ADAPTIVE CHANNEL-SUPERFRAME ALLOCATION (ACSA) FOR 60 GHZ WIRELESS NETWORKS

by

Majid Veyseh
B.Sc., Amirkabir University of Technology, Tehran, Iran, 2007

A thesis
presented to Ryerson University
in partial fulfilment of the
requirement for the degree of
Master of Applied Science
in the program of
Computer Networks

Toronto, Ontario, Canada, 2009

Author's Declaration

I hereby declare that I am the sole author of this thesis or dissertation.

I authorize Ryerson University to lend this thesis or dissertation to other institutions or individuals for the purpose of scholarly research.

Signature

I further authorize Ryerson University to reproduce this thesis or dissertation by photocopying or by other means, in total or in part, at the request of other institutions or individuals for the purpose of scholarly research.

Signature

Abstract

Adaptive Channel-Superframe Allocation (ACSA) for 60 GHz Wireless Networks

©Majid Veyseh 2009

Master of Applied Science
Computer Networks
Ryerson University

Millimeter-wave (MMW) systems are high frequency wireless systems with a center frequency of around 60 GHz. This thesis deals with adaptive channel-superframe allocation (ACSA) for such system. An adaptive bandwidth or channel allocation algorithm is utilized in the piconet controller (PNC) and a new superframe structure is designed in order to distribute bandwidth among real-time (RT) and non-real-time (NRT) flows. We propose to serve RT and NRT flows separately in different channels instead of serving them in different times. We also propose to change the sliced superframe of IEEE 802.15.3 to an adaptive unsliced superframe in order to decrease the TCP round-trip time. We simulated a MMW system with appropriate parameters using 802.15.3 MAC as well as ACSA MAC. We measured three performance metrics (throughput, delay and fairness), which we aimed to improve in our superframe design. The simulation results show that the adaptive superframe structure could provide throughput improvements not only for NRT flows, but also for RT flows. The control algorithm in PNC could manage the bandwidth allocation in superframe and improve the throughput of RT flows. The channel access delay is improved by providing an unsliced superframe, which eliminated an imposed delay on TCP connections. Finally, the better distribution of bandwidth in ACSA MAC improves the fairness of the system. As a brief, the simulation results support the analysis of the proposed adaptive channel-superframe allocation algorithm, which could generally improve the quality of service for MMW systems.

Acknowledgment

I would like to thank my supervisor Dr. Alagan Anpalagan for his time, intellectual mind, the excellent guidance and continuous support throughout my research.

I would also like to thank Dr. Boby Ma who gave me the honor to pursue my Master's degree at Computer Networks program at Ryerson University. I benefited from the perfect infrastructure, the nice working atmosphere, and the large experience and knowledge gathered there, which all served as a stable basis for my scientific research.

I would also like to acknowledge the Computer Networks Department and the School of Graduate Studies at Ryerson University for their financial support and providing me the opportunity to work as a teacher assistant.

In addition, I would like to thank the members of WAN (Wireless Access and Networking) Research Group. We had such a friendly and peaceful work environment together.

Last but not least, I would like to thank my family, without whom I would never have been able to achieve my longtime goals.

Contents

List of Figures

List of Tables

Nomenclature

3G	3rd Generation
ACK	Acknowledgment
ACSA	Adaptive Channel-Superframe Allocation
ADD	Automatic Device Discovery
BER	Bit Error Rate
BIFS	Back-off Inter-Frame Space
BP	Beacon Period
BPSK	Binary Phase Shift Keying
CAP	Contention Access Period
CSMA/CA	Carrier Sense Multiple Access With Collision Avoidance
CTA	Channel Time Allocation
CTAP	Channel Time Allocation Period
CTS	Clear to Send
DC-OFDM	Dual Carrier-Orthogonal Frequency Division Multiplexing
DMAC	Directional MAC
EHF	Extremely High Frequency
FTP	File Transfer Protocol
HDTV	High-Definition Television
HTTP	Hypertext Transfer Protocol
IP	Internet Protocol

JFR	Job Failure Rate
LOS	Line of Sight
MAC	Medium Access Control
MBS	Memorized Back-off Scheme
MBS	Mobile Broadband System
MMW	Millimeter-Wave
MPDU	MAC Protocol Data Unit
MSDU	MAC Service Data Unit
NRT	Non-Real-Time
NS2	Network Simulator 2
OOK	On-Off Keying
PDX	Polarization Diversity Extension
PHY	PHY Protocol Data Unit
PHY	Physical
PNC	Piconet Coordinator
PSNR	Peak Signal Power to Noise Ratio
PTDBS	Pseudo-Time Division Backoff Scheme
RMS	Root-Mean-Square
RT	Real-Time
RTS	Request to Send
RTT	Round-Trip Time
SIFS	Short Inter-Frame Space
SNR	Signal to Noise Ratio
SUBAA	Short Unique Back-off Allocation Algorithm
SYN	Synchronization Packet

TCP	Transmission Control Protocol
TDMA	Time Division Multiple Access
UMTS	Universal Mobile Telecommunications System
UWB	Ultra Wide Band
WLAN	Wireless Local Area Network
WPAN	Wireless Personal Area Networks

Chapter 1

Introduction

The new trend of home/office networking is expected to migrate to very high speed wireless communications and 60 GHz wireless systems could support the required bandwidth requirements. 60 GHz frequency band is unlicensed and is too wide. It starts from 57GHz and ends at 66GHz. This band has a wavelength of ten to one millimeter, giving it the name millimeter band or millimeter wave, sometimes called MMW. Compared to lower bands, terrestrial radio signals in this band are extremely prone to atmospheric attenuation, making them of very little use over long distances. In this thesis we elaborate some important characteristics of 60 GHz signals and explain the reasons for its popularity. Both physical layer and MAC layer of MMW system are subject to further study and research. The possible applications of MMW systems are in WLAN, mobile networks and fixed wireless networks. These applications are taking advantage of the high rate of transmission and new features in MMW systems, which do not exist in legacy 802.11 wireless networks. Based on that, some particular functions are added to the MAC layer of Ultra Wide Band (UWB) and MMW, which was not a part of MAC responsibility in 802.11 wireless systems. Thus, the MAC layer is such an important part of design in UWB and MMW systems

Considerable number of studies are published which address the design issues of MMW radio systems. The IEEE 802.3.15c draft version is the published IEEE standard for MMW physical layer, but there is not a published standard for MMW MAC. However, different studies reveal that 60 GHz MAC could be designed based on UWB MAC (IEEE 802.15.3) [7] [12] [13]. They explain that the basic design of superframe and functions of piconet controller of UWB systems could be used without any changes in MMW systems, yet system throughput, device discovery, hidden terminal problem, directional transmission and efficiency are still potential research areas.

Therefore, in this study we direct our research toward MAC layer issues since all the mentioned potential research areas are related to MAC layer functions. We converge some of those issues in efficiency (such as throughput, fairness and delay) into one problem and provide a system model to address the shortcomings. Then we propose a superframe structure controlled by an adaptive channel-superframe allocation (ACSA) algorithm in MAC layer.

In this thesis, we present the following contributions to improve the performance of a MMW system, and to quantify the effectiveness of the proposed ACSA algorithm. In particular, we

- Developed a novel structure for superframe of MMW systems.
- Proposed an adaptive channel-superframe allocation algorithm (ACSA), which adaptively associates channels and superframes to different flows based on the type of the flows in order to improve the performance in terms of throughput, fairness and delay.
- Evaluated the performance of the proposed algorithm, and compared the results with UWB MAC (IEEE 802.15.3 MAC) with various number of flows and superframe length and data rates.

The thesis is organized as follows.
In **Chapter 2**, we introduce the 60 GHz system and signal characteristics in this band. Moreover, we explain possible applications of a MMW system and the functions of IEEE 802.15.3 MAC layer. Following the general information on UWB and 60 GHz systems, a literature survey is provided in **Chapter 3**. In this section, a summary of previous studies on MMW MAC layer is provided. We categorized the studies into sections and each section contains a summary of related studies claiming to improve one of the performance metrics such as throughput, delay and fairness. In **Chapter 4**, the novel channel-superframe allocation algorithm is presented for MMW MAC. We analyzed the performance of UWB MAC layer and discovered the possible issues which may rise if it would be used for MMW systems. Then, description of necessary modifications on UWB MAC is provided to make it a potential MAC layer for MMW systems. Consideration on the bandwidth requirements of wireless applications is an important part of this modification. The system parameters and numerical results for this algorithm are provided in **Chapter 5**. We simulated the proposed MAC with Network Simulator (NS2) and discussed the results. This includes the throughput, channel utilization, delay and fairness performance of the proposed systems with various types and numbers of flows. We conclude in **Chapter 6**, where we summarize the benefits of the proposed algorithm, and point out directions for future research.

Chapter 2

60 GHz Systems

The need for deploying high speed wireless systems for different kinds of applications (e.g. , WLAN) is increasing and recent studies show that intention for customers in retail market is rising up to get higher rates of wireless transmission. Those retail markets are mostly for home users and office networking specialists. This claim is based on the accelerating sales of IEEE 802.11 family of WLAN hardware [1].

Today, we gained 54 Mb/s in regular WLAN 802.11 wireless systems and most of the commercial wireless access points are capable of handling this amount of traffic in a desired speed. However, this speed is only enough for regular home and office users and obviously, it is insufficient for future demanding applications such as wireless high-quality video conferencing, multiple simultaneous wireless IEEE 1394 (Firewire) connections, wireless LAN bridges across network segments and wireless HDTV , since they need multiple gigabits per second bandwidth.

Looking for the high speed wireless networks, extremely high frequency radio signals (EHF) are proper candidates, above which are called terahertz radiation. The range of EHF is between 30 to 300 gigahertz which includes an unlicensed 60 GHz band. The 60 GHz band has a wavelength of 10 to 1 millimeter and therefore, it is called millimeter band or millimeter wave (MMW).

2.1 Technology

60 GHz wireless communication is the new wireless technology coming to the world of research. The name of 60 GHz millimeter-wave band is used for a huge unlicensed available spectrum, which is from 57GHz up to 66GHz. This allows a very high data rate wireless communication by using new transceiver modules. Researchers in recent years challenged with some specific characteristics and issues of high frequency wireless communications. They improve reliable methods and algorithms, used for lower frequency regions and then utilized the knowledge in designing and deploying new techniques for MMW wireless communication.

Compared to lower frequency signals, radio signals in this band (60 GHz) are extremely prone to atmospheric attenuation. Specifically, oxygen molecules absorb the energy of 60

Table 2.1: Frequencies Available for 60 GHz Wireless Communications Worldwide.

Country / Region	Frequency Range [GHz]	Purpose
Europe	62 to 63 or 65 to 66	MBS
Europe	59 to 62	WLAN
USA	59 to 64	General purpose
Japan	59 to 66	Wireless Communication

GHz radio signals much more than that of lower frequency signals. This physical effect (very high rate of attenuation) on 60 GHz signals is a considerable effect even for short distance communications. In addition, there is another problem which is called "rain fade". When humidity is high, or if it is raining, there will be a noticeable decrease in 60 GHz signal strength. Although this is an energy absorption effect which limits 60 GHz signals to practically function in only limited distances, it is possible for 60 GHz signals to have smaller frequency reuse distances than lower frequencies. The small wavelength allows modest size antennas to have a small beam width, further increasing frequency reuse potential.

In summary, what makes 60 GHz millimeter wave systems a very attractive solution for the purposes described earlier, are given below:

(a) There is about 9 GHz spectrum available from 57 GHz up to 66 GHz unlicensed for all kinds of short range wireless communication. This enables densely situated, non-interfering wireless networks to be used in the most high demanding bandwidth applications. Table 2.1 provides the situation of licensing for 60 GHz band. In Europe, the frequency ranges 62-63 GHz and 65-66 GHz are reserved for wideband mobile networks (MBS, Mobile Broadband System), whereas 59-62 GHz range is reserved for unlicensed wideband wireless local area networks. In the United States, the frequency range 59-66 GHz is a generally unlicensed range. In Japan, 59-66 GHz is reserved for wireless communications [11] [14].

(b) The propagation characteristics of 60 GHz signals in terms of attenuation rate and the power of passing through walls allow very high speed wireless networks to be setup very close to each other without congestion.

(c) 60 GHz signals are automatically under control based on the attenuation rate opposing to oxygen molecules. Therefore, these signals bring the higher security in wireless networks than before. Today, security is getting more important attention when all of the local networks are changing from wired to wireless. More specifically, sniffing is the problem and is the reason which makes 802.11 less secure than wired networks. Therefore, using a wireless system that cannot be sniffed from behind your office walls is much more desired.

(d) Metropolitan wireless networks are required to handle many users and applications simultaneously and, there is a gigabit-scale bandwidths offering by 60 GHz wireless signals in fixed outdoor wireless networks. Therefore, 60 GHz has a high potential to be at service for some metropolitan wireless networks.

(e) 60 GHz is not a competing technology for 3G systems, but instead a building block for future wireless networks [1]. Deploying 60 GHz wireless networks in indoor environments is highly required, and for outdoor is a complementary solution for 3G and high speed wireless technologies used in metropolitan systems.

2.2 Applications

2.2.1 General Applications

Scientific Research This band is commonly used in radio astronomy and remote sensing. Satellite-based remote sensing near 60 GHz can determine temperature distributions in the upper atmosphere by measuring radiation emitted from oxygen molecules that is a function of temperature and pressure. 60 GHz band is used for this purpose because of the high energy absorption by oxygen molecules facing 60 GHz signals. There are many applications for this band in climate sensing systems [15].

Telecommunications 60 GHz band can be used for unlicensed short range 1.7km data links with data throughput up to 2.5Gbps. It is used commonly in flat terrain in United States.

Security Systems A recent development has been the imagers for security applications as clothing and other organic materials are translucent in some mm-wave atmospheric windows.

2.2.2 Wireless Applications

Wireless Local Area Networks (WLAN)

An important usage of 60 GHz is to establish Wireless Local Area Networks (WLAN). Fig. 2.1 illustrates this application in two different states. 60 GHz is a solution for home or office. Once you establish a 60 GHz WLAN at your home, you can integrate your WLAN with UMTS (cellular network) and wired infrastructure (Cable or Fiber Optic). There are many kinds of devices at home or in the office which can be connected together without wiring in very high speed communication. A user can send his/her captured video to TV and watch it. Also it is possible to connect laptops and audio players and cell phones together and synchronize all them. It means that for example, if you like to watch a movie then you have the choice to watch it on your TV or laptop or cell phone or iPod. You can transfer data from your laptop to other stations as fast as over gigabits per second. Possibly, you can watch HD movies on you TV while they are stored in your laptop hard drive, without prerecording and just in the same time you are broadcasting from your laptop to your TV. In fact, there are so many possibilities for WLAN application scenarios and there are much

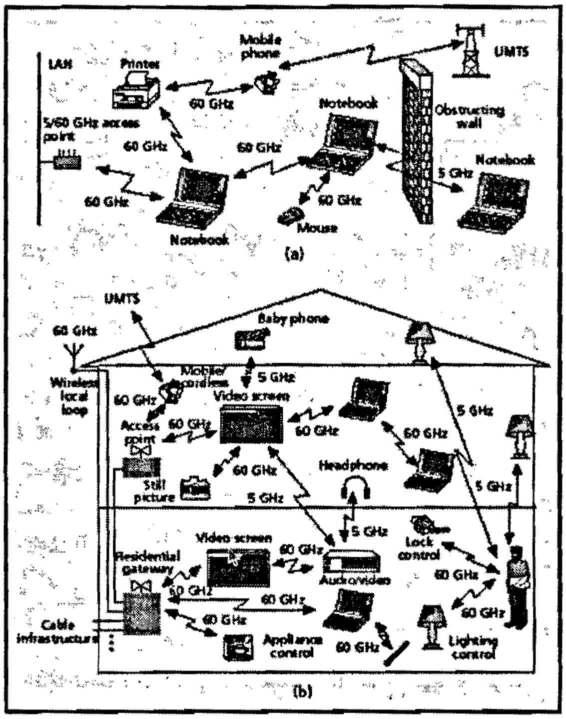

Figure 2.1: Home and Office Application Scenarios for 60 GHz or Combined 5-60 GHz Systems [1].

more demand for it. For example, in Canada we get a bundled program from Rogers that includes cable TV, high speed Internet and home phone. Then, we can only connect a 60 GHz access point to the incoming cable to our home. Then all these services would work simultaneously from anywhere at our home or office.

Mobile Networks

Technically, 60 GHz system is not a suitable candidate for cellular systems due to the higher rate of attenuation compared to lower frequency signals. Actually, higher frequency signals above 60 GHz do not have this attenuation problem and there may be a cellular application for those frequencies in the future. 60 GHz MMW systems is proposed for the 4G communications technology by some researches [1], but a stronger argument would place the 60 GHz in WLAN and Fixed Wireless applications.

Fixed Wireless

The most promising application for 60 GHz MMW technology is fixed wireless communication. Unfortunately, it is not possible for long distance communications as the attenuation effect is very influential on signal power. 60 GHz signals with less than 500 mW output power can be widely used without licensing. In addition, in many geographical areas only two or three carriers are allowed to operate on licensed spectrum areas.

Today, 60 GHz transceivers are available for fixed wireless communication. They cover communications between offices, buildings, database systems and bank branches and its usage is growing fast. Fig. 2.2 shows this kind of usage. The fact is that Ultra-Wide Band systems are used for the same purposes and are preparing the atmosphere for bringing higher frequency technologies like 60 GHz for metropolitan applications.

Another issue is reliability and interference. Compared to UWB communications, 60 GHz MMW systems make use of a proven technology with little worries about interference. Up to now, UWB has shown great potential as far as power consumption, frequency reuse and manufacturing costs are considered but still unresolved questions exist about interference and regulations.

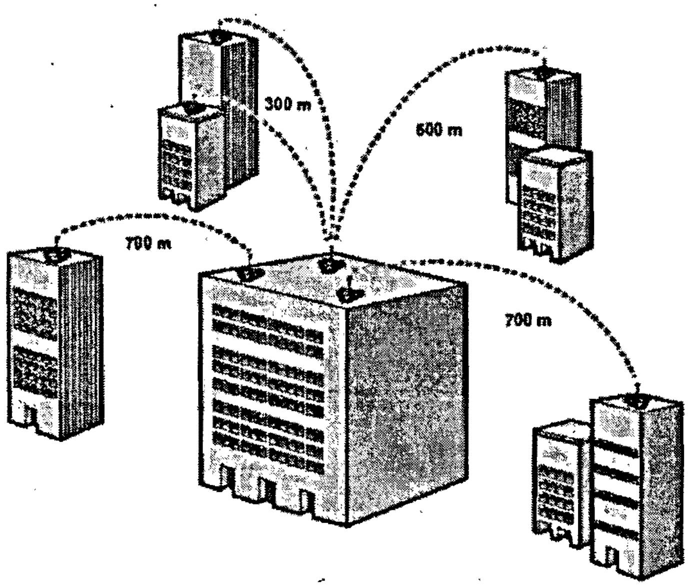

Figure 2.2: Wireless 60 GHz Communication in Metropolitan Area [1].

Along with all the research on 60 GHz wireless communication, one attractive issue to work on is the integration of 60 GHz radio frequency circuitry with the antenna so that intra connectivity losses could be reduced and size/costs optimized. The user hardware antenna for 60 GHz WLANs should also be optimized for better behavior when line-of-sight is lost [2].

The evolution or commercial roll-out, has started with fixed wireless metropolitan area solutions, and it is likely to continue with WLAN-like applications later on. Both may also be able to survive: UWB in home applications and customer electronics, 60 GHz in corporate use with higher security and reliability requirements. We think that, 60 GHz systems will eventually make their way to wide-band wireless market.

Combined 5/60 GHz Radio Hardware

One possible scenario for quicker 60 GHz adoption is 5 GHz / 60 GHz combo hardware that could solve two problems at the same time: the backward compatibility of WLAN systems (e.g. 5 GHz IEEE 802.11a) and the problems with 60 GHz in situations when line-of-sight or adequate amount of reflected signals is lost. With a 5/60 combo hardware, 5 GHz network could operate as a backup channel. Fig. 2.3 illustrates this point [2].

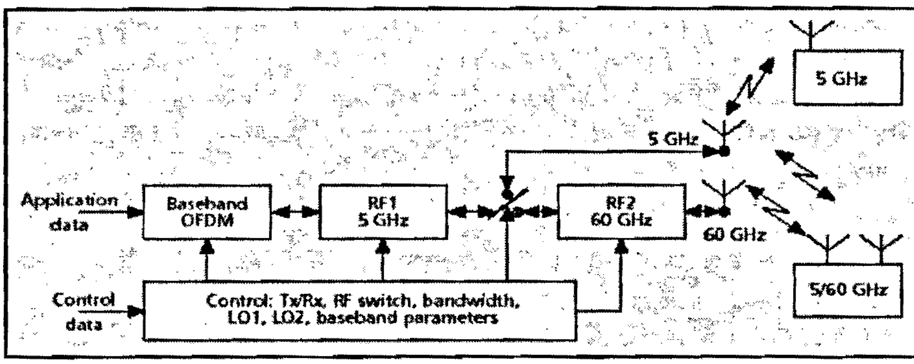

Figure 2.3: 5/60 GHz Combo Hardware [2].

In addition to backward compatibility and improved network redundancy, a 5/60 GHz system could also be rather highly integrated: if 5 GHz is used as an intermediary frequency, a lot of circuitry can be used for both 5 GHz and 60 GHz functions minimizing size and cost (see Fig. 2.3).

