IMPLEMENTATION OF CSMA/CA BASIC ACCESS in SOFTWARE DEFINED RADIO

A Project Report

submitted by

SOURAV CHATTERJEE

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THESIS CERTIFICATE

This is to certify that the thesis titled **IMPLEMENTATION OF CSMA/CA BASIC ACCESS in SOFTWARE DEFINED RADIO**,submitted by
Sourav Chatterjee(EE12M016), to the Indian Institute of Technology, Madras, for the award of the degree of **Master Of Technology**, is a bonafide record of the research work done by him under our supervision. The contents of this thesis, in full or in parts, have not been submitted to any other Institute or University for the award of any degree or diploma.

Prof. 1
Dr. Radhakrishna Ganti
Assistant Professor
Dept. of Electrical Engineering
IIT-Madras, Chennai-600 036

Place: Chennai

Date: 16th June 2014

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The place i loved the the most during the past 1 year was my lab. The lab is ideal for work and i would like to thank all my labmates specially Arjun,Vipul,Reshma and Praveen for the help they provided in my project. I will specially thank Praveen for providing the receiver code without which my project was at standstill at one point.

ABSTRACT

KEYWORDS: Software Defined Radio;USRP;CSMA;OFDM.

Software Defined Radio is a communications platform in which all the parts of a Communication System can be realised with the help of Software. It consists of two parts, a Universal Software Radio Peripheral(USRP) and a Computer. The Computer performs all the signal processing steps required for a particular application. The processed data is sent to a RF Frontend which converts it into passband analog signal for transmission.

USRP is the frontend used for realizing Software Defined Radio. It consists of an RF passband to baseband(and vice-versa) converter circuit and a digital signal processor for sampling the baseband signal before sending it to the computer for application specific processing.Every parameter of the USRP can be set the Computer via USB or LAN cable which is also used for sending and receiving samples.

Carrier Sense Multiple Access(CSMA) is a protocol for allowing multiple nodes in a network to communicate by sharing a common channel.The protocol defines a channel access mechanism by which multiple transmitters can operate with minimum collision between the packets in the channel.

The focus of this implementation is CSMA with Collision Avoidance(CSMA/CA). In this protocol the stations are allowed to sense the channel prior to transmission. Based on channel sensing they either transmit in that slot or defer the transmission to a later slot.

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ABBREVIATIONS

CSMA/CD	Carrier Sense Multiple Access with Collision Detection
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
IEEE	Institute of Electrical and Electronics Engineers
USRP	Universal Software Radio Peripheral
OFDM	Orthogonal Frequency Division Multiplexing
STA	Wireless Station
BSS	Basic Service Set
ESS	Extended Service Set
AP	Access Point
DS	Distributed System
MIMO	Multiple Input Multiple Output
MU-MIMO	Multiuser MIMO
DCF	Distributed Coordination Function
CW	Contention Window
SIFS	Short Inter Frame Space
DIFS	Distributed Inter Frame Space
RTS	Request To Send
CTS	Clear To Send
ACK	Acknowledgement
ICI	Inter Carrier Interference
CS	Channel Sensing
NAV	Network Allocation Vector

CHAPTER 1

INTRODUCTION

IEEE 802.11 standard is a set of MAC and PHY Layer specifications defined by that allows mobile stations to function in a coordinated fashion. They are created and maintained by the IEEE LAN/MAN Standards Committee (IEEE 802). It consists of a series of half-duplex over-the-air modulation techniques that operate in the 2.4, 3.6, 5 and 60 GHz frequency bands. 802.11-1997 was the first wireless networking standard in the family, but 802.11b was the first widely accepted one, followed by 802.11a, 802.11g, 802.11n and 802.11ac. Other standards in the family (c−f, h, j) are service amendments and extensions or corrections to the previous specifications.

802.11 standards and ammendments:

1. **802.11-1997 (802.11 legacy)**- This specified three alternative physical layer technologies: diffuse infrared operating at 1 Mbit/s; FHSS operating at 1 Mbit/s or 2 Mbit/s; and DSSS operating at 1 Mbit/s or 2 Mbit/s. It used microwave transmission over the ISM band at 2.4 GHz.

2. **IEEE 802.11a**- The 802.11a standard uses the same data link layer protocol and frame format as the original standard, but an OFDM based air interface (physical layer). It operates in the 5 GHz band with a maximum net data rate of 54 Mbit/s, plus error correction code.

   Due to the use of 5GHz band centered at 5.8GHz the interference from other devices is less than that of the 2.4GHz band. But due to increase in frequency attenuation is more and hence the system has a short range of operation.

3. **IEEE 802.11b**- This uses the same media access method defined in the original standard.It has a maximum data rate of 11Mb/s. Due to the sufficient increase in throughput in the 2.4GHz band this is the preffered more than the original standard.

4. **IEEE 802.11g**- This uses OFDM as transmission scheme,same as 802.11a and works in the 2.4GHz band. It offers a maximum data rate of 54Mb/s with FEC thus resulting in average data rate of 22Mb/s.Like 802.11b, 802.11g devices suffer interference from other products operating in the 2.4 GHz band.

5. **IEEE 802.11n**- This is an ammendment to the original 802.11a which adds support for MIMO. It operates in both 2.4GHz and 5GHz bands and provides a maximum net data rate from 54 Mbit/s to 600 Mbit/s.

6. **IEEE 802.11ac**- This is an ammendment to the 802.11-2012 and builds on 802.11n. It includes support for wider channels(40MHz,80MHz and 160MHz) as opposed to 40MHz for 802.11n in the 5GHz band.The spatial streams for MIMO has been increased from 4 to 8. and MU-MIMO is supported. The modulation scheme used is same as 802.11n with support for hihger constellations(up to 256-QAM).This yielded a data rate of up to 433.3 Mbit/s per spatial stream, 1300 Mbit/s total, in 80 MHz channels in the 5 GHz band.

7. **IEEE 802.11ad(future ammendment)**- This ammendment is expected to define a new physical layer at 60GHz band and offer a transmission rate in the range of Gb/s.

1.1 Components of 802.11 architecture

1.1.1 Wireless Station(STA)

A STA is the smallest addressable unit in the architecture. Physical and operational characteristics are defined by modifiers that are placed in front of the term STA. For example, in the case of location and mobility, the addressable units are the fixed STA, the portable STA, and the mobile STA. A STA might take on multiple distinct characteristics, each of which shape its function. For example, a single addressable unit might simultaneously be a portable STA, a quality-of-service (QoS) STA, a dependent STA, and a hidden STA.

1.1.2 Basic Service Set

The BSS is the basic building block of 802.11 architecture. It is a collection of STA's situated close to each other so that they can connect either directly or via an Access Point(AP). BSS may or may not refer to the area where the STA's belonging to it are located but it is often a good assumption to refer BSS as a Basic Service Area(BSA). There are 2 types of BSS:

I) **Infrastructure BSS**-In this type of BSS there is an AP associated with each BSS. A STA trying to communicate with another STA has to send its packets first to the AP. The AP then forwards the packets to the destination which may be in the same BSS or different BSS. For this type, association,joining and authentication requests has to be made by the STA prior to packet transmission.

