OFDM TECHNIQUE IN PACKET CAPTURED SLOTTED ALOHA MOBILE COMMUNICATION SYSTEMS

By

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PREFACE

This Thesis proposes a combination of OFDM systems with Packet Captured Slotted ALOHA System. The synchronization of OFDM systems is aided by the Slotted-ALOHA, which utilizes its inherent timing mechanism. Compared to [28], where OFDM was applied over ALOHA, the proposed system has an obvious two fold gain in multiple access utilization due to the inherent slotted-timing scheme. An SNR comparison in terms of the loss incurred by using OFDM frames in Pure ALOHA and Slotted-ALOHA is provided. It is shown that Slotted-ALOHA with OFDM has a better SNR performance compared to the Pure ALOHA with OFDM modulation. Also, the packet capture properties are discussed in detail and a performance comparison is provided, comparing the Slotted-ALOHA systems with packet capture to the Pure ALOHA system. Also, the obtained performance results are compared with the Carrier Sense Multiple Access (CSMA) technology, which is the fundamental access technology for the IEEE 802.11 industry standard

Orthogonal Frequency Division Multiplexing (OFDM) technique is a novel method for transmitting messages through a channel with the sole purpose of reducing the inter-symbol interference (ISI) and the inter-carrier interference (ICI). OFDM systems are very attractive for mobile communication applications due to the inherent capability to eliminate frequency selective fading. Wireless applications of OFDM technology is emerging as one of the core technologies that will be employed in the near future has technical developments permit. Some examples exist in 4^{th} generation wireless mobile communication systems, as well as IEEE 802.11a wireless local area network (LAN) devices, that provides 54 Mbps at the 5.7 GHz industrial, scientific, and medical (ISM) frequency bands.

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FOREWORD

Orthogonal Frequency Division Multiplexing (OFDM) technique is a novel way of transmitting data over a communicating channel. The idea of such a system is to reduce the effect of the channel on the signals that are being transmitted. The OFDM system achieves this by modulating the data on to a number of frequency carriers that are closely placed in the frequency domain. OFDM gets its name because each of its subcarriers is orthogonal to each other.

As it is well known that any system cannot be perfect, OFDM systems too suffer from various problems. The part of this research is devoted to addressing some of the main problems that plague the OFDM systems the most. They are the carrier frequency offset and the frame synchronization problems.

The OFDM modulation is then applied to the Slotted ALOHA Mobile Communication systems with the Packet Capture technique to estimate the performance of the Slotted ALOHA over the Unslotted ALOHA technique. Performance results show that Slotted ALOHA systems perform much better in the presence of the same channel conditions than the Unslotted ALOHA systems. Infact the same basic two fold gain in channel access utilization holds, apart from the potential for even more gain because of the use of OFDM system as a modulation technology.

This thesis is organized in to five chapters. Chapter I introduces the ALOHA Systems and the Orthogonal Frequency division Systems in detail. Chapter II focuses in detail about the background research conducted for this thesis. Chapter III focuses on the OFDM Frame synchronization and the problems associated with it. Chapter IV discusses the advantages of using such a combination of systems and technologies. Chapter V provides the derivations underwent and the simulations and results.

The report concludes with a discussion of the work so far done and also explores future research possibilities in this topic. References follow this.

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CHAPTER I

INTRODUCTION

This chapter gives the reader a general idea about a Mobile Communication System and introduces ALOHA techniques used in the wireless communication systems. Also the chapter introduces the idea of using the combination of a slotted ALOHA multiple access scheme with a very efficient and robust modulation scheme called the Orthogonal Frequency Division Multiplexing (OFDM).

1.1 Mobile Communication System Overview

The Bell laboratories developed the framework for the development of the wireless mobile communication systems. The wireless mobile communication systems were based on the concept of frequency reuse, where by a same set of radio frequencies are to be used repeatedly in a cell group. The cells are arranged to form a cellular structure. Also the concept of handoff between the cells was developed to take in to account the user mobility.

The mobile systems were categorized in to generations of development. The first generation mobile systems were analog in nature in that the modulation and other technologies were analog. The multiple access technology used by these systems was Frequency Division Multiplexing (FDM). These systems were bulkier and highly unwieldy. But due to the rapid growth and development in the semiconductor industry, and by using ASIC designs, the size of these cellular phones were reduced to that of a small handset. The second generation (2G) standards were developed based on the digital technology. Various standards like the Global Systems for Mobile communication systems (GSM), IS-136 (Time Division Multiple Access), and IS-95 (Code Division Multiple Access) were developed. The foray in to the digital technology not only improved the quality of the voice and the video services but also reduced the cost of the mobile systems to a considerable extent.

Now the latest development in the mobile communication systems is the development of the third generation (3G) networks. The 3G systems focus on improving the spectral efficiency and the data rates. Also this generation of technology aims to introduce more functionality in to the user handset like PDA etc.

1.1.1 Frequency Spectrum Allocation

Due to the enormous growth rate in terms of number of users expected in the future, the spectral efficiency that can be achieved through various technologies alone cannot ensure to support this many subscribers. Hence apart from the existing frequency range of 800-900 MHz allotted for the 2G systems, we need to allocate additional frequency bands for the 3G systems. The frequency bands allocated by various countries differ. Usually The International Telecommunication Union (ITU) and International Mobile Telecommunication (IMT 2000) regularize the choice of the frequency bands.

1.1.2 Mobile Standards

The 3G systems are based on the existing 2G systems that have different multiple access and air interface standards. For 2G systems that are based on TDMA or GSM standards, their maturity in to the 3G systems is slated to be through an Enhanced Data for GSM Evolution (EDGE) standard [4]. For 2G systems that are based on IS-95 CDMA standards, their evolution to 3G will be through CDMA 2000 and IS-2000 technologies. The 3G Universal Mobile Telecommunication System (UMTS), is based on the Wideband CDMA (W-CDMA) multiple access technology. This 3G standard uses approximately 5 MHz of bandwidth in both the links of the duplex connection. The 3G systems that are based on CDMA 2000 use 1.25 MHz in each of the duplex links.

To fasten the evolution of the third generation systems, programs were developed to focus on the CDMA 2000 and the UMTS standardization. A partnership program called 3GPP (3G Partnership Project) was developed for UMTS. A similar project called 3GPP2 was developed for CDMA 2000. Continents like United States and Europe and countries like China, Japan and Korea support and develop these projects.

The goals of these projects include specifying the air interface standards but not the protocols that are needed for the core network so that the major network standards can be developed independently. A Network – to – Network Interface (NNI) was specified by the International Telecommunication Union (ITU) to allow mobile users to operate between those cells that are connected to two completely different networks. The development of the 3G systems places one delicate situation when different systems supporting different interface standards using different frequency bands are developed and employed. The problem is that the complexity of the mobile handset is now increased, as it should be designed to support multiple standards when roaming between cells that support different standards. A software solution to this standard is suggested. The technique is called a Software Defined Radio (SDR), which necessitates a single hardware that is capable of supporting different software that implements different frequency bands and different standards.

1.1.3 Antenna Considerations

Because of the rapidity in the development of the mobile systems, there is a need to increase the system capacity and to specify appropriate frequency bands. Antenna design is an important issue related to increasing the capacity of the system. [4]. There are two techniques namely the steered-beam approach and the switched-beam approach. The first approach uses phased-array antennas, with multiple columns of equally spaced antenna arrays in pairs. This creates a narrow beam that is focused toward the mobile user. Switched-antenna beam approach is somewhat a static approach in that the steering of the antenna as the mobile user moves, is not adaptive but is simply switched from one beam to the other. Also in this method, the antenna aperture is fixed and is not a function of the channel condition, which is not the case in the previous approach. Apart from these techniques, some dynamic processing is also used to increase the gain over other similar techniques. [4] Also Bell labs have proposed a powerful intelligent antenna array technique called BLAST (Bell labs Layered Space Time). This technique employs simultaneous channels belonging to the same frequency band, but without any increase total transmitted power.

1.2 ALOHA Systems

This is a kind of channel access scheme that is employed in a packet broadcasting system [1]. Multiple access techniques like TDMA and FDMA have some disadvantages that prevent them to be an efficient channel access technique. So the packet broadcasting system uses a simple and efficient multiplexing system, where each user transmits the packets without any coordination or synchronization, over the wireless channel. If the repetition rate of each of the users is low, then the probability that a packet from one user interfering with a packet from another user is small provided that the number of users in the channel is are not very high. Thus the importance is placed on determining how many users on average can share the system without degrading the performance of the system.

1.2.1 Channel Capacity for ALOHA Systems

The channel capacity of the ALOHA system can be analyzed as follows. There are two types of errors that occur when a packet is transmitted and it is received incorrectly or lost in transit. They are (1) Random noise errors and (2) Packet Overlap errors. Due to the random nature of the type one error, we focus on the type 2 errors to improve the performance of the system. Before that, some of the basic results relating the throughput and channel traffic are derived.

The analysis starts with the assumption that the packet starting times approximate a discrete Poisson random process. Let the parameter λ be the rate at which the packets are sent in packets/seconds. If the duration of the packets is given by τ seconds, the normalized channel traffic is given by [1]

$$G = \lambda\tau \tag{1.1}$$

If it is assumed that the packets that are received correctly are those packets that do not overlap with any other packet in the system, then the normalized channel throughput is given by

$$S = \lambda' \tau \qquad (1.2)$$

Where $\lambda' < \lambda$, is defined as the rate of occurrence of the packets that are received correctly at the receiver.

The probability that the packets do not overlap is based on the probability that no packets are transmitted τ seconds sooner or later after the first packet is sent. From the Poisson distribution, this probability is given by

$$e^{-2\lambda\tau} \text{ Or } e^{-2G} \qquad (1.3)$$

Hence the throughput accordingly is given by,

$$S = Ge^{-2G} \qquad (1.4)$$

The following figure shows the plot of the channel throughput versus the channel traffic in an ALOHA system.

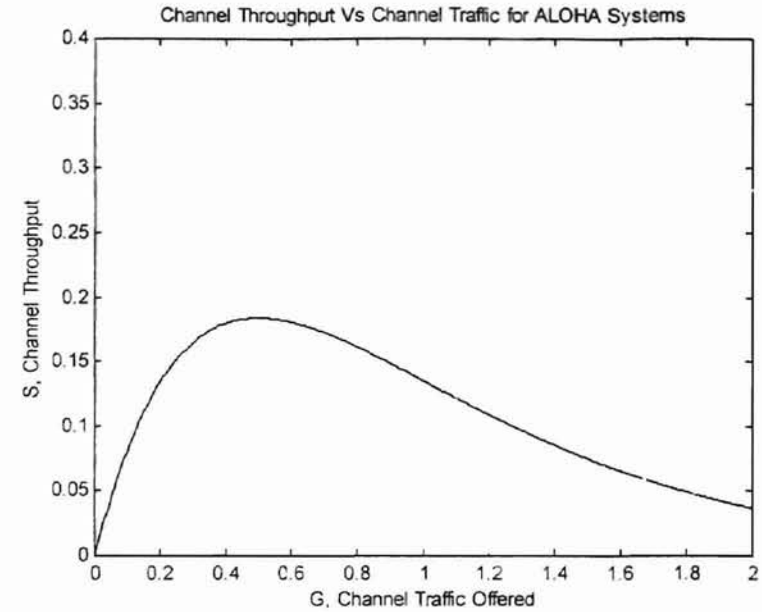

Fig.1-1 Throughput Versus Offered Load for Pure ALOHA

From this plot, we can see that as the channel traffic increases, the throughput also increases till it reaches a maximum value at $S = 1/2e =$ 0.184 after which the throughput decreases as the load on the traffic increases furthermore. This value of the throughput gives the channel capacity of the ALOHA system. The corresponding value of the channel traffic is 0.5.

1.2.2 Lost Packet Recovery

To recover from the possible packet losses over the ALOHA channel due to the packet overlap, we use different techniques of which some are discussed here.

1. Positive Acknowledgements (POSACKS): This method sends out an acknowledgement for each of the received packet. The transmitted packet is stored in a transmit buffer after it is transmitted until a POSACK for that

particular packet is received by the transmitter. If for a given amount of time the POSACK is not received, then the transmitter resends the packet again and waits for the POSACK. This POSACK can either be transmitted in the same channel as the data or can be transmitted in a separate control channel.

2. Transponder Packet Broadcasting: Satellites transponders use a mechanism of receiving a packet in one frequency and transmitting it in a different frequency. This technique is applied in the ALOHA packet broadcasting network where each of the units receive a packet and check for overlap and repeat the packet if there is an overlap.

3. Carrier Sense Packet Broadcasting: Usually in most of the ground-based packet broadcasting networks, the propagation time of the packet over the channel is much less than the duration of the packet itself. In such a situation it is practical and feasible to provide each transmitting unit with a device to prevent the unit from transmitting in to the channel if it detects that other transmitter is currently utilizing the channel. This type of sensing the channel increases the channel throughput considerably.

4. Packet Recovery Codes: While the first three methods use a kind of a feedback channel to the transmitter, this method is based on the coding mechanism. If a long user file data is transmitted as a sequence of smaller packets, to recover from any possible overlap, these packets are encoded using burst error-correcting codes or cyclic product codes. But this method may not be as efficient as the others unless some more research has been done in this field.

1.3 Slotted ALOHA Systems

The unsynchronized ALOHA systems can be modified in to a more of an organized multiple access system so that the throughput can be increased to a considerable extent. In a pure ALOHA system, the users transmit the packets as soon as they are ready to transmit, without any coordination among other users of the system. This method leads to inefficiency of channel utilization. So an improved version of ALOHA called a Slotted ALOHA system is proposed where each user starts transmitting the data at fixed instants of time. This method achieves the improvement in the channel throughput, as we will see later. In this slotted ALOHA channel, a central clock sets up a time base for a sequence of slots that are of same duration as that of the packet transmission. That is each slot is of the exact duration as the length of each packet. Once the clock releases a set of slots, if a user has some packets to be sent, he synchronizes the start of the transmission of the packets to the start of a slot. Hence if there is any overlap, that is, if two users transmit at the start of the slot, then they will completely overlap for the duration of the slot or slots. Thus there is no partial overlap of data. The slotted ALOHA channel can be analyzed as follows.