At last, in order to demonstrate the required bandwidth for different kinds of wireless applications, Table 2.2 is provided in [1] containing a list of all possible important wireless applications and the average required bandwidth for each of them. This table is helpful for finding the possible combination of usages for a 60 GHz transceiver at home or office or metropolitan area. In designing systems and protocols for 60 GHz MMW communications, the main guidelines for architecture design should include affordability, scalability, modularity, extendability and interoperability.

2.3 MAC Protocols

MAC (Medium Access Control) for 60 GHz wireless systems is a new design challenge for researchers because of high frequency and high rate of data transmission which requires

Table 2.2: Demonstrates bandwidth usage of wireless applications [11]

Application	**Capacity per user [Mb/s]**
Wireless LAN bridge, e.g., for interconnecting GigaEthernet LANs in different buildings	100 to 1000
Wireless virtual reality allowing free body movements	450
Wireless IEEE 1394	100,200,400
Wireless TV high-resolution recording camera	150 to 270
Wireless trading terminal having multiple video channels that can be viewed simultaneously for monitoring world news next to stock quote information	50 to 100
Wireless news tablet, a very thin, possibly flexible device that provides the user with a newspaper, e.g., the possibility of activating images to see video impressions	50 to 100
Wireless (high-quality) videoconferencing	10 to 100
Wireless Internet download of lengthy files	10 to 100
Wireless ad hoc communications, i.e., direct communication between notebooks, between notebook and nearby printer, etc.	0.1 to 100
Wireless interactive design	20 to 40
Hospital bedside application allowing wireless retrieval of patients status including Xray pictures	10
Wireless surveillance cameras allowing face and number plate recognition at long distances	4 to 10
Patient monitoring (patients can walk freely around in the hospital or even at home) with devices that transmit ECG, blood pressure information, etc.	2
Wireless videophone	1.5
Wireless connection between domestic appliances and the Internet e.g. a refrigerator scans its contents and orders the nearby supermarket to deliver what is missing	0.1
Wireless billing (e.g., automatic payment for petrol service via wireless connection between car and filling station)	0.1
Road pricing	0.1
Wireless burglar alarm (wireless window sensors, etc.)	0.01
Remote control (TV, lighting, door or window lock)	0.01
Wireless embedded systems (e.g., in car between oil filter and dashboard)	0.01

new error detection, error correction and retransmission algorithms. Although some physical features of 60 GHz wireless systems are different than regular UWB systems, it is still kind of an UWB wireless system in terms of having a wide range of available frequencies. UWB wireless MAC is a standard published by IEEE for high rate and short range wireless communication which is suitable for 60 GHz wireless systems but with some changes and modifications [7] [12] [13].

For high data rate (11 to 55 Mbps) Wireless Personal Area Networks (WPAN), we have to refer to 802.15.3 standard that has four categories. First is 802.15.3 which is the general standard for high data rate MAC and PHY. The second one, 802.15.3a, is for higher speed UWB PHY. The third one, 802.15.3b, is amendment for MAC in order to correct some errors and preserve backward compatibility. The fourth one, 802.15.3c, is for the PHY of MMW wireless communication (60 GHz).

In summary, we do not have a standard dedicated to 60 GHz wireless MAC. Today, researchers are trying to design the MMW MAC layer based on general 802.15.3 MAC layer design. The smallest unit of UWB network is piconet containing a controller (Piconnet Controller), which is called PNC, with surrounding devices. A piconet is distinguished from other types of data networks in that communications are normally confined to a small area around person or object that typically covers at most 10 m in all directions and envelops the person or a PC or a mobile device [3]. A piconet is a wireless ad hoc data communications

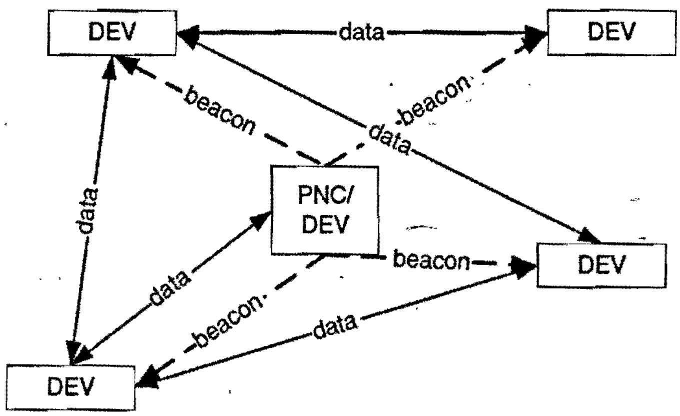

Figure 2.4: Piconet Elements [3].

system which allows a number of independent data devices to communicate with each other in a small area. Piconet is in contrast to local area network (LAN), metropolitan area network (MAN), and wide area network (WAN), each of which covers a larger geographic area, such as a single building or a campus.

The component of an UWB piconet (PNC and surrounding devices) are shown in Fig.2.4. In fact the PNC is a device chosen among others to provide basic timing for the piconet with the beacon , manage the quality of service (QoS) requirement, power saving modes and access control to the piconet. The timing is an important responsibility of PNC that is studied in this thesis in terms of superframe size. The timing effects the QoS and performance of the entire network.

We discussed the some characteristics and applications of 60 GHz systems in this chapter. We explained that one the important usages of MMW networks is short range WLAN and based on this, we mentioned that the basic functionalities of 802.15.3 MAC is to serve the connections and data transmissions among devices in an ad hoc topology. We explained that piconet is the smallest unit of a 802.15.3 network containing PNC and several surrounding devices. In the next chapter, we would review some studies which analyzed the performance of MMW and UWB systems utilizing 802.15.3 MAC layer. We would also explain some proposed algorithms by recent studies aimed to improve the performance of MAC layer.

Chapter 3

Literature Survey

Over the past years, a considerable amount of research has been performed regarding MMW MAC layer. As discussed previously, MMW is considered a specific type of UWB as is in the form of pulse radio wherein each transmitted pulse instantaneously occupies the UWB bandwidth, which is an aggregation of at least 500MHz worth of narrow carriers. Therefore, studies which are done around UWB physical and MAC layer are also related to that of MMW systems. In this thesis, we would use the results from studies done for physical layer in our system model and will direct our research toward MAC layer. The MAC model of UWB and MMW systems are different from legacy 802.11 systems in terms of functions and components. Piconet is an ad hoc network and is defined as the smallest unit of a 802.15.3 network containing a PNC and several surrounding devices. Considering the new structure of MAC in 802.15.3, we provide a general review around MMW and UWB systems and their performance improvements.

3.1 Throughput

Throughput is the average rate of successful message delivery over a communication channel. However, throughput may be measured in different ways. In order to clarify this, we define throughput in the following terms [20]:

- **Maximum theoretical throughput** is closely related to the channel capacity of the system, and is the maximum possible quantity of data that can be transmitted under ideal circumstances.

- **Maximum achievable throughput** takes into account handshake and control packets, which reduce the amount of channel space available for data packets, as well as considerations such as reduced data packet length. Additionally, this value takes into account hardware limitations of the systems on both ends of the channel.

- **Peak measured throughput** is measured by a real, implemented system, or a simulated system. The value is the throughput measured over a short period of time mathematically.

- **Maximum sustained throughput** is the throughput averaged or integrated over a long time or infinity.

Throughput is sometimes normalized and measured in percentage :

- **Channel utilization**, also known as **bandwidth utilization efficiency,** in percentage is the ratio of achieved throughput and maximum theoretical throughput.

The term "throughput" usually refers to the maximum sustained throughput which is a result of simulations. However, many studies analyzed the maximum theoretical throughput (capacity) and many analyzed the maximum sustained throughput. We provide a review from studies about both terms in the two following sections.

3.1.1 Maximum Theoretical Throughput (Capacity)

Studies in [4] are utilizing different techniques to improve maximum achievable throughput. The proposed technique there includes the following modifications in MAC layer;

1. An automatic device discovery using directional antenna (ADD) .
2. A method to remove unnecessary packet headers (Aggregation).

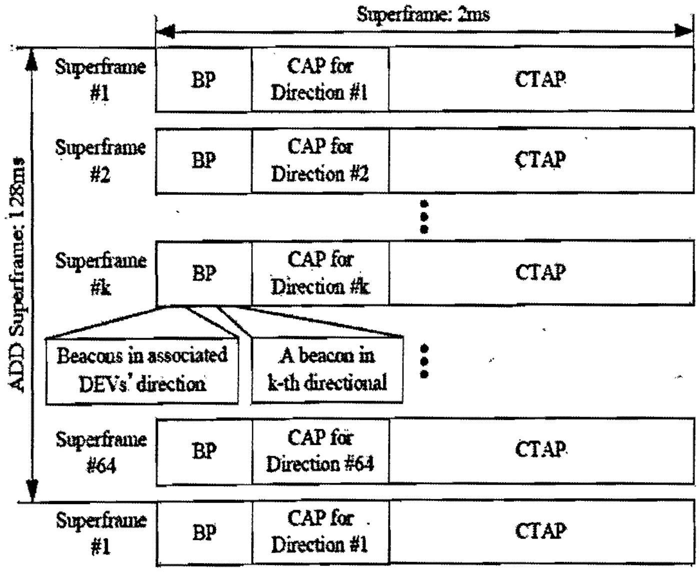

Figure 3.1: Automatic device discovery superframe [4].

As it can be seen in Fig. 3.1, the beacon period (BP) is the shared time for broadcasting beacons to all the nodes from PNC. Beacon contains the timing information and is a coordinator as it is used to synchronize all the nodes with PNC. Contention access period (CAP) is the time for devices which are transmitting non-real-time flows. These flows are supposed to access the channel randomly without any reservations. The MAC layer protocol in CAP is CSMA/CA. Channel time allocation period (CTAP) is the time for transmission of real-time flows. Nodes need to send registration request and transmit in an specific channel time allocation (CTA).

As discussed previously, MMW signals have high attenuation rate, therefore, directional tranmission is needed in a MWW system. One solution to make a directional transmission possible in a WLAN is to provide a LOS (Line of Sight) direction for each pair of devices. This method is not emphasized in [4] for many reasons such as high complexity and costs. However, ADD is introduced as a proper candidate to be used for directional transmissions. This technique is running in MAC layer and is a new type of superframe structure. Superframe is handling the coordination process with the help of two protocols (CSMA/CA and TDMA) for different kinds of transmission in each superframe cycle. In order to implement ADD, PNC sends the superframe in 64 directions as shown in Fig. 3.1, thus beacons for devices in each direction are included in the superframe. Numerical analysis in [4] reveals that maximum of 64 directions can be managed by PNC simultaneously. In this scenario, the length of superframe would be $2 \times 64 = 128$ms.

In order to measure the system capacity, a MMW system is simulated in [4]. Simulations are done under two use cases. Use case one (U1) is an uncompressed video streaming and use case two (U2) is a kiosk file downloading which are shown in Fig. 3.2. These use cases are defined scenarios for MWW data transmission in IEEE 802.15.3c standard. Also proper acknowledgment techniques are used for each of the use cases. No-Ack and Delay-Ack are used for use case UM1 and UM2 respectively. Results shown in Fig. 3.3 and Fig. 3.4 are

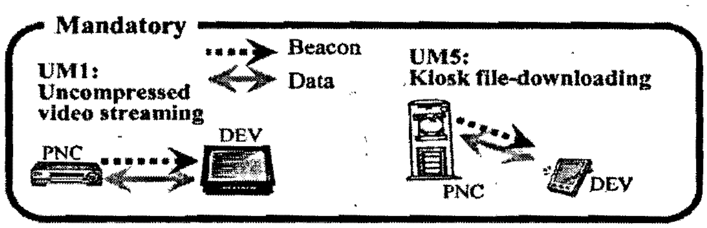

Figure 3.2: Usage model [4]

MAC data rate (capacity) versus data payload size. It can be seen that payload size is not a bottleneck in U1 but, the optimum payload size for U2 is 10 kilo octets. The reason is as follows: Retransmission of packets would reduce the capacity for a certain SNR and BER, hence longer data packets need longer time for retransmission and consequently cause reduction in capacity. Based on this U1 utilizing No-Ack method which has no retransmission has better data rate results which is 4 Gbps data rate and U2 utilizing Delay-ACK method results in 2 Gbps data rate. Another simulation is also done based on different aggregation schemes.

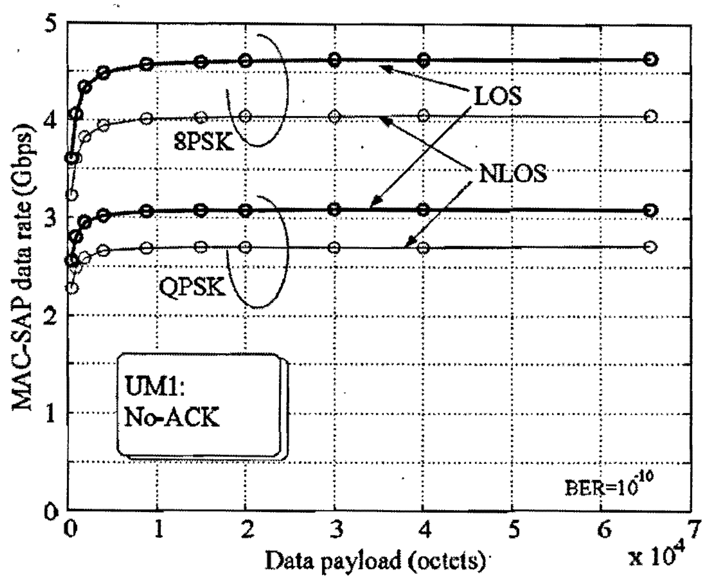

Figure 3.3: MAC data rate analysis design, U1 [4].

Several aggregation types are MSDU (MAC Service Data Unit) aggregation, MPDU (MAC Protocol Data Unit) and PPDU (PHY Protocol Data Unit). Interesting results by frame aggregation are provided in [4]. The aggregation results of MPDU and PPDU show that aggregation of MSDU could not improve the capacity and results in 2 Gbps data rate loss, but MPDU packet aggregation results in 2 Gbps improvement in capacity.

Moreover, some studies are focusing on how the network topology influence the throughput in UWB and MMW networks. For example, a meshed network topology may contain enormous number of devices and if MMW device is used in a meshed wireless topology, the capacity of superframe might not scale to higher number of devices that send multimedia traffic. In [5], some methods are proposed on how we could solve the superframe limitations in a meshed wireless network and how to maximize the capacity (maximum theoretical throughput) of a meshed MMW network.

This paper analyzed the throughput of RT flows in CTA period. It presents a theoretical relation among the number of devices in the entire system, length of each CTA period and the number of frames transmitted in each CTA period. Based on their study, by increasing the number of frames per CTA, the overhead for each CTA reduces but the number of supported devices decreases. This is due to the fact that the ratio $\frac{n \times n_{frag}}{n_{CTA}}$ increases with the increase in number of frames sent per CTA where n is the number of fragments sent per

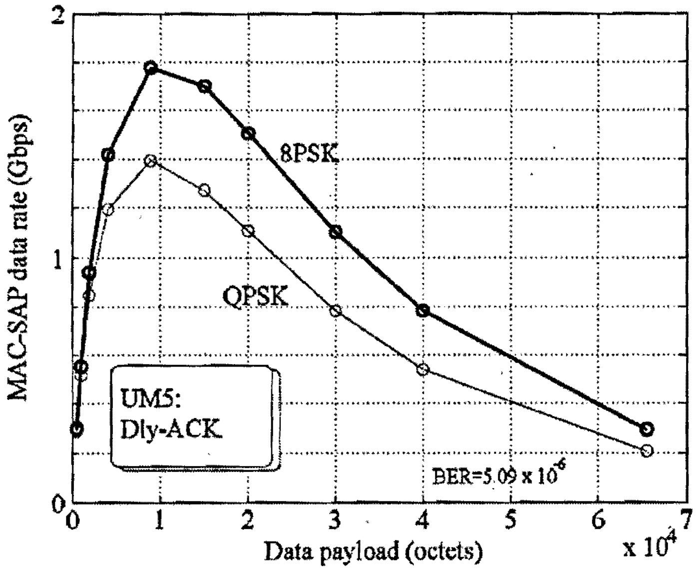

Figure 3.4: MAC data rate analysis design, U2 [4].

CTA, n_{frag} is the time duration required to send each fragment and n_{CTA} is the total time duration for each CTA which includes the overhead. On the other hand, to keep the high quality of service in multimedia transmission, it is better to transmit data in larger fragment sizes. Based on this, they analyzed an UWB system (IEEE 802.15.3) where devices send data with a highest fragment size and they observed that by sending up to 3 frames per CTA, only 8 devices can be accommodated in the superframe. This number is different for different types of application. For example, if only voice traffic is transmitted in the system, the number of supported device could be up to 40 devices.

They have extended this analysis to a system with higher transmission rates such as MMW systems (IEEE 802.15.3c). Two of the relevant graphs presented in [5] are shown in Fig. 3.5 and Fig. 3.6 for illustration. It can be seen that wireless systems with different transmission rates are able to support different number of devices. From both figures, we can see the effect of number of frames in CTA period on the entire number of devices in the system wherein a system with 800 Mbps transmission rate could support 10 to 40 devices and a system with 3000 Mbps transmission rate could support 30 to 120 devices depending on the number of frames in a CTA time slot.

3.1.2 Maximum Sustained Throughput

In [16], an analysis is done for an UWB system to measure the successful requests from users in CAP periods. They estimated the rate of successful requests of IEEE 802.15.3 accounting for Dual Carrier-Orthogonal Frequency Division Multiplexing (DC-OFDM-UWB). This

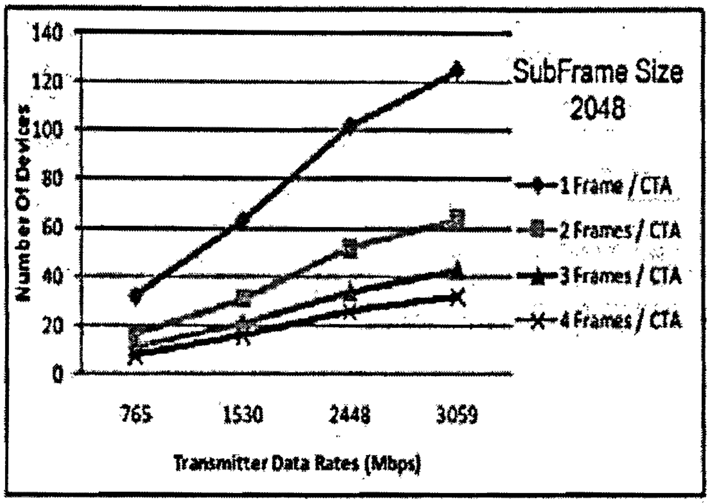

Figure 3.5: Superframe Capacity with frame aggregation [5].

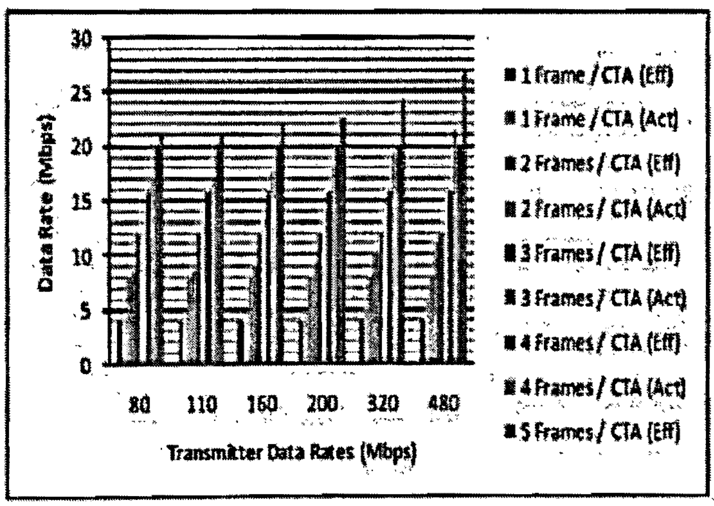

Figure 3.6: Achieved throughput with IEEE 802.15.3a [5].

analysis provides helpful insight to design the superframe in MMW systems. In [16], the throughput of NRT flows in CAP period is defined as the ratio between the time in which the channel is used for successful transmission of data frame and the total channel time. Thus the throughput S is denoted in (3.1).

$$S = \frac{T_{data}}{T_r}, \tag{3.1}$$

where T_{data} is the duration of the payload and T_r is the renewal interval which consists of idle slots followed by a successful or failed transmission attempt. These times are pertaining to the deployment of CSMA/CA protocol. And more computations based on this provide the average number of successful requests per device as shown in (3.2).

$$N_r = \frac{T_{cap} \times S}{T_{data} \times n}, \tag{3.2}$$

where T_{cap} is the duration of CAP and n is the number of flows. Thus, the proposed method for estimating the duration of CAP is to specify a certain value for N_r (number of successful transmissions) and compute the duration of contention free period (CTAP) by subtracting CAP duration from the entire length of superframe as shown in (3.3).

$$\text{Length of CTAP} = \text{Total Length of Superframe} - \text{Length of CAP}. \tag{3.3}$$

Simulaton results in this study show that N_r or the probability of successful transmissions decreases rapidly from 8% to 1% when number of flows increases and, this is due to the back-off mechanism in CSMA/CA protocol which takes effect from the shorter length of contention access period comparing to the length of contention free period.

This paper [16] shows the relationship among the throughput of NRT flows and the ratio between CAP and CTAP. The authors claim that a longer CAP would increase the NRT throughput while it should keep the minimum required bandwidth for CTAP.

