Controlling Frame Aggregation Rate in IEEE 802.11n WLANs to Obtain Optimized Frame Size

A Thesis Presented By

Srijan Gupta

Roll No.: CS1505

in Partial Fulfillment of the Requirements for the

Master of Technology in Computer Science

degree of the

Indian Statistical Institute, Kolkata

Under the supervision of:

Dr. Sasthi C. Ghosh

Associate Professor

Advanced Computing and Microelectronics Unit

Certificate of Submission

This is to certify that *Srijan Gupta (Roll No.: CS1505)* has fulfilled the partial requirements for the degree of *Master of Technology in Computer Science* from the *Indian Statistical Institute*, Kolkata. The thesis entitled, *"Controlling the Frame Aggregation Rate in IEEE 802.11n WLANs to Obtain Optimized Frame Size"* was carried out under my direct supervision. No part of the thesis was submitted for the award of any degree prior to this date.

(Thesis Supervisor)
Dr. Sasthi C. Ghosh
Associate Professor
Advanced Computing and Microelectronics Unit
Indian Statistical Institute, Kolkata

Declaration

I hereby declare that the all the work depicted in the thesis was carried out by me as part of my Master of Technology in Computer Science studies under the supervision of Dr. Sasthi C. Ghosh, Associate Professor, Advance Computing and Microelectronics Unit, Indian Statistical Institute.

The thesis has been completely prepared and presented in accordance with academic rules and ethical conduct. All the materials and methods that are not original to this work, have been properly cited and referenced.

I also declare that no part of the thesis has been submitted for the award of any other degree prior to this date.

Roll No.: CS1505
Thesis Title: *Controlling the Frame Aggregation Rate in IEEE 802.11n WLANs to Obtain Optimized Frame Size*

(Signature of the Candidate)
Srijan Gupta
Indian Statistical Institute, Kolkata

Acknowledgments

I would like to express my deep sense of gratitude to Dr. Sasthi C. Ghosh, Associate Professor, Advanced Computing and Microelectronics Unit, Indian Statistical Institute,Kolkata, for giving me the opportunity to work under his guidance. His inspiring presence, motivation, and untiring efforts in providing technical advice helped me to go through various problems in course of thesis work.

Also, my sincere obligation goes to all the teachers for providing me the technical skill that will always remain as my asset.

Finally, I would like to thank my parents, friends for their continuous support throughout the tenure of my thesis work.

Regards,
Srijan Gupta

Abstract

The newly adopted IEEE 802.11n protocol has the scope of providing a much higher data rate, around 600 Mbps, with respect to the previous counterparts such as IEEE 802.11a/b/g. To provide this high speed, the IEEE 802.11n protocol has to adopt some key concepts like frame aggregation and channel bonding. In this work, we have developed an analytical framework to control the rate an individual user should aggregate frames in the presence of some important factors like partially overlapping channel, channel bonding, varying bit error rate over time, and bandwidth variation. The frame aggregation rate is chosen such that the throughput becomes maximum. We have considered two different frame aggregation techniques namely MAC Protocol Data Unit Aggregation (A-MPDU) and MAC Server Data Units Aggregation (A-MSDU). After showing when a user should use which frame aggregation scheme under partially overlapping channel condition, the effect of channel bonding is shown. A greedy algorithm is proposed that suggests which APs should be given the opportunity for channel bonding in order to increase the throughput. Then the results are validated by simulating the conditions assumed in the analytical model.

Contents

List of Figures

List of Tables

Chapter 1

Introduction

IEEE 802.11a/b/g networks have given us very successful Wireless Local Area Network (WLAN) protocols for communication between device and data exchange. But in recent times the number of applications using real time data has increased. With the increasing number of clients as well as the increasing demand of data rate, the IEEE committee introduced another protocol that is known as IEEE 802.11n. This protocol has several new technologies with respect to its predecessors in order to achieve a higher data rate, a maximum of 600 Mbps with respect to previous best of 54 Mbps (for IEEE 802.11a/g) or 11 Mbps (IEEE 802.11b).

One of the main techniques incorporated in IEEE 802.11n protocol is frame aggregation. Using this technique the smaller frames can be aggregated into one large frame. Because of that, there is a significant reduction in header information and contention period for channel access. As WLANs use Carrier-sense Multiple Access with Collision Avoidance (CSMA/CA) protocol to have error free transmission of frames, reducing the number of contending frames gives a huge boost to the achieved data rate.

Along with this, one more concept that helps in enhancing data rate is channel bonding. This feature is introduced in IEEE 802.11n and further work about this feature is going on for IEEE 802.11ac protocol. In channel bonding, if there is provision for an AP to use more than one non overlapping channels, then the AP can be assigned a bonded channel, where more than one channels are clubbed together. The clients connected to the AP can use the bandwidth of both the channel [5] [9].

1.1 Background

The IEEE 802.11n protocol can support two kind of frame aggregation techniques namely A-MSDU and A-MPDU [16] [4] [14] [11]. In the first technique, MAC Server Data Units (MSDU) are concatenated with individual subframe header and padding (if needed) to form a subframe. These subframes are concatenated one after another to form the payload of larger MAC Protocol Data Unit (MPDU). Then the MAC layer header is added in front of the MPDU and the frame check sequence (FCS) is concatenated with the MPDU to form Physical Service Data Unit (PSDU). With the PSDU, the Physical layer (PHY) header is added to form the *MSDU Aggregation* (A-MSDU). It is shown in Fig 1.1 [13].

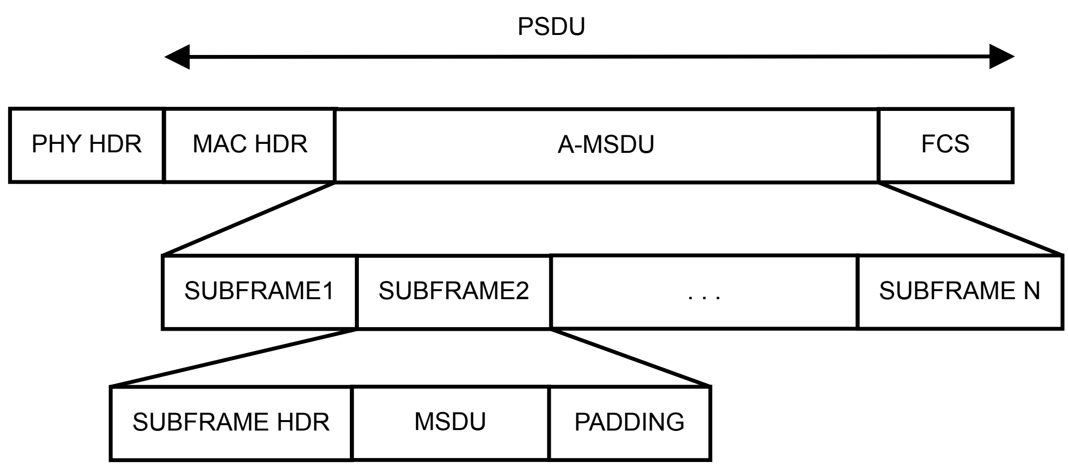

Figure 1.1: A-MSDU frame format

In the second technique, each MSDU is processed first. For each MSDU its FCS is calculated. The MAC layer header and the FCS is then appended with each MSDU to form a sub-MPDU. MPDU delimeter is then added with these sub-MPDUs and some padding bits are also added to make the size of sub-MPDUs a multiple of 4 Bytes. After this all the sub-MPDUs are concatenated with each other and they form a large PSDU. With the PSDU, physical layer header is added to form the *MPDU Aggregation* (A-MPDU). The frame format is shown in Fig 1.2 [11] [13].

In the two techniques, the algorithm for handling errors are also different. In A-MSDU technique, if one bit error occurs, then the full super frame needs to be

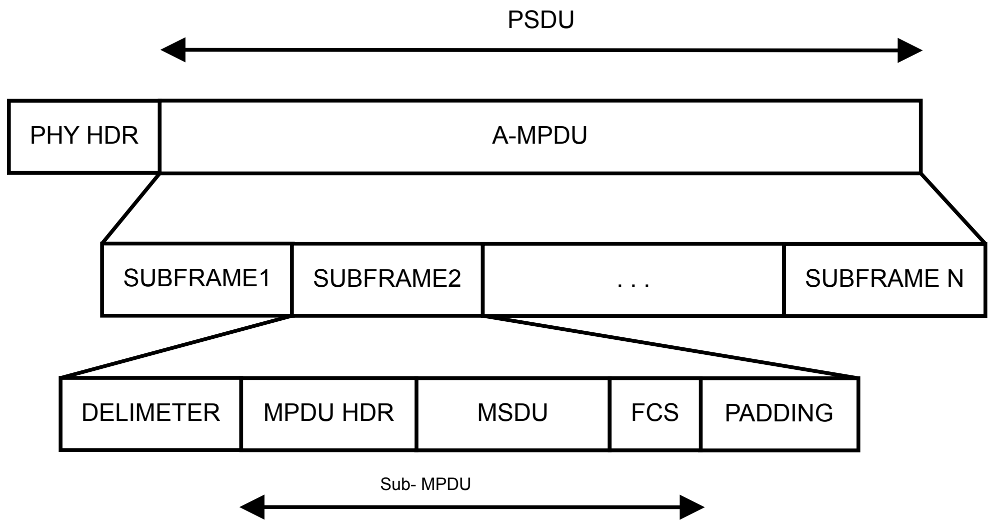

Figure 1.2: A-MPDU frame format

retransmitted. Whereas, in the A-MPDU technique, it can be easily checked, in which subframe the error occurred. Only that subframe needs to be retransmitted.