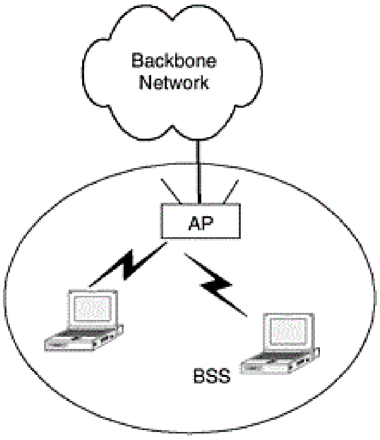

Figure 1.1: An infrastructure BSS

II) **Independent BSS**- In this type of BSS there is no AP. The STA's can directly communicate with each other. This network is temporary and requires no association request. This is often referred to as *ad-hoc network.*

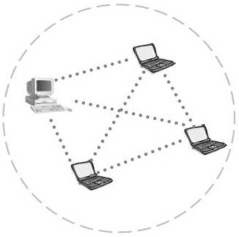

Figure 1.2: An independent BSS or ad-hoc network

1.1.3 Extended Service Set(ESS)

An ESS is a set of individual infrastructure BSS which operates collectively as a single network. An ESS is required because sometimes the BSS cannot include some STA's in their coverage area due to distance limitations. The APs in a ESS perform the communication via an abstract medium called the distribution system (DS). To network equipment outside of the ESS, the ESS and all of its mobile stations appears to be a single MAC-layer network where all stations are physically stationary. Thus, the ESS hides the mobility of the mobile stations from everything outside the ESS.

The BSS's present in a ESS may be physically disjoint or overlapping to provide a large, continuous coverage area.

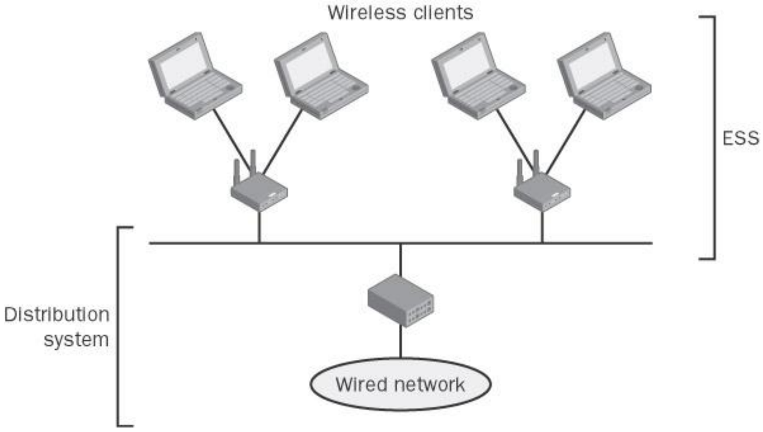

Figure 1.3: An Extended Service Set

1.1.4 Distributed System(DS)

A DS is a system which helps to connect different wireless AP's of different BSS's to function as a single network visible to the outside world.All AP's in a DS must be configured to use the same radio channel, method of encryption (none, WEP, or WPA) and the same encryption keys. They may be configured to different service set identifiers. DS also requires every base station to be configured to forward to others in the system. However DS may suffer from a disadvantage which is decrease in throughput due to multiple hopping of packets from one AP to another.

CHAPTER 2

Carrier Sense Multiple Access(CSMA)

2.1 Introduction

CSMA is a random access protocol which is defined in the MAC layer. This protocol allows simultaneous packet transmissions in a network with multiple nodes with a target of achieving minimum packet collision probability. The main concept behind this is channel sensing which allows the stations to maximise throughput by either detecting the collisions or avoiding it by detecting other incoming packet transmissions before transmitting its own packet.Based on this,there are two types of CSMA protocols

a) **CSMA with Collision Detection(CSMA/CD)**- In this scheme the station transmits after a random period of time and senses the channel after transmission for any possible collision. If it detects collision it retransmits the same packet.

b) **CSMA with Collision Avoidance(CSMA/CA)**- In this scheme the station senses the channel prior to transmission and if the channel is free at the time of transmission then only the packet is sent. By this way, collision probability is reduced as collision more or less avoided.

2.1.1 CSMA/CD

The flowchart for CSMA/CD protocol is given below,

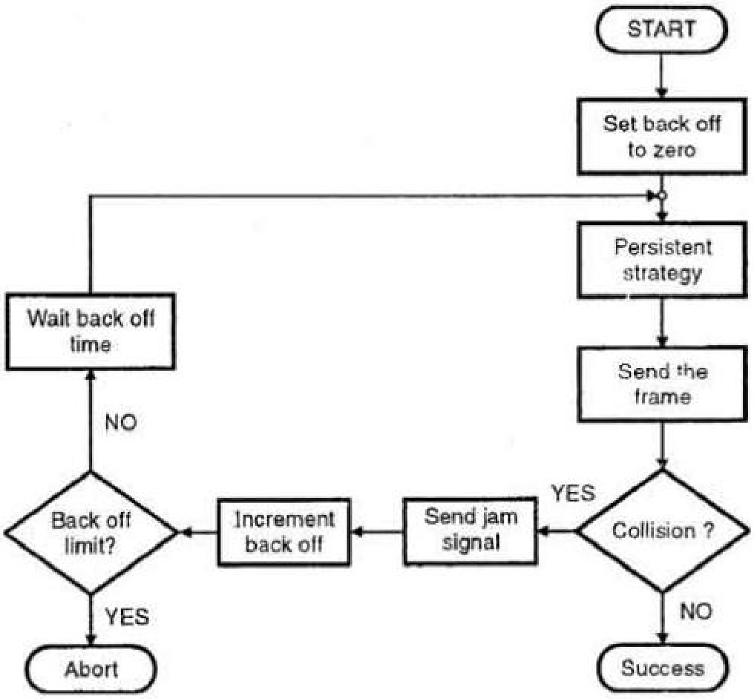

Figure 2.1: Flowchart for CSMA/CD protocol

As seen from figure 2.1 the main difference of CSMA/CD from CSMA is that in CSMA/CD channel is continued to be sensed even after transmission. So, whether it is 1-persistent or p-persistent, the station first senses transmits the packet with a zero backoff counter. If collision is detected by channel sensing, then a jam signal is sent by the corresponding transmitter. This allows all the transmitters to know of the collision. Upon reception of the jamming signal all the stations increment their backoff counters so that further collision does not occur. If no collision is detected then the backoff counter remains at zero for the next transmission.