Another study investigating an optimum length of superframe is in [6]. In this study, an UWB system with different ratios between CAP and CTAP in superframe is simulated and results provide us a good insight for designing the superframe in MMW systems. They varied the length of the superframe while keeping the length of the CAP constant at 1ms. Fig. 3.7 shows the average throughput of the real-time flows when allocated CTAs with time duration of five, seven and nine milliseconds. It can be seen that the throughput is decreasing as the superframe duration is increasing. This is due to the increase in waiting time for a real-time flow. This result shows that if a real-time flow is limited by channel time, every one millisecond increment in superframe duration corresponds to an average reduction of 0.2Mbps in throughput. In the second case, the authors simulated the system with several different

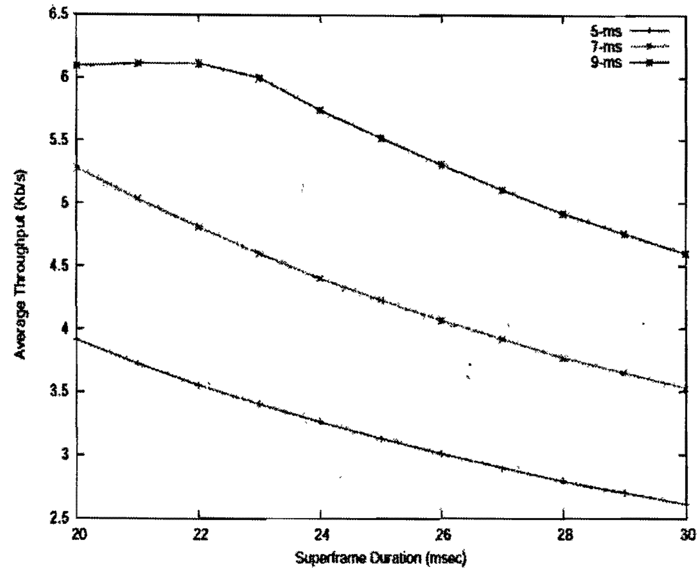

Figure 3.7: Average throughput of real-time flows versus superframe duration. The graph also shows the average throughput for each of the two real-time flows under varying CTA slot times [6].

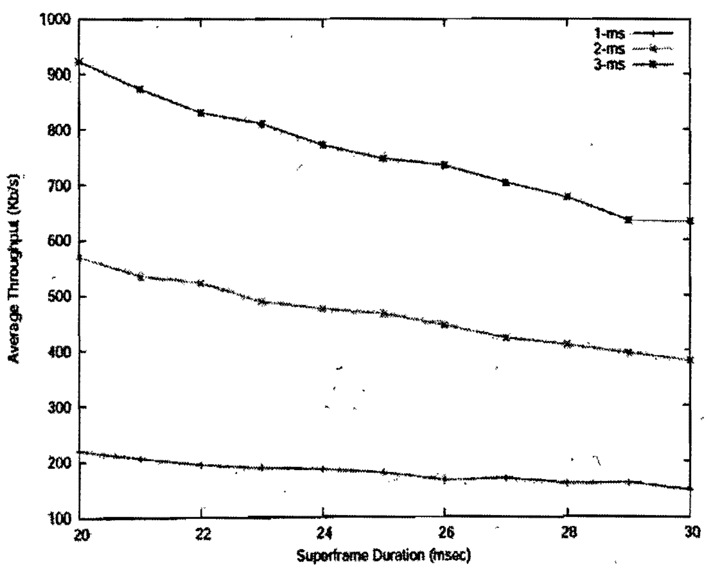

Figure 3.8: Average TCP throughput versus superframe duration for different CAP durations [6].

CAP sizes and measure TCP throughput (non-real-time flows) as showin in Fig. 3.8. It can be seen that TCP throughput is increasing from 200kbps to 900kbps as the CAP length is increasing from 1ms to 3ms. Moreover, similar to the previous case for real-time flow, the TCP traffic shows a reduction in throughput as the entire superframe period increases.

In summary, since the superframe time must be shared between the CAP and the CTAP, the throughput of the TCP flows could be improved by increasing the CAP duration. However, if this is done, less time will be available to the real-time flows in the CTAP. Therefore, we should find the optimum point for the ratio between CAP ad CTA period in order to maximize the system capacity.

The fairness among TCP flows is studied in [6]. As shown in Fig. 3.9 and it can be

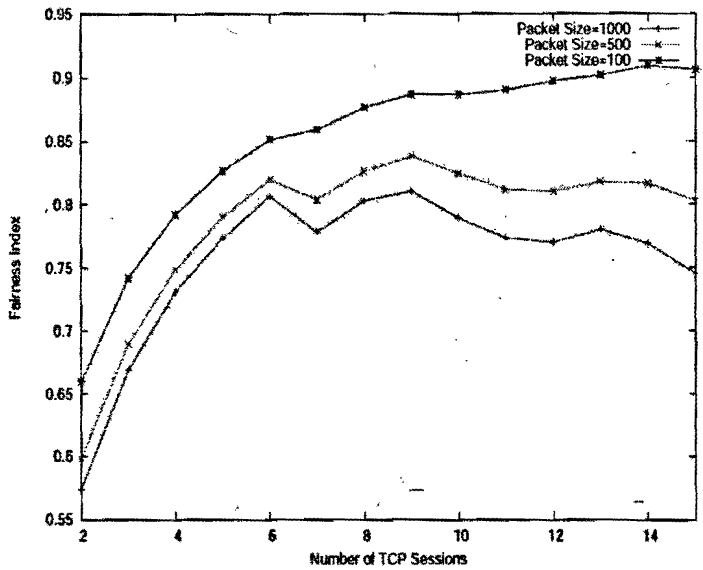

Figure 3.9: Fairness index versus increasing TCP sessions (CAP set to 1ms). Packet size is in bytes [6].

seen that the fairness index is low when there are a small number of connections, yet fairness improves with increasing number of flows and starts declining again when there are a high number of flows. This is due to the back-off mechanism in CSMA/CA protocol. The higher the number of TCP flows are, the higher contentions occur in the system and flows should waits for the longer back-off times. Consequently, some flows could take the chance to transmit and some could not. Therefore, the fairness among TCP flows decreases.

As discussed previously, MMW systems are used to transmit data and multimedia in very high data rates, thus researchers are looking for the transmission bottlenecks in order to improve the throughput of a MWW system and reach the required quality of service. Some studies are directed toward this matter and investigated the bottlenecks in the MAC layer of a MMW system. Superframe is the main element of MAC layer in IEEE 802.15.3 where different flows could be distributed whithin a superframe period and transmit data based on TDMA or CSMA/CA protocol. In [7], the performance of CSMA/CA in a MMW system is studied. They analyzed the performance of CSMA/CA protocol when the data rate is high (at least 1Gbps). In this study, the authors define three allocation metrics in order to measure the CSMA/CA performance which are as follows:

- *Normalized Throughput*: It is defined as the ratio of the throughput to the data rate of 1Gbps.
- *Average Access Delay*: It is defined as the avergae time delay of a frame from creating to transmit it.
- *Completion Rate*: It is defined as the number of tranmistted frames successfully over the total number of tranmissted frames.

Normalized throughout in [7] is the same as channel utilization which we defined earlier. Average access delay is an important metric for estimating the quality of service for multimedia tranmission. Completion rate is the reverse of another metric which is called job failure rate (JFR) and we would use it for our performance measurements in this thesis. In [7], the

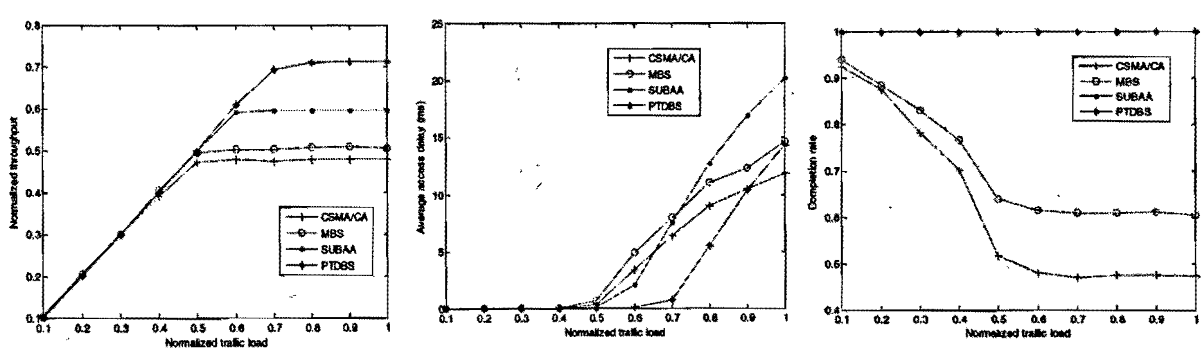

Figure 3.10: (left) Normalized throughput, (middle) average access delay, (right) completion rate [7].

authors compared the performce of CSMA/CA with other channel access protocols such as MBS (Memorized Back-off Scheme), SUBAA (Short Unique Backoff Allocation Algorithm) and their proposed channel access algorithm is called PTDBS (Pseudo-Time Division Backoff Scheme). The technical difference among mentioned algorithms are the back-off strategy when contention happens. As shown in Fig. 3.10, this PTDBS could gain higher throughput, lower delay and the completion rate of close to 1 when working in a MMW system. From [7] we could obtain an insight about the performance of CAP period in superframe and all three of the simulation results are being referenced in our analysis in this thesis.

TCP flows (non-real-time flows) are supposed to be sent during the contention access period. However, some studies proposed a new method in which, non-real-time flows could be sent during CTA period. This method is apart from the throughput analysis of CTA period wherein real-time flows are being transmitted. In [6] and [8], this method was proposed and is extensible to all the non-real-time flows which may be transmitted using TDMA protocol and not using CSMA/CA protocol. In this method, a TCP flow would send a message to PNC to register a CTA time slot, then keeps the data in a buffer until the next time slot in which it has the right to transmit. In this case, the receiver should also register for a time slot in order to be able to send back acknowledgment packets.

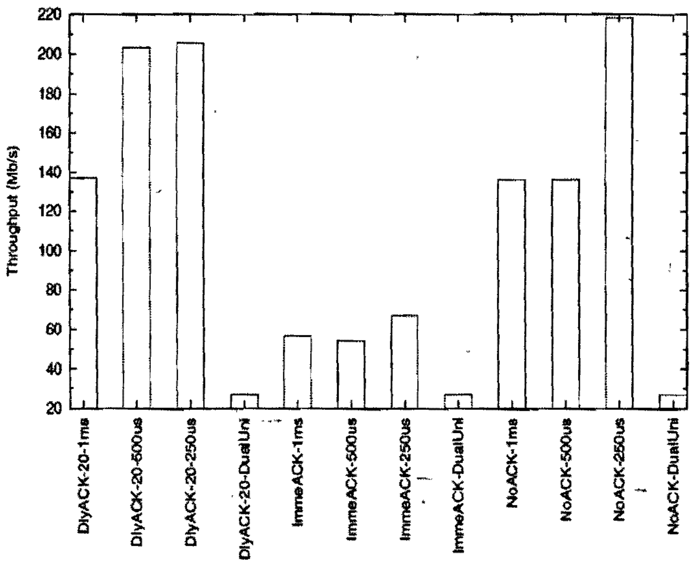

Figure 3.11: The maximum throughput achieved for different CTA allocation strategies and acknowledgment policies [8].

As it can be seen from Fig. 3.11, achieved throughput varies by different acknowledgment techniques. Also duration of CTA time slot in which the data is being transmitted is important. Those flows which are using No-Ack or Delay-Ack gain better throughput than those which are using Immediate-ACK. The closer the size of a CTA time slot is to the size of the TCP contention window, the better throughput would be achieved.

The analysis and simulations in [8], are beneficial when no improvement in TCP throughput could be gained using CSMA/CA protocol and thus, a wider bandwidth utilized by TDMA protocol could be a solution.

3.2 Power Consumption

Power consumption is an important issue for mobile hand-held devices. The better performance in power consumption results in a longer time in using a hand-held device without a permanent power source.

Authors of [17] claim that the power consumption of the entire system could be reduced by choosing a device as the piconet coordinator (PNC), which is located in the center of the network topology. This means that the average distance between each node and PNC could be reduced and therefore, PNC and other devices could communicate with an optimized signal power. They propose to utilize the direction information extracted from beam formed transmissions in finding the piconet coordinator automatically in a 60 GHz wireless network.

Authors of [17] used an algorithm for utilizing the direction information besides using the distance information discussed in [18]. In this algorithm, P_i is assigned to the received power for device i. The transmitter must guarantee a minimum threshold for all the P_i values. In other words, it must provide a minimum SNR for all the receivers. In this case, we have to minimize the transmitting power of a potential coordinator in order to satisfy that condition. This minimization needs the distance information which is discussed in [18]. Directional transmission of 60 GHz is utilized in extracting the direction information that each device resides relative to other devices. In this algorithm, a coordinator-capable device calculates D_i, T_i for all $i = 1$ to $N - 1$ (D: Meter, T: Radian, N: Number of devices, i: Index number of a device). The basic idea of the discussed algorithm is to find a rectangle, which covers all the points, then choosing a device that is closest to the center of the rectangle.

The simulation results show that this approach results in less power consumption compared to distance-based algorithm and is more likely to find the best device as a coordinator. In Fig. 3.12, the improved algorithm is compared to original algorithm. It is obvious that the maximum distance to the coordinator in the original algorithm is more than the case which is proposed in [17].

After optimizing the distance of the coordinator, we can expect that PNC can operate longer and the whole network would continue to operate for a longer time without any interruption. Another improvement which is achieved by this proposed algorithm is solving the hidden terminal problem. Since it is much less possible for a coordinator to lose its connection with a device or not be able to cover the whole area of wireless network. By using this algorithm, this can be assured that all the devices are covered by the coordinator.

There is also another improvement in power consumption proposed in [9], which discusses about a new algorithm for a directional MAC. This directional MAC takes advantage of

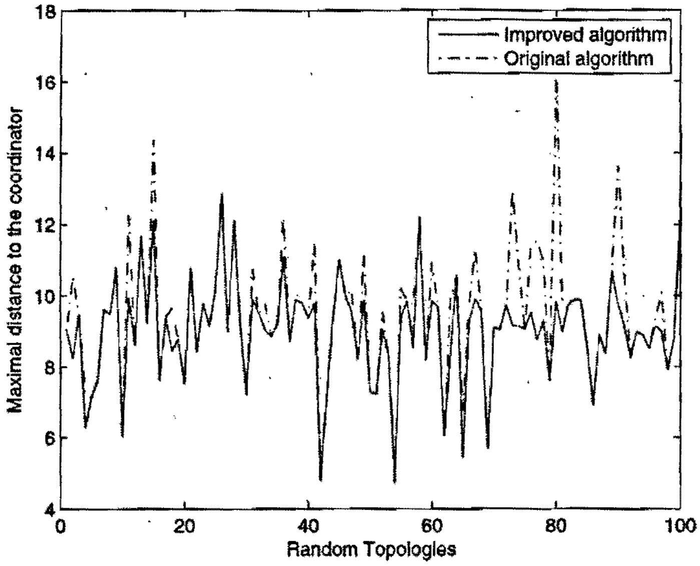

Figure 3.12: Improved algorithm vs. original algorithm [4]

different signal polarization and distant information simultaneously in order to find the dominant path for the connection between transmitter and receiver. As we discussed earlier about the penetration loss in 60 GHz signals, we need to make sure that mobile nodes are receiving data from the best direction by means of a DMAC protocol (Directional MAC). The proposed DMAC in [9] also improves RMS (Root-Mean-Square) delay spread. Delay spread is a type of distortion that is caused when an identical signal arrives at different times at its destination.

We can conclude the following about power saving methods. Firstly, by using an efficient coordinator selection algorithm proposed in [17], average distance to the coordinator can be reduced by about 2 meters and consequently reduction of power consumption for sending data to all the nodes guarantying a minimum threshold of PSNR. Secondly, by using directional antenna along with a directional MAC proposed in [9], received signal power for each single node could be increased by −15 dBm.

3.3 Directional Transmission

In [9] directional antenna supported by MAC layer is introduced as a solution for hidden terminal problem, saving power, increasing peak signal power to noise ratio (PSNR) and also reducing RMS delay [9]. In this case, a MAC protocol suitable for directional antenna should be employed while taking the specific characteristics of 60 GHz signals into consideration.

Author of [9] discusses that earlier studies about directional MAC has been around lower frequency wireless networks and most of them could not propose a proper and practical direction-finding algorithm. However, the authors of [9] claim that they present the best direction-finding algorithm. The proposed algorithm in [9] is taking advantage of the fact that reflected signals, which are received by the receiver, have different powers under different polarizations. Thus we can see whether LOS exists between transmitter and receiver or not. Another step forward in order to benefit from this algorithm is to use smart antenna in which polarization choices can be adaptively selected for different communication scenarios. The proposed algorithm is as follows. Two different test signals in terms of polarization are propagated simultaneously in all the directions. Then the different received power for these two signals in the receiver is a measure to determine a LOS or NLOS situation. After the LOS or NLOS estimations, the receiver would respond with an acknowledgment packet containing LOS or NLOS information. Transmitter upon receiving the ACK would adjust a circular polarization for LOS scenario or linear polarization for NLOS scenario to obtain a reliable wireless connection. In order to keep transmission in a certain LOS or NLOS dominant path, transmitter and receiver would synchronize a time stamp. The discussed algorithm is simulated versus traditional MAC (omni-MAC) and interesting results obtained as shown in Fig. 3.13.

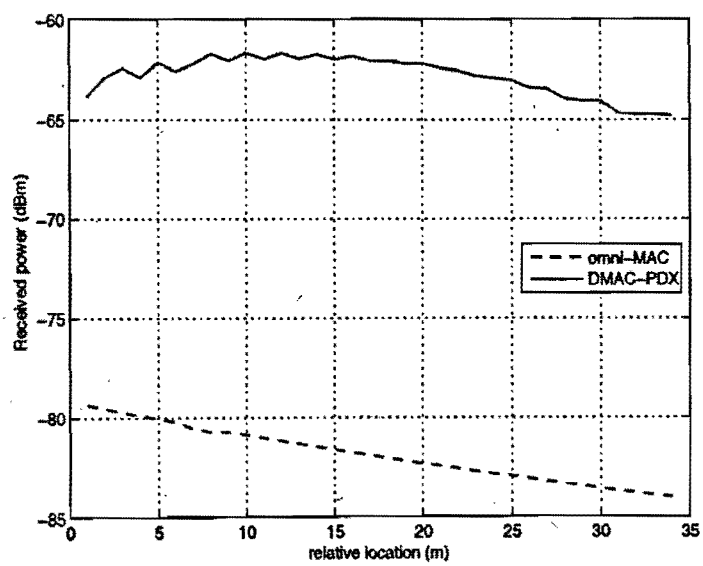

Figure 3.13: Comparison of the traditional MAC (omni-MAC) and the proposed DMAC-PDX algorithm in terms of received power from the most dominant path in NLOS [9].

As discussed previously in subsection 3.1, another method for directional transmission is proposed in [4]. In this method which is called automatic device discovery, PNC transmits superframes in 64 directions. This superframe contains beacons for all the 64 directions. Therefore, the length of the superframe is 64 times longer than a simple superframe. For example, if the simple superframe has 20msec length the ADD superframe length would be $20 \times 64 = 1280$msec. This method is less complex and more extendible, yet it compensates more bandwidth for directional transmission comparing to the previous discussed method.

We reviewed some recent studies around 802.15.3 networks in this chapter. The improvement and diminish in capacity, sustained throughput, delay and power consumption are briefly explained while using DMAC or omni-MAC or different CAP and CTAP in terms of superframe sizes. Some techniques in MAC layer are proposed by some studies in order to improve the performance of TCP or UDP flows in an UWB network, which we also briefly explained. As explained earlier in this chapter, studies which are done around UWB physical and MAC layer are related to that of MMW systems since MMW MAC layer design is based on the UWB MAC layer. Therefore, in the next chapter, we would analyze the performance of UWB MAC utilized for a MMW system. We would then propose necessary modifications on 802.15.3 as an algorithm and analyze the performance of a MMW MAC using our proposed algorithm.

Chapter 4

Adaptive Channel-Superframe Allocation (ACSA)

In Chapter 3, some MAC layer algorithms for MMW systems were reviewed. The new ideas and algorithms proposed so far in the literature related to superframe design, acknowledgment techniques and directional transmission could give considerable improvements to a MMW system in terms of throughput, delay, fairness and power consumption. In some cases, throughput is improved by utilizing a directional antenna or a novel acknowledgment technique. A directional antenna could utilize a complex algorithm in MAC layer as discussed in [9] in order to locate devices around it. Another approach proposed in [4] is broadcasting 64 superframes in all the directions instead of locating the devices. Above techniques could also improve SNR significantly as well as reduce power consumption. Even though some papers achieved favorable results, we realized that not much attention is paid to the types of applications using a MMW system. We need to look at the type of applications that would use a 60 GHz system. There are different real-time and non-real-time applications which should take advantage of the new possibilities of 60 GHz systems (wider range of frequency and higher bandwidth available). The above attention results in this thesis to the proposal of a novel superframe structure, which is necessary to support different kinds of flows. In this chapter we discuss ways that we could modify the superframe of the MAC layer and the PNC functions of a MMW system in order to lessen some of the shortcomings.