Let G_i be the probability that the i^{th} user will transmit a packet in some given slot. Also each user is assumed to be operating independently of each other. The transmission of packet in a given slot is not dependent on the state of any previous slot. If there are n users in the system, then the normalized channel traffic for the slotted ALOHA system is given by [1]

$$G = \sum_{i=1}^{n} G_i \tag{1.5}$$

Let $S_i \leq G_i$ be defined as the probability that a user sends a packet in a slot and that this is the only packet in that slot. That is there is no overlap in that slot. Then if there are n users in the system, the normalized channel throughput S for the slotted ALOHA system is given by

$$S = \sum_{i=1}^{n} S_i \tag{1.6}$$

Here $S \leq 1$ and $S \leq G$. The probability that a packet from the i^{th} user will not be interfered by the packets from the other users is given by,

$$\prod_{\substack{j=1 \\ j \neq i}}^{n} (1 - G_j) \tag{1.7}$$

Thus the relationship between the throughput and the traffic rate for the i^{th} user can be written as [1]

$$S_i = G_i \prod_{\substack{j=1 \\ j \neq i}}^{n} (1 - G_j) \tag{1.8}$$

If all the users in the system are identical, then

$$S_i = \frac{S}{n} \qquad \text{And} \tag{1.9}$$

$$G_i = \frac{G}{n} \tag{1.10}$$

Hence the above equation can be written as

$$S = G\left(1 - \frac{G}{n}\right)^{n-1} \tag{1.11}$$

And as $n \to \infty$, we get

$$S = Ge^{-G} \tag{1.12}$$

The above equation is plotted in the following figure. It can be seen that the throughput of the slotted ALOHA system reaches a maximum capacity of $1/e = 0.368$, which is twice as much as the pure ALOHA system.

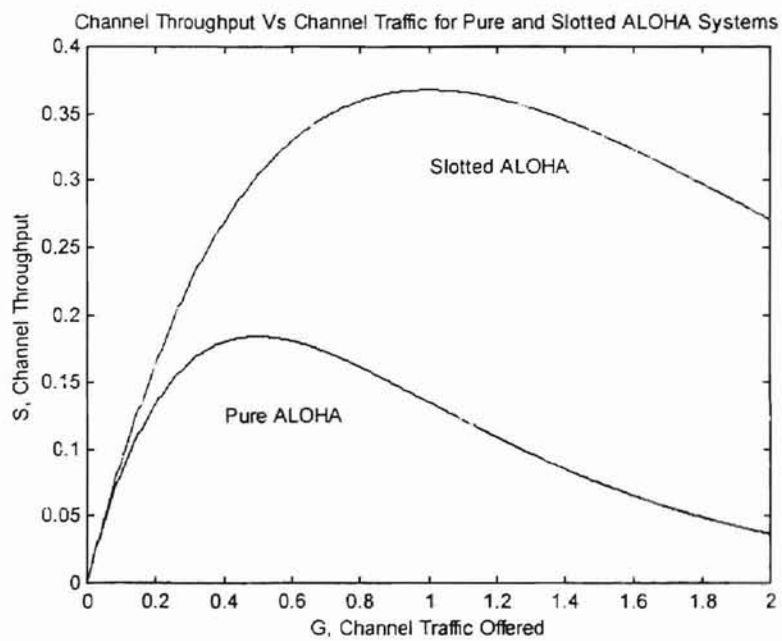

Fig.1-2 Throughput Versus Offered Load for Pure and Slotted-ALOHA

1.4 OFDM

Orthogonal Frequency Division Multiplexing (OFDM) is a modulation technique that is widely being recognized as a robust technique against multipath fading and Intersymbol interference. Applications of OFDM have been in the Digital Subscriber

Lines (DSL). Also OFDM has been used in the wireless Local Area Networks. This technique gets its name because all of the subcarriers in this system are orthogonal to each other. This property of orthogonality is due to the use of Inverse Fast Fourier Transform at the OFDM transmitter. Some of the OFDM properties that make it so popular are [12]

1. The symbol period of this multicarrier system is long compared to that of a single carrier system that makes it robust to channel fading and immune against impulse noises.
2. With OFDM, different subcarriers can use different modulation techniques using a concept called dynamic bit allocation.
3. The channel effects can be easily taken care of in OFDM by employing a simple equalizer (one-tap) for each subband.
4. The hardware implementation of OFDM can be greatly simplified because of its efficient implementation of the FFT pairs even for the large number of subcarriers.

1.4.1 OFDM Transmitter

The figure shows a simple OFDM transmitter. [12]. The input to the system is a series of user symbols arriving at the rate of R symbols/second. These are then sent through a serial-to-parallel block that produces an output on M lines with a data rate of R/M symbols/second on each of the M outputs.

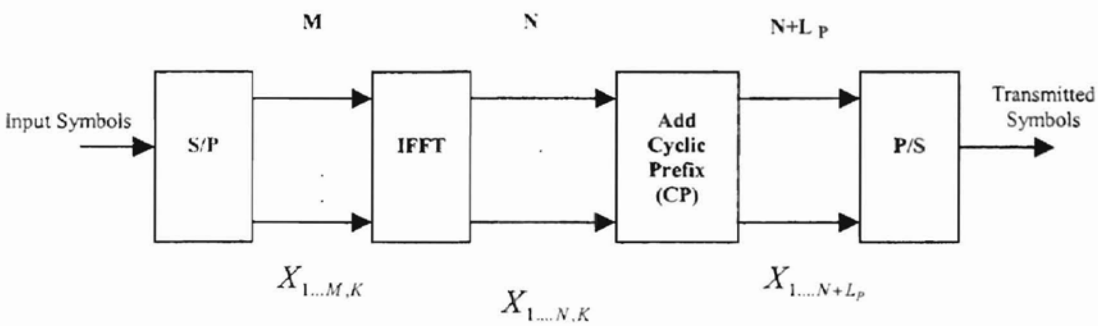

Fig.1-3 OFDM Transmitter

Let the symbols that come out of the serial to parallel block be represented as $X_{i,k}$. i denotes the subband to which the OFDM symbols belong and it ranges from 1 to M. k denotes the k^{th} set of M symbols. $X_{1,k,}X_{2,k}$ etc can be called as subsymbols. This set of subsymbols ($X_{1,k}$ to $X_{M,k}$) form a single OFDM symbol. This set of M symbols is sent to the point N IFFT block. This converts the incoming data symbols from the frequency domain in to N time domain samples ($X_{1,k}$ to $X_{N,k}$).

1.4.2 Cyclic Prefix

To the N time domain samples, a Cyclic Prefix (CP) is added. Lets denote the length of the cyclic prefix to be L_p samples. These samples are attached to the beginning of the N samples to form a cyclically extended symbol. Hence the length of the OFDM symbol now becomes $N + L_p$ samples. This mechanism of pre-pending the OFDM symbol with a cyclic prefix helps to remove the effects of the channel at the receiver. These samples are then sent through a parallel-to-series converter block and are sent through the channel after filtering and bandpass modulation. The cyclic prefix makes the OFDM symbol appear periodic over a duration of time. In discrete time domain, a circular convolution takes place instead of linear convolution.

The channel affects the transmitted OFDM symbols with its impulse response. The result is the circular convolution of the transmitted OFDM symbols with that of the channel impulse response. This is equivalent to multiplying the OFDM symbol's frequency response with that of the channel's frequency response. It is worthwhile to notice that the frequency response of the OFDM symbol gives the original user data in the frequency domain. Thus the effect of the channel on the OFDM symbol is to multiply each of the original data symbol with that of a single complex number, which is the DFT of the channel at that corresponding frequency, thus greatly simplifying the equalization process.

1.4.3 OFDM Receiver

The following figure shows a simple OFDM receiver. The receiver operation does just the inverse of what the transmitter does. Let the received signal samples be represented as $Y_{1 \ldots N+L_{p,K}}$ in the time domain. The channel affects the received samples. The samples are first sent through a serial-to-parallel converter. Then they are sent to a block, which removes the cyclic prefix (L_P). Since the CP contains data samples that are redundant, its removal does not affect the OFDM data symbols in any way. But it helps in identifying the effects of the channel.

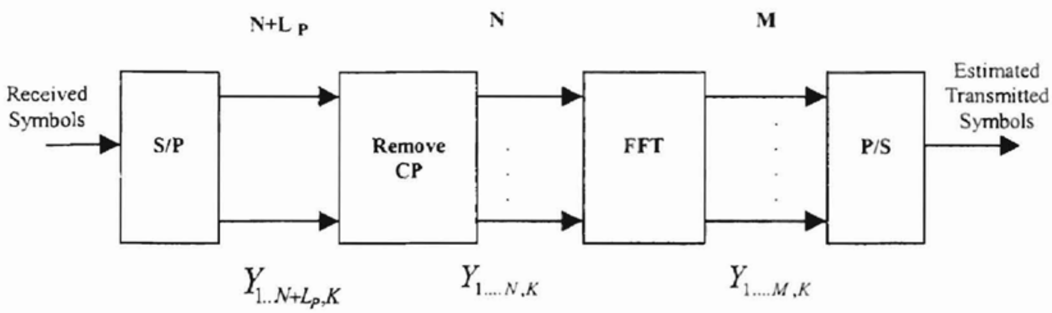

Fig.1-4 OFDM Receiver

Also the CP apart from identifying the channel effects helps in avoiding the effect of the Intersymbol interference (ISI), if the L_P samples are greater than the memory of the channel. That is greater than the channel impulse response time. ISI is a major cause of concern in the multipath fading channels, that occurs when signals get delayed traveling through multiple channel paths and interfere with other OFDM symbols in the receiver. So if the CP is larger than the channel response time, the ISI affects only the CP part of the received symbol, which is removed anyway at the receiver. Thus OFDM symbols are immune to the ISI even though they are affected by it.

After this, the remaining samples are sent through a N-point Fast Fourier Transform (FFT) block to get the data back to the frequency domain. From this the received samples are formed from the first M samples of the FFT block output and the remaining $N-M$ samples are omitted. The samples after the FFT block $Y_{1\ldots M,K}$ still suffer from the channel effects.

After the FFT the received signals can be represented as a frequency domain multiplication as follows [12]

$$Y = CX + \eta \tag{1.13}$$

Where Y is the frequency response of the received signal, C is the channel's frequency response and X is the frequency response of the transmitted signal and η represents additive noise. To rectify the channel effects the equalization process is performed as follows [12]

$$Y_{equalized} = Y / C = X + \eta / C \tag{1.14}$$

Thus the result of this equalization is the original transmitted symbol and a noise term. For very large fades, the channel's frequency response will have small values resulting in the noise term taking prominence. Now equalizing the received signal will therefore amplify the noise in the signal and producing distorted output. So to reduce these effects at the receiver, practical applications make use of a threshold value that determines the equalization process. Those carriers with very small values will not be equalized thus reducing the distortion at the output. The parallel-to-series converter block follows the FFT block. The output of the P/S block gives the estimate of the transmitted OFDM symbol.

1.5 OFDM Over Slotted ALOHA Technique

This report proposes a novel technique for achieving frame synchronization for Orthogonal Frequency Division Multiplexing (OFDM) systems. The proposed scheme uses the inherent timing mechanism of the Slotted ALOHA multiple access technology. The OFDM frames exist in the synchronized time slots from the slotted ALOHA multiple access technology. This kind of transmission of OFDM frames in the ALOHA slots achieves coarse synchronization by itself. Now the fine-tuning is required to perfectly detect the frame synchronization pattern, which will dynamically change due to the user mobility and the transmission signal multipath and shadowing effects.

OFDM technique is a novel method for transmitting messages through a channel with the sole purpose of reducing the Inter-symbol interference (ISI) and the Inter-carrier interference (ICI). OFDM systems are very attractive for mobile communication applications due to the inherent capability to eliminate frequency-selective fading. Already OFDM has found applications in the third generation (3G) and fourth generation (4G) wireless communication technologies.

Frame synchronization is one of the most critical processing functions that an OFDM system has to accomplish in order provide high quality data rate transmission. Frame synchronization problems arise when the receiver cannot estimate the exact starting position of the frame from a sequence of frames transmitted. Carrier offset arises because of the mismatch between the transmitter and the receiver tuning circuits. These synchronization problems lead to ISI and ICI and hence the loss of data.

Most proposed schemes for OFDM frame synchronization use two steps: Coarse Synchronization and Fine synchronization. The first step is used to limit the timing error to a certain range. Within this range fine synchronization is done to find the exact starting point of the OFDM symbols. Traditionally, coarse synchronization has been based on correlation method by using the cyclic prefix part of the OFDM symbol. In [30], the author uses a supplementary guard interval to do the coarse synchronization. A metric like $\Delta = ab - \frac{1}{2}(a^2 + b^2)$ is used. Where a and b are the samples of the supplementary guard interval and the OFDM data respectively. This metric gets its maximum value of zero when $a = b$. This value of the maximum will be the coarse estimation of the start of the OFDM symbol. But this method does not give a unique point but only gives a range. The other drawback is that the metric is very flat when a and b is close to each other. The result is a large standard deviation from the average. In [28], the author proposes a frequency domain synchronization scheme for both coarse and fine synchronization. This coarse synchronization can achieve synchronization to within $\pm\frac{1}{2}$ of a sample period. But this method has a probability of failure of about 10^{-2} to 10^{-3} when using 15 and 18 synchronization tones. This failure probability affects the overall performance of the system.

Compared to these schemes, the proposed Slotted ALOHA-OFDM architecture does not have the possibility of coarse OFDM frame synchronization failure. Every frame will exactly start at the frame boundary. Instead of coarse synchronization, what we should do is just to detect the presence of an OFDM symbol and if there is a collision due to more than one mobile station that are sending the symbols in the same slot. After the symbol detection, the fine-tuning mechanism of the frame synchronization mechanism, which is similar to the method applied in [30], will be done. This enables the proposed OFDM system to perform better as well as have a much simpler processing architecture.

Suppose the mobile users in a Slotted ALOHA system is spatially distributed around a base station in a annular ring with the maximum radius R transmitting over a Rayleigh fading channel, then the signal power of a user within the annular ring defined by $0 < r \leq x \leq R$ may be modeled by propagation law as

$$E = x^{-\alpha} \tag{1.15}$$

Where E is the received power at the base station, x is the distance of the mobile user to the base station and $2 < \alpha < 5$ for ultra high frequency (UHF) applications. The average SNR that is received at the base station can be given by the propagation law as follows

$$\Gamma_i = (\frac{R}{x_i})^4 \Gamma(R) \tag{1.16}$$

Where x_i is the distance from the i^{th} transmitting user to the base station. [31].

In Slotted ALOHA multiple access networks, there are two possibilities of a packet being transmitted over the Rayleigh fading channel. One, the packet can be received correctly at the base station without any collision and two, the packet can collide with packets from other users in the system. So, we need two power level thresholds for the two cases: a Symbol detection threshold and a Collision detection threshold. The base station will monitor the power from the beginning of each slot and compare them with the two thresholds:

$$\text{If } \Gamma(R)\beta_1 > \Gamma_i \qquad \text{the packet does not exist} \tag{1.17}$$

$$\text{If } \Gamma(R)\beta_1 < \Gamma_i < \Gamma(R)\beta_2 \qquad \text{the packet exists and no collision} \tag{1.18}$$

$$\text{If } \Gamma(R)\beta_2 < \Gamma_i \qquad \text{the collision exists} \tag{1.19}$$

Where β_1 and β_2 are real numbers which can be obtained by the heuristics to be $0.3 \sim 0.5$ and $1.2 \sim 1.3$ respectively. When a signal is detected, the system will directly perform a fine synchronization. If a collision is detected, this slot will be discarded and the symbol will be transmitted again.

