Performance of Slotted ALOHA in OFDM Based Multichannel System

by

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A thesis submitted to the Department of Electrical and Electronic Engineering

of

Bangladesh University of Engineering and Technology

in partial fulfillment of the requirement for the degree of

MASTER OF SCIENCE IN ELCTRICAL AND ELECTRONIC ENGINEERING

বাংলাদেশ প্রকৌশল বিশ্ববিদ্যালয়
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তাং 15.11.05
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October, 2005

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Dedicated to

My Parents,

My Teachers,

and

My Country

Acknowledgements

All praises and gratitude to Almighty Allah, without whose desire this thesis would have never been published.

I would like to express my sincere and profound appreciation to my supervisor, Dr. Satya Prasad Majumder, Professor, Department of Electrical and Electronic Engineering (EEE), Bangladesh University of Engineering and Technology (BUET), Dhaka, Bangladesh, for his invaluable guidance, edifying suggestions and continuous encouragement throughout the course of this research work.

I am highly grateful to Md. Ataur Rahman Sarker, Assistant Professor, Department of EEE, BUET, for his invaluable help in collecting the journals, transactions and conference papers. Special thanks go to Dr. M. Nazrul Islam, Associate Professor, Department of EEE, BUET and Mr. Jahangir H. Sarker, Communications Laboratory, Helsinki University of Technology, Finland, for their continuous suggestions and co-operations in different important points regarding my thesis.

Finally, I would like to convey my deepest gratitude to my parents and my country whose contributions are latent in everywhere of this work.

Contents

List of Figures

List of Tables

List of Symbols

α	Raised cosine filter pulse shaping constant
σ	RMS delay spread
b	Number of information bits in a packet
B_c	Coherence bandwidth
BW_{RF}	Radio frequency bandwidth
D	Delay
e	Number of correctable erroneous bits in a packet
E_m	Effective number of copies of a packet
f_0	Carrier spacing
f_{max}	Nyquist frequecny
G	Actual number of packets offered to the system
G_0	Total number of packets offered to the system
k	Data symbol size of Reed Solomon code
L	Retransmission deadline
m	Number of copies of a packet
m_K	Number of copies of a packet transmitted at the K-th transmission
M	Maximum number of copies of a packet transmitted upto retransmission deadline
M_K	Total number of packets transmitted upto K-th transmission
n	Number of interfering packets / Codeword symbol size of Reed Solomon code
N	Number of channels / Number of DFT samples
N_S	Number of subcarriers
P_B	Bit error rate

$P_{CI}(Su/n)$	Probability of success of a packet in capture type I with n other interfering packets
$P_{CII}(Su/n)$	Probability of success of a packet in capture type II with n other interfering packets
P_{EI}	Probability of error of a single packet after decoding
P_{Em}	Probability of error of a packet at retransmission deadline
P_K	Probability of success after K-th transmission
P_n	Combined power of n interfering packets
P_{SI}	Probability of success of a single packet after decoding
P_{SB}	Probability of a decoded packet with no bit error
P_{SC}	Probability of successful transmission of a single packet in colliding environment
P_{Sm}	Probability of successful transmission of at least one copy of a packet at retransmission deadline
P_{SmC}	Probability of success for coded system
P_{SmU}	Probability of success for uncoded system
P_t	Power of a test packet
r	Code efficiency or code rate
R	Data rate of a single carrier
$R_{CARRIER}$	Data rate of a sub-carrier
S_C	Normalized throughput for coded system
S_U	Normalized throughput for uncoded system
T	Symbol period
$T_{CARRIER}$	Subcarrier bit duration
T_{FFT}	Size of the FFT

T_G	Length of the guard period
T_R	Single carrier bit duration
w	Number of bits in a symbol of Reed Solomon code
z	Capture ratio

List of Abbreviations

2G	Second Generation
3G	Third Generation
3GPP	Third Generation Partnership Project
3GPP2	Third Generation Partnership Project 2
4G	Fourth Generation
ADSL	Asymmetric Digital Subscriber Line
BER	Bit Error Rate
CDMA	Code Division Multiple Access
CSMA	Carrier Sense Multiple Access
DAB	Digital Audio Broadcasting
DAMA	Demand Assigned Multiple Access
DFT	Discrete Fourier Transform
DMT	Digital Multi-Tone
DTTB	Digital Terrestrial Television Broadcasting
DVB	Digital Video Broadcasting
ETSI	European Telecommunications Standards Institute
FDM	Frequency Division Multiplexing
FDMA	Frequency Division Multiple Access
FFT	Fast Fourier Transform
FM	Frequency Modulation
FSK	Frequency Shift Keying
IEEE	Institute of Electrical and Electronic Engineers
ICI	Inter Carrier Interference
IDFT	Inverse Discrete Fourier Transform
IFFT	Inverse Fast Fourier Transform
IQ	In-phase and Quadrature-phase
ISI	Inter Symbol Interference
LAN	Local Area Network
MAN	Metropolitan Area Network
OFDM	Orthogonal Frequency Division Multiplexing
QAM	Quadrature Amplitude Modulation

QoS	Quality of Service
RF	Radio Frequency
SNR	Signal-to-Noise-Ratio
TDMA	Time Division Multiple Access

Abstract

In this work slotted ALOHA protocol, OFDM system and the OFDM channel model for the random access purpose is demonstrated. The advantages of OFDM channel model compared to conventional FDMA/CDMA model are discussed. Channel traffic, packet success rate, bit error rate, throughput and delay are the vital Quality of Service (QoS) parameters used for performance measurement for any network access scheme. In this research analytical formulations of these QoS parameters for the proposed ALOHA scheme are derived. Numerous performance improvement techniques are introduced to optimize QoS parameters. Different network parameters are evaluated in case of single packet transmission per user. Analysis for simultaneous multicopy transmissions per user is carried out also. Different multicopy transmission schemes like fixed multicopy transmission and hybrid single-multicopy transmission scheme are analyzed. The effect of capture in the retransmission environment to improve network performance is investigated. Effect of number of channels is also shown. Effect of Reed Solomon coding on system performance is compared with the performance of uncoded system in different channel bit error rate conditions. The analytical expressions become more complex continually with the incorporation of different performance enhancement techniques. Moreover, the proposed system provides significant performance improvement without requiring any sophisticated control and still preserving the simplicity of ALOHA algorithm. Therefore, OFDM based slotted ALOHA seems to be a strong candidate for the future high-speed communication system.

Chapter 1
Introduction

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1.1 Introduction

Communication plays a vital role in modern world. The tremendous advancement and numerous discoveries in wireless communication have lead to its immense popularity even among common people. Wireless communication has become one of the fastest growing sectors of the world. Inspite of remarkable development, this technology cannot keep pace with people's ever-increasing demand. Second generation (2G) wireless communication is going to be the past; third generation (3G) is expanding rapidly. It seems that the world will soon switch to very high-speed fourth generation (4G) wireless communication system.

Orthogonal Frequency Division Multiplexing (OFDM) is a representative technology of ensuing 4G system. It eliminates the drawbacks of other multicarrier modulation schemes (like multipath, Inter Symbol Interference) while providing very high data rate. It has already been adopted as the standard modulation technique for Digital Audio Broadcasting (DAB), Digital Video Broadcasting (DVB) and wireless LAN (IEEE 802.11a).

ALOHA is one of the oldest random access schemes which did not loose its appeal even in 3G systems. There are many random access schemes like Carrier Sense Multiple Access (CSMA), Demand Assigned Multiple Access (DAMA), token ring but all are much complex to implement. Because of simple algorithm and popularity, ALOHA is still a strong candidate for 4G communication system.

Since future communication systems seems to be based on OFDM approach. It is required to analyze the performance of ALOHA protocol in such system.

1.2 Literature Review

Before proceeding further, it is necessary to focus a little bit on the history and researches done on ALOHA and OFDM.

1.2.1 History of ALOHA

One of the early computer networking designs, the ALOHA network was created at the University of Hawaii in 1970 under the leadership of Norman Abramson [1]. Like the ARPANET group, the ALOHA network was built with DARPA funding [2]. Similar to the ARPANET group, the ALOHA network was built to allow people in different locations to access the main computer systems. But while the ARPANET used leased phone lines, the ALOHA network used packet radio.

Abramson was a professor of engineering at Stanford, but he joined the University of Hawaii in 1970 and started working on a radio-based data communications system to connect the Hawaiian Islands together. On July 1970 first packet radio -ALOHANET became operational using the ALOHA concept of random packet transmission.

Lawrence Roberts suggested slotted ALOHA for short traffic and packet Reservation for long blocks on July1971. He proposed a new version of ALOHA, called slotted ALOHA and introduced the concept of capture in ALOHA protocol [3]. Fratta and Sant also worked extensively on capture effect in packet radio network especially in ALOHA [4].

In late 90s Pountourakis and Sykas proposed multichannel ALOHA network and analyzed the stability and optimization of system performance of this network [5].

Another important improvement was proposed by Wong and Yum. They worked on multicopy transmission in ALOHA network and suggested a performance optimization technique [6].

There has been much research work on ALOHA protocol and still a huge research is going on.

1.2.2 History of OFDM

In late 1950s concept of Frequency Division Multiplexing (FDM) evolved which lead to the introduction of OFDM. Since 1960s multicarrier transmission is introduced [7]. In 1966, R.W. Chang at Bell Labs first proposed OFDM and also patented it [8]. Weinstein & Ebert proposed use of FFT and guard interval in 1971 [9]. In 1985 L.J. Cimini describeed use of OFDM for mobile communications [10]. The reseach on OFDM gained momentum in the 1990s. Below is a short chronology of development in OFDM [11].

- 1987: Alard & Lasalle: OFDM for digital broadcasting
- 1995: ETSI DAB standard: first OFDM based standard
- 1997: ETSI DVB-T standard

- 1998: Magic WAND project demonstrates OFDM modems for wireless LAN
- 1999: IEEE 802.11a wireless LAN standard (Wi-Fi)
- 2000: proprietary fixed wireless access (V-OFDM, Flash-OFDM, etc.)
- 2002: IEEE 802.11g standard for wireless LAN
- 2004: IEEE 802.16-2004 standard for wireless MAN (WiMAX)
- 2004: ETSI DVB-H standard
- 2004: Candidate for IEEE 802.15.3a standard for wireless PAN (MB-OFDM)
- 2004: Candidate for IEEE 802.11n standard for next generation wireless LAN
- 2005: Candidate for 3.75G mobile cellular standards (3GPP & 3GPP2)
- 2005: Candidate for 4G standards.

1.3 Objectives of the Thesis

The main objective of this thesis is to analyze the performance of a slotted ALOHA network in the OFDM based multichannel system. In this research analytical formulation for channel traffic, packet success rate, bit error rate, throughput and delay for the proposed ALOHA scheme are derived considering an OFDM system with different number of channels. Different network parameters are evaluated in case of single packet transmission per user. Analysis for simultaneous multicopy transmissions per user is carried out also. Different multicopy transmission schemes like fixed multicopy transmission and hybrid single-multicopy transmission scheme are analyzed. The effect of capture in the retransmission environment to improve network performance is analyzed and investigated. The analytical results are validated by computer simulation. The results of this work are expected to contribute in the design of future high-speed communication system, especially in cellular mobile and satellite networks.

1.4 Outline of the Thesis

The remaining chapters of this thesis are organized as follows:

Chapter 2 introduces the ALOHA protocol and describes different types of ALOHA schemes. Capture phenomenon is also discussed here.

In Chapter 3, concepts of OFDM are presented. OFDM generation, transmission and reception with system block diagram are discussed in detail. Advantages of OFDM with other system are also depicted here.

Chapter 4 presents proposed system model and analyzes the model. Analytical formulations of different network parameters like channel traffic, packet success rate, bit error rate, throughput and delay are developed here.

In chapter 5, simulation results are presented with detail discussion. Effect of different number of channels, channel condition, capture, coding and multicopy transmission are thoroughly investigated here.

Chapter 6 summarizes the results of this research work and gives future research directions.

Chapter 2

ALOHA Protocol

2.1 Introduction

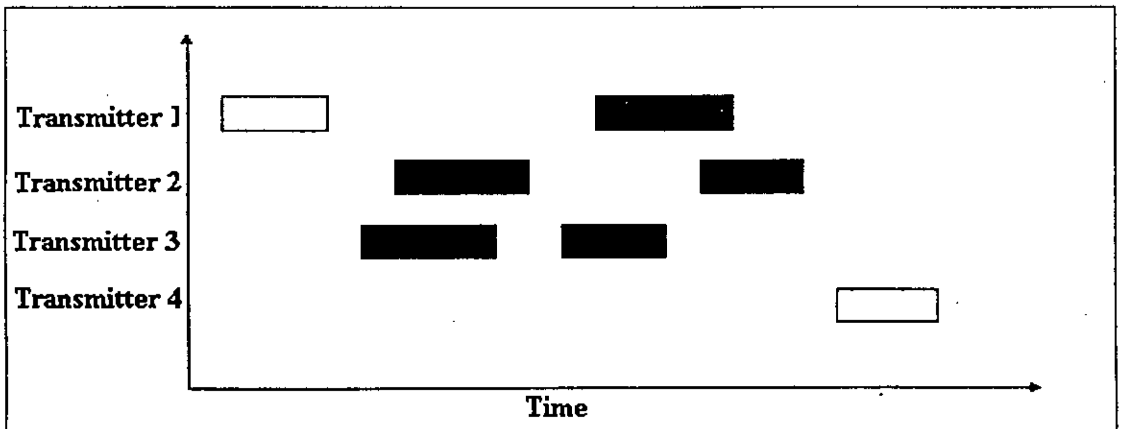
Broadcast media, such as satellite, ground radio, and multipoint cable channels, can easily provide full connectivity for communication among geographically distributed users. One of the most important problems in the design of networks (referred to as packet broadcast networks) that can take practical advantage of broadcast channels is how to achieve efficient sharing of channels. Many multiple access protocols, or algorithms, for packet broadcast networks have been proposed such as fixed assignment protocol, random access protocol, demand assignment protocol etc. In this chapter we will deal with random access protocol, specifically the ALOHA protocol.

2.2 Random Access Protocols

Random access protocol is the most suitable when the traffic is bursty. With bursty traffic a station may use available channel for a specific time and idle for the remaining time. If channel assignment is fixed, as in FDMA or TDMA, the user occupies the channel for the whole period irrespective of whether it has some data to transmit or not. Therefore, those idle periods corresponds to wasted bandwidth even though other stations may have data to send. Random access protocol like ALOHA protocol addresses this problem. ALOHA refers to a simple communications scheme in which each source (transmitter) in a network sends data whenever there is a data packet to send. If the packet successfully reaches the destination (receiver), the next packet is sent. If the packet fails to be received at the destination, it is sent again [1].

2.3 Pure ALOHA

In pure ALOHA there is no co-ordination among the transmitters. Every transmitter sends data when required. Hence there exists high possibility of collisions among the data packets. If two or more transmitters transmit packets at the same time, collision occurs and all the packets will be lost. This scheme is graphically represented as in the figure 2.1.



(Shaded slots indicate collisions and all the collided data packets are lost)

Figure 2.1: Classical (Pure) ALOHA protocol.

Pure ALOHA works well for light load and does not require any complicated access mechanisms. For example, in a wireless broadcast system or a half-duplex two-way link, ALOHA works perfectly. But as networks become more complex, for example in an Ethernet system involving multiple sources and destinations in which data travels many paths at once, trouble occurs because data frames collide (conflict). The heavier the communications volume, the worse the collision problems become. The result is degradation of system efficiency, because when two frames collide, the data contained in both frames is lost.

Pure ALOHA has the advantage of its adaptability to varying number of stations. On the other, its main problem is the collision. Assuming Poisson arrival of the packets, maximum throughput is achieved is only 18.4 % [12].

2.4 Slotted ALOHA

To minimize the number of collisions, thereby optimizing network efficiency and increasing the number of subscribers that can use a given network, a scheme called slotted Aloha was developed.

In pure ALOHA data packets can be either of the same length or not. But in slotted ALOHA all the packets are of same length. The transmission time is broken into same length of interval which is known as time slot. The packet length equals to the time slot.

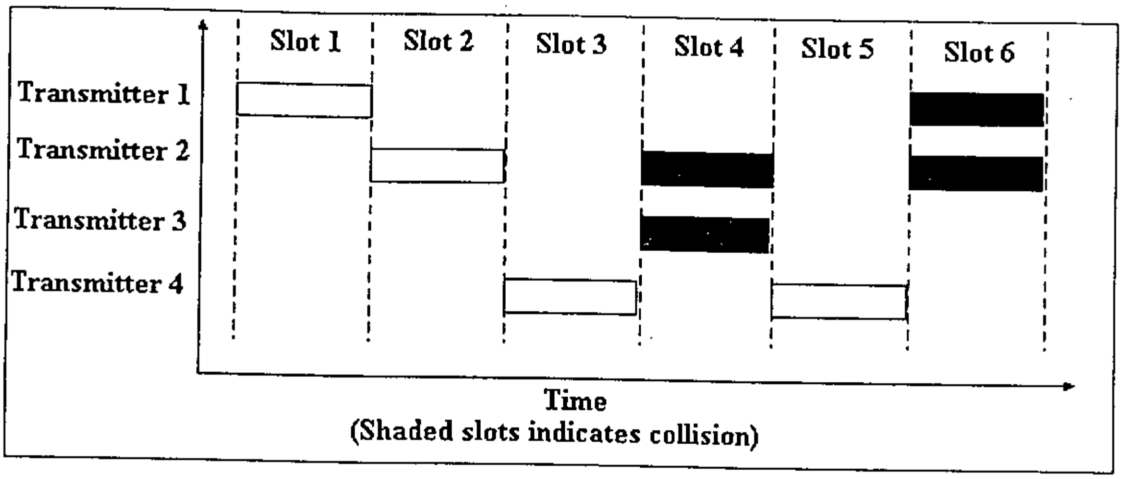


Figure 2.2: Slotted ALOHA protocol.

In the slotted ALOHA, each transmitter is allowed to transmit data packet at the beginning of a time slot. If two or more packets are transmitted during the same time slot collision will occur. Any collision of packets will overlap completely, not partially as in the pure ALOHA protocol. This condition is shown in the figure 2.2. Here the shaded packets represent colliding packets. When collision occurs, all of the transmitted packets may be lost or one of the colliding packets may be successfully transmitted depending on whether capture is introduced or not. Hence slotted ALOHA can be further divided into two categories as below [13]

(i) Slotted ALOHA without capture phenomenon, and

(ii) Slotted ALOHA with capture phenomenon.

2.4.1 Slotted ALOHA without Capture Phenomenon

In case of slotted ALOHA without capture, when collision occurs between two or more packets, all the packets will be lost. These collided packets need to be transmitted again later after a random number of time slots. Due to the use of the slotted ALOHA without capture, maximum throughput doubles then that of pure ALOHA. That means maximum throughput becomes equal to 36.8% [12]. Slotted ALOHA without capture also improves other Quality of Service (QoS) parameters such as probability of successful transmission of a packet, access delay, rejection probability etc. than that of pure ALOHA.

2.4.2 Slotted ALOHA With Capture Phenomenon

In this case a technique called capture is introduced. In case of pure ALOHA or slotted ALOHA without capture, the successful transmission of a packet will occur if and only if one user transmits a packet in the given time slot. All the packets that are attempted to transmit during the same time slot will collide and lost. The collided packets are required to be retransmitted later. But in case of slotted ALOHA with capture, if certain condition is satisfied, one of the packets may survive among the other interfering packets into the same time slot and hence successfully transmitted. The necessary condition can be categorized into two capture types namely [13]

(i) Capture type I

(ii) Capture type II

2.4.2.1 Capture Type I

Let, the probability of success of a test packet in capture type I with n other interfering packets is $P_{CI}(Su/n)$. The capture type I is defined as follows: in case of a packet collision, a test packet is captured if its power P_t is larger than z times of the combined power of n other interfering packets P_n, i.e.,

$$P_{cI}(Su/n) = P_r(P_t > z\,P_n), \text{ for } 0 \leq n \leq N-1 \tag{2-1}$$

where z ($z \geq 1$) is the capture ratio, N is the total number of transmitters and P_n is the combined power of n other interfering packets.

