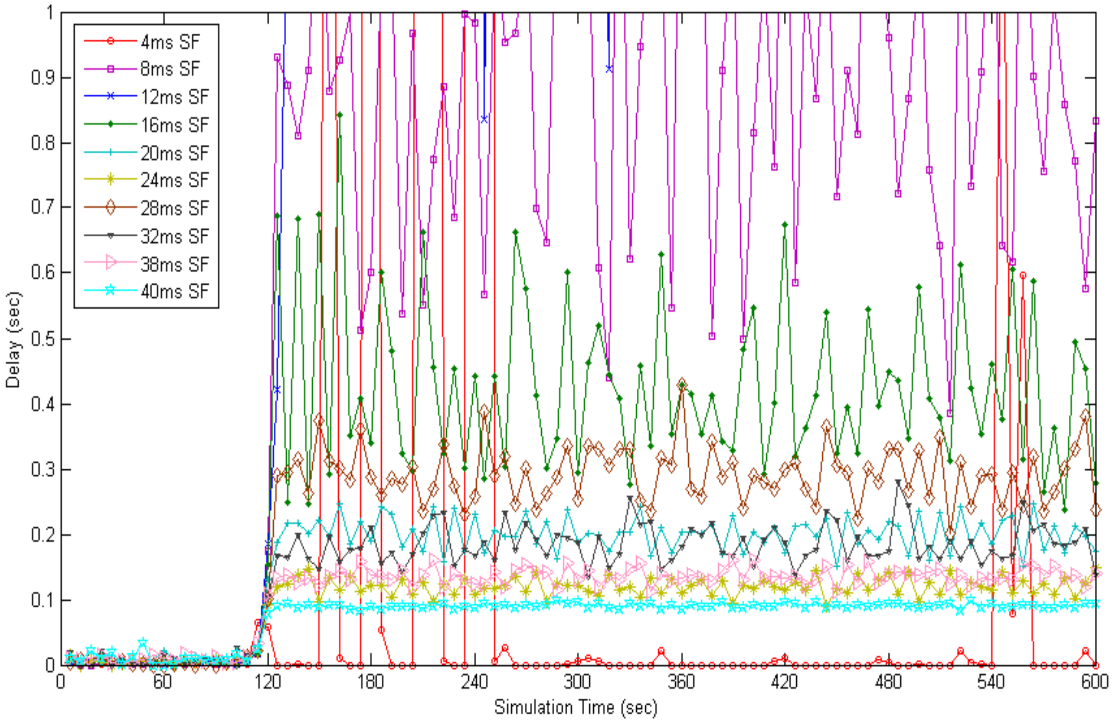


Figure 5-11 Delay as Result of Varying SF Duration

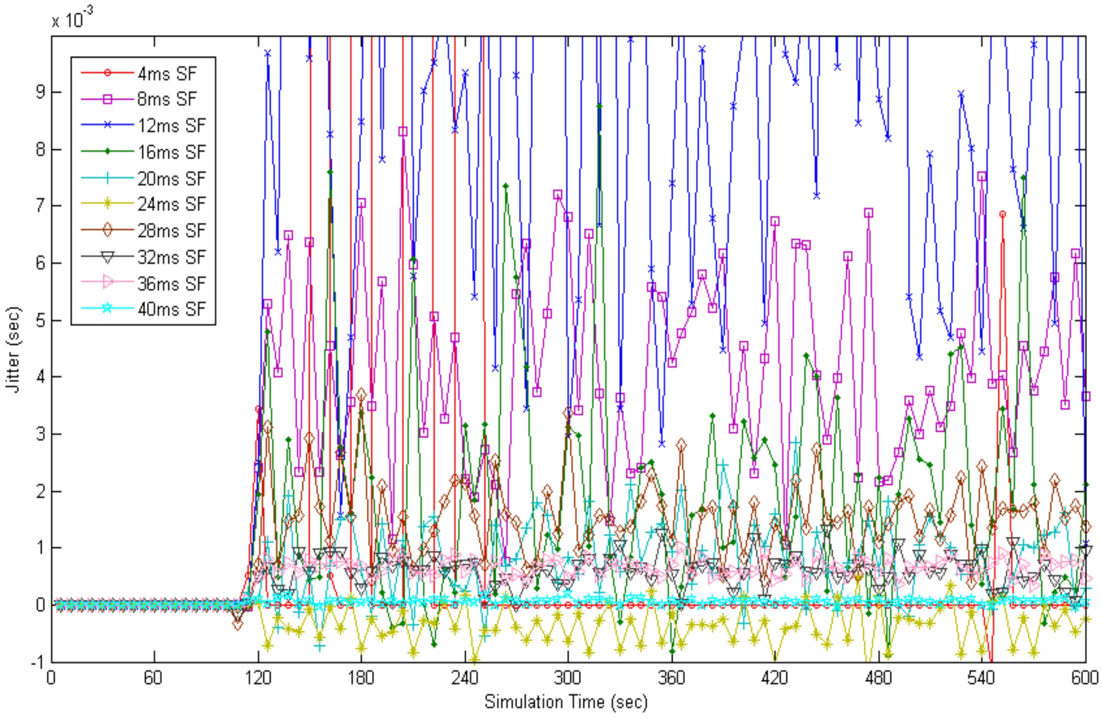


Figure 5-12 Jitter as Result of Varying SF Duration

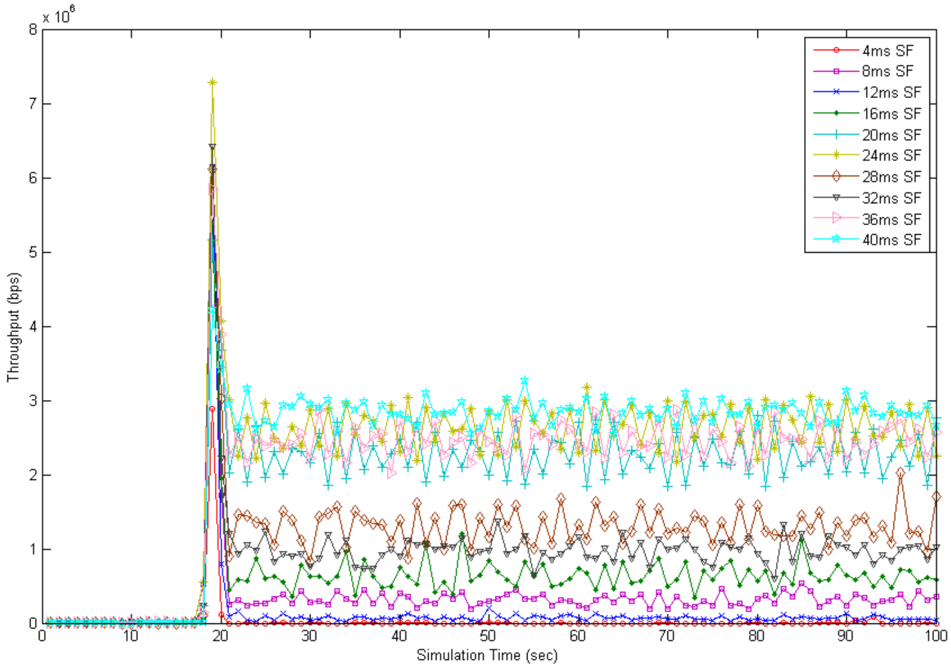


Figure 5-13 Throughput as Result of Varying SF Duration

The trend for average delay, jitter and throughput can be seen in Figures 5-14 to 5-16.

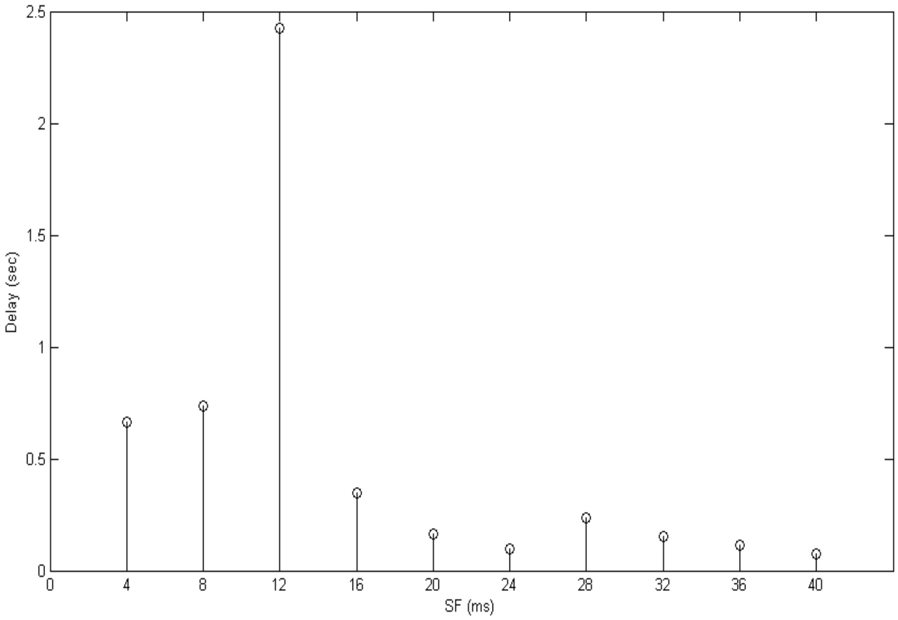


Figure 5-14 Average Delay as a Result of Varying SF Duration

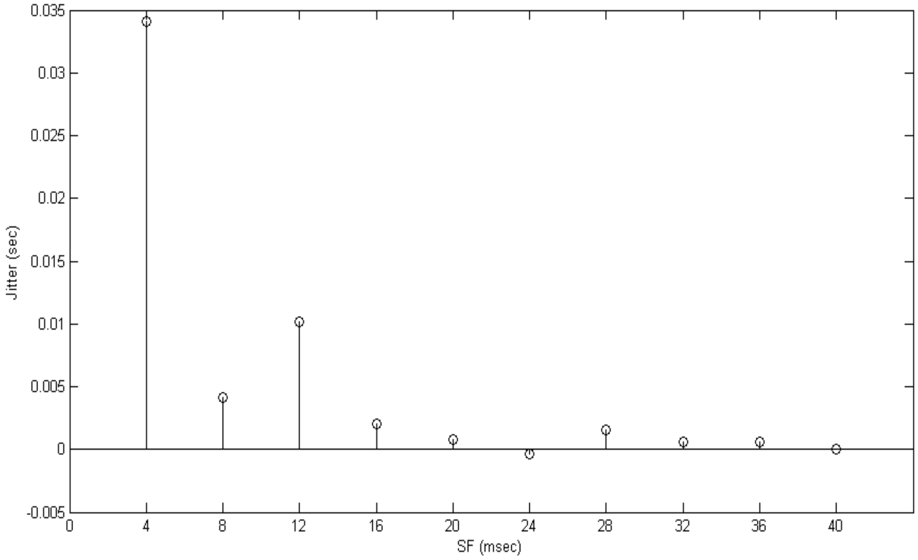


Figure 5-15 Average Jitter as a Result of Varying SF Duration

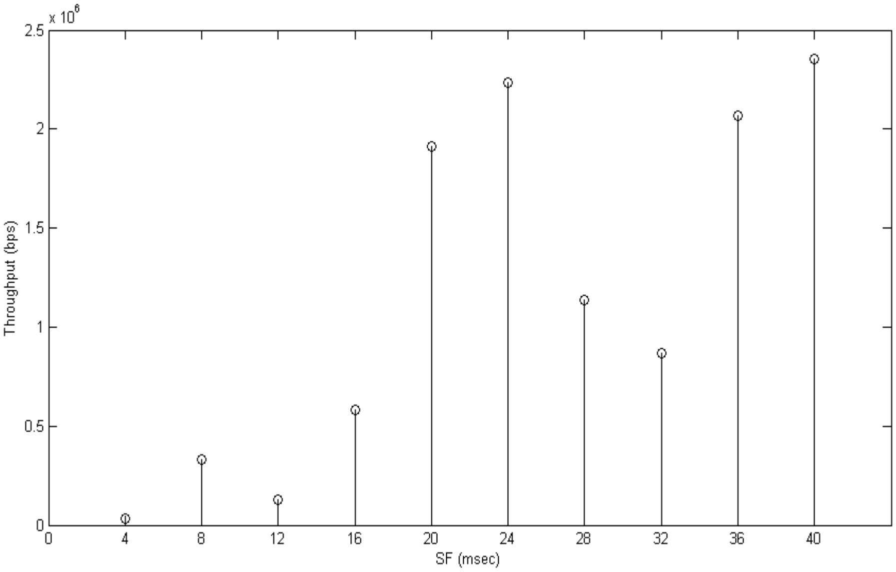


Figure 5-16 Average Throughput as a Result of Varying SF Duration

Delay and jitter metrics have decreasing trend as SF duration increases. Throughput also increases as SF duration increases. An overall conclusion will be made at the end of this section.

5.5.3. Varying IMM (Fixed SF 20 msec and CFP 10 msec)

In this final scenario IMM nodes are varied while SF and CFP durations are set at the default values of 20 msec and 10 msec respectively.

Figure 5-17 Delay as a Result of Varying IMM nodes

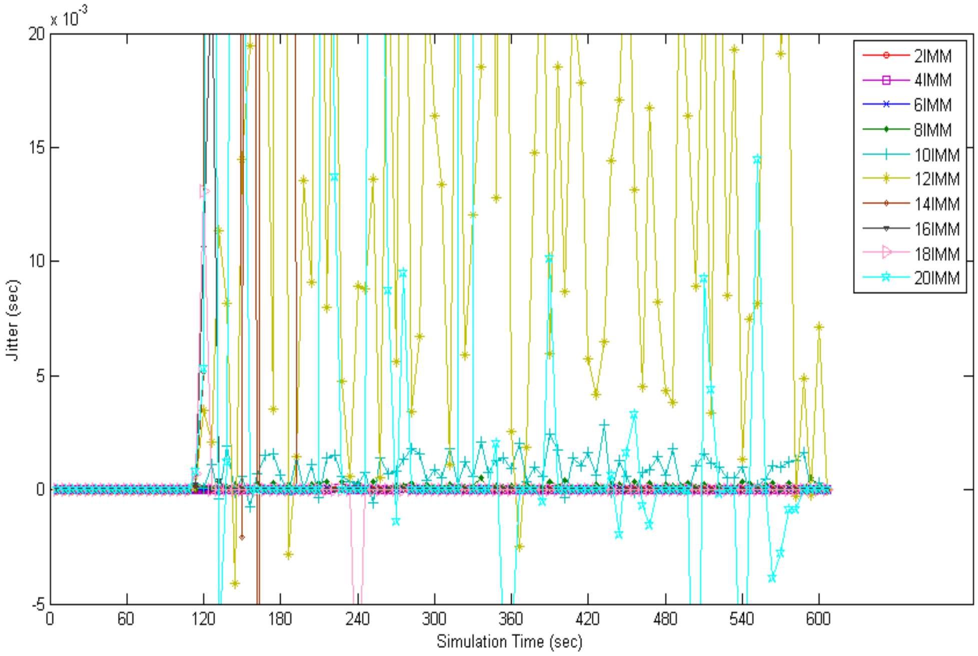


Figure 5-18 Jitter as a Result of Varying IMM nodes

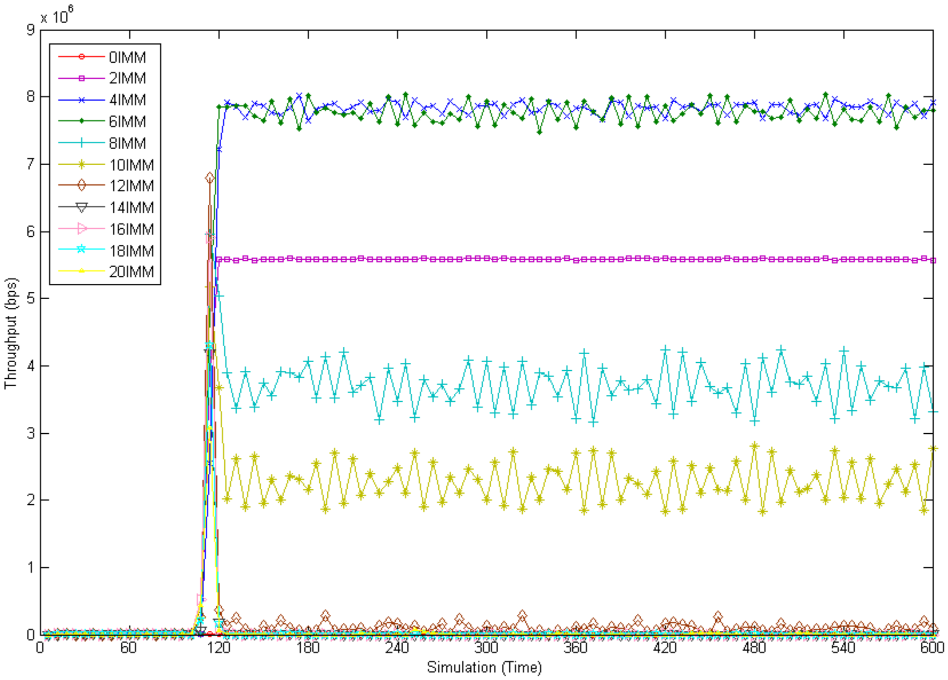


Figure 5-19 Throughput as a Result of Varying IMM nodes

Figures 5-17 to 5-19 show delay, jitter and throughput statistics as a result of varying IMM nodes in the WLAN. Figures 5-20 to 5-22 show average delay, jitter and throughput in the form of a stem graph to demonstrate the trend in these scenarios.