The environment in which all the users and the APs work together can be of two types, non-overlapping channel (NOC) and partially overlapping channel (POC). In NOC environment, only those channels are used, which are non overlapping to each other. They are assigned to the APs in such a way that none of the APs cause interference to other AP. But the clients can have interferences from other APs who use the same channel as the server AP of the client. Such client interference must be under some predefined threshold. In the POC environment, channels which are partially overlapping to each other, can also be used. In this case also APs are assigned channels in such a way, that none of the APs cause interference to another AP. But for the end users, sometimes the interference gets more due to the use of POC. In this case too, the client interference should be below a predefined threshold. There are 14 channels $[1, 2, ..., 14]$ in the 2.4 GHz for transmission. But, consideration of the first 11 channels is sufficient to discuss all possible scenarios. So, in this thesis, we will be dealing with first 11 channels. Any two consecutive channel's central frequencies are separated by 5 MHz. Two channels are known as POC if their difference in channel number is less than 5, in other word the difference between central frequencies are less than 20 MHz. On the other hand, two channels are considered to be NOC if their difference in channel number is 5 or more. Hence there can be at most 3 NOCs regardless of 11 or 14 channels are available. In the POC environment the use of channel bonding technique increases the interference to its legacy client (e.g., IEEE 802.11b/g). In channel bonding technique, more than one channel is assigned to an AP. Those two channels must be separated by 5 or more channels. We will look into these concepts in greater detail in the next chapter, Chapter 2.

Now increasing bandwidth or aggregating smaller frames increase the data rate but these have some trade off also. Aggregating frames increase the length of the packet, which in turn increase the expected number of retransmissions to successfully send a packet in an error prone channel. As the bit error rate can vary, the number of retransmission will also change from client to client and it will be a factor of time also, as the client can be in a moving situation. Bonding the channels also have issues like increased interference in legacy clients, increase in bit error rate (BER) as the data rate gets increased. Increase in BER also result in degradation of performance.

1.2 Motivation

In the previous works [13] [2], for a particular user, the optimal frame length is being measured depending upon the BER of the channel. The change of BER over time and the effect of change in bandwidth over time is not taken into account. In our work, we try to model an algorithm which can incorporate these limitations. In this thesis, we will show how the optimal size of frame aggregation changes over time for one particular user. More over in the previous works, the size of one frame length for both A-MPDU and A-MSDU is considered as same. But as the two schemes have two different error handling mechanism, i.e, in case of A-MSDU if there is some error, then the whole super-frame needs to be retransmitted, but in case of A-MPDU only the sub-frame in which the error occurred needs to be retransmitted. So, the frame size could be different in two schemes. Moreover, the sub-frame size for A-MPDU is taken as fixed in the previous works [13]. And based on that the number of frames that should be aggregated is controlled. But we have tried to show that for A-MPDU scheme, the sub frame length is a more important factor than number of frames that are aggregated. So, in this work, we gave a detailed analysis on how to vary the sub-frame length in A-MPDU scheme in order to get optimal throughput.

Along with these factors, we have also added the effect of POC [3] [6] [24] and channel bonding while measuring the optimal frame size. However, the issue of channel bonding is a very significant one. Since the number of NOCs is very limited, only 3 in this case, we need to assign them to APs very carefully so that it leads to overall throughput gain. Since bonded channel assignment is in general an NP-complete problem[15], we have given a greedy algorithm which assigns the extra channel to an AP by checking the throughput gain greedily.

1.3 Related Literature

Relating these issues there are some previous works. Yuxia Lin and Vincent W.S. Wong, in their paper [13] showed a method of how to compute optimal frame length for A-MSDU with a fixed bit error rate. In their paper they have taken into account error due to both collision and bit error rate. They have also shown what should be the optimal frame length with different number of users present in the network. They have taken into consideration both one-way transmission and two-way transmission of traffic. They have assumed all the frame lengths for both A-MSDU and A-MPDU is same and the network situation is also not changing.

In the paper [23] how a frame size can be optimized that is shown for IEEE 802.11 protocol. They have taken an error prone channel and then showed for optimizing the frame length how to deal between the two main concern as stated below:

- Decrease the overhead due to header files by increasing payload and
- Decrease number of retransmission due to error by decreasing payload.

Shin et al. showed how the performance of A-MSDU can be increased by a technique called physical layer link adaptation [21]. In their paper they have proposed a new link adaptation mechanism over IEEE 802.11 networks. Their technique does a cross layer interaction between MAC layer and Physical layer to jointly optimize different parameter and calculates a new packet error rate (PER). They have given an algorithm on how using the PER the throughput of MAC layer can be optimized.

In the paper [1] the authors gave a mechanism that helps to reduce the number of collision so that the throughput of IEEE 802.11n can increase. As high data rate is provided in IEEE 802.11n, one can be able to serve many clients, which leads to more collision of packets. In the paper the authors, Zakhia Abichar and J. Morris Chang proposed a group based MAC protocol, where stations are divided into groups and only the group leader will contend for a slot. Thus the waiting period and collision both will be reduced.

In paper [10] the authors have shown how the frame aggregation increases the throughput of IEEE 802.11b protocol. They have also proposed a simple mechanism to implement frame aggregation in practical scenarios.

The authors of the paper [5] identified the factors that affect the performance of the channel bonding in IEEE 802.11n. They have designed a network detector that can successfully identify interference conditions that affect channel bonding decision. Using that they have designed algorithms for channel bondings in an optimized way.

1.4 Scope of the thesis

The thesis is divided into different chapters. Chapter 2 which is followed by Chapter 1 (Introduction), lists some generalized terms and concepts related to wireless communication, wireless channels, different technologies associated with WLAN.

Once the commonly used terminologies and concepts are introduced, we begin with the model we are proposing and mathematical analysis of our model in chapter 3. In this chapter, we have three different sub-sections. In 3.1 we have discussed the assumptions that we made for our model. In the next section 3.2 we have discussed about the mathematical concepts for our work. In the beginning the equations behind the calculation of BER for an user over time is discussed. Then we have discussed about the parameters which decides the frame aggregation level. In our case the main parameter considered is throughput of the system. Then we have shown the throughput analysis of the A-MSDU and A-MPDU schemes.

An algorithm is proposed in 3.3 to show the step by step proceedings to calculate the mathematical model discussed, practically and the algorithm outputs the level of frame aggregation along with the aggregation mode.

After this, in Chapter 4 we have given some insight on how bonded channel work for IEEE 802.11n. Then we have discussed how channel bonding can effect the frame aggregation concept in section 4.1. At the end of the section we have given an greedy algorithm which identifies the APs that should be given extra channels for bonding based on the output of frame aggregation concept.

In the next chapter, Chapter 5 the results of the simulations done by us is shown. Using the simulation results we can compare between different modes of frame aggregation by tuning other parameters.

The last one is Chapter 6 about the future scope of the project and the conclusion that we can draw from the work of this thesis.

Chapter 2

Terminologies and Basic Concepts

This chapter actually discusses certain terms that are essential in understanding WLAN work environment, their effecting factors and theories behind the mathematics discussed later.

1. **WLAN [22] [12]:** *WLAN* is a computer network where the devices are not connected via wires but they are linked using a wireless distribution method within a limited area. The wireless links use high frequency radio waves. The main feature of the WLAN is that an user can move around and they still be connected with the network until they get outside the coverage area. Generally there are Access Points (AP) which work as center of attachment for the users. WLAN is known as IEEE 802.11 protocol.

   There are two different modes to connect clients with one another:

   - **Infrastructure Mode:** In this model each client is connected with an AP and they are in turn connected with other network. Thus a interconnected network is formed. Each client has to be connected to an AP in order to send or receive any data.
   - **Ad Hoc Mode:** In this model a set of computers associate with each other so that they can transfer frames to each other. In this mode clients do not need to join with any central entity in order to share data between themselves. This model is also known as **Peer-to-Peer or P2P** model.

   In the perspective of WLAN protocols, Infrastructure mode is more important than Ad Hoc mode. So,in my thesis I have also taken all the assumptions based on the protocols for Infrastructure mode. The two modes are shown in the Figure 2.1.

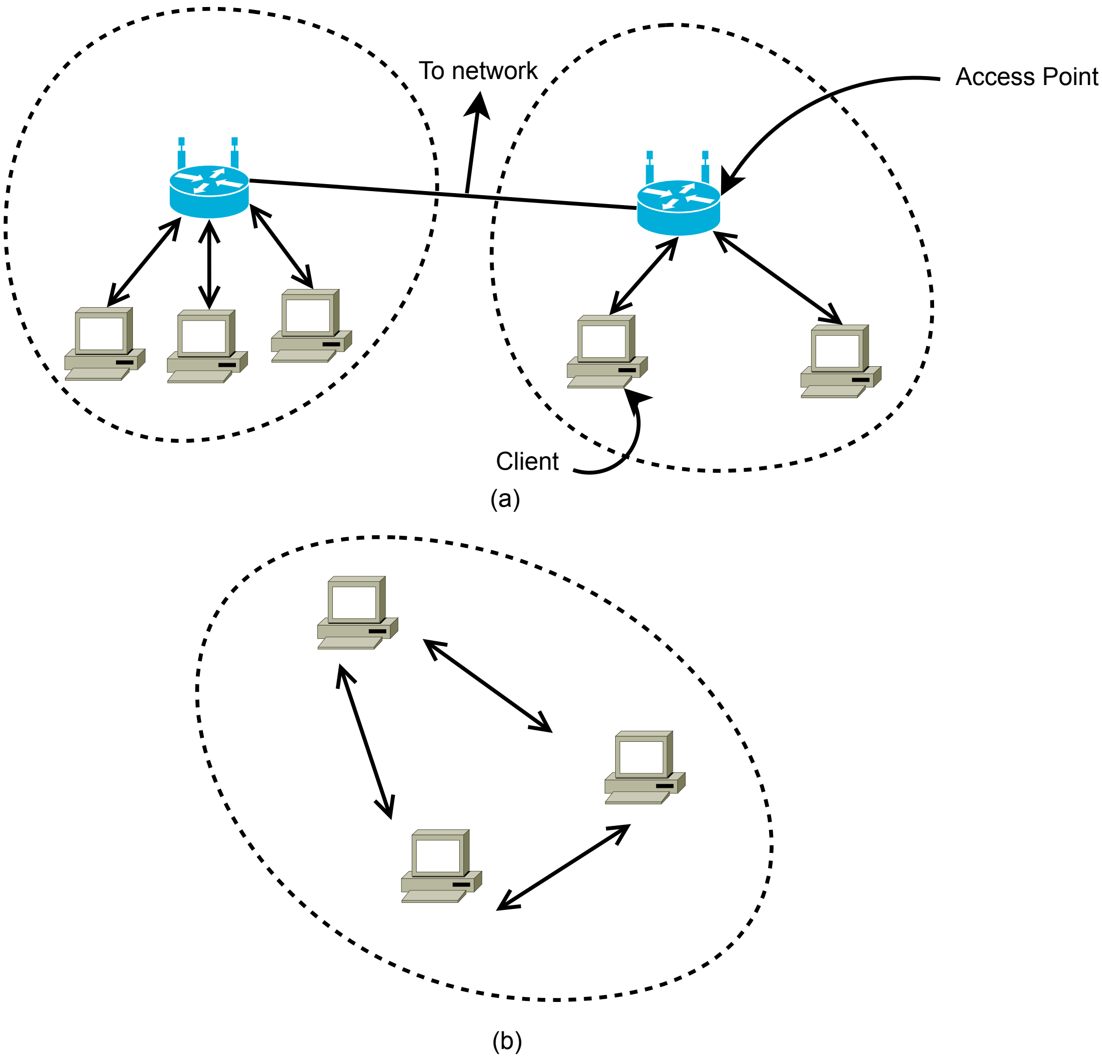

Figure 2.1: 802.11 architecture (a)Infrastructure Mode (b)Ad Hoc Mode

2. **Bit Error Rate (BER) [13]:** *BER* is the percentage of bits that is altered while transmitting some data through a channel due to noise, interference or distortion. BER is generally calculated using the simple formula:

$$BER = \frac{\text{number of bits altered}}{\text{number of bits sent}} \tag{2.1}$$