The expected conclusion, compared to the performance obtained in [28], shows that the proposed novel method of applying OFDM over Slotted –ALOHA scheme possesses more than a two-fold gain in multiple access utilization due to the inherent slotted-timing scheme.

CHAPTER II

BACKGROUND RESEARCH

This chapter discusses about the research conducted on OFDM systems and CDMA and Wideband CDMA systems. This chapter also discusses the implementation of the OFDM systems in IEEE 802.11a systems. The remaining part of the chapter discusses in length about the multipath fading effects that play a major role in any wireless communication systems and the last part addresses the synchronization issues related to the OFDM systems.

2.1 Orthogonal Frequency Division Multiplexing (OFDM)

Orthogonal Frequency division Multiplexing technique is a multicarrier digital modulation technique. The basic technique of OFDM is to divide a high-speed data transmission coming in to the system in to a number of low speed streams that are transmitted simultaneously over a number of subcarriers. Hence, since the transmission for the lower rate symbols are increased in this way, and also since OFDM systems use a concept of Cyclic Prefix, the effects of the multipath delay spreads leading to Intersymbol Interference (ICI) are reduced. Also instead of wasting the guard time, the OFDM signal is cyclically extended in this period to avoid Intercarrier Interference (ICI). The popularity of this system has increased multifold in recent years because of some of the useful properties of the system.

1. OFDM has a long symbol period (compared to an equivalent single carrier system) that allows OFDM systems to be more robust to impulse noise and fast channel fades.
2. The ability to efficiently implement an N-point Fast Fourier transform in hardware for a large N greatly simplifies the practical implementation of OFDM system.
3. OFDM is very robust against narrow-band channel interference as it affects only a small proportion of the subcarriers and hence the possibility of the data being reconstructed successfully at the receiver. But this is not the case in a single carrier system where data is lost in the carrier is knocked off due to a frequency-selective fade.

Usually OFDM is viewed as a collection of transmission techniques. When applied in a wireless environment such as in radio broadcasting it is usually called as OFDM. However in a wired environment, such as in asymmetric digital subscriber lines (ADSL), the term Discrete Multitone (DMT) is more appropriate.

OFDM's history dates back to the 1960's when Chang published a paper on synthesis of bandlimited signals for multichannel transmission. He presented a principle of transmitting messages through a bandlimited channel without any Inter Carrier Interference (ICI) or Inter Symbol Interference (ISI). After this Saltzberg analyzed and concluded that the means to design an efficient system that can transmit different signals simultaneously through a bandlimited channel should aim to reduce the crosstalk between adjacent channels as this is the one that dominates at higher frequencies.

A major contribution to OFDM was presented in 1971 by Weinstein and Ebert, who used the Discrete Fourier Transform (DFT) to perform the baseband modulation and demodulation. To counter the problems of ISI and ICI, they used both a guard space between the symbols and a raised-cosine windowing in the time domain. Although their system did not obtain perfect orthogonality between subcarriers under dispersive channel conditions, theirs still was a major contribution to OFDM systems.

Peled and Ruiz made another notable contribution in 1980, when they introduced the concept of Cyclic Prefix (CP) or cyclic extension, solving the orthogonality problem. Instead of using an empty guard space, they filled the guard space with a cyclic extension of the OFDM symbol. This effectively simulates a channel performing cyclic convolution, which implies orthogonality over dispersive channels when the CP is longer than the impulse response of the channel.

2.1.1 Generation of an OFDM Signal

The following figure shows the block diagram of an OFDM modulator that is used as a high end modulating system. The input to the OFDM system is a series of Phase Shift Keying (PSK) data or Quadrature Amplitude Modulation (QAM) data. These sequences of data are used to modulate the orthogonal carriers of the OFDM system to generate the OFDM data that is transmitted through the air. The mathematical representation of the OFDM system can be represented as from [34].

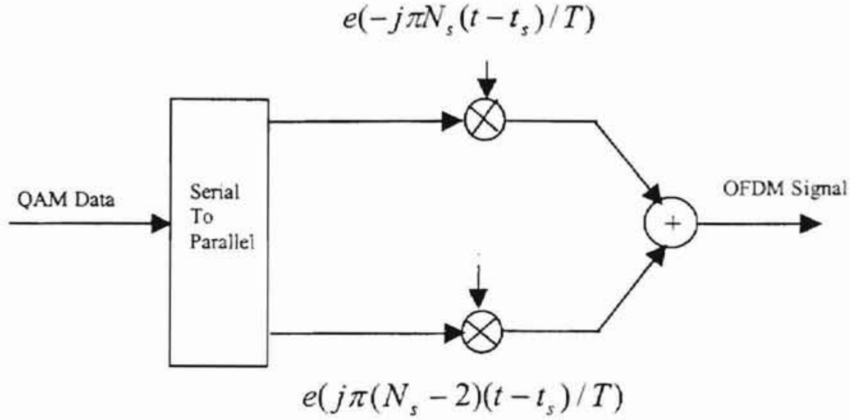

Fig. 2-1 OFDM Modulator

$$s(t) = \sum_{i=\frac{-N_s}{2}}^{\frac{N_s}{2}-1} d_{i+\frac{N_s}{2}} \exp(j2\pi \frac{i}{T}(t-t_s)), \qquad t_s \le t \le t_s + T \tag{2.1}$$

$$s(t) = 0, \qquad t < t_s \wedge t > t_s + T \tag{2.2}$$

Where N_s is the number of subcarriers, T is the symbol duration and t_s is the starting time of one OFDM symbol. The exponential in the above equation indicates the carriers with their frequencies spaced $1/T$ apart that modulate the incoming stream of QAM /PSK data. The above equation can be achieved using an inverse Discrete Fourier Transform (IDFT) to generate the samples at each of the carrier frequencies. This is given by,

$$s(t) = \sum_{i=0}^{N_s} d_i \exp(j2\pi \frac{i}{N} n) \tag{2.3}$$

Where the time t is replaced by the sample number n. The practical implementation of the OFDM system is done in the same manner but using a technique called Inverse Fast Fourier Transform (IFFT) which reduces the computational complexity of the IDFT system to a great extent and simplifies the overall system implementation of the OFDM modulation.

2.1.2 OFDM Symbol: Guard Time and Cyclic Extension

The main impetus to do OFDM is the way in which this technology deals with multipath delay. OFDM introduces the guard symbol duration to completely eliminate the problem of Intersymbol Interference (ISI). The selection of the guard time is made such that it is longer than the estimated delay spread of the channel, such that multiple path components of one signal cannot interfere with the other signal. In order to get rid of the problem of the Intercarrier Interference (ICI), which arises due to the problem of crosstalk between different subcarriers there by causing nonorthogonaltiy between carriers, the OFDM symbol is cyclically extended in the guard time. Without any cyclic extension in the OFDM symbol, a delayed subcarrier might have an interference with the subcarrier the OFDM receiver is demodulating, because there will not be any integer number of cycles difference between the two subcarriers in the FFT window. The cyclic extension ensures that delayed replicas of OFDM symbol always have an integer number of cycles with in the FFT window, as long as the delay is with in the guard interval. As a result multipath signals with delays that are smaller than the guard time cannot cause ICI.

The following figure shows the effect of the multipath on the OFDM symbol with an empty guard interval. The next figure shows a cyclically extended guard interval in an OFDM symbol.

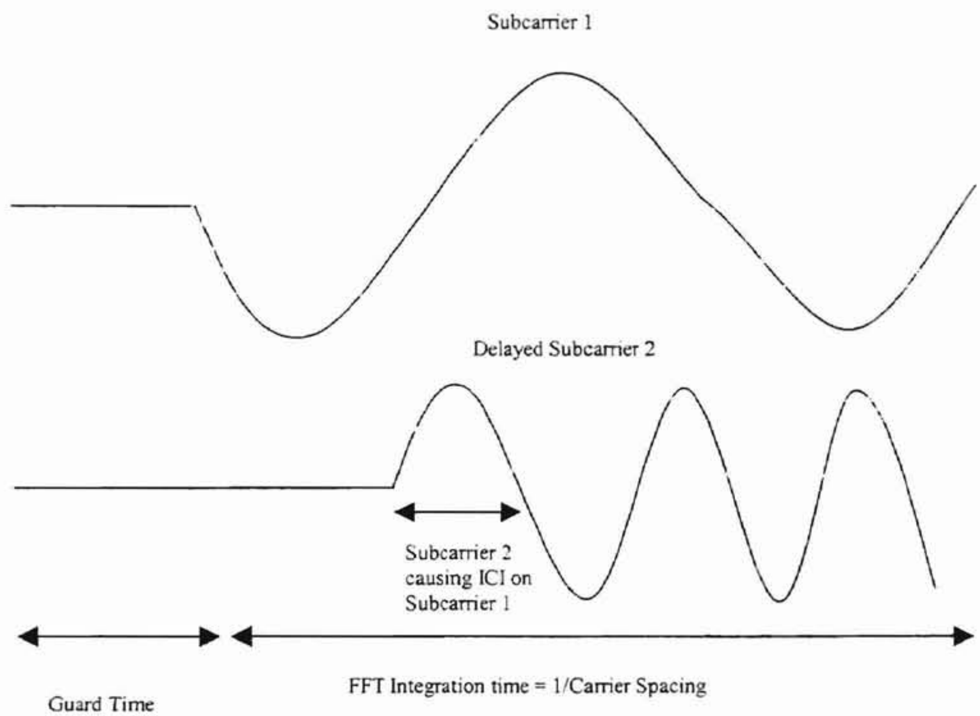

Fig.2-2 Effect of Multipath on OFDM Signals

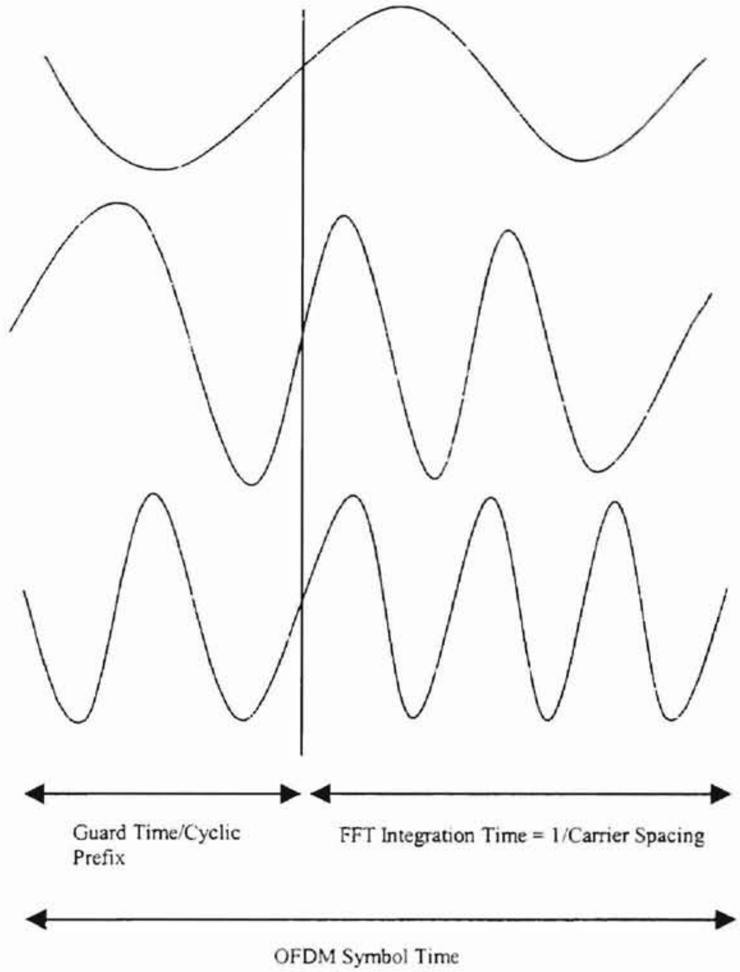

Fig. 2-3 OFDM Symbol with Cyclic Extension

2.1.3 OFDM System Environments

According to [36], OFDM technique is discussed in terms of the two major system environments namely the wired and the wireless environments.

2.1.3.1 Wireless Systems

In wireless radio systems, changes in the physical environment cause the channel to fade. These changes include both relative movement between transmitter and receiver and moving scatterers/reflectors in the surrounding space. In theoretical studies of wireless systems, the channel models are usually chosen so that they result in tractable analysis. The two major classes of fading characteristics are the Rayleigh and Rician. A Rayleigh-fading environment assumes no line-of –sight and no fixed reflectors/scatterers. The expected value of the fading is zero. If there is a line-of-sight, this can be modeled by Rician-fading, which has the same characteristics as the Rayleigh-fading except for a non-zero expected value. We will be dealing with these fading effects in more detail in the later sections of this chapter.

Often properties of a theoretical model are characterized by only a few parameters, such as power-delay profile and maximal Doppler frequency. The power-delay profile $\rho(\cdot)$ depends on the environment and a common choice is the exponentially decaying profile

$$\rho(\cdot) = e^{-\tau/\tau_{rms}}, \tag{2.4}$$

Where τ is the time delay and τ_{rms} is the root mean-squared (RMS) value of the power-delay profile. The maximal Doppler frequency $f_{d,\max}$ can be determined by

$$f_{d,\max} = f_c \frac{\upsilon}{c} \quad , \tag{2.5}$$

Where the carrier frequency is f_c Hz, the speed of the receiver is υ m/s, and the speed of light is $c \approx 3\times10^{8}$ m/s. Isotropic scattering is normally assumed, i.e. the received signal power is spread uniformly over all angles of arrival, which results in a U-shaped Doppler spectrum. Here we discuss the wireless environment in two contexts: the transmission from a base-station to mobile terminals (downlink) and the transmission from mobile transmission to a base-station (uplink).

Downlink

A picture of the downlink transmission is shown in the Figure .In this case, the mobile terminal number n receives the signal $s(t)$ transmitted from the base station through its own channel $g_n(t)$, and the received signal $r_n(t)$ is given by

$$r_n(t) = (s * g_n)(t)\,. \tag{2.6}$$

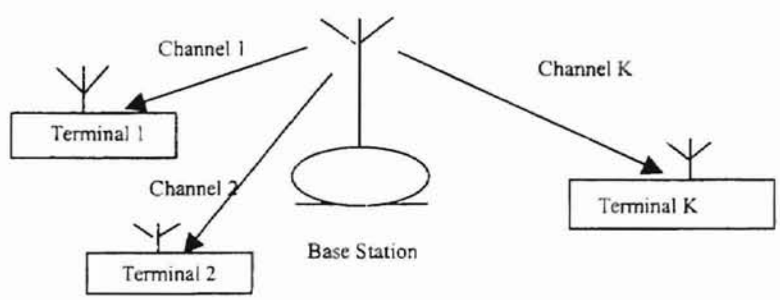

Fig.2-4 Wireless Downlink Environment

This environment implies that each receiver (terminal) only has to synchronize to base station and, from its point of view, the other terminals do not exist. This makes synchronization relatively easy and all pilot information transmitted from the base station can be used for channel estimation and synchronization.