We discuss the properties of the superframe introduced in 802.15.3 MAC in section 4.1. Then we would answer the question that how well the network is providing application requirements in terms of throughput, bandwidth and maximum possible delay. Following this, we provide the MAC model for the proposed adaptive channel-superframe allocation in section 4.2. Finally, we present a detailed explanation of the adaptive channel-superframe allocation algorithm.

4.1 IEEE 802.15.3 Superframe Analysis

The UWB MAC layer introduced in the standard has been considered as the first reference for designing 60 GHz MAC. According to this standard, different devices with different types of flows are distinguished based on real-time and non real-time transmissions. The TV set, camera, PC or a laptop are capable of sending high data rate real-time flows and others are capable of sending non-real-time flows as shown in Fig. 4.1. CAP section of the superframe pertains to non-real time flows (commands and/or asynchronous data) and CTAP section of the superframe pertains to real-time flows (commands, isochronous and asynchronous data) as depicted in Fig. 4.2.

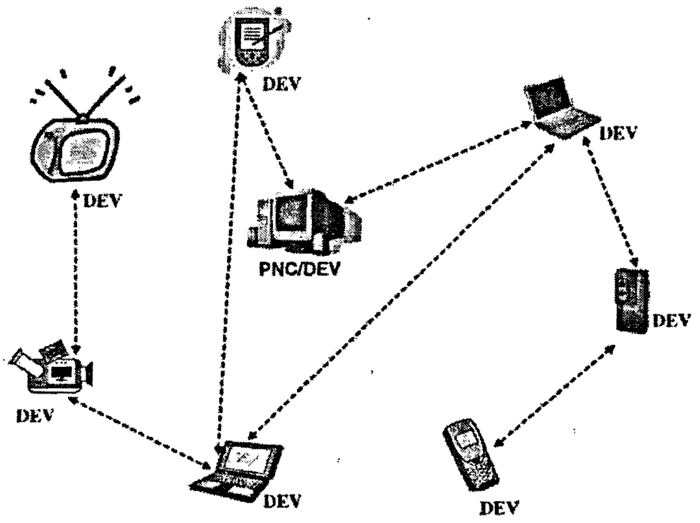

Figure 4.1: Different devices and flows in 60 GHz system.

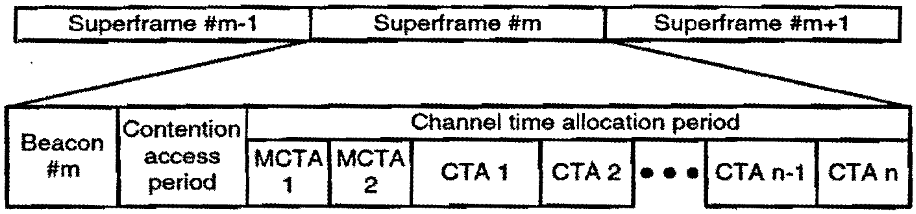

Figure 4.2: Superframe structure for IEEE 802.15.3.

Having a superframe split in two parts for real-time and non-real-time flows would support expected functions of IEEE 802.15.3 and IEEE 802.15.3c MAC layer. Yet, the performance results are very different from these two standards due to the higher data rates and directional transmission in MMW system. These two important factors are making 802.15.3 networks very different from 802.15.3c networks. As discussed previously, in many analysis, reduction in performance metric values are observed due to the use of the 802.15.3 MAC in place of the 802.15.3c MAC; hence, many studies emphasized on more considerations in the design of 802.15.3c MAC [5] [10] [12] [17] [19] in order to improve the performance metrics.

In this chapter, we would explain the properties of IEEE 802.15.3 MAC, such as throughput, delay and fairness, when it is used for MMW systems in section 4.1.1, 4.1.2 and 4.1.3 respectively.

4.1.1 Capacity

Throughput is an important metric in quality of service and particularly for real-time flows and multimedia transmission. In this section, we analyze the theoretical throughput (capacity) of the system. Superframe of 802.15.3 has two major properties. This includes the length of the superframe and proportion of CAP and CTAP. As discussed in Chapter 3, studies in [6] show that the length of the superframe can not be more than a threshold since it would increase the waiting time for all the flows. On the other hand, the proportion of CAP and CTAP is really important since a longer CAP results in a shorter CTAP and consequently, real-time flows would suffer. Also increase in CTAP would cause non-real-time flows to suffer from inaccessibility to channel. In particular, the effect of transmission in time slots spaced far from each other for non-real-flows is not well investigated so far.

In order to measure the MAC layer capacity we need to have a proper estimation of the following parameters:

- *Modulation*
  In the proposed channelization in [10], four channels are considered for the wide range of spectrum in a MMW system and, based on different modulations, different data rates are possible for PHY layer. OOK and BPSK provide 1.560Gbps, and QPSK provides 3.120Gbps. In addition to the higher capacity available in QPSK mode, the authors claim that QPSK is better to be chosen because of a better hardware compatibility among wireless devices. Therefore, we assume a 3.120Gbps data rate for the PHY layer.

- *Superframe length*
  As discussed in Chapter 3, different studies estimate different values for an optimum length of the superframe. In [6], the entire length of superframe is assumed to be 30msec. Earlier in this section, we have chosen ADD as the directional transmission method, which requires a superframe containing 64 simple superframes. Thus, if we assume the length of the simple superframe equal to 30msec, we would have an ADD superframe equal to $30 \times 64 = 1920$msec. In this case, both NRT and RT flows would suffer from a long waiting time because each flow should wait for the entire superframe duration in order to resume transmission. Therefore, the proposed superframe length

in [6] is not acceptable. Another proposition in [10] gives a 2msec period of time to the simple superframe and broadcasts this in 64 directions using ADD method. The result is a superframe with 128msec duration which is much less than the previous case. We would choose this value for the length of superframe in our MAC model since the shorter superframe length would lessen the waiting time of the flows.

- *Frame aggregation*
  Frame aggregation methods are not considered in capacity analysis since they provide small changes in maximum achievable throughput [4].

- *CAP and CTAP ratio*
  The bandwidth distribution in [6] is as follows:

$$CAP = 3\text{msec and } CTAP = 27\text{msec}.$$

Thus, we could obtain certain values as shown in Table 4.1 for the capacity analysis. This distribution results in less throughput for NRT flows compared to RT flows. This

Table 4.1: MAC model parameters

Total Capacity	3120Mbps
Superframe Length	30msec
Beacon Length	0.2msec which is small and ignored
CAP Length	$0.1 \times 30 = 3$msec
CTAP Length	$0.9 \times 30 = 27$msec
CAP Capacity	$\frac{3}{30} \times 3120 = 312$Mbps
CTAP Capacity	$\frac{27}{30} \times 3120 = 2808$Mbps

is not acceptable for the following reasons:

- Based on the studies in [8], the required bandwidth for all of the NRT flows in the piconet could be as much as 300Mbps. If we consider the handshake and acknowledgement packets in CAP which require at least 40% of the entire capacity [6], then the obtained value for NRT capacity (312Mbps) would not be enough.
- Based on the studies in [1], the required bandwidth for each of the NRT flows could be up to 50Mbps. Considering that an average number of NRT flows in the piconet equals to 10, there would be 500Mbps capacity required for CAP, which is far more than the obtained value.
- We have chosen the length of the superframe equal to 2msec, then the CAP based on [6], would be around 200μS. This short duration of CAP is sometimes even less than the RTT (Round-Trip Time) of a TCP connection. Therefore, this duration of CAP is not acceptable for NRT flows.

– As discussed in Chapter 3, a throughput definition for NRT flows is presented in (3.1) and the average number of successful requests using CSMA/CA protocol is defined in (3.2). Thus, we can obtain the duration of CAP (T_{CAP}) based on the number of flows (n), average number of successful request (N_r) and the renewal interval time (T_v) as shown in (4.1).

$$T_{CAP} = N_r \times T_v \times N. \tag{4.1}$$

Equation (4.1) shows that for a certain number of devices, a minimum length of CAP is required and for a certain value of CAP duration, a maximum number of devices could be supported in the system. Analysis in [16] show that for a CAP with the length of 4096μS and average number of successful requests per device equal to 2, only 10 devices could be supported in the CAP. We can conclude that shorter CAPs will reduce the number of supported devices to less than 10 which is not acceptable for home or office scale networks.

Therefore, we choose a different ratio between CAP and CTAP than what is proposed in [6] in order to obtain a longer CAP. In [10], the proposed superframe length is equal to 2msec wherein 0.2msec is for the beacon. We choose this length of superframe and assign 25% of the remaining time to CAP and 75% of the remaining time to CTAP. This ratio is slightly in favor of NRT flows compared to the previous case. In summary, we would have the certain values as shown in Table 4.2 for the capacity analysis.

Table 4.2: MAC model parameters.

Total Capacity	3120Mbps
Superframe Length	2msec
Beacon Length	0.2msec
Total Available Data Rate	$\frac{1800}{2000} \times 3120\text{Mbps} = 2808\text{Mbps}$
CAP Length	$0.25 \times 1800 = 450\mu\text{S}$
CTAP Length	$0.75 \times 1800 = 1350\mu\text{S}$
CAP Capacity	$\frac{450}{1800} \times 2808 = 702\text{Mbps}$
CTAP Capacity	$\frac{1350}{1800} \times 2808 = 2106\text{Mbps}$

According to the the bandwidth requirements of different applications provided in [1], real-time applications require 100Mbps on average. Therefore, the maximum possible number of real-time flows being served in the system would be $\lfloor 2106/100 \rfloor = 21$. This is a significant number of high data rate devices which could be supported in a wireless network.

4.1.2 Delay

There are two types of delays to be considered in 802.15.3 MAC:

1. *Registration delay*: Both real-time and non-real-time flows need to register in PNC before sending any data. This process happens in CAP using CSMA/CA protocol. A real-time flow would receive beacon information and start sending data in CTAP and a non-real-time flow (i.e., TCP connections for web browsing) would send asynchronous data in the CAP. We analyze both cases in the following.

   (a) Real-time flow: Based on 802.15.3 MAC, a real-time flow sends a registration request in the CAP and waits for a long time to start transmission. Fig. 4.3 shows the process.

   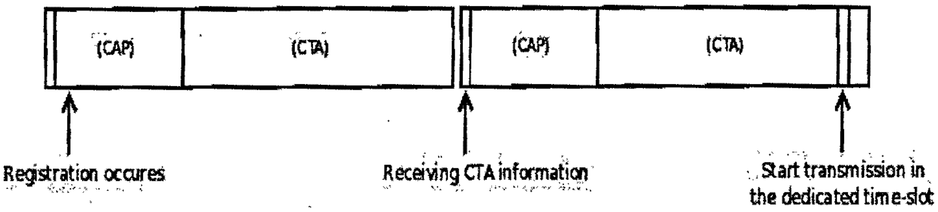

   Figure 4.3: Registration of a real-time flow.

   In the worst case scenario, the registered application should wait until it receives the next beacon containing CTA information and starts transmission in the following superframe. Therefore, there would be

   $$Registration\ delay = 2 \times superframe\_length \tag{4.2}$$

   before transmission starts.

   (b) Non-real-time flow: A non-real-time flow need to synchronize with beacon since CSMA/CA protocol takes care of the transmission, there is the same registration delay for non-real-time flows.

2. *Protocol delay*: We define protocol delay as the delay imposed by MAC layer protocols. The MAC layer of IEEE 802.15.3 imposes delay on TCP flows due to the CTA waiting time in superframe. In order to clarify the causes of protocol delay, we explain the properties of CSMA/CA and TCP in the following.

   (a) CSMA/CA is responsible to control the physical layer in accessing the channel. The transmission sequence in CSMA/CA protocol is "RTS-CTS-DATA-ACK" and, based on the studies in [16] and [20], a successful transmission period is the time for a sequence of transmissions and delays as shown in (4.3).

   $$T_{success} = RTS + SIFS + CTS + SIFS + DATA + SIFS + ACK + BIFS, \tag{4.3}$$

where SIFS is the short inter-frame space and BIFS is the back-off inter-frame space. And a failed transmission period in CSMA/CA is shown in (4.4)

$$T_{fail} = RTS + BIFS \tag{4.4}$$

The failed transmission period in (4.4), is related to the function of CSMA/CA protocol. Hence, this is not an imposed delay on TCP connections. We would take this delay into account in designing the ACSA MAC later. However, an imposed delay may be possible if the CAP expires and an unfinished transmission is going on. Then, there would be four cases of failure in transmission, which are RTS error, CTS error, data error and ACK error. The consequence delays are:

$$Delay_{f1} = RTS + BIFS$$

$$Delay_{f2} = RTS + SIFS + CTS + BIFS$$

$$Delay_{f3} = RTS + SIFS + CTS + SIFS + DATA + BIFS$$

$$Delay_{f4} = RTS + SIFS + CTS + SIFS + DATA + SIFS + ACK + BIFS$$

Above delays are possible for non-real-time flows when the CAP expires, yet they could be ignored since they are so small and only occur once during a superframe cycle. However, another important reason of delay is the sliced superframe, which increases the probability of collision in CSMA/CA protocol. CSMA/CA would allow all the flows to try to access to channel and if collision happens for a flow, it will wait for a certain period of time based on back-off mechanism. Therefore, as the time pass, all of the flows would find a chance to transmit and the probability of a successful transmission of a flow would increase. This means that when the traffic load is high the longer CAP results in higher probability of successful transmission. However, UWB superframe would keep this probability very low by slicing the CAP. At the beginning of a CAP all the flows, which are waiting to transmit would try to access to channel. Therefore, many collisions would happen. Then, many of the flows would keep out of transmission mode based on CSMA/CA protocol and then some flows would be able to successfully transmit over time. In this case, if the CAP expires, all the waiting nodes must wait for the next coming CAP and meanwhile, all the flows would come out of back-off mode and try to transmit at the beginning of the next coming CAP, which again results in many collisions.

(b) TCP protocol in transport layer is responsible to deliver the packets in a reliably established connection. Many applications rely on the connection-oriented services (such as HTTP, FTP, and pure IP) which are offered by TCP. When these applications are launched, the TCP stack on the local device must establish a connection with the TCP stack on the destination device. The handshake process is based on three steps. Fig. 4.4 is a graphical illustration of TCP handshaking. The handshake process involves the following steps:

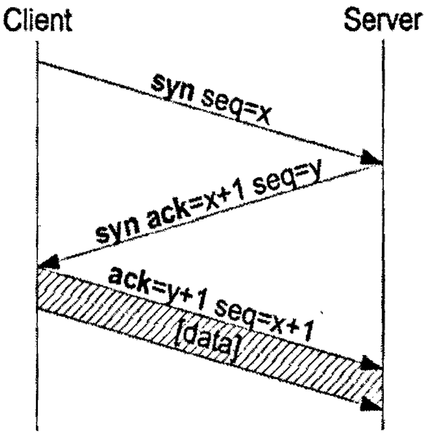

Figure 4.4: TCP handshaking

- Source sends synchronization packet containing its TCP sequence number and maximum segment size to destination.
- Destination responds by sending acknowledgment packet containing its sequence number and maximum segment size to source.
- Source acknowledges receipt of the sequence number and segment size information.

During TCP transmission, this process remains the same. There would be an acknowledgment in response to each transmitted packet. The delay of this process is defined as RTT. This is the time from the SYN packet is sent until the time that ACK packet is received. Some studies reveal that RTT for SYNACK and ACK packets could best predict the average RTT during the entire transmission [21]. According to this study, the average RTT for SYNACK and ACK packets is 145.7 msec. Having this high proportion of time as RTT delay for making a TCP connection would considerably affect web browsing specially when using IEEE 802.15.3 MAC layer. The reason is the follows. Consider that a source decides to send a SYN packet to the destination, then it must wait for the next CAP in the superframe cycle. In the worst case, it must wait for the entire duration of CTAP which is much longer than the entire CAP duration. Thus, a long delay is possible prior to sending a packet. Moreover, same delay exists when the source receives an ACK packet. The ACK packet may arrive earlier than the next CAP in the superframe cycle. Thus, a longer delay, which could be equal to the entire CTA duration is possible while receiving the ACK.

Therefore, sending and receiving a TCP packet in IEEE 802.15.3 takes more time than a normal RTT. Taking the superframe imposed delay into account, we can

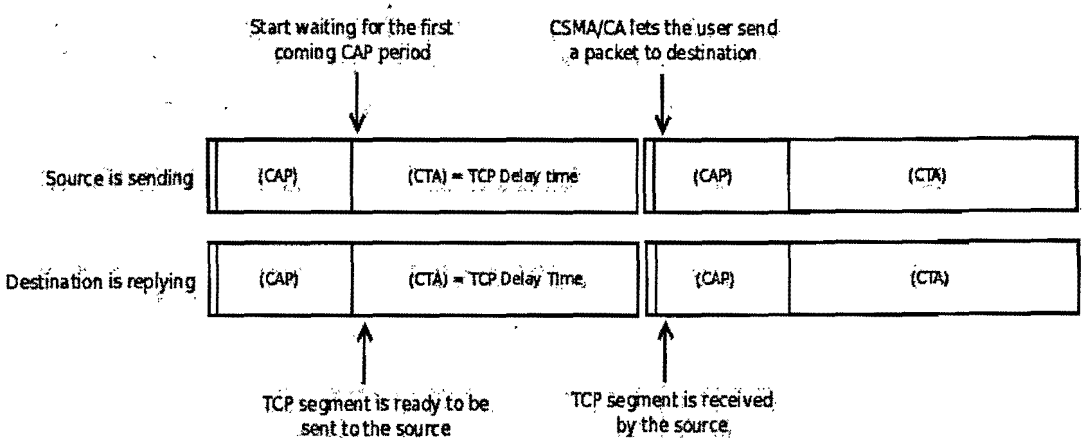

Figure 4.5: Protocol delay in 802.15.3 superframe.

derive the new average RTT as given in (4.5)

$$New\ RTT = 2 \times \left[\frac{Normal\ TCP\ RTT}{2} + \frac{CTA\ Duration}{Superframe\ Length} \times \frac{CTA\ Duration}{2}\right] \tag{4.5}$$

Considering an average value of normal $RTT = 145.7$msec [21], we would have new RTT for the two previously discussed superframe durations as the following:

- CTA Duration = 27msec and Superframe Length = 30msec

$$\text{New Average RTT} = 2 \times (\frac{145.7}{2} + \frac{27}{30} \times \frac{27}{2}) = 170\text{msec}$$

- CTA Duration = 1.35msec and Superframe Length = 2msec

$$\text{New Average RTT} = 2 \times (\frac{145.7}{2} + \frac{1.35}{2} \times \frac{1.35}{2}) = 146.61\text{msec}$$

For Internet applications, requests are commonly around 500 bytes [22]. Therefore, increment in TCP RTT delay should significantly impact loading small objects and it means that the page load is bottlenecked.

4.1.3 Fairness

Fairness is another consideration in many performance studies. Particularly in distributed systems, where a set of resources is to be shared by a number of users, fair resource allocation is important. In computer networks, we can study fairness of different metrics. The popular allocation metrics to study in computer networks are response time, throughput, power and TCP variable window flow control [23]. We need to find an appropriate allocation metric applicable for MMW system applications. Based on superframe structure in IEEE 802.15.3

MAC, a discrimination between real-time and non-real-time exists. Here, we assume a certain number of flows in the system and analyze the superframe to figure out whether it is fair in allocating bandwidth to different flows. Therefore, we choose the bandwidth as the allocation metric. To measure fairness, we use fairness index defined in [23]. If a system allocates sources to n contending users, such that the i^{th} user receives an allocation x_i, then the following fairness index is defined for the system:

$$\text{Fairness Index} = f(x) = \frac{[\sum_{1 \leq i \leq n} x_i]^2}{n \sum_{1 \leq i \leq n} x_i^2} \quad \text{where} \quad x_i \geq 0. \tag{4.6}$$

This index measures the "equality" of resource allocation x. If all users get the same amount, i.e., $x_i s$ are all equal, then the fairness index is 1, and the system is 100% fair. As the disparity increases, fairness decreases and a scheme which favors only a selected few users has a fairness index near 0. This fairness index is dimensionless and independent of scale and is bounded between 0 and 1, and it is continuous so that any slight change in x_i changes the index. In our case which we have different types of flows we need to measure the inter-class fairness. As discussed earlier, NRT flows need 50Mbps bandwidth on average and RT flows need 100Mbps bandwidth on average. Therefore, we define a weight for NRT flows in order to assume equal type of flows with equal bandwidth requirements. We multiply the measured throughput of NRT flows by two in the fairness equation to be able to deal with all the throughput values equally.
Therefore, substituting the values of Table 4.2 for x_i and assuming 10 RT and 10 NRT flows in the system, we obtain:

Inter-class fairness index of bandwidth allocation for 802.15.3 superframe

$$= \frac{(70.2 \times 2 \times 10 + 210.6 \times 10)^2}{20 \times ((70.2 \times 2 \times 10)^2 \times 10 + 210.6^2 \times 10)} = 0.62 \quad \text{which is equal to } 62\%$$

4.1.4 Acknowledgment

The observations in [6] show that the overhead due to the acknowledgment messages are marginal as shown in Table 4.3. This means that efforts to reducing the quantity of acknowledgment messages can gain only a slight increase in throughput. Moreover, there is a new proposed acknowledment method by the same author in order to improve throughput. This method is splitting a CTA slot in down-link and up-link channels for use of TCP flows [6] [8]. This method has a few problems described below:

- This method means that we need to compensate 40% of CTA time for acknowledgment packet transmission. This would highly decrease the available bandwidth for real-time flows. The same authors propose another method to eliminate this problem, which is using CAP for transmission of acknowledgment packets. This method would lead to the trade off between CAP and CTAP. As we increase the CAP up to a certain threshold, the acknowledgment traffic is well served and throughput increases. Yet

Table 4.3: Percentage of acknowledgment traffic versus data traffic [6].