2.4.2.2 Capture Type II

Capture type II effect can be defined as follows: in case of a packet collision a test packet is captured if its power P_t, exceeds the power of each of the interfering packets.

Therefore the probability of success of a test packet with n other interfering packets will be

$$P_{cII}(Su/n) = \Pr(P_t > P_1)\Pr(P_t > P_2)\Pr(P_t > P_3)......\Pr(P_t > P_n) \tag{2-2}$$

The throughput of the system is much better in the capture environment than in the no capture environment. Also the other QoS parameters as mentioned above show better improvement in the capture environment than the no capture environment.

2.5 Multichannel ALOHA

So far all the discussion was based on single channel ALOHA. Recently, multichannel ALOHA has drawn attention [14]-[17]. In single channel ALOHA, there is only one channel. So at an instant, all users have only one time slot for data transmission. If number of users increases then system throughput falls drastically. Multichannel ALOHA was proposed to solve this problem. In multichannel ALOHA there is more than one channels, therefore users have a number of time slots available at an instant. A User chooses a channel randomly and sends the packet. Therefore, probability of collision is reduced and success rate increases.

2.6 Summary

In this chapter ALOHA random access protocol has been discussed thoroughly. Classification of ALOHA protocol is presented and relative advantages and disadvantages are compared. Capture phenomenon is analyzed and different types of capture are investigated. Finally, the chapter ends with the description of a recent ALOHA protocol called multichannel ALOHA.

Chapter 3
Orthogonal Frequency Division Multiplexing

3.1 Introduction

One of the challenges in today's wireless network is the ability to deploy and operate the system while maintaining good performance. Modern telecommunication users demand very high rate of data transmission. In order to transmit data at high rate, short symbol periods must be used. The symbol period is the inverse of the baseband data rate, so as data rate increases, symbol period must decrease. Natural and man-made obstacles affect performance of the wireless systems by causing multipath. In a multipath environment, a shorter symbol period leads to a greater chance for Inter-Symbol Interference (ISI) and hence degrades performance.

Efficient use of radio spectrum includes placing modulated carriers as close as possible without causing Inter-Carrier Interference (ICI). Optimally, the bandwidth of each carrier would be adjacent to its neighbors, so there would be no wasted spectrum. In practice, a guard band must be placed between each carrier bandwidth to provide a space where a filter can attenuate an adjacent carrier's signal. These guard bands cause wastage of bandwidth.

Orthogonal Frequency Division Multiplexing (OFDM) addresses both of these problems. OFDM achieves high-speed data rates without ICI and overcomes multipath and hence prevents ISI. OFDM is, therefore, considered as a candidate modulation technique in a broadband, multi-path environment.

3.2 OFDM Overview

OFDM is a modulation technique where multiple low data rate carriers are combined by a transmitter to form a composite high data rate transmission. Digital signal processing makes OFDM possible. To implement the multiple carrier scheme using a bank of parallel modulators would not be very efficient in analog hardware. However, in the digital domain, multi-carrier modulation can be done efficiently with currently available DSP hardware and software. Not only can it be done, but it can also be made very flexible and programmable. This allows OFDM to make maximum use of available bandwidth and to be able to adapt to changing system requirements.

Each carrier in an OFDM system is a sinusoid with a frequency that is an integer multiple of a base or fundamental sinusoid frequency. Therefore, each carrier is like a Fourier series component of the composite signal. In fact, an OFDM signal is created in the frequency domain, and then transformed into the time domain via the Inverse Discrete Fourier Transform (IDFT).

3.3 Difference between OFDM and FDM

There are many differences between Orthogonal Frequency Division Multiplexing (OFDM) and Frequency Division Multiplexing (FDM). In conventional broadcasting each radio station transmits on a different frequency, effectively using FDM to maintain a separation between the stations. There is however no coordination or synchronization between each of these stations. With an OFDM transmission such as Digital Audio Broadcasting (DAB), the information signals from multiple stations are combined into a single multiplexed stream of data. This data is then transmitted using an OFDM band that is made up from a dense packing of many subcarriers. All the subcarriers within the OFDM signal are time and frequency synchronized to each other, allowing the interference between subcarriers to be carefully controlled. These multiple subcarriers overlap in the frequency domain, but do not cause Inter-Carrier Interference (ICI) due to the orthogonal nature of the modulation. Typically with FDM the transmission signals need to have a large frequency guard-band between channels to prevent interference. This lowers the overall spectral efficiency. However with OFDM the orthogonal packing of the subcarriers greatly reduces this guard band, improving the spectral efficiency.

Each of the carriers in a FDM transmission can use an analog or digital modulation scheme. There is no synchronization between the transmission and so one station could transmit using FM and another in digital using FSK. In a single OFDM transmission all the subcarriers are synchronized to each other, restricting the transmission to digital modulation schemes. OFDM is symbol based, and can be thought of as a large number of low bit rate carriers transmitting in parallel. Since these multiple carriers form a single OFDM transmission, they are commonly referred to as 'subcarriers', with the term of 'carrier' reserved for describing the RF carrier mixing the signal from base band.

3.4 Definition of Orthogonal

Signals are orthogonal if they are mutually independent of each other. Orthogonality is a property that allows multiple information signals to be transmitted perfectly over a common channel and detected, without interference. Loss of orthogonality results in blurring between these information signals and degradation in communications. The subcarriers in an OFDM signal are spaced as close as is theoretically possible while maintain orthogonality between them. OFDM achieves orthogonality in the frequency domain by allocating each of the separate information signals onto different subcarriers. OFDM signals are made up from a sum of sinusoids, with each corresponding to a subcarrier. The baseband frequency of each subcarrier is chosen to be an integer multiple of the inverse of the symbol time, resulting in all subcarriers having an integer number of cycles per symbol. As a consequence the subcarriers are orthogonal to each other. Figure 3.1 shows the construction of an OFDM signal with four subcarriers.

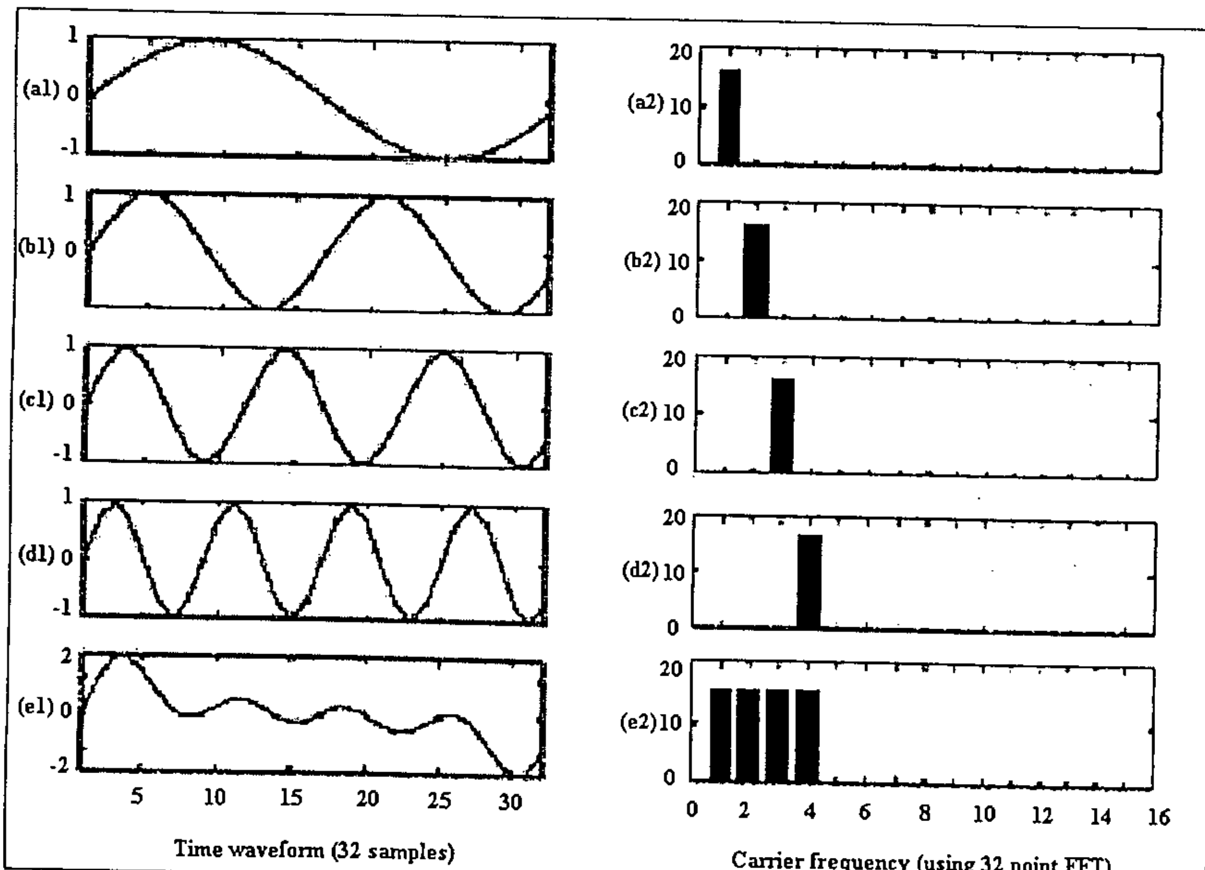


Figure 3.1: Time domain construction of an OFDM signal.

In figure 3.1, (a1), (b1), (c1) and (d1) show individual subcarriers, with 1, 2, 3, and 4 cycles per symbol respectively. The phase on all these subcarriers is zero. Each subcarrier has an integer number of cycles per symbol. (a2), (b2), (c2) and (d2) show the FFT of the time waveforms in (a1), (b1), (c1) and (d1) respectively. (e1) and (e2) shows the result for the summation of the 4 subcarriers.

3.4.1 Orthogonality in Time Domain

Sets of functions are orthogonal to each other if they match the conditions in equation (3.1). If any two different functions within the set are multiplied, and integrated over a symbol period, the result is zero, for orthogonal functions. Another way of thinking of this is that if we look at a matched receiver for one of the orthogonal functions, a subcarrier in the case of OFDM, then the receiver will only see the result for that function. The results from all other functions in the set integrate to zero, and thus have no effect.

$$\int_0^T s_i(t)s_j(t)dt = \begin{cases} C & i = j \\ 0 & i \neq j \end{cases} \tag{3-1}$$

Equation (3.2) defines a set of orthogonal sinusoids, which represent the subcarriers for an unmodulated real OFDM signal.

$$s_k(t) = \begin{cases} \sin(2\pi k f_o t) & 0 < t < T \quad k = 1, 2, \ldots M \\ 0 & otherwise \end{cases} \tag{3-2}$$

where f_0 is the carrier spacing, M is the number of carriers, T is the symbol period. Since the highest frequency component is Mf_0 the transmission bandwidth is also Mf_0.

These subcarriers are orthogonal to each other because when we multiply the waveforms of any two subcarriers and integrate over the symbol period the result is zero. Multiplying the two sine waves together is the same as mixing these subcarriers. This results in sum and difference frequency components, which will always be integer subcarrier frequencies, as the frequency of the two mixing subcarriers has integer number of cycles. Since the system is linear we can integrate the result by taking the integral of each frequency component separately then combining the results by adding the two sub-integrals. The two frequency components after the mixing have an integer number of cycles over the period and so the sub-integral of each component will be zero, as the integral of a sinusoid over an entire period is zero. Both the sub-integrals are zeros and so the resulting addition of the two will also be zero, thus we have established that the frequency components are orthogonal to each other.

3.4.2 Orthogonality in Frequency Domain

Another way to view the orthogonality property of OFDM signals is to look at its spectrum. In the frequency domain each OFDM subcarrier has a *sinc*, *sin(x)/x,* frequency response, as shown in figure 3.2. This is a result of the symbol time corresponding to the inverse of the carrier spacing. As far as the receiver is concerned each OFDM symbol transmitted for a fixed time (T_{FFT}). This symbol time corresponds to the inverse of the subcarrier spacing of $1/T_{FFT}$ Hz. This rectangular, boxcar, waveform in the time domain results in a *sinc* frequency response in the frequency domain. The *sinc* shape has a narrow main lobe, with many side-lobes that decay slowly with the magnitude of the frequency difference away from the centre. Each carrier has a peak at the center frequency and nulls evenly spaced with a frequency gap equal to the carrier spacing. The orthogonal nature of the transmission is a result of the peak of each subcarrier
corresponding to the nulls of all other subcarriers. When this signal is detected using
a Discrete Fourier Transform (DFT) the spectrum is not continuous as shown in figure 3.2, but has discrete samples. The sampled spectrums are shown as 'o's in the figure. If the DFT is time synchronized, the frequency samples of the DFT correspond to just the peaks of the subcarriers, thus the overlapping frequency region between subcarriers does not affect the receiver. The measured peaks correspond to the nulls for all other subcarriers, resulting in orthogonality between the subcarriers.

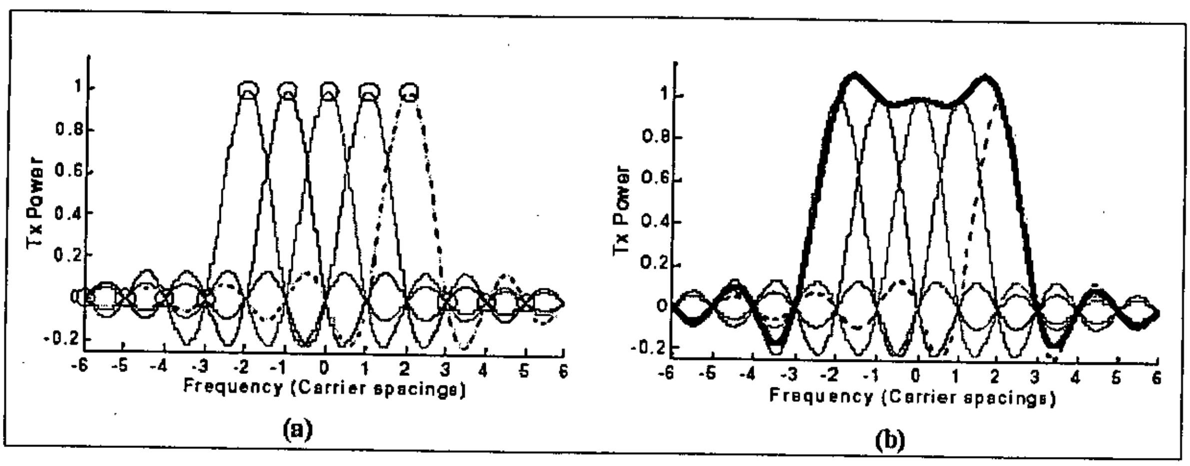


Figure 3.2: Frequency response of the subcarriers in a 5-tone OFDM signal.

Figure 3.2(a) shows the spectrum of each carrier, and the discrete frequency samples seen by an OFDM receiver. Note, each carrier is *sinc*, *sin(x)/x*, in shape. Figure 3.2(b) shows the overall combined response of the 5 subcarriers (thick black line).

3.5 OFDM Operation

When the DFT (Discrete Fourier Transform) of a time signal is taken, the frequency domain results are a function of the time sampling period and the number of samples. The fundamental frequency of the DFT is equal to $1/NT$ (1/total sample time). Each frequency represented in the DFT is an integer multiple of the fundamental frequency. The maximum frequency that can be represented by a time signal sampled at rate $1/T$ is $f_{max} = 1/2T$ as given by the Nyquist sampling theorem. This frequency is located in the center of the DFT points. All frequencies beyond that point are images of the representative frequencies. The maximum frequency bin of the DFT is equal to the sampling frequency ($1/T$) minus one fundamental ($1/NT$).

The IDFT (Inverse Discrete Fourier Transform) performs the opposite operation to the DFT. It takes a signal defined by frequency components and converts them to a time signal. The parameter mapping is the same as for the DFT. The time duration of the IDFT time signal is equal to NT.

Figure 3.3 summarizes the above-mentioned process. It is perfectly valid to generate a signal in the frequency domain, and convert it to a time domain equivalent for practical use. However, the frequency domain is a mathematical tool used for analysis. Anything usable by the real world must be converted into a real, time domain signal.

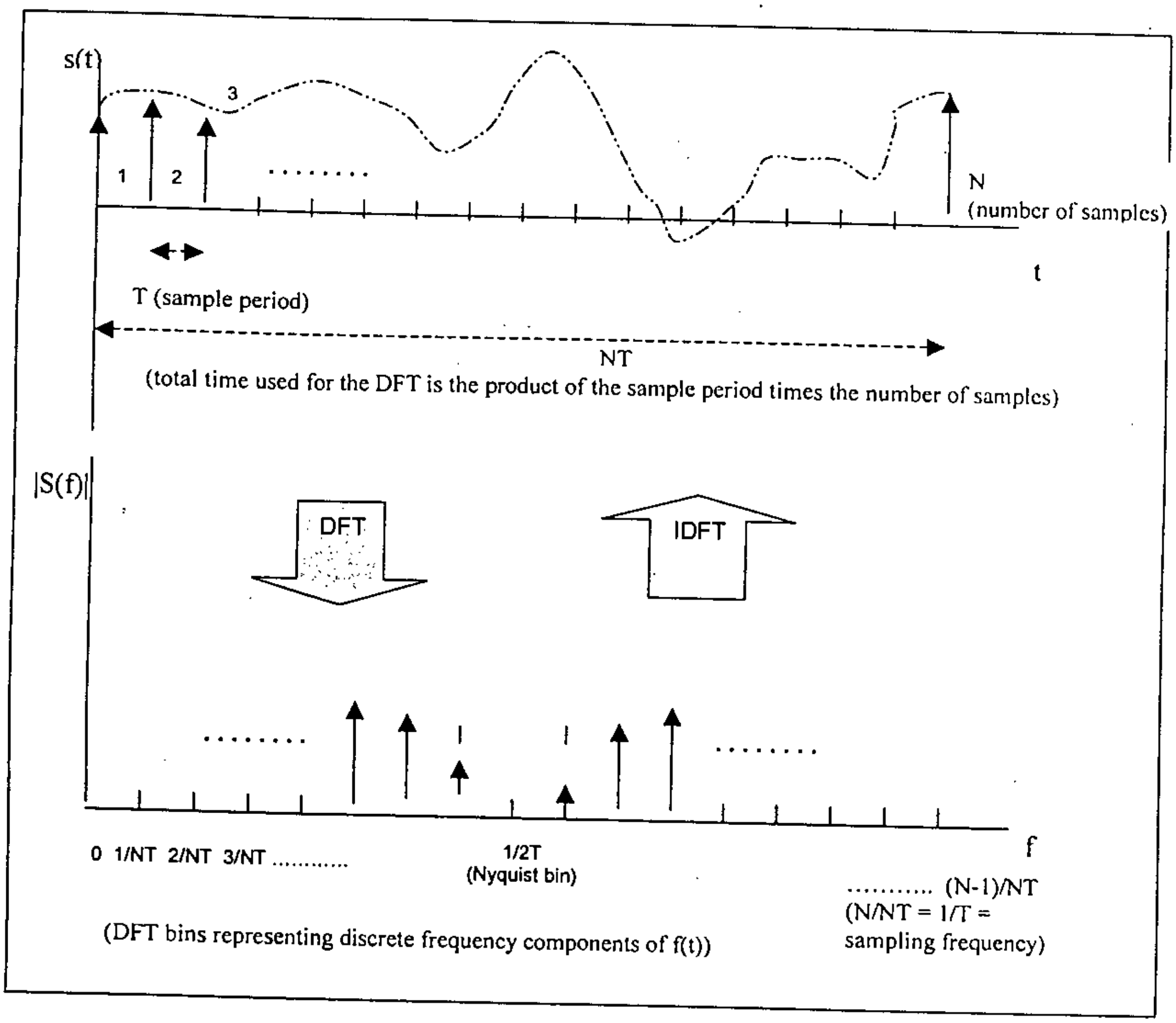


Figure 3.3: Parameter Mapping from Time to Frequency for the DFT.