Uplink

A picture of the uplink system is shown in Figure. In this case, the base station receives the transmitted signal $s_n(t)$ from mobile terminal n through channel $g_n(t)$, and the total received signal $r(t)$ at the base station is a superposition of signals from all mobile terminals.

$$r(t) = \sum_{n=1}^{K} (s_n * g_n)(t) \tag{2.8}$$

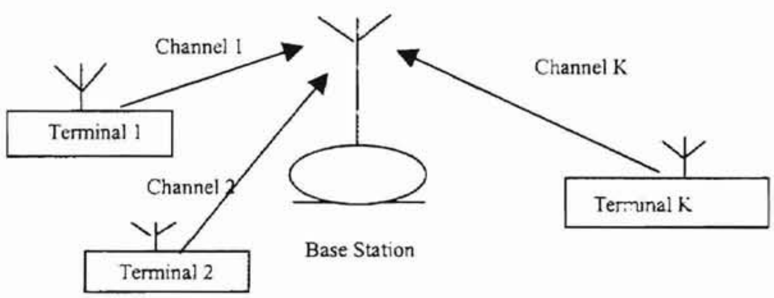

Fig.2-5 Wireless Uplink Environment

The major problem here is the superposition of signals arriving through different channels. For the base station to be able to separate the signals from each receiver, a sufficient orthogonality between received signals, from different terminals, has to be achieved. Several methods for obtaining this have been proposed. These include combinations of OFDM and code-division, time-division and frequency-division multiple access (CDMA, TDMA and FDMA respectively). Independent of the method selected to separate signals from different terminals, the system synchronization is one of the major design issues. To avoid interference, all mobile terminals have to be jointly synchronized to the base station. Further in the case of coherent modulation being used, different channels from the users have to be estimated separately.

2.1.3.2 Wired Systems

A distinction usually is made between shielded cables (like coaxial cables) and unshielded cables (like twisted pairs), when studying the transmission characteristics of cables. Coaxial cables have much better transmission properties for broadband signals than do wire pairs. Wire pairs are the dominating cable type in the case of telephone networks that are built for point-to-point and two-way communication. Coaxial cables are usually used for the cable TV systems, a network that is primarily intended for broadcasting. We focus on the wire pairs from now on.

The copper wires are considered a stationary channel, as they do not change their physical behavior significantly with time. This makes it possible to use a technique called Bit loading, which makes good use of the spectrally shaped channel. When bit loading is used in a wired OFDM system, it is often referred to as DMT. Since OFDM in combination with bit loading makes efficient use of the available bandwidth it has become a good candidate for digital subscriber-line (DSL) systems otherwise called as digital high-speed communication in the telephone access network. When the bit rate offered in downstream direction (to the subscriber) is larger than the bit rate in the upstream direction (to the base station), it is called an asymmetric digital subscriber-line (ADSL). ADSL is suitable for applications like video on demand, games, virtual shopping, Internet surfing etc., where most of the data goes from the base station to the subscriber. The downstream bit rates are usually from 1.54 to 6.1 Mbits /s in the USA. The upstream bit rates usually range between 9.6 and 192 Kbits /s.

Noise and Crosstalk

The most important noise sources in the subscriber line environment are crosstalk from other wire pairs in the same cable, radio frequency (RF) noise from nearby radio transmitters, and impulse noise from relays, switches, electrical machines, etc. AWGN is generally not a limiting factor in digital subscriber-lines for short cables, but becomes more important with increasing cable length.

Impulse noise is difficult to characterize completely but some efforts has been made to model these kinds of disturbances. The normal way to counter the effects of impulse noise on a DMT system is to add 6-12 dB to the system margin and to use specially designed codes. It should be noted that DMT is more resistant to impulse noise than the single carrier systems. The impact of RF noise on a DSL system can be reduced significantly with OFDM and bit loading. RF noise can be modeled as narrowband disturbance with known spectral density and the bit-error rate (BER) can be preserved by transmitting fewer (sometimes zero) bits on the disturbed subchannels.

There are basically two different types of crosstalk: near-end crosstalk (NEXT) and far-end crosstalk (FEXT). NEXT occurs at the base station when the weak upstream signal is disturbed by strong downstream signals. FEXT is crosstalk from one transmitted signal to another, in the same direction and appears at both ends of the wire loop.

The OFDM topic is revisited at the end of this chapter where issues related to synchronization are discussed.

2.2 Code Division Multiple Access (CDMA) and Wideband CDMA (WCDMA)

Multiple access is the basic requirement for any air interface design. Multiple access is the technique that is concerned about how the common transmission medium is shared between users. There are mainly three different types of multiple access techniques. They are (1) Frequency Division Multiple Access (FDMA), where the total system bandwidth is divided in to number of frequency channels and each user is allotted a different channel. (2) Time Division Multiple Access (TDMA), where each frequency channel is divided in to time slots and each user is allotted a slot of time in that frequency channel. And (3) Code Division Multiple Access (CDMA), where each user is assigned an unique pseudo-random code (also called pseudo-noise code due to noise-like autocorrelation properties), which has good auto and cross correlation properties. This code is used to transform the user's signal bandwidth to a wideband signal called a spectrum spread (SS) signal. This is a wideband signal whose bandwidth is many times larger than the user's baseband signal. At the receiver, the same pseudo-random code is used to transform the wideband signal to the original signal bandwidth. The wideband signals due to other users remain as wideband signals. This SS signal has a high immunity against narrow band interference. Also it is highly secure because the signals appear as noise for the unintended receivers.

Usually, we do not have the entire bandwidth spectrum to play with. The bandwidth spectrum is already divided in frequency before being allotted to different applications. Hence the CDMA or TDMA use FDMA to divide the bandwidth in to smaller frequency channels that are then divided in a time division or code division fashion. The CDMA technique is the widely used form of multiple access in wireless devices, as it is evident from the fact that all the third generation (3G) networks that are now being developed are based on the wideband-CDMA (WCDMA) technology.

2.2.1 Classification of CDMA

CDMA systems are classified according to the modulation method used to obtain the wideband signal. Based on this there are three types of CDMA. They are (1) Direct-Sequence CDMA (DS-CDMA), where the spectrum is spread by multiplying the baseband information signal with the pseudo-noise signal to obtain a wideband signal. (2) Frequency-Hopping CDMA (FH-CDMA), where the pseudo-noise sequence defines the instantaneous transmission frequency. The frequency hopping can be slow (several symbols in one hop) or fast (several hops in one symbol). And (3) Time-Hopping CDMA (TH-CDMA), where the pseudo-noise sequence defines the instantaneous transmission time of the signal. DS-CDMA is the technique that is used widely in generating the wideband signal. All 3G networks use DS-CDMA for wideband signal generation. [33] defines Wideband CDMA as a direct sequence spread spectrum multiple access scheme where the information is spread over a bandwidth of approximately 5 MHz or more.

2.2.2 Basic CDMA Principles and Elements

The multiple access capability of the CDMA is achieved by the Spread Spectrum (SS) technique. There are some requirements of the spread spectrum signals that are to be met for a wideband CDMA system. These requirements are

1. The transmission bandwidth of the signal should be much larger than the information bandwidth.
2. The RF transmission bandwidth is statistically independent of the information signal as the transmission bandwidth is determined by the pseudo-random code.

The processing gain of the spread-spectrum system is defined as the ratio of the transmitted bandwidth to the information bandwidth. This is given by,

$$G_P = \frac{B_t}{B_i} \tag{2.9}$$

Where B_t is the transmission bandwidth, B_i is the information bandwidth and G_P is the processing gain of the system.

Because of the resulting enlarged bandwidth after coding using the pseudo-noise sequence, the SS signals have a number of properties that are different from those of the narrowband signals. These properties are

1. Multiple Access capability: Even if many number of users transmit the signals at the same time and at the same frequency, the receiver will still be able to detect the users individually, provided that each user has a unique pseudo-noise code that has sufficiently low correlation with the other codes. Correlating the received signal of a large bandwidth with the spreading code of a particular user will result in the despreading of that particular user. Thus in the information bandwidth, the

power of that particular user will be larger than that of the other interfering users and the desired signal can thus be extracted.

2. Multipath Interference Rejection: Multipath interference is due to the fact that there are many paths in which a signal can arrive at the receiver. The signals of different paths are replicas of the original signal but with different amplitudes, phases, delays and arrival angles. Adding these may result in destruction of the original signal at certain frequencies. Spread Spectrum effectively reduces these effects due to multipath but the way in which this is achieved depends on the type of modulation that is used.

3. Security: The SS mechanism is very secure and reliable in that the transmitted signal can only be despread and detected if the spreading code is known to the receiver. Otherwise the signal appears as noise to the unintended receiver.

4. Narrowband interference rejection: The receiver sees the narrowband interference as the sum of the transmitted wideband signal and the interfering signal. The receiver despreads the SS signal while the interference signal is spread making it appear as a background noise at the receiver.

5. Low Probability of interception (LPI): Because the signal resides below the noise level, there is a very low probability that an unintended receiver will detect the signal.

Since DS-CDMA is the widely used technique, we will restrict our focus to this system only.

2.2.2.1 DS-CDMA Modulator

In DS-CDMA, the input to the modulator is mostly a digital signal. This signal is directly multiplied by the code signal and the resulting signal modulates a wideband carrier. It is from this direct multiplication operation that the Direct Sequence CDMA gets its name. The following figure shows the block diagram of the DS-CDMA modulator. The chipping code is selected such that it is higher than the rate of the information signal to obtain the desired spreading.

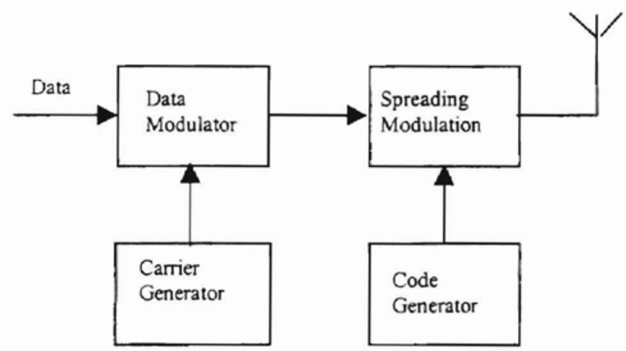

Fig.2-6 DS-CDMA Modulator

For the spreading modulation, the wideband modulator usually employs some form of Phase Shift Keying (PSK) techniques like binary PSK (BPSK), differential binary PSK (D-BPSK), quadrature PSK (QPSK) or minimum shift keying (MSK) technique.

2.2.2.2 DS-CDMA Demodulator – The RAKE Receiver

A DSSS signal is resolved in to a multipath signal when traveling through a multipath channel. From each multipath component's point of view, other components are considered as interference. This can be suppressed using the processing gain. But a greater benefit can be obtained if the resolved multipath signals are combined using the RAKE receiver. The RAKE receiver is shown in the following figure.

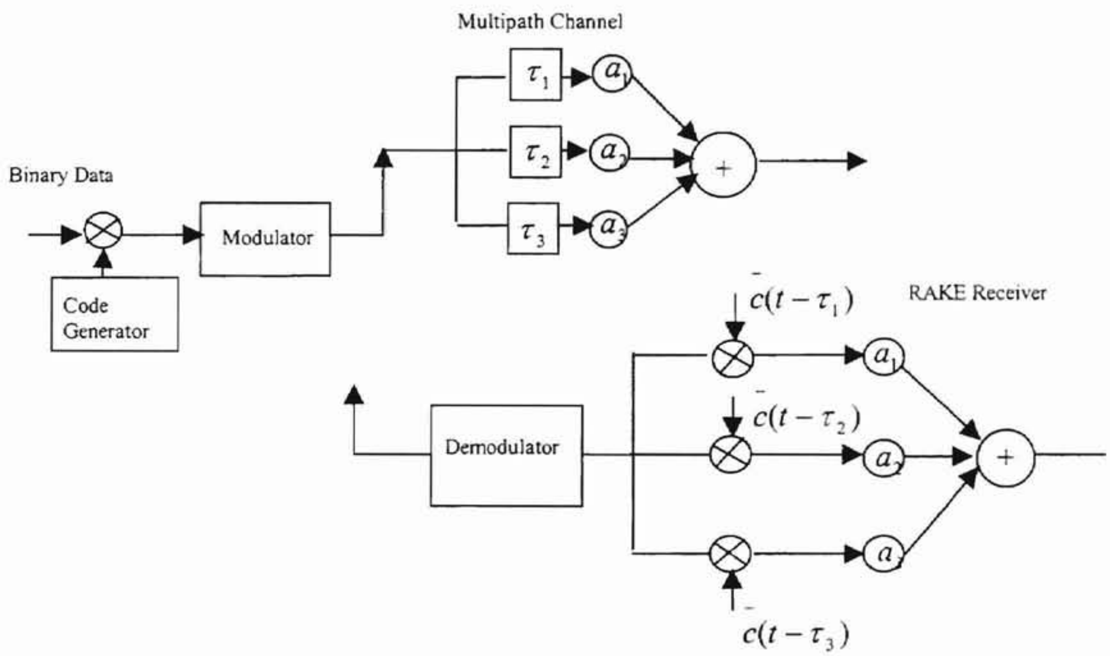

Fig.2-7 The RAKE Receiver Principle

RAKE receivers consist of correlators, each receiving a multipath component. After the signal is spreaded and modulated, the signal is transmitted on to the medium. The signal passes through a multipath channel. Since the transmission bandwidth is very much greater than the channel coherence bandwidth, the effect of the channel becomes a frequency selective fade that has little or no influence on the transmitted SS signal. To study the characteristics of the RAKE receiver, the multipath channel can be approximated as a tapped delay line. The RAKE receiver has a receiver finger for each multipath component. In each finger the received signal is correlated by a spreading code, which is time-aligned with the delay of the multipath signal. After dispreading, the signals are weighted and combined using the maximal ratio combining. Due to the mobility, the scattering environment will change and thus the delays and the attenuation factors will also change. Therefore, it is necessary to measure delay profile for the changing conditions and to reallocate the RAKE fingers whenever the delays have changed by a significant amount. A code-tracking loop takes care of small-scale changes, less than one chip, which tracks the time delay of each multipath signal.

2.2.2.3 Power Control

All users in a DS-CDMA system transmit the messages by using the same bandwidth at the same time and therefore users interfere with each other. Due to this kind of transmission mechanism, the signal received by the base station from a transmitter close to it will be stronger than the signal received from a transmitter located at the boundary region. Hence the close users will dominate the distant users. This is called the near-far effect. To combat this problem, all the users, irrespective of their distances, must be received at the base station with the same mean power. To achieve this, power control is done, which attempts to obtain a constant mean power for each user.