But when these numbers are unknown to us, we have to calculate BER using different mechanism. That method needs the value of Signal to Noise Ratio (SNR) value of the channel, bandwidth of the channel etc. We have explained that method later in 3.2.

3. **Channel Throughput [22] [12]:** In the networks and communication systems channel throughput refers to the effective data rate that one user is getting while using a particular channel. Channel throughput is often measured in bits/second. But current channel throughput can also be measured as the ratio of number of successful bits sent and total number of bits sent.

$$Throughput = \frac{\text{Number of successful bits sent}}{\text{total number of bits sent}} \tag{2.2}$$

   It gives the percentage value, that is currently achievable by the channel with respect to the maximum achievable rate.

   Some of the important factors that effect the throughput are:

   - **Noise:** Presence of noise effects badly on channel throughput. Presence of noise increases the chance of errors, which leads to more number of retransmission. That decreases the throughput value.
   - **Interference:** Interference from other devices also increases the chance of errors in the receiver side of a channel. So, the overall performance i.e throughput gets affected.
   - **Load in the system:** If the environment is highly loaded then error due collision increases, moreover the contention period of each user gets increased. So, the throughput gets affected badly.

4. **Signal to Interference plus Noise Ratio (SINR) [17] [7]:** *SINR* is measured for a device to calculate how much data rate it can achieve. It is the ratio of how much power a device is getting from its server and how much interference along with background noise it is getting from other devices and the channel. So, the simple formula for calculating SINR is

$$SINR = \frac{P}{I + N} \tag{2.3}$$

   where P is the power given to the device, I is the interference caused by other devices and N is the channel noise. SINR value is measured in dB. The channel throughput can be measured using the SINR value. Using Shannon's throughput limit the throughput can be calculated as 2.4

$$Throughput = W \log_2(1 + SINR) \tag{2.4}$$

5. **Different IEEE 802.11 protocols [22] [12] [25]:** The IEEE 802.11 protocols are known as wireless LAN protocols. Till today IEEE worked on different protocols in order to increase the data rate for users. The most widely used IEEE 802.11 protocols are IEEE 802.11a/b/g. But in recent times a huge increase in number of users and data requirement forced IEEE to work on a new protocol named IEEE 802.11n. This protocol has many concepts which are not implemented in any previous version of IEEE 802.11. A study of data rates achievable by different IEEE 802.11 protocols are shown in Table 2.1. The 802.11 protocol stack is shown in the Figure 2.2.

Protocol Name	Maximum Data Rate	Frequency Works On
802.11a	54mbps	5GHz
802.11b	11mbps	2.4GHz
802.11g	54mbps	2.4GHz
802.11n	300mbps	2.4GHz

Table 2.1: Comparison between different IEEE 802.11 protocols

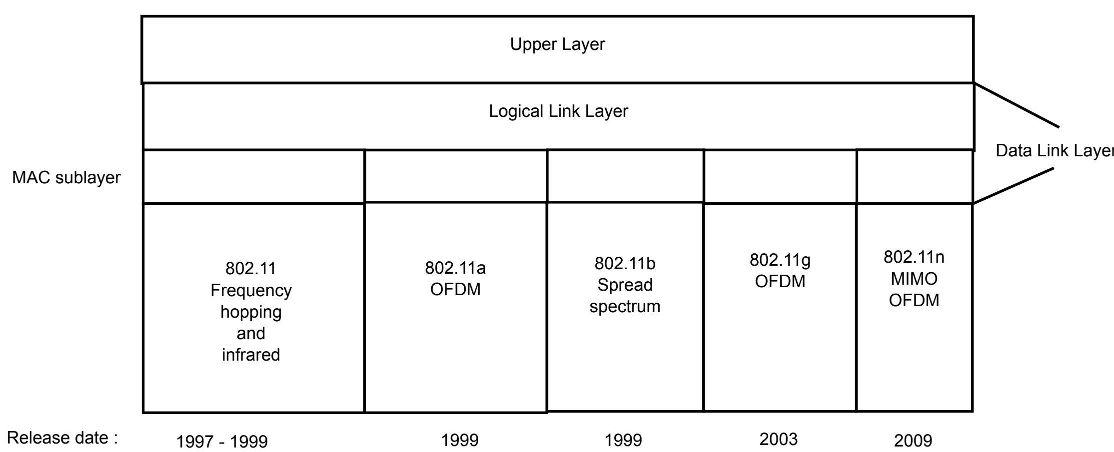

Figure 2.2: Part of the 802.11 protocol stack

IEEE recently is working on another IEEE 802.11 protocol named 802.11ac. But discussing the details of this protocol is out of the scope of this thesis. The different concepts that are playing role in IEEE 802.11n achieving a high data rate are:

- **Frame Aggregation:** One key concept that is introduced in the IEEE 802.11n is frame aggregation. This concept helps in reducing number of contending frames as well as redundant header bits. So it results in providing a high data rate.
- **Channel Bonding:** In IEEE 802.11n if at some point of time one channel becomes idle, that can be used by some other eligible AP. So, the devices who are incorporated with MIMO technique only they can utilize this facility. In later section we have discussed this issue in more details.

6. **Multipath Effect [19] [22] [13]:** In conventional wireless communication one sender antenna sends electromagnetic signal and that is received by the receiver antenna. Now in the signal propagation path if the signal is obstructed by hills, buildings or some other things, then the signal gets scattered. Now the different portion of the scattered signal follows different paths to reach to the destination. For that, different portion reaches at different time. This causes several problems such as fading, cut off and intermittent reception. In digital wireless communication systems this effect cause reduction in data speed and increase in errors.

7. **Antenna technologies for WLAN [19]:** Number of antennas used in receiver side and sender side in a wireless communication is known as antenna technology. There are four types of antenna technology:

   - **SISO:** This is the classical antenna model. Both the receiver and the sender uses single antenna. So, the technique is named Single Input Single Output. SISO is the simplest technique used. The throughput of the system depends upon bandwidth and signal to noise ration only. In this technique multipath effect creates problems.

   To get rid of these problems and to increase data rate multiple antenna techniques are adopted.

   - **SIMO:** In this technique instead of one receiver antenna, there are multiple receiver antennas, and one sender antenna. Which helps in sending a signal in different path to a receiver. SIMO systems usually were used for short wave listening and receiving stations to counter the effects the ionosphere fading.

- **MISO:** In this technique, sender side has more than one antenna to send signals. If some signal gets scattered others can reach the destination intact which helps in discarding scattered signals and thus the chance of error reduces.
- **MIMO:** Receiver and sender both have more than one antenna to send and receive signal. The IEEE 802.11n protocol uses this technique to reduce error and multipath effect.

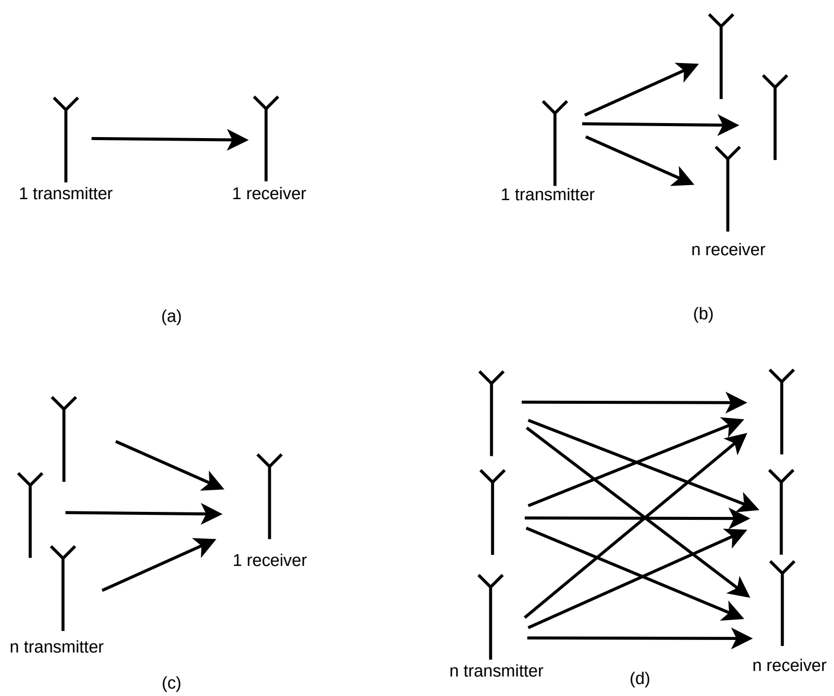

Figure 2.3: Different antenna technologies (a) SISO (b) SIMO (c) MISO (d) MIMO

8. **Carrier Sense Multiple Access (CSMA) [22] [12]:** CSMA are the protocols in which stations/clients listen to the carrier and accordingly tries to send the data. In this protocols all the stations can sense when the carrier

is busy sending another frame or when it is idle. Depending upon the condition the station decides whether to transmit the data of its own. In this protocol one station only sends data when it senses the channel as idle. But still there can be collisions as signal has a propagation delay. The situation is explained in Figure 2.4. So, it may happen station B started sending some

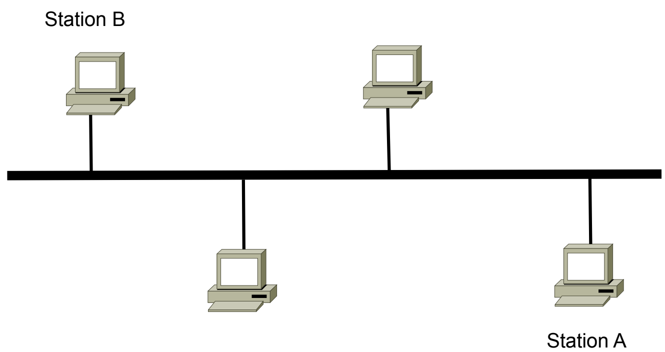

Figure 2.4: Collision situation in CSMA

data at time t which need Δt time to reach station A. Now if station A, at time t senses the carrier, it will think the carrier is idle. And station A can also start sending its data on the same carrier, and station A needs more than Δt time to transmit its data. Then there will be a collision between station A's data and station B's data. To avoid the costly resource loss, for Ethernet a new kind of protocol is used, namely CSMA/CD (CSMA with Collision Detection). When a collision is detected in the carrier, this protocol let the senders know about it, so that the senders can abort their transmission.