3.5.1 Block Diagram of a Basic OFDM Transceiver

OFDM signals are typically generated digitally due to the difficulty in creating large banks of phase lock oscillators and receivers in the analog domain. Figure 3.4 shows the block diagram of a typical OFDM transceiver. The transmitter section converts digital data to be transmitted, into a mapping of subcarrier amplitude and phase. It then transforms this spectral representation of the data into the time domain using an Inverse Discrete Fourier Transform (IDFT). The Inverse Fast Fourier Transform (IFFT) performs the same operations as an IDFT, except that it is much more computationally efficient, and so is used in all practical systems. In order to transmit the OFDM signal the calculated time domain signal is then mixed up to the required frequency.

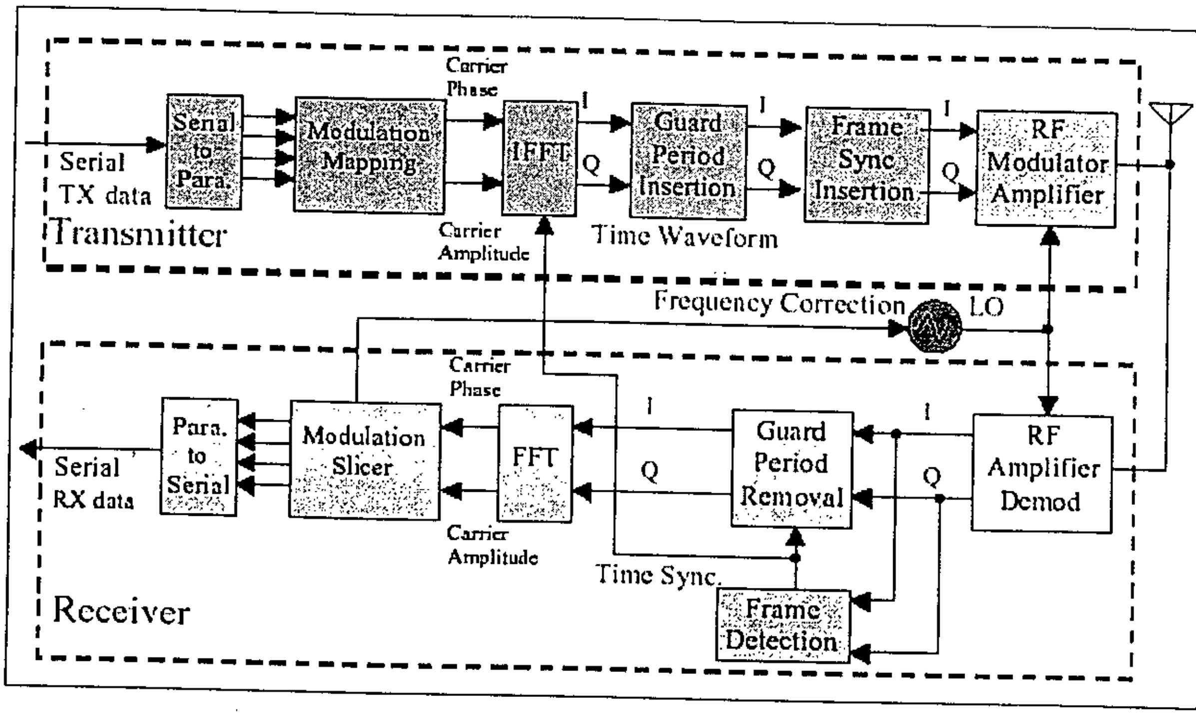


Figure 3.4: Block diagram showing a basic OFDM transceiver.

The receiver performs the reverse operation of the transmitter, mixing the RF signal to base band for processing, then using a Fast Fourier Transform (FFT) to analyze the signal in the frequency domain. The amplitude and phase of the subcarriers is then picked out and converted back to digital data. The IFFT and the FFT are complementary function and the most appropriate term depends on whether the signal is being received or generated.

3.5.2 Serial to Parallel Conversion

Data to be transmitted is typically in the form of a serial data stream. In OFDM, each symbol typically transmits 40 - 4000 bits, and so a serial to parallel conversion stage is needed to convert the input serial bit stream to the data to be transmitted in each OFDM symbol. The data allocated to each symbol depends on the modulation scheme used and the number of subcarriers. For example, for a subcarrier modulation of 16-QAM each subcarrier carries 4 bits of data, and so for a transmission using 100 subcarriers the number of bits per symbol would be 400.

At the receiver the reverse process takes place, with the data from the subcarriers being converted back to the original serial data stream.

3.5.3 Subcarrier Modulation

Once each subcarrier has been allocated bits for transmission, they are mapped using a modulation scheme to a subcarrier amplitude and phase, which is represented by a complex In-phase and Quadrature-phase (IQ) vector. Figure 3.5 shows an example of subcarrier modulation mapping. This example shows 16-QAM, which maps 4 bits for each symbol. Each combination of the 4 bits of data corresponds to a unique IQ vector, shown as a dot on the figure. A large number of modulation schemes are available allowing the number of bits transmitted per carrier per symbol to be varied.

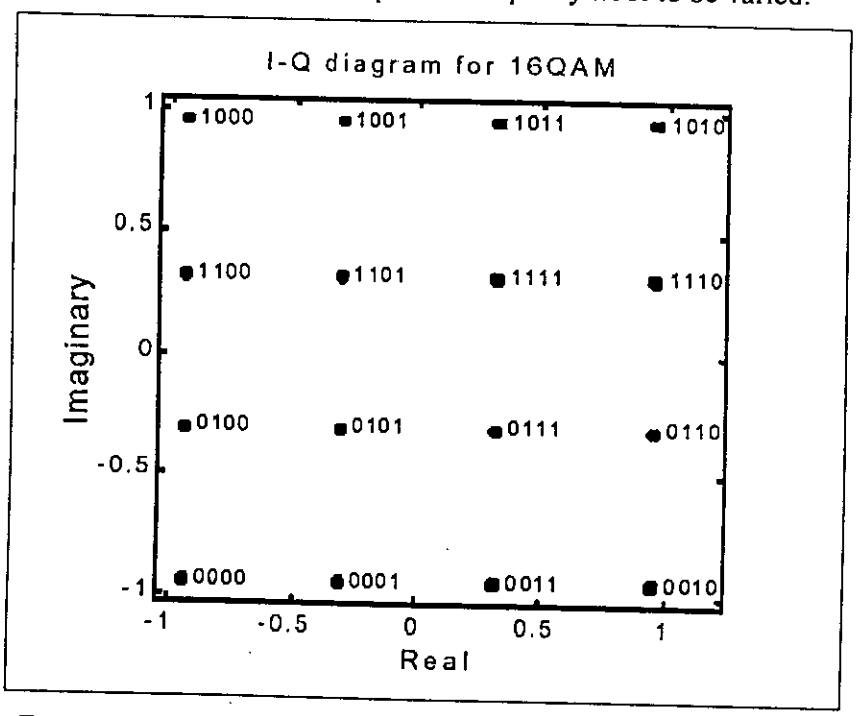


Figure 3.5: Example IQ modulation constellation. 16-QAM, with gray coding of the data to each location.

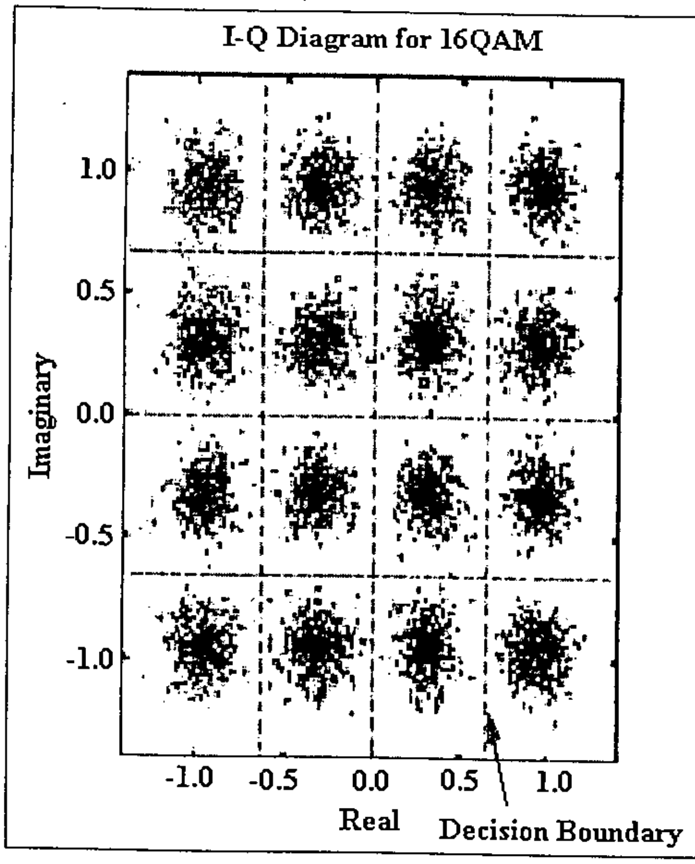


Figure 3.6: IQ plot for 16-QAM data with added noise.

In the receiver, mapping the received IQ vector back to the data word performs subcarrier demodulation. During transmission, noise and distortion becomes added to the signal due to thermal noise, signal power reduction and imperfect channel equalization. Figure 3.6 shows an example of a received 16-QAM signal with a SNR of 18 dB. Each of the IQ points is blurred in location due to the channel noise. For each received IQ vector the receiver has to estimate the most likely original transmission vector. This is achieved by finding the transmission vector that is closest to the received vector. Errors occur when the noise exceeds half the spacing between the transmission IQ points, making it cross over a decision boundary.

3.5.4 Frequency to Time Domain Conversion

After the subcarrier modulation stage each of the data subcarriers is set to an amplitude and phase based on the data being sent and the modulation scheme; all unused subcarriers are set to zero. This sets up the OFDM signal in the frequency domain. An IFFT is then used to convert this signal to the time domain, allowing it to be transmitted. Figure 3.7

shows the IFFT section of the OFDM transmitter. In the frequency domain, before applying the IFFT, each of the discrete samples of the IFFT corresponds to an individual subcarrier. Most of the subcarriers are modulated with data. The outer subcarriers are unmodulated and set to zero amplitude. These zero subcarriers provide a frequency guard band before the Nyquist frequency and allows for a realistic roll off in the analog anti-aliasing reconstruction filters.

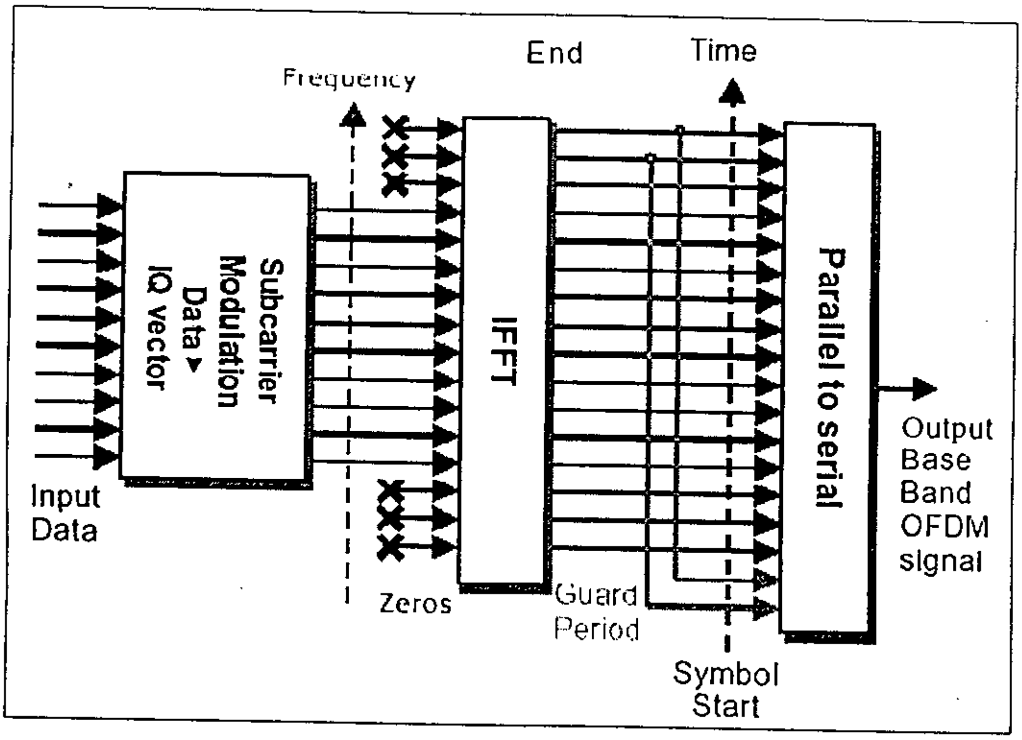


Figure 3.7: OFDM generation, IFFT stage.

3.5.5 Guard Period Insertion

OFDM demodulation must be synchronized with the start and end of the transmitted symbol period. If it is not, then ISI will occur (since information will be decoded and combined for 2 adjacent symbol periods). ICI will also occur because orthogonality will be lost (integrals of the carrier products will no longer be zero over the integration period).

In order to avoid ISI and ICI, the guard period must be formed by a cyclic extension of the symbol period. This is done by taking symbol period samples from the end of the period and appending them to the front of the period. When the IFFT is taken for a symbol period (during OFDM modulation), the resulting time sample sequence is technically periodic because the IFFT/FFT is an extension of the Fourier Transform which is an extension of the Fourier series for periodic waveforms.

With the cyclic extension, the symbol period is longer, but it represents the exact same frequency spectrum. As long as the correct number of samples are taken for decoding, they may be taken anywhere within the extended symbol. Since a complete period is integrated, orthogonality is maintained. Therefore, both ISI and ICI are eliminated.

Figure 3.8 and 3.9 show the insertion of a guard period. The total length of the symbol is $T_s = T_G + T_{FFT}$, where T_s is the total length of the symbol in samples, T_G is the length of the guard period in samples, and T_{FFT} is the size of the FFT used to generate the OFDM signal.

However, it is to be noted that some bandwidth efficiency is lost with the addition of the guard period (symbol period is increased and symbol rate is decreased).

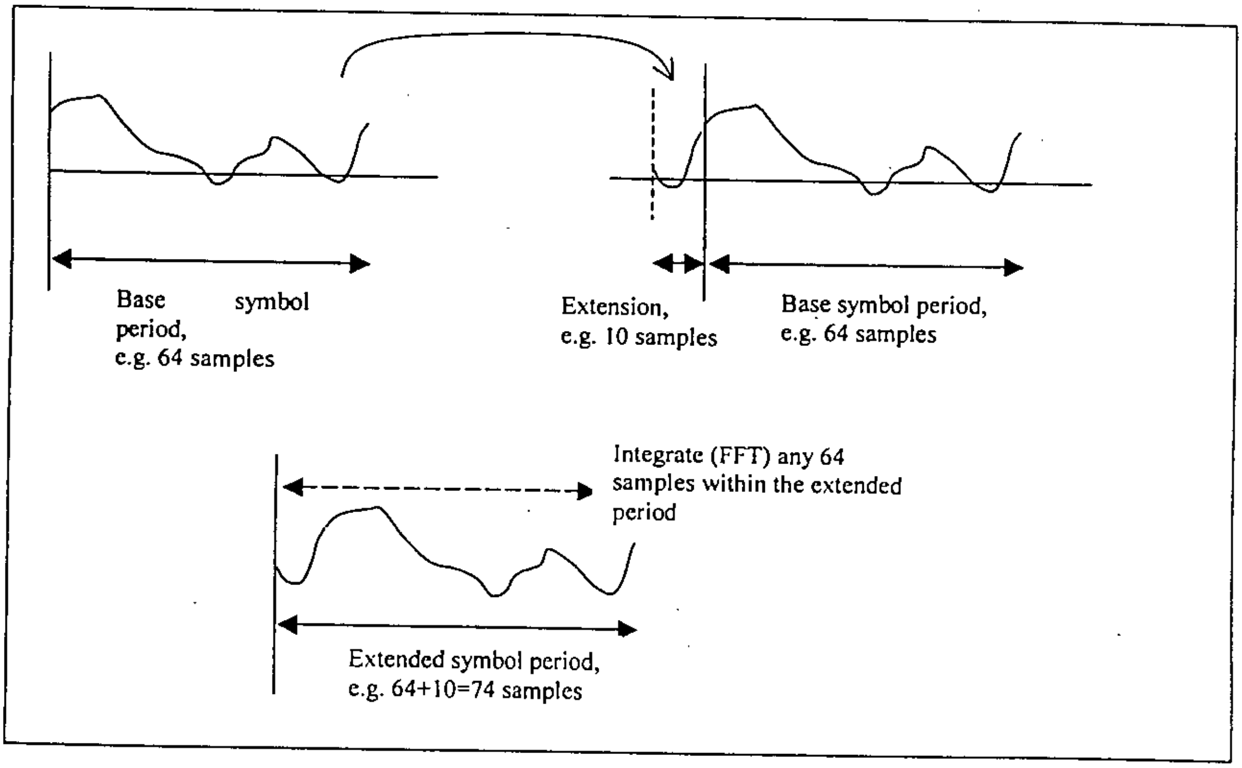


Figure 3.8: Guard Period via Cyclic Extension.

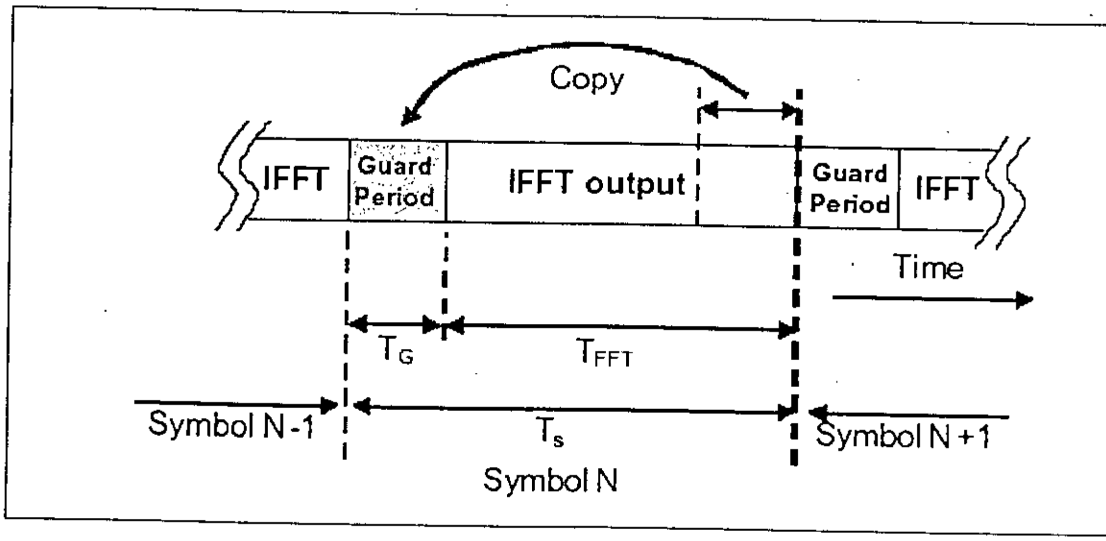


Figure 3.9: Addition of a guard period to an OFDM signal.

3.5.6 RF Modulation

The output of the OFDM modulator generates a base band signal, which is at low frequency and hence not suitable for transmission. Therefore, the baseband must be mixed up to the required transmission frequency. This can be implemented using the technique as shown in Figure 3.10.