2.1.2 CSMA/CA

Distributed Coordination Function(DCF)

The CSMA/CA protocol described in 802.11 standard(1) is called DCF. By 'Distributed' we mean that the control is not centralized and is distributed over the transmitters in the network. So, each transmitter behaves independently and follows the same protocol. The DCF allows a mechanism to be followed by every transmitter to avoid collision. The mechanism is known as contention. By contention we mean that within a particular duration all the transmitters compete for the channel and whichever has shortest contention window(CW) wins and gets an opportunity to transmit. All the other transmitters remain idle for a period of time and again go into contention after the ongoing transmission is over. The DCF defines two types of access, basic access and RTS/CTS based access.

A) **Basic Access**- The Basic Access is the core mechanism of DCF.For DCF the time is divided into slots, with every slot being either busy,idle or in collision. There are mainly two types of idle times present in DCF Basic Access. The other idle times shown in figure 2.2 are used in Point Coordination Function(PCF) and Hybrid Coordination Function(HCF). The two idle times are,

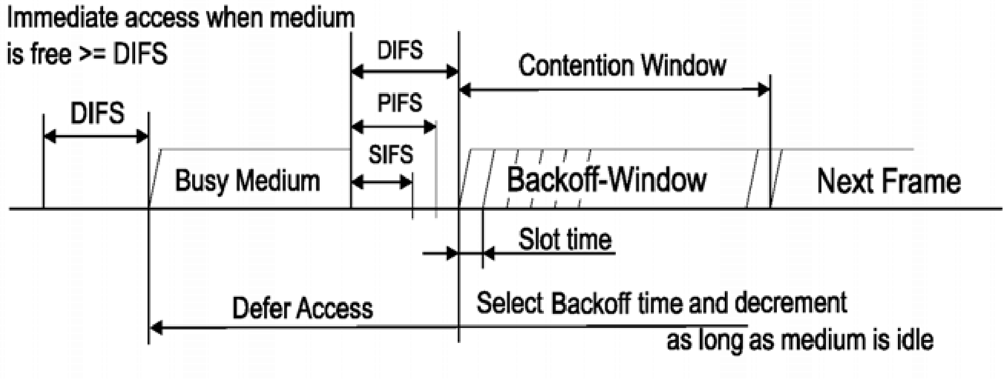

Figure 2.2: DCF Basic Access Procedure

i) **Short Inter Frame Space(SIFS)**-SIFS is the gap between the transmission of a Data Frame and reception of an Acknowledgement Frame.It is measured from the point of the last symbol of the transmitted frame to the first symbol of the preamble of the received frame as seen from the air interface. The receiver waits this amount of time after reception of Data Frame before it transmits an ACK Frame. The transmitter also starts its `ACK_Timeout` counter after this time from transmitting the Data Frame.SIFS is the shortest of the IFS's between transmissions from different STA's. SIFS is used to maintain continuity between frame transmission and reception of acknowledgement. It is kept the the smallest so that other stations which are idle do not transmit in between the transmission of packet and reception of ACK frame.

ii) **Distributed Inter Frame Space(DIFS)**-DIFS is the gap between end of a successful transmission period and begining of contention period. DIFS is used so that there is a gap between the transmitted frames so that the receiver can process the previously transmitted Data Frame. It is generally ,

$$DIFS = SIFS + 2 * \texttt{Slot\_Time} \tag{2.1}$$

The DCF starts with the backoff procedure immediately after the DIFS. The contention period is divided into time slots with each slot having length `Slot_Time`. At the beginning of each slot the STA will perform Channel Sensing(CS) mechanism. The contention period has a duration which is chosen from a Random Backoff interval. At first the `Backoff_Counter` is started at the beginning of the contention period. For every idle slot the `Backoff_Counter` is decremented by 1. When the `Backoff_Counter` reaches 0 the packet is transmitted. If the CS mechanism returns 'busy' for any slot, the `Backoff_Counter` is freezed at that point and the STA goes into idle state. The `Backoff_Counter` is again started from the freezing point in the next contention period after DIFS.

Backoff Procedure- The `Backoff_Counter` is chosen as a random integer between [0,CW] where CW=Contention Window. The CW is an integer which is given as,

$$CW = 2^m CW_{min} - 1 \tag{2.2}$$

where m is the backoff stage.The CW varies between CW_{min} and CW_{max} where the values of CW_{min},CW_{max} are defined in the 802.11 Standard for different Physical Mediums.

After transmitting at the end of `Backoff_Counter` the transmitter waits for SIFS amount of time before checking for ACK Packets. If no ACK Packet is found within a `ACK_Timeout` then it is assumed that there is a collision and packet is lost. So, in the next contention, the CW goes to the next stage $i.e.m$ increases by 1. If ACK is received then the packet transmission is successful and CW is made as $CW_{min} - 1$. A comprehensive view of the Data and ACK packets transfer is shown below,

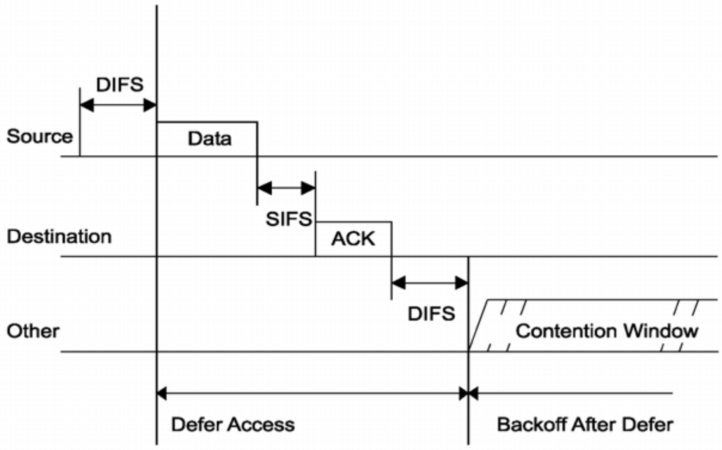

Figure 2.3: DCF Basic Access Procedure full process

B) **RTS/CTS Based Access-** This access procedure is advantageous than basic access because of the lower time the channel remains idle in case of detection of collision. The procedure is shown below, In this, the transmitter first sends a RTS(Request To Send) Frame instead of directly the Data Frame just after backoff window. The RTS Frame is devoid of any data and contains a special header.If the RTS Frame is correctly received, then the receiver will send a CTS(Clear To Send) Frame after waiting for SIFS amount of time. If the transmitter does not receive the CTS Frame correctly, it will assume that the channel is busy and will defer its Data Frame transmission to the next contention period. If the CTS Frame is received correctly then after waiting for SIFS time the transmitter sends the Data Frame. The rest of the procedure is similar to basic access.