But in the IEEE 802.11 protocols, which uses wireless medium as transmitter, can not use CSMA/CD because of several reasons :

- Radios are nearly always half duplex, that mean they can not transmit and listen to the noise burst at the same time on a single frequency.
- The collision signal, that the stations receive generally are a million times weaker than the signal sent, so they ignore it as background noise.

Because of this, IEEE 802.11 protocols try to avoid the collision with a protocol called **CSMA/CA (CSMA with Collision Avoidance)**. In this protocol, also stations sense the channel before sending data and waits for a certain period, i.e, back off time after collision. The stations which has a frame to send, starts with a random back off time. Usually the number of slots chosen for back off period is between 0 to 15 in case of orthogonal frequency division multiplexing (OFDM). The channel waits for a short period of time called Distributed coordination function Inter-frame Space (DIFS), after sensing the channel is idle. Then it counts down idle slots and pauses when frames are sent. The frame is sent when the count reaches 0. If the frame reaches the destination successfully, a short ACK (acknowledgement) frame is sent back by the receiver. If the ACK frame is not received by the sender, it is considered that a collision has happened or some error occurred. The sender then doubles its back off time and try to retransmit the frame. This process continues until the frame is successfully transmitted or maximum number of retransmission is reached.

9. **Channels in IEEE 802.11 [17] [6] [18]:** In the 802.11 protocol, a station or client is served an AP. An AP can serve the clients associated with it using a frequency channel. In 2.4 GHz there are 11 frequency channels each of them 20 MHz wide. The channels are shown in the Figure 2.5. The

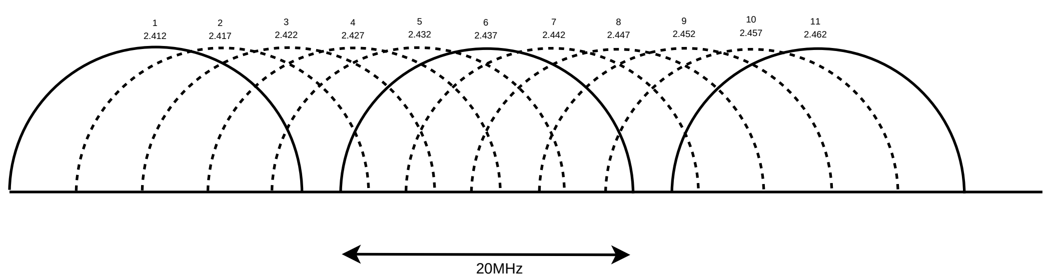

Figure 2.5: 2.4 GHz channels graphical representation

central frequency of any two consecutive channel is separated by 5 MHz. Now, there there can be two kind of channels:

- **Non Overlapping Channel (NOC):** If we take any two channel $channel_i$ and $channel_j$ from the channel set, then they are called non overlapping channels iff they have a channel separation of 5 or more. So, condition for NOC is :

$$|channel_i - channel_j| \geq 5$$

From this condition it can be easily seen that for 2.4 GHz spectrum, we can have at most 3 NOCs working at the same time, and they are $\{1, 6, 11\}$.

- **Partially Overlapping Channel (POC):** If the channels taken have a channel separation lesser than 5 then they are known as partially overlapping channels. POCs create interference to each other. Generally, lesser the separation between channels, more is the interference done by the channels. But that interference also depends on distance between the two APs that are using the channels. If the distance increases, then the interference decreases. If the two APs are situated at a suitable distance, they can use POCs without causing any interference to each other. The distance at which there will be no interference for POCs can be measured by a simple process described below [6]. At

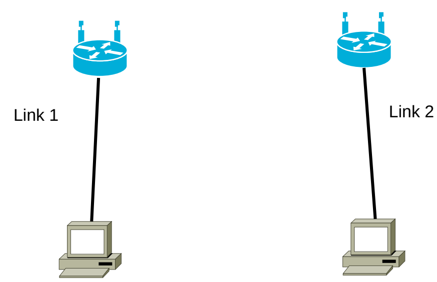

Figure 2.6: POC interference calculation set-up

first, the throughput of $link1$ and $link2$ are calculated as $throughput_1$ and $throughput_2$ by turning off the other link. After that turning on both the links, their new throughputs are measured $throughput'_1$ and $throughput'_2$. Now the Interference Factor is calculated as :

$$IF = \frac{throughput_1 + throughput_2}{throughput'_1 + throughput'_2} \tag{2.5}$$

Now, if the IF is equal to 1 then we say the links do not create any interference to each other. And when IF is lesser than 1, then the links create interference to each other. Now, by varying the distance between links in the case of POCs we can check when the IF is becoming 1, that is the distance of the links in which the two channels can operate without any interference. The distances for which there will be no interference for two POC is given in the Table 2.2 [6]. So, from

	I_0	I_1	I_2	I_3	I_4	I_5
2M	2R	1.125R	0.75R	0.375R	0.125R	0
5.5M	2R	R	0.625R	0.375R	0.125R	0
11M	2R	R	0.5R	0.375R	0.125R	0

Table 2.2: Partially overlapping channel interference range

the table it can be seen if two POC remain in 2R distance then they can operate on same channel without any interference. Where R is the radius of the user up to which the user can send data. So, 2R is the upper bound of distance between two POC channel to work without interferences. If the channel distance is equal to 5, then they can be used at same place.

10. **Channel Bonding [5] [18] :** In IEEE 802.11n the APs have an option to operate on a wider channel by using the technique of channel bonding. In this method instead of one 20 MHz wide channel, one AP can combine two non overlapping channel and operate on 40 MHz channel. The central frequency of the bonded channel then changes. One of these two channels are assigned as primary channel to the user and the other channel is secondary channel.

    Though combining two channels seem to have advantages over single channel, there are issues such as Increase in Bit Error Rate (BER), more interference on neighboring APs, decrease in number of channels which can be assigned to neighboring APs etc.

    The stations which follows IEEE 802.11n protocol can transmit in 20 MHz/40 MHz channels depending on the situation. The rules for this kind of transmission are as follows :

    - The station checks if the primary channel is idle for DIFS period and the back off counter time.
    - The station checks if the secondary channel is also idle for a PIFS time immediately preceding the back off time.
    - If both channels are idle for the required period of time, the station can start transmitting with 40 MHz channel.
    - If the secondary channel is not idle, then the station has two choices :

- The station can start sending the data in the 20 MHz wide primary channel.
- The station can choose a random back off time again and restart the waiting procedure for both the channel to send data using 40 MHz wide channel.

Chapter 3

Mathematical Analysis of the Proposed Model

In this chapter, the assumptions, mathematical models and analysis are shown at first. Then an algorithm is proposed in order to take the decision individually for each user. In first section the assumptions that we have taken is explained. In the second section the process of calculating SINR, in a partially overlapping channel condition is shown. Then how BER is calculated from SINR, that is shown. After that, the effect of BER on the packet size is calculated along with channel throughput. The calculations are done for A-MSDU and A-MPDU separately. In the second section a pseudo-code is given which shows step by step process of all the calculations.

3.1 Assumptions of the proposed model

We have assumed in this paper that each user has infinite number of frames to be sent. The time slots are divided into smaller slots. Lets assume the slot length is ΔT. The upper layer protocol has a maximum data unit for one packet. No packet can have size more than that. If a packet size is lesser, then by padding the bits are filled without disturbing the protocol. The size of one subframe is fixed for all the user and that defined as l_{MSDU}, for the A-MSDU model. For A-MPDU model the length of one subframe will vary depending upon BER b_i. We denote the subframe length of A-MPDU model as l_{MPDU} For both the model the protocol defined maximum length of a packet is L_{size}. We have also assumed that there are two kind of nodes available in the environment, one is APs and the other is

Clients (users). The set of APs are denoted by S and let there are m APs. So, $S = \{s_1, s_2, s_3..., s_m\}$. Under each AP there are a number of Clients who are being served by that AP. Now, let R_i is the maximum data rate achievable by user i. And required rate by i^{th} user is r_i.

3.2 Mathematical Analysis of Proposed Model

We define the frame aggregation rate ρ as:

$$\rho = \frac{\text{number of frames that should be aggregated}}{\text{maximum number of frames that can be aggregated}} \tag{3.1}$$

For i^{th} user the aggregation rate is defined as ρ_i. The value of ρ_i varies from $1/N_i$ to 1. N_i is the maximum number of frames that can be aggregated for the i^{th} user. Now for each user, over time the bit error rate and the bandwidth will change. Depending upon that over time for each user the value of N_i will change and so will the aggregation rate. For gaining the required data rate the expected number of retransmission for a frame lesser than a threshold value. And for some user the application may not even start because of high expected number of retransmission due to high BER. For the i^{th} user, we need to take the minimum of L_{size} and $\Delta T \times R_i$ in order to get the maximum frame size that can be sent by the user as well as can be handled by the protocol. Now for the A-MPDU frame aggregation method, the value of N_i is given by:

$$N_i = \lfloor \frac{min(\Delta T \times R_i, L_{size}) - L_{phyhdr}}{l_{MPDU\,i}} \rfloor \tag{3.2}$$

where the L_{phyhdr} denotes the length of physical layer header.