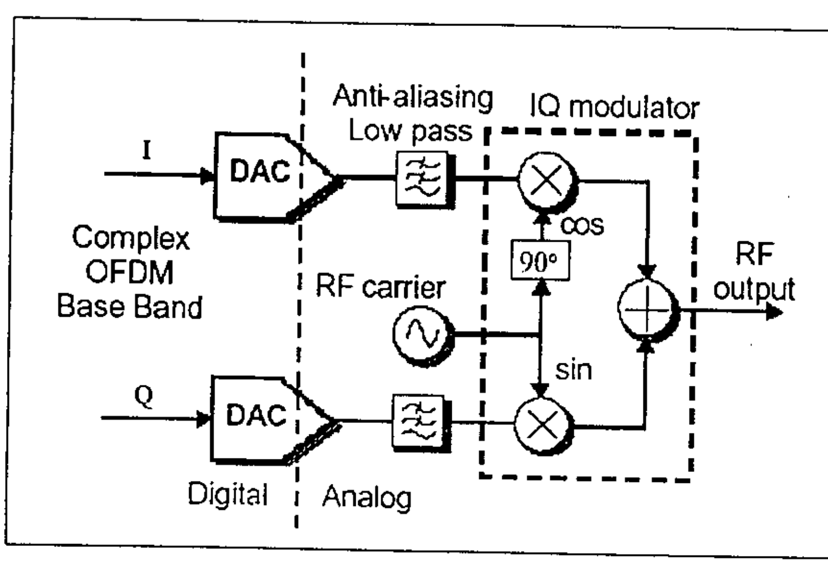


Figure 3.10: RF modulation of complex base band OFDM signal.

3.6 Multipath Characteristics of OFDM

In an OFDM signal the amplitude and phase of the subcarrier must remain constant over the period of the symbol in order for the subcarriers to maintain orthogonality. If they are not constant it means that the spectral shape of the subcarriers will not have the correct *sinc* shape, and thus the nulls will not be at the correct frequencies, resulting in Inter-Carrier Interference. At the symbol boundary the amplitude and phase change suddenly to the new value required for the next data symbol. In multipath environments ISI causes spreading of the energy between the symbols, resulting in transient changes in the amplitude and phase of the subcarrier at the start of the symbol. The length of these transient effects corresponds to the delay spread of the radio channel. The transient signal is a result of each multipath component arriving at slightly different times. Figure 3.11 shows this effect. Adding a guard period allows time for the transient part of the signal to decay, so that the FFT is taken from a steady state portion of the symbol. This eliminates the effect of ISI provided that the guard period is longer than the delay spread of the radio channel.

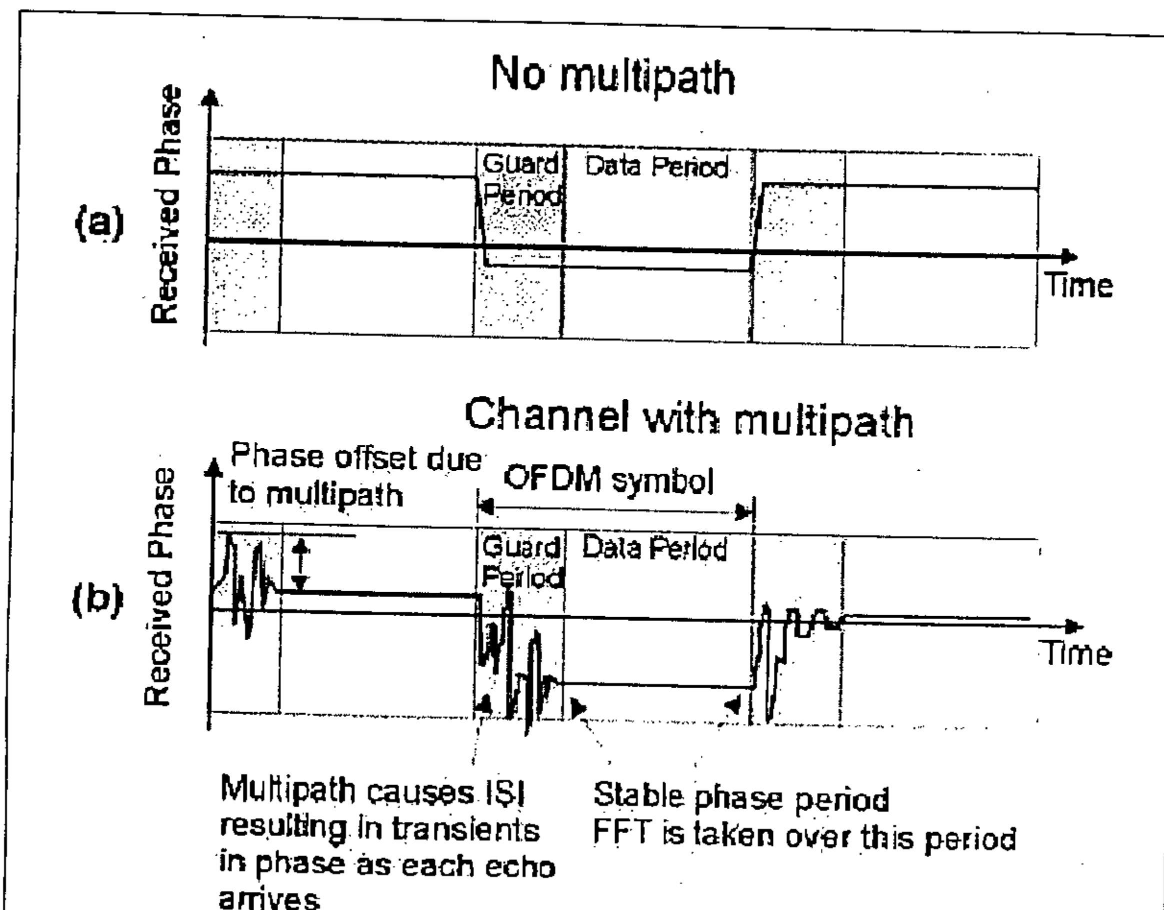


Figure 3.11: Function of the guard period for protecting against ISI.

For a given transmission channel and a given source data rate, OFDM can provide better multipath characteristics than a single carrier. This is illustrated in Figure 3.12. OFDM avoids frequency selective fading and ISI by providing relatively long symbol periods for a given data rate.

Single Carrier →

- 56 Kbps data rate (R)
- $T_R = 1/R = 17.86\ \mu sec$

Outdoor Transmission Channel →

- $\sigma = 10.0\ \mu sec$ (RMS delay spread)
- $B_c = 1/5\sigma = 20$ KHz (Coherence Bandwidth)

- $\sigma > T_R/10$*
- **Frequency Selective Fading**
- ISI

Multiple Carrier (OFDM) →

- 56 Kbps data rate (R)
- 112 carriers
- $R_{CARRIER} = 500$ bps
- $T_{CARRIER} = 2000\ \mu sec$

Outdoor Transmission Channel →

- $\sigma = 10.0\ \mu sec$ (RMS delay spread)
- $B_c = 1/5\sigma = 20$ KHz (Coherence Bandwidth)

- $\sigma < T_R/10$
- **Flat Fading**
- reduced ISI
- add guard period and completely eliminate ISI

*note: rule of thumb. if $\sigma > T_R/10$. then channel has frequency selective

Figure 3.12: OFDM vs. single carrier, multipath characteristic comparison.

3.7 Bandwidth Comparison

A comparison of RF transmission bandwidth between OFDM and a single carrier is shown in Figure 3.13 (using the same example parameters as in Figure 3.12). The calculations show that OFDM is more bandwidth efficient than a single carrier. Note that another efficient aspect of OFDM is that a single transmitter's bandwidth can be increased incrementally by addition of more adjacent carriers.

Single Carrier

- 56 Kbps data rate (R)
- $T_R = 1/R = 17.86\ \mu sec$
- Raised Cosine Filter pulse shaping, $\alpha = 0.3$
- $BW_{RF} = R(1+\alpha) = 56\ Kbps\ (1+0.3)$
- $BW_{RF} = 72.8$ KHz

Multiple Carrier (OFDM)

- 56 Kbps data rate (R)
- n = 112 carriers
- $R_{CARRIER} = 500$ bps
- $T_{CARRIER} = 2000\ \mu sec$
- N = IFFT bin count
- T = base period
- NT = symbol period = $T_{CARRIER} = 2000\ \mu sec$
- $BW_{RF} = (n+1)/NT$
- $BW_{RF} = (112+1)/2000\ \mu sec$

OFDM is 29% more bandwidth efficient for this example (72.8/56.5 = 1.2885)

Figure 3.13: OFDM Bandwidth Efficiency.

3.8 Applications of OFDM

OFDM applications include the following:

1. Digital Audio Broadcasting (DAB), wireless CD-quality sound transmission
2. Digital Video Broadcasting (DVB), specifically, Digital Terrestrial Television Broadcasting (DTTB)
3. Wireless LAN (IEEE 802.11a)
4. ADSL (Asymmetric Digital Subscriber Line), also called DMT (Digital Multi-Tone)

3.9 Summary

In this chapter, a brief overview of OFDM system is given. The property of orthogonality is explained and how OFDM is different from FDM has been discussed. Basic idea of OFDM generation has been presented. A discussion on the structure of OFDM transmitter and receiver is given. Then some performance characteristics of OFDM like multipath, ISI, data rate and bandwidth efficiency have been compared with single carrier transmission scheme. Finally some practical applications of OFDM are mentioned.

Chapter 4

ALOHA in OFDM

4.1 Introduction

ALOHA is one of the most important yet simple random access schemes, which have been successfully implemented in practical packet radio systems [18]. Recently ALOHA is being considered for multiple channels with multicopy transmission [19]-[20]. It has been shown that if multiple copies of a packet are transmitted in slotted ALOHA channels at lightly loaded condition, the probability of success and throughput increases [21]. In previous works multichannel ALOHA is achieved through either FDMA (Frequency Division Multiple Access) [5], [18], [22], [23] or CDMA (Code Division Multiple Access) [24]. In FDMA, the frequency spectrum is segmented into channels, while in CDMA, users employ different spreading codes. Both the FDMA and CDMA methods have drawbacks. In CDMA, the interference between users is inevitable. Even with orthogonal codes, orthogonality is lost at the base station because of multipath. The interference among users increases packet error rate and reduces the effective throughput. For FDMA, guard bands have to be inserted between neighboring channels. The use of guard band reduces spectrum utilization. These difficulties are non-existent with OFDM (orthogonal frequency division multiplexing) [25]. Again, multicopy slotted ALOHA is suitable only for large number of channels [6], [26]. In today's communication system spectrum utilization is a vital consideration. The requirement of large number of channels along with efficient spectrum utilization can be resolved by using Orthogonal Frequency Division Multiplexing (OFDM) based multichannel system. The OFDM system is composed of a single carrier divided into multiple subcarriers through fast Fourier transform (FFT). With accurate synchronization, each subcarrier is orthogonal to other subcarriers. Therefore, multi-channel ALOHA can be achieved through partitioning all subcarriers into subcarrier groups, with each group being a channel for random access.

OFDM enables multiple users to simultaneously access the media by using different orthogonal subcarriers without any significant interference [27]. Most importantly, the feedback message (like acknowledgement of packet success) from the base station for all the channels can be received by all users. Also traffic density of all the channels can be informed easily to all the users and hence users can choose optimal multicopy transmission scheme. For instance, in peak times channels are heavily loaded with traffic and hence users can stick to the optimal scheme for the peak multicopy scheme. Similarly, for off peak or lightly loaded condition users can choose another optimal scheme. These lead to the convenient realization of multichannel ALOHA [28].

4.2 The System Model

In OFDM based multichannel system, the subcarriers are grouped into non-overlapping sets of subcarriers. One set of subcarriers form a channel, as shown in figure 4.1. Since subcarriers are orthogonal, different channels are also orthogonal. Mobile terminals access network resource through a base station. OFDM is used for both uplink (from mobiles to base station) and downlink (from base station to mobiles), Let the total number of subcarriers be N_S. The N_S subcarriers are divided into N channels. Therefore each channel consists of N_S/N contiguous subcarriers. Users transmit in fixed length time slots. A time slot lasts for several OFDM symbol time. Figure 4.1 shows a time slot with 16 subcarriers divided into 4 channels.

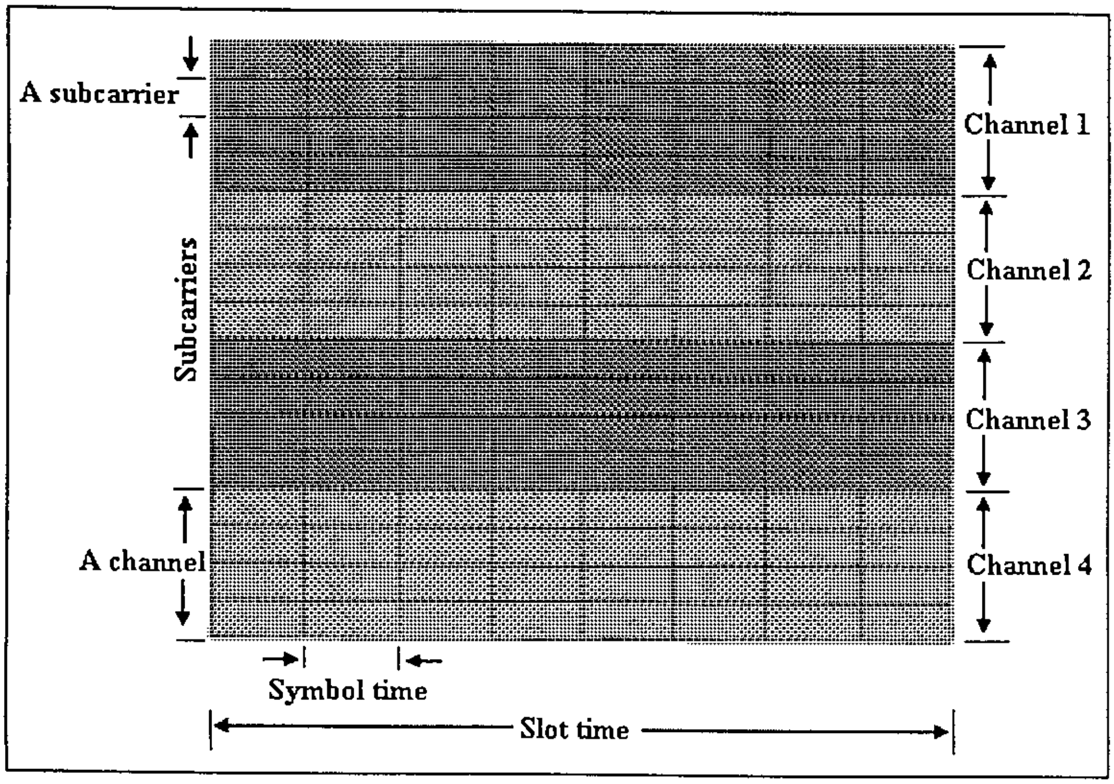


Figure 4.1: Channel and slot structure based on OFDM.

At the base station, after receiving uplink user packets, a feedback frame is sent over the downlink. This feedback frame contains information about packet transmission success or failure and the channel traffic condition. This feedback frame is received by all users. In OFDM, the data on all subcarriers is available to every user. Thus a user has the information of the feedback messages for all channels. This is different From FDM. In FDM, the channels are physically separated by different frequency bands. Thus a user is unaware of the channel status of other channels that is not operating on the same frequency band. Although it is possible for the feedback messages to be broadcast over all channels, the approach is inefficient, especially when contrasted with OFDM.

All active users transmit their packets only at the starting of a slot period. Hence N parallel slots are available to each user. Each user selects m of the N channels randomly and transmits a copy of the packet in each of the m channels. The optimal value of m is decided based on the channel traffic condition which is informed to the users through the feedback frame. For example, when channels are heavily loaded with traffic then optimal value of m may be 1 but in other loading conditions optimal value of m may be greater than 1. If user's packet is transmission is unsuccessful, the user retransmits the packer(s) on randomly chosen channels. The packet is rejected after transmission of L times, where L is called retransmission deadline. However it is assumed that the channels are memoryless, i.e., any copy of a packet experiences collision uncorrelated with its other copies of packets.

4.3 System Analysis

The system performance is measured in terms of Qos parameters like Bit Error Rate (BER), packet success rate, throughput and delay. Implementation of error correction technique affects QoS parameters also. In OFDM system, Reed Solomon forward error correction is widely used. So system performance is to be evaluated in coded and uncoded environment with different channel BER conditions.

4.3.1 Analytical Formulation of Bit Error Rate

In an uncoded system if a packet contains no bit error, the packet is successfully received. Otherwise packet is corrupted and failure occurs. For a coded system, a packet is successfully received if the decoder can correct all the erroneous bits in the transmitted packet.

Let the probability of a decoded packet with no bit error is P_{SB}. If a packet contains b information bits then for uncoded system [29],

$$P_{SB} = (1 - P_B)^b \tag{4-1}$$

where P_B is bit error rate (BER).

In coded system let us assume that the (n, k) singly extended Reed Solomon coding is applied for forward error correction. Therefore, the encoder takes k data symbols of w bits each and adds parity symbols to make n symbol codeword, where $n=2^w$. Figure 4.2 illustrates the case.

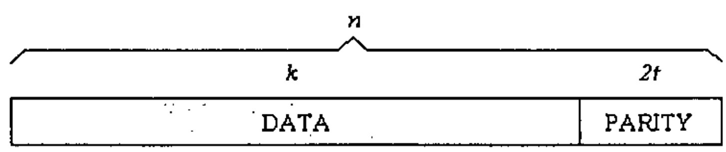


Figure 4.2: A typical Reed Solomon codeword.

The packet success rate of the coded system is given by [29],

$$P_{SB} = \sum_{x=0}^{e} \frac{b!}{(b-x)!x!} P_B{}^x (1 - P_B)^{b-x} \tag{4-2}$$

which can be expanded and simplified in the following way,

$$P_{SB} = \frac{b!}{(b-0)!0!} P_B{}^0 (1-P_B)^{b-0} + \frac{b!}{(b-1)!1!} P_B{}^1 (1-P_B)^{b-1} + \ldots\ldots + \frac{b!}{(b-e)!e!} P_B{}^e (1-P_B)^{b-e}$$

$$= (1-P_B)^b + \left[\frac{b!}{(b-1)!1!} P_B{}^1 (1-P_B)^{b-1} + \frac{b!}{(b-2)!2!} P_B{}^2 (1-P_B)^{b-2} + \ldots\ldots + \frac{b!}{(b-e)!e!} P_B{}^e (1-P_B)^{b-e} \right]$$

$$= (1-P_B)^b + \left[1 - (1-P_B)^e \right]$$

i.e.

$$P_{SB} = 1 + (1-P_B)^b - (1-P_B)^e \tag{4-3}$$

here, e is the number of correctable erroneous bits in a packet. Each packet contains b/wk codewords and in each codeword $w(n-k)/2$ bits can be corrected [30]. Therefore,

$$e = \frac{b}{wk} \left(\frac{n-k}{2} \right) w$$

i.e.

$$e = \frac{b}{k} \left(\frac{n-k}{2} \right) \tag{4-4}$$

4.3.2 Analytical Formulation of Packet Success Rate

Let us first consider that the packet collision in a slot is the only cause of packet failure. In such environment, the probability of successful transmission of a single packet is P_{SC}. Since P_{SC} and P_{SB} are independent and since a packet is successfully transmitted if and only if the packet is error free after decoding, hence the probability of success of a packet is,

$$P_{S1} = P_{SC}.P_{SB} \tag{4-5}$$

Therefore, probability of error of a single packet,

$$P_{E1} = 1 - P_{S1} = 1 - P_{SC}.P_{SB} \tag{4-6}$$

Now transmission of same packet(s) can occur upto L times and in such case the probability of error of a packet is,

$$P_{Em} = (P_{E1})^M = (1 - P_{SC}.P_{SB})^M \tag{4-7}$$

where M is maximum number of copies of a packet transmitted upto L times and therefore is given by,

$$M = \sum_{i=1}^{L} m_i \tag{4-8}$$

where m_i is number of copies of a packet in i-th transmission.

Value of M depends on which multicopy transmission scheme is followed during transmission. Here two multicopy transmission schemes are analyzed:

(i) Pure multicopy in which m copies are transmitted in each attempt. In such case,

$$M=mL \tag{4-9}$$

(ii) Multicopy in last attempt in which single copy is transmitted upto $L-1$ transmission attempt and in L-th or last attempt m copies are transmitted. In such case,

$$M=m+L-1 \tag{4-10}$$

Now the probability of success of at least one copy is

$$P_{Sm} = 1 - P_{Em} = 1 - (1 - P_{SC}.P_{SB})^M \tag{4-11}$$

Let the number of packets offered to the system is Go. It is assumed that the total packet arrival distribution at the receiver, including multicopy, is Poisson. Then from [20], probability of successful transmission of a single packet is,

$$P_{SC} = \exp\left(- \frac{Go}{N} \frac{z}{1+z} \right) \tag{4-12}$$

where z is the capture ratio. The capture ratio used here is of capture type I which is defined as follows:

In case of packet collision, a packet is captured if its power P_t is larger than z times the combined power, P_n, of n other interfering packets, where z ($z \geq 1$) is the capture ratio.