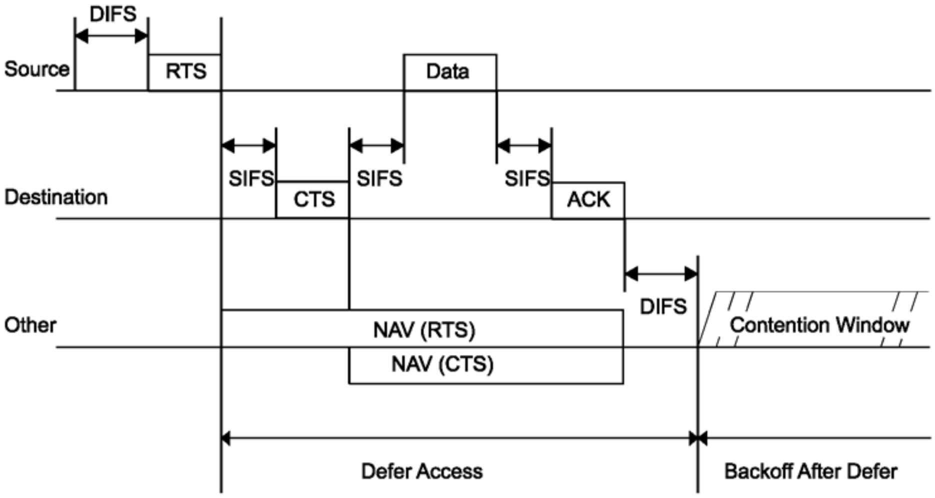

Figure 2.4: DCF RTS/CTS based Access Procedure

Advantage of RTS/CTS based access

The major advantage of RTS/CTS procedure is to avoid the Hidden Node problem. The Hidden Node problem can be shown from the figure below. Suppose there are 3 nodes A,B and C with B being the Access Point.The transmission ranges of the nodes are such

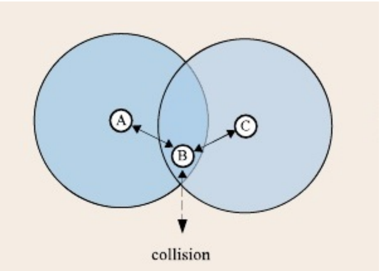

Figure 2.5: Hidden Node problem

that nodes A and C cannot sense the transmission of each other or are 'hidden' from each other. But B can receive packets from both A and C since it falls in the transmission range of both. So A and C can simultaneously send packets because they are unaware of each other and can sense the channel idle. But at B there will be a collision. This is the Hidden Node problem. Now, if any of A or C sends an RTS packet first then B will know the impending transmission from that node and hence can reserve the channel for that node by not accepting frames from other transmitters. By not accepting it means B will not send the CTS packet for all other RTS packets.

2.2 Performance Analysis of CSMA/CA

In this project a two transmitter, one receiver model has been implemented with Basic Access protocol. The main performance measure of CSMA/CA is the saturation throughput. Saturation throughput is defined as the throughput under Saturated Conditions means there is always a packet available to transmit at the input buffer of every transmitter. The assumption has been made according to the work done by Bianchi(2). In this paper the input data is assumed to have a poisson arrival process with some arrival rate. Accordingly, at any point of time the probability that a packet is present for transmission is given by a Bernoulli Distribution.

2.2.1 Transmission Probability

At any slot when the transmitter gets senses the channel idle and has a `Backoff_Counter` value 0, it transmits with a probability τ. This is a close approximation to the model proposed in the paper having a poisson arrival process. The transmission probability is given as,

$$\tau = 1 - (1-p)^{1/n-1} \tag{2.3}$$

where p is the unconditional probability that a packet transmitted in air at any slot will collide and n is the number of stations in the network. This comes from the simple theory that there will be a collision if atleast one node transmits. So p is given as,

$$p = 1 - (1-\tau)^{n-1} \tag{2.4}$$

from which Eq. 2.3 comes.

2.2.2 Throughput

Throughput of the system is defined as the fraction of time the channel is used to successfully transmit a payload or Frame. The expression for throughput is given as,

$$Throughput = S = \frac{P_s P_{tr} E[P]}{P_s P_{tr} T_s + (1 - P_{tr})\sigma + P_{tr}(1 - P_s)T_c} \tag{2.5}$$

The numerator of the above expression denotes the amount of payload transmitted in a slot during a successful transmission. The term $P_s P_{tr}$ denotes probability of successful transmission and $E[P]$ denotes the expected payload length. For constant packet size as assumed in this project $E[P]$=packet size(in bytes)(assumed here). The denominator is the average time of a slot.

The quantities in the equation are expressed as,

$$P_{tr} = 1 - (1 - \tau)^{n-1} \tag{2.6}$$

$$P_s = \frac{n\tau(1 - \tau)^{n-1}}{P_{tr}} \tag{2.7}$$

For Basic Access scheme,

$$T_s = \texttt{Packet\_Header} + \texttt{Payload\_Length} + SIFS + \texttt{ACK\_Length} + DIFS \tag{2.8}$$

$$T_c = \texttt{Packet\_Header} + \texttt{Payload\_Length} + DIFS \tag{2.9}$$

For RTS/CTS based Access scheme,

$$T_s = \texttt{RTS\_Length} + SIFS + \texttt{CTS\_Length} + SIFS + \texttt{Packet\_Header} + \texttt{Payload\_Length} + L \tag{2.10}$$

$$T_c = \texttt{RTS\_Length} + DIFS \tag{2.11}$$

where

RTS_Length=Time taken by RTS Packet to travel through the buffer and air-interface

CTS_Length=Time taken by CTS Packet to travel through the buffer and air-interface

Payload_Length+Packet_Header=Time taken by Data Packet to travel through the buffer and air-interface

ACK_Length=Time taken by ACK Packet to travel through the buffer and air-interface

The denominator of Eq. 2.5 can be explained in the following way. Every slot is either idle which occurs during the backoff period with Backoff_Counter not equal to zero. The probability of the slot being idle is that none of the transmitters transmit which is equal to $1 - P_{tr}$. When a transmitter gets an idle slot with zero Backoff_Counter, it transmits. The slot contains a successful transmission when a Data Frame is transmitted properly and ACK Frame is received properly by the transmitter.This happens only when there is only 1 transmission in that slot. The probability of that happening is the unconditional probability of having only 1 transmitter transmitting which is given by $P_s P_{tr}$. Finally, there is collision if more than 1 transmitter transmits in a slot. This occurs with a probability that atleast 1 transmitter transmits and more than 1 transmitter transmits. This is equal to $P_{tr}(1 - P_s)$.

CHAPTER 3

Basics of OFDM

3.1 Introduction

Orthogonal Frequency Division Multiplexing can be used both as modulation method or as a multiplexing method. OFDM is a type of frequency division multiplexing in which data is transmitted in different subcarriers or pulses having different carrier frequencies which are orthogonal to each other. A simple illustration showing difference between OFDM and normal Frequency Division Multiplexing is given below, For normal FDM system the symbols are modulated with subcarriers which are spaced far apart so that at the receiver they can be isolated with BPF's. But in OFDM the subcarriers overlap so that overall bandwidth reduces in order to make it more bandwidth efficient. So in order to separate the contents of the subcarriers, they have to be orthogonal so that a simple correlation detector is sufficient.