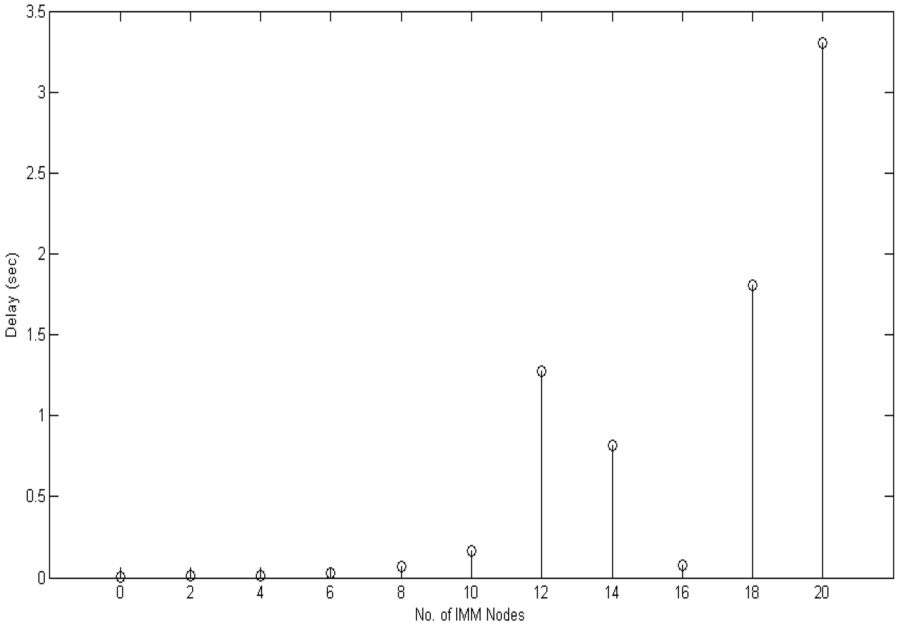


Figure 5-20 Average Delay as a Result of Varying IMM nodes

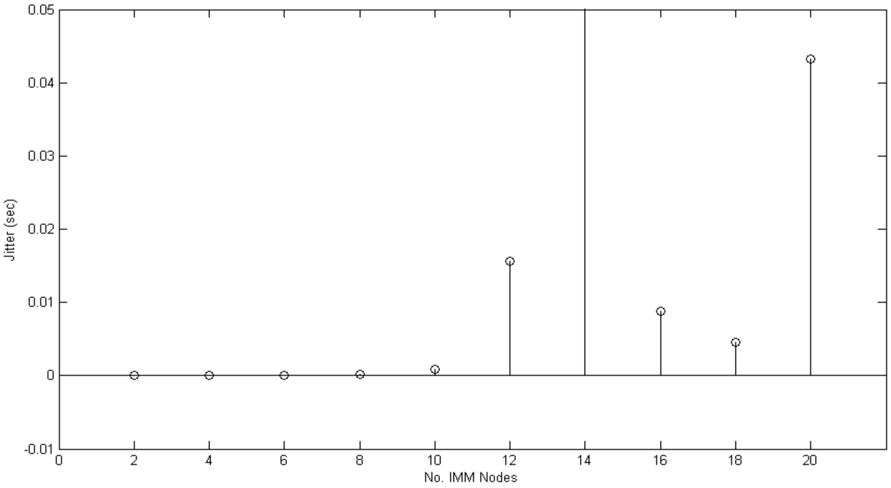


Figure 5-21 Average Jitter as a Result of Varying IMM nodes

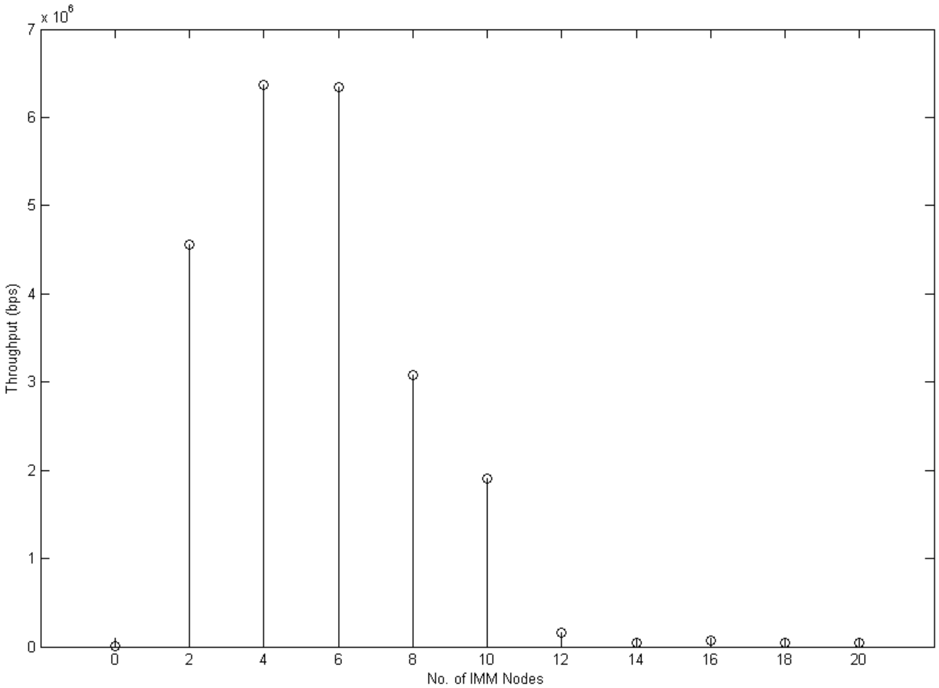


Figure 5-22 Average Throughput as a Result of Varying IMM nodes

In summary the three scenarios show the relevant trends in performance of WLAN in varying SF, CFP and number of IMM nodes. It is evident that increasing SF and CFP will decrease delay and jitter while increasing throughput. Increasing number of IMM nodes, however, will increase delay and jitter while decreasing throughput. CFP cannot use up the entire SF (CFP=SF) as the standard allocates CPmin duration in SF.

5.6. CFP Selection using Look-up Table

It can be argued based on the simulation results presented in the previous sections that for every combination of IMM and data nodes there will be CFP duration to achieve the best QoS metrics. A look-up table of the desired CFP duration can be assembled based on extensive simulation to assist the PC to select the right balance of CP/CFP. In this section we have employed a dynamic PCF based scheme where the PC employs a look-up table to decide on the CFP duration based on the number of IMM nodes. We assume that there are ten non real time data nodes while IMM nodes are varied from 2 to 20. The QoS metrics for this dynamic scheme are compared with those of a standard PCF scheme with CFP duration of 10msec and 20msec SF. Default to dynamic SF configuration settings on OPNET are shown in Appendix B.