For, A-MSDU frame aggregation method, the equation becomes:

$$N_i = \lfloor \frac{MSDU_{size}}{l_{MSDU}} \rfloor \tag{3.3}$$

where $MSDU_{size} = min(\Delta T \times R_i, L_{size}) - L_{phyhdr} - L_h - L_{fcs}$. And L_{phyhdr} means the size of physical layer header, L_h denotes size of mac layer header length and L_{fcs} denotes the length of Check Sum.

Now, we have to calculate the SINR value for each user in order to calculate the BER [7]. The SINR calculation for NOC environment is as follows:

$$SINR(s,i) = \frac{F_s d(s,i)^{-\alpha}}{\sigma_n^2 + \sum_{v \in S, v \neq s} F_v d(v,i)^{-\alpha}} \tag{3.4}$$

Here F_s is the power used by the s^{th} AP and σ_n^2 is the white Gaussian noise present in the environment, $d(s,i)$ is the Euclidean distance between s^{th} AP and i^{th} client and s^{th} AP is the server AP of i^{th} client.

We calculate the distance using the equation

$$d(v,i) = \sqrt{(v_x - i_x)^2 + (v_y - i_y)^2} \tag{3.5}$$

where (v_x, v_y) is the co-ordinate of v_{th} AP and (i_x, i_y) is the co-ordinate of i_{th} client.

If we use the POC conditions then the SINR calculation will look like [24]:

$$SINR(i) = \frac{F_s d(s,i)^{-\alpha}}{\sigma_n^2 + \sum_{v \in S, v \neq s} F_v d(v,i)^{-\alpha} \gamma(u,v)} \tag{3.6}$$

Here $d(u,v)$ denotes the Euclidean distance from u to v. The $\gamma(u,v)$ is the channel overlapping degree of the channels used by u and v. α is the path loss index of the channel. ${\sigma_n}^2$ is the white noise present in the background. For a particular client, any AP that using a channel j, will not create any interference to it, if the channel distance between j and the channel the client is using has a channel difference equal to or more than 5. Moreover if the AP uses same channel as the client, and if their distance is more than 2R then also there will be no interference caused. For other channels also there is a certain distance which we have shown in the Table 2.2 Here R is the maximum distance of the AP and client.

The value of $\gamma(u,v)$ is given in the Table II [3] [24]. This table gives us the γ

Channel Distance	0	1	2	3	4	5
Overlapping Degree	1	0.727	0.271	0.037	0.005	0

Table 3.1: Overlapping degree with channel distance

values Now, the interference factor depends on both the channel distance(δ) and physical distance between two nodes. So, interference factor is defined as [8] [17]:

$$IF_{i,j} = \begin{cases} \frac{IR(\delta)}{d_{i,j}} & \text{when } 0 \leq \delta < 5 \text{ and } d_{i,j} < IR(\delta) \\ 0 & \text{otherwise} \end{cases} \tag{3.7}$$

Here $IR(\delta)$ is the interference range for channel separation $\delta = |i - j|$. $d_{i,j}$ is the distance between the two users operating on channel i and j.

Now we know that [22] [20],

$$SINR = \frac{E_b}{N_o} \times \frac{R_b}{B} \tag{3.8}$$

where E_b is energy per bit, N_o is noise power spectral density, R_b is bit rate in bits/second and B is noise bandwidth. From equation 3.8 we can easily calculate the $\frac{E_b}{N_o}$ value for i^{th} user. The relation between BER and $\frac{E_b}{N_o}$ is given by the following equation [22]:

$$BER = 0.5 \times erfc(\sqrt{\frac{E_b}{N_o}}) \tag{3.9}$$

So, using equation 3.9 we can easily determine the BER b_i for each user. Now we define the throughput in terms of expected number of retransmission needed for a packet for each user. The function $erfc(x)$ is defined as, for $x > 0$

$$erfc(x) = \frac{\Gamma(\frac{1}{2}, x^2)}{\sqrt{\pi}} \tag{3.10}$$

where $\Gamma(a, b)$ is the incomplete gamma function.

We define $E[X_i]$ as the expected number of retransmission needed for i^{th} user. If the user uses A-MSDU technique then one bit error results in retransmission of entire packet. But in case of A-MPDU only the packet which was damaged is needed to be sent. So, for A-MPDU we will consider one packet is successfully sent if and only if all the subframes are sent successfully individually. Now for the i^{th} user the expected number of retransmission is calculated differently for A-MPDU and A-MSDU technique.

In case of A-MSDU, lets assume at an particular instance the BER is b and total number of bits sent is n. Now let us assume that p is the probability of sending a frame consisting of n bits successfully. So, the probability p is calculated using equation

$$p = (1 - b)^n \tag{3.11}$$

Hence, $1 - p$ is the probability of failure of a frame of length n bits. Now, let us assume, a particular frame gets successfully delivered at the k^{th} attempt. The probability of this event is denoted by P_k. So,

$$\begin{aligned} P_k &= \text{Probability that a frame takes } k \text{ attempts to get delivered} \\ &= \text{The frame takes } k-1 \text{ failed attempts before 1 successful attempt} \\ &= p \times (1-p)^{k-1} \end{aligned}$$

So, the probability P_k is calculated by the equation

$$P_k = p \times (1-p)^{k-1} \tag{3.12}$$

Now we can calculate the expected number of retransmission required to successfully transmit a frame by calculating $\sum_{k=1}^{\infty} kP_k$. Now,

$$\begin{aligned} \sum_{k=1}^{\infty} kP_k &= \sum_{k=1}^{\infty} kp \times (1-p)^{k-1} \\ &= p \times \sum_{k=1}^{\infty} k(1-p)^{k-1} \\ &= p \times \frac{1}{p^2} \\ &= \frac{1}{p} \end{aligned}$$

So, the expected number of retransmission is

$$E[Retrans] = \frac{1}{p} \tag{3.13}$$

Then the expected number of retransmission for the i^{th} user is

$$E[Retrans_i] = \frac{1}{(1-b_i)^{n_i}} \tag{3.14}$$

where n_i is the length of frame for i^{th} user in bits.

The throughput for the A-MSDU is given by the formula stated below

$$\begin{aligned} T_{A-MSDUi} &= \frac{actual\, number\, of\, data\, bits\, sent}{total\, number\, of\, bits\, sent} \\ &= \frac{MSDU_{size}}{E[Retrans_i] \times Total\, frame\, length} \\ &= \frac{n_i - L_{phyhdr} - L_{fcs} - L_{machdr}}{E[Retrans_i] \times n_i} \end{aligned}$$

So, the throughput is calculated as

$$T_{A-MSDUi} = \frac{n_i - L_{phyhdr} - L_{fcs} - L_{machdr}}{E[Retrans_i] \times n_i} \tag{3.15}$$

Depending upon the values of n_i and $E[Retrans_i]$ the value of $T_{A-MSDUi}$ will change. We will take the n_i value for which the throughput becomes maximum for the BER b_i. So, after getting the n_i value we can determine the number of frames that should be sent for the i^{th} user as follows:

$$N_{maxi} = \lfloor \frac{n_i - L_{phyhdr} - L_{fcs} - L_{machdr}}{l_{MSDU}} \rfloor \tag{3.16}$$

Here the L_{phyhdr}, L_{fcs} and L_{machdr} are the length of physical header, FCS value and MAC layer. l_{MSDU} is the length that a subframe can have.

For, A-MPDU the calculation is a bit different. In this mode suppose the super frame is generated by concatenating r sub frames. Lets, assume all the sub frames have length l_{MPDUi} for the i^{th} user. All the sub frames in this mode have self error detecting mechanism, i.e, each sub frame has its own FCS value. So, if one sub frame is erroneous, it can be retransmitted separately. In this case the expected number of retransmission needed for a super frame is calculated using the expected number of retransmission of a subframe.

$$\begin{aligned} E[Retrans_{Superframe}] = max(&E[Retrans_{subframe1}], \\ &E[Retrans_{subframe2}], ..., E[Retrans_{subframer}]) \end{aligned} \tag{3.17}$$

Now for all the subframes, as their size is same, the $E[Retrans_i]$ value will be same for all subframes. Hence, in this mode the expected number of retransmission($E[Retrans_i]$) needed for a super frame is as same as that needed for a subframe. We need to determine the subframe length for which the $E[Retrans_i]$ becomes least. In this case the subframe length is not fixed. The subframe length can vary depending upon the value of b_i in order to minimize $E[Retrans_i]$ value. The $E[Retrans_i]$ value for one sub frame is

$$E[Retrans_i] = \frac{1}{(1-b_i)^{l_{MPDUi}}} \tag{3.18}$$

The throughput is given as

$$T_{A-MPDUi} = \frac{l_{MPDUi} - L_{delimeter} - L_{mpduhdr} - L_{fcs}}{E[Retrans_i] \times l_{MPDUi}} \tag{3.19}$$

We calculate the value of l_{MPDUi} for which the value of T_{A-MPDU} becomes maximum. That will be the length of one sub frame for the BER b_i. So the number of subframes that we can aggregate in the A-MPDU scheme is

$$N_{A-MPDUi} = \lfloor \frac{L_{size}}{l_{MPDUi}} \rfloor \tag{3.20}$$

Now, if we consider the situation where both the schemes can send a packet with no padding, i.e, they utilize the maximum packet length fully, then using A-MSDU is better than A-MPDU. Because in A-MPDU header informations are more in size than A-MSDU. This situation can happen when the BER is very low. But when the BER becomes higher, A-MSDU starts using more padding bits instead of data bits. But A-MPDU handles this problem by adding more subframes. As, adding subframes also does not change the value of $E[Retrans_{MPDU}]$ So, we can tell that, for lower bit rate using A-MSDU can be beneficial, but when the BER is high then using A-MPDU gives better throughput.