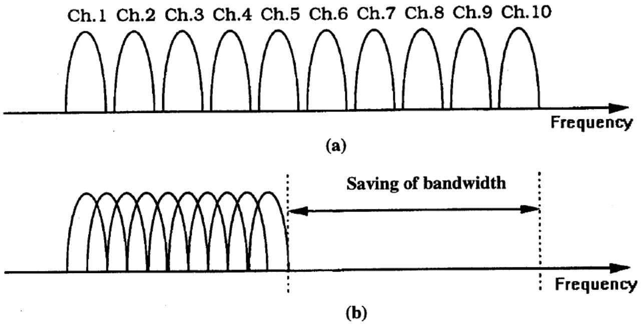

Figure 3.1: Comparison of OFDM vs normal FDM

3.2 Structure of OFDM signal

The general time domain representation for an OFDM spread over a time **T**and starting at $t = t_s$ is given as,

$$s(t) = \sum_{i=-\frac{N_s}{2}}^{\frac{N_s}{2}-1} d_{i+N_s/2} exp(j2\pi\frac{i}{T}(t - t_s)) \quad , \, t_s \leq t \leq t_s + T \tag{3.1}$$

$$s(t) = 0 \, , \quad t < t_s \, and \, t > t_s + T \tag{3.2}$$

where,

$d_{i+N_s/2}$=Complex frequency domain constellation symbols(PSK or QAM)

N_s=Number of subcarriers

t_s=Starting time of the OFDM symbol

$exp(j2\pi\frac{i}{T}(t - t_s))$is the complex $i + N_s/2$ 'th subcarrier. These are the basis functions. In the demodulator, in order to decode $d_{i+N_s/2}$ we perform correlation with the $i+N_s/2$ 'th basis function. If we pass $s(t)$ through the correlator,

$$\int_{t_s}^{t_s+T} exp(-j2\pi\frac{j}{T}(t - t_s)) \sum_{i=-\frac{N_s}{2}}^{\frac{N_s}{2}-1} d_{i+N_s/2} exp(j2\pi\frac{i}{T}(t - t_s))dt$$

$$=\sum_{i=-\frac{N_s}{2}} \frac{N_s}{2} - 1 d_{i+N_s/2} \int_{t_s}^{t_s+T} exp(j2\pi\frac{i-j}{T}(t - t_s))dt$$

$$=d_{j+N_s/2}T$$

This happens because the term within the integral is non-zero only when $i = j$. In other cases there is a complex sinusoid with frequency $\frac{i-j}{T}$ which contains integral number of full cycles within the interval T which when integrated turns out to be 0.So, only one term of the N_s terms is non-zero. This is the result of orthogonality between the subcarriers.

3.3 Practical Implementation of OFDM

Practically, the basis functions bring forward the analogy of DFT. Since, Eq.3.1 is similar to IDFT operation, a direct N_s point IDFT can be performed on the parallel data coming from the constellations. The transmitted nth OFDM symbol is obtained as,

$$s(n) = \sum_{i=0}^{N_s-1} d_i exp(j2\pi\frac{in}{N_s}) \tag{3.3}$$

At the receiver , DFT operation is performed to extract the ith symbol

$$d(i) = \frac{1}{N_s}\sum_{n=0}^{N_s-1} exp(-j2\pi\frac{in}{N_s}) \tag{3.4}$$

The block diagram for the OFDM system based on this is given below,

3.4 OFDM Guard Time and Cyclic Extension

In OFDM there are two types of interference-Inter Symbol Interference(ISI) and Inter Carrier Interference(ICI). As we know, OFDM is quite robust to multipath delays. This robustness comes due to addition of a Guard Interval before each symbol. However the Guard Interval should have certain properties. For example, if it contains just zeros then due to multipath delay, different subcarriers will be delayed by different amounts and in the receiver correlator of Section 3.2 there will not be integral number of cycles within T so for every j so there will be non-zero contribution from all the subcarriers. This is called ICI. It is illustrated below for 2 subcarriers, Now the Guard Time is kept as a cyclic extension of the OFDM symbol,i.e a part of the OFDM symbol from the end repeated in the front, this will effectively mean that a part, from the end of every subcarrier is repeated in the front.By doing this we observe that even if multipath delay is there, if that delay is less than the part used as Guard Time, then at the receiver, still all the subcarriers will be orthogonal and the output will be contributed by only one subcarriers. This happens because the part used as Guard Time is a cyclic extension of every subcarrier added together and this maintains the fact that integral number of

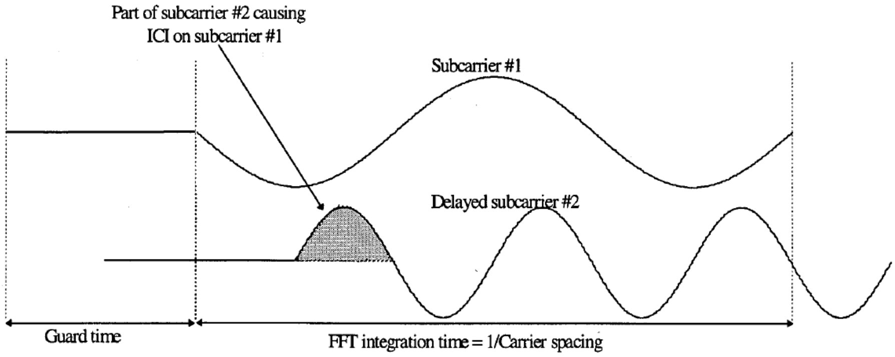
Part of subcarrier #2 causing ICI on subcarrier #1
Subcarrier #1
Delayed subcarrier #2
Guard time
FFT integration time = 1/Carrier spacing

Figure 3.2: Inter Carrier Interference due to multipath

cycles occur in the cross-terms($i \neq j$) of Eq.3.1. This is called cyclic prefix and is shown as,