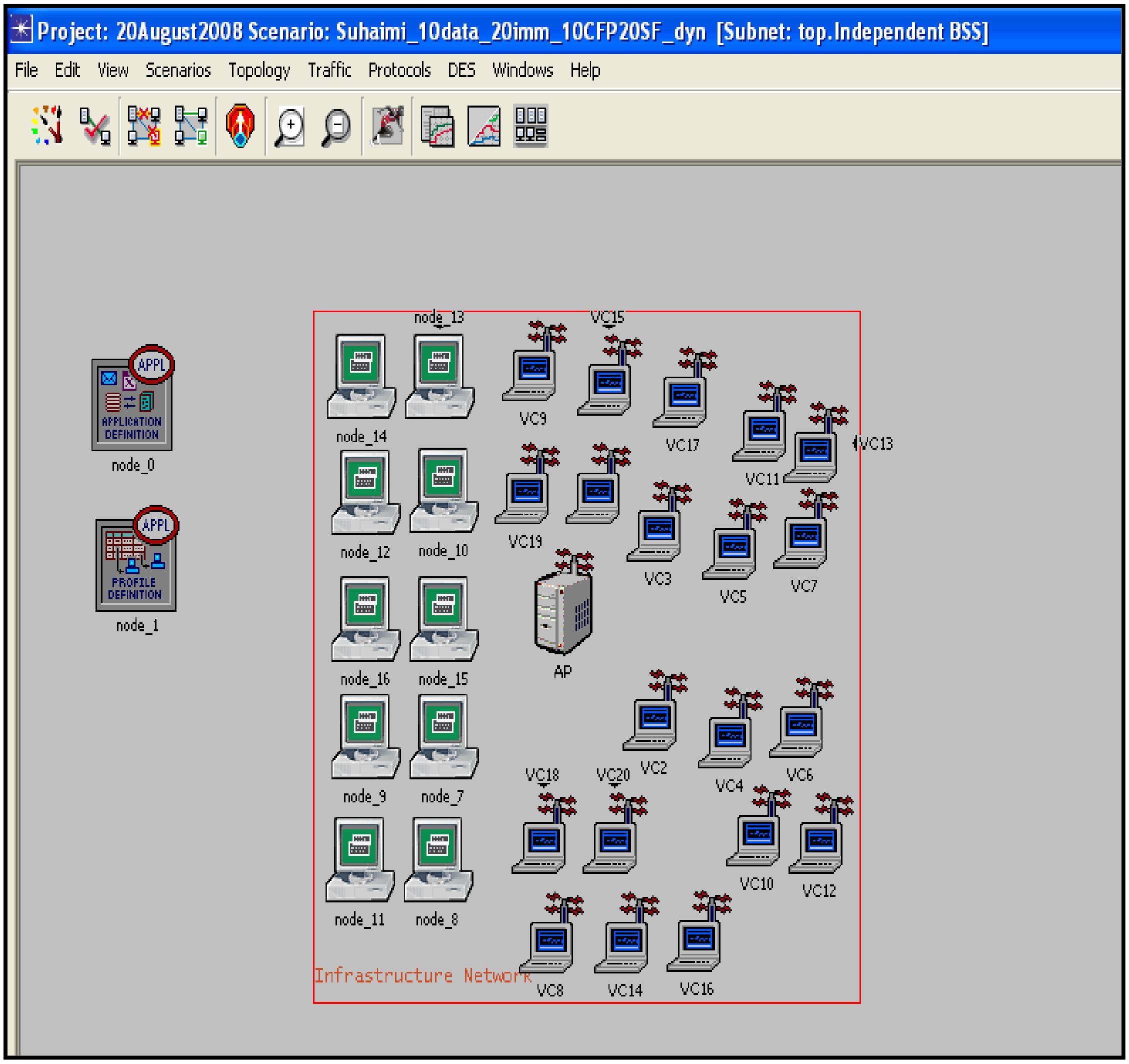


Figure 5-23 Simulation Setup to Model Dynamic PCF Scheme

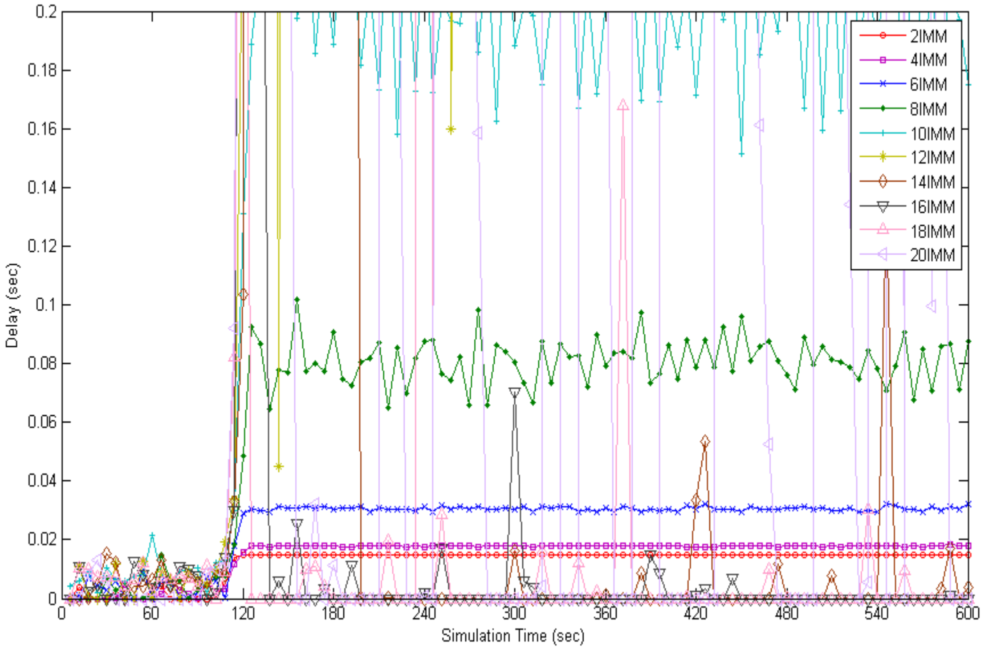


Figure 5-24 Delay Statistics as a Result of Default SF Configuration

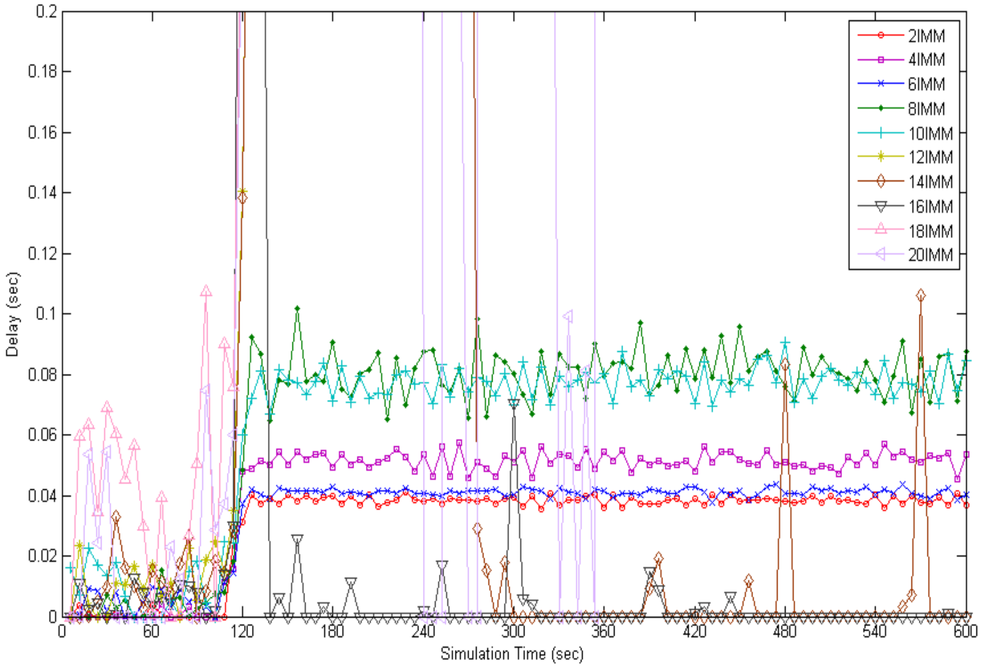


Figure 5-25 Delay Statistics as a Result of Dynamic SF Configuration

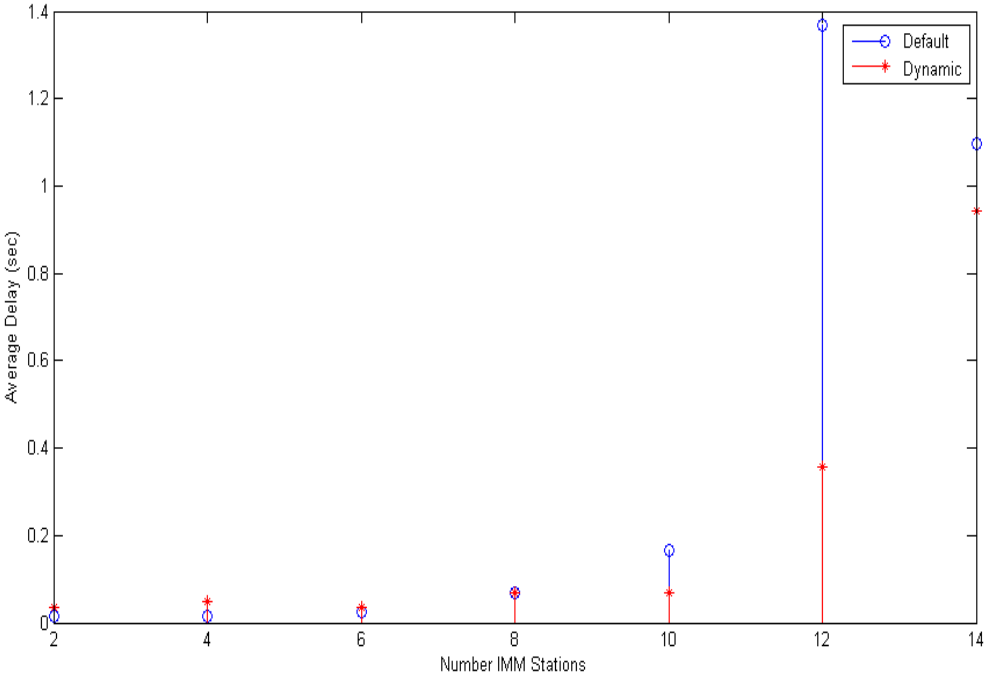


Figure 5-26 Average Delay Trend for Dynamic and Default SF Configuration

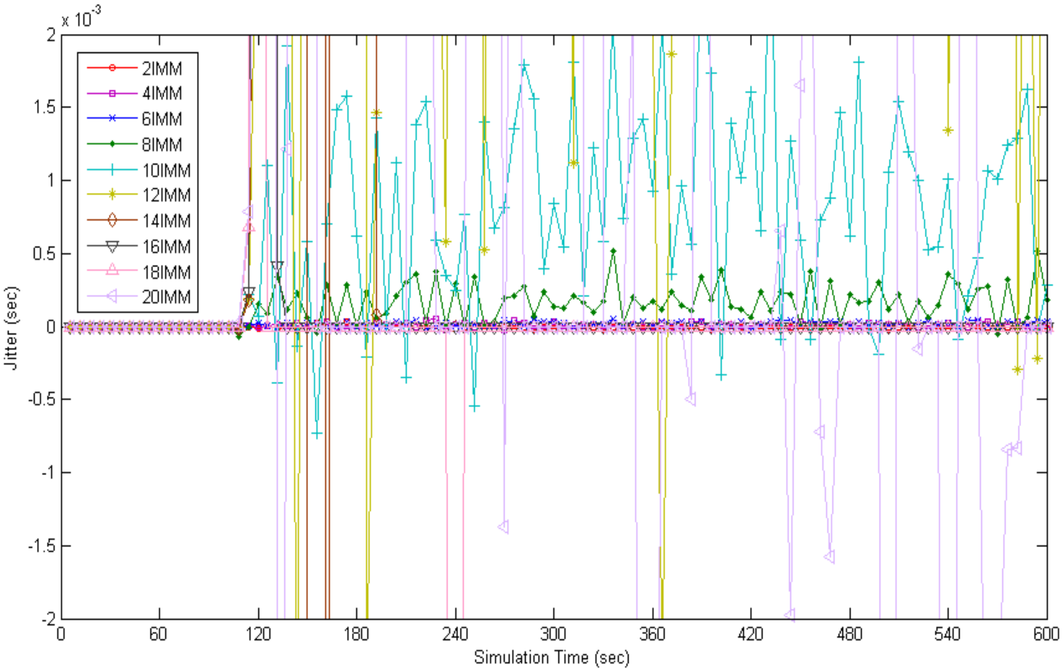


Figure 5-27 Jitter Statistics as a Result of Default SF Configuration

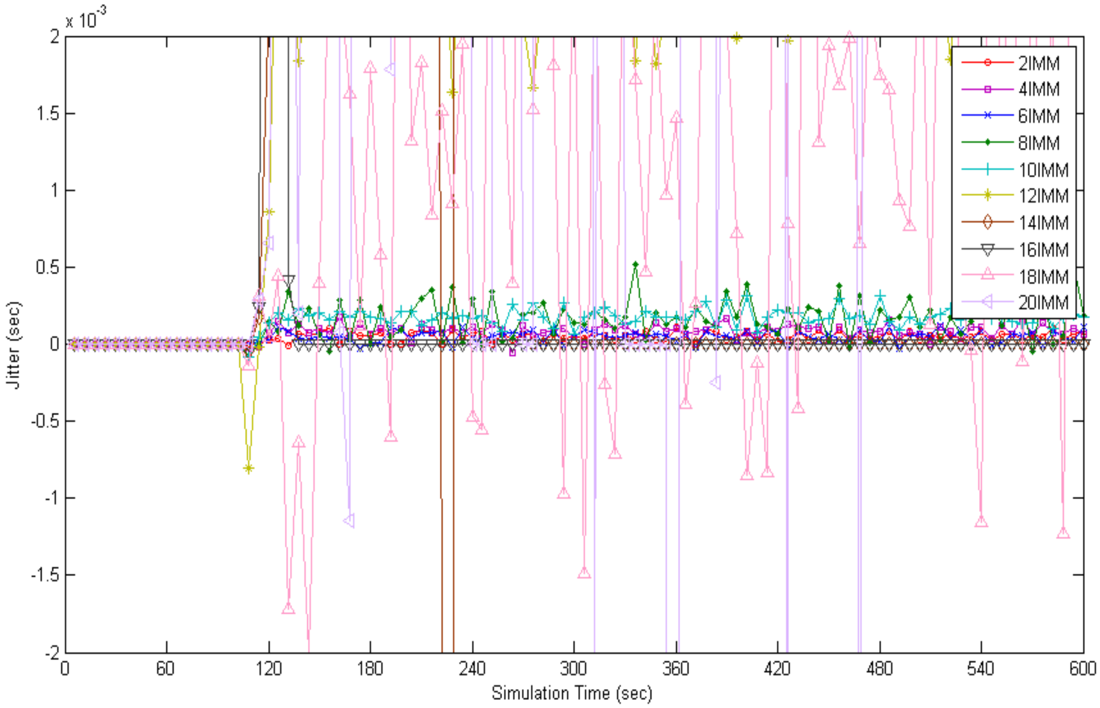


Figure 5-28 Jitter Statistics as a Result of Dynamic SF Configuration

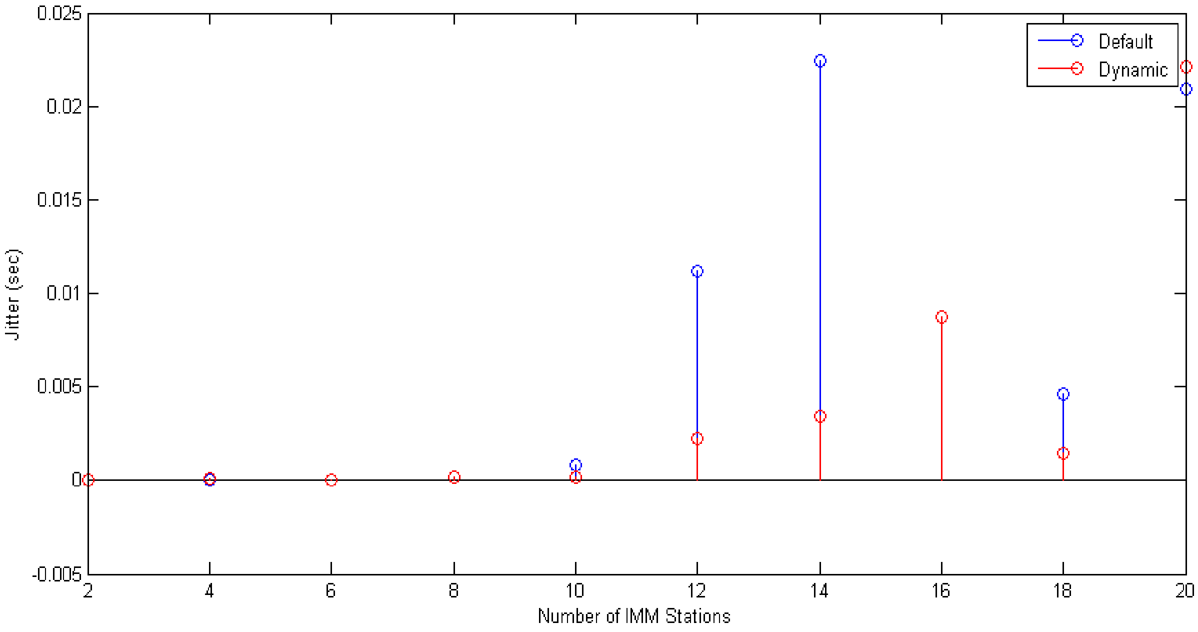


Figure 5-29 Average Jitter Trend for Dynamic and Default SF Configuration

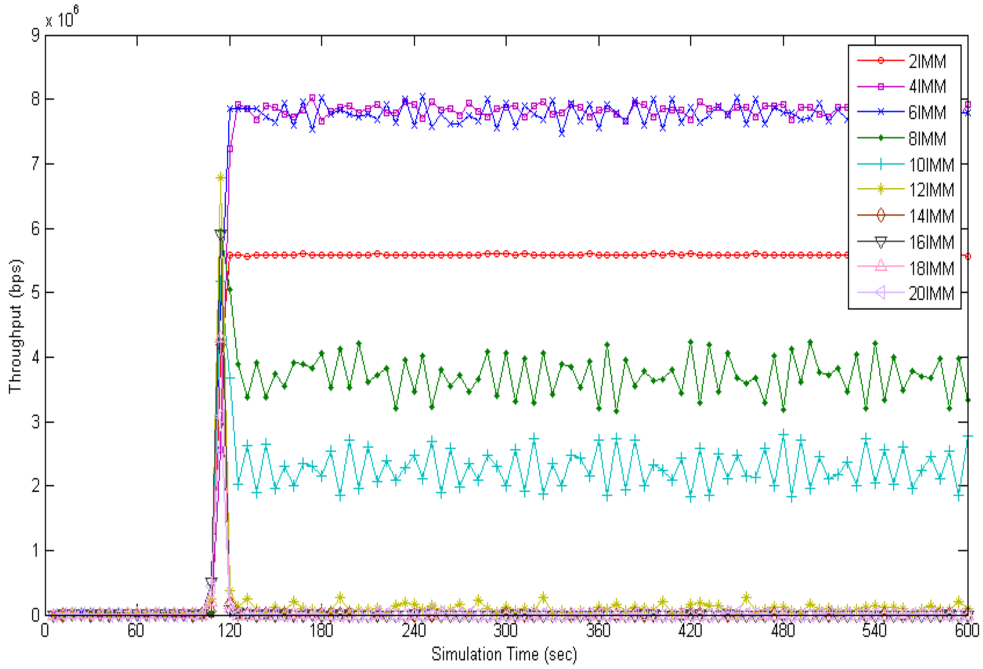


Figure 5-30 Throughput Statistics as a Result of Default SF Configuration

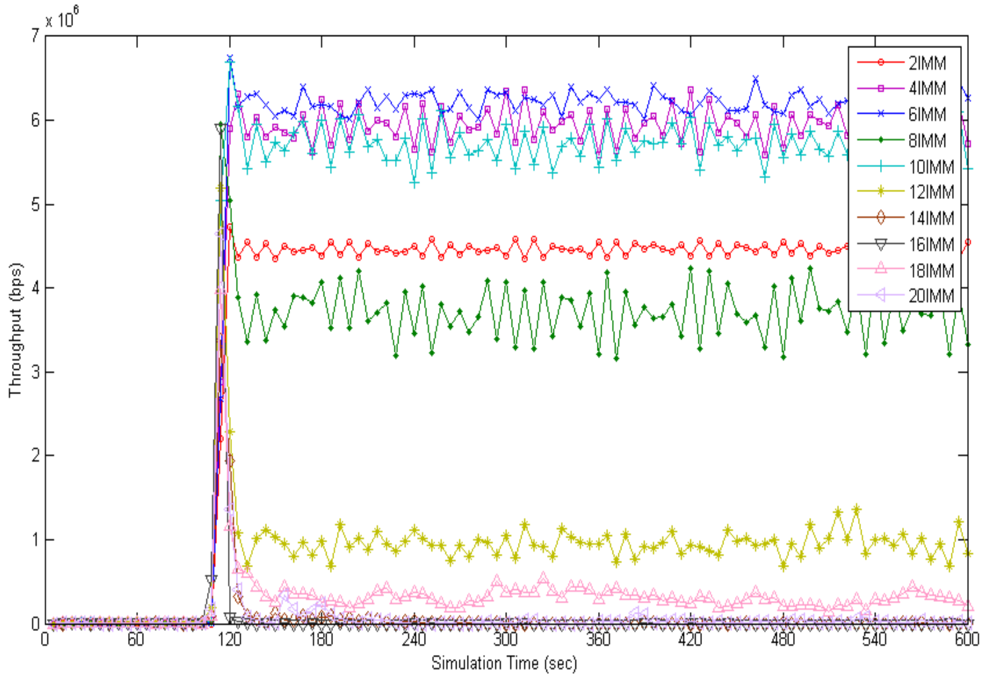


Figure 5-31 Throughput Statistics as a Result of Dynamic SF Configuration

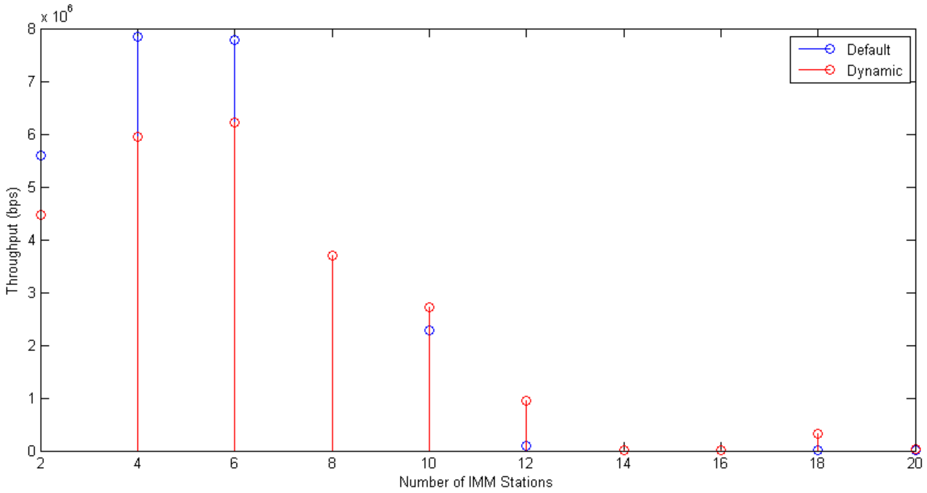


Figure 5-32 Average Throughput Trend for Dynamic and Default SF Configuration

In Figure 5-8, OPNET simulation results shows that the default PCF configuration produced high delays (beyond 0.1seconds) with more than 8 IMM nodes in the network. Dynamic configuration, however, has managed to keep the delay below 0.1seconds with an additional two IMMs in the network, making ten IMMs. Figure 5-10, shows the graphical representation of average delays comparison for default and dynamic comparison. Figure 5-11, shows the jitter statistics for the WLAN under default PCF configuration. Jitter statistics went beyond the 0.5msec mark when there are more than eight IMM nodes in the network. In Figure 5-12, however, jitter statistics went beyond the 0.5msec mark when there are more than ten IMM nodes in the network. This is clearly an improvement to the number of IMM nodes that could be made available with dynamic configuration, when delay and jitter is concerned. As for throughput, both default and dynamic PCF configuration produced complimentary results. Default PCF configuration produced higher throughput for six and lesser number of IMM nodes while dynamic PCF configuration produced relatively higher or equal number of average throughput for eight and more IMM nodes. However, this case is true in larger number of IMM nodes which most relevant and consistent with our objectives in this research.

We can clearly observe that the dynamic scheme has lower delay and jitter while providing the same throughput.

5.7. Validation of Simulation Models

Model Validation, as defined by Biachi [114], is substantiating that the model, within its domain of applicability, behaves with satisfactory accuracy consistent with the study objectives. One of the techniques in model validation is by substantiating that the input-output transformation of the model has sufficient accuracy in representing the input-output transformation of the system. Model validation deals with building the right model. It is conducted by running the model under the same input conditions that drive the system and by comparing model behaviour with the corresponding system behaviour.

Various validation and verification techniques mentioned by Biachi and also found in the literature include animation, degenerate tests, event validity, face validity, internal validity and multistage validation. The most suitable one for this research is the “comparison to other models” technique.

The voice, data and video traffic models used in the simulations presented in this thesis are taken from the predefined traffic generator in OPNET. Interactive voice and interactive video models are used to represent an IMM traffic mix. Voice traffic is modelled as conversation speech with periods of silence and encoded with the standard G.711 scheme for voice over internet protocol (VoIP). Interactive video is modelled with a frame interarrival rate of 10 frames per second. VBR video sequence in OPNET is obtained from video conference encoded in H.263. An example of the video traffic sent, is shown in Figure 5-33. On the average, 12MB traffic at the rate of 10 kbps is transmitted on to the WLAN during the simulation. This is consistent with that found in published literatures.

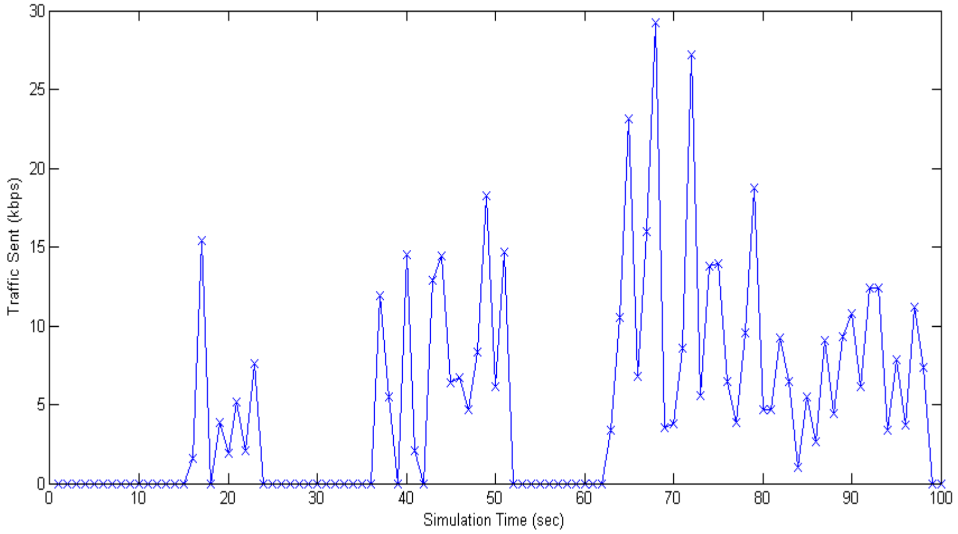


Figure 5-33 Typical Rate of Video Traffic Sent to the WLAN

On top of the consistency in the traffic simulated, OPNET being proprietary software built by a large team of programmers is understood to have taken necessary steps to ensure that computerized models are adequately accurate. Following this, statistical results obtained from our models are consistent with other published (validated) models. Therefore, the consistent simulation results prove the validity of these models.