Now, as the parameters vary with time, by using our algorithm, all the users can dynamically decide about the mode and optimal size of the frame. So, for each point of time the users will be able to achieve maximum throughput that can be achieved in that situation. Also this is achieved without disturbing the upper layer protocol as we have the provision for padding in case of smaller frame size.

3.3 Algorithm

In the previous section the mathematical formulas and their significances are discussed. In this section I am going to give an algorithm with the help of the mathematics discussed in the previous section on based on situation which aggregation scheme we should use and what should be the subframe length. The length of the subframe will be determined based on throughput value of the channel. As input we take the set of AP, S along with the co-ordinates of all the AP S_{pos} and the powers they are working with. We take the powers of all the AP is same and that is P. Then we will have the client set C and the co-ordinates of the clients are also given in the set C_{pos}. The client radius R is given. Bandwidth B of the channel is fixed as we have taken only 2.4 GHz case. Data rate for each user is also given as set r. Other values that are provided are :

- Maximum data unit size that can be handled by protocol, $MAXLEN$
- Subframe length for A-MSDU model, l_MSDU
- Physical layer header length, L_{phyhdr}
- MAC layer header length, L_{machdr}
- FCS length for each subframe in A-MPDU model $L_{fcs_{MPDU}}$
- FCS length for the super frame in A-MSDU model $L_{fcs_{MSDU}}$
- Delimiter length for each subframe in A-MPDU model $L_{delimiter}$
- Header length for each subframe in A-MPDU model $L_{mpduhdr}$

The algorithm outputs which mode one user should use, i.e, A-MSDu or A-MPDU, maximum achieved throughput T_{MSDU} or T_MPDU based on the previous decision and number of frames that can be aggregated N_i. Using the algorithm each user can dynamically change their frame aggregation rate to achieve maximum throughput and they can easily select which method they should select for aggregating. In case of A-MPDU the subframe length is varying, but to maximize gain in this mode, the super-frame is fully loaded with the subframes.

Algorithm 1: *Mode selection and Optimal frame length*

$MaxThroughput_{MPDU} = 0, MaxThroughput_{MSDU} = 0, OptLen_{MPDU} = 0, OptLen_{MSDU} = 0, OptLen = 0, MinDist = 0$;

for $i = 1, 2, \ldots n$ **do**

  Calculate the distance of c_i from all the AP s_j, $j \in \{1, 2, \ldots m\}$ as $d(i, j)$ from Equation 3.5;

  Set $MinDist_{i,j} = min(d(i, 1), d(i, 2), \ldots d(i, m))$;

  if $MinDist > R$ **then**

    then output *This user is not connected to any AP* ;

    Exit ;

  Calculate $SINR(i, j)$ using the Equation 3.6 and 3.10. The value of $\gamma(a, b)$ is taken from the table3.1 ;

  Calculate $\frac{E_b}{N_o}$ using the equation 3.8;

  Calculate BER using the equation 3.9;

  for $k = 1, 2, \ldots MAXLENGTH$ **do**

    Calculate successfull transmission probability p for frame length k using the Equation 3.11;

    Calculate expected number of retransmission $E[Retrans_{MSDU}]$ for A-MSDU scheme using the Equation 3.13 ;

    Calculate throughput for $A - MSDU$ model $ThroughPut$ for frame length k using the Equation 3.15 ;

    if $ThroughPut \geq MaxThroughput_{MSDU}$ **then**

      $MaxThroughput_{MSDU} = ThroughPut$;

      $OptLen_{MSDU} = k$;

    Calculate expected number of retransmission $E[retrans_{MPDU}]$ for $A - MPDU$ model using Equation 3.18 for k bit long sub-frame;

    Calculate throughput for $A - MPDU$ model $ThroughPut$ for sub-frame length k bit, using the Equation 3.19 **if** $ThroughPut \geq MaxThroughput_{MSDU}$ **then**

      $MaxThroughput_{MSDU} = ThroughPut$;

      $OptLen_{MPDU} = k$;

    if $MaxThroughput_{MSDU} > MaxThroughput_{MPDU}$ **then**

      $OptLen = OptLen_{MSDU}$;

      The number of frames can be calculated from $OptLen$ using Eqation 3.16 ;

    else

      $OptLen = OptLen_{MPDU}$;

      The number of frames can be calculated from $OptLen$ using Eqation 3.20 ;

  Output the selected mode and number of frames.

Chapter 4

Channel Bonding and Its effect

In this chapter we will see how another aspect of IEEE 802.11n, that is channel bonding works along with frame aggregation. We will see how it effects the throughput of the users and how we can choose APs which will be eligible for channel bonding by suggesting a greedy algorithm.

4.1 Effect of Channel Bonding

One very important feature of IEEE 802.11n is channel bonding [9] [5]. In this technique more than one channel is assigned to one AP to provide a higher bandwidth to its connected users. The change in bandwidth also brings change in BER for users connected with that AP. In this work we also have checked whether an IEEE 802.11n AP should be given the opportunity of channel bonding or not. We have analyzed the effect of channel bonding on a AP and its neighbors, and then we have proposed a greedy algorithm for the assignment of Channel Bondings. For all APs the channel bonding technique may not improve the throughput as it should. Depending upon the BER, throughput might get decreased for some user.

We assume that there are a total of n APs. Among them m APs are using the channel bonding technique. We denote the set of APs who use channel bonding as $CBAP = \{cbap_1, cbap_2, ..., cbap_m\}$ So the rest of the $n - m$ number of users are named as normal users, and the set is $NAP = \{nap_1, nap_2, ..., nap_n\}$. $cbap_{i_{m\_n}}$ denotes that i^{th} AP uses the two channel (m, n). While calculating the interference we denote

$$channel\_dist(m\_n, l\_k) = min[channel\_dist(m, l), channel\_dist(n, l), \\ channel\_dist(m, k), channel\_dist(n, k)] \quad (4.1)$$

where $channel\_dist(a, b)$ denotes the channel distance between two single channel a and b, $channel\_dist(a\_b, c\_d)$ calculates the channel distance between two bonded channel user AP who are using channel a_b and c_d. Similarly we can denote $channel\_dist(a\_b, c)$ to calculate channel distance between a bonded channel using AP and a normal AP. After getting the channel distances, the SINR can be found using a equation similar to Equation 3.6. Here the γ value will be depend upon $channel\_dist$. Now we use greedy approach to find whether an AP should be given bonded channel or not.

4.2 Proposed Algorithm

Initially we will consider all the APs as normal AP who are using a single channel. Then we choose an AP randomly and see if there is a potential idle channel that can be assigned to it for channel bonding. For finding the potential idle channel we have taken two conditions that needs to be satisfied-

- If the channel number of the AP is i then only the channel(s) j_i will be one potential candidate for idle channel if $|i - j_i| = 5$.
- If any of the neighbouring AP of the candidate AP do not use the channel j_i only then j_i can be considered as potential idle channel.

For example if an AP is working on channel number 6, then there can be two potential channels - 1 and 11, that can be assigned to that AP for channel bonding. Suppose one of the neighbor of that AP uses channel 1 and no other neighbor uses channel 11, then channel 11 will be a potential idle channel for the AP. We can calculate the SINR of a client in bonded channel environment using the same equation as 3.6 where the channel distance is calculated using 4.1. If there are more than one potential idle channel then we have calculated the total interference caused to neighbors for all the cases and choose the channel which causes least interference. In that way, channel bonding scheme can be applied to the network by taking care of throughput of the concerned AP and its neighbors. The algorithm is as follows :

Input: Set of APs $AP_{normal} = \{AP_1, AP_2, \ldots, AP_n\}$ along with their channels and co-ordinates

Output: Set of APs AP_{bonded} which are using channel bonding and their channel number

Set $ChannelSet = \phi, AP_{bonded} = \phi, throughput = 0, gain = 0, channel = 0$;

for $i = 1, 2, \ldots n$ **do**

    c_{AP} = Channel number of AP_i;

    The potential idle channel set is $ChannelSet = \{j_k\}$ where $|c_{AP} - j_k| = 5$ and j_k is not used by any neighbor of AP_i. $c_{AP}, j_k \in \{1, 2, \ldots, 11\}$;

    for *Each channel k in ChannelSet* **do**

        Calculate gain in throughput of the clients of AP_i ;

        Calculate loss in throughput of the clients of the neighbors of AP_i;

        Calculate *gain* as average gain in throughput - average loss in throughput ;

        if $gain > throughput$ **then**

            $channel = k$;

    $AP_{bonded} = AP_{bonded} \cup AP_{i,channel}$;

Output AP_{bonded} set

The algorithm selects the APs suitable for channel bonding and assigns channels to them accordingly. After the assignments are made we calculate the SINR for users and depending upon that, their frame aggregation rate is changed again.

Chapter 5

Results

In this section we have discussed about the implementation of the mathematical model that we have presented in the Chapter 3. At first the practical scenario that I have taken is explained. Then the results are shown which gives a comparative study between previous assumptions taken and our proposed method.

In our setup we have considered same WLAN environment as [17]. A number of APs and clients are distributed uniformly over a $100 \times 100\ m^2$ area.