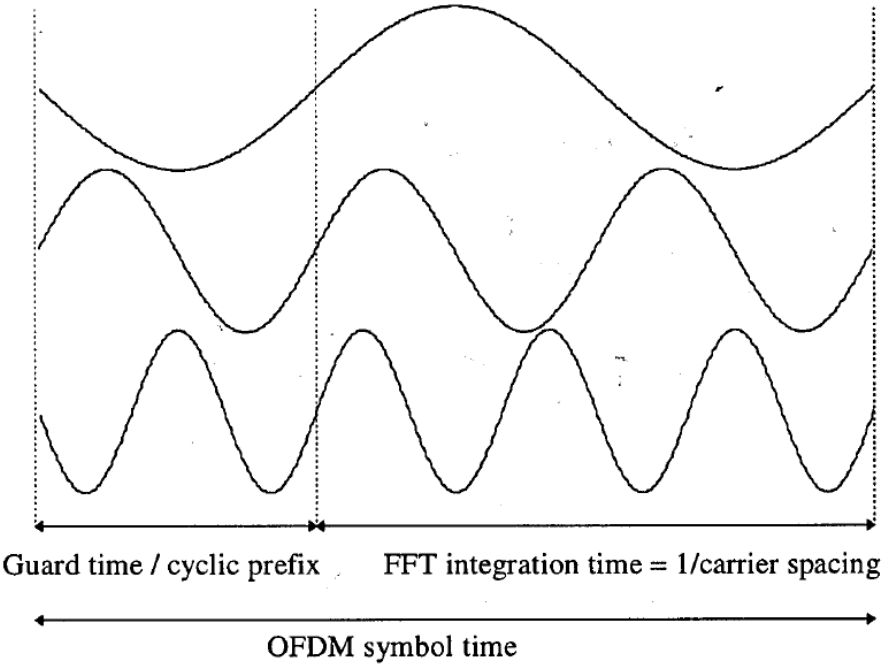
Guard time / cyclic prefix
FFT integration time = 1/carrier spacing
OFDM symbol time

Figure 3.3: Cyclic Prefix in OFDM Symbol

The block diagram of the practical OFDM transmitter is given below.

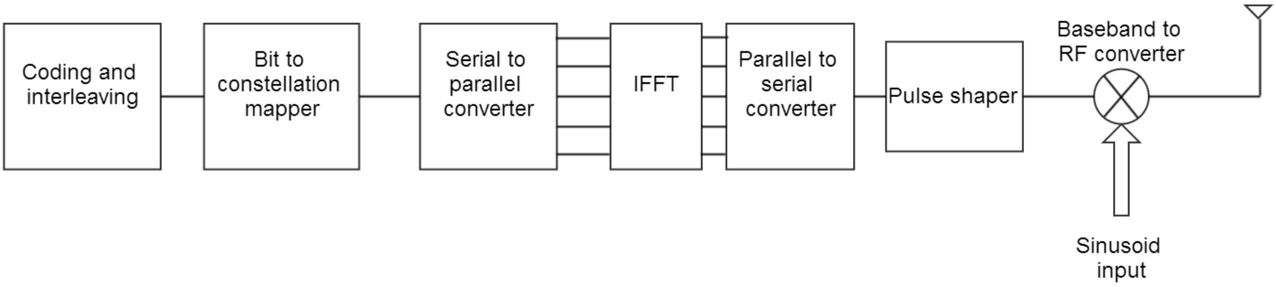
Coding and interleaving
Bit to constellation mapper
Serial to parallel converter
IFFT
Parallel to serial converter
Pulse shaper
Baseband to RF converter
Sinusoid input

Figure 3.4: OFDM Transmitter

CHAPTER 4

Channel Sensing

4.1 Introduction

In CSMA the channel sensing(CS) mechanism is used to detect the presence of any frames transmitted by other transmitters in a particular slot. There are two types of CS mechanism involved:-

1. **Physical CS**-This CS mechanism involves sensing the channel for a change in its properties.For example methods like energy detection can be used to detect change in energy for a possible detection of Frames. Other methods like feature based detectors are also used.
2. **Virtual CS**-This is used along with physical CS to prevebt the stations from transmitting. The packets contain a counter called NAV which is a measure of their duration. When a station senses this NAV in the packet, it goes into idle for these many (NAV) number of slots. The NAV is set during the detection of the packet and is subsequently decremented for each slot.

4.2 Preamble based detection used for Channel Sensing

In this project a detection scheme based on the preamble is used to detect the presence of Data Frames and ACK Frames in Basic Access. The method is actually a modified version of the Schmidl-Cox method for timing synchronization which is used here in a modified application. The preamble for this system is shown as,

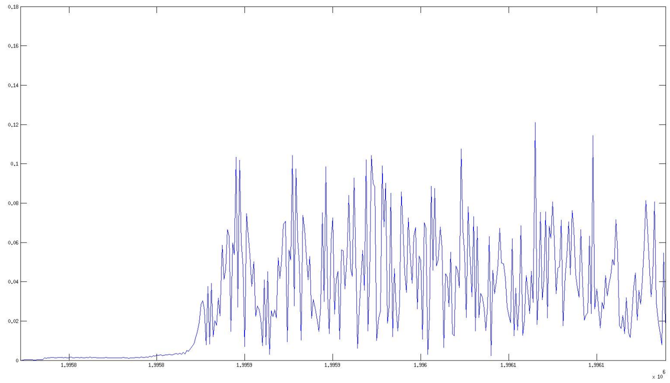

Figure 4.1: Preamble used in this project

We see that the preamble, which is the first part of the Frame received as shown above consists of two identical parts. This is the feature used for finding out the start of the preamble. At first a window of size `PREAMBLE_LENGTH` is slided over the received data whose size is kept as `2*OFDM_FRAME_SIZE`.

Let `leftavg` and `rightavg` be the average value of the left and the right halves of the preamble. Though the halves are identical, `leftavg` and `rightavg` may be different after the preamble is passed though the channel. The idea is to perform correlation explicitly between two shifted samples, so that when the shifted samples are identical because of repetition the correlation maximizes. This happens only at the start of the preamble,so the location of the beginning of the Frame can be found. The window mentioned before is slided through the received buffer and the correlation for each delay is calculated.The correlation is done for the samples after removing the DC component. The metric is given as,

$$M_1[k] = \frac{\sum_{i=1}^{L/2} |(X_{k+i} - leftavg)(X_{k+L/2+i} - rightavg)^*|}{\sqrt{\sum_{i=1}^{L/2} |X_{k+i} - leftavg|^2}\sqrt{\sum_{i=1}^{L/2} |X_{k+i} - rightavg|^2}} \tag{4.1}$$

where, L=`PREAMBLE_LENGTH`

k here varies from 0 to a number called `TIME_CORRELATION_LENGTH`. $M_1[k]$ is calculated for all values of k. For each $M_1[k]$ it is checked whether it lies between $1 - \delta$ and $1 + \delta$, where δ = `PACKET_THRESHOLD`. If it lies then the next step of fine correlation is done otherwise that k is discarded and the next $M_1[k]$ is calculated. The position k obtained from Eq.4.1 may or may not indicate the start of the preamble.

This is because, high value of noise and data itself which is not fully random can have slightly cyclic properties which causes false detection through the first step. In order to remove that ambiguity, the numerator of Eq.4.1 is compared with a threshold whether it is higher than that.

After the initial estimate from Eq.4.1, a fine autocorrelation is performed to ensure that the point of starting of the preamble and not something else. The algorithm above is run iteratively until the final value of k is found. The fine metric is given as,

$$M_2 = \frac{\sum_{j=1}^{L/2} P_j X_{j+L_1}}{sqrt \sum_{j=1}^{L/2} |X_j|^2 * (\textbf{PREAMBLE-AUTOCORRELATION})} \tag{4.2}$$

where, **PREAMBLE-AUTOCORRELATION**=autocorrelation value of half of the preamble.