MDSS generates an optimum SF configuration according to the QoS requirements of traversing IMM traffic. The scheme will provide optimum bandwidth for IMM traffic while giving appropriate bandwidth allocation to non-IMM traffic. MDSS will be implemented in the AP as a dynamic lookup table indexed by number of IMM traffic currently in the polling list. In practicality the station sends an associate request to join the polling list of the AP. Once this information is received the AP forms the polling list, then it sends a beacon frame with CFP period and beacon interval to all stations in the base station (BSS). MDSS could reside in the AP or outside. The lookup table could be generated online or offline with information required by the AP like QoS delay/jitter budget and number of IMM and non-IMM nodes are updated. The MDSS will generate an optimal SF/CFP duration suitable for the traversing traffic.

3.6. Chapter Summary

In summary, as discussed in this chapter the trend of research, are focused on the scheduling issues rather than polling. The published literatures on polling were trying to deal with the transmission times of the information packets and overhead due to polling of stations that has no frames to transmit. An open issue on improving WLAN to enable it to carry interactive multimedia applications is via SF configuration. This paper has presented a promising technique to improve the performance of IMM transmission over WLANs using the same. Our focus is on the PCF, which can be seen as a specialized case of the more rigid and complex coordination function of IEEE 802.11e. The effects on the QoS metrics measured, following variation of superframe and contention free period durations were investigated via simulation of typical mixed traffic WLANs scenario on OPNET. The configuration of this superframe, was proven to, directly affect the system's ability to support the mixed traffic effectively. If the configuration is injudicious, QoS requirements of the multimedia traffic will be missed and/or the data traffic will be starved of access. Therefore, tuning the CFP and SF duration can attain better QoS profiles. A correct choice of CFP duration with respect to the SF has considerable impact on the performance of the WLAN in transmitting interactive multimedia traffic. The study has clearly demonstrated that a scheme that dynamically selects CFP and SF will ensure QoS in transmitting interactive multimedia traffic.

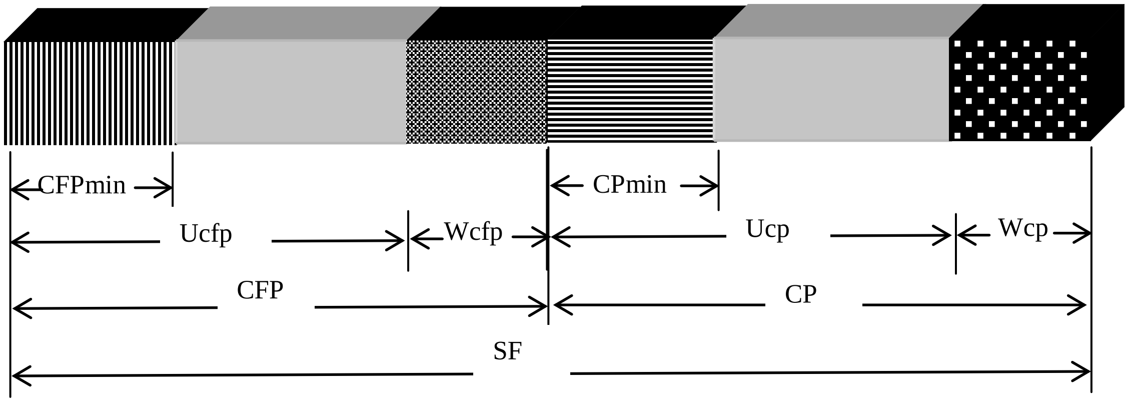
Chapter 6
Analytical Model of MDSS

Previous sections have established the importance of polling in transmitting interactive multimedia (IMM) traffic. We have shown, through simulation results that the partitioning of superframe (SF) into appropriate contention-free period (CFP) and contention period (CP) duration is essential for efficient transfer of real time (RT) delay sensitive traffic. If this partitioning is adversely erroneous, the QoS requirements of the IMM applications will not be fulfilled and the non-IMM applications will be starved. Further to that, a dynamic approach to the configuration of SF, with respect to the number of IMM nodes and the traffic mix, would ensure a sustainable supply of optimum CFP and CP combinations. An auto-generated optimum parameter set, which consists of CFP and SF combinations, is essential to ensure that both IMM and non-IMM traffic achieve fairness in utilizing the wireless local network (WLAN) resources. In order to find this optimum parameter set, a basic mathematical model will be needed to represent the phases in the SF i.e., the point coordination function (PCF) and distributed coordination function (DCF). A suitable optimization algorithm will be implemented to find a solution to the model. This optimum parameter set will be fed into the simulation scenarios to demonstrate the efficiency of our solutions.

This chapter aims at mathematically modelling the two access schemes i.e., PCF and DCF of a WLAN medium access control (MAC), in order to generate the optimum parameter set. The optimum value for CFP and CP is achieved by maximizing utilization of the PCF access schemes while providing fairness to the DCF scheme during each SF.

6.1. CFP and CP Optimization

In this research the term optimization, refers to a method in mathematical programming, to the study of problems to minimize the objective function by systematically choosing the values from within an allowed constraint. Optimizing the use of CFP and CP by minimizing their wastage would be one of the ways to model the SF. The lengths of the CFP and the CP are controlled explicitly by the contention free period repetition interval (CFPri). Here the CFPri duration is where the PCF and DCF alternate an SF. Utilisation of the polled IMM and contending non-IMM traffic could be a suitable representation to model our scheme. This could be calculated from the amount of time incurred at the frame level i.e., ratio of CFP and CP overheads to their actual allocated time. What is left of the wastage is taken as the utilization period.



Where,

SF: Superframe (msec)
CFP: Contention free period (msec)
CP: Contention period (msec)
CFPmin: Minimum CFP duration
CPmin: Minimum CP duration
Wcfp: CFP wastage
Wcp: CP wastage
Ucfp: CFP utilization
Ucp: CP utilization

Figure 6-1 Diagram of SF Partitioning into CFP and CP

Both the CFPmin and CPmin, in Figure 6-1, are minimum duration constraints for CFP and CP respectively. The IEEE standard has defined that the CFP has to be at least big enough to contain one polled exchange comprising the largest payload possible in each direction, plus a Beacon and a `CF-End`. The CP has to be large enough to contain an acknowledged exchange of the largest payload possible. Ucfp and Ucp are representatives of the utilization in CFP and CP respectively.

6.2. Analytical Model

The bandwidth utilization of the two phases in one SF can be modelled as that done by Haines et. al. [96]. They have modelled these two phases as the objective function of their optimization analysis. We will have utilized their work and possibly make substantial improvements to it. Their optimization solution was developed on the basis that both the phases form the objective function and respective utilizations are optimized.

The utilization in CFP and CP over the entire SF are shown in Equations (6-1) and (6-2).

$$\text{Ucfp} = \frac{Np(Cb - Ca)}{xy} \qquad \text{Equation (6-1)}$$

$$\text{Ucp} = \frac{\Pr Nc(Ms - Hs)}{1 - x} \qquad \text{Equation (6-2)}$$

where,

Np: Number of polled stations
Cb: Entire polled exchange duration (msec)
Ca: Polled exchange overhead (msec)
x: CFP (%) i.e., Ucfp+Wcfp as in Figure 6-1
y: SF (msec) i.e., Ucp+Wcp as in Figure 6-1
Pr: Contending traffic packet generation rate (packets per sec)
Nc: Number of contending stations
Ms: Standardized contended exchange overhead (msec)
Hs: Entire standardized contended exchange duration (msec)

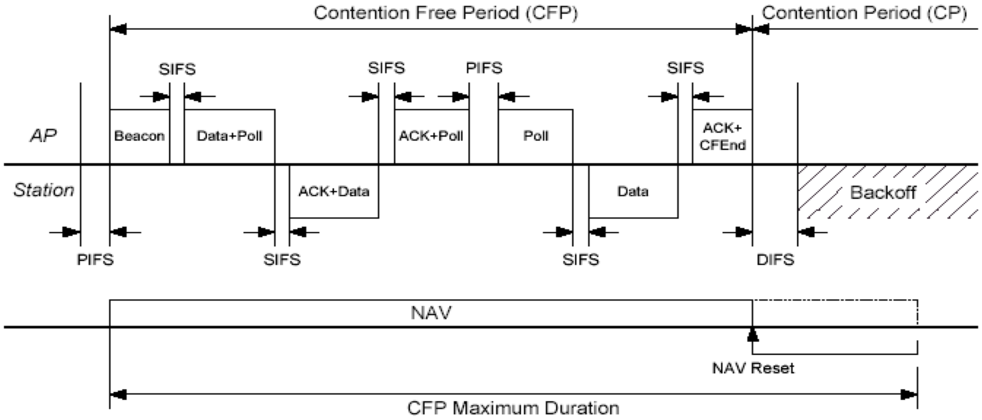


Figure 6-2 SF model showing wastages in CFP and CP [1]

Wastage in CFP, Wcfp, comes from the standard inter-frame space (SIFS) as shown in Figure 6-2. The polled exchange overhead, Ca, is twice the SIFS duration as in Equation 6-3.

$$Ca= 2 * SIFS \qquad \text{Equation (6-3)}$$

Utilization of the CFP, Ucfp, is the difference between the allocated polled exchange duration Cb and the polled exchange overhead, Ca. For all polled stations, Np, in the CFP, the utilization can be defined as in Equation (6-1). Wastage in CFP, Wcfp, as depicted in Figure 6-1 can be expressed as in Equation (6-4).

$$Wcfp = 1 - \frac{Np(Cb - Ca)}{xy} \qquad \text{Equation (6-4)}$$

Wastage in CP, Wcp comprises of the overhead, Ms which could be expressed as in Equation (6-5).

$$Ms = DIFS + Backoff + SIFS + ACKFrame \qquad \text{Equation (6-5)}$$

Utilization of the CP, Ucfp, is the difference between the allocated standardized contended exchange duration, Hs and overhead, Ms. For all contended stations, Nc, in the CP arriving at the rate, Pr, the utilization can be defined as in Equation (6-1). Wastage can be expressed as in Equation (6-6).

$$\mathrm{Wcp} = 1 - \frac{\mathrm{Pr}\,\mathrm{Nc}(\mathrm{Ms} - \mathrm{Hs})}{1 - \mathrm{x}} \qquad \text{Equation (6-6)}$$

Our scheme aims to maximize the utilization of CFP while providing fairness to CP i.e., minimizing Wcfp while providing a threshold value for Wcp, therefore providing QoS. For this, the objective function could be defined as in Equation 6-7 with only the Wcfp while the Wcp term is made into a constraint.

$$f(x,y) = 1 - \frac{\mathrm{Np}(\mathrm{Cb} - \mathrm{Ca})}{\mathrm{xy}} \qquad \text{Equation (6-7)}$$

Haines et. al, has taken results from Li et. al, [117] as their benchmark. Li has proposed a self-adaptive scheme to configure the PCF superframe. They have selected the parameters from pre-defined look-up tables indexed by a quantized number of active polled stations and stepped values for the maximum allowable delay of the active applications. The values populating the look-up tables are derived through experimental simulation results. In practicality, the size of the look-up table will be directly proportional to the size and traffic combination in the WLAN. However, it will be an inefficient use of computational resources having such large look-up tables in order to accommodate the various sizes and traffic combinations. A dynamic scheme would be essential, hence MDSS.

6.3. A Model of the MDSS

We aim to maximize the utilization of CFP while providing fairness to CP. In other words we will ultimately minimize CFP wastage while providing threshold value in CP.

The objective function is left with only the CFP term while the CP term is made into a constraint. The basis of this is the importance of IMM traffic in the contemporary and next generation networks, if not all, most of the traffic are and will be IMM. It is only logical to us that fulfilling the requirements of the IMM traffic be the main objective of this optimization scheme. However, possibly providing fair resources to time insensitive non-IMM traffic is a constraint to be considered.

Published materials [22, 23, 103, 118, 119] predict the high volume of multimedia traffic on shared media. It is only realistic that our solution optimizes the duration utilized by IMM i.e., the CFP duration while providing fair resource to the traditional data traffic. Treating IMM traffic as data traffic or equating both IMM and non-IMM together could compromise a fair treatment that should be given to these QoS sensitive traffic.

Our MDSS scheme will virtually generate a logical lookup table indexed by active IMM nodes in the network. The optimization process will be automated by a simple C/C++ programme to populate the lookup table. Unlike Li's work, however, the logical lookup table created automatically when there is a change in traffic mix would be a more dynamic approach. This would be more scalable and less resource hungry.

For that, we have reduced the objective function to an optimization of the CFP duration, while CP duration is taken as a constraint, hence optimization of variables *x* and *y* hence the Equation (6-7).

Possible solutions for Equation (6-7) are constrained as shown in Equations (6-8) to (6-12). These are namely CFPmin cannot be bigger than CFP i.e., x*y, CPmin not bigger than CP i.e., (1-x)*y, since x a percentage should be between the range of 0 and 1, y is bounded by the delay budget D and finally the threshold value for Wcp.

$$\mathrm{CFP}\min - xy \le 0 \qquad \text{Equation (6-8)}$$

$$\mathrm{CP}\min - (1-x)y \le 0 \qquad \text{Equation (6-9)}$$

$$0 \le x \le 1 \qquad \text{Equation (6-10)}$$

$$0 \le y \le D \qquad \text{Equation (6-11)}$$

$$Wcp \geq 0 \quad \text{Equation (6-12)}$$

On top of constraints defined in equations 6-8, 6-9, 6-10 and 6-11 the CP wastage is added as in Equation (6-12). This will provide the fairness needed by non-IMM traffic. It will be the constraint in fulfilling the requirements of the objective function.

As discussed in the previous section, wastage in CP has to leave enough time for CPmin. We have to limit the size of Wcp so that I does not go beyond CP-CPmin or in other words, Ucp has to be at least equals to CPmin, as shown in Equation 6-13 and Equation 6-14.

$$Wcp \leq CP - CP\min \quad \text{Equation (6-13)}$$

$$Ucp \geq CP\min \quad \text{Equation (6-14)}$$

Table 6-1 shows typical and standard values for the objective function and constraints. These values are obtained experimentally. The constants are compiled according to the concrete values and expressions explained and developed in [83, 120]. These constant values will be used in the optimization process that follows.

Table 6-1: Constants for Optimization Function [117]

Constant Parameter	***Value***
Contending data exchange overhead, Ms	0.674 msec
Contending data exchange duration, Hs	4.978 msec
Contending data packet generation rate, Pr	0.0075 s^{-1}
Polled exchange overhead, Ca	0.02 msec
Polled change duration, Cb	2.228 msec
CFPmin	39.922 msec
CPmin	21.404 msec

Choice of the starting point in an interior point method for solving non linear optimization problems is crucial [96, 121, 122]. The objective function, in Equation 6-7 is not convex so feasible starting points must be determined to pick the appropriate local minima. The starting points in Table 6-2 can be considered as suggested in [96].

Table 6-2: Feasible Starting Points

Delay Budget (msec)	x_o	y_o
75	0.6	70
100	0.7	145
150	0.8	145
200	0.8	195

Values of **y_o** is chosen close to D and will define constraint in Equation 6-15, where y= **y_o**. **x_o** is defined as in Equation 6-16.

$$x_0 \in \left(\frac{\mathrm{CFP}_{\min}}{y}, 1 - \frac{\mathrm{CP}_{\min}}{y} \right) \quad \text{Equation (6-15)}$$

$$CFP_{\min} \langle y_0 - CP_{\min} \quad \text{Equation (6-16)}$$

6.4. Algorithm Complexity

In implementing our scheme, a typical question of computational complexity would be central. It is essential that the complexity is kept to minimal as the main functionality of the AP in managing the WLAN resources should not be deprived. The running time and memory requirements of the algorithm, the implications and ramifications are fundamental in the success of our scheme. In other words, the theory, among other things, the scalability of computational solutions and algorithms has to be justified. In order to achieve this, we had utilised the time complexity technique. In this a problem is the number of steps that it takes to solve an instance of the problem as a function of the size of the input (usually measured in bits), using the most efficient algorithm.

The main justification for using Interior Point or Barrier Methods, for solving our problem has to be its efficiency (fast convergence, scalable, etc.) in solving non-linear optimization problems as detailed in [123-126] and references therein. Interior point method achieves optimization by going through the middle of the solid defined by the problem rather than around its surface. It is also chosen for comparison with results obtained by Haines.

6.5. Comparison of Optimization Models

We have utilized the Interior point method in the Optimization tool box available in MATLAB. Results of the optimization from Haines and Li are shown in Table 6-3 and 6-4 respectively. For the purpose of comparisons, the assumptions and parameters used by them have been adopted here. These include a WLAN setup with 10 data stations contending for access in CP with varying number of voice stations polled in CFP. Results of our optimization are shown in Table 6-5.