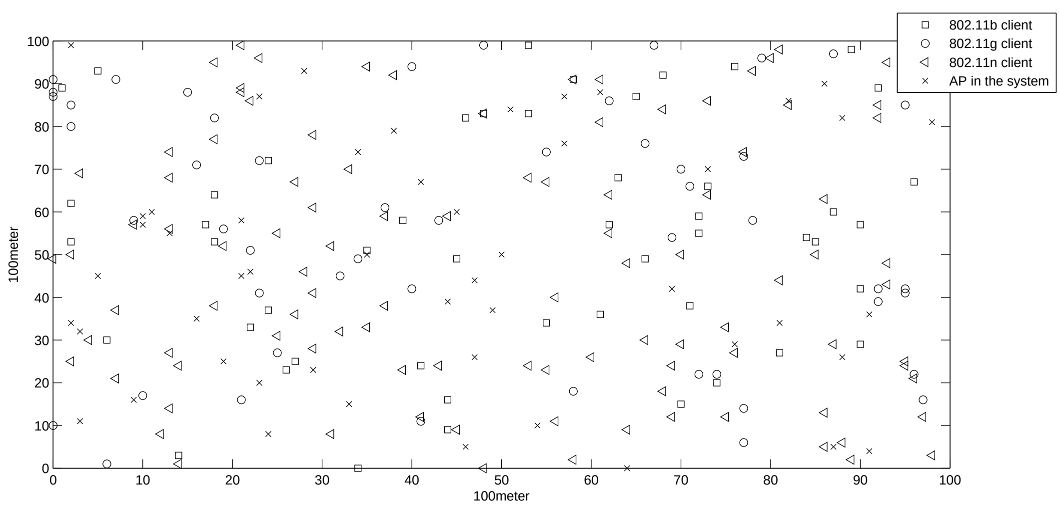

Figure 5.1: Different devices are scattered in the rectangular area

The number of AP is taken as 50 and the number of client is 200. In our simulation we considered NOC condition as there are only 3 channel available namely $\{1, 6, 11\}$. When we have considered POC environment, then the channel set we have taken is $\{1, 2, \ldots, 11\}$. We have considered a heterogeneous type of clients. 25% of the clients are IEEE 802.11b, 25% of the clients are IEEE 802.11g and rest 50% clients are IEEE 802.11n. Type of each client is selected randomly. One possible set-up of the environment is shown below in Figure 5.1

We have calculated the bit error rate each user is getting for a particular set-up. We have calculated the BER for each user in NOC mode, in NOC+POC mode and NOC+POC+Bonded Channel mode. The BER is dependent upon how much interference it is getting due to POC and Bonded Channel.

Now, when the things get boiled down to a single factor, that is BER, we have shown the results that our method is achieving.

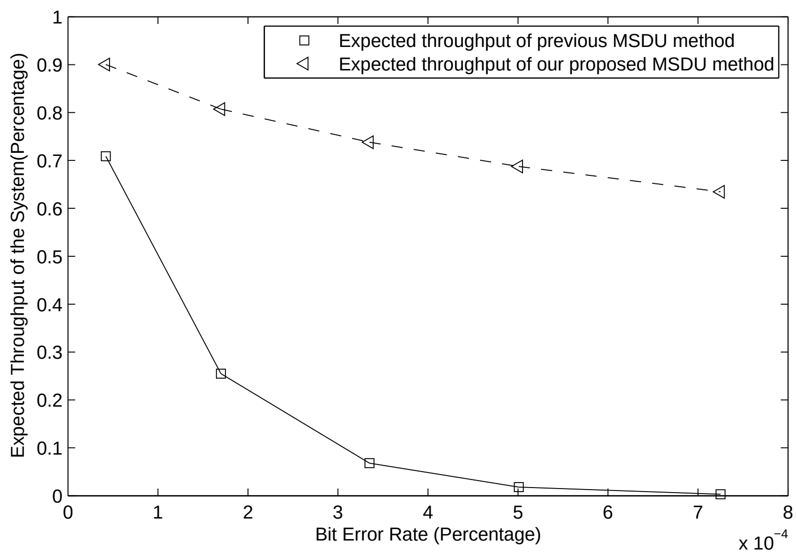

Figure 5.2: Proposed throughput gain for A-MSDU

The figure 5.2 shows the throughput comparison for A-MSDU mode, between our proposed method and the classical frame aggregation method that aggregates a fixed length or full length and does not vary over time. In this test I have taken BER in order of 10^{-4} as the WLAN generally have BER in the order of 10^{-3} – 10^{-5}. From the figure we can conclude see that as the BER is decreasing both the methods are gaining more and more throughput. But as the BER is increasing, then our proposed method is giving a much higher throughput than the previous method. While in the low BER zone, a slight increase in BER is causing massive downfall in throughput for the former method, but in our proposed method the rate of decrease is more stable.

The next figure 5.3 shows the throughput gain for A-MPDU mode.

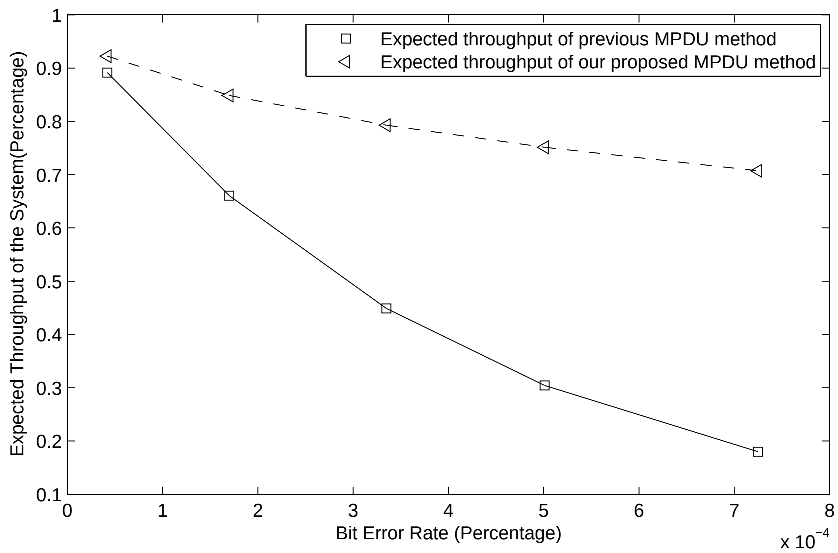

Figure 5.3: Proposed throughput gain for A-MPDU

Next I have compared how the throughput varies for A-MPDU and A-MSDU mode in my proposed method in Figure 5.4. The result shows that in the higher bit error rate A-MPDU has a higher throughput than A-MSDU. That happens because expected retransmission increases more for A-MSDU than A-MPDU. However, as the BER decreases their respective throughput comes closer. At a very low BER, the throughput of A-MSDU becomes more than that of A-MPDU.

Next the graph 5.5 is shown about the frame size variation of A-MSDU mode with the BER. As the BER is increasing, the optimal length gets decreased. This is very easy to understand. Greater size means greater the chance of being erroneous, and that increases expected number of retransmission. So, to have optimal throughput, the optimal length needs to change with BER over time for a single user.

Similar thing happens for the A-MPDU mode also. The result in the graph form is shown in Figure 5.6.

In the last graph, Figure 5.7 I have compared how the frame length for A-MSDU and subframe length for A-MPDU varies. It represents a stark contrast

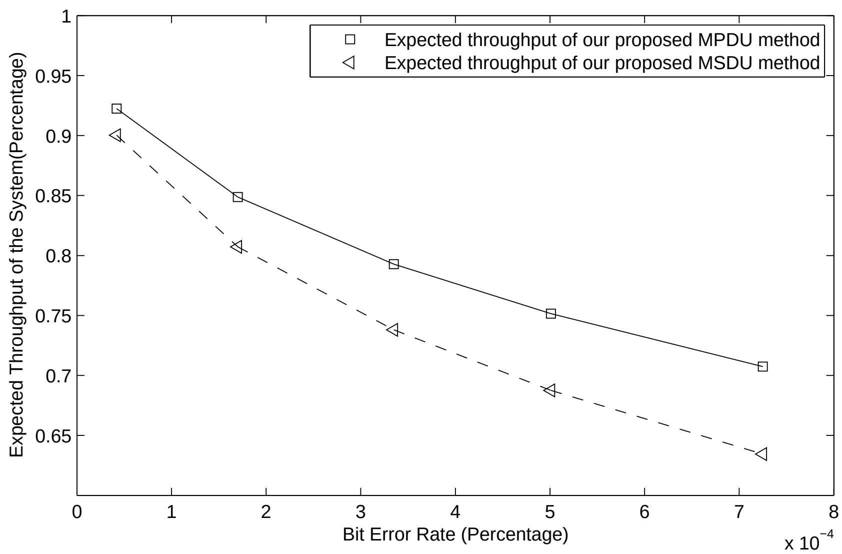

Figure 5.4: Comparison of throughput gain between A-MPDU and A-MSDU

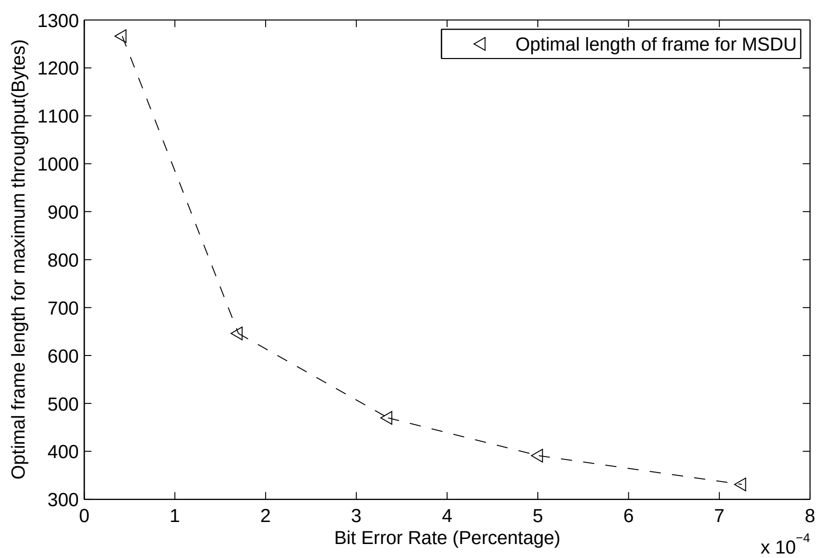

Figure 5.5: Optimal length of frame in A-MSDU

with former method, where the subframe length of A-MSDU and A-MPDU was

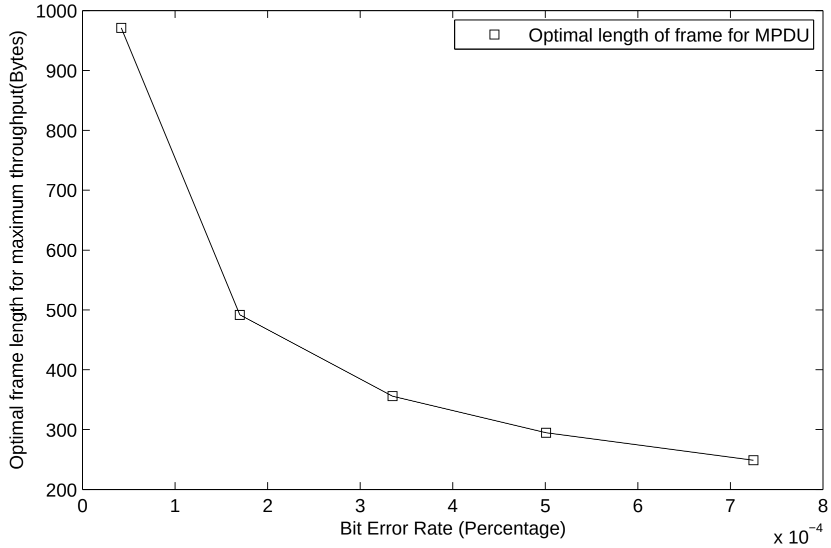

Figure 5.6: Optimal length of frame in A-MPDU

taken as same. The result is showing for achieving high throughput, A-MPDU needs a subframe length almost as similar as a frame length of A-MSDU mode in similar conditions.