After M_2 is calculated it is seen whether it is greater than a quantity `THRESHOLD_FINE`.If it is, then it is compared with the previous $M_1[k]$ (i.e. $M_1[k]$ for the previous value of k). If it is greater than that value then the search is even more refined.

The above algorithm is repeated for all values of k within [0,`TIME_CORRELATION_LENGTH`].Since only that k is accepted at which the above conditions of both the metrics are satisfied , the choice of k stops after a certain time and no newer k satisfies that.This is the final k obtained which indicates the start of the preamble.

CHAPTER 5

Implementation method

5.1 Parameters Used

The parameters used in this project are shown in the table,

$Parameter$	$Value$
OFDM-LENGTH	512
CYCLIC-PREFIX-LENGTH	32
`PREAMBLE_LENGTH`	64
CYCLIC-PREFIX-PREAMBLE-LENGTH	6
`OFDM_FRAME_SIZE`	4416
`PACKET_THRESHOLD`	0.2
`THRESHOLD_FINE`	0.3
`TIME_CORRELATION_LENGTH`	4416
POWER-PREAMBLE	2
PREAMBLE-AUTOCORRELATION	65536

Table 5.1: Parameters used in this project

In addition, the following parameters were used in USRP,

Tx_Gain=20dB

Rx_Gain=20dB

$Tx\_Center\_Frequency$=1.9GHz

$Rx\_Center\_Frequency$=1.9GHz

$Tx\_Bandwidth$=20MHz

$Rx\_Bandwidth$=20MHz

Constellation used=BPSK

$Tx\_Antennae$=TX/RX

$Rx\_Antennae$=RX2

5.2 Method

- The normal $OFDM\_RX.cpp$ program is modified to have both RX and TX capabilities. In the CSMA transmitter, it is required that the USRP continues to sense the channel by the scheme described in Section 4.2 always. There should also be a transmitter part which should implement all the delays and Frame creation involved. These two operations must run in parallel, so that channel sensing continues during idle period of the transmitter. To implement that we have used multithreading(5) where the receiver part has been kept in a thread using $'boost :: thread'$ object. The creation of threads has provided a certain degree of parallelism.

- Since for CSMA the USRP has to take the decision for transmission based on detection of Frames from the receiver, there has to be some form of communication between the threads. This is done by a simple control variable $packet\_found$ which is checked by the transmitter for 'True'. This variable is set by the receiver on detection of start of preamble.

- The packets are numbered by a Sequence Number which lies in [0,`SEQ_NO_MAX`. This is required for the generation of ACK packets and also to calculate throughput.

- Since there are multiple transmitters, the data sent by each transmitter has a unique sequence by which the receiver knows from which transmitter the packet is from.

- In order to calculate the quantities in Eq. 2.5 a variable *'slot_counter'* is kept to count the number of slots. The variable is incremented for every idle slot, every slot where packet is transmitted and every slot where packet is not transmitted on detection of busy channel. To calculate probability of success P_sP_{tr}, a counter *'success'* is kept which increments only when the packet is transmitted and an ACK Frame is received with the same Sequence Number as the transmitted Frame. For probability of collision, similarly a counter *Collision'* is kept which increments when no ACK is received for a transmitted packet. It is also incremented when the channel is sensed busy and a packet is ready to get transmitted. Lastly, for idle probability $1 - P_{tr}$ a counter *idle* is kept. An Idle slots are during backoff and when the transmitter does not have a packet at the time of transmission. The probabilities are given as,

$$P_sP_{tr} = \frac{success}{slot\_time}$$
$$1 - P_{tr} = \frac{idle}{slot\_time}$$
$$P_{tr}(1 - P_s) = \frac{collision}{slot\_time}$$

- T_s,T_c and T_i are calculated by calculating the average time the channel is busy during a successful transmission, collision and idle during backoff. In the expression 2.5 σ is the average length of slot time during backoff. Since that is constant ,it can take a constant value. The other times are calculated using 'std::chrono::high_resolution_clock' object(6) from std::chrono library. Time time duration is 'duration_cast' to 'std::chrono::milliseconds' objects to find time in milliseconds.

- Since the packets are sent with probability τ ,this is implemented using 'std::bernoulli::distribution' object and generating random numbers according to bernoulli distribution. These random quantities are either 'True' or 'False'. When 'True' packet is transmitted on channel idle. When 'False' packet is not transmitted even when channel is idle and that is counted as an idle slot.

- The estimated packet size is 4486 samples.So, the throughput calculated is not in proper units. The behaviour of throughput is important and not units.

CHAPTER 6

Results

The throughput behaviour of the system depends on several parameters. This includes throughput behaviour with applied load which indicates upto how much the system can be pushed to its limits. This is important because it helps to analyse the behaviour at various system load and also helps to calculate the maximum saturated throughput. Next, throughput also depends on the arrival rate of the poisson traffic. This Since the arrival rate depends on the transmission probability in every slot, throughput also depends on transmission probability. Throughput also depends heavily on the backoff procedure. The backoff indicates how much time the channel is kept idle and wasted for just sensing mechanism.The backoff is directly related to the Contention Window and so throughput depends on the Contention Window.

6.1 Throughput vs applied load

The plot of backoff vs applied rate is given below,

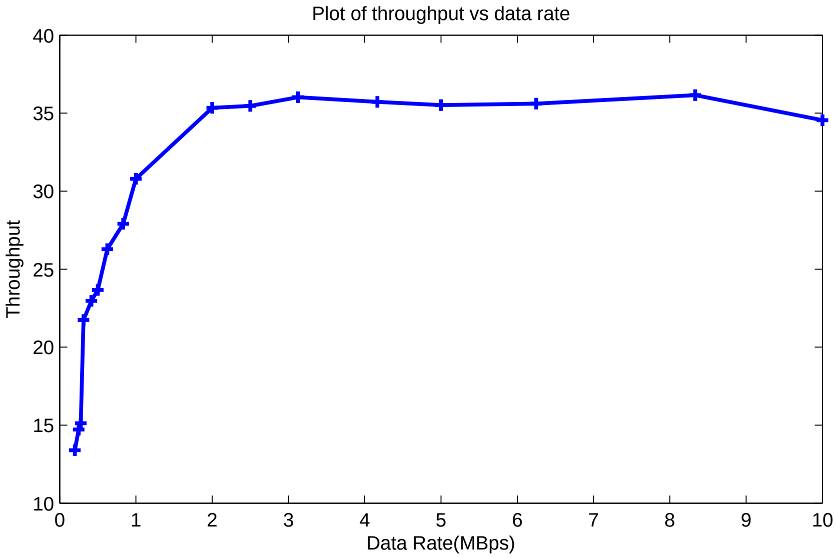

Figure 6.1: Variation of Throughput with applied load for 2 transmitters

The applied load here is related to the sampling rate of the USRP which is taken as the data rate through the channel. We see that Throughput increases at first with increase in applied load and after that it saturates. If we increase the load further then it starts to decrease indicating that the system has a maximum saturation throughput in which it can perform. If we increase the load beyond that it Throughput starts to decrease. Since this is for only 2 transmitters the behaviour is not that prominent. The result is more prominent for more transmitters.