Table 6-3 Haines Optimization Results [96]

		No. of IMM nodes									
		2	4	6	8	10	12	14	16	18	20
D= 200 (msec)	SF(msec)	173.00	184.00	181.00	182.00	184.00	168.00	165.00	162.00	148.0	152.0
	CFP(msec)	50.69	45.26	43.80	44.41	44.34	76.10	62.70	84.24	51.21	58.52
D= 150 (msec)	SF(msec)	126.00	130.00	135.00	125.00	132.00	126.00	139.00	92.00	108.00	125.00
	CFP(msec)	61.36	68.64	76.82	51.50	46.60	55.82	83.96	46.64	54.76	60.25
D= 100 (msec)	SF(msec)	100.00	91.00	90.00	89.00	90.00	78.00	85.00	100.00	100.00	100.00
	CFP(msec)	39.90	45.68	46.80	47.88	46.71	39.78	42.84	39.90	39.90	39.90
D= 75 (msec)	SF(msec)	75.00	75.00	75.00	75.00	75.00	75.00	75.00	75.00	75.00	75.00
	CFP(msec)	39.90	39.90	39.90	39.90	39.90	39.90	39.90	39.90	39.90	39.90

Table 6-4 Li Optimization Results [117]

		No. of IMM nodes									
		2	4	6	8	10	12	14	16	18	20
D= 200 (msec)	SF(msec)	28.00	22.00	21.00	100.00	55.00	40.00	33.00	27.00	24.00	75.00
	CFP(msec)	5.60	6.60	8.40	40.00	27.50	24.00	23.10	21.60	21.60	67.50
D= 150 (msec)	SF(msec)	30.00	23.00	24.00	20.00	100.00	50.00	39.00	33.00	29.00	42.00
	CFP(msec)	6.00	6.90	9.60	10.00	50.00	30.00	27.30	26.40	26.10	39.90
D= 100 (msec)	SF(msec)	40.00	30.00	32.00	27.00	25.00	20.00	80.00	50.00	40.00	100.00
	CFP(msec)	8.00	9.00	12.80	13.50	15.00	14.00	56.00	40.00	36.00	95.00
D= 75 (msec)	SF(msec)	50.00	50.00	55.00	45.00	40.00	35.00	32.00	30.00	60.00	100.00
	CFP(msec)	10.00	15.00	22.00	22.50	24.00	24.50	25.60	27.00	57.00	95.00

Table 6-5 Proposed Optimization Results

		No. of IMM nodes									
		2	4	6	8	10	12	14	16	18	20
D= 200 (msec)	SF(msec)	63.93	62.97	62.66	62.56	62.54	62.59	62.72	63.09	64.31	67.95
	CFP(msec)	43.28	42.63	42.42	42.35	42.34	42.37	42.46	42.71	43.54	47.50
D= 150 (msec)	SF(msec)	63.91	62.96	62.65	62.56	62.54	61.96	62.72	63.08	64.30	74.39
	CFP(msec)	43.27	42.63	42.42	42.35	42.34	41.95	42.46	42.71	43.53	50.36
D= 100 (msec)	SF(msec)	63.86	62.95	62.65	62.55	62.54	66.12	66.72	63.07	66.54	81.28
	CFP(msec)	43.23	42.62	42.41	42.35	42.34	44.76	45.17	42.70	45.05	55.03
D= 75 (msec)	SF(msec)	70.08	69.80	69.53	62.54	69.30	62.56	69.42	62.15	67.76	70.16
	CFP(msec)	47.44	47.26	47.07	42.34	46.92	42.35	47.00	42.08	45.87	47.50

A general observation of the optimization results, show that Haines scheme has relatively larger SF and CFP durations as compared to that of Li and the MDSS scheme. Smaller SF sizes will generally result to a shorter waiting time for the polled stations to utilize the network resources. For the proposed scheme, this has been anticipated, as our objective function is based on solely minimizing the overheads of real time traffic.

6.6. Simulation Model and Results

Several scenarios have been setup to test the effects of our optimization theory. Values achieved from various schemes are extracted and compared. The network setup is kept consistent in all the scenarios for the purpose of comparisons. Figure 6-3 shows the OPNET network simulation design for all the scenarios in this section. For the purpose of research we have assumed that the WLAN nodes are not mobile hence on one BSS is involved. Power and security issues are not central to this research but would be a good consideration for future work.

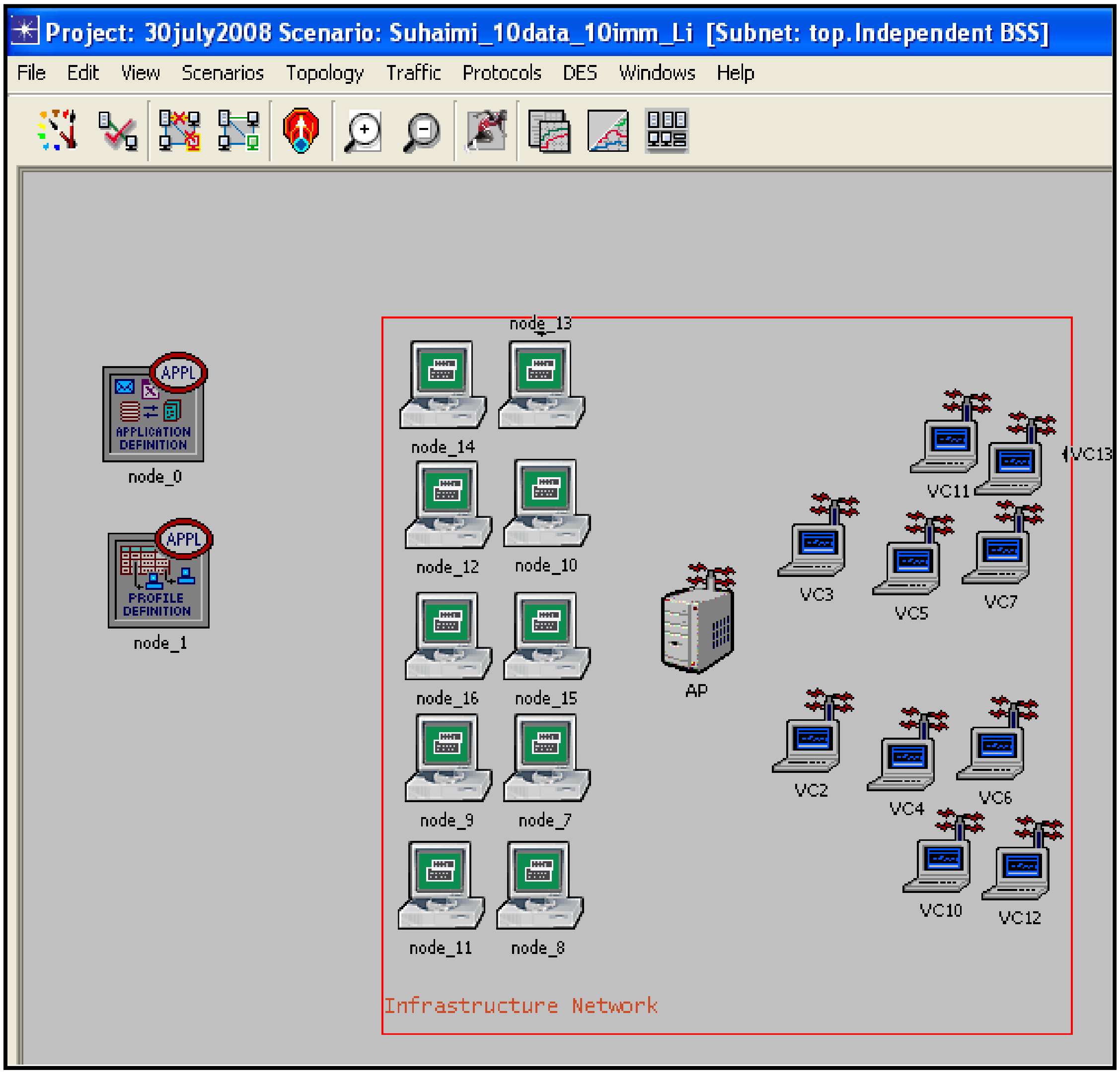
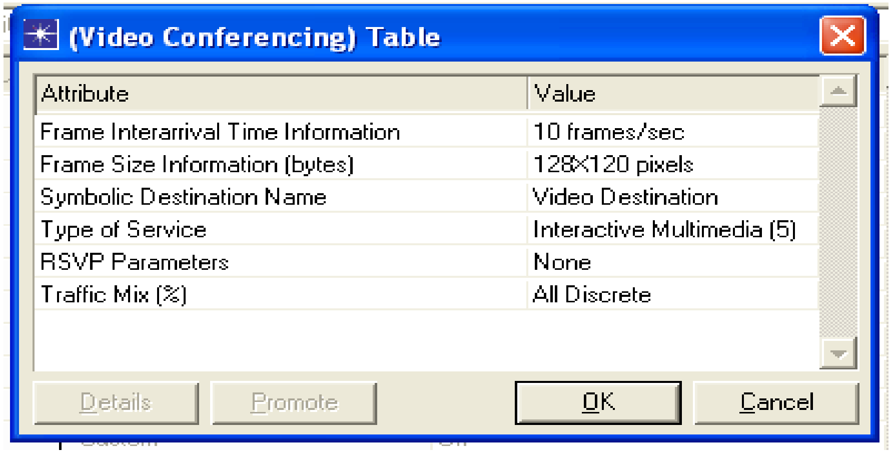


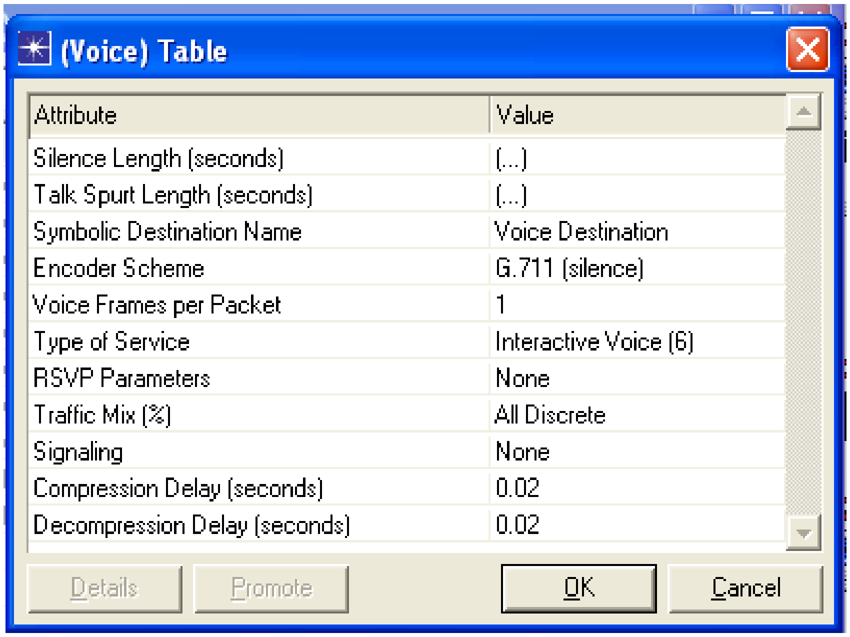
Figure 6-3 Network Setup in OPNET for Performance Study of MDSS

Five WLAN scenarios of 10 IMM and 10 non-IMM nodes each are setup. All of the IMM nodes carry both video and voice traffic. These traffic types are defined as in Figure 6-4 and 6-5 respectively. Default values for the parameters are chosen to represent a typical mixed traffic environment. The non-IMM nodes carry default best-effort data traffic.



Attribute	Value
Frame Interarrival Time Information	10 frames/sec
Frame Size Information (bytes)	128X120 pixels
Symbolic Destination Name	Video Destination
Type of Service	Interactive Multimedia (5)
RSVP Parameters	None
Traffic Mix (%)	All Discrete

Figure 6-4 Video Conferencing Node Parameters



Attribute	Value
Silence Length (seconds)	(...)
Talk Spurt Length (seconds)	(...)
Symbolic Destination Name	Voice Destination
Encoder Scheme	G.711 (silence)
Voice Frames per Packet	1
Type of Service	Interactive Voice (6)
RSVP Parameters	None
Traffic Mix (%)	All Discrete
Signaling	None
Compression Delay (seconds)	0.02
Decompression Delay (seconds)	0.02

Figure 6-5 Voice Node Parameters

A summary of the IMM node and network parameters specified in OPNET is presented in Table 6-4 and 6-5. 802.11b is chosen for its stability and support given OPNET version 11.5. Default CFP/CP duration is 50% with 20 msec SF.

Table 6-6 System Parameters

IMM node	Physical layer descriptions
Video Conferencing: Interarrival rate: 10 frames/sec Frame Size Information (bytes): 128 x 120 pixels	Physical Characteristics: Extended Rate PHY (802.11b)
Voice: Encoder Scheme: G.711 (silence) Voice frames per Packet: 1	Data Rate (bps): 11Mbps
Type of Service: Interactive Voice/Video	Superframe and CFP duration: varying

For comparison purposes, SF and CFP duration obtained for 10 data and 10 non real time nodes were extracted from Table 6-3, 6-4 and 6-5. A scenario with parameters in Table 6-3 for WLAN simulations was setup. Several OPNET simulations were run 30 minutes for each. Probes were used throughout the simulation to gather the needed statistics. The delay budget 150ms was taken as it is the most suitable delay budget for our IMMs applications. Cisco equipments, for example, recommend designing to the ITU standard of 150 msec for multimedia applications to be used with their switches. The QoS metric gathered from simulation runs, shown in Figures 6-6, 6-7 and 6-8 for Haines and Li schemes and also for 802.11e enabled environment.

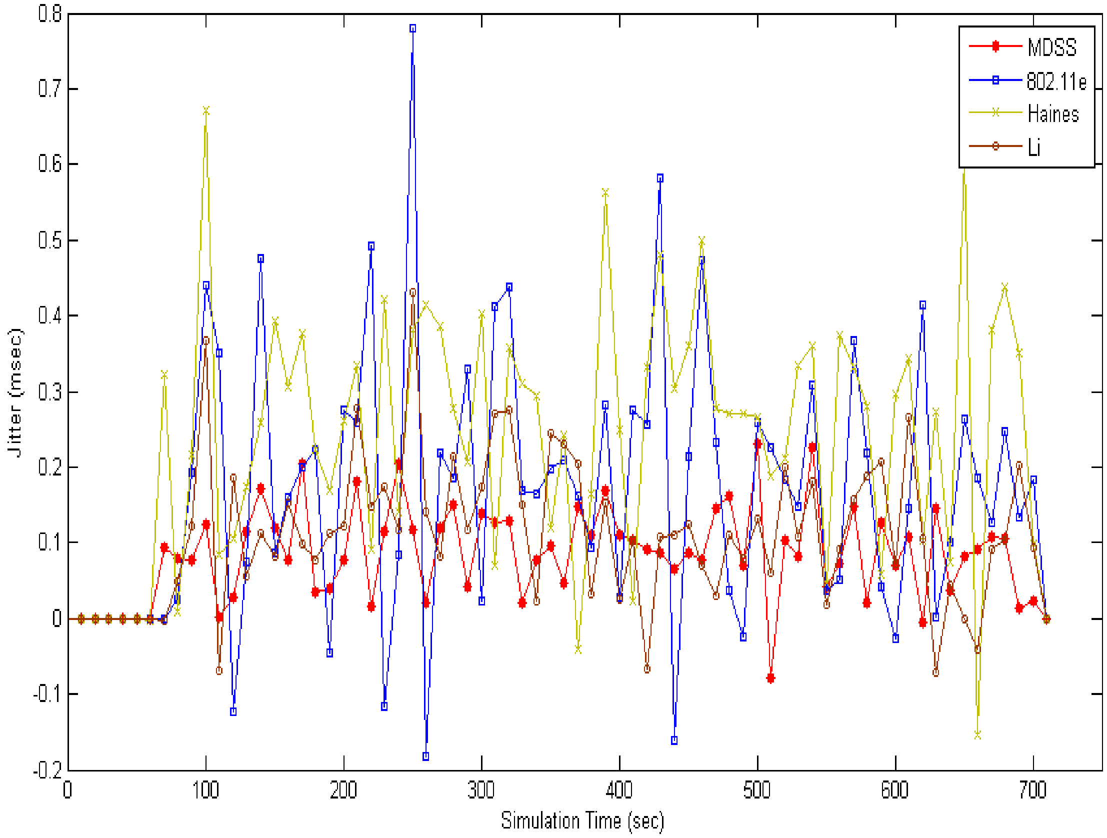


Figure 6-6 Jitter Statistics as a Result of Implementing Various Schemes

Jitter statistics for all the four schemes compared are shown in Figure 6-6. Jitter is critical for most voice applications. All four schemes have successfully capped average jitter well below 30 msec QoS required mark because the network is not at a saturation point. Scheme by Haines produced relatively most jitter at 0.26 msec, followed by 802.11e at 0.18 msec, then Li at 0.12 msec. MDSS produced the least jitter averaging to 0.09 msec. MDSS could bring jitter down by 50% off 802.11e.