ACK Policy	Percentage of Total Traffic
No-ACK	0%
Immediate-ACK	0.92% which is small and ignored
Delay-ACK-10	0.59%
Delay-ACK-20	0.47%
Delay-ACL-30	0.43%

increasing CAP length means decreasing CTAP length and is equal to compensate some bandwidth of real-time flows for acknowledgment packet transmission. Therefore, increasing the CAP length more than a threshold, would decrease the average throughput among real-time and non-real-time flows.

- The nature of TCP flows is to transmit data in various sizes per transmission cycle, which means that a TCP flow may transmit an object from a few Kbytes upto hundreds of Kbytes in size. Thus, if CTA slot registration is required for sending very small packets, i.e., a 2Kbyte packet, there would be a bottleneck in the system since the registration delay is much more than the transmission time.

- Any method which needs more signaling and controlling messages between PNC and other devices, would provide overhead and complexity in MAC protocol. In this method, every device using CSMA/CA protocol should keep the TCP data in a buffer and send during a CTA slot using TDMA protocol. Therefore, PNC is required to inform other devices about this change and other devices should be able to modify the MAC protocol in order to match these changes. Hence, complexity is another problem of this method.

4.2 Channel-Superframe Allocation Algorithm

In this section, we describe an adaptive channel-superframe allocation(ACSA) algorithm. The goal of this algorithm is to lessen the deficiencies of 802.15.3 superframe. Improvement in throughput, fairness and delay are the desired achievements in our design. We propose a new superframe in compliant with MMW system channels while not interfering with basic functionalities of the MAC layer. Based on our analysis, the fairness of bandwidth allocation while using 802.15.3 superframe for MMW system is poor, as well as the possible throughput of non-real-time flows. However, the wide range of available bandwidth utilized by a new structure of superframe would be able to compensate these shortcomings..

4.2.1 System Model for Adaptive Channel-Superframe Allocation

The MAC layer of MMW system in our study is modeled based on 802.15.3 MAC layer, yet we made some modifications in the design of superframe and provided an algorithm in order to manage different flows transmitted using this superframe. Detailed parameters of MAC layer in simulation environment are discussed in Chapter 5. The physical layer of MMW system works in a particular situation in terms of attenuation and frequency. We discuss next the physical layer and channel model related to our studies.

Channel Model

Among wireless systems, attenuation is a very important factor of physical layer analysis which indirectly affects MAC layer design. MMW systems have very high attenuation as described earlier. Therefore, we presume that MMW systems would mostly be available for devices inside home and office. Accordingly, we have analyzed some applications running at a home or an office, which leads to finding a minimum required bandwidth for each single application. In summary, we designed our proposed algorithm in order to obtain the following:

- Benefit from the entire wide range of available bandwidth in 60 GHz systems.
- A dedicated channel for non-real-time application (CAP).
- Reduce channel access time for different kinds of applications and traffic.

Having the above as requirements of MAC design and based on the proposed channelization in [10], we assume four channels out of the wide range of frequency in 60 GHz system (57 GHz to 66 GHz) as shown in Fig. 4.6 . The above mentioned channelization supports four channels over 9 GHz bandwidth. Basic parameters and data rate modes of this channelization are presented in Table 4.4 and 4.5.

4.2.2 The Algorithm

We mentioned that devices intended to operate on 60 GHz systems need a minimum bandwidth equal to 50Mbps for NRT flows and 100Mbps for RT flows. However, the bandwidth distribution of 802.15.3 superframe is not fair as discussed earlier. Moreover, one characteristic of a TCP flow is the round-trip-time, which could be increased significantly by a sliced contention access period and consequently decrease the service quality. In addition, we know that real-time and non-real time applications must operate one at a time distinguished by CAP and CTAP and can not be multiplexed. Here we analyze the 802.15.3 superframe in order to find an optimum ratio between CAP and CTAP. As shown in (4.1), the length of CAP could be obtained based on the number of flows (n), average number of successful requests (N_r) and the renewal interval time (T_v). The results in [16] show that for $N = 10$, $N_r = 2$ could be achieved. Thus, we need to find the value of T_v for our analysis. T_v is the

Channel Number	Low Freq. (GHz)	Center Freq. (GHz)	High Freq. (GHz)	Nyquist BW (MHz)	Roll-Off Factor
A1	57.200	58.240	59.280	1664	0.25
A2	59.280	60.320	61.360	1664	0.25
A3	61.360	62.400	63.440	1664	0.25
A4	63.440	64.480	65.520	1664	0.25

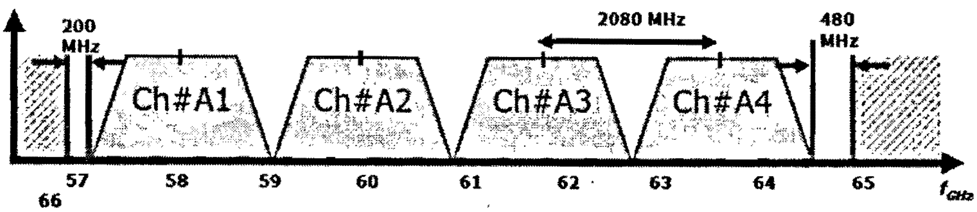

Figure 4.6: MMW Channelization adopted from [10].

Table 4.4: Basic Parameters for MMW System Channel [10].

Modulation Scheme	OOK	BPSK	QPSK
Bits per symbol	1	1	2
Detection	Non-coherent/Coherent	Coherent	Coherent
FEC	Reed Solomon (255,239)	Reed Solomon (255,239)	Reed Solomon (255,239)
PHY header	Shorten Reed Solomon (32, 16)	Shorten Reed Solomon (32, 16)	Shorten Reed Solomon (32, 16)
PHY payload data rate	1.560 Gbps	1.560 Gbps	3.120 Gbps

Table 4.5: Data Rate Modes for MMW System Channel [10].

Mode	Modulation	Detection	FEC Scheme	PHY-SAP payload rate [Gbps]
1.1	OOK	Noncoherent	RS(255,239)	1.560
1.2	OOK	Coherent	RS(255,239)	1.560
2	BPSK	Coherent	RS(255,239)	1.560
3	QPSK	Coherent	RS(255,239)	3.120

renewal interval which is made up of idle slots followed by a successful or failed transmission attempt as shown in (4.7).

$$T_v = (N_c + 1).T_{idle} + N_c T_f + T_s, \tag{4.7}$$

where T_{idle} is the duration of the idle period after last transmission, N_c is the number of failed transmissions in a renewal interval, T_f and T_s are the duration of a failed and successful transmission respectively. Earlier we defined T_s and T_f in (4.3) and (4.4). We need to assume a certain number of failed transmissions, (N_c) in order to obtain the value of T_v. We choose different values for N_c from 0 to 10 which simulates the increasing traffic in the network. We also ignore the T_{idle} since it is so small compared to T_s and T_f.

- $N_c = 0$

$$\begin{gathered} T_{v1} = T_S = RTS + SIFS + CTS + SIFS + DATA + SIFS + ACK + BIFS \\ \text{Based on [16], we consider RTS = CTS = ACK = 14 Bytes,} \\ \text{DATA = 100 Bytes, SIFS = } 10\mu\text{S and BIFS = } 15.8\mu\text{S, then we would have:} \\ T_{v1} = \frac{(14 \times 3 \times 8) + 100}{2808\text{Mbps}} + (3 \times 10\mu\text{S}) + 15.8\mu\text{S} \\ T_{v1} = 46\mu\text{S} \end{gathered} \tag{4.8}$$

- $N_c = 1$

$$\begin{gathered} T_{v2} = T_{v1} + T_f = T_{v1} + RTS + BIFS \\ T_{v2} = 0.92\text{msec} + \frac{14 \times 8}{2808\text{Mbps}} + 15.8 \\ T_{v2} = 77.8\mu\text{S} \end{gathered} \tag{4.9}$$

- $N_c = 2$

$$\begin{gathered} T_{v3} = T_{v1} + 2 \times T_f = T_{v1} + 2 \times (RTS + BIFS) \\ \text{The CSMA/CA protocol would increase BIFS after each failed transmission.} \\ \text{Therefore, we assume the second BIFS is twice of the previous} \\ \text{BIFS [16] then we would have } T_{v3} \\ T_{v3} = 125.2\mu\text{S} \end{gathered} \tag{4.10}$$

- $N_c = 5$

$$T_{v4} = T_{v1} + 5 \times T_f = T_{v1} + 5 \times (RTS + BIFS)$$
$$\text{Similar to the previous equations, we would have } T_v$$
$$T_{v4} = 362.4\mu\text{S} \tag{4.11}$$

Based on the above values for T_v, we could obtain T_{cap} by having $N = 10$ and $N_r = 2$.

$$T_{CAP} = N_r \times N \times T_v$$
$$Tv = 46\mu\text{S then } T_{CAP} = 2 \times 10 \times 46\mu\text{S} = 0.92\text{msec}$$
$$Tv = 77.8\mu\text{S then } T_{CAP} = 2 \times 10 \times 77.8\mu\text{S} = 1.55\text{msec}$$
$$Tv = 125.2\mu\text{S then } T_{CAP} = 2 \times 10 \times 125.2\mu\text{S} = 2.5\text{msec}$$
$$Tv = 362.4\mu\text{S then } T_{CAP} = 2 \times 10 \times 362.4\mu\text{S} = 7.24\text{msec} \tag{4.12}$$

Therefore, the duration of CAP could be from 0.92msec upto 7.25msec. This range of CAP is obtained based on the different average number of failed transmissions (N_c) for a TCP flow. In addition, the number of devices initiating TCP connections in the system could be from zero to 10 or 20 or even more. Assuming different values for the average number of devices, the minimum and maximum length of CAP could be obtained as the following:

$$\text{Minium length of CAP is when } N = 0$$
$$\text{then } T_{CAP} = 0\text{msec}$$
$$\text{Maximum length of CAP is when } N = 20 \text{ and } T_v = 362.4\text{msec}$$
$$\text{then } T_{CAP} = 14.5\text{msec} \tag{4.13}$$

In the analysis above, we have a guaranteed quality of service defined as a fixed number of successful request per flow per CAP ($N_r = 2$). Then we changed other parameters to discover a proper length for CAP. Precisely, with a variable number of devices (N) and a variable average number of failed transmissions per flow (N_c), different values for length of CAP from 0msec to 14.5msec is obtained. We should also mention that the length of CAP and N_c are related. By increasing the length of CAP, N_c would decrease since the longer CAP would increase the probability of successful transmission in CSMA/CA protocol and as a result the number of failed transmission (N_c) would become less. In conclusion, a guaranteed quality of service is possible if we could adapt the length of CAP in different situations. As it can be observed from (4.13), the range of this adaptability could be wider. In other words, a variable ratio between CAP and CTAP durations is needed in order to keep the quality of service for NRT flows. Even though the variable length of CAP would favor NRT flows, there would be some drawbacks as we explain in the following:

1. Considering a fixed length for superframe, there must be a maximum length of CAP. Otherwise, the entire superframe would be assigned for NRT flows. In this case, there is no possibility to increase the CAP length up to the desired value and consequently quality of service is not guaranteed.

2. Considering a fixed length for the superframe, a longer CAP results in a shorter CTAP, which consequently diminishes the quality of service of RT flows.

3. Considering a variable length for superframe, a longer CAP results in a longer superframe cycle and if we keep the ratio between CAP and CTAP a fixed value, we obtain a very long superframe length. For example, keeping the ratio as a fixed ratio (25% and 75%) and CAP = 14.5μS, we would have a superframe with length of $14.5 + 43.5 = 58$msec. This long superframe cycle would result in longer waiting times for both RT and NRT flows in each cycle, which would diminish the throughput and quality of service of real-time flows. Also longer waiting times require higher buffer sizes in devices.

4. Considering a variable length for superframe requires more signaling and control messages. Each time the length of superframe changes, all the transmitting and receiving devices in the piconet should be notified. For example, each time a device is joining the piconet or leaving the piconet, all other devices should be notified from the changes in timing because a shift in time slot of each flow would be necessary.

Therefore, **we present the idea to separate CAP and CTAP by channels** as shown in Fig. 4.7, **instead of separating them into different times.** The reasons are the followings.

1. Analysis in (4.8)-(4.11) shows that a big share of time is spent on BIFS (wasted time) for each transmission interval using CSMA/CA protocol. This means that no matter how much the capacity is, the waste of time is constant in CSMA/CA protocol. The ratio of BIFS over T_v for different N_c's are as follows:

   - $N_c = 0$

$$\frac{BIFS}{T_v} = \frac{15.8\mu\mathrm{S}}{46\mu S} = 34\%$$

   - $N_c = 1$

$$\frac{BIFS}{T_v} = \frac{15.8\mu S + 31.6\mu\mathrm{S}}{77.8\mu S} = 60\%$$

- $N_c = 2$

$$\frac{BIFS}{T_v} = \frac{15.8\mu S + 31.6\mu S + 47.4\mu S}{125.2\mu S} = 75\%$$

- $N_c = 5$

$$\frac{BIFS}{T_v} = \frac{15.8\mu S + 31.6\mu S + 47.4\mu S + 63.2\mu S + 79\mu S + 94.8\mu S}{362.4\mu S} = 91\%$$

It can be seen that the wasted time could be as much as 90% of the entire transmission interval (T_v). Using CSMA/CA protocol, if traffic load is high and collisions happen most of the time, then the wasted time could be as much as 100% of the entire time. Therefore, when the traffic load is high, the less available capacity dedicated to NRT flows result in the less wasted bandwidth. For example, if NRT flows use a channel with 2000Mbps capacity, the wasted bandwidth could be upto 1800Mbps and even more; but if the NRT flows use a channel with 1000Mbps capacity, the wasted bandwidth could be upto 1000Mbps which is much less than the previous case. Therefore, we need to find the maximum necessary capacity needed for NRT flows and assign it to NRT flows. This method could be implemented by separating the entire bandwidth into different channels and assigning one of them to NRT flows.

2. As discussed earlier, a MAC with a fixed or variable length of superframe could not guarantee the quality of service for different possible situations. Because there is always a limit in the length of superframe or a limit in the length of CAP and CTAP. Thus, dedicating a channel for NRT flows would eliminate the time limits such as the maximum length of CAP or CTAP. In this case, bandwidth is split into channels and each channel is assigned to a CAP or CTAP.

This idea of separating CAP and CTAP by channels could not be applied for 802.15.3 for the following reason. In contrary to the wide range of bandwidth in 802.15.3c (3120Mbps), the maximum capacity in 802.15.3 is 55Mbps. Thus, a frequency division scheme could not be acceptable since it would decrease the available bandwidth in the channel to less than 55Mbps, which could not support the data rate of wireless applications. However, the idea of separating CAP and CTAP by channels could be applied on MMW system because the wide range of bandwidth in IEEE 802.15.3c provides the chance of using a frequency division scheme. In this case, the superframe has the following properties.

1. There is one superframe for each one of the channels. For example, if the spectrum is divided into four channels, there would be four superframes in MAC layer.

2. The length of superframe is defined as the time difference between two consecutive transmitted beacons from PNC.

3. The constraints in the length of CAP or CTAP could be resolved since an entire superframe of channel# 4 is dedicated to contention access period. In contrast to standard MAC, a longer CTAP is not equal to shorter CAP.

4. The superframe would not be sliced and consequently, the probability of starving for channel access would be less.

5. A channel/superframe is dedicated for each type of flow and we have them (RT and NRT) served simultaneously.

6. The superframe structure is adaptive based on the number and types of the flows in the networks.

We discuss the adaptive superframe structure and improvements in performance metrics.

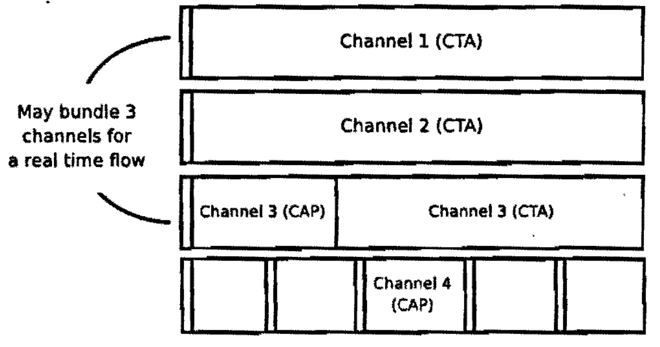

Figure 4.7: Proposed Superframe Structure.

802.15.3c Superframe Design

Based on the channelization described previously in section 4.2.1, we assign one of the four channels for CAP, one channel shared for CAP and CTAP and two channels for CTAP as shown in Fig. 4.7. Consequently, we need to have four different superframes for 4 channels. The length of superframes is not necessarily equal. As mentioned earlier, there are four similar channels. Therefore, the available bandwidth for each channel is 3.120Gbps/4 = 780Mbps. The proposed superframe is taking advantage of these four channels and may stand in one of the four different states as shown in Fig 4.8. The **A** state is when non-real time flows occupy both channel# 4 and # 3. The **B** state is when there are real-time and non-real-time flows sharing the channel# 3. The **C** state has no shared channel. The **D** state has three CTA channels and a shared channel. Among these states, **A** has the more bandwidth for non-real-time flows and **D** has the less bandwidth respectively. Depending on the number of real-time and non-real time applications, one of these four states may occur for the system. In the above channelization, each channel has 780Mbps capacity. Based on the studies in [11], non-real-time applications in a MMW system require 50Mbps on average. However, this is not enough bandwidth for the case that channel# 4 would be over-utilized. In our proposed MAC, we considered the flexibility to change the state of channel# 3 and channel# 4 to shared channels in order to overcome the bandwidth shortage

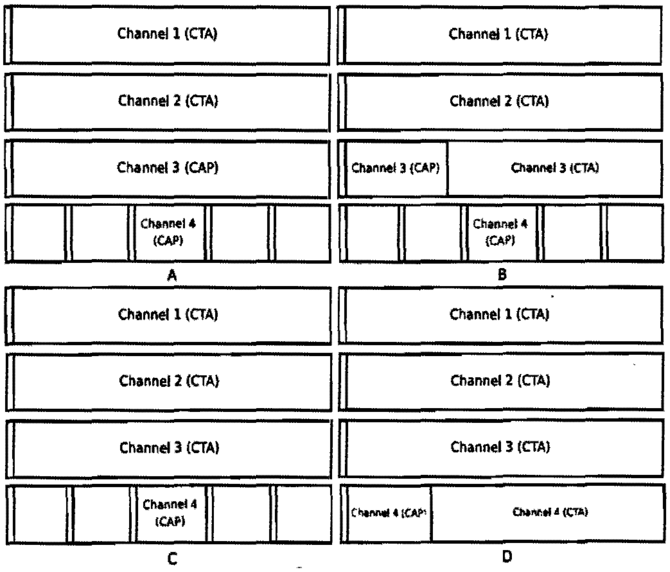

Figure 4.8: Four States of 802.15.3c Superframe.

in different situations. PNC is responsible to change the superframe structure and provide this mentioned flexibility, which is explained in the next section.

PNC Control Algorithm

PNC is responsible to control all the channels and manage the timings based on the traffic load and occupied bandwidth in each channel. Any application which wants to join the network must be registered in PNC. The request could be sent through channel# 4 (CAP). The duration of the superframe is defined as the time difference between two consecutive transmitted beacons from PNC. The PNC should inform all the nodes about the frequency of sending beacons. Following is the function of PNC upon receiving a request from RT or NRT flows.

- Upon receiving a time-slot request from a RT flow, PNC would examine the minimum length of time needed for the flow based on the required bandwidth in the request. PNC would assign the flow to channel# 1, # 2 or # 3 according to the available bandwidth in each of the channels. If there is not enough bandwidth in one of them, PNC may bundle 2 or 3 channels to prepare more bandwidth for the flow.

- Upon receiving a request for using CAP, PNC would assign the flow to the channel #3 or #4 based on the available bandwidth in each of them.

The step-by-step description of the PNC algorithm is presented in the following pseudo-code, which explains the decisions made by PNC throughout the entire process. This algorithm is designed to satisfy the following important quality measures.

1. Real time video streaming and gaming have the highest priority to be served.
2. Some applications (such as interactive gaming, photo share, random access for video and audio, uploading and downloading for non-real time systems) should be able to have a reasonable and minimum bandwidth (around 50Mbps) and have access to channel in the less possible time.