Using (4-1) and (4-12) in (4-11), the probability of success for uncoded system is,

$$P_{SmU} = 1 - \left\{1 - \exp\left(-\frac{Go}{N}\frac{z}{1+z}\right)(1-P_B)^b\right\}^M \tag{4-13}$$

Using (4-3) and (4-12) in (4-11), the probability of success for coded system is,

$$P_{SmC} = 1 - \left\{1 - \exp\left(-\frac{Go}{N}\frac{z}{1+z}\right)\overline{1+(1-P_B)^b-(1-P_B)^c}\right\}^M \tag{4-14}$$

4.3.3 Analytical Formulation of Normalized Throughput

The normalized throughput used throughout this paper refers to the number of successfully received information bits per time slot per channel.

Due to multicopy transmission, actual number of packets is not the same as the total number of offered packets. In single transmission scheme actual packet generation rate is simply *Go/m* where *Go* is the total packet generation rate [20]. However, in retransmission environment an effective packet generation rate is used to estimate actual packet generation rate which is given by [19]

$$G = \frac{Go}{E_m} \tag{4-15}$$

where E_m is effective number of copies of a packet and is given by [19]

$$E_m = m_1 + \sum_{i=2}^{L} m_i \left(1 - P_{SC}P_{SB}\right)^{\sum_{j=1}^{i-1} m_j} \tag{4-16}$$

For pure multicopy ALOHA $m_1 = m$, $m_i = m$ for any i and $m_j = m$ for any j.

If we replace $(1-P_{SC}P_{SB})$ by x then we can write,

$$E_m = m_1 + \sum_{i=2}^{L} m_i (x)^{\sum_{j=1}^{i-1} m_j}$$

$$= m_1 + m_2(x)^{m_1} + m_3(x)^{m_1+m_2} + m_4(x)^{m_1+m_2+m3} + \ldots + m_L(x)^{m_1+m_2+\ldots+m_{L-1}}$$

$$= m + m(x)^m + m(x)^{2m} + m(x)^{3m} + \ldots + m(x)^{(L-1)m}$$

$$= m\left[1+(x)^{m}+(x)^{2m}+(x)^{3m}+\ldots+(x)^{(L-1)m}\right]$$

$$= \frac{m\{1-x^{mL}\}}{\{1-x^{m}\}}$$

Therefore for pure multicopy ALOHA

$$E_m = \frac{m\left\{1-\left(1-P_{SC}P_{SB}\right)^{mL}\right\}}{\left\{1-\left(1-P_{SC}P_{SB}\right)^{m}\right\}} \tag{4-17}$$

For multicopy at last attempt we have $m_1=1$, $m_i=1$ for $i=1$ to L-1, $m_j=1$ for $j=1$ to L-1 and $m_L=m$.

$$E_m = m_1 + \sum_{i=2}^{L} m_i (x)^{\sum_{j=1}^{i-1} m_j}$$

$$= 1+(x)+(x)^2+(x)^3+\ldots+(x)^{L-2}+m(x)^{L-1}$$

$$= 1+(x)+(x)^2+(x)^3+\ldots+(x)^{L-2}+(x)^{L-1}-(x)^{L-1}+m(x)^{L-1}$$

$$= \left\{1+(x)+(x)^2+(x)^3+\ldots+(x)^{L-2}+(x)^{L-1}\right\}+m(x)^{L-1}-(x)^{L-1}$$

$$= \left\{\frac{1-x^{L}}{1-x}\right\}+(m-1)(x)^{L-1}$$

$$= \left\{\frac{1-\left(1-P_{SC}P_{SB}\right)^{L}}{1-\left(1-P_{SC}P_{SB}\right)}\right\}+(m-1)\left(1-P_{SC}P_{SB}\right)^{L-1}$$

Therefore for multicopy at last attempt E_m is given by

$$E_m = \frac{\left\{1-\left(1-P_{SC}P_{SB}\right)^{L}\right\}}{P_{SC}P_{SB}}+(m-1)\left(1-P_{SC}P_{SB}\right)^{L-1} \tag{4-18}$$

The normalized throughput for uncoded system is given by,

$$S_U = \frac{G}{N}P_{SmU} \tag{4-19}$$

Using (4-13) this can be written as,

$$S_U = \frac{G}{N}\left[1-\left\{1-\exp\left(\frac{Go}{N}\frac{z}{1+z}\right)\left(1-P_B\right)^{b}\right\}^{M}\right] \tag{4-20}$$

The normalized throughput for coded system is given by,

$$S_C = \frac{G}{N}P_{SmC} \tag{4-21}$$

Using (4-14) this can be written as,

$$S_c=\frac{G}{N}\left[1-\left\{1-\exp\left(-\frac{G\sigma}{N}\frac{z}{1+z}\right)\overline{1+(1-P_B)^b-(1-P_B)^c}\right\}^M\right] \tag{4-22}$$

4.3.4 Analytical Formulation of Delay

In single channel ALOHA if the transmission of a packet (or packets) is (are) unsuccessful then the next transmission is attempted after certain randomization of time slots. That is, transmission of next packet(s) occurs after several time slots. However, in OFDM based system multichannel system channel randomization instead of time slot randomization can be used. In OFDM system users know whether any copy of the packet on any of the channels has succeeded. In case of failure users take the next transmission attempt instantly over randomly chosen channel(s) instead of waiting for several time slots.

The packet delay or simply delay corresponds to the mean number of time slots required for successful transmission of a packet.

Now, the probability that the packet will be successfully transmitted after K-th $(1 \le K \le L)$ transmission is given by [13],

$$P_k = P_{S1}(1-P_{S1})^{M_k - m_k} \tag{4-23}$$

where M_K is the total number of packets transmitted upto K-th transmission, m_K is number of copies of a packet transmitted at the K-th transmission and P_{SI} is the probability of success of a packet given by equation (4-5).

For retransmission deadline equal to L, the mean number of transmission needed for a successful transmission is from 1 to L. Therefore the modified distribution is [13]

$$P_K' = \frac{P_K}{\sum_{K=1}^{L} P_K} \tag{4-24}$$

This can be simplified as

$$P_K' = \frac{(1-P_{S1})^{(M_k - m_k)}}{\sum_{J=1}^{L}(1-P_{S1})^{(M_J - m_J)}} \tag{4-25}$$

The delay incurred for a successful transmission of a packet is given by [13],

$$D = \sum_{K=1}^{L} K P_K' \tag{4-26}$$

Using (4-25) in (4-26), we get

$$D = \sum_{K=1}^{L} \frac{K(1-P_{S1})^{(M_K - m_K)}}{\sum_{J=1}^{L}(1-P_{S1})^{(M_J - m_J)}} \tag{4-27}$$

4.4 Summary

This chapter introduces with the discussion of benefits of ALOHA in OFDM system compared to other systems like FDMA and CDMA. Then the system model is represented and thoroughly analyzed. The OFDM based channel structure is depicted with details. Different multicopy transmission schemes, retransmission deadline, message feedback etc. have been discussed. Finally analytical formulation of bit error rate, packet success rate, normalized throughput and delay is developed for the proposed system.

Chapter 5

Results and Discussion

5.1 Introduction

This chapter presents simulation results based on the system model analyzed in the previous chapter. The simulations are carried out by well known MATLAB (version 7.0) software. A typical channel bit error rate in communication system is as low as 10^{-6}. Therefore packet success rate, normalized throughput and delay performance are evaluated in these channel BER conditions. To investigate worse channel BER condition on system performance, QoS parameters are also evaluated at a high BER of 10^{-3}. Effect of using capture is compared with uncaptured system. Reed Solomon coding is employed to improve system performance. Effect of using multicopy transmission on system performance is also investigated.

5.2 System Parameters

The system parameters that are used constantly throughout the simulation are listed below

Table 5.1: Constant System Parameters used throughout the Simulation

Parameter	Value
Retransmission deadline, L	3
Packet size, b (bits)	1200
Code word symbol size, n	32
Data symbol size, k	24
Number of copy in L-th transmission in multicopy at last attempt scheme	2

5.3 Packet Success Rate Performance

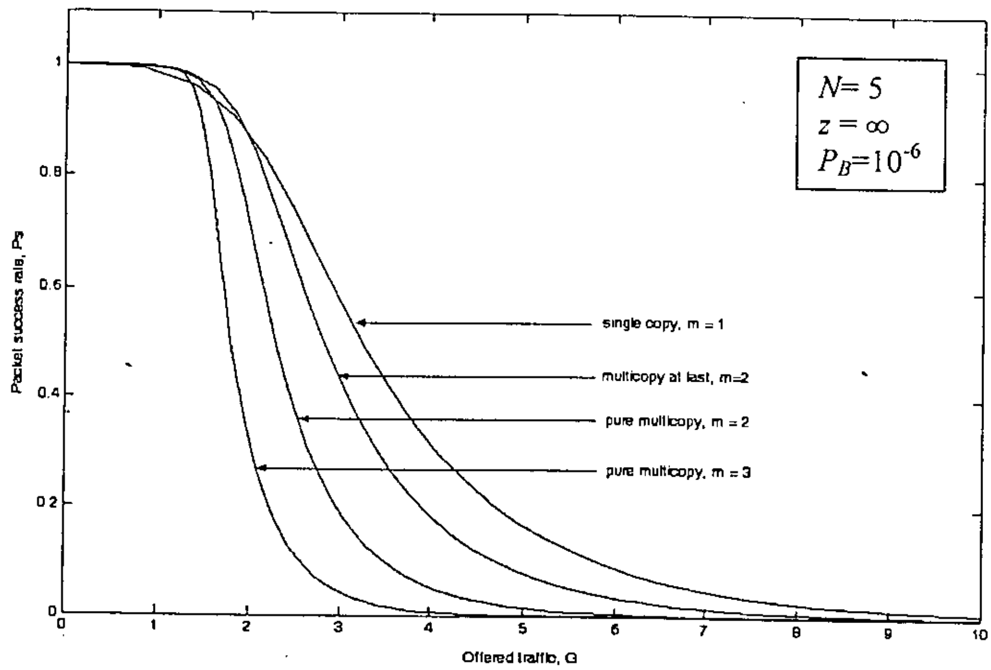
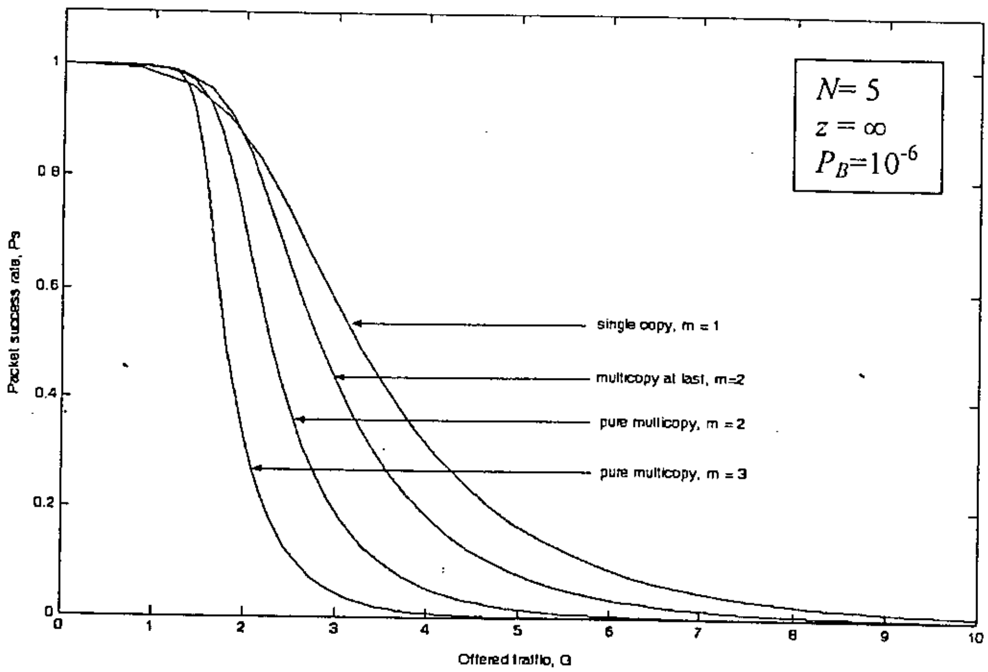


Figure 5.1: Packet success rate vs. offered traffic in uncoded uncaptured system for small number of channels and low bit error rate.

Figure 5.1 shows packet success rate in uncoded system for different traffic load in the channels when total channel number is 5 and channel bit error rate is 10^{-6}. Here capture has not been used. It is seen that when channel traffic G is around 1, all packet is successfully transmitted. However when G exceeds 1, packet success rate falls with traffic. Especially for pure multicopy i.e. m>1 this fall is drastic. For 1<G<2, multicopy at last attempt scheme exhibits the better packet success rate. However, when G>2, single copy transmission is the best policy.

5.3 Packet Success Rate Performance



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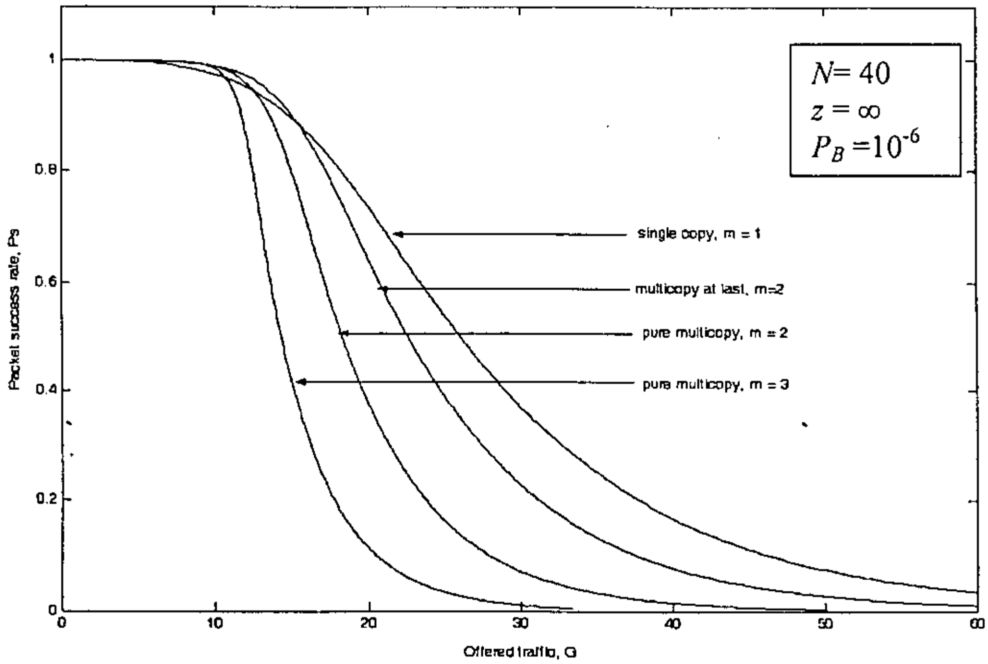

Figure 5.2: Packet success rate vs. offered traffic in uncoded uncaptured system for large number of channels and low bit error rate.

In figure 5.2, effect of increase of number of channel is shown. Here N is increased from 5 to 40.Upto G=5, packet success rate is 1. From G=5 to G=15.5, multicopy at last attempt scheme gives the better success rate. Beyond G=15.5, single copy transmission is the best policy. Therefore, increasing number of channel, better packet success rate can be achieved at higher traffic load.

Figure 5.3 and figure 5.4 show the effect of Reed Solomon coding on packet success rate with other conditions being same as in the system of figure 5.1 and figure 5.2 respectively. It is seen that there is no significant improvement in packer success rate. So for a BER of 10^{-6} and without capture, no improvement can be achieved in packet success rate by implementing coding.

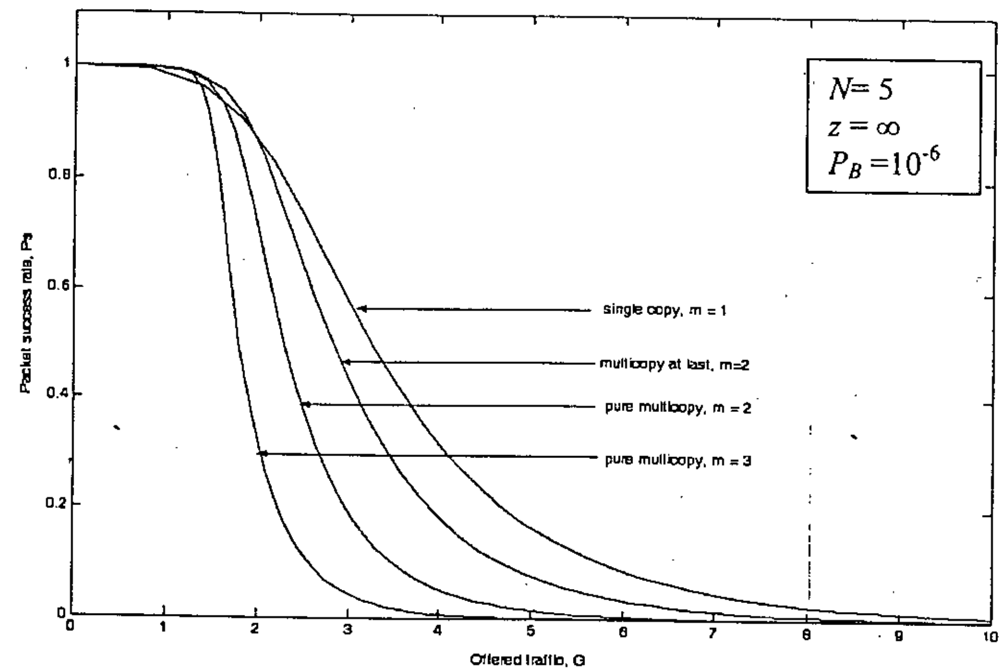

Figure 5.3: Packet success rate vs. offered traffic in coded uncaptured system for small number of channels and low bit error rate.

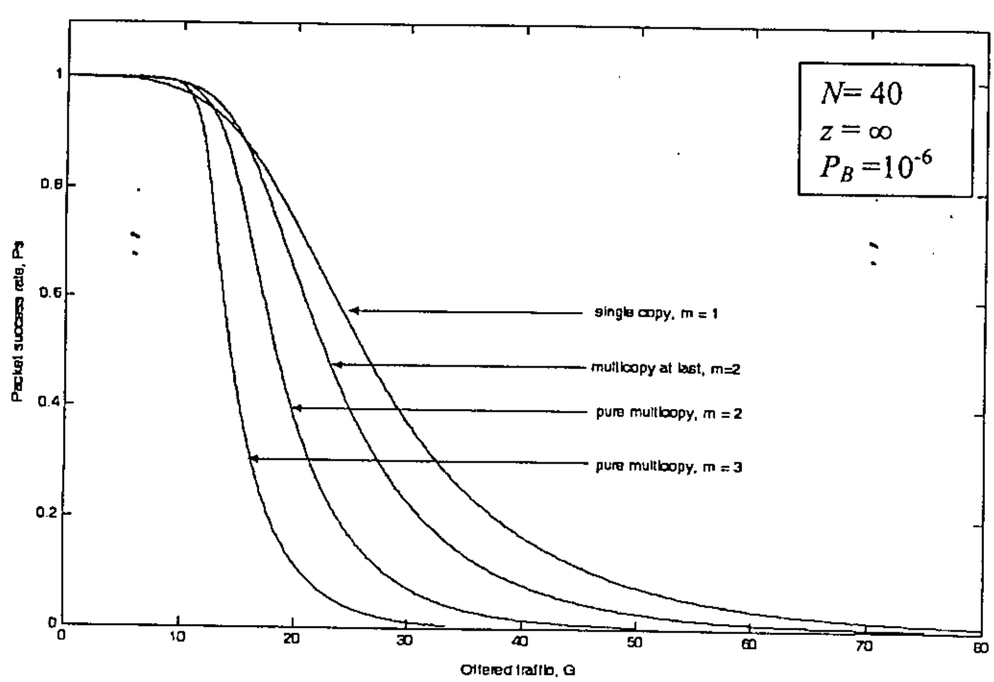

Figure 5.4: Packet success rate vs. offered traffic in coded uncaptured system for large. number of channels and low bit error rate.