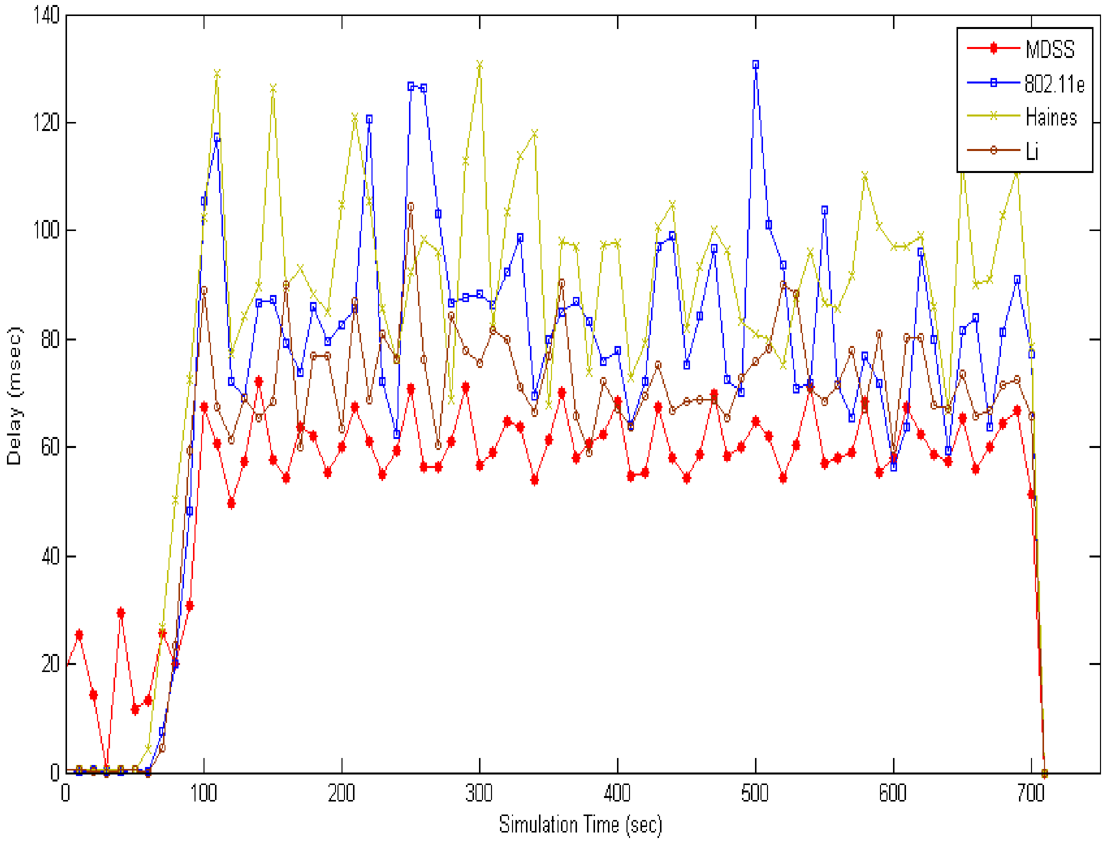


Figure 6-7 Delay Statistics as a Result of Implementing Various Schemes

All schemes managed to keep delay under 150 msec (delay budget set for the optimization constraint) as required by the QoS. Figure 6-7 shows that our scheme gives the least delay at an average of 52 msec, 802.11e gives 69 msec, Li gives 71 msec and Haines with the most delay at 78 msec. MDSS managed to reduce delay further by 25% off 802.11e. Figure has been generated from MS-Excel from 0 to 700sec, hence the sudden drop of statistics to zero beyond the 700sec mark.

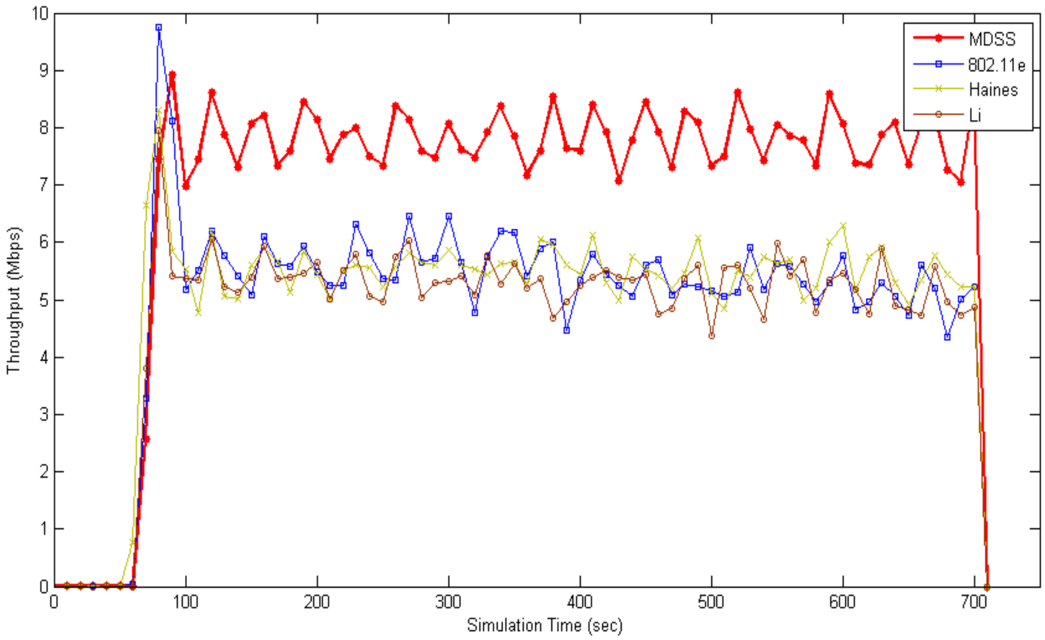


Figure 6-8 Throughput Statistics as a Result of Implementing Various Schemes

Throughput statistics in Figure 6-8 show that MDSS managed to forward the most amount of data i.e., 8 Mbps compared to 5.5 Mbps by all the other three schemes. In OPNET "throughput" represents the total number of bits (in bits/sec) forwarded from wireless LAN layers to higher layers in all WLAN nodes of the network (as explained in Chapter 4.5).

Table 6-7 Comparison Table for Schemes – 10 Non-IMM & 10 IMM Nodes

	Delay (msec)	Jitter (msec)	Throughput (Mbps)
MDSS	52	0.09	7.7
802.11e	69	0.19	5.5
Li	71	0.12	5.3
Haines	78	0.27	5.5

Table 6-7 shows that the performance of our scheme is comparable to that of 802.11e and Li. Jitter and delay for our scheme and 802.11e is relatively lower than Haines'. Our optimization results achieved the highest throughput followed by 802.11e then Haines'. These are results expected of MDSS consistent with what we have established earlier that CFP improves WLAN performance if its duration is between 50% and 100% of SF. MDSS has optimally set the CFP/SF duration according to delay budget required of the most critical traffic traversing on the WLAN while giving a fair treatment to the non-IMM applications. Therefore WLAN performance improves. Haines results also exceed the 150ms delay budget targeted for by the optimization process. In practice if constraints exist and a delay target cannot be met, the delay boundary can be extended to 200 msec without significant impact on voice quality. However, lower jitter is essential to IMM for higher quality output.

6.7. Summary of Model and Results

It can be summarized here that a theoretical model has been developed to model the two phases of WLAN medium access control (MAC) schemes namely the PCF and DCF. With this model an optimum value of the CFP was calculated to meet the QoS requirement of IMM traffic being transmitted. A self-adaptive scheme that generates an optimum SF configuration according to the QoS requirements of traversing IMM traffic at that particular SF is shown to provide a more efficient transmission on WLAN. We have made improvements to available schemes, and proven via simulation that our scheme works better.

Chapter 7
Implementation of MDSS

7.1. Variation of the Models

In the previous chapters, we have developed and tested our scheme on a model suitable for comparison with existing published schemes. In this chapter we will vary the model parameters to represent various scenarios, verifying the results we obtained in previous chapters. We will discuss necessary adjustments to the scheme in order to implement it on a real network.

The following variations were introduced into the models and simulated with a setup as shown in Figure 7-1.

1. Models using fixed number of non-IMM and increasing number of IMM nodes with implementation of the MDSS scheme
2. Models with fixed number of non-IMM and IMM nodes with implementations of MDSS, 802.11e, Li and Haines schemes.

These model variations were made to validate the performance of MDSS with respect to the existing schemes. In the first scenario, the numbers of non-IMM nodes were fixed while numbers of IMM nodes were increased until the resulting jitter and delay of the WLAN get very large. Here we can see the IMM and non-IMM combinations that MDSS could manage to keep the QoS requirements within the required level. Then following this, in the next scenario, we take the IMM and non-IMM combination and apply all the other existing schemes. Their performances are analyzed. Homogeneous WLAN scenarios were also simulated to analyse the performance of MDSS in a purely non-IMM and a purely IMM network. The simulation results show that MDSS performs comparatively similar to 802.11e and better than the other schemes.

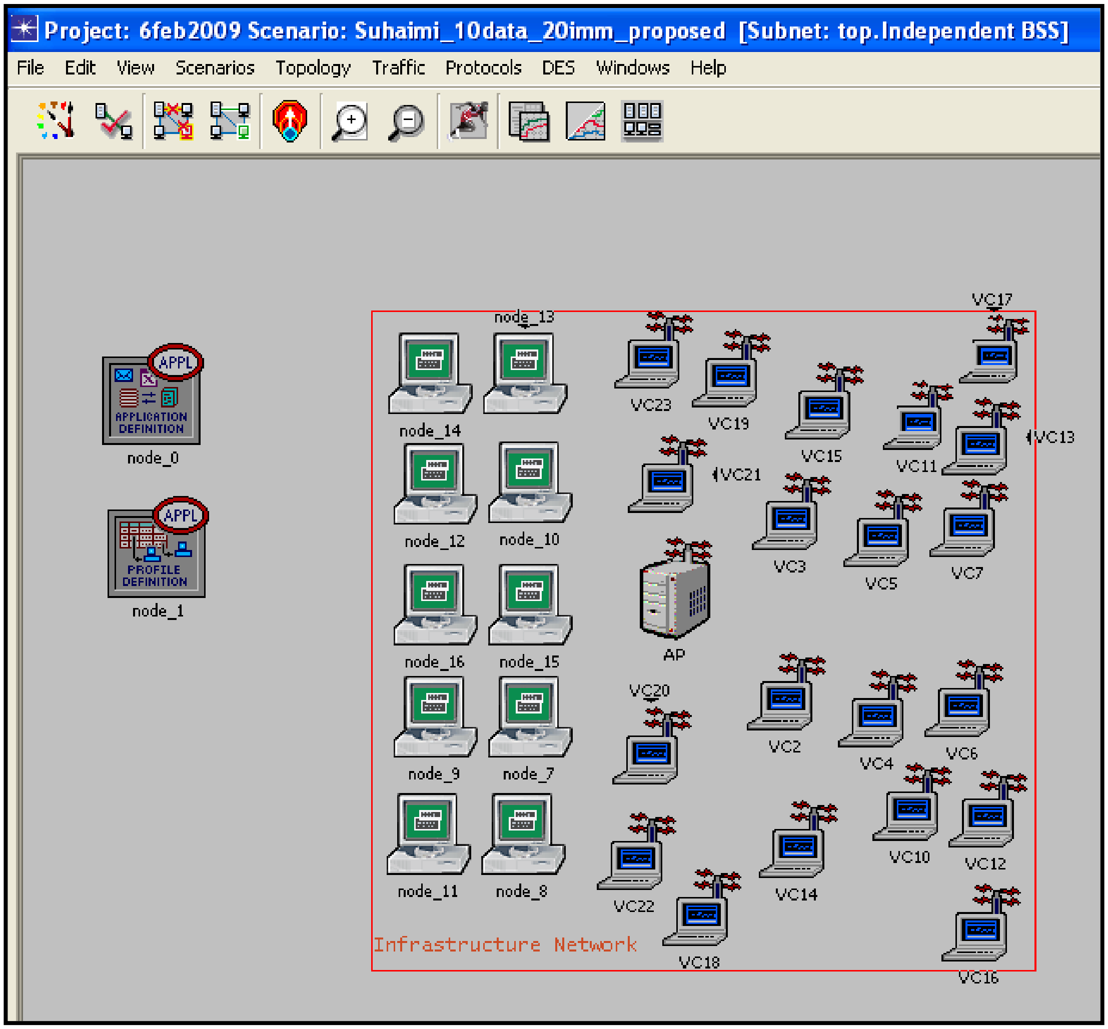


Figure 7-1 IMM and non-IMM scenario setup in WLAN

7.2. Models with Fixed Number of non-IMM and Increasing Number of IMM Nodes

In this scenario, 10 non-IMM applications were simulated into the heterogeneous network with 10, 12, 14, 16, 18 and 20 IMM nodes. Essential QoS metrics were collected. The scenarios were simulated for 700 seconds each, although it takes about 100 seconds for the WLAN to populate. IMM applications delay budget was set to

150ms. The OPNET setup of Figure 7-1 with the voice and video terminal attributes shown in Figure 6-4 and 6-5 of Chapter 6 were used.

3.1.1. Results of Simulation Model

Figures 7-2 to 7-4 clearly show the respond of jitter, delay and throughput to the increasing number of nodes in the WLAN. These results validate the MDSS scheme.

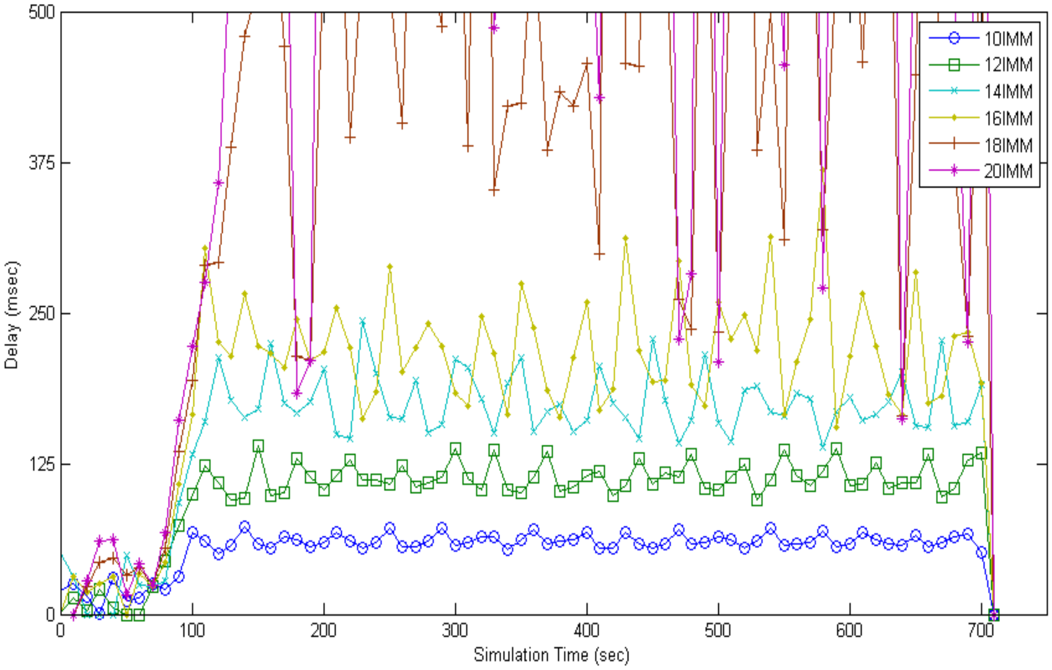


Figure 7-2 Delay Statistics as a Result of Increasing Number of IMM Nodes with MDSS

Delay statistics shown in Figure 7-2, show increasing delay with more IMM nodes added into the network. 10 IMM nodes give 52 msec (as reported in Chapter 6.6). 12 IMM nodes give 110 msec, 14 IMM nodes give 155 msec and more than that gives unacceptable delay exceeding the QoS requirements.

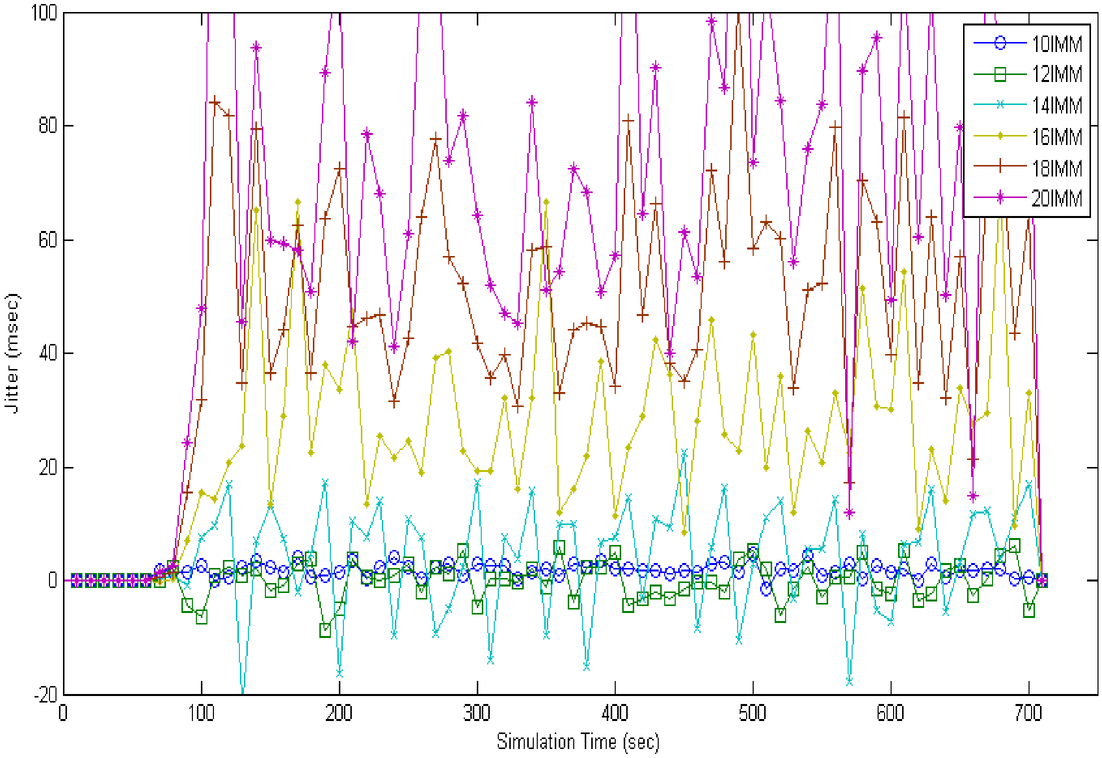


Figure 7-3 Jitter Statistics as a Result of Increasing Number of IMM Nodes with MDSS

In Figure 7.3, jitter for 10 IMM nodes is 9 msec, 10 msec for 12 IMM nodes, 15 msec for 14 IMM nodes, 40 msec for 16 or more IMM nodes and more for 18 and 20 IMM nodes. Jitter is unacceptable with 16 IMM nodes in WLANs.