Algorithm 1 Piconet Coordinator Decision Making

Require:
F: An incoming flow - B: Required bandwidth for F
Ensure:
Required and fair bandwidth is given to F in an acceptable time
if F is NRT **then**
    Go to NRT flows procedure
else
    Go to RT flows procedure
end if
return true

Algorithm 2 NRT Flow Procedure

```
Check Channel# 3 and 4 (CAP channels) status
if Channel 4 is fully used by NRT flows then
    if Channel 3 is fully used by NRT flows then
        if Channel 3 is partially used by RT flows then
            Split channel 3 for use of NRT and RT flows
            Add F to channel 3
            Superframe State is B {No NRT flow is starving for channel}
            return true
        else
            Add F to channel 3
            Superframe State is B {No NRT flow is starving for channel}
            return true
        end if
    else
        Add F to channel 4
        Superframe State is A {Some NRT flows are starving for channel}
        return true
    end if
else
    Add F to channel 4
    Superframe State is C or D {No NRT flow is starving for channel}
    return true
end if
```

Algorithm 3 RT Flow Procedure

```
Check Channel# 1, 2 and 3 (CTA channels) status
if Channel 1 is fully used by RT flows then
  if Channel 2 is fully used by RT flows then
    if Channel 3 is fully used by RT flows then
      if 10% reserved bandwidth for RT flows in channel 4 is fully used then
        if Priority of F is higher than any RT flow in channel 1,2,3,4 then
          Deduct from time of the flow with less priority
          Add F in place of the victim flow {Some RT flows are starving for channel}
          return true
        else
          Reject F {This RT flow registration request is denied}
          return false
        end if
      else
        Add F to to channel 4 in place of 10% reserved bandwidth
        Superframe State is D {This RT flow may be starving for channel}
      end if
    else
      Push some NRT flows to channel 4 to provide bandwidth in channel 3
      Split channel 3 for use of NRT and RT flows
      Add F to channel 3
      Superframe State is B {Some NRT flows are starving for channel}
      return true
    end if
  else
    Add F to channel 2
    Superframe State is A,B,C or D {No RT flow is starving for channel}
    return true
  end if
else
  Add F to channel 1
  Superframe State is A,B,C or D {No RT flow is starving for channel}
  return true
end if
```

The ACSA algorithm follows a particular pattern in assigning the channels and adapt the superframe structure accordingly. This pattern could be summarized in the following items:

1. Any application would send the first request through channel# 4 and continue syncing with PNC through the assigned channel later.
2. Real-time flows have the highest priority and PNC may push down some of the non-real-time flows from channel# 3 to channel# 4 if there was a shortage in bandwidth for real-time flows. (Superframe state is **B** or **C**)
3. PNC may change the state of channel# 3 to complete CTA period and make channel# 4 as a shared channel if there was still shortage in bandwidth for real-time flows. (Superframe state is **D**)
4. Sending beacons on channel# 4 is more frequent in order to reduce waiting times for synchronization of incoming flows as well as flows while being served.

A graphical representation of ACSA is shown in Fig. 4.9, which explicitly defines the ACSA algorithm using an adaptive superframe structure. Based on the ACSA rules, RT flows are able to occupy three channels. Each time an incoming RT flow is added to the superframe, an increase in duration of superframe cycle is necessary. However, this increase in the length of superframe should not violate the bandwidth requirements of the other RT flows. Equation in (4.14) shows how PNC should verify the possibility of increasing the length of superframe.

$$\frac{S_i \times T_s}{\text{The Increased Length of Superframe}} = \text{Minimum bandwidth requirement of flow } i, \tag{4.14}$$

where S_i is the throughput of flow i and T_s is the duration of the time slot assigned to the flow i. If this verification is passed for all the existing flows, then an increase in length of superframe is possible and the new flow could be added to the superframe. Besides, if one of the flows could not pass the above verification and consequently there was no possibility to add the incoming flow to channel# 1, # 2 or # 3, adaptive channel-superframe allocation algorithm would dedicate a maximum 10% of bandwidth in channel# 4, to RT flows. On the other hand, NRT flows can use a guaranteed bandwidth equal to 90% of channel# 4. This idea would resolve the problem of long waiting time for NRT flows. The length of the superframe assigned to channel# 4, which is a CAP, is calculated based on (4.15).

$$T_{CAP} = N_r \times \frac{T_{DATA}}{S} \times N, \tag{4.15}$$

where N_r is the average number of successful requests per flow per superframe. PNC could estimate this number at any point of time using the channel# 4. T_{DATA} is the duration of payload transmission excluding the handshakes and failed transmissions. S is the throughput of the system and N is the number of NRT flows in the system. If NRT flows demand for more bandwidth, ACSA is flexible to assign the entire channel# 3 to NRT flows as long as there is no demand for that share of bandwidth from RT flows. Otherwise, NRT flows would not be able to use the entire duration of superframe of channel# 3 and may share the superframe with RT flows. And if more RT flows demand for bandwidth, NRT flows would be pushed down to channel# 4 and the entire duration of superframe of channel# 3 would be utilized by RT flows.

In summary, we set certain parameters for 802.15.3 MAC and analyzed the performance of a MMW system in terms of capacity, delay and fairness using 802.15.3 MAC. We investigated about the applications of a MMW system (NRT and RT flows) and analyzed the performance of the system. Taking the type of the applications of a MMW system into account, we discussed the existing shortcomings in 802.15.3 MAC. We explained RT an NRT required quality of services and discussed on how the sliced superframe and the limits in the length of superframe, CAP and CTAP are preventive measures in providing the quality of service for RT and NRT flows. Then, we presented a new unsliced superframe structure in this chapter, which eliminates the limits in the length of superframe, CAP and CTAP. The proposed superframe is adaptive and is controlled by an algorithm in PNC. We called this algorithm adaptive channel-superframe allocation algorithm. We explained some specific reasons which based on them, ACSA could provide better throughput for NRT flows by compensating some available bandwidth for RT flows. We also discussed on how the unsliced superframe could improve the channel access delay in a MMW system.

In the next chapter, we would discuss the system parameters in order to simulate our proposed MAC with the adaptive channel-superframe allocation and superframe structure. We would simulate both the standard MAC and ACSA MAC and compare the results to verify the benefits and drawbacks of our proposed algorithm.

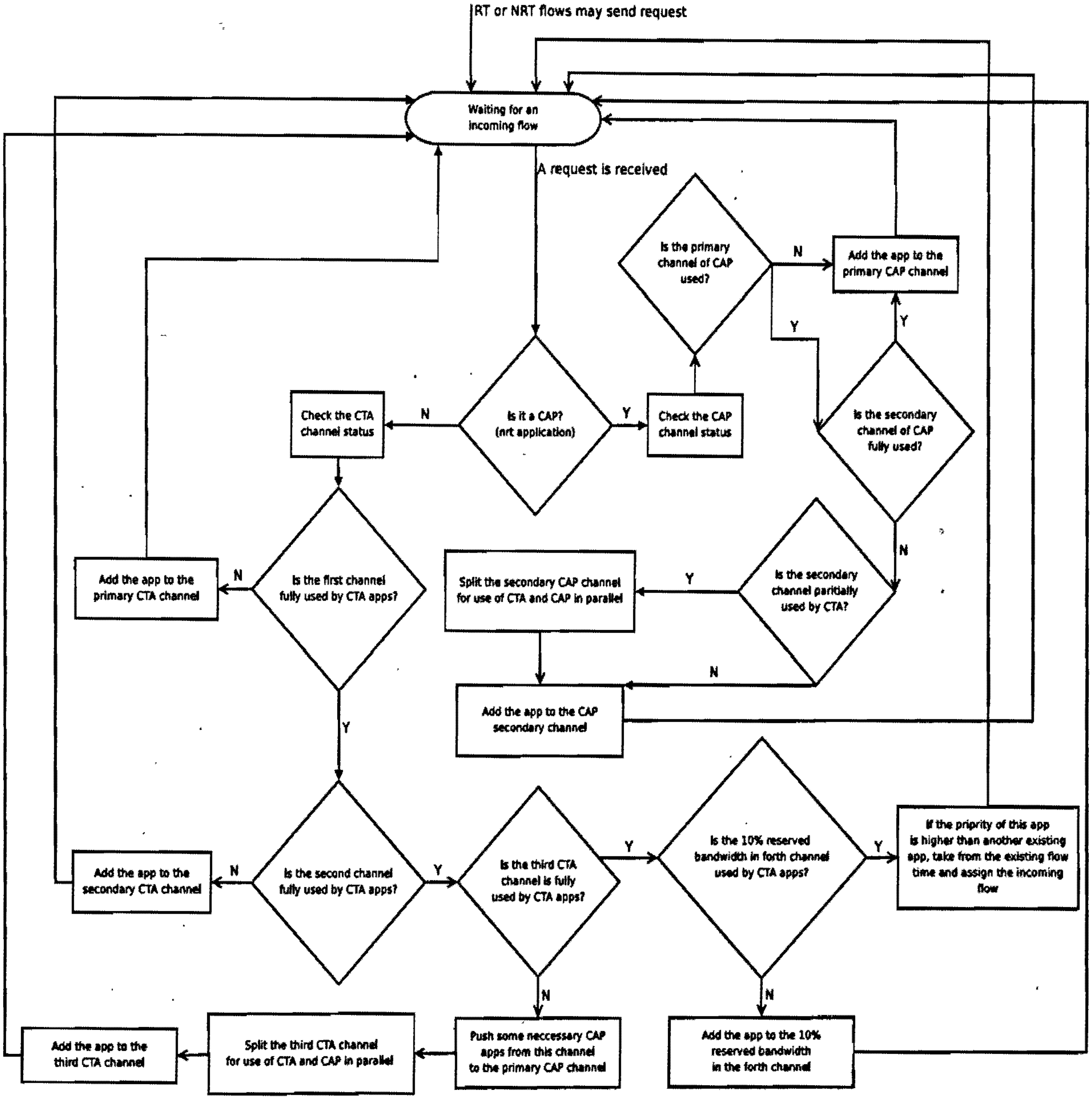

Figure 4.9: Graphical representation of ACSA using an adaptive superframe structure.

Chapter 5

Numerical Results and Discussion

In this chapter we analyze the performance metrics such as throughput, delay and fairness of the proposed adaptive channel-superframe allocation algorithm (ACSA) by simulation over a shared medium. Parameters are chosen for the ACSA system that are typical for MMW systems. MMW systems are becoming a new field of research, yet there is no software package specifically designed for simulations of MMW systems. Our investigation on comparing different simulation tools such as OPNET and network simulator (NS2) shows that NS2 is a well structured simulation tool used in network research and is better to be used for simulation of MMW system because it benefits from C, C++, TCL and OTCL programming languages, which provide the possibility to develop a wide range of networking modules. We used the package published in [24] intended for simulation of IEEE 802.15.3 and is partially tested by some developers. However, we changed the physical layer appropriately in order to obtain the required capacity in 60 GHz system. We also modified the MAC layer to suit for the proposed superframe and ACSA algorithm.

System Parameters and Tools

We explain the variable parameters of MAC layer used in our simulation. Then we present the simulation results and compare them in terms of allocation metrics. The variable simulation parameters in our study are as follows:

1. Number of flows
2. Data rate
3. CAP and CTAP length and ratio
4. Packet size

In network simulations, particularly when higher rate of data transmission is simulated, longer times are required for processing in the simulator. Thus, in order to speed up the simulation, we did not use a 3120Mbps available bandwidth of the MAC layer. Instead, we needed to find a smaller value for the maximum capacity of the MAC layer. We used the "trial and error" method to find a range of bandwidth for our simulations, which could be possible to scale down to four channels as well as scale up to higher capacities. We realized that if we choose the capacity of 800Mbps for the system, we could fully utilize the entire channel with 20 flows (10 NRT and 10 RT), each with 10 Mbps data rate. The error pattern applied to the system would waste considerable amount of bandwidth. The benefit of utilization of channels is that we can observe the variation in throughput and delay more accurately when the number and data rate of flows change. A simulation of a channel, which is not utilized would not provide adequate results to observe the effects of MAC layer algorithms. Hence, we set the values of mentioned parameters as shown in Table 5.1. Later, we modify them to study possible changes in results.

Table 5.1: Variable Parameters in 60 GHz system MAC simulation.

MAC Bandwidth	800Mbps
Error Rate	0% PER
Payload Length	2Kilo Bytes
Data Rate	10Mbps
# of NRT flows	10
# of RT flows	10
Warm Up Time	10mS
Steady State Time	10S
Superframe Size	5msec

Network Topology

The network topologies used to this study are simple topology as shown in Fig. 5.1 and NS2 auto-configured random topologies as shown in Fig. 5.2. Our simulation results show that different topologies do not affect the results since we are not studying directional transmission or mobile nodes.

5.1 Throughput Measurements

As discussed earlier, throughput may stand for different definitions. We compare IEEE 802.15.3 MAC and ACSA MAC in terms of capacity and maximum sustained throughput which is the throughput analysis based on simulation results.

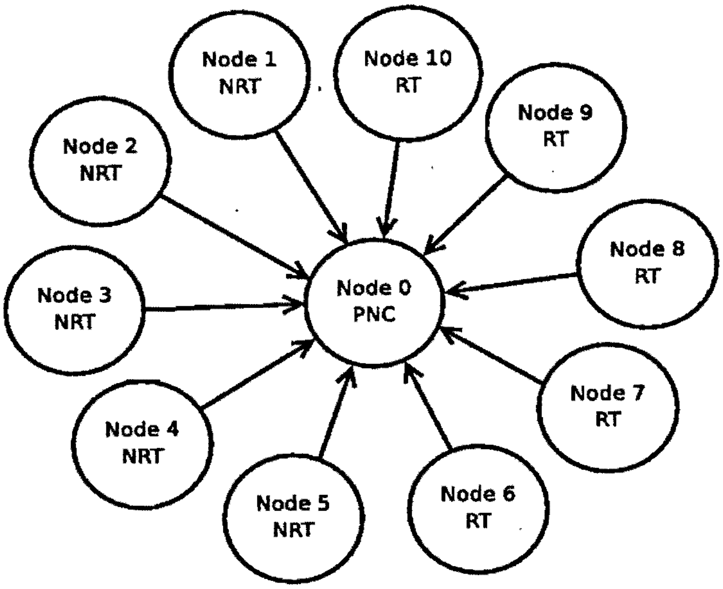

Figure 5.1: MMW simple network topology.

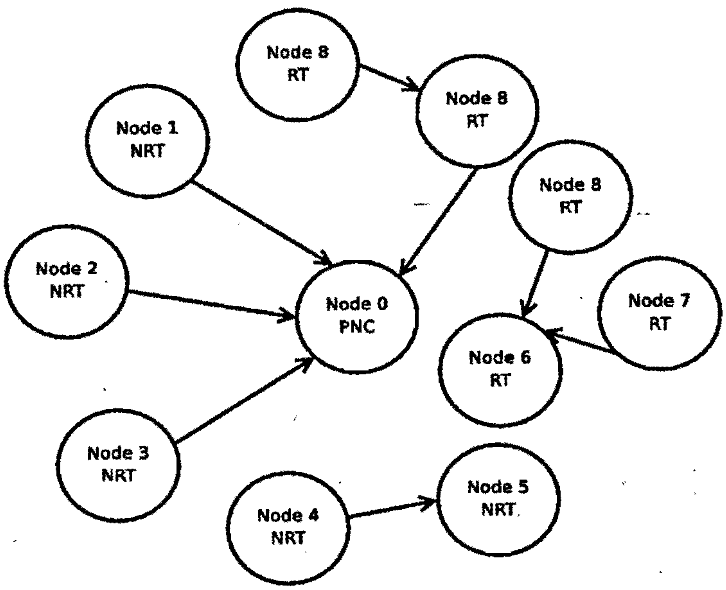

Figure 5.2: An example of MMW random topology.

5.1.1 Capacity Analysis

In order to make both standard and ACSA MAC similar in terms of available capacity and to illustrate the advantage of ACSA MAC, we would modify the capacity assignment of standard MAC in a way comparable to ACSA MAC. Therefore, we set the same ratio of CAP and CTAP in standard MAC as defined in ACSA MAC. Although the capacity of CAP and CTAP are fixed for standard MAC, ACSA would modify the superframe structure to keep a fair distribution among CAP and CTAP. We set 1250μS for CAP and 3750μS for CTAP. This would provide 25% of the entire capacity for non-real-time flows and 75% of the entire capacity for real-time flows respectively.

We cross reference the ratio of CAP and CTAP with our simulations parameters provided in Table 5.1 and obtain the following results:

$$CAP\ Bandwidth = 0.25 \times 800\,\text{Mbps} = 200\text{Mbps} \tag{5.1}$$

$$CTA\ Bandwidth = 0.75 \times 800\,\text{Mbps} = 600\text{Mbps} \tag{5.2}$$

In summary, we modified our simulation parameters in order to match the capacity analysis.

5.1.2 Sustained Throughput Analysis

As discussed previously, the values for simulations parameters are explained in Table 5.2. The values above are modified for both standard MAC and ACSA MAC in our throughput

Table 5.2: Modified Variable Parameters in 60 GHz System MAC Simulation.

MAC Bandwidth	800Mbps
CAP Bandwidth	200Mbps
CTAP Bandwidth	600Mbps
Error Rate	0% PER
Payload Length	2KiloBytes
Data Rate	10Mbps
# of NRT flows	10
# of RT flows	10
Simulation Duration	10mS

analysis. We compared the throughput of 10 NRT and RT flows in standard MAC and ACSA MAC, and the results are shown in Fig. 5.3. It can be seen that the throughput of a NRT flow is increasing from 1.8 Mbps (standard MAC) to 7 Mbps (ACSA MAC). This is not because of giving more bandwidth to NRT flows in ACSA MAC, but it is caused by the new channel-superframe assignment. The significant improvement in NRT throughput is

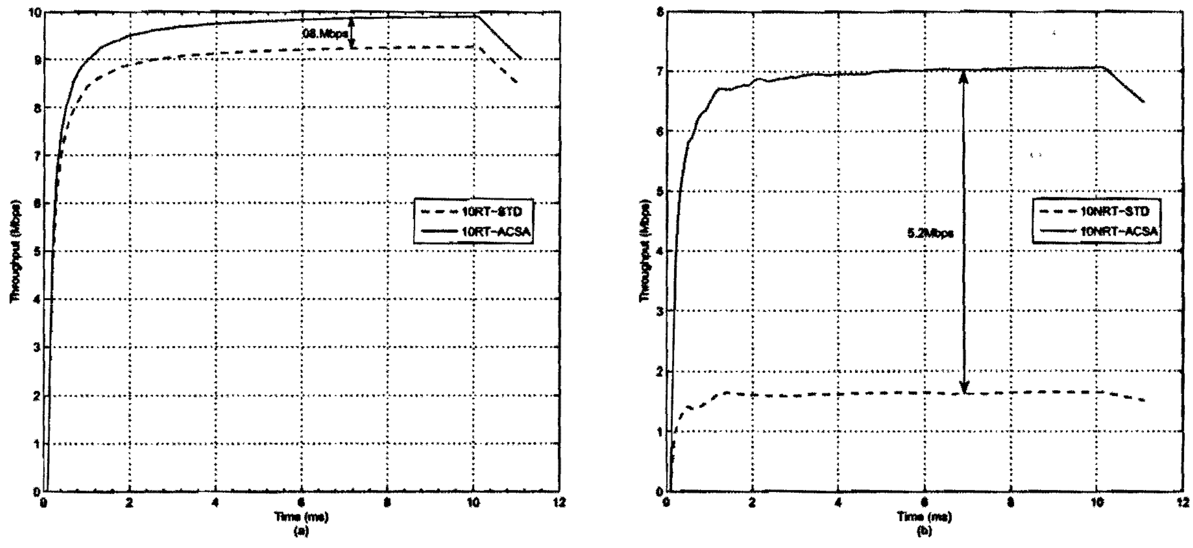

Figure 5.3: Throughput comparison of (a) 10 RT and (b) 10 NRT flows (10 Mbps each).

because of the longer unsliced CAP. Consequently, the total time spent on data transmission over the entire CAP would be higher than that of standard MAC.

There is also another small improvement in throughput of RT flows, which is from 9 Mbps (standard MAC) to 10 Mbps (ACSA MAC). This improvement in throughput of RT flows in our proposed MAC is provided by two features in the modified superframe design.

1. First is the priority we have given to RT flows. Any RT flow which is trying to get access to channel could be granted to use the required bandwidth. There is only one case that a RT flow may be rejected to access the channel, which is in the case that considerable amount of bandwidth is used by RT flows. This is not a common case in MMW networks since we have made channel#1, #2, #3 and 10% of channel#4 (78% of the entire bandwidth) available for RT flows to occupy.

2. Second is the unsliced superframe for channel#1 and channel#2 , which is keeping RT flows away from starving state for channel.

The average JFR (Job Failure Rate) values for this simulation are shown in Table 5.3. JFR is the rate at which video frames are dropped due to missing their deadlines. Thus, the higher JFR means the higher channel access delay. Accordingly, the objective of our superframe design is to reduce JFR as much as possible, namely to increase Job Success Rate as much as possible.

An important question is what if we use a dynamic ratio between CAP and CTAP in standard MAC. In this case ACSA would be still outperform standard MAC? In order to find

Table 5.3: Job failure rate (10NRT and 10RT, 10Mbps each)

-	Standard	ACSA
NRT	83.7%	29.7%
RT	8.2%	1.9%

the answer and clarify the effectiveness of ACSA algorithm, we changed the ratio between CAP and CTAP dynamically while the length of superframe is 5msec. We measured the performance of standard MAC as shown in Fig. 5.4. It can be seen that NRT throughput would increase by increasing the length of CAP. However, the diminish in RT throughput is significant. We set a maximum value for the length of CAP wherein the throughput of RT flows would not become less than the throughput of NRT flows, otherwise we do not provide an acceptable QoS for RT flows. We can see that the throughput of NRT in the standard MAC could not reach the throughput of NRT in ACSA MAC. We also observe a significant reduction in RT throughput while trying to improve the NRT throughput in standard MAC. In conclusion, the ACSA MAC throughput measurements as shown in Fig. 5.3 outperform the standard MAC throughput measurement with dynamic ratio between CAP and CTAP.