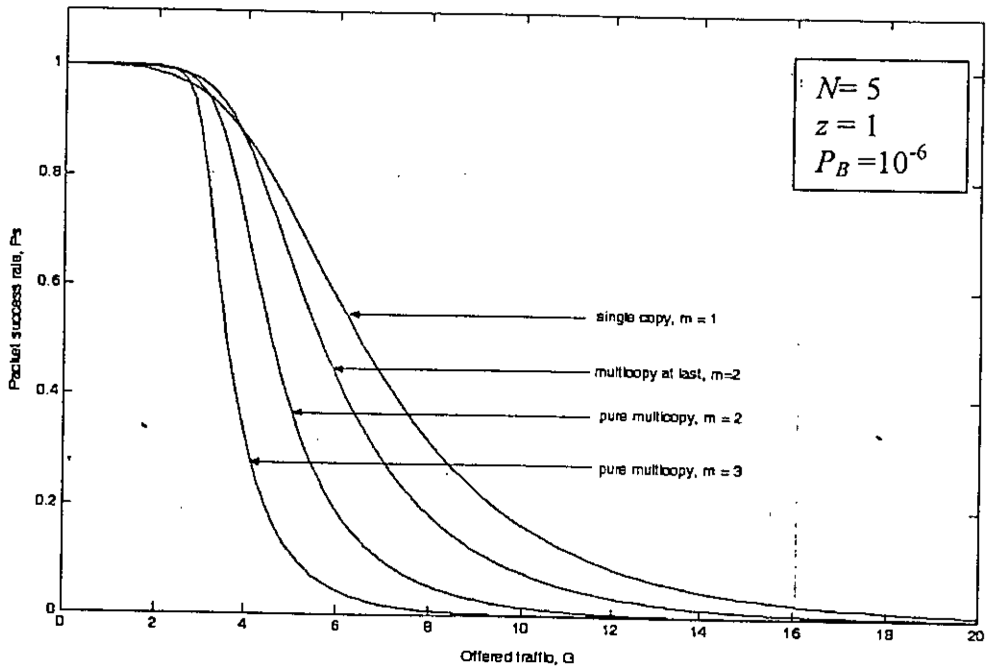

Figure 5.5: Packet success rate vs. offered traffic in uncoded captured system for small number of channels and low bit error rate.

Effect of capture on the system of figure 5.1 is shown in figure 5.5. It is seen that for traffic $G<2$, packet success rate is around 1 for all transmission scheme. For $2<G<4$, multicopy at last attempt gives the better performance. When $G>4$, single copy transmission is the best scheme. Therefore with capture, system can allow more traffic to be successfully transmitted.

Effect of increase of number of channel is shown in figure 5.6. Here increase of number of channel allows more traffic in the system than in the system of figure 5.5. For instance, here packet success rate is around 1 for traffic G upto 20.For $20<G<30$, multicopy at last attempt shows the better performance and above G=30, single copy gives the best success rate.

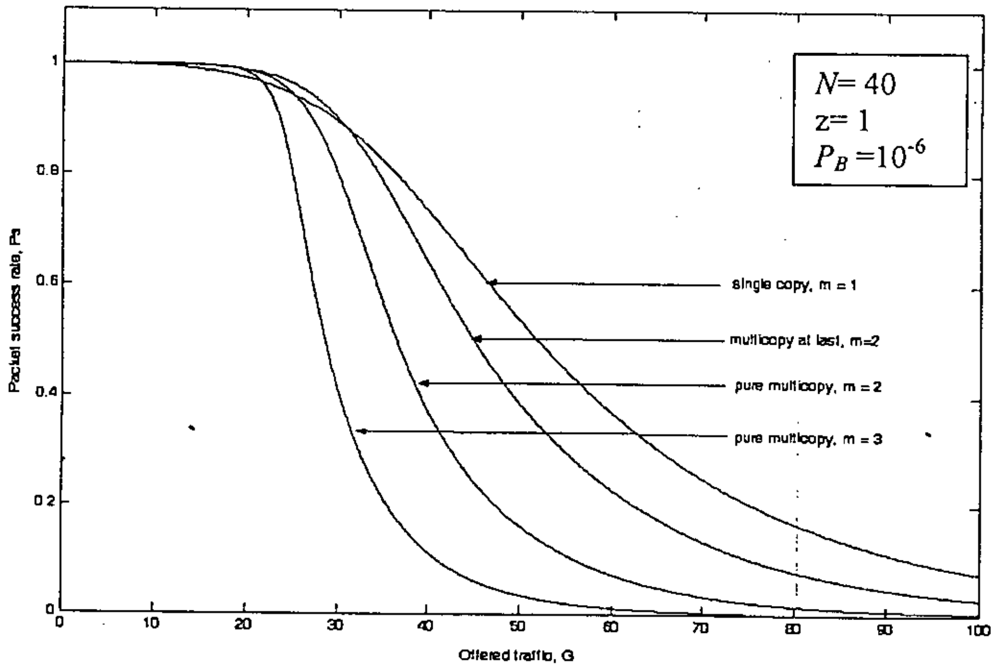

Figure 5.6: Packet success rate vs. offered traffic in uncoded captured system for large number of channels and low bit error rate.

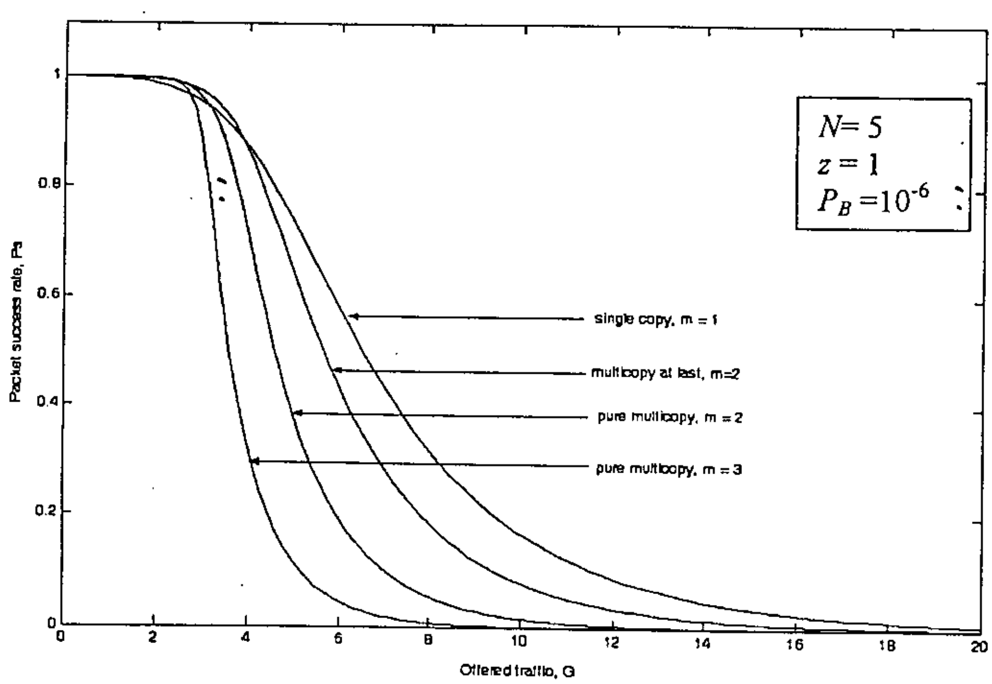

Figure 5.7: Packet success rate vs. offered traffic in coded captured system for small number of channels and low bit error rate.

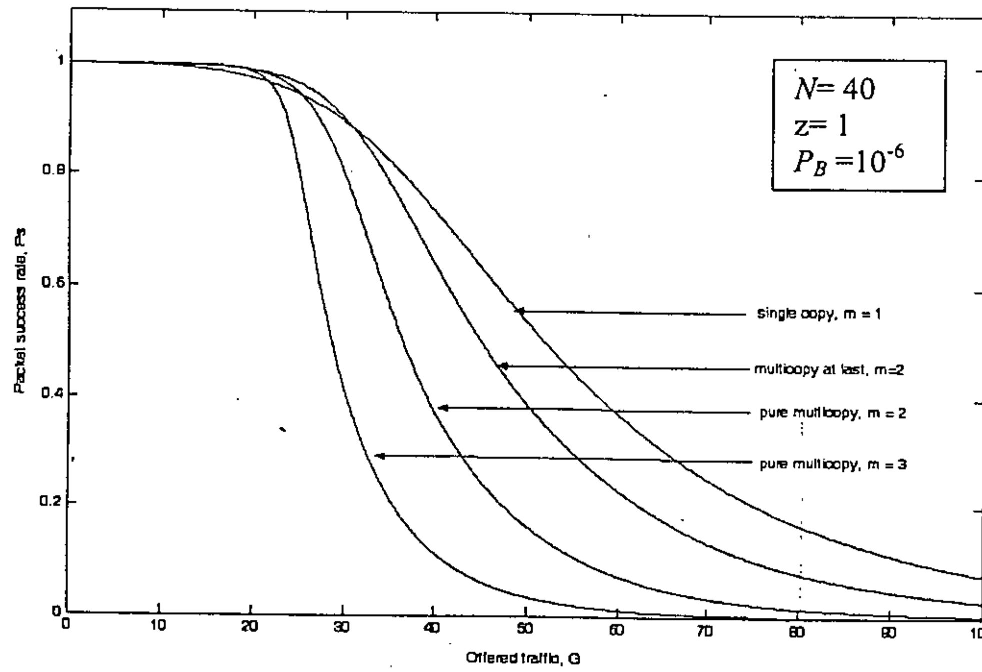

Figure 5.8: Packet success rate vs. offered traffic in coded captured system for large number of channels and low bit error rate.

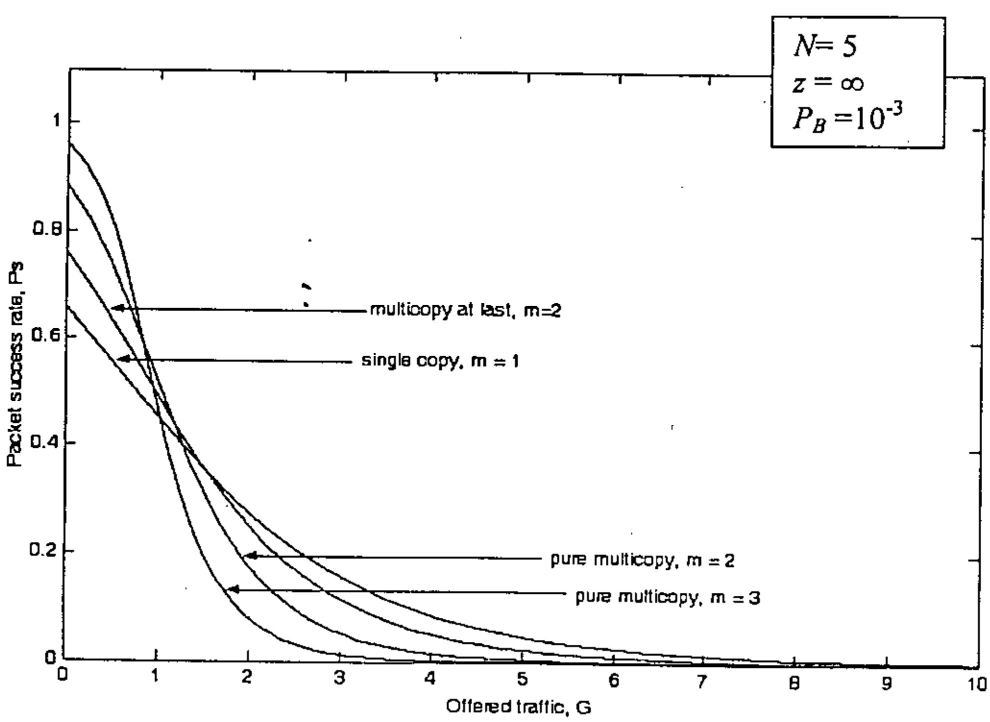

Figure 5.9: Packet success rate vs. offered traffic in uncoded uncaptured system for small number of channels and high bit error rate.

Figure 5.7 and figure 5.8 show effect of Reed Solomon coding on the packet success rate with other conditions being as same as in the system of figure 5.5 and figure 5.6 respectively. It is obvious that coding can contribute no significant improvement here.

Figure 5.9 shows packet success rate in uncoded system for different traffic load in the channels when total channel number is 5 and channel bit error rate is 10^{-3}. When G is around 1, pure multicopy schemes give the best rate i.e. the higher the number of copy, the higher the success rate. However, success rate is never around 1 for any of the packet transmission scheme. There is a crossover near G=1. That is when G exceeds 1, single copy gives the best performance. Success rate falls drastically after crossover for other transmission schemes.

Figure 5.10 shows effect of increase of number of channel from 5 to 40. Now the system can allow much more traffic load. However success rate is never 1 here. Upto traffic load of 6.5, pure multicopy schemes show better performance. Beyond this traffic, success rate improves for single copy transmission scheme and degrades with increase of number of copies.

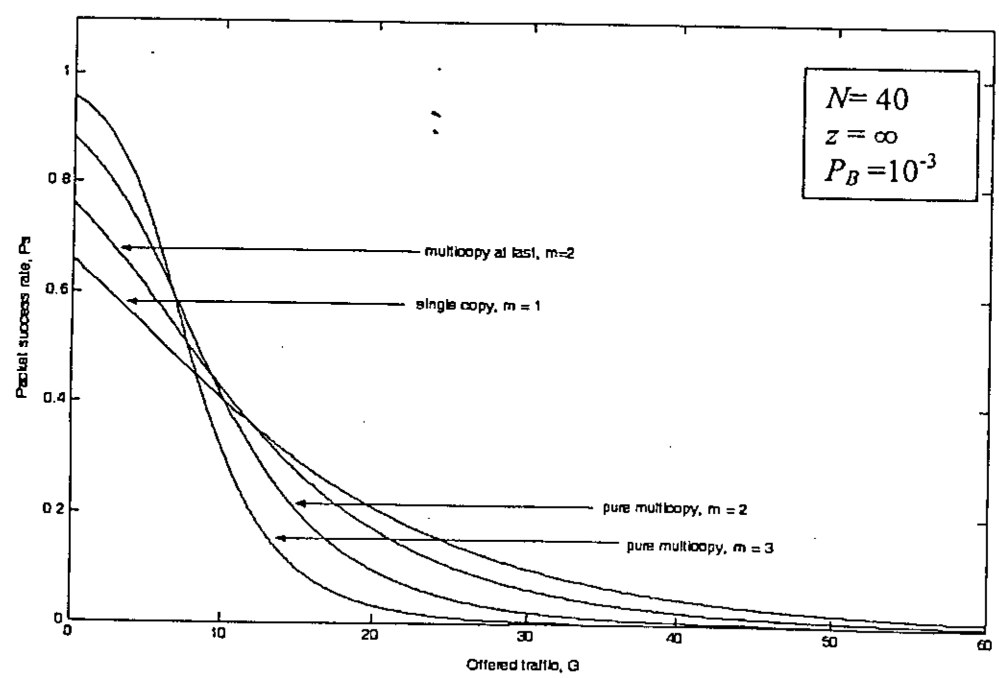

Figure 5.10: Packet success rate vs. offered traffic in uncoded uncaptured system for large number of channels and high bit error rate.

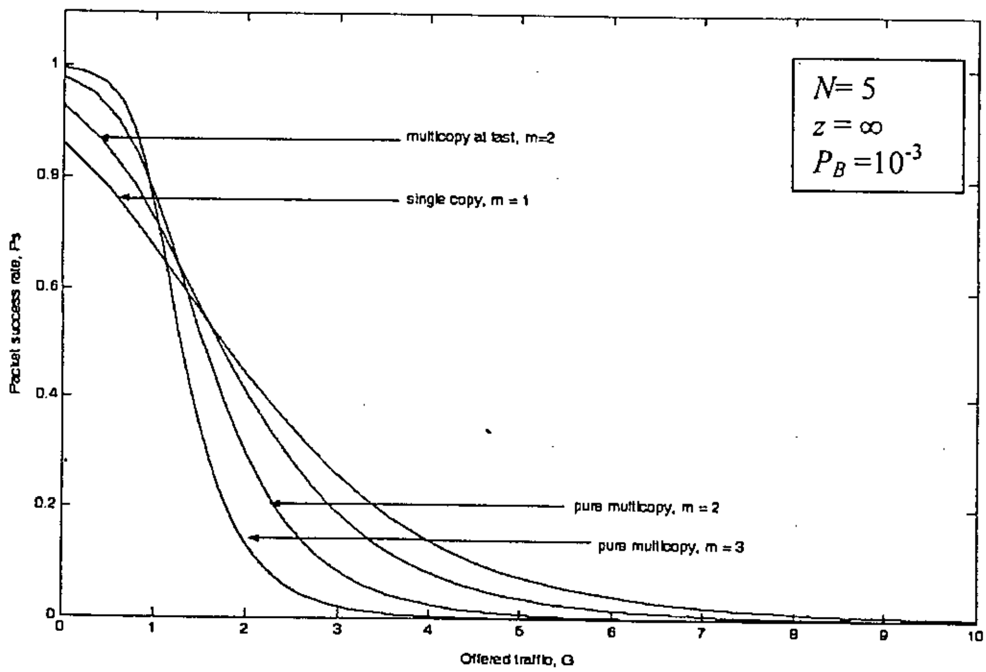

Figure 5.11: Packet success rate vs. offered traffic in coded uncaptured system for small number of channels and high bit error rate.

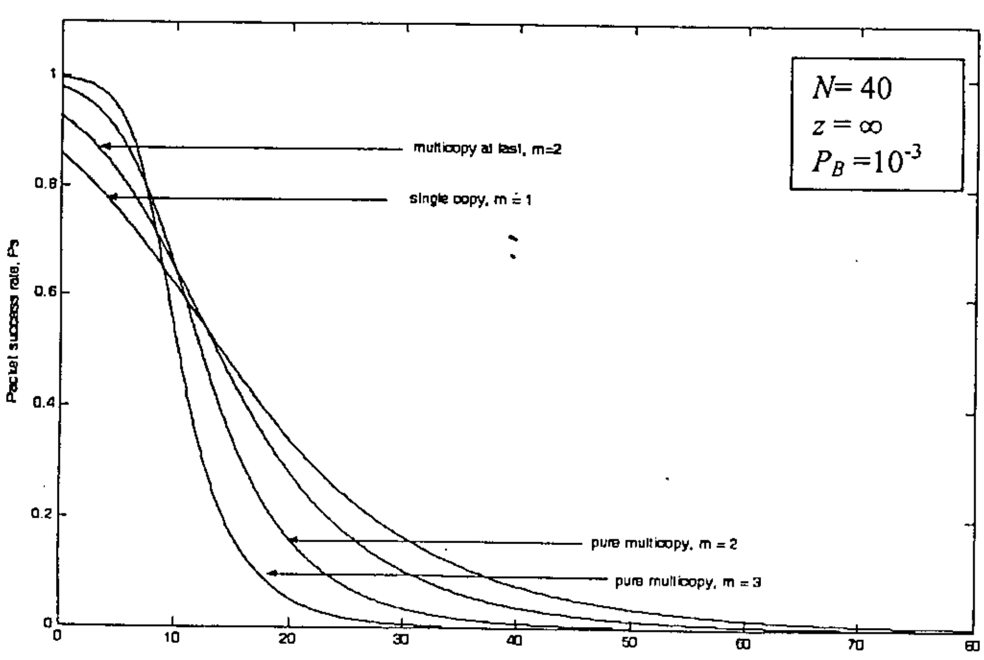

Figure 5.12: Packet success rate vs. offered traffic in coded uncaptured system for large number of channels and high bit error rate.