In contrast to the uplink, the base station transmits equally to all the users in the downlink. In the downlink all signals propagate through the same channel and thus are received by the mobile station with equal power. Hence here no power control is needed here. However power control is required to minimize interference with other cells and interference from other cells. There are two types of power control. They are (1) Open loop, which measures the interference from the channel and adjusts the transmitted power to achieve the required Bit Error Rate (BER) or Frame Error Rate (FER). (2) Closed loop power control performs better than the open loop. It measures the signal-to-interference (SIR) and sends commands to the transmitter on the other end to adjust the transmission power.

2.2.2.4 Soft Handover

Soft handover is a process in which a mobile station is connected to more than one base station simultaneously. This is used in CDMA systems to reduce interference with other cells and to improve performance. Hard handover is the process in which handover is done if the signal strength of the neighboring cell exceeds the signal strength of the current cell with a given threshold. But since in a CDMA system, frequency reuse concept is used, the neighboring cell frequencies are the same as that of the current cell frequencies. This might cause excessive interference in to the neighboring cell and hence a degradation in capacity and performance. Thus mostly the soft handover mechanism is used in CDMA.

2.2.2.5 Multiuser Detection

The capacity of a DS-CDMA system depends on the interference limit that the system can tolerate. Thus when a new user or interferer enters the network, other users quality of service will degrade. The more the network can resist the interference more users can be serviced. Multiuser detection, also called joint detection and interference cancellation (IC), provides means of reducing the effect of multiple access interference, and hence increases system capacity. In addition to this, Multiuser Detection also reduces the near/far problem by tracking the base station that blocks the whole cell traffic by using a high transmission power. If this user is detected first and removed from the input signal, the interference with other users is eliminated. Some of these detectors that are used in practice are decorrelator and linear minimum mean square error (LMMSE) detectors.

2.3 Random Multiple Access Scheme

The Random multiple access scheme is a method in which the users transmit information in the form of a packet over a common channel. This system is similar to the CDMA system in that it is a multiple access technology but here, the information signals of the users are not spread in frequency. That is this technology is based in the time domain, where, the users transmit the information when they have one or more packets to send. When more than one user attempts to transmit a packet, a collision occurs. That is, the packets from different users overlap with each other in time, thereby causing a loss of information. This conflict has to be resolved to achieve an efficient communication. There are several protocols that are developed to avoid packet collision in time. The prominent ones are, (1) ALOHA Systems and Protocols and (2) Carrier Sense Systems and Protocols. [37]

2.3.1 ALOHA Systems and Protocols

When a user transmits a packet and no other user transmits any information for the entire duration of that packet, then the packet is considered to be successfully transmitted. However, when other users transmit a packet at the same time as that of the first user, then a collision occurs. If there is a mechanism by which the user can know of the status of the packet that was transmitted, that is, whether the packet has collided or it has been successfully transmitted, then the efficiency of the transmission can be greatly increased. Channel Access Protocols are devised for this purpose that enables the user to retransmit a packet that has collided. The feedback to the users can be provided in a number of ways. In a radio type broadcast system, the feedback is given in the form of a broadcast, where a base station sends the packets to all the users in the downlink. The following information about whether no packet was transmitted, or a packet was transmitted successfully or there was a collision in the transmission can be given in the feedback. This type of feedback is called a (0,1,c) feedback.

There are two types of ALOHA systems. They are (1) Synchronized or slotted ALOHA and (2) Unsynchronized or unslotted ALOHA. In an unslotted ALOHA system, the user can transmit data in the form of packets at any time. Based on the feedback received, the sender makes the decision as to whether a retransmission is required or not. Whereas in a Slotted ALOHA system, the packets are transmitted in time slots. These slots have specified starting and ending times. The feedback mechanism is the same as above. The two systems are characterized based on their efficiency. It is found that a Slotted ALOHA system performs better than the Unslotted ALOHA system, under similar load conditions. The following figure shows the plot between the throughput versus the offered traffic for both the slotted and the unslotted ALOHA systems. The throughput is defined as the number of frames successfully transmitted in one second.

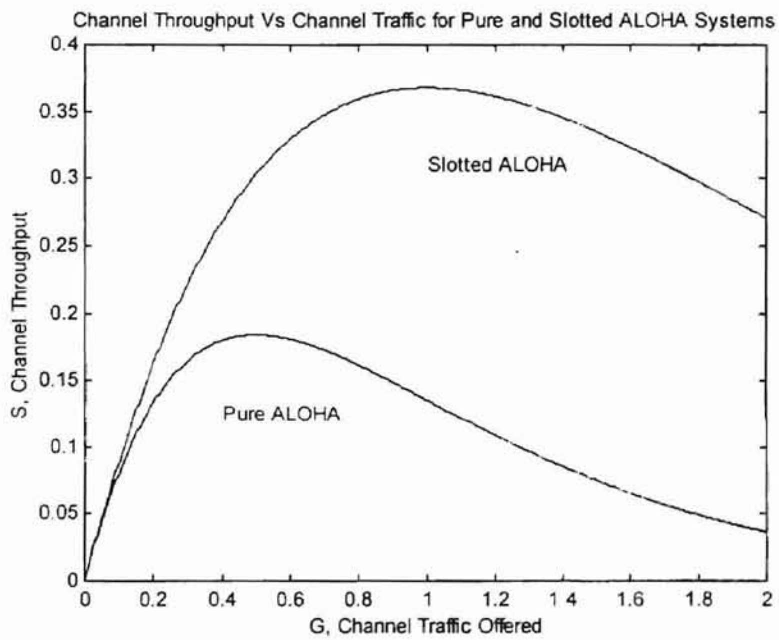

Fig. 2-8 Throughput Versus Offered Load for Pure and Slotted-ALOHA

The maximum efficiency of the Slotted ALOHA system is 36.8% and is achieved when on average, only one sender transmits in a time slot. The efficiency for the Unslotted ALOHA system is 18.4%. But this efficiency is insufficient for a multiple access system, hence some modifications are done to the above system to achieve a better throughput. We have discussed the ALOHA systems in the previous chapter in more detail.

2.3.2 Carrier Sense Systems and Protocols

In channel mediums, where the delays are very less, it is possible to design a random access protocol that can increase the throughput. One such a protocol is called a Carrier Sense Multiple Access with Collision Detection (CSMA/CD) protocol. Here the transmission mechanism is such that all the users in the system listen for any transmissions in the channel. A user, who needs to transmit information, seizes the channel when it senses that the channel is idle. This system does not ensure perfect collision avoidance as it suffers from the following problem. Assume that two users are separated by a maximum distance in the network. And let 'τ_d' be the propagation delay for the signal from one user to the other user. Then the maximum time required to sense an idle channel is τ_d. Suppose that the user 1 transmits a packet of duration T_p. Then it would take τ_d seconds after T_p for the user 2 to seize the channel and transmit. But user 1 would not know of this transmission until τ_d seconds after user 2 begins transmission. Hence the time interval $2\tau_d$ is defined as the maximum time interval to detect a collision. Whenever a collision occurs, a jam signal is transmitted to all users that serve to notify of the collision and abort their transmission. If it is assumed that the time to send a jam signal is negligible, the CSMA/CD protocol yields a high throughput when $2\tau_d << T_p$.

There are several protocols that may be used to reschedule transmissions whenever a collision occurs. They are

Non-persistent Strategy: In this protocol, a user that has a packet to be transmitted senses the channel and performs the following

(1) The user transmits the packet, if the channel is idle.

(2) If the channel is sensed busy, the user schedules the packet transmission at a later time, which is decided by some delay distribution. After the delay period, the user senses the channel repeats the above steps.

1-Persistent strategy: This protocol is designed to achieve a high throughput by not allowing the channel to go idle when some user has a packet to be sent.

(1) If the channel is idle, the user transmits the packet with a probability of one.

(2) If the channel is sensed to be busy, the user waits until the channel is idle and transmits a packet with a probability of one. This protocol has a high probability of collision.

P-Persistent Strategy: To decrease the collision rate in the 1-persistent protocol and to increase the throughput, this protocol follows the following procedures

(1) If the packet is to be transmitted and the channel is sensed to be idle, the user transmits the packet with the probability of 'p', and delays the transmission of the packet by τ with a probability of (1 - p), where p is selected in such a way that it reduces the probability of collisions while the idle period between the consecutive transmissions is kept small.

(2) If at time t = τ, the channel is sensed to be still idle, then the above step is performed. If a collision is detected, the users schedule a retransmission based on a certain delay distribution.

(3) If at time t = τ, the channel is sensed to be busy, the user waits until it is idle and then performs the above steps.

Exponential Back off: The transmission time is randomly selected in the exponential units of time like 1,2,4,8,16..,. That is each user after detecting a collision waits for an exponential amount of time before sending a retransmission.

2.4 IEEE 802.11a

This is a standard for Wireless LAN Medium Access Control and Physical Layer Specifications. The reason for discussing IEEE 802.11a standard at this point is because this standard uses the Orthogonal Frequency Division Multiplexing (OFDM) technique as the modulation technique for the physical layer in a wireless LAN. The RF wireless LAN system operates at the 5 GHz unlicensed band in the range of 5.15-5.25, 5.25-5.35 and 5.725-5.825 GHz. The OFDM at the physical layer provides a wireless LAN with a data payload communication capacity of 6, 9, 12, 18, 24, 36, 48, and 54 Mbits/s. According to [40], the system uses 52 orthogonal subcarriers that are modulated using one of the Phase Shift Keying (PSK) techniques like binary PSK (BPSK) or quadrature PSK (QPSK) or 16-quadrature amplitude modulation (QAM), or 64-QAM. The coding used at the physical layer is a Forward Error Correction coding using a convolutional code with the coding rate of $1/2$, $2/3$, or $3/4$.

The OFDM physical layer consists of two protocol functions. These are discussed briefly below.

(1) A PHY convergence function, which adapts the capabilities of the physical medium dependent (PMD) system to the PHY service. A layer procedure called the Physical Layer Convergence Procedure (PLCP), is supported by the physical layer, that defines a method of mapping the IEEE 802.11 PHY sublayer service data units (PSDU) into a frame format that is suitable for sending and receiving user data and management information between two or more stations, each using the associated PMD system.

(2) A PMD system whose function defines the characteristics and method of transmitting and receiving data through a wireless medium between two or more stations, each using the OFDM system.

The OFDM PHY physical layer contains three main functional entities: the PMD function, the PHY convergence function, and the layer management function. The PMD function provides a means to send and receive data between two or more stations. This layer is concerned with the 5 GHz band using OFDM modulation. And the PHY Management Entity (PLME) performs management of the local PHY functions along with the MAC management entity.

We will limit ourselves to a very brief overview of this standard. For more information on this material please refer to [40].

2.5 Multipath Fading

The radio environment is usually affected by the channel characteristic that leads to the fading of the signal. Any signal fading in the RF environment can be modeled as a large-scale path loss component together with a slow varying medium-scale component that can be modeled by a log-normal fading channel and a fast changing component that can be modeled by a Rician or a Rayleigh fading channel. Hence any wireless environment is analyzed with respect to these channel models.

Usually the received signal power decreases with distance due to the reflection, refraction and diffraction around structures. This phenomenon is called the path loss. Large-scale propagation models can be used to determine the path loss, as they characterize the received signal strength by averaging the power level over large separation distances between the transmitter and the receiver. There will be a change in the mean power of the received signal due to the movement of the receiver over distances of the order of tens or hundreds of meters. This phenomenon is called shadowing. Shadowing occurs due to the obstruction of the signal by trees or foliage. Medium-scale propagation models are used to determine this change in the received signal mean power. Also, over short distances of the order of a few wavelengths or over a short duration of time of the order of seconds, the signal strength may vary very fast. This is due to the multipath reflection of the signal by scatterers such as tall buildings, houses and other man-made objects or natural objects like forests etc. surrounding the mobile receiver. Small-scale models are used to approximate these conditions.

Rayleigh multipath fading channel is a severe radio environment channel in which the fading dips are very deep. In a Rayleigh fading channel, it is assumed that all multipath components are independent and there is no dominant path. The Rician fading channel has a distribution that characterizes a shallow fading dip channel. This assumes the presence of a dominant path in addition to the scattered multipath components. There are certain concepts related to the multipath channel. They are the delay spread, coherence bandwidth, Doppler spread, and coherence time. These concepts are used to model the various effects of the multipath channel. [33].

The root-mean-square (rms) value of the delay spread is a statistical measure that tells the spread of the multipaths about the mean delay of the channel. The maximum spread gives the delay difference between the first and the last multipath components in a power delay profile. The coherence bandwidth is defined as the maximum frequency difference for which the signals are still strongly correlated. The coherence bandwidth is inversely proportional to the delay spread. If the transmission bandwidth of the signal is greater than the coherence bandwidth, then the signal undergoes a frequency selective fading. But if the coherence bandwidth is larger than the transmission bandwidth, the signal experiences a uniform flat fading and the channel becomes a flat fading channel. The coherence bandwidth gives a measure of the diversity of the RAKE receiver or any equalized receiver. The smaller the coherence bandwidth is, higher is the order of diversity. The Doppler spread arises due to the movement of the mobile receiver. Doppler spread is measured as the width of the observed spectrum, when an unmodulated carrier wave is transmitted. The Doppler spread is zero, when there is only a single path from the source to the receiver, although there might be a slight shift in the carrier frequency called the Doppler frequency shift. The Doppler frequency varies depending on the angle of the mobile station movement also. The coherence time is defined as the reciprocal of the Doppler shift. Doppler spread can be measured as the range of values over which the Doppler power spectrum is non-zero.

2.5.1 Radio Environments and Distribution Functions

Usually the wireless systems are analyzed in three general type radio environments namely, (1) Vehicular radio environment (2) Outdoor to indoor and pedestrian radio environment and (3) Indoor office radio environment. These environments correspond respectively to the Macrocell, Microcell and Picocell. These cell structure analyses are each studied with respect to the large-scale propagation models that calculate the path loss component due to the large separation distances between the transmitter and the receiver, the medium-scale models that approximate the log-normal shadowing effects and the small-scale models that approximate the Rayleigh and Rician fading channels.

There are three main distribution functions that play an important role in these cell analyses. The log-normal probability density function is used to model the shadowing and the Rayleigh and Rician probability density functions are used to model the distribution of the instantaneous power in small-scale fading models.

The log-normal pdf is given by,

$$f_X(x) = \frac{1}{(x-a)\sigma\sqrt{2\pi}} \exp\left[-\frac{(\ln(x-a)-b)^2}{2\sigma^2}\right] \tag{2.10}$$

Where σ is the logarithmic standard deviation of the shadowing and the mean is given by $a + \exp(b + \sigma^2/2)$.