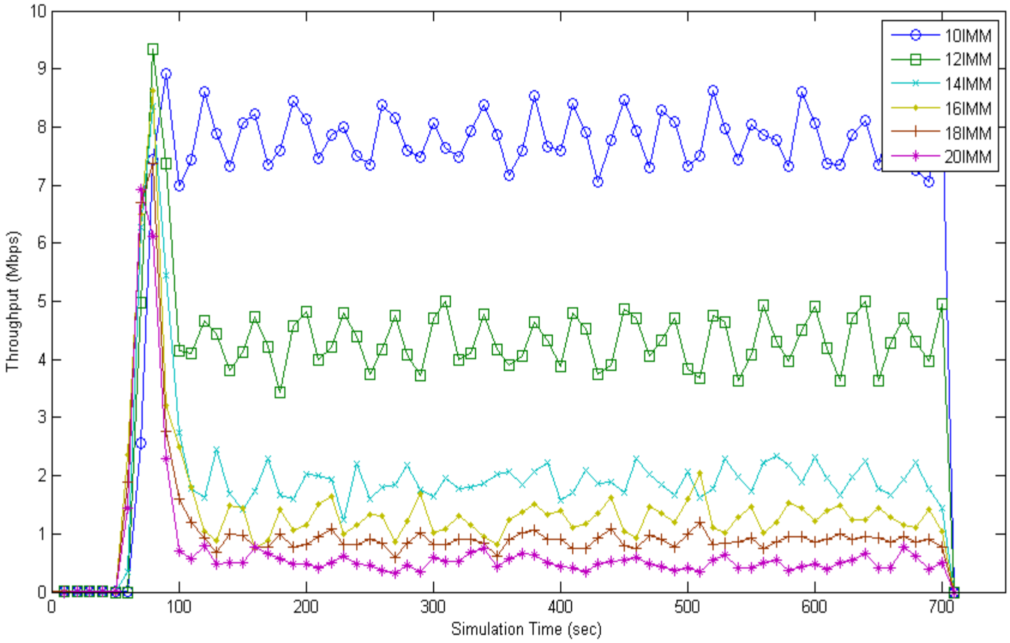


Figure 7-4 Throughput Statistics as a Result of Increasing Number of IMM Nodes with MDSS

Throughput statistics in Figure 7-4 re-affirms the other two statistics obtained where performance of the scheme deteriorates with 16 and more IMM nodes in the WLAN (including 10 data nodes). Therefore, MDSS can maintain a good QoS provision with a mix of up to 50% more IMM nodes to non-IMM nodes.

On this note, the next section will further analyze the performances of the other three schemes namely 802.11e, Haines and Li with 10 non-IMM and 14 IMM mix in the network.

7.3. Models with Fixed Number of non-IMM and IMM Nodes

10 non-IMM applications were assumed active in the heterogeneous network with 14 IMM nodes available. Essential QoS metrics were collected. The scenarios were simulated for 700 seconds, as previous simulations. IMM applications delay budget was

set to 150ms. The OPNET setup can be seen as in Figure 7-1 while the voice and video terminal attributes shown in Figure 6-4 and 6-5 of Chapter 6 were used.

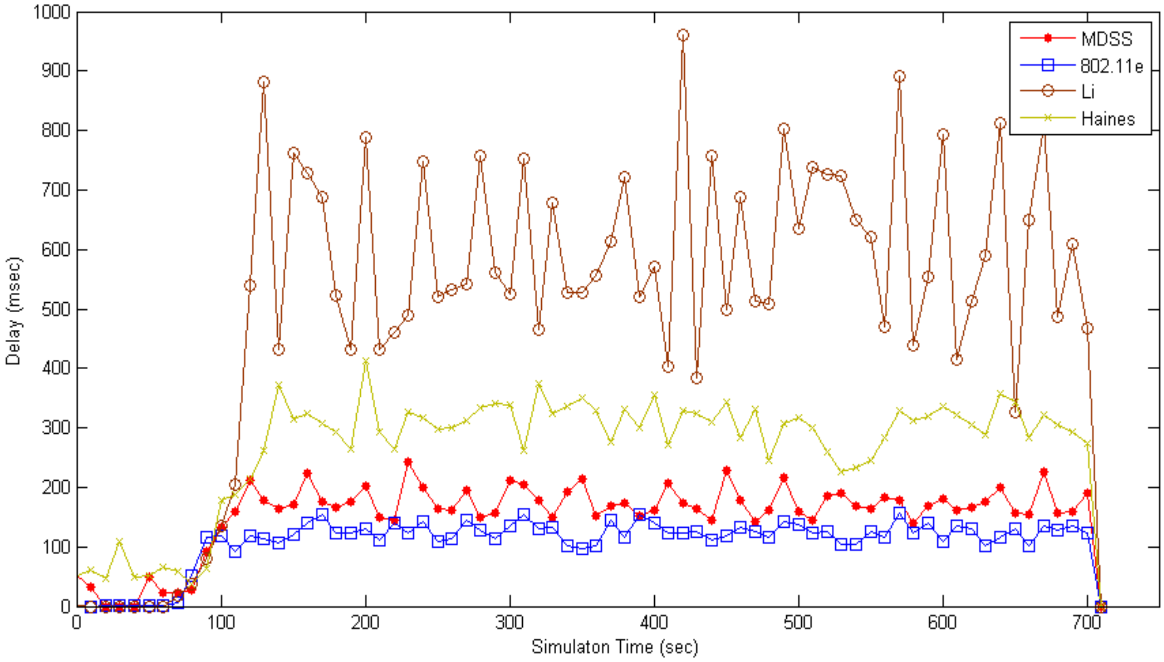


Figure 7-5 Delay Statistics as a Result of 14 IMM Nodes Implementing MDSS, 802.11e, Li and Haines Schemes

7.3.1. Results of Simulation Model

Figure 7-5 clearly shows that Li scheme goes beyond the required delay budget of 150 msec with 14 IMM nodes in the WLAN. Haines scheme, with the optimization scheme implemented, gives an average delay of 285 msec. MDSS barely meets the delay requirements at 155 msec. 802.11e is the only scheme out of the four to give an average delay lesser than the required delay budget of 117 msec.

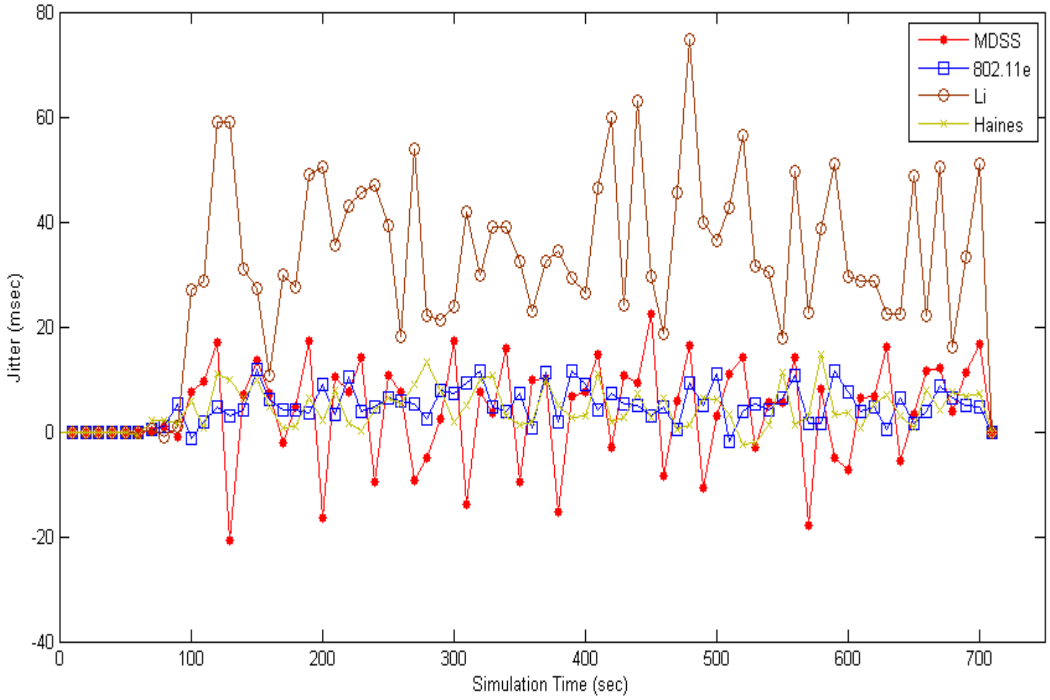


Figure 7-6 Jitter Statistics as a Result of 14 IMM Nodes Implementing MDSS, 802.11e, Li and Haines Schemes

Jitter statistics for 10 non-IMM and 14 IMM nodes depicted in Figure 7-6, shows that only Li scheme went beyond the 40 msec mark. The other three schemes including MDSS perform well in this front at below 10 msec.

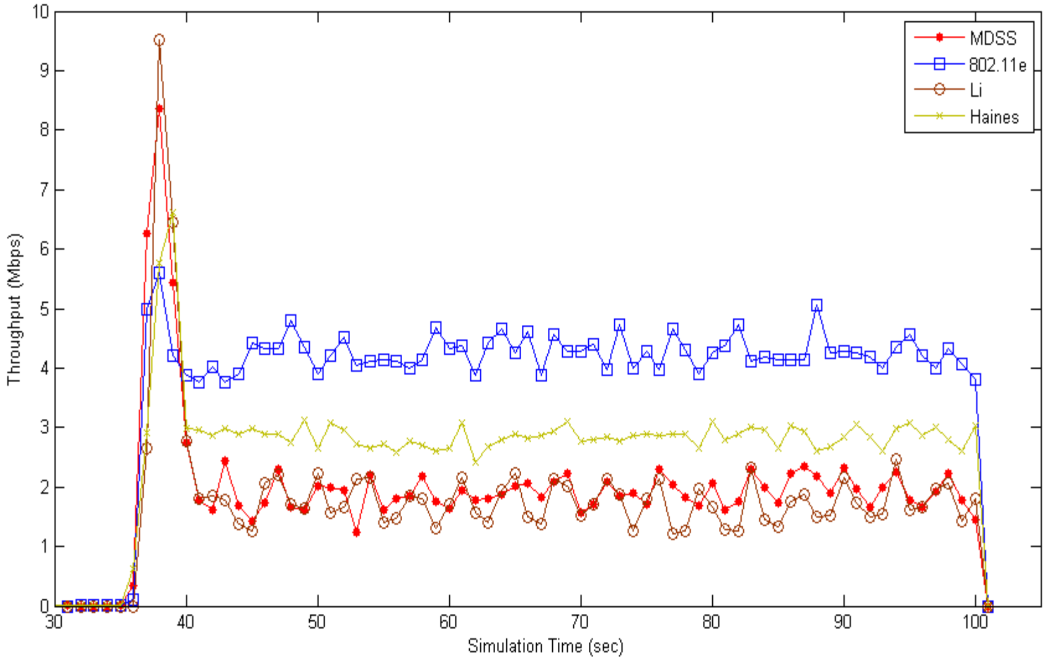


Figure 7-7 Throughput Statistics as a Result of 14 IMM Nodes Implementing MDSS, 802.11e, Li and Haines Schemes

Figure 7-7 shows the throughput statistics for the 10 non-IMM and 14 IMM nodes network. 802.11e has the highest throughput at 4.1 Mbps, Haines follows with 2.8 Mbps, MDSS with 2.0 Mbps and finally Li with 1.9 Mbps.

7.4. Homogeneous WLAN Setup with Only non-IMM nodes

10 non IMM applications were modelled into the heterogeneous network with no IMM available. Essential QoS metrics were observed. The scenarios were simulated for 60 minutes each. IMM applications delay budget was set to 150ms. The proposed schemes were compared to 802.11e with AP setup shown in Figure 7-2. The WLAN scenario setup was shown in Figure 7-1 while the voice and video terminal attributes shown in

Figure 6-4 and 6-5 respectively. Delay and throughput statistics of the simulation results are shown in Figure 7.8 and 7.9.

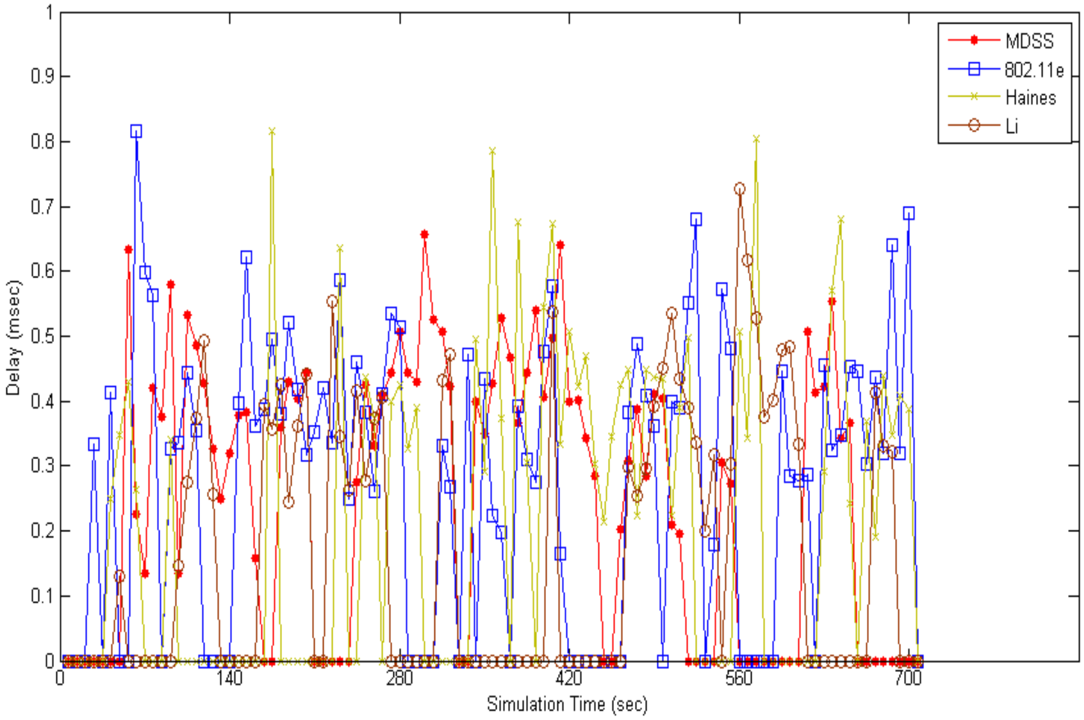


Figure 7-8 Delay Statistics as a Result of 0 IMM Nodes

7.4.1. Results of Simulation Model

Following delay statistics in Figure 7-8, we note that in a homogeneous WLAN with purely non-IMM nodes, the average delay is a mere 0.5 msec, which is very negligible. Jitter statistics is not available as no IMM nodes are considered in the WLAN.

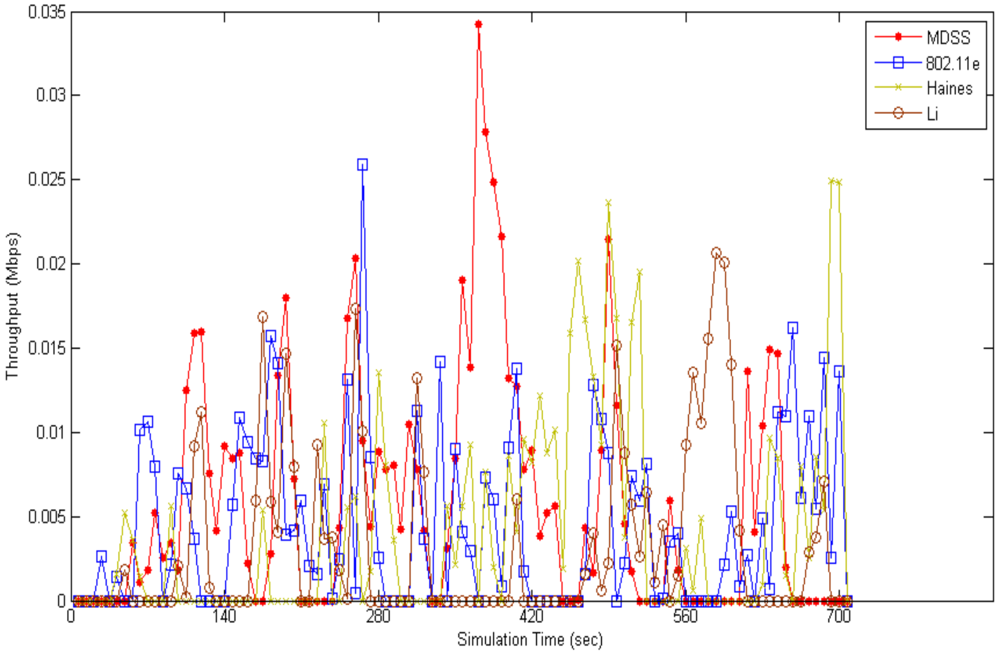


Figure 7-9 Throughput Statistics as a Result of 0 IMM Nodes

Throughput is fairly equal amongst the 4 schemes as shown in Figure 7-9. These statistics validate the impact of IMM nodes in the WLAN.