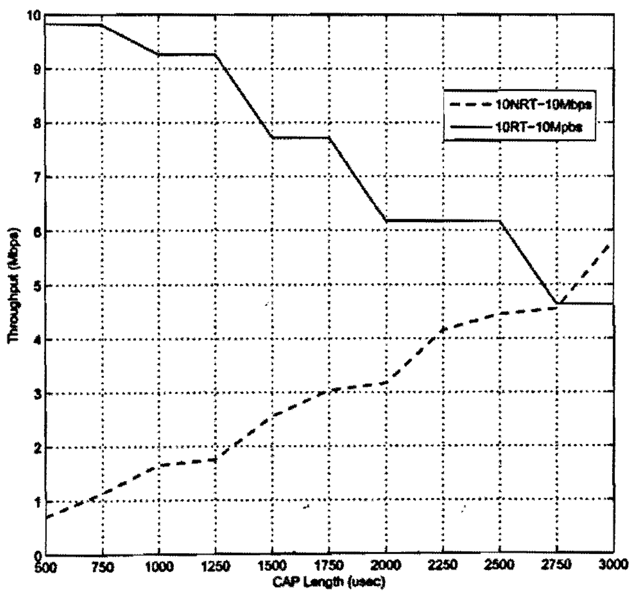

Figure 5.4: Throughput measurements of standard MAC with dynamic ratio between CAP and CTAP length.

In order to study the robustness of the ACSA MAC, we increased the packet error rate and compared the results as shown in Fig. 5.5. The simulation results with higher packet error rates show that there is still significant improvements in throughput of NRT flows using ACSA MAC. However, the standard MAC and ACSA MAC, which are simulated with higher PER experience diminish in throughput of both RT and NRT flows. As it can be seen in Fig. 5.5(b), throughput of RT flows with 10% PER using ACSA MAC is less than the throughput of RT flows with 1% PER using standard MAC. Therefore, ACSA MAC with higher packet error rates can not outperform standard MAC, which means that the higher PER could reduce the performance of the system significantly either using standard or ACSA MAC.

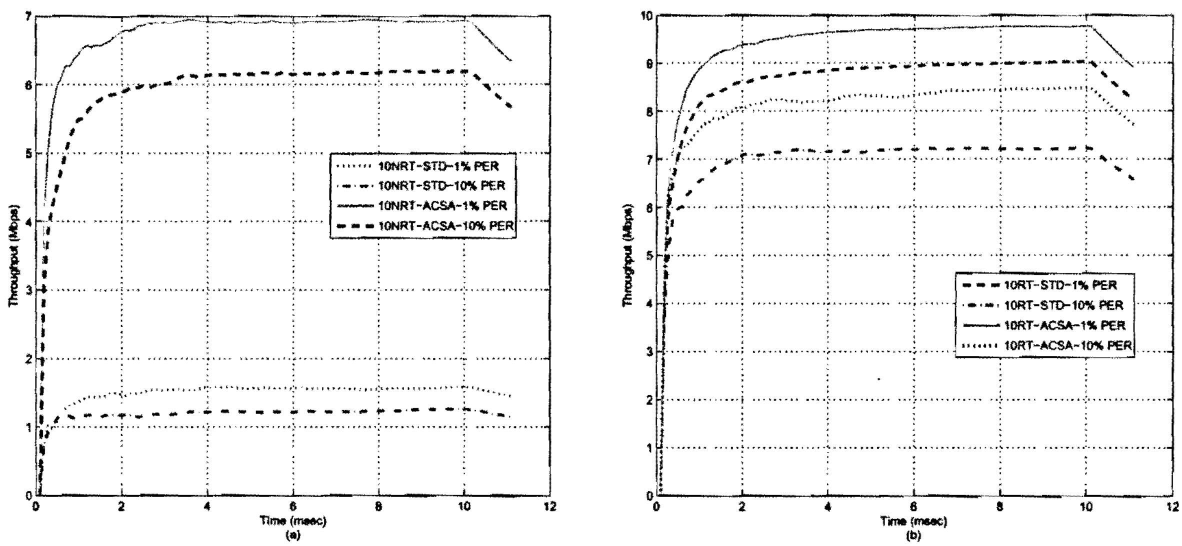

Figure 5.5: Throughput comparison of (a) NRT and (b) RT flows with different packer error rates.

We increased the number of flows and observed the changes in results as shown in Fig. 5.6. This pair of results are interestingly illustrating the trade off between RT flows and NRT flows. There are four important observations in Fig. 5.6:

1. NRT throughput in both standard and ACSA MAC is reduced when number of flows is increased.

2. The average NRT improvement is reduced from 5 Mbps to 2.5 Mbps.

3. RT throughput in both standard and ACSA MAC is reduced when number of flows is increased.

4. The average RT throughput of ACSA MAC is not improved and is 0.5 Mbps less than the average RT throughput of standard MAC.

The reasons for the above observations are:

- The decrease in throughput value for both RT and NRT flows in both MAC protocols, is the effect of higher number of flows, which is twice of the previous case. Therefore, the average available bandwidth for each flow is about half of the previous case. The values in graphs are illustrating this change.

- The reason that we observe improvement in NRT throughput and diminish in RT throughput is that we are compensating available bandwidth of RT flows for NRT flows transmission. Therefore, while there are higher number of flows, NRT throughput is still improving at the cost of diminished RT throughput. This is tolerable since the improvement in NRT throughput is 2.5 Mbps which overcomes the 0.5 Mbps decrease in RT throughput.

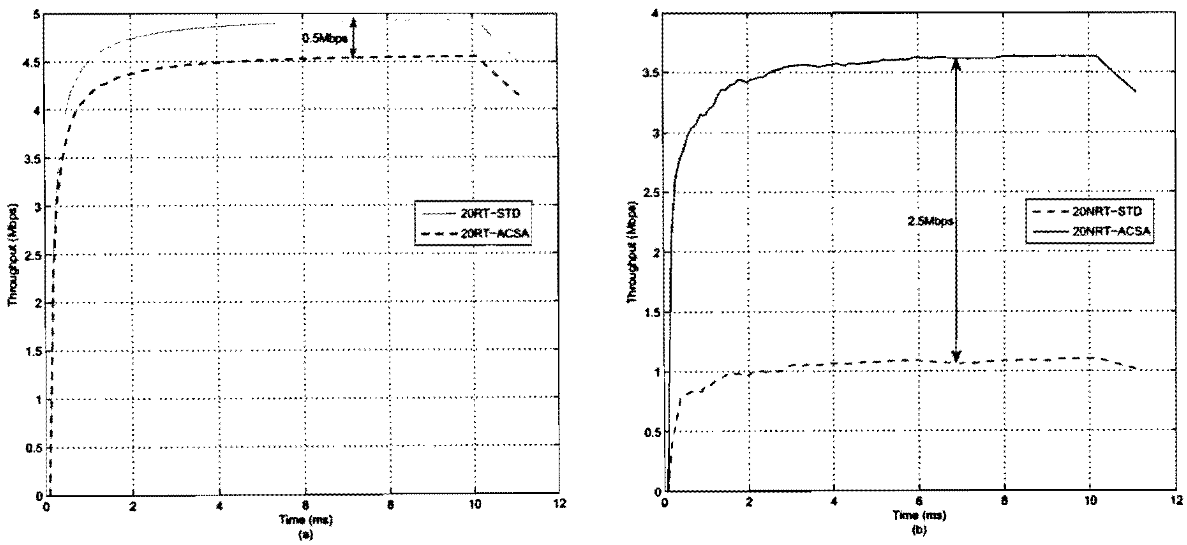

Figure 5.6: Throughput comparison of (a) 20 RT and (b) 20 NRT flows (10 Mbps each).

The JFR for this simulation is shown in Table 5.4. We see that the JFR for all the flows are increased specially for RT flows. This is because of the high number of flows in the system. Therefore, we do not increase the number of flows more than 40, which is also not an acceptable number for MMW network in reality.

Next, we increased the data rate of flows from 10 Mbps to 20 Mbps in our simulation. The results are shown in Fig. 5.7, which are comparing the performance of standard MAC

Table 5.4: Job Failure Rate (20NRT and 20RT, 10Mbps each)

-	Standard	ACSA
NRT	89%	63.8%
RT	51.1%	54.8%

with ACSA MAC. We can observe the same improvement in throughput of NRT flows as we could observe in simulation of NRT flows with 10 Mbps data rate which is shown in Fig. 5.3(b). On the other hand, we can observe the more improvement in throughput of RT flows comparing to the case of RT flows simulation with 10 Mbps data rate which is shown in Fig. 5.3(a). Therefore, we only have small improvement in throughput of RT flows while increasing the data rate of flows. In order to better observe this improvement and discuss about it we compared the simulation results of different data rates (10 Mbps and 20 Mbps) using ACSA MAC in Fig. 5.8.

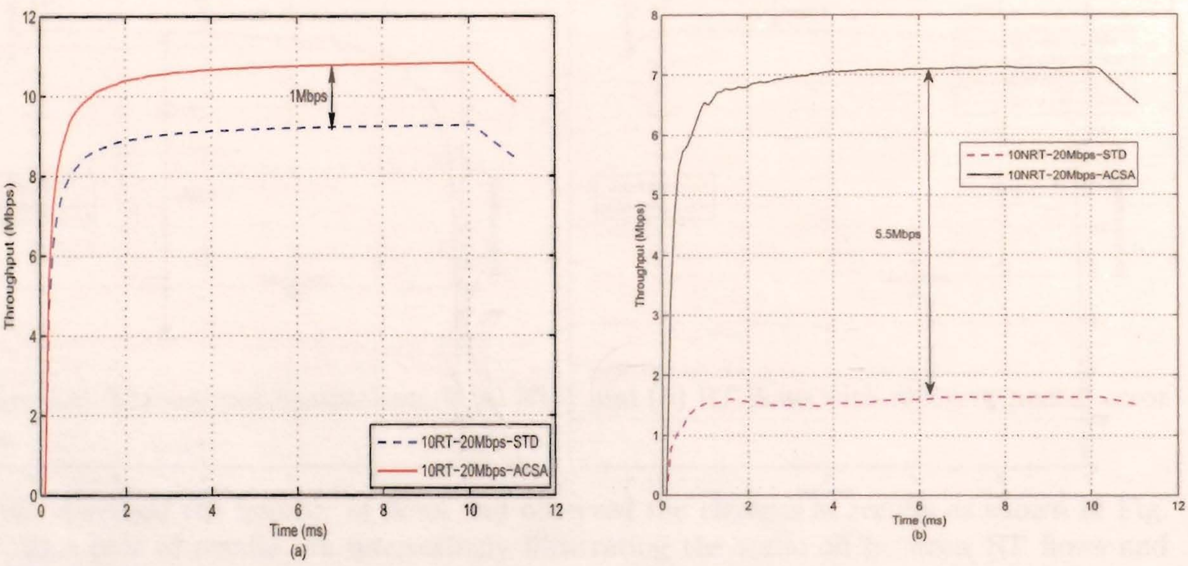

Figure 5.7: Throughput comparison of (a) 10 RT flows and (b) 10 NRT flows (20 Mbps each).

It can be observed that there is no change between simulation of NRT flows with 10 Mbps and 20 Mbps, but small increase in throughput of RT flows can be seen. The reason is that the channel for NT flows is fully utilized and transmission by a higher data rate would still keep utilizing the channel and the throughput results remain the same. Therefore, if the link is utilized then the only consequence of sending packets by a higher data rate is having longer queues in network nodes and more failure in sending and receiving acknowledgments. Consequently, the JFR values increase as shown in Table 5.5 comparing to JFR values shown in Table 5.3. The higher JFR values show the effect of longer queues and higher rate of transmission. The longer queue shows the more waiting time for transmission. Thus, more packets will miss the transmission deadline (lifetime) and will be dropped.

Table 5.5: Job Failure Rate (10NRT and 10RT, 20Mbps each)

-	Standard	ACSA
NRT	91.9%	64.8%
RT	54.1%	46.3%

There is also a huge gap in JFR values specially for RT flows. The reason is that the deadline for RT packets to be received is much less than the deadline of NRT packets. In fact, real-time flows would be time sensitive and they are usually high quality multimedia. Therefore, we set the deadline for RT flows equal to the length of the superframe which means that if a flow looses the chance of sending a queued packet in the current active superframe, it will drop the packet. On the other hand, the deadline for non-real time packets is 10 times of the superframe length.

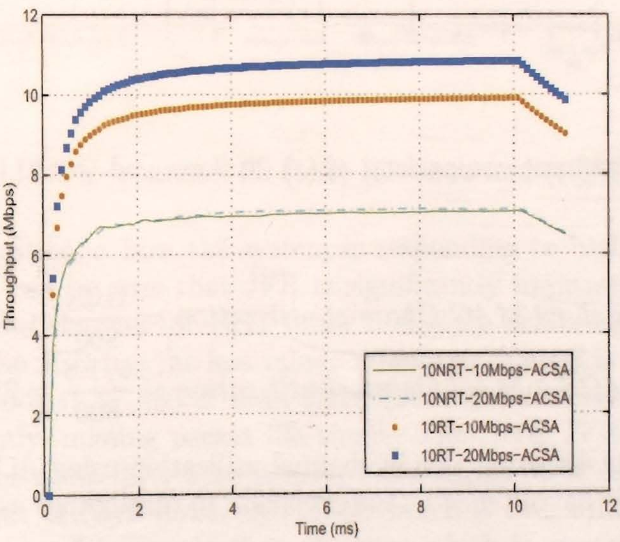

Figure 5.8: Throughput comparison of RT and NRT flows for 10 and 20 Mbps using ACSA MAC.

5.2 Channel Utilization

The channel utilization is defined in section 3.1. We compute the percentage of bandwidth used efficiently in both of the standard and ACSA MAC.

We need to measure the aggregate throughput of all the flows in the system in order to obtain bandwidth utilization efficiency. We simulated the system containing 20 flows each with 10 Mbps data rate as shown in Fig. 5.9(a) and 40 flows each with 10 Mbps data rate as shown in Fig. 5.9(b). In (5.3) we provide the computations for channel utilization.

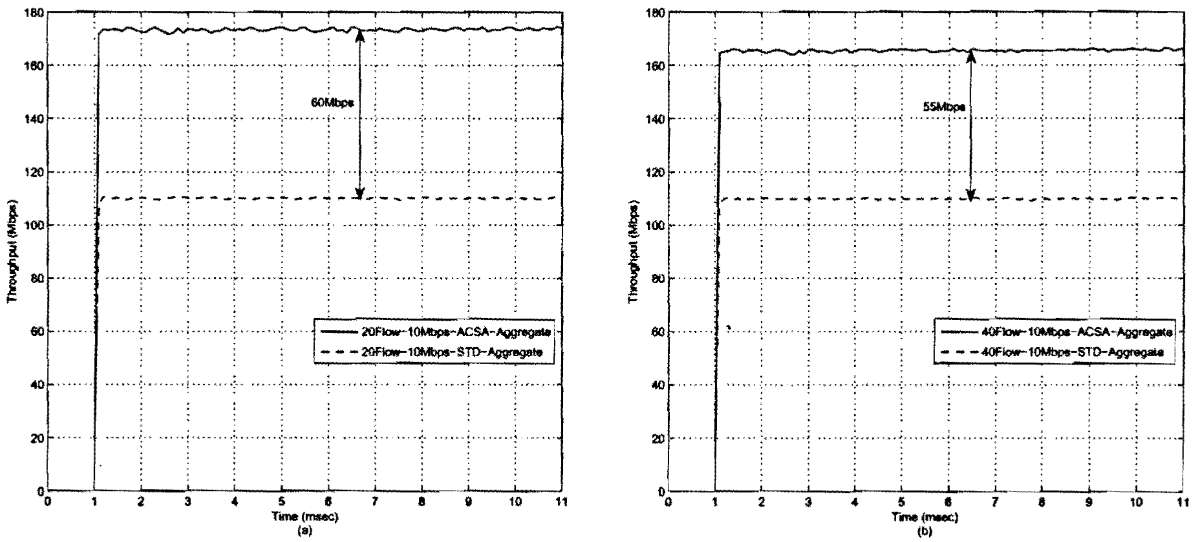

Figure 5.9: Total throughput comparison of (a) 20 flows and (b) 40 flows (10 Mbps each).

$$Standard\ MAC\ Channel\ utilization = \frac{110.4}{800} = 13\% \tag{5.3}$$

$$ACSA\ MAC\ Channel\ utilization = \frac{171.8}{800} = 21\% \tag{5.4}$$

We can see a significant improvement in channel utilization using ACSA MAC. This result could be expected while having such an improvement in throughput as explained in previous section.

$$Standard\ MAC\ Channel\ utilization = \frac{110.32}{800} = 13\% \tag{5.5}$$

$$ACSA\ MAC\ Channel\ utilization = \frac{166.2}{800} = 20\% \tag{5.6}$$

The throughput results discussed earlier for 40 flows were significantly different to 20 flows. Yet, there is not much difference in channel utilization since the aggregate throughput of the entire system is the same for both cases. As we discussed earlier, we are compensating some available bandwidth of RT flows for NRT flows transmission and may observe different results in throughput in terms of RT and NRT types. However, the total received bandwidth of entire system is almost equal to the previous case. In summary, an increase in number of users would not logically and practically affect the way that users are utilizing the channel. In order to show the trend of increase in job failure rate, we provided a graph as shown

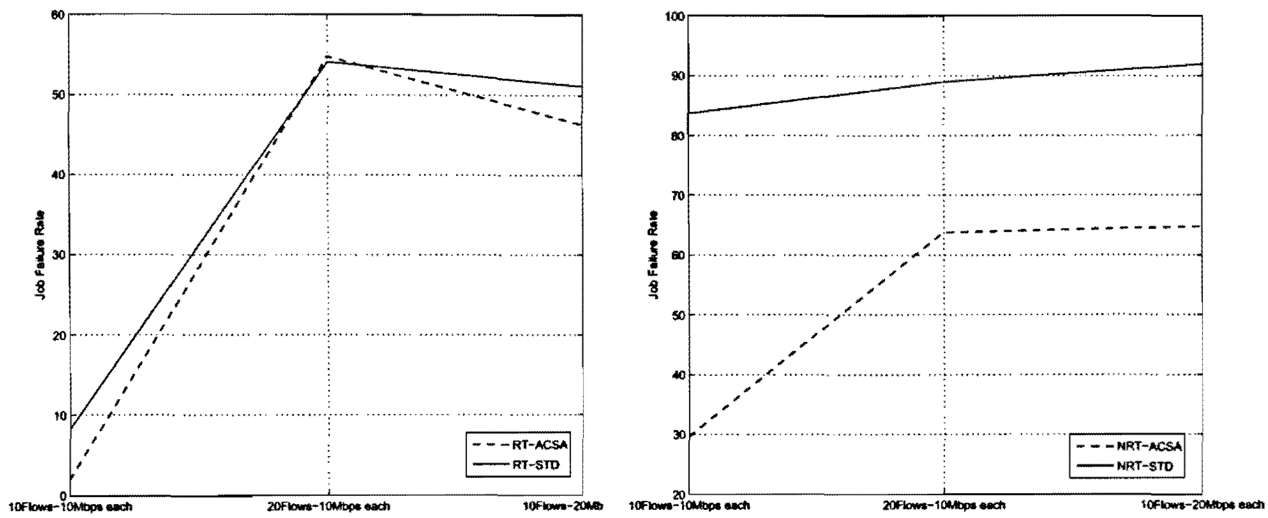

Figure 5.10: JFR comparison for (left) RT flows and (right) NRT flows.

in Fig. 5.10. The question is how the system is responding to higher number of flows or higher data rates? It can be seen that JFR is significantly improved for NRT flows using ACSA MAC. When the channel for NRT flows is slightly utilized in the system (10NRT flows, 10Mbps each) the JFR has the less value. When the channel is over-utilized by higher number of flows (20 flows) or higher data rates (20Mbps) the system experience longer queues and consequently, missing packet life times. Therefore, JFR significantly increases when the system is over-utilized for both standard and ACSA MAC. JFR results of RT flows is very different to that of NRT flows. A system which is over-utilized by higher number of flows has higher packet failure rate than the time which is over-utilized by higher rates of transmission. The reason is that a system with 20 RT flows would assign a shorter time to each flow because the CTAP should be separated into 20 CTA slots. Therefore, each flow has less time to transmit and because the channel is completely utilized, the chance of transmission would decrease. Consequently longer queues are possible and packet life times could be reached. In summary, JFR values have different trends for RT and NRT values. We see a significant improvement in NRT flows while the JFR for RT flows are slightly improved.

5.3 Delay Measurements

5.3.1 Simulation-based Study

Based on our discussion in section 4.1.2, there are two types of delay which we aimed to reduce in our MAC protocol design. First is the registration delay and the second is imposed MAC protocol delay on TCP connections. We simulated the MMW system and measured both delays on MAC protocols. The results are shown in Table 5.6 and Table 5.7 respectively. We modified the simulation parameters to be the same as the values in Table 5.2 except for the length of the superframe. We kept the ratio between CAP and CTAP periods same as before (CAP = 25% and CTAP = 75%), yet the length of superframe takes different values from 5000μS up to 30000μS.