Figure 5.11 shows effect of Reed Solomon coding on the system of figure 5.9. Packet success rate is improved here. For example for G=1 and m=3, packet success rate is 0.75 in coded system whereas for uncoded system it is 0.475. Therefore coding can increase packet success rate when BER is high.

Effect of increase of number of channel on coded system is shown in figure 5.12. It is seen that both packet success rate and traffic load is improved. For G=7.5 and m=3, packet success rate is around 0.8 here but in uncoded system it is 0.52 which is quite low.

Figure 5.13 shows effect of capture on the system of figure 5.9. With capture channel can allow more traffic at higher packet success rate. Upto traffic G=2, pure multicopy (m=2,3) gives the better performance. Beyond G=2, single copy transmission is the better policy.

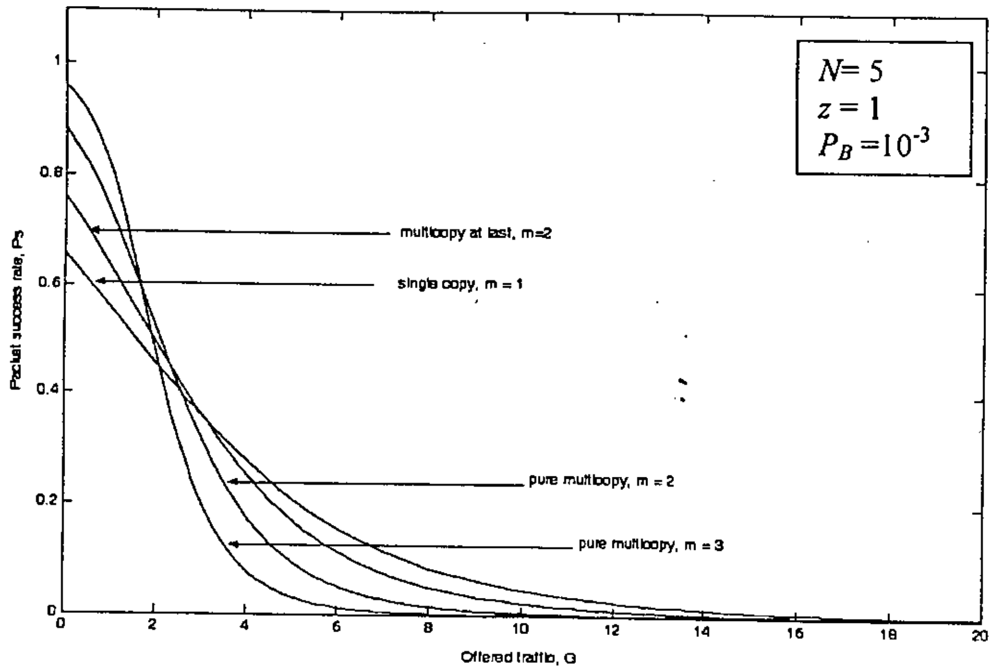

Figure 5.13: Packet success rate vs. offered traffic in uncoded captured system for small number of channels and high bit error rate.

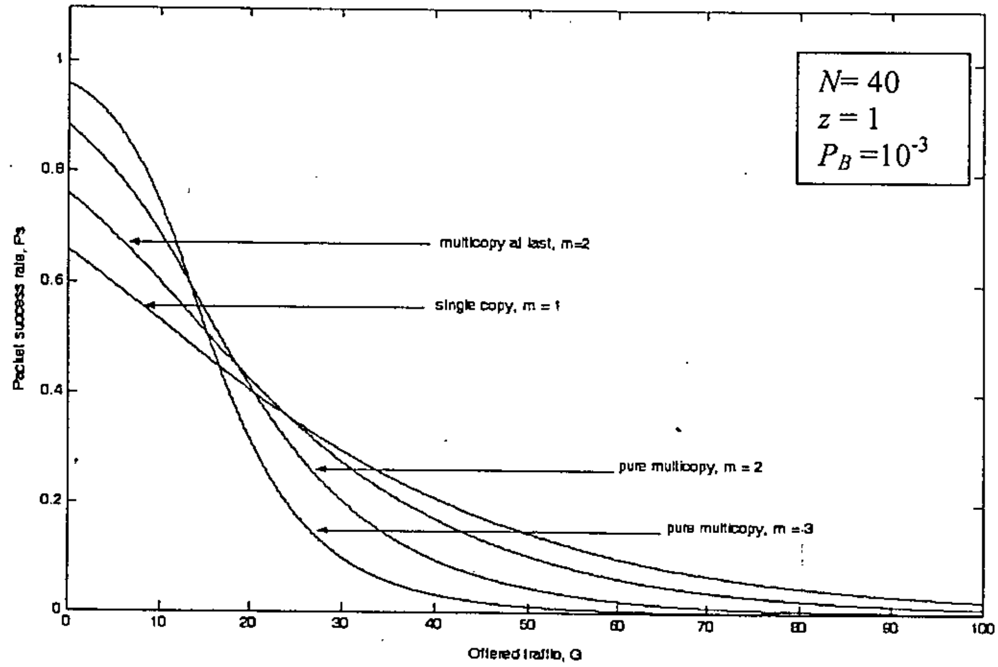

Figure 5.14: Packet success rate vs. offered traffic in uncoded captured system for large number of channels and high bit error rate.

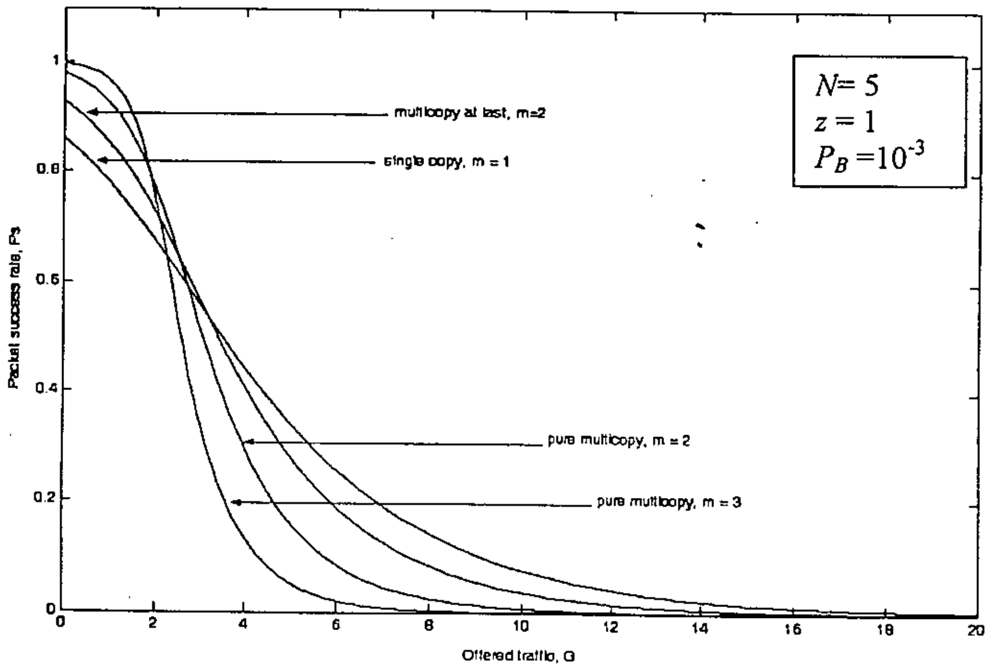

Figure 5.15: Packet success rate vs. offered traffic in coded captured system for small number of channels and high bit error rate.

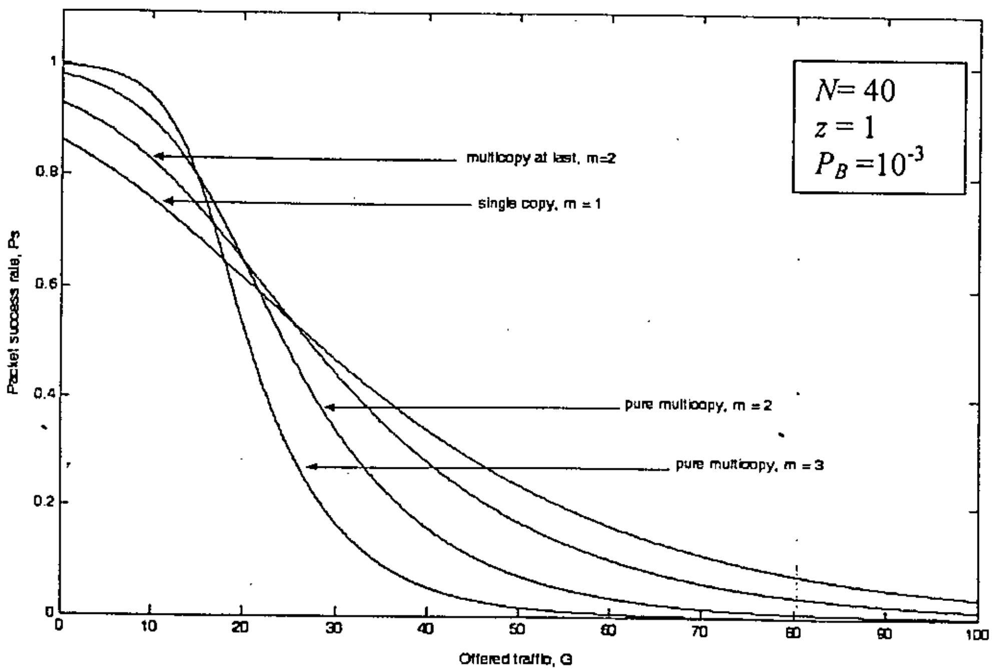

Figure 5.16: Packet success rate vs. offered traffic in coded captured system for large number of channels and high bit error rate.

Figure 5.14 shows effect of increase of channel on captured system. Upto G=13, m=3 is the best transmission scheme. Between G=13 and G=20, m=2 is better but beyond G=25, single copy transmission is the best policy.

Figure 5.15 shows the effect of Reed Solomon coding on captured system of figure 5.13. It is seen that packet success rate is improved significantly. When m=3, success rate is 0.475 at G=2 for uncoded system of figure 5.13, whereas it is 0.75 at same traffic for coded system in figure 5.15. Again, when m=1, success rate is 0.276 at G=4 for uncoded system, whereas it is 0.443 at same traffic for coded system. So packet success rate almost doubles when coding is employed. It is also to be noted that for traffic upto G=13.5, m=3 is better scheme. From G=13.5 to G= 18.5 m=2 is better and from G=18.5 to G=24, multicopy at last attempt gives the better success rate. When G>24, single copy gives the best performance.

Effect of Reed Solomon coding on captured system of figure 5.14 is shown in figure 5.16. Here again packet success rate is significantly improved. For example when G=10 and m=3, success rate is 0.74 in uncaptured system of figure 5.14 and for coded system of figure 5.16 it is 0.94. However, for traffic upto G=15, m=3 gives better success rate, from G=15 to G=20 m=2 gives the better performance and from G=20 to G=25, multicopy at last attempt gives the better rate. Beyond G=25, single copy transmission is the best scheme.

5.4 Normalized Throughput Performance

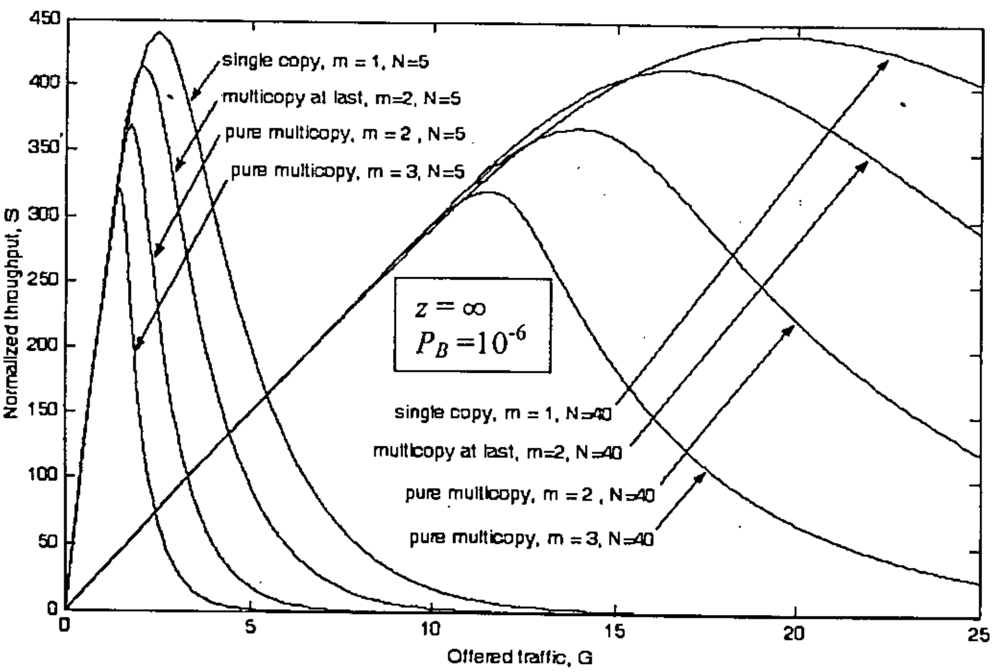

Figure 5.17: Normalized throughput vs. offered traffic in uncoded uncaptured system for low bit error rate.

Figure 5.17 shows the normalized throughput plotted against offered traffic in no capture uncoded system for different number of channel in low bit error rate (=1×10^{-6}) condition. Here, it is seen that pure multicopy transmissions are of no practical importance for lower number of channel (N=5) and it contributes to throughput a little in case of higher number of channel (N=40). Only multicopy at last attempt shows better normalized throughput upto traffic G=15.41 and beyond this traffic single copy is the best scheme.

Effect of coding on the system of figure 5.17 is shown in figure 5.18. It is observed that in case of low channel BER condition, coding contributes nothing to traffic density and system throughput. The curves are as same as in the previous figure.

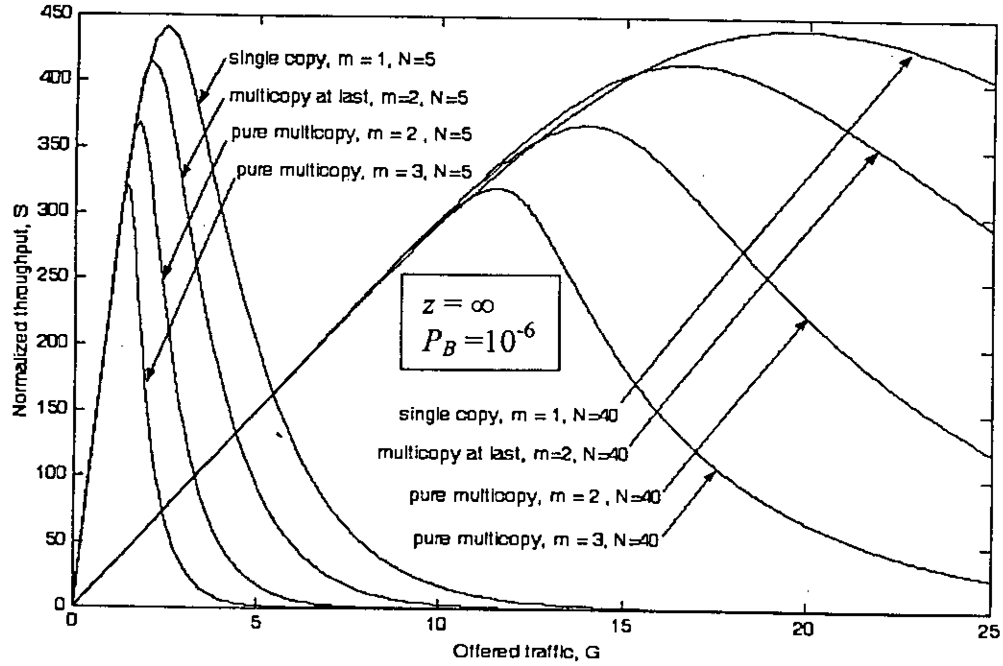

Figure 5.18: Normalized throughput vs. offered traffic in coded uncaptured system for low bit error rate.

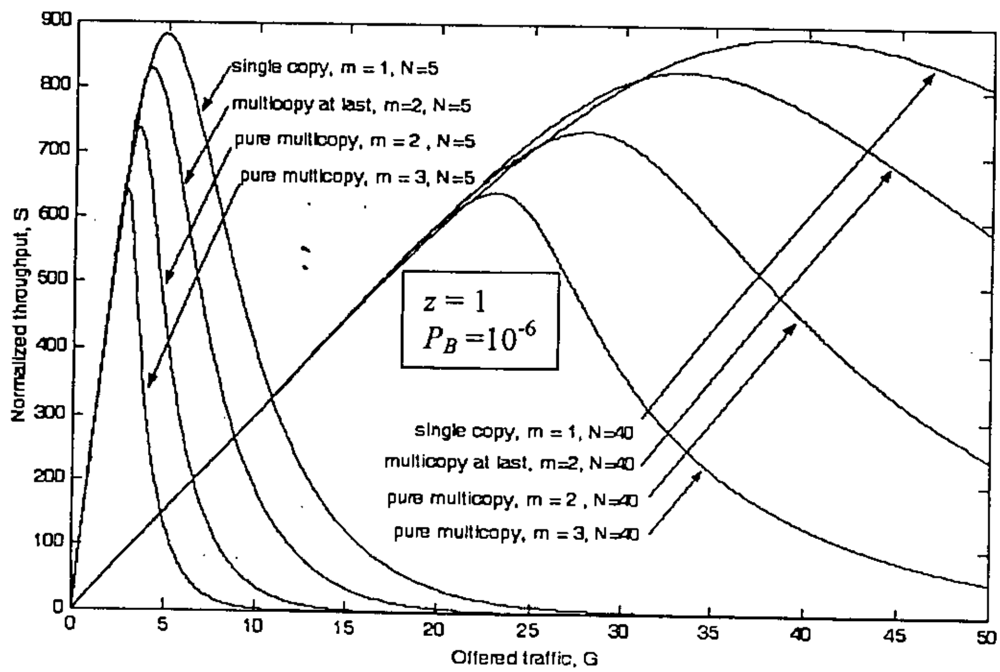

Figure 5.19: Normalized throughput vs. offered traffic in uncoded captured system for low bit error rate.

Effect of capture on uncoded system is shown in figure 5.19. The normalized throughput now enhanced significantly and shifted rightward for all multicopy transmission schemes.

This indicates higher throughput at higher traffic condition. For example, with N=40, in uncoded/coded no capture low BER channels, multicopy at last attempt exhibits better normalized throughput upto G=19.6 (where S=409 bits) whereas in captured system the normalized throughput is better upto G=30.8 (where S=819 bits). However, pure multicopy is of no use.

Figure 5.20 demonstrates the effect of combined implementation of capture, coding and increase of number of channel in low channel BER condition. Again it is observed that coding results no significant improvement in the system performance.

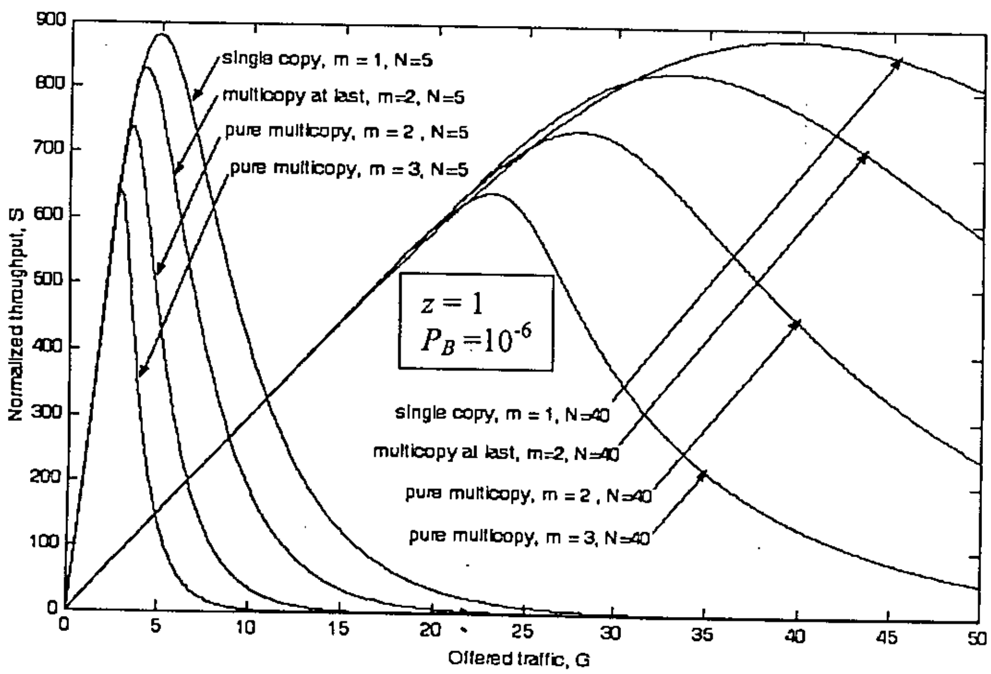

Figure 5.20: Normalized throughput vs. offered traffic in coded captured system for low bit error rate.