The Rician pdf is given by

$$f_R(r) = \frac{r}{p_0^{'}} \exp\left[-\frac{r^2+s^2}{2p_0^{'}}\right] I_0\left(\frac{r^2}{p_0^{'}}\right) \qquad \text{for } s \geq 0, r \geq 0 \qquad (2.11)$$

$$= 0 \qquad \text{for } r \leq 0 \qquad (2.12)$$

Where r is the amplitude of the received signal, $I_0(\ )$ is the modified Bessel function of the first kind and zeroth order, s is the peak value of the specular radio signal, $p_0^{'}$ is the average power of the signal, where $p_o^{'} = p_o/(k+1)$, p_o is the total received local-mean power, and k is the Rician factor, which is defined as the ratio of the average specular power and the average fading power received over specular paths, given by

$$k = \frac{s^2}{2p_o^{'}} \qquad (2.13)$$

When the direct signal, which is the dominant component of the multipath signal, does not exist, (i.e., when s is zero and hence k is zero) the above equation becomes a Rayleigh density function given by.

$$f_R(r) = \frac{r}{p_o^{'}} \exp\left[-\frac{r^2}{2p_o}\right] \qquad \text{for } r > 0 \qquad (2.14)$$

$$= 0 \qquad \text{otherwise.} \qquad (2.15)$$

2.6 Synchronization in OFDM Systems

The OFDM receiver has to perform the important task of synchronization of the received signal, before the OFDM carriers could be demodulated. First the receiver has to find out where the OFDM symbol boundaries are and what the optimal timing instants are, to minimize the effects of the intercarrier interference (ICI) and the intersymbol interference (ISI). Usually the single carrier systems are more robust against the phase noise and the frequency offsets than the multicarrier OFDM systems, as they only give degradation in the received SNR instead of causing Interchannel Interference. Also the carrier frequency offset if any, needs to be taken care of, because it can also cause ICI. Hence in this section we will discuss the effects of the phase, frequency and the timing offset on the performance of the OFDM receiver.

2.6.1 Phase Noise Sensitivity

Phase noise arises due to the non-ideality of the receiver oscillator. In [34] the received power density spectrum of the signal at the receiver along with the phase noise is modeled as a Lorentzian spectrum given by the squared magnitude of the first order low pass filter transfer function, reproduced here by,

$$S_S(f) = \frac{2/\pi f_l}{1 + f^2/f_l^2}, \tag{2.16}$$

Where f_l is the –3dB linewidth of the oscillator signal. For practical purposes, the double sided spectra are measured, the bands centered on the carrier frequency f_c. The bandwidth is doubled and hence the amplitude of the spectrum is divided by two in order to keep the total normalized to one.

$$S_S(f) = \frac{1/\pi f_l}{1 + |f - f_c|^2/f_l^2} \tag{2.17}$$

A change in phase leads to a change in the frequency. Hence one of the serious problems of the phase noise is that it causes ICI because the subcarriers in the OFDM symbol are no longer orthogonal i.e., the spacing between them is no longer exactly equal to $1/T$. One of the possible ways to reduce the effect of the phase noise is to employ a Phase Locked Loop (PLL) at the receiver oscillator to generate a stable carrier frequency with little or no phase changes.

2.6.2 Frequency Offset Sensitivity

We know that for the OFDM subcarriers to be orthogonal, there should be an integer number of cycles difference between the adjacent subcarriers with in an FFT window. A shift in the frequency means that the number of cycles in the FFT window is not an integer anymore. This leads to ICI after the FFT operation. The FFT output for each subcarrier has interference from the adjacent subcarriers. The interference will be greater for the subcarriers in the middle of the symbol because of the interference from subcarriers on either of the sides, than for those at the end of the OFDM symbol. Obviously, the interfering components have power that is inversely proportional to the frequency spacing between the carriers. That is, greater the spacing between carriers lesser is the interference. Pollet et.al have derived the expression for the degradation in the received SNR, due to the frequency offset. It is given by,

$$D_{freq} \cong \frac{10}{3\ln 10}(\pi \Delta f T)^2 \frac{E_S}{N_O}, \tag{2.18}$$

Where Δf is the frequency offset and T is the OFDM symbol duration, and E_S / N_O is the signal energy to the noise power ratio.

A lot of frequency synchronization techniques are developed to reduce the frequency offset at the receiver. These operations are carried out before the FFT block at the receiver to reduce the ICI. We will be discussing about more of these techniques in the next chapter.

2.6.3 Timing Offset Sensitivity

We just briefly discussed about how an offset in the phase or frequency can give rise to Intercarrier Interference (ICI). However, OFDM systems are more robust against the timing offset problems. This is because there exists a guard interval in the OFDM symbol that enables to reduce the interference between adjacent symbols. The symbol timing offset can vary over an interval equal to the guard interval without causing any Intercarrier Interference (ICI) or Intersymbol Interference (ISI). Also the OFDM receiver is robust against multipath interference as long as the multipath components arrive within the guard interval.

It is worthwhile to notice that as the OFDM symbol timing changes, the phase of the demodulated subcarriers also change. The relation between the phase ϕ_i of the subcarrier i and the timing offset τ is given by

$$\phi_i = 2\pi f_i \tau \qquad (2.19)$$

Where f_i is the frequency of the i^{th} subcarrier before sampling. Hence we can observe that even a small change in the time offset gives rise to a large phase shift between the subcarriers. Also an error can occur in the sampling frequency that might lead to ICI or can produce time-varying phase changes. Usually channel estimation is done at the coherent OFDM receiver to estimate these phase changes and equalize them. More techniques that deal with these offset problems are discussed in detail in the next chapter.

CHAPTER III

OFDM Frame Synchronization

This chapter deals with the synchronization issues associated with the OFDM systems. The OFDM systems are very sensitive to the synchronization errors and hence considerable research is being conducted of late to take care of this problem. Hence this chapter discusses the frame synchronization issues.

3.1 Symbol Synchronization

3.1.1 Timing Errors

The use of Cyclic Prefix in OFDM systems has relaxed the timing requirements related to the OFDM symbol synchronization. The aim is to determine the starting point of the OFDM symbol. A timing offset in the symbol gives rise to a phase rotation of the subcarriers. The orthogonality of the OFDM systems is maintained if the timing offset is small enough to keep the impulse response of the channel to within the cyclic prefix. The channel estimator can estimate the phase rotations here. However if the timing offset error is longer than the cyclic prefix, then ISI will occur.

Basically there are two ways to do timing synchronization. They are (1) by using a Pilot symbol or (2) by using a Cyclic Prefix. Warner and Leung [28] have suggested a scheme in which the transmitter encodes a number of subchannels (reserved) with known phases and amplitudes. There are three phases in their algorithm namely the Power detection, coarse synchronization and fine synchronization. The power detection phase detects the presence or absence of an OFDM symbol by comparing it to a threshold value. The coarse synchronization phase is used to obtain timing alignments to within a half a sample of duration. Even though this phase simplifies the symbol tracking procedure, the performance of this is not acceptable. Normally correlation is done between this received signal and a copy of the transmitted pilot synchronization symbol. The correlation peak determines the coarse estimate of the symbol starting position. After this, the fine synchronization is done by equalizing the subchannels with the pilots with the channel estimate obtained from the pilots.

The techniques based on the Cyclic Prefix compute the difference between the received samples that are spaced N samples apart, where N is the number of samples in the OFDM symbol. When one of the samples in the difference belongs to the cyclic prefix part and the other belongs to the OFDM symbol part from which it was formed, then the difference should be small. When the samples are from different symbols then the difference will be larger on average. By using a window of length equal to that of this cyclic prefix, to window the difference between the samples, the new OFDM symbol starting position can be estimated by looking for the minimum of the difference.

Timing synchronization in the uplink direction is more difficult than in the downlink direction or during broadcasting. This is mainly due to separate timing offsets for each user. The use of a random access scheme to synchronize the mobile and the base station is also considered in some of the researches.

3.1.2 Carrier Phase Noise

This is due to the imperfections in the clocks of the transmitter and the receiver oscillators. An analysis of the OFDM system in the presence of the carrier phase offset was done by [22]. The phase offset here is modeled as a Weiner process $\theta(t)$ with a mean of zero and a variance of $4\pi\beta|t|$, where β (in Hz) denotes the one-sided 3 dB linewidth of the Lorentzian power spectrum of the carrier generator. The SNR loss, i.e., the additional SNR needed to compensate for the error, can be approximated as

$$D(dB) = \frac{11}{6\ln 10}\left(4\pi N \frac{\beta}{W}\right)\frac{E_S}{N_O} \tag{3.1}$$

Where W is the bandwidth and E_S / N_O is the per-symbol SNR. As the number of subcarriers is increased, the degradation increases. The analysis for the single carrier and the multicarrier system as a function of the carrier phase noise is given in the above figure.

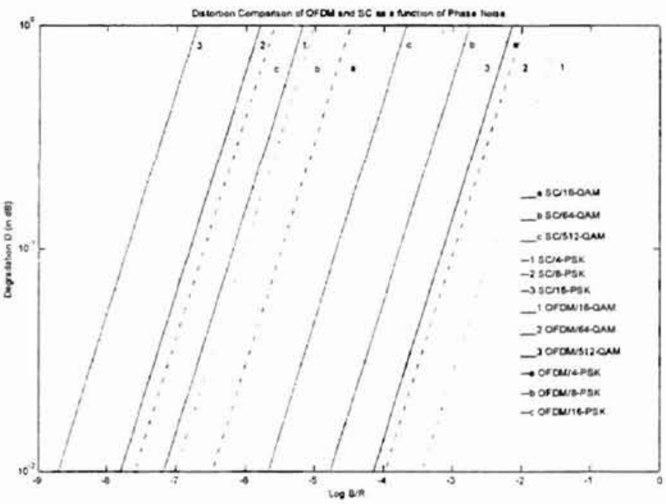

Fig. 3-1 Degradation as a function of Phase Noise

3.2 Sampling - frequency Synchronization

The received signal is continuous in nature. The receiver clock determines the sampling instants of the received signal. The receiver clock should be in alignment with the transmitter clock in order to prevent a clock frequency offset. There are two methods to achieve this. In synchronized sampling systems, a timing algorithm is used to control the VCO in order to align the receiver and the transmitter clocks. The other method is by non-synchronized sampling, where the sampling rate is fixed. But this requires some post-processing in the digital domain. The frequency offset leads to the signal component getting rotated and attenuated. Also this leads to the problem of ICI. The non-synchronized sampling systems are much more sensitive to the sampling frequency offset compared to the synchronized sampling systems.

3.3 Carrier Frequency Synchronization

3.3.1 Frequency Errors

The frequency offsets are generated by Doppler shifts, phase noise or due to the difference in the clock oscillators in the transmitters and the receivers. The carrier frequency offsets cause the reduction of the signal amplitude and introduction of ICI from other channels due to the loss of orthogonality. In [22] the degradation of the BER due to the presence of the carrier frequency offset in an AWGN channel is analyzed. It is found that the multicarrier system is much more sensitive to the single carrier system. The frequency offset, normalized by the subcarrier spacing is given by $\Delta f = \dfrac{\Delta F}{W / N}$, where ΔF is the frequency offset and N is the number of subcarriers. The degradation in SNR can then be given as

$$D(dB) = \frac{10}{3\ln 10}\left(\pi \frac{N.\Delta F}{W}\right)^2 \frac{E_S}{N_O} \tag{3.2}$$

Note that, in this case the degradation increases with the square of the number of the subcarriers.

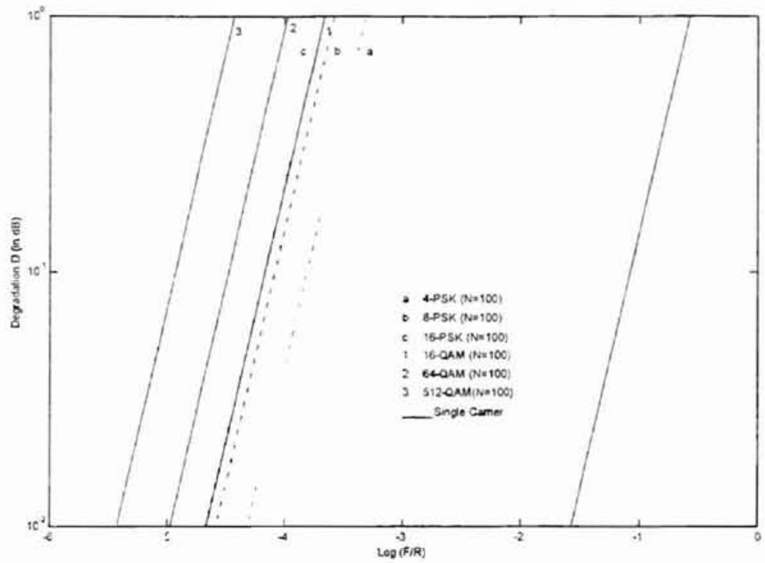

Fig.3-2 Degradation as a function of Frequency Offset

The analysis shows the effect of frequency offset on both the single carrier and the multicarrier system. The multicarrier systems are found to be more sensitive to the offset in the frequency.

3.3.2 Frequency Estimators

There are numerous research material in literature as far as carrier frequency synchronization schemes are concerned. The symbol synchronization schemes can be divided in to two categories. They are based on pilot carriers or on the cyclic prefix. The following discussion provides a short overview of these techniques.

In pilot-aided algorithms, the some of the reserved subcarriers are used for the transmission of the pilots. With the knowledge of these symbols, the phase rotations caused by the frequency offsets can be determined. There is a one-to-one relation between the frequency offset and the phase rotation, under the assumption that the frequency offset is less than half the subcarrier spacing. For this to happen, acquisition algorithms should be developed. [38] has developed such an algorithm by forming a function which is sinc-shaped and has a peak for the value of $f-\hat{f}=0$. The acquisition could be obtained by maximizing the function by solving the function at points that are $0.1/T$ apart.

The cyclic prefix has the redundant information from the OFDM symbol. And this redundancy could be used in several ways, i.e., by developing a function that peaks at zero offset and determining the maximizing value or by doing a maximum-likelihood estimation (ML).

It is worthwhile to notice the relationship between the timing and the frequency synchronization. If the frequency synchronization is the problem, then it can be reduced by reducing the number of the subcarriers that form the OFDM symbol. Lowering the number of subcarriers increase the subcarrier spacing. This will however place a demand on the timing synchronization, due to the reason that the OFDM symbol length gets shorter. This will lead to the occurrence of a relatively large timing error. Thus the synchronization of the OFDM symbol in the time and frequency domains are closely related to each other.

CHAPTER IV

OFDM IN PACKET CAPTURED SLOTTED ALOHA

This chapter focuses on the advantages of using OFDM in Slotted ALOHA systems. The main issues concerning packet capture and throughput, SNR gain and easy synchronization are discussed in this chapter. The proposed system combination of Slotted-ALOHA with packet capture with OFDM modulation technique has three distinct advantages, they are

(1) The proposed system uses less overhead compared to [30] that uses a supplementary guard interval in each of the first OFDM symbol of a burst data frame apart from a normal guard interval.

(2) More throughput compared to the system used in [28] that used a Pure ALOHA with the OFDM Modulation technique.

(3) The proposed system model also improves the OFDM frame synchronization because it utilizes the inherent timing mechanism of the Slotted-ALOHA system.