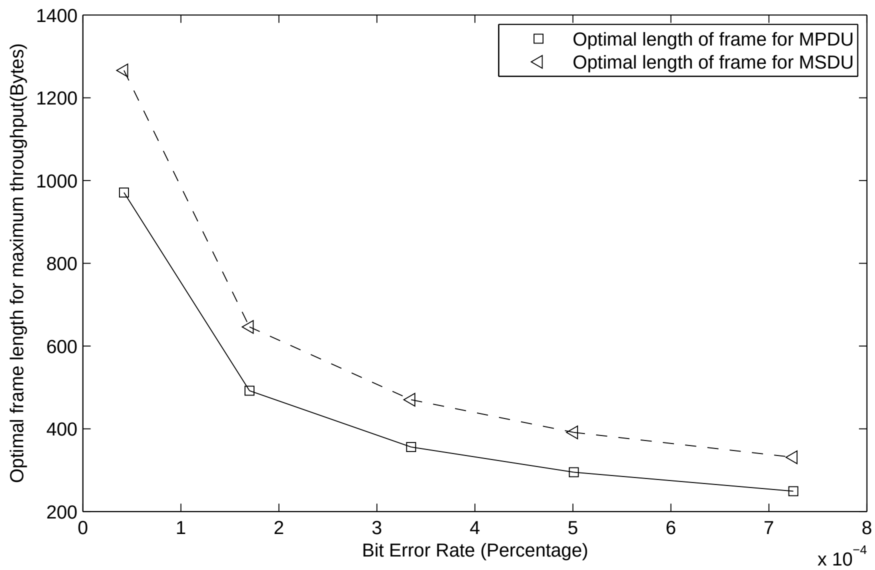

Figure 5.7: Optimal length comparison between A-MPDU and A-MSDU

Chapter 6

Conclusion and Future Scope

In this chapter I have tried to summarize the thesis work I have done. Also further enhancements and few other generalizations that can be done is discussed.

6.1 Conclusion

In the entire thesis work I have tried to shown the necessity of using dynamic frame aggregation model in order to achieve maximum throughput. In our day to day scenario where clients are very much mobile in the WLAN environment, and they require much higher data rate for more than one application, the frame aggregation method should be so much adaptive that it can handle all the newly proposed enhancements of the protocol as well as provide a higher throughput.

In Chapter 3 we have studied how the IEEE 802.11n protocol should adapt its frame aggregation techniques with the changing environment, i.e, in NOC or NOC+POC or NOC+POC+Bonded channel, changing required data rate, changing Bit Error Rate etc. as a function of time. Every now and then some of these factors can change for a user and the user must change its frame aggregation technique readily.

In Chapter 4 I have talked about the Bonded Channel environment and how it affects the throughput of a client. Moreover a greedy algorithm for assigning channel bondings to the APs is also proposed in the light of frame aggregation.

However, the main summary of this thesis work is, in IEEE 802.11n the frame aggregation methods should know how much they should aggregate in different situations, where factors can change over time, rather than sticking to aggregate fully all the time or compute the optimal frame length once.

6.2 Future Research Scope

The concept of frame aggregation and its application in IEEE 802.11n is very significant. We can study this concept in a more generalized way.

In this thesis we have assumed that a particular user is using only one application and only that application needs some minimum data rate to perform at a particular time instance. Now, if we think the scenario that a particular user is using different applications and each application is producing data that needs to be transmitted and each application needs different data rates, the problem gets more generalized and complicated. So, what could be a possible area to look at in future is, as I call it, *Heterogeneous Frame Aggregation Models*.

We have to study how combining heterogeneous subframes are affecting the throughput of one user. Not only that, what should be the ratio of the different frames, that will also be a big question. And depending upon environmental situation that ratio can also change. Whenever a new application starts sending data or an application stops sending data, how to cope up with those situations will also be a very interesting topic to look at future.

Bibliography

[1] Zakhia Abichar and J Morris Chang. Group-based medium access control for ieee 802.11 n wireless lans. *IEEE Transactions on Mobile Computing*, 12(2):304–317, 2013.

[2] Giuseppe Bianchi. Performance analysis of the ieee 802.11 distributed coordination function. *IEEE Journal on selected areas in communications*, 18(3):535–547, 2000.

[3] Michell Burton. Channel overlap calculations for 802.11 b networks. *White Paper, Cirond Networks Inc*, 2002.

[4] Enhanced Wireless Consortium et al. Ht mac specification, 2006.

[5] Lara Deek, Eduard Garcia-Villegas, Elizabeth Belding, Sung-Ju Lee, and Kevin Almeroth. Intelligent channel bonding in 802.11 n wlans. *IEEE Transactions on Mobile Computing*, 13(6):1242–1255, 2014.

[6] Yong Ding, Yi Huang, Guokai Zeng, and Li Xiao. Using partially overlapping channels to improve throughput in wireless mesh networks. *IEEE Transactions on Mobile Computing*, 11(11):1720–1733, 2012.

[7] Daquan Feng, Lu Lu, Yi Yuan-Wu, Geoffrey Ye Li, Gang Feng, and Shaoqian Li. Device-to-device communications underlaying cellular networks. *IEEE Transactions on Communications*, 61(8):3541–3551, 2013.

[8] Mohammad A Hoque, Xiaoyan Hong, and Farhana Afroz. Multiple radio channel assignement utilizing partially overlapped channels. In *Global Telecommunications Conference, 2009. GLOBECOM 2009. IEEE*, pages 1–7. IEEE, 2009.

[9] Lei Jiao, Vicent Pla, and Frank Y Li. Analysis on channel bonding/aggregation for multi-channel cognitive radio networks. In *Wireless Conference (EW), 2010 European*, pages 468–474. IEEE, 2010.

[10] Youngsoo Kim, Sunghyun Choi, Kyunghun Jang, and Hyosun Hwang. Throughput enhancement of ieee 802.11 wlan via frame aggregation. In *Vehicular technology conference, 2004. VTC2004-Fall. 2004 IEEE 60th*, volume 4, pages 3030–3034. IEEE, 2004.

[11] Jyoti Kolap, Shoba Krishnan, and Ninad Shaha. Frame aggregation mechanism for high-throughput 802.11 n wlans. *International Journal of Wireless & Mobile Networks*, 4(3):141, 2012.

[12] James F Kurose and Keith W Ross. *Computer Networking : A Top-Down Approach*. Pearson, sixth edition, 2013.

[13] Yuxia Lin and Vincent WS Wong. Wsn01-1: frame aggregation and optimal frame size adaptation for ieee 802.11 n wlans. In *Global telecommunications conference, 2006. GLOBECOM'06. IEEE*, pages 1–6. IEEE, 2006.

[14] Josip Lorincz and Dinko Begusic. Physical layer analysis of emerging ieee 802.11 n wlan standard. In *Advanced Communication Technology, 2006. ICACT 2006. The 8th International Conference*, volume 1, pages 6–pp. IEEE, 2006.

[15] Arunesh Mishra, Suman Banerjee, and William Arbaugh. Weighted coloring based channel assignment for wlans. *ACM SIGMOBILE Mobile Computing and Communications Review*, 9(3):19–31, 2005.

[16] SA Mujtaba. Ieee p802. 11âĂŤwireless lans, tgn sync proposal technical specification. *The Institute of Electrical and Electronics Engineers, Inc., doc.: IEEE*, 802(04):1–131, 2004.

[17] Sreetama Mukherjee and Sasthi C Ghosh. Throughput improvement using partially overlapping channels in wlan with heterogeneous clients. In *International Conference on Wired/Wireless Internet Communication*, pages 335–347. Springer, 2016.

[18] Minyoung Park. Ieee 802.11 ac: Dynamic bandwidth channel access. In *Communications (ICC), 2011 IEEE International Conference on*, pages 1–5. IEEE, 2011.

[19] Kritika Sengar, Nishu Rani, Ankita Singhal, Dolly Sharma, Seema Verma, and Tanya Singh. Study and capacity evaluation of siso, miso and mimo rf wireless communication systems. *arXiv preprint arXiv:1403.7774*, 2014.

[20] Rishad Ahmed Shafik, Md Shahriar Rahman, and AHM Razibul Islam. On the extended relationships among evm, ber and snr as performance metrics. In *Electrical and Computer Engineering, 2006. ICECE'06. International Conference on*, pages 408–411. IEEE, 2006.

[21] Cheolkyu Shin, Hyuncheol Park, and Hyuck M Kwon. Phy-supported frame aggregation for wireless local area networks. *IEEE Transactions on Mobile Computing*, 13(10):2369–2381, 2014.

[22] Andrew S Tanenbaum and David J Wetherall. *Computer Networks*. Pearson, fifth edition, 2012.

[23] Jun Yin, Xiaodong Wang, and Dharma P Agrawal. Optimal packet size in error-prone channel for ieee 802.11 distributed coordination function. In *Wireless Communications and Networking Conference, 2004. WCNC. 2004 IEEE*, volume 3, pages 1654–1659. IEEE, 2004.

[24] Kunxiao Zhou, Xiaohua Jia, Yanan Chang, and Xing Tang. Partially overlapping channel assignment for wlans using sinr interference model. *International Journal of Communication Systems*, 27(11):3082–3095, 2014.

[25] Hua Zhu, Ming Li, Imrich Chlamtac, and Balakrishnan Prabhakaran. A survey of quality of service in ieee 802.11 networks. *IEEE Wireless Communications*, 11(4):6–14, 2004.