Table 5.6: Registration Delay Measurement

Delay	Standard	ACSA
NRT Registration	9440μS	646μS
RT Registration	9440μS	646μS

The results of registration delay shows that our ACSA MAC is reducing the registration delay significantly and the users would feel this result while browsing websites. Later, we discuss how effective this improvement is in service quality. The results of protocol delay

Table 5.7: Imposed MAC Protocol Delay Measurement

NRT Protocol Delay (TCP Delay) for Standard MAC	Superframe Length
61450μS	5000μS
69340μS	10000μS
85250μS	20000μS
114130μS	40000μS

illustrate the effect of having a channel-superframe dedicated to NRT flows (TCP flows). We had the same number of TCP flows in our simulations with different lengths of superframe. As we increase the length of superframe, the TCP delay is increasing and the reason for this diminish in quality of service is the starving state of TCP packets while going back and forth. As discussed previously in section 4.1.2, by theoretical analysis, the RTT (Round

Trip Time) would increase by twice of the length of CTA period while using standard MAC. We think that this amount of delay would significantly reduce the service quality and the simulation results are showing it evidently. The effect of this delay on throughput is really important to be investigated. We simulated the system for different sizes of superframe (5000μS, 10000μS and 20000μS and 40000μS) and provided the results for NRT throughput in Fig. 5.11. The line with square marks in Fig. 5.11 shows that there is no change in NRT

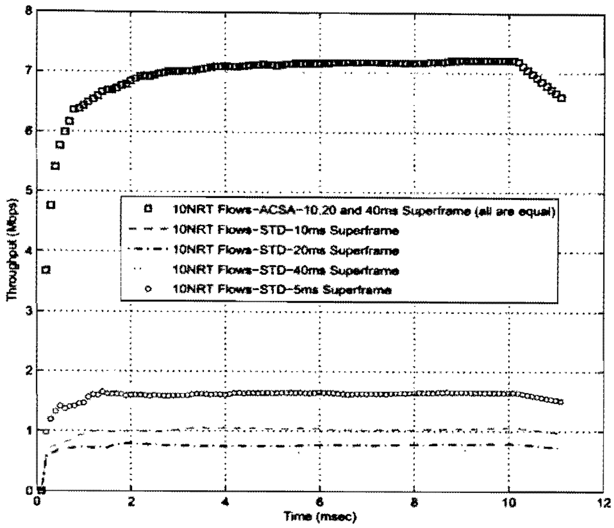

Figure 5.11: NRT Throughput Comparison for 10msec, 20msec and 40msec Superframe Sizes

throughput of ACSA MAC for different superframe sizes. The reason is that there is not any imposed delay from superframe design in ACSA MAC. There is no sliced CAP superframe for NRT flows and an increase in the length of superframe will not affect TCP RTT delay. On the other hand, increasing superframe length from 5000μS to 40000μS (the line with circle marks) is diminishing the NRT throughput. This is because of the increasing RTT of TCP connections. The longer CTAP length would increase the waiting time of TCP packets. Each TCP packet sent from the source needs to be acknowledged. Then, if a packet is sent from the source and the transmission is suspended for a CTAP length prior to receive the ACK, then it would increase the waiting time significantly. As discussed earlier, a longer waiting time is equal to longer queues and more packets missing their transmission deadline. Therefore, the NRT throughput is diminished while increasing the length of superframe. The results of RT throughput for different superframe sizes are shown in Fig. 5.12. The graph shows that ACSA has about 4.8Mbps, 2.6Mbps and 1.7Mbps higher throughput results for RT flows when superframe sizes are 10msec, 20msec and 40msex respectively. However, both MAC protocols experience a significant diminish in RT throughput while increasing the superframe size. The reason is the long delays between CTA slots of a flow. It means

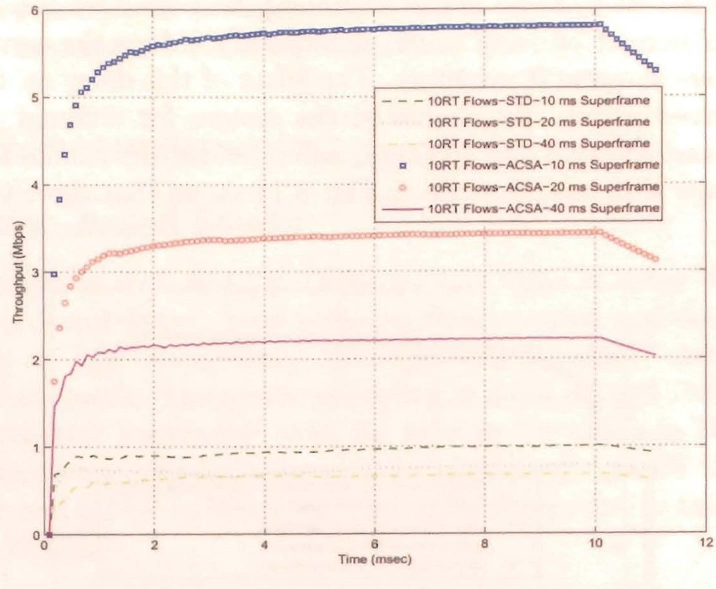

Figure 5.12: RT Throughput Comparison among 10msec, 20msec and 40msec Superframe Sizes.

that a flow should wait a longer time for the next CTA slot while having a longer superframe size. We compute the waiting times for a RT flow in (5.7) and (5.8).
RT waiting time for Superframe Size $= 10000\mu$S

$$= 10000\mu\text{S} - \frac{Total\ CTA\ Length}{Number\ of\ CTA\ Flows} = 10000\mu\text{S} - 800\mu\text{S} = 9200\mu\text{S} \tag{5.7}$$

RT waiting time for Superframe Size $= 40000\mu$S

$$= 40000\mu\text{S} - \frac{Total\ CTA\ Length}{Number\ of\ CTA\ Flows} = 40000\mu\text{S} - 3200\mu\text{S} = 36800\mu\text{S} \tag{5.8}$$

It can be seen that a very long delay is possible when using a longer superframe. The available bandwidth in a channel for RT flows is important because RT flows are bandwidth intense. However, they are also time sensitive. Therefore, giving a wider range of bandwidth to RT flows alone will not keep up the service quality, but a more frequent available CTA slot could give a better chance to a RT flow to send the data and consequently increase the quality of service.

5.3.2 Experiment-based Study

In order to understand how TCP delay would affect service quality in practical cases, we measured the page loading time of different websites using a website load-time test [25]. We measured the TCP connections delay while browsing websites. Fig. 5.13 and Fig. 5.14 show the loading time of Google web page. We did not use the 802.15.3 MAC or ACSA MAC in this experimental test. This test could be an evidence supporting our claim that TCP RTT is an important parameter in quality of service of NRT flows and it should be taken into account for future wireless systems such as MMW systems. In our design, we aimed to keep RTT in a less value as is in 802.15.3 MAC.

Figure 5.13: Loading time and number of TCP connections while loading Google.

Each line in Fig. 5.13 shows a TCP connection downloading a section of the website. The yellow part is the time for establishing the connection. The green part is the delay before receiving the first byte and the blue part is the time during which data is being received. The result is:

- Two sequential TCP connections
- Approximately 100msec TCP delay for loading page

The same test is done for browsing Youtube web page as a commonly used website, which is shown in Fig. 5.15 Fig. 5.16. The result is:

- Two sequential levels for TCP connections
- Approximately 600msec TCP delay for loading page

We intend to observe and analyze the average delay of TCP connections while loading different web pages. We have done the above tests for some other popular websites such as Microsoft, Yahoo and Facebook and measured the average delay of a TCP connection and it is equal to 695msec. Two important points in our observations are as follows:

- Many TCP connections are needed for loading a web page.
- At least 50% of the loading time is for TCP connections delay.

Considering the fact that Internet users are browsing many web pages in a short period of time (establishing many TCP connections), and also considering that TCP connection delay is at least 50% of loading time, we realize that the TCP delay is a bottleneck in loading web pages. Therefore, our superframe design, which keeps the TCP RTT at the

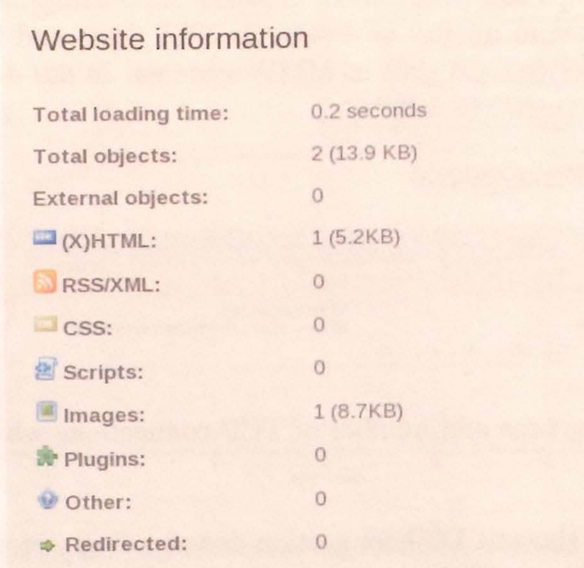

Figure 5.14: Information about loaded object from Google.

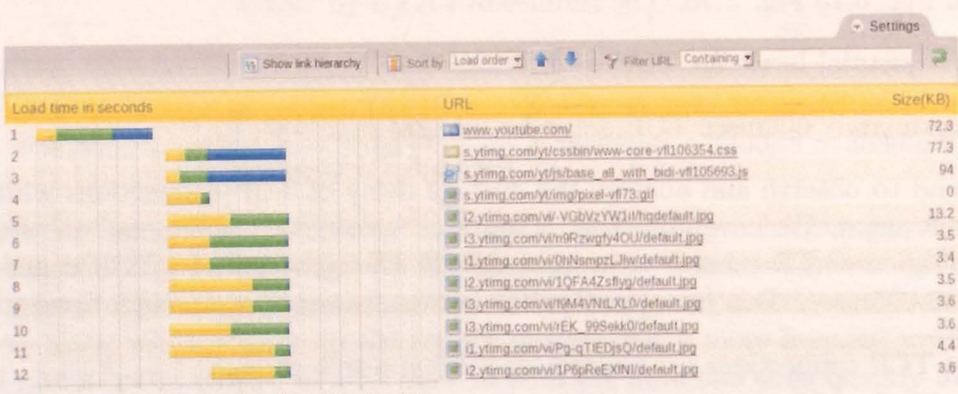

Figure 5.15: Loading time and number of TCP connections while loading Youtube.

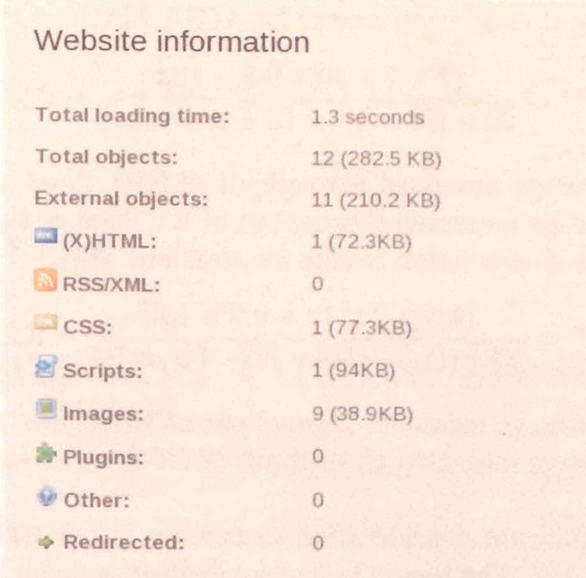

Figure 5.16: Information about loaded objects from Youtube.

minimum value and eliminates the superframe imposed delay would help to overcome this bottleneck in MMW systems. Knowing that TCP delay is a considerable share of time while loading a web page, we should prevent this TCP delay to become more. **We changed the sliced superframe to an adaptive unsliced superframe in order to prevent the TCP delay to increase.** The improvements in our simulation results are confirming this experiment as the TCP delay is improved in ACSA MAC.

5.4 Fairness Measurements

We defined a fairness index in section 4.1.3. In addition we defined a weight equal to two for NRT flows throughput in our inter-class fairness measurement to be able to deal with NRT and RT flows equally. The theoretical and practical inter-class fairness measurements of standard MAC and ACSA MAC considering the system parameters in Table 5.2 are as follows:

Inter-class fairness index of capacity allocation for standard and ACSA MAC:

$$f_{ACSA}^{analysis} = f_{STD}^{analysis} = \frac{(20 \times 2 \times 10 + 60 \times 10)^2}{20 \times ((20 \times 2)^2 \times 10 + 60^2 \times 10)} = 0.96 = 96\% \tag{5.9}$$

where 20 Mbps is the available bandwidth for a NRT flow and 60 Mbps is the available bandwidth for a RT flow.

Inter-class fairness index of simulation results for ACSA MAC:

$$f_{ACSA}^{Simulation} = \frac{(7 \times 2 \times 10 + 9.8 \times 10)^2}{20 \times ((7 \times 2)^2 \times 10 + 9.8^2 \times 10)} = 0.97 = 97\% \tag{5.10}$$

where 7 Mbps is the average measured throughput of NRT flows as shown in Fig. 5.3(b) and 9.8 Mbps is the average measured throughput of RT flows as shown in Fig. 5.3(a).
Inter-class fairness index of simulation results for standard MAC:

$$f_{STD}^{Simulation} = \frac{(1.6 \times 2 \times 10 + 9.2 \times 10)^2}{20 \times ((1.6 \times 2)^2 \times 10 + 9.2^2 \times 10)} = 0.81 = 81\% \tag{5.11}$$

where 1.6 Mbps is the average measured throughput of NRT flows as shown in Fig. 5.3(b) and 9.2 Mbps is the average measured throughput of RT flows as shown in Fig. 5.3(a).

As it can be seen, a significant consideration to fairness in our ACSA MAC is illustrated comparing (5.10) and (5.11). The reason is that we dedicate enough bandwidth for non-real-time flows to be transmitted and we provided an algorithm to keep the priority of real-time flows higher than the priority of non-real-time flows. Therefore, we gained a fairness value of 97%, which is very close to one and is significantly improved. ACSA fairness is even better than theoretical fairness value and the reason is that real-time flows could not use as much bandwidth as we assumed to be used in theoretical case. Therefore, the throughput results of RT and NRT flows are even closer than the values in theoretical analysis. The difference between average RT throughput and NRT throughput in analysis was 40 Mbps as shown in (5.9) while this difference in our simulation results is reduced to 2 Mbps as shown in (5.10). If the bandwidth usage of RT flows is closer to bandwidth usage of NRT flows, the fairness of the system is closer to one. Therefore, we obtained a better fairness index based on the measured values in our simulation.

In summary, we simulated a MMW system with certain simulation parameters using 802.15.3 MAC as well as ACSA MAC. We measured three performance metrics (throughput, delay and fairness) which we aimed to improve in our superframe design. These allocations metrics are proper illustrations of quality of service in a network. The simulation results show significant improvement in throughput of NRT flows. The simulation results show that the adaptive superframe structure could provide throughput improvements not only for NRT flows, but also for RT flows. The control algorithm in PNC could manage the bandwidth allocations in superframe and improve the throughput of RT flows. Basically, we gained higher average throughput using ACSA MAC. The channel access delay is also improved by providing an unsliced superframe which eliminated an imposed delay on TCP connections. Finally the better distribution of bandwidth in ACSA MAC could improve the entire fairness of the system. As a brief, the simulation results could support the analysis of the proposed adaptive channel-superframe allocation algorithm, which could generally improve the quality of service for MMW systems.

Chapter 6

Conclusions and Future Research

6.1 Conclusions

In this thesis, we studied the performance of a MMW system MAC layer in terms of throughput, delay and fairness. An adaptive bandwidth allocation algorithm (ACSA) is utilized in the PNC and a new superframe structure is designed in order to distribute bandwidth among real-time and non-real-time flows. A fair distribution of bandwidth in ACAS could improve the fairness index of a MMW system significantly. We compensate some available bandwidth of RT flows in order to improve the throughput of NRT flows. Therefore, the throughput of RT flows could be improved only for a number of RT flows less than about 20. However, NRT flows take advantage of an unsliced superframe and consequently encounter less channel access delay and less TCP round-trip time.

The complexity of ACSA MAC is more than IEEE 803.15.3 MAC and specifically an adaptive bandwidth allocation algorithm is more difficult to be implemented in PNC compared to implementation of 802.15.3 functions in PNC. For instance, the number of superframes, which are broadcast in 802.15.3 MAC is 64 superframes in 64 directions while it is increased to 256 superframes in ACSA MAC (4 superframes per directions).

6.2 Future Research

In future, ACSA MAC could be studied for a MMW system in an ad hoc topology taking the mobility into the account. Therefore, the directional transmission could be studied as an important feature of MMW system along with ACSA MAC. Moreover, acknowledgment techniques and power saving methods are still potential research areas which could be studied over ACSA MAC.

Bibliography

[1] E. Kivisaari, "60 GHz MMW Applications," *Helsinki University of Technology, Telecommunications Software and Multimedia Laboratory*, Apr. 2003.

[2] P. Xia, Q. Xiangping, and et. al., "Short range gigabit wireless communications systems: potentials, challenges and techniques," *IEEE International Conference on Ultra-Wideband*, pp. 872–7, Sept 2007.

[3] "Wireless medium access control (MAC) and physical layer (PHY) specifications for high rate wireless personal area networks (WPANs), IEEE Standard 802.15.3," *IEEE, New York*, Sept 2003.

[4] F. Kojima, C. W. Pyo, and et. al., "Necessary Modifications on Conventional IEEE802.15.3b MAC to Achieve IEEE802.15.3c Millimeter Wave WPAN," *Personal, Indoor and Mobile Radio Communications, 2007. PIMRC 2007. IEEE 18th International Symposium on*, pp. 1–5, Sept. 2007.

[5] S. A. Mahmud, S. Khan, and et. al., "Capacity issues in meshed high data rate WPANs," *22nd International Conference on Advanced Information Networking and Applications*, pp. 1285 – 90, 2008.

[6] K. Chin and D. Lowe, "A Simulation Study of the IEEE 802.15.3 MAC," *The IEEE Conference on local Computer Networks, 2005. 30th Anniversary*, Sept. 2005.

[7] K. Gan, P. Xu, and et. al., "A novel MAC protocol for IEEE 802.15.3C," *Intelligent Signal Processing and Communication Systems, 2007. ISPACS 2007. International Symposium on*, pp. 702–705, Nov. 2007.

[8] K. Chin and D. Lowe, "A Simulation Study of TCP over the IEEE 802.15.3 MAC," *The IEEE Conference on local Computer Networks, 2005. 30th Anniversary*, pp. 476–477, Nov. 2005.

[9] F. Yildirim and H. Liu, "Directional MAC for 60 GHz Using Polarization Diversity Extension (DMAC-PDX)," *Global Telecommunications Conference, 2007. GLOBECOM '07. IEEE*, pp. 4697–4701, Nov. 2007.

[10] T. Matsumoto, S. Fujita, and et. al., "Panasonic PHY and MAC Proposal to IEEE802.15 TG3c CFP," *In response to TG3c Call for Proposals (IEEE P802.15-07-0586-02-003c)*, July 2007.

[11] P. Smulders, "Exploiting the 60 GHz Band for Local Wireless Multimedia Access: Prospects and Future Directions," *IEEE Communications Magazine*, pp. 140–147, Jan. 2002.

[12] H. Singh and H. N. et. al., "Supporting uncompressed HD video streaming without retransmissions over 60 GHz wireless networks," *IEEE Wireless Communications and Networking Conference*, pp. 1939–44, Mar 2008.

[13] H. Singh, Q. Xiangping, and et. al., "Support of uncompressed video streaming over 60 GHz wireless networks," *5th IEEE Consumer Communications and Networking Conference*, pp. 243–8, Jan 2007.

[14] J. Mikkonen, "Emerging Wireless Broadband Networks," *IEEE Communications Magazine*, pp. 112–117, Feb. 1998.

[15] I. C. 2007, "Comments of IEEE Geosciences and Remote Sensing Society," *IEEE Communications Magazine*, pp. FCC RM–11104, Nov. 2007.

[16] Y. Zhang, P. Xu, X. Chen, and S. Huang, "Analysis of IEEE 802.15.3 CAP Based on DC-OFDM-UWB," *Wireless Communications, Networking and Mobile Computing, 2006. WiCOM 2006.International Conference on*, pp. 1–4, Sept. 2006.

[17] X. Qin, P. Xia, and et. al., "Autonomous coordinator selection in Beamformed 60 GHz wireless networks," *IEEE Wireless Communications and Networking Conference*, pp. 1815–1820, March 2008.

[18] S. Khuller and Y. Sussmann, "The capacitated K-center problem," *Algorithms - ESA '96. Fourth Annual European Symposium. Proceedings*, pp. 152–66, Sept 1996.

[19] H. Kim, H. Woesner, and A. Wolisz, "A medium access control protocol for radio over fiber wireless LAN operating in the 60-GHz band," *5th European Personal Mobile Communications Conference 2003*, pp. 204–8, Apr 2003.

[20] X. Dong and P.Varaiya, "Saturation Throughput Analysis of IEEE 802.11 Wireless LANs for a Lossy Channel," *IEEE Communications Letters*, vol. 9, Feb. 2005.

[21] P. Sessini and A. Mahanti, "Observations on Round-Trip Times of TCP Connections," *Society for Computer Simulation*, vol. 38, pp. 347–353, 2006.

[22] A. Hopkins, "Optimizing page load time," Oct 2006. `http://www.die.net/musings/page_load_time`.

[23] R. Jain, D. Chiu, and W. Hawe, "A Quantitative Measure Of Fairness And Discrimination For Resource Allocation In Shared Computer Systems," *DES Research Report TR-301*, vol. 38, Sept 1984.

[24] S. Datta, I. Seskar, M. Demirhan, and et. al., "Ad-hoc extensions to the 802.15.3 MAC protocol," *Sixth IEEE International Symposium on a World of Wireless Mobile and Multimedia Networks-WoWMoM*, pp. 293–298, Jun 2005.

[25] "Full page test," Jun 2009. `http://tools.pingdom.com`.