7.5. Homogeneous WLAN Setup with Only IMM nodes

10 IMM applications were simulated next into the heterogeneous network with no non-IMM available. Essential QoS metrics were observed. The scenarios were simulated for 700 seconds each. IMM applications delay budget was set to 150ms. The proposed schemes were compared to 802.11e with AP setup shown in Figure 7-2. The WLAN scenario setup was shown in Figure 7-1 while the voice and video terminal attributes shown in Figure 6-4 and 6-5 respectively. Delay and throughput statistics of the simulation results are shown in Figure 7.8 and 7.9. The simulation results showing jitter, delay and throughput statistics are shown in Figures 7.10, 7.11 and 7.12.

7.5.1. Results of Simulation Model

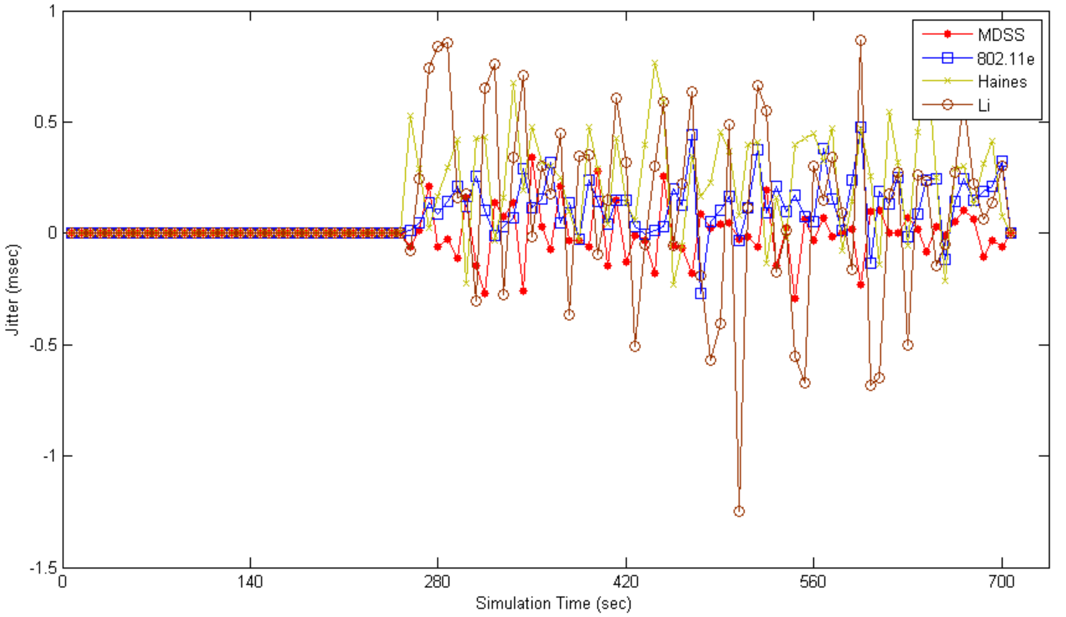


Figure 7-10 Jitter Statistics as a Result of Exclusively 10 IMM Nodes

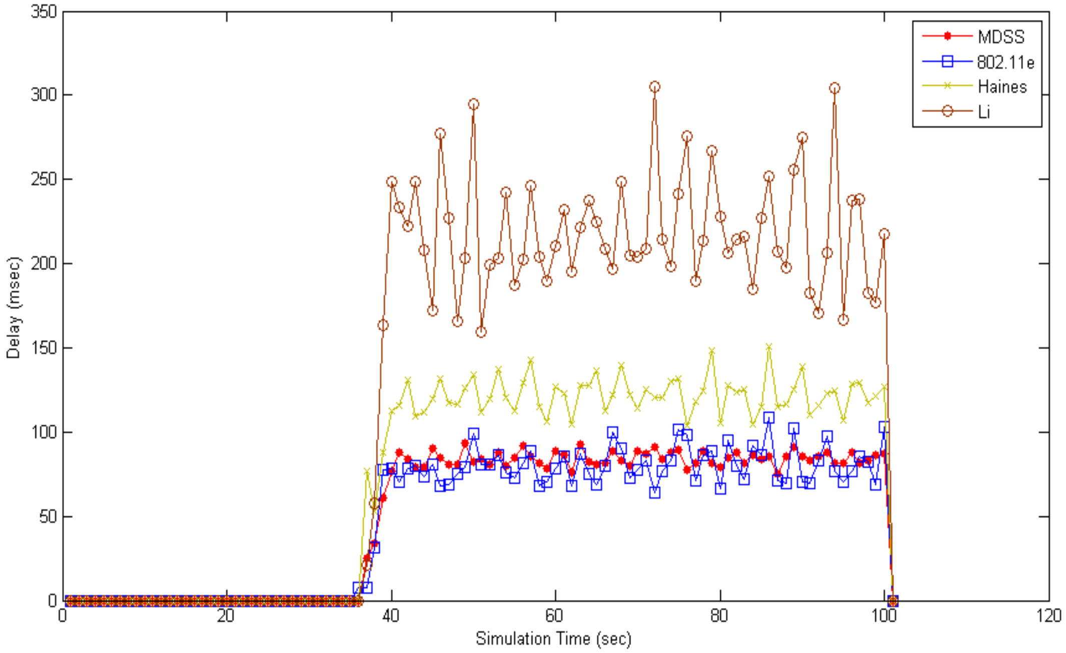


Figure 7-11 Delay Statistics as a Result of Exclusively 10 IMM Nodes

Jitter statistics in Figure 7-10 shows arbitrarily small jitter generated by all four schemes. Delays statistics in Figure 7-11 re-affirms the performance of MDSS in transmitting IMM traffic in WLAN. MDSS scheme gives a delay of 82 msec, 802.11e with 80 msec, Haines gives 120 msec and Li a staggering 212 msec which is beyond the target delay budget.

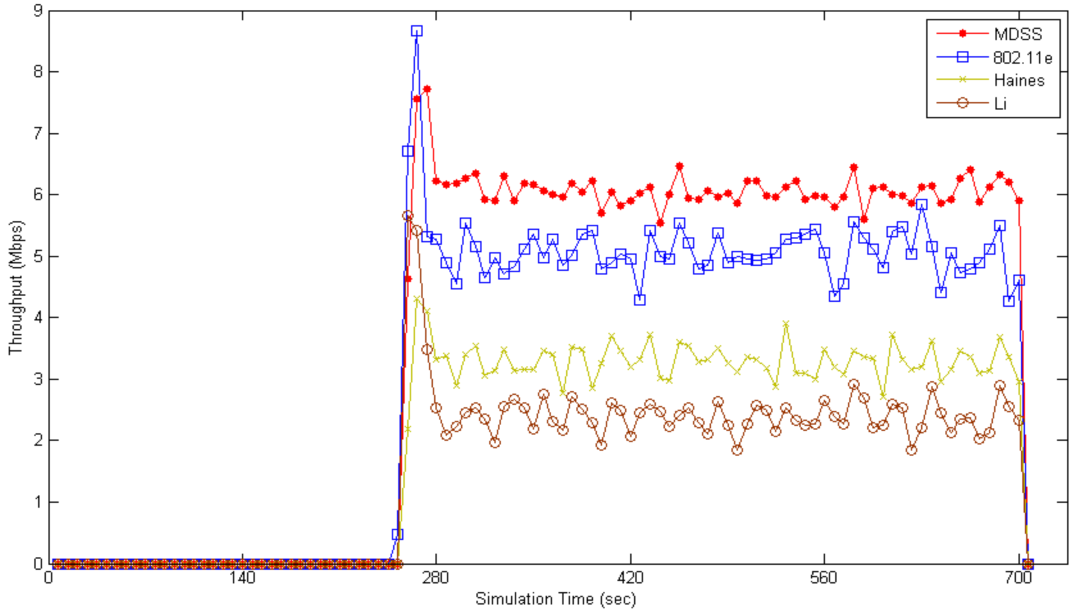


Figure 7-12 Throughput Statistics as a Result of Exclusively 10 IMM Nodes

In the absence of data traffic in the WLAN, MDSS gives a throughput of about 6 Mbps as shown in Figure 7-12. 802.11e gives 5.1 Mbps, Haines gives 3.2 Mbps and Li the lowest at 2.4 Mbps.

7.6. Summary of the Results

The average values of all data collected in Figures 7-5 to 7-12 are presented in Tables 7-1 to 7-3.

Table 7-1 Comparison Table for Schemes – 10 Non-IMM And 14 IMM Nodes

	Delay (msec)	Jitter (msec)	Throughput (Mbps)
MDSS	155	<10	2.0
802.11e	117	<10	4.1
Li	700	40	1.9
Haines	285	<10	2.8

Table 7-2 Comparison Table for Schemes – 10 non-IMM and 0 IMM Nodes

	Delay (msec)	Jitter (msec)	Throughput (Mbps)
MDSS	0.26	n/a	0.02
802.11e	0.28	n/a	0.015
Li	0.18	n/a	0.009
Haines	0.22	n/a	0.01

Table 7-3 Comparison Table for Schemes – 0 non-IMM and 10 IMM Nodes

	Delay (msec)	Jitter (msec)	Throughput (Mbps)
MDSS	82	0.128	6.07
802.11e	80	0.131	5.12
Li	212	0.13	2.49
Haines	120	0.257	3.29

It can be concluded here that MDSS has performed well in both homogeneous and heterogeneous scenarios, as can be seen from the results in Table 7-1 and 7-2 and 7-3 respectively. Delay profile for MDSS and 802.11e can be seen as comparable in homogeneous scenario although MDSS delivers relatively higher throughput as compared to 802.11e. MDSS has also improved results obtained its predecessors Li and Haines.

Chapter 8
Conclusions and Directions for Future Works

The growth of communicating high volume of real time multimedia traffic over high-speed wireless networks is one of the major driving forces behind the evolution on wireless network. Ubiquitous access to the network and internet with minimum delay is becoming crucial every day. Published researches on WLANs and the approval of the 802.11e provide a scope to improve the transmission of IMM traffic in WLAN.

Most researchers focus on improving the functionalities in WLANs by working on the DCF. We have demonstrated that there is a scope of improving the WLAN QoS in IMM mixed traffic environment via improvement of the legacy PCF. It is well recognized that the PCF was developed to cater for real time traffic in WLAN. Its implementations have been deterred due to various performance inefficiencies such as the overheads in polling. This work has investigated extensively the results of published research on QoS provisioning on high-speed wireless networks and developed, modelled and performed performance studies on a proposed access scheme that improves the transmission of real time multimedia application on WLAN.

8.1. Research Contributions

This thesis proposes a solution to improve efficient communication of IMM traffic on WLAN by optimizing the duration utilized by IMM i.e., the CFP duration while providing fair resource to the traditional data traffic. Treating IMM traffic as data traffic or equating both IMM and non-IMM together could compromise a fair treatment that should be given to these QoS sensitive traffic. Through the course of this thesis various tools have been discussed and developed to show that with significant improvements in the PCF, more reliable and efficient transmission of IMM in a mixed traffic scenario in WLAN could be made.

A channel access protocol called the MAC with Dynamic Superframe Selection (MDSS), for IMM communication over WLAN under heterogeneous or mixed traffic scenario has been proposed, by leveraging the inherent advantages of both the contention based and the polling schemes. This was achieved by balancing the contention and contention free phases of the super frame. Various critical QoS metrics such as throughput, delay and jitter was taken into consideration during the development of the protocol. Performance improvements over current schemes in terms of these QoS metrics for mixed traffic environment through extensive simulation have been constructed.

A comprehensive review of published literature established the fact that solutions to the improvement of IMM transmission over WLAN are concentrated on the DCF scheme. Time sensitive IMM applications are treated as data and consequently are forced to compete for resources in the shared medium. This provided a scope in providing a solution to QoS provisioning to IMM transmission in WLAN via scheduling and polling scheme in PCF. There are several researchers that have looked into this solution; however, most have not considered the mixed traffic conditions, hence again treating IMM traffic like any other data traffic. Furthermore, in reality communication on the network and internet is of heterogeneous in nature not homogeneous. The large number of research on improving 802.11e shows that the scheme is far from being settled. The proposed scheme is developed from legacy 802.11 hence relatively simpler to implement and would be a complementary technique to 802.11e in improving IMM transmission in WLAN.

Simulation models were developed to test the potentials of PCF in providing QoS for IMM traffic. QoS statistics for an arbitrary number of wireless nodes with half of them transmitting IMM traffic were collected and analysed. The jitter and delay experienced by PCF-enabled nodes were observed to be lower than DCF-enabled nodes. Variation of the durations for CFP, CP and SF was also simulated and results show that variation of duration does impact the QoS provisioning. This is consistent with our original hypothesis that SF configuration could present a considerable impact on the QoS delivery of the WLAN. SF duration is found to be proportionally close to the delay requirements of the WLAN. Therefore, for every IMM/non-IMM combination there is an appropriate CFP/CF duration with respect to the QoS requirements and number of IMM traffic traversing the WLAN.

The two phases of WLAN MAC schemes, namely the PCF and DCF was modelled according to their bandwidth utilization. This model maximizes the CFP duration bounded by a number of constraints including the CP duration. This way IMM application will be given priority while providing fairness to the non-IMM traffic. With this model an optimum set of values of the CFP could be automatically generated to meet the QoS requirement of IMM traffic being transmitted. This self-adaptive scheme, called MAC with Dynamic Superframe Selection (MDSS) scheme, generates an optimum SF configuration according to the QoS requirements of traversing IMM traffic. That particular SF is shown to provide a more efficient transmission, with respect to the QoS metrics, on WLAN. Results have shown that MDSS brings jitter and delay down by 50% and 25% off standard scheme, respectively. MDSS also forwards the most amounts of data hence achieving highest throughput compared to other schemes. Its performance is comparable to that of 802.11e, which is taken as the benchmark for comparison. In providing optimum bandwidth to IMM applications, the non-IMM applications have not been starved. We have achieved considerable improvements to available schemes, and proven via simulation that our scheme works better.

The 802.11e scheme was taken as the performance benchmark for the MDSS. The implementation of MDSS with varying scenario shows that it performed comparatively similar to 802.11e. Given the complexity of 802.11e and the major task of backward compliancy to the legacy protocol, the MDSS is a alternative scheme which gives simpler implementation with comparable outcome.

Given the numerous researches on the problems of DCF carrying IMM traffic seen in published literatures, the general direction of DCF improvements shows that researchers are bringing DCF closer in functionality to PCF. An obvious emulation of the PCF functionality in DCF is in the implementation of 802.11e. The IEEE guideline in handling IMM traffic is still inadequate in fulfilling the QoS requirements of IMM traffic in a heterogeneous network. This is proven by the number of researches published trying to rectify the weakness of the 802.11e scheme. The more settled 802.11 with PCF which was created to handle IMM traffic, has its development and implementation daunted by the overheads posed by the polling mechanisms. We have proven that improving the polling mechanisms in the WLAN MAC gives a relatively cheaper solution compared to the 802.11e scheme.

Implementing this scheme in the AP could be made with simple modification of the legacy MAC. The dynamic referencing of the optimization engine in configuring the SF size could be done within or outside the AP. The MDSS scheme will auto-generate a lookup table according to the QoS requirements of the prevailing IMM traffic at a particular SF. The point coordinator (which resides in the AP) will refer to this lookup table to decide on a suitable SF configuration and adjust it accordingly. The advantage of implementing the MDSS outside the AP is that it saves memory resources although advanced technology in electronics makes memory allocation issues in computers a minor one.

8.2. Directions for Future Works

As with any form of research answering one question typically generates many more unanswered questions. The development work that goes into answering the original questions is, never finished and can always be improved upon. To that end, the continuous improvement of MDSS is imperative to the success of IMM transmission in WLANs.

Further reduction in bandwidth cost, delay and jitter could become technically feasible by further defining the overheads during CP and CFP. A detailed mathematical model of each phase outlining more possible overheads that could represent wastages would be useful in the optimization process. Various optimization techniques including neural networks and the Sequential Linear-Quadratic Programming (SLQP) could be employed to further support the model.

Cross layer optimization technique, as discussed in section 2.4, is probably the next step for the improvements of this protocol. Cross layer strategies can be employed with the MDSS if the scheme is executed outside the AP. Cross layer tunnelling is a viable candidate for the implementation strategy. Full network design and optimization could be a challenge. This is particularly true when attempting a real time self-adaptive optimization. Further research may be initiated to determine the design methodologies that encompass flexibility for cross-layer optimization. The metrics used in the optimization should accommodate the individual optimization criteria of various network layers. For example, the physical-layer is primarily focused on minimizing the bit error-rate, MAC layer is concerned with node throughput or channel availability and network layer, delay or routing efficiency. A set of metric that represents all these concerns will be a major challenge to uptake. In the context of dynamic optimization, information passed between the network layers, should not create extra delay and large computational costs and also should not be too simple to have too little information.

An extension to this research would be the active information exchange between the PHY and MAC layer. For example, it would be interesting to assess the effects of MDSS over multiple-input and multiple-output (MIMO) wireless links. Synergizing the MAC mechanisms control, at the link layer and MIMO at the physical layer provides a scope in a novel cross layer protocol to improve the IMM transmission by making use of the multi-channel transmission features. Considerably more work will need to be done to determine the effects of cross-layer feedback between the PHY and MAC layer.

Improvements of optimization method in the analytical model could be explored further. Constraints depicting the requirements of IMM traffic flows could further be defined in the model. More information on factors contributing to the objective function would help to establish a greater degree of accuracy on this matter.

Further scrutiny of the polling scheme could enhance the MDSS scheme in transmitting IMM traffic. For example, a scheme that uses a multi-level polling mechanism with the QoS classes differentiated would be worth exploring. Stations with higher priority traffic form the highest level to access the wireless channel through an appropriately defined policy followed by other lesser priority traffic.

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