6.2 Throughput vs Transmission Probability

The plot of Throughput with respect to transmission probability is given below, Transmission probability is a factor which models the arrival process for the traffic. The arrival rate of the poisson process is directly related to this probability. If the arrival rate increases, then the throughput increases at first an then becomes saturated. Here we see that the throughput increases with the transmitter probability and nearly goes into saturation. This is because if the transmission probability keeps on increasing there will be more collisions which will prevent the throughput from increasing further. If number of transmitters are increased it may also decrease.

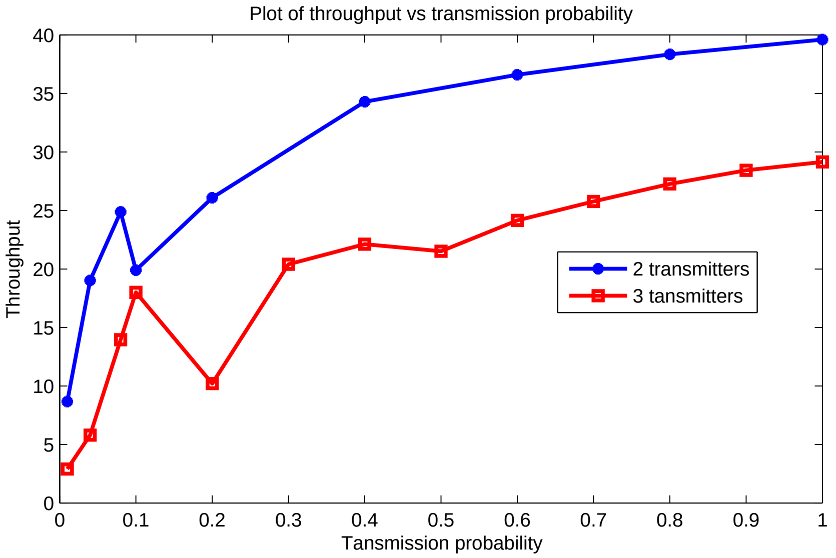

Figure 6.2: Variation of Throughput with transmission probability

6.3 Throughput vs Backoff

Since backoff is related to Contention Window ,the plot of Throughput vs minimum Contention Window shows the Throughput behaviour with respect to backoff.

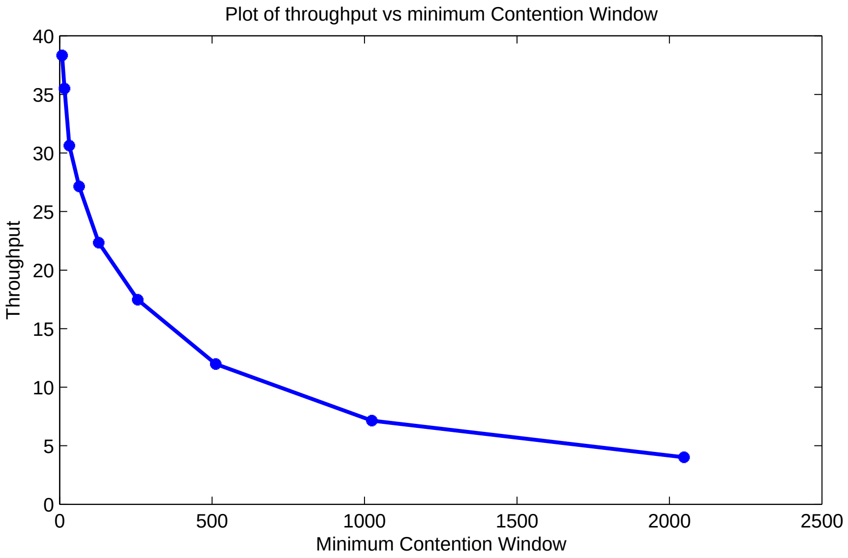

Figure 6.3: Variation of Throughput with minimum Contention Window for two transmitters

As the Contention Window increases the backoff durations also increase which results in more idle slots. So, probability of idle increases and this results in decrease in Throughput. But actually, with increase in backoff there will be less number of transmissions so a packet will have less chance of collision. Due to this backoff should increase at first and then start to decrease after reaching a maximum point due to the countereffect as discussed before. The graph shows a behaviour which is not fully correct.

From the above plots we see that the behaviour of the CSMA system depends upon many parameters like applied load, backoff scheme used, arrival process etc. All these things are essential for finding out the maximum throughput supported by the system under various conditions.

CHAPTER 7

Conclusions and Future Work

In this project a CSMA/CA Basic Access system has been implemented using USRP's as multiple transmitters and 1 receiver. The channel sensing mechanism used here is a Frame Detection algorithm based on the timing and frequency synchronization algorithm for OFDM using the preamble. Multithreading has been implemented in C++ language using Boost libraries to enable two parallel threads for channel sensing and transmission. The threads communicate with each other using simple control variables which is sufficient for this purpose. The Throughput for this system has been calculated and the behaviour with various parameters has been analysed.

The timing parameters used in this system are not perfect and USRP round trip time has not been taken care of. The timing calculations have to be improved to get more accurate results. Also, RTS/CTS based access is more advantageous and has higher Throughput which can be implemented here. The 802.11 standard defines many other methods for scanning, joining, many PHY layers all of which can be implemented for a more complete Wi-Fi system using USRP.

APPENDIX A

Appendix A

A.1 Problems faced during the project

- The USRP faces interference at some frequencies like 2.4GHz(Wi-Fi signals),900MHz(GSM signals) etc. Hence care should be taken when choosing the center frequency. For this '*uhd_fft*' can be used to view all the frequency spectrum at one particular center frequency.
- Multithreading is not easy to handle. The threads do not run perfectly in parallel and steps have to be taken so that no object is accessed concurrently by multiple threads. For this semaphores are used.
- The *'cpu_format'* and *'wire_format'* used by the *'usrp'* object has to be carefully chosen. Problems were occurring because the receive buffer and these formats were not matching. This is a subtle thing to notice but it is very important.
- The receiver code was earlier implemented using a different algorithm for Frame detection. The code was not optimized and hence the receiver went into overflow. The newer has been optimized to a great extent.
- The USRP does not support all sampling rates. Those rates are only supported which give even or multiples of 4 upon diving 100 MSPS with them. 100 MSPS is the maximum sampling rate of the FPGA in the USRP.Also rates less than 0.2 MSPS are not supported. Also rates upto 10 MSPS have been used by us. Beyond that the USRP goes into overflow.
- The latency experienced by the USRP has not been calculated and no proper way has been found out to calculate it. Knowing the latency will help in improving throughput.

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