The first advantage of this system is quantized in terms of a Signal to Noise ratio loss calculations in dB, comparing the OFDM frame structure of the Slotted-ALOHA system with that of the OFDM frame structure proposed by the [30]. The second advantage listed above is quantized in terms of a throughput versus offered Channel traffic comparing the Slotted-ALOHA system with that of the classical or Pure ALOHA system and with the Carrier Sense Multiple Access (CSMA) system, which is the fundamental access technique for the IEEE 802.11 industry standard. This chapter discusses in detail about the comparison of the SNR loss in both the systems. The next chapter discusses in detail about the throughput performances of the Slotted-ALOHA system, Pure ALOHA system and the CSMA system. The topic concerning the OFDM frame synchronization in Slotted ALOHA multiple access technology is explored as a potential topic for future research.

4.1 SNR Comparison for OFDM systems

This section discusses the loss in the Signal to Noise Ratio (SNR) due to the inclusion of the Cyclic Prefix in the OFDM symbol. The frame structure of the OFDM symbol used in our proposed system model is given in the following figure.

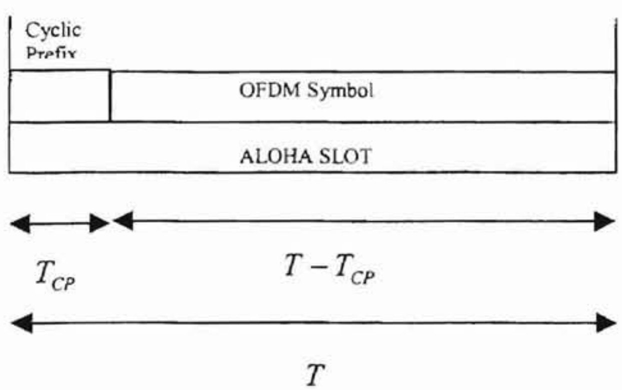

Fig. 4-1 OFDM Symbol with Cyclic Prefix

The symbol is encompassed in an ALOHA slot. There may be more symbols in one slot. But for simplicity, we assume one OFDM symbol in one ALOHA slot. The duration of the OFDM symbol is given by T seconds of which T_{CP} forms the cyclic prefix part.

The OFDM symbol according to [30] is given by the following figure.

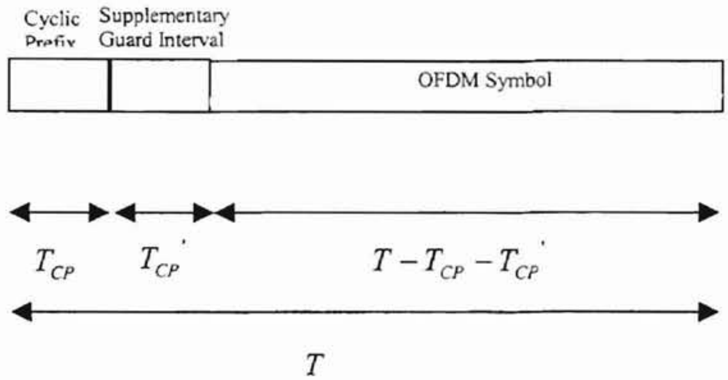

Fig. 4-2 OFDM Burst Frame Format

The above frame format is different from the previous format in that a supplementary guard interval is used here for doing the coarse frequency synchronization. The SNR loss due to the usage of cyclic prefix can be quantized as follows.

From [36], the relationship between the SNR loss and the cyclic prefix can be given as

$$SNR_{loss} = -10\log_{10}(1-\gamma) \tag{4.1}$$

Where γ is given by $\gamma = T_{CP}/T$, and is the relative length of the cyclic prefix. The longer the duration of the cyclic prefix, larger the SNR loss. This loss as we calculate and show, is better for the frame format used by our proposed OFDM over Slotted-ALOHA system.

For the format without the supplementary guard interval, the SNR loss can be written as

$$SNR_{loss} = 10\log_{10}(T/(T-T_{CP})) \tag{4.2}$$

For the system used in [30], the SNR loss can be written as

$$SNR_{loss} = 10\log_{10}\left(\frac{T}{T - T_{CP} - T_{CP}'}\right) \tag{4.3}$$

Where T_{CP}' is the duration of the supplementary guard interval. The performance of these two OFDM systems under a practical environment is considered here for analysis. The choice of various OFDM parameters basically depends on three main requirements. They are the bandwidth, bit rate and the delay spread of the medium. The choice of the channel delay spread directly influences the choice of the OFDM guard interval. As a rule, usually the OFDM guard interval should be at least two to four times the duration of the estimated channel delay spread to take in to account the effect of the multipath channels.

Once the value of the guard interval is fixed, the symbol duration can be fixed. In order to minimize the signal to noise ratio loss caused by the inclusion of the redundant cyclic prefix, it is desirable to have the symbol duration to be very large compared to that of the guard interval. However the length of the OFDM symbol cannot be arbitrarily large compared to that of the guard interval because larger symbol duration means more number of subcarriers with small subcarrier spacing, a larger implementation complexity and more sensitivity to timing and frequency offsets. Hence a practical choice of the symbol duration would be to select a length that is at least five times the guard symbol duration.

We now show the performance analysis of the two systems with different frame formats here. A practical value of 250 nanoseconds is assumed for the channel delay spread.

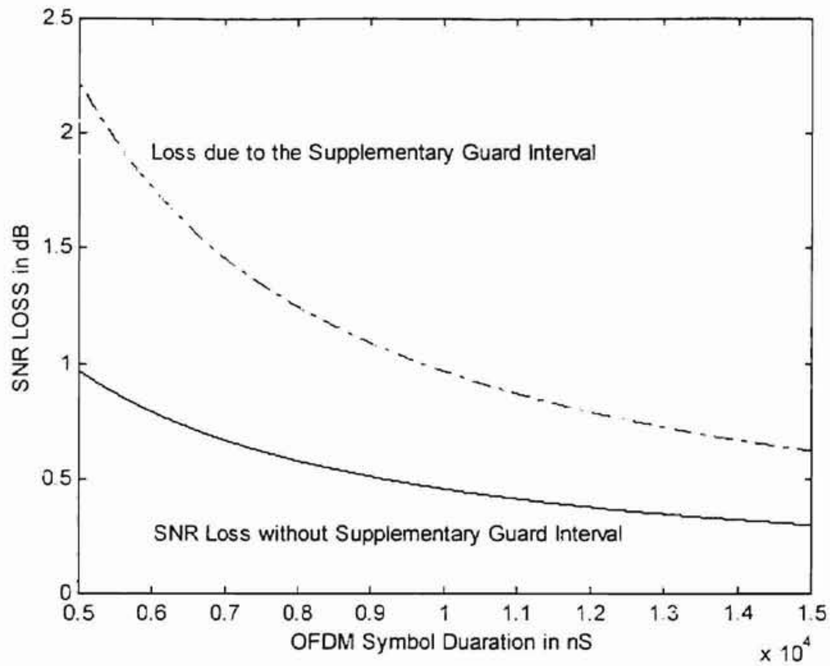

Fig. 4-3 Performance Comparison of the SNR loss for the OFDM-Slotted ALOHA System and the system in [30]

It can be seen clearly from the performance graphs that the OFDM frame format using a supplementary guard interval has twice as much loss compared to the frame format used in our proposed OFDM-Slotted ALOHA system with packet capture. Also apart from the SNR loss, the system in [30] suffers from the coarse synchronization failure problems, which is not present in our system. Because the OFDM- Slotted ALOHA system uses the inherent timing mechanism of the slotted aloha system to estimate the starting point of the OFDM symbol. Because here, the OFDM symbol almost always starts at the beginning of the ALOHA slot. After this, our system uses a fine synchronization technique to exactly determine the starting point of the OFDM symbol. The method of performing this frame synchronization and determining the exact symbol starting position can be explored as a future research topic from this thesis.

The performance of the proposed OFDM-Slotted ALOHA system in terms of the channel utilization is analyzed in detail in the next chapter. The technique of packet capture is employed to improve the throughput performances.

CHAPTER V

MULTIPLE ACCESS PROTOCOLS AND THEIR THROUGHPUT ANALYSIS RESULTS AND OBSERVATION

5.1 Channel Capacity for ALOHA Systems

The channel capacity of the ALOHA system can be analyzed as follows. There are two types of errors that occur when a packet is transmitted and it is received incorrectly or lost in transit. They are (1) Random noise errors and (2) Packet Overlap errors. Due to the random nature of the type one error, we focus on the type 2 errors to improve the performance of the system. Before that, some of the basic results relating the throughput and channel traffic are derived.

The analysis starts with the assumption that the packet starting times approximate a discrete Poisson random process. Let the parameter λ be the rate at which the packets are sent in packets/seconds. If the duration of the packets is given by τ seconds, the normalized channel traffic is given by [1]

$$G = \lambda \tau \tag{5.1}$$

If it is assumed that the packets that are received correctly are those packets that do not overlap with any other packet in the system, then the normalized channel throughput is given by

$$S = \lambda' \tau \tag{5.2}$$

Where $\lambda' < \lambda$, is defined as the rate of occurrence of the packets that are received correctly at the receiver.

The probability that the packets do not overlap is based on the probability that no packets are transmitted τ seconds sooner or later after the first packet is sent. From the Poisson distribution, this probability is given by

$$e^{-2\lambda\tau} \text{ Or } e^{-2G} \tag{5.3}$$

Hence the throughput accordingly is given by,

$$S = Ge^{-2G} \tag{5.4}$$

5.2 Slotted ALOHA Systems

The unsynchronized ALOHA systems can be modified in to a more of an organized multiple access system so that the throughput can be increased to a considerable extent. In a pure ALOHA system, the users transmit the packets as soon as they are ready to transmit, without any coordination among other users of the system. This method leads to inefficiency of channel utilization. So an improved version of ALOHA called a Slotted ALOHA system is proposed where each user starts transmitting the data at fixed instants of time. This method achieves the improvement in the channel throughput, as we will see later. In this slotted ALOHA channel, a central clock sets up a time base for a sequence of slots that are of same duration as that of the packet transmission. That is each slot is of the exact duration as the length of each packet. Once the clock releases a set of slots, if a user has some packets to be sent, he synchronizes the start of the transmission of the packets to the start of a slot. Hence if there is any overlap, that is, if two users transmit at the start of the slot, then they will completely overlap for the duration of the slot or slots. Thus there is no partial overlap of data. The slotted ALOHA channel capacity can be given as follows from [1].

$$S = G\left(1-\frac{G}{n}\right)^{n-1} \tag{5.5}$$

And as $n \to \infty$, we get

$$S = Ge^{-G} \tag{5.6}$$

It can be shown that the throughput of the slotted ALOHA system reaches a maximum capacity of $1/e = 0.368$, which is twice as much as the pure ALOHA system.

5.3 Carrier Sense Systems and Protocols (CSMA)

In channel mediums, where the delays are very less, it is possible to design a random access protocol that can increase the throughput. One such a protocol is called a Carrier Sense Multiple Access with Collision Detection (CSMA/CD) protocol. Here the transmission mechanism is such that all the users in the system listen for any transmissions in the channel. A user, who needs to transmit information, seizes the channel when it senses that the channel is idle. This system does not ensure perfect collision avoidance as it suffers from the following problem. Assume that two users are separated by a maximum distance in the network. And let 'τ_d' be the propagation delay for the signal from one user to the other user. Then the maximum time required to sense an idle channel is τ_d. Suppose that the user 1 transmits a packet of duration T_p. Then it would take τ_d seconds after T_p for the user 2 to seize the channel and transmit. But user 1 would not know of this transmission until τ_d seconds after user 2 begins transmission.

Hence the time interval $2\tau_d$ is defined as the maximum time interval to detect a collision. Whenever a collision occurs, a jam signal is transmitted to all users that serve to notify of the collision and abort their transmission. If it is assumed that the time to send a jam signal is negligible, the CSMA/CD protocol yields a high throughput when $2\tau_d << T_p$.

There are several protocols that may be used to reschedule transmissions whenever a collision occurs. They are

Non-persistent Strategy: In this protocol, a user that has a packet to be transmitted senses the channel and performs the following

(1) The user transmits the packet, if the channel is idle.

(2) If the channel is sensed busy, the user schedules the packet transmission at a later time, which is decided by some delay distribution. After the delay period, the user senses the channel repeats the above steps.

1-Persistent strategy: This protocol is designed to achieve a high throughput by not allowing the channel to go idle when some user has a packet to be sent.

(1) If the channel is idle, the user transmits the packet with a probability of one.

(2) If the channel is sensed to be busy, the user waits until the channel is idle and transmits a packet with a probability of one. This protocol has a high probability of collision.

P-Persistent Strategy: To decrease the collision rate in the 1-persistent protocol and to increase the throughput, this protocol follows the following procedures

(1) If the packet is to be transmitted and the channel is sensed to be idle, the user transmits the packet with the probability of 'p', and delays the transmission of the packet by τ with a probability of (1 - p), where p is selected in such a way that it

reduces the probability of collisions while the idle period between the consecutive transmissions is kept small.

(2) If at time t = τ , the channel is sensed to be still idle, then the above step is performed. If a collision is detected, the users schedule a retransmission based on a certain delay distribution.

(3) If at time t = τ , the channel is sensed to be busy, the user waits until it is idle and then performs the above steps.

Exponential Back off: The transmission time is randomly selected in the exponential units of time like 1,2,4,8,16. That is each user after detecting a collision waits for an exponential amount of time before sending a retransmission.

5.4 Packet Capture and its Effect on the Throughput

In the packet broadcasting networks using the ALOHA protocols a collision is said to occur when two or more packets are said to be transmitted concurrently. The classical model of the ALOHA systems assume that when such a collision occurs, the packets are always discarded. The throughput of the system in such a case suffers very badly. But in the presence of a fading channel medium, the packets coming from different transmitters have different fading characteristics and the receiver can still extract the packet with a higher power than the others without discarding all the transmissions. This technique of packet capture greatly increases the throughput of the system. The following analysis considers this packet capture effect on the throughput for both the Slotted and

Pure ALOHA systems and is compared to the CSMA systems that are widely used in the commercial IEEE 802.11 standards.

5.4.1 Slotted ALOHA Analysis

The analysis starts with the assumption of a Poisson-distributed offered load to the channel. The average offered load in packets per unit time slot is given by G. Let the probability that there are n packets in one slot be represented as $P(n)$ and is given by

$$P(n) = \frac{G^n}{n!}e^{-G} \tag{5.7}$$

Let us introduce the packet capture effect by assuming that a packet received at the receiver with a power P_c will be detected and demodulated by the receiver if the ratio of the powers

$$\frac{P_c}{P_t} \geq \Gamma_n \tag{5.8}$$

Where P_t is the total sum of the powers of all the concurrently received packets in the same slot and Γ_n is the threshold value called the packet capture ratio. The value of Γ_n depends on the type of coding and modulation characteristics.

Then the probability of correct capture of the subject packet is defined as

$P_{capture} = P($ subject packet is received with n other packets concurrently) AND

$$P\left(\frac{P_c}{P_t} \geq \Gamma_n\right) \tag{5.9}$$

That is

$$P_{capture} = \sum_{n=1}^{\infty} P(n) * P\left(\frac{P_c}{P_t} \geq \Gamma_n\right) \tag{5.10}$$

$$P_{capture} = 1 - \sum_{n=1}^{\infty} P(n) P\left(\frac{P_c}{P_t} < \Gamma_n\right) \tag{5.11}$$

Substituting for the value of $P(n)$, we get the packet capture probability as

$$P_{capture} = 1 - \sum_{n=1}^{\infty} \frac{G^n}{n!} e^{-G} P\left(\frac{P_c}{P_t} < \Gamma_n\right) \tag{5.12}$$

The throughput in terms of the packet capture probability can be quantized for Slotted ALOHA as [33]

$$S' = GP_{capture} \tag{5.13}$$

Substituting the value of the capture probability in the above equation, we get,

$$S' = G.\left[1 - \sum_{n=1}^{\infty} \frac{G^n}{n!} e^{-G} P\left(\frac{P_c}{P_t} < \Gamma_n\right)\right] \tag{5.14}$$

$$S' = G.\left[1 - \sum_{n=1}^{\infty} \frac{G^n}{n!} e^{-G} F_{\Gamma_n}(\Gamma(n))\right] \tag{5.15}$$

Where $F_{\Gamma_n}(\Gamma(n))$ is the Cumulative distribution function and is given by, for the coherent addition of the interfering signals as [33]

$$F_{\Gamma_n}(\Gamma(n)) = \frac{n\Gamma_n}{n\Gamma_n + 1} \tag{5.16}$$

This increase in throughput for the case of Slotted ALOHA with packet capture can be written in terms of throughput for Slotted ALOHA without packet capture as

$$S' = S + \Delta S \tag{5.16}$$

$$S \quad = Ge^{-G\Gamma_\infty/(\Gamma_\infty+1)} \tag{5.17}$$

Where S is the throughput of the classical infinite user model Slotted ALOHA when $\Gamma_\infty \to \infty$ and ΔS is the increase in throughput due to the packet capture.

The value of the increase in throughput can be determined for the coherent addition of the interfering signals as follows [32]

$$\Delta S = Ge^{-G}\sum_{n=1}^{\infty}\frac{G^n}{n!}F_{\Gamma_n}(\Gamma(\infty)) - Ge^{-G}\sum_{n=1}^{\infty}\frac{G^n}{n!}F_{\Gamma_n}(\Gamma(n)) \tag{5.18}$$

$$= Ge^{-G}\sum_{n=1}^{\infty}\frac{G^n}{n!}\left[F_{\Gamma_n}(\Gamma(\infty)) - F_{\Gamma_n}(\Gamma(n))\ \right] \tag{5.19}$$

For a finite number of interferers, the value of ΔS becomes

$$\Delta S = Ge^{-G}\sum_{n=1}^{N}\frac{G^n}{n!}\left[\frac{n\Gamma_\infty}{n\Gamma_\infty+1} - \frac{n\Gamma_n}{n\Gamma_n+1}\right] \tag{5.20}$$

Now substituting the value of ΔS in S' we get the increase in throughput due to the packet capture in slotted ALOHA as

$$S' = Ge^{-\frac{G\Gamma_\infty}{(\Gamma_\infty+1)}} + Ge^{-G}\sum_{n=1}^{N}\frac{G^n}{n!}\left[\frac{n\Gamma_\infty}{n\Gamma_\infty+1} - \frac{n\Gamma_n}{n\Gamma_n+1}\right] \tag{5.21}$$

The Values of n, Γ_∞, Γ_n are taken from [42] for the performance analysis.

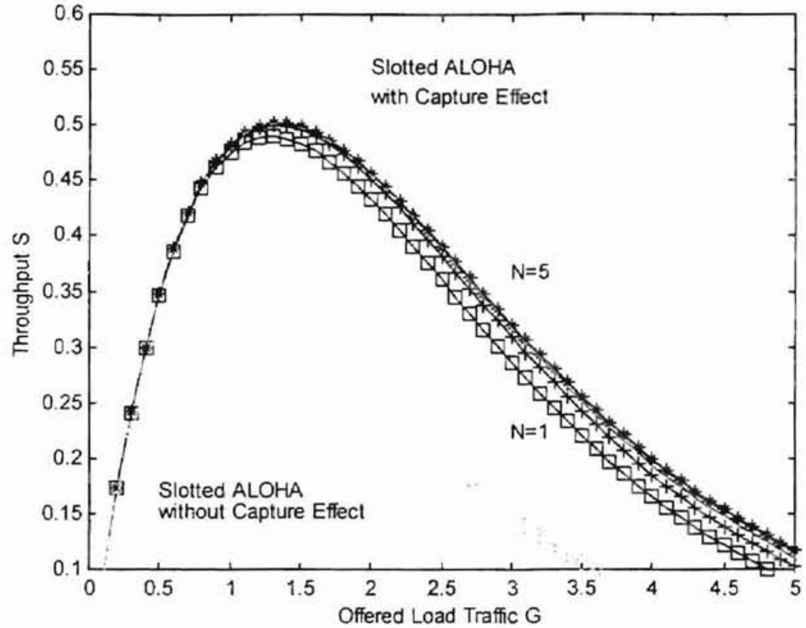

Fig.5-1 Throughput as a function of Offered Traffic with and without Packet Capture for Slotted-ALOHA System.

5.4.2 Pure ALOHA Throughput Analysis

For Pure ALOHA System, using the same analysis, we can find out the increase in the throughput for the system with packet capture enabled as follows

$$S' = Ge^{-2G(\Gamma_\infty/\Gamma_\infty+1)} + Ge^{-2G}\sum_{n=1}^{\infty}\frac{G^n}{n!}\left[\frac{n\Gamma_\infty}{n\Gamma_\infty+1} - \frac{n\Gamma_n}{n\Gamma_n+1}\right] \tag{5.22}$$

The Values of n , Γ_∞, Γ_n are taken from [42] for the performance analysis.

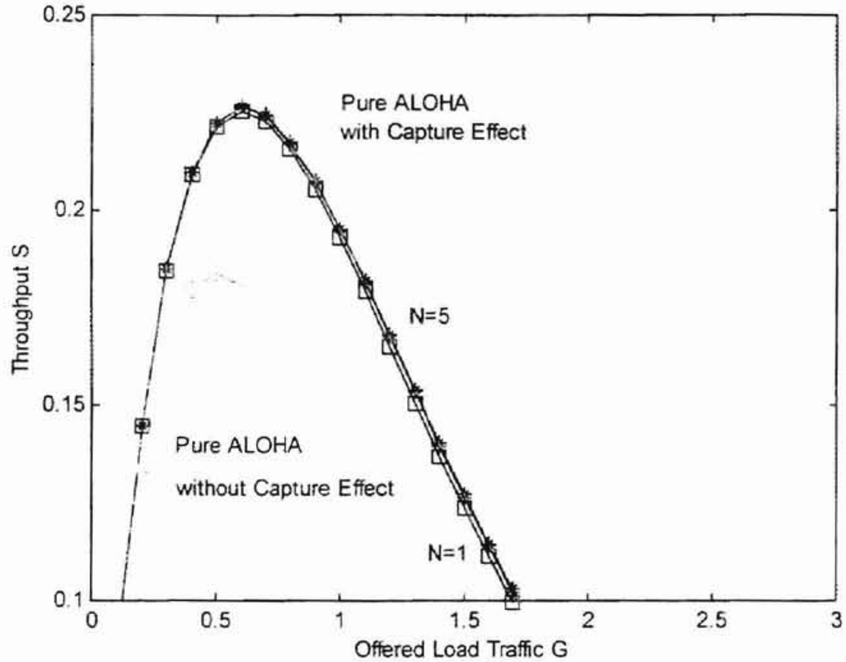

Fig. 5-2 Throughput as a function of Offered Traffic with and without Packet Capture for Pure ALOHA System.

5.4.3 CSMA Throughput Analysis

This is the fundamental multiple access technique for the IEEE 802.11 original standard. The IEEE 802.11 standard uses the Carrier Sensing mechanism based on a Back-off time algorithm. The transmitter senses the carrier to be either busy or idle. If the channel is sensed to be busy, the transmitter will defer the transmission to until a DCF IFS (D Inter Frame Space) period. Also if the channel is sensed to be idle, the transmitter will defer the transmission to until an EIFS (Extended Inter Frame Space) period. After the DIFS or EIFS period, the transmitter will still defer the transmission for a period equal to the Back-off time.

$$\text{Back-off Time} = \text{Random ()} * \text{a slot time} \tag{5.23}$$

Where Random () is a pseudo-random integer drawn from a uniform distribution over the interval [0, CW], where CW is the Contention Window ($CWmin \leq CW \leq CWmax$).

The performance of the IEEE 802.11 CSMA is similar to that of the Non-persistent CSMA, [35] where a user that generated a packet and found that the medium to be busy, restricts itself from transmitting the packet. Then the packet transfer is scheduled for a future time. This time is chosen randomly.

The parameter of interest in the CSMA technique is denoted by 'a', and is defined as

$$a = \text{Maximum Propagation Delay in the System } (\tau) \text{ / Packet Length } (T)$$

The throughput for the non-persistent CSMA in terms of the offered load can be given by

$$S = \frac{Ge^{-aG}}{G(1+2a)+e^{-aG}} \tag{5.24}$$

The plot for different values of the parameter 'a' for the non-persistent CSMA is shown below.

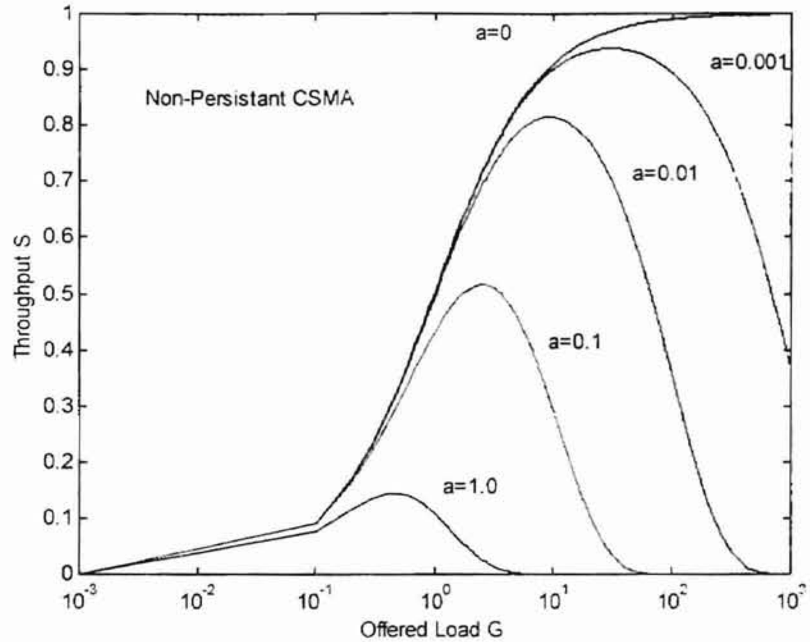

Fig. 5-3 Throughput-Load Characteristics of Non-Persistent CSMA

5.5 Numerical Results and Conclusion

Figure 5.1 shows the throughput-load characteristics of the Slotted-ALOHA systems with packet capture in case of a coherent addition of Rayleigh-fading interference signals. The plot shows that the throughput performances of the Slotted-ALOHA with packet capture far exceed the performance of the Slotted-ALOHA system without packet capture under similar load conditions. Figure 5.2 shows that the Pure ALOHA systems with capture also perform better than the classical ALOHA system. The performance of the CSMA system, which is the fundamental access technique for the IEEE 802.11, is shown in the figure 5.3. Here we can see that under similar load conditions the performance of the Slotted-ALOHA systems with packet capture is better than the CSMA system. But for higher loads, the CSMA performance exceeds the

Slotted-ALOHA system. The following table lists out the channel utilization for different Multiple access techniques.

	Multiple Access Protocols	Channel Utilization
1.	Pure ALOHA	18%
2.	Slotted- ALOHA	36%
3.	Pure ALOHA with Packet Capture	23%
4.	Slotted-ALOHA With Packet Capture	50%
5.	CSMA	~50% At $a = 0.1$

Table 5-1 Channel Utilization Comparison for different Channel Access Techniques

From the table, we can see that the channel access utilization for the classical Slotted-ALOHA system is almost twice that of the classical Pure ALOHA system. Generally, by introducing the packet capture effect, the channel access utilization for both the ALOHA protocols is increased. Particularly, the Slotted-ALOHA utilization is more than twice the performance of the Pure ALOHA system and almost equals the performance of the CSMA multiple access technique that is used in the widely popular IEEE 802.11 industry standard.

CHAPTER VI

CONCLUSION AND FUTURE RESEARCH

This OFDM-Slotted ALOHA system with Packet Capture discussed about three distinct features namely the OFDM modulation technique, the slotted ALOHA system with packet capture and the synchronization issues related to the combination of the two systems. The distinct advantages of using such a combination of systems is listed as follows

(1) The proposed system uses less overhead compared to [30] that uses a supplementary guard interval in each of the first OFDM symbol of a burst data frame apart from a normal guard interval.

(2) More throughput compared to the system used in [28] that used a Pure ALOHA with the OFDM Modulation technique.

(3) The proposed system model also improves the OFDM frame synchronization because it utilizes the inherent timing mechanism of the Slotted-ALOHA system.

The first advantage of this system was quantized in terms of a Signal to Noise ratio loss calculations in dB, comparing the OFDM frame structure of the Slotted-ALOHA system with that of the OFDM frame structure proposed by the [30]. The second advantage listed above was quantized in terms of a throughput versus offered Channel traffic comparing the Slotted-ALOHA system with that of the classical or Pure ALOHA system and with the Carrier Sense Multiple Access (CSMA) system, which is the fundamental access technique for the IEEE 802.11 industry standard. Chapter 4 discussed in detail about the comparison of the SNR loss in both the systems. Chapter 5 discussed in detail about the throughput performances of the Slotted-ALOHA system, Pure ALOHA system and the CSMA system.

The topic concerning the OFDM frame synchronization in Slotted ALOHA multiple access technology is explored as a potential topic for future research. Numerous background researches have been conducted on this topic and were presented in this thesis work as a chapter. Basically the frame synchronization issues can be resolved in to two categories namely frequency domain synchronization and time domain synchronization. Various authors have addressed these issues separately by assuming either frequency or time domain synchronization to be present in the system.

This topic can be further extended to include these synchronization issues as a performance criterion for estimating the efficiency of the Slotted ALOHA system and the Pure ALOHA system. Also the OFDM modulation technique can be applied over other multiple access techniques like Time Division Multiple Access (TDMA) and its performance compared to the classical TDMA can be evaluated in the future.

CHAPTER VII

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