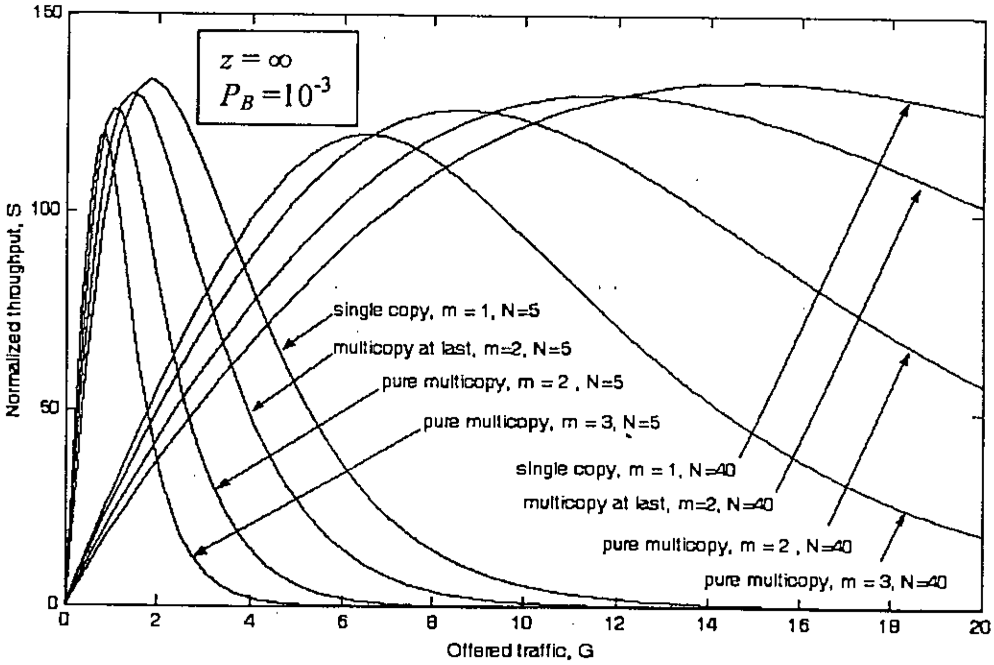

Figure 5.21: Normalized throughput vs. offered traffic in uncoded uncaptured system for high bit error rate.

From figure 5.21, it is seen that in no capture (z=∞) uncoded worse channel condition where BER is 1×10^{-3}, use of smaller number of channels allows system to exhibit peak normalized throughput at only at light traffic condition. But for large number of channels, the normalized throughputs peak at heavier traffic conditions. In both cases, multicopy transmission or multicopy at last attempt give higher throughput at lower traffic compared to single copy transmission. For example, when N=40, m=3 gives better throughput upto traffic G=6.93 (where S=139 bits), m=2 gives better throughput from G=6.93 upto traffic G=9.54 (where S=148 bits) and multicopy at last attempt gives better throughput from G=9.54 upto G=12.25 (where S=153 bits). When G>12.25, single copy transmission is the best policy and this scheme exhibits peak throughput of 158 bits at G=15.2.

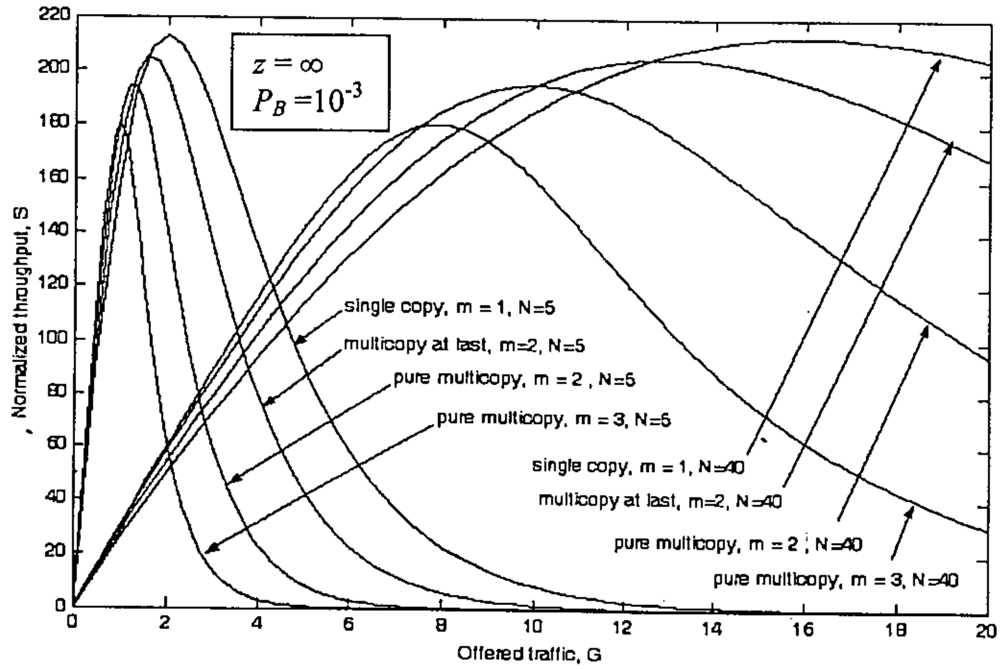

Figure 5.22: Normalized throughput vs. offered traffic in coded uncaptured system for high bit error rate.

In figure 5.22, effect of Reed Solomon error correction coding is shown. The peak throughput points are shifted rightward slightly. The normalized throughput is also increased. When N=40, m=3 gives better throughput upto traffic G=7.66 (where S=189 bits), m=2 gives better throughput from G=7.66 upto traffic G=10.14 (where S=207 bits) and multicopy at last attempt gives better throughput from G=10.14 upto G=12.75 (where S=218 bits). These indicate that coding allows more traffic to be successfully transmitted

Capture effect (z=1) on uncoded system with high BER channel has been shown in figure 5.23. The performance is better than even no capture coded system. Here, For N=40, m=2 gives better normalized throughput from G=13.8 upto G=19 (where S=296 bits). Multicopy at last attempt gives better normalized throughput from G=19 upto G=24.5 (where S =307 bits). So system allows more traffic density and exhibits higher throughput compared to earlier two systems.

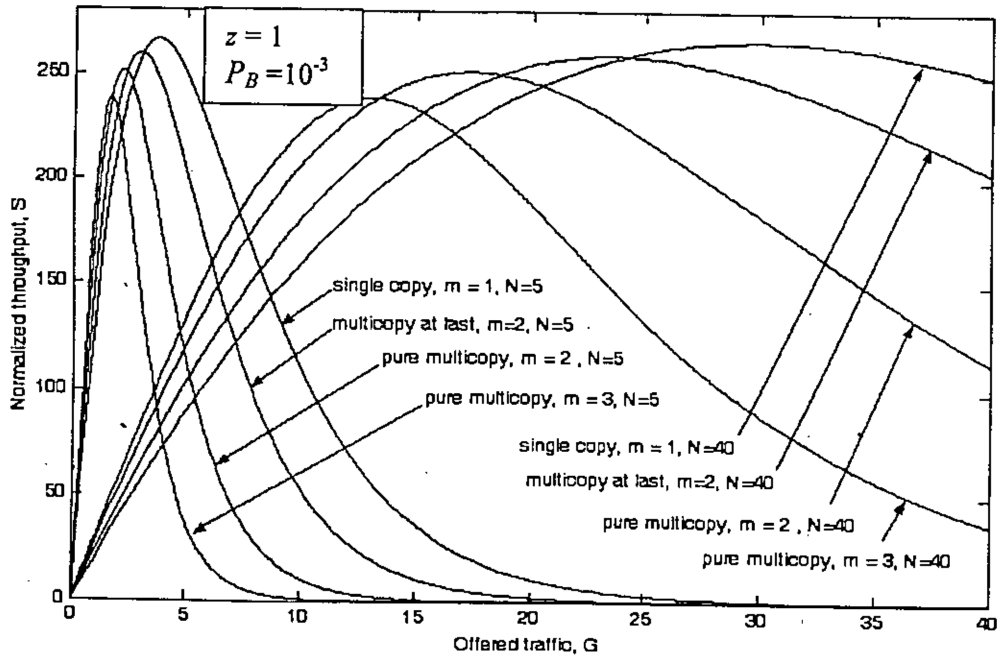

Figure 5.23: Normalized throughput vs. offered traffic in uncoded captured system for high bit error rate.

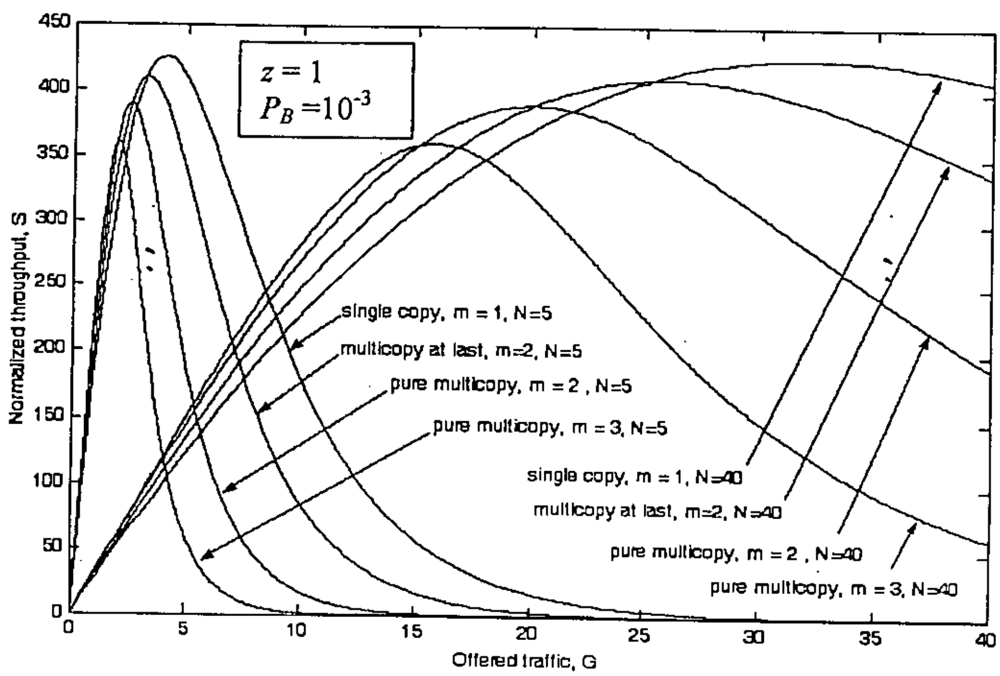

Figure 5.24: Normalized throughput vs. offered traffic in coded captured system for high bit error rate.

Figure 5.24 shows the combined implementation of capture, coding and increase of number of channel in high BER ($=1\times10^{-3}$) channel condition. Here, all the transmission schemes show better normalized throughputs compared to earlier three systems. For N=40 the normalized throughput for multicopy at last attempt is better from G=20.28 upto G=25.5 and normalized throughput is 437 bits at G=25.5. With respect to both traffic density and normalized throughput, this is the better than all previous schemes. Similarly it is observed that for m=1, 2 and 3, the system of figure 5.24 is better than previous three systems.

5.5 Delay Performance

Figure 5.25 shows delay plotted against channel traffic for number of channel 5 and bit error rate 10^{-6} with no capture. When traffic exceeds 2, delay increases sharply. Especially, for pure multicopy this rise is sharper. When G=4, delay for single copy transmission is 1.9 whereas for m=2, delay is 2. After G=12, delay for all transmission scheme saturates to 2.

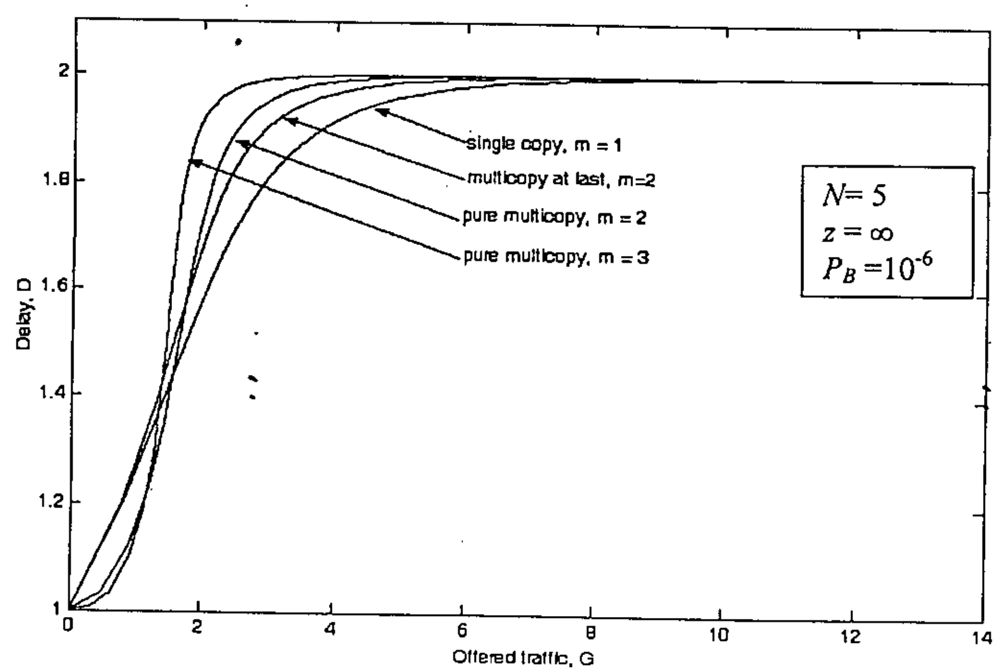

Figure 5.25: Delay vs. offered traffic in uncoded uncaptured system for small number of channels and low bit error rate.

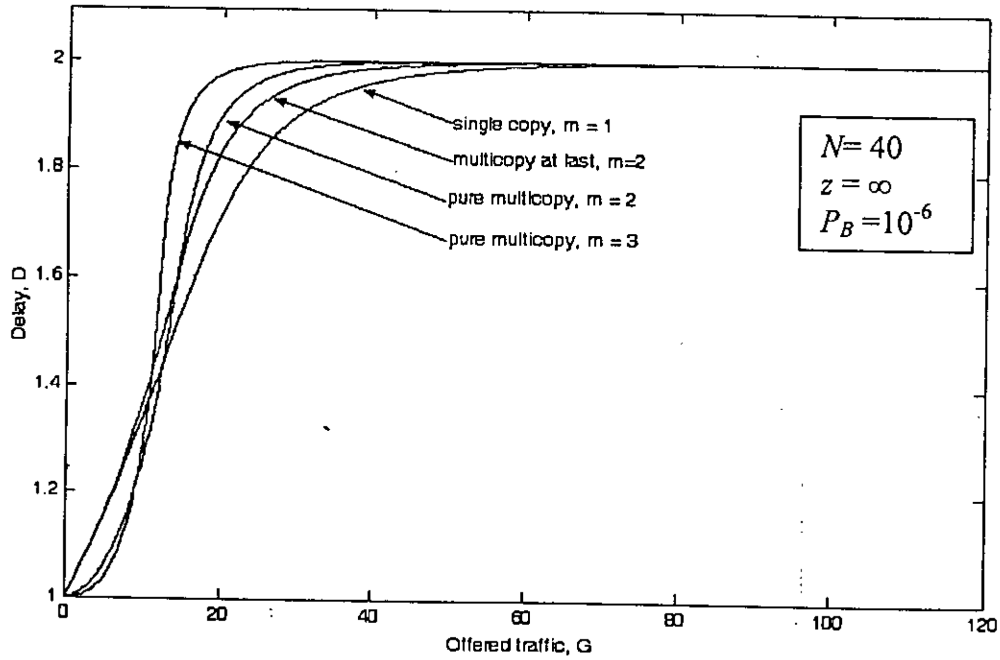

Figure 5.26: Delay vs. offered traffic in uncoded uncaptured system for large number of channels and low bit error rate.

Figure 5.26 shows delay performance for increased number of channel. The shapes of the curves are unchanged but now the system can allow higher traffic load. For example, when G=30, delay is 1.9 for single copy and is 2 for m=3. Delay saturates to 2, after channel traffic of G=100. It is also to be noted that upto traffic G=9, m=3 gives the least delay. From G=9 to G=13.5, m=2 gives the least delay and beyond G=13.5, single copy transmission gives the least delay.

In figure 5.27 effect of capture on delay performance with number of channel 5 is shown. The delay curves are now as same as in figure 5.25, but now channel traffic density doubles at same delay. When G=8, delay for single copy transmission is 1.9 whereas for m=2, delay is 2. Upto traffic G=2.5, m=3 gives the best delay performance. From G=2.5 to G=3.2, m=2 gives the least delay and after G=3.2, single copy gives the least delay. After G=25, delay for all transmission scheme saturates to 2.

Figure 5.28 shows effect of increase of number of channel in capture environment. At the same delay system allows almost double traffic load than in no capture environment.

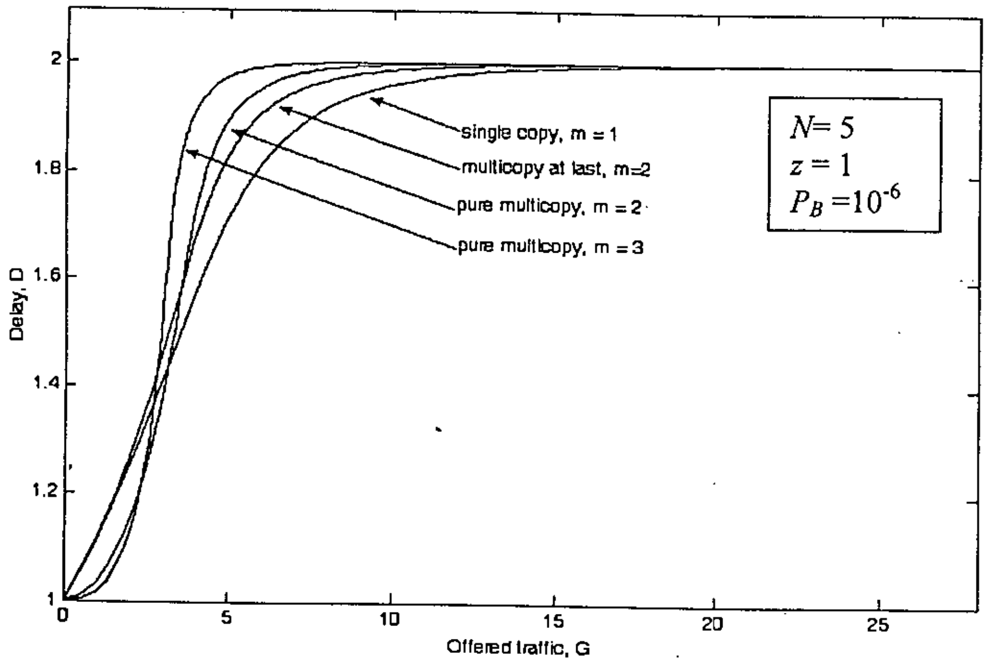

Figure 5.27: Delay vs. offered traffic in uncoded captured system for small number of channels and low bit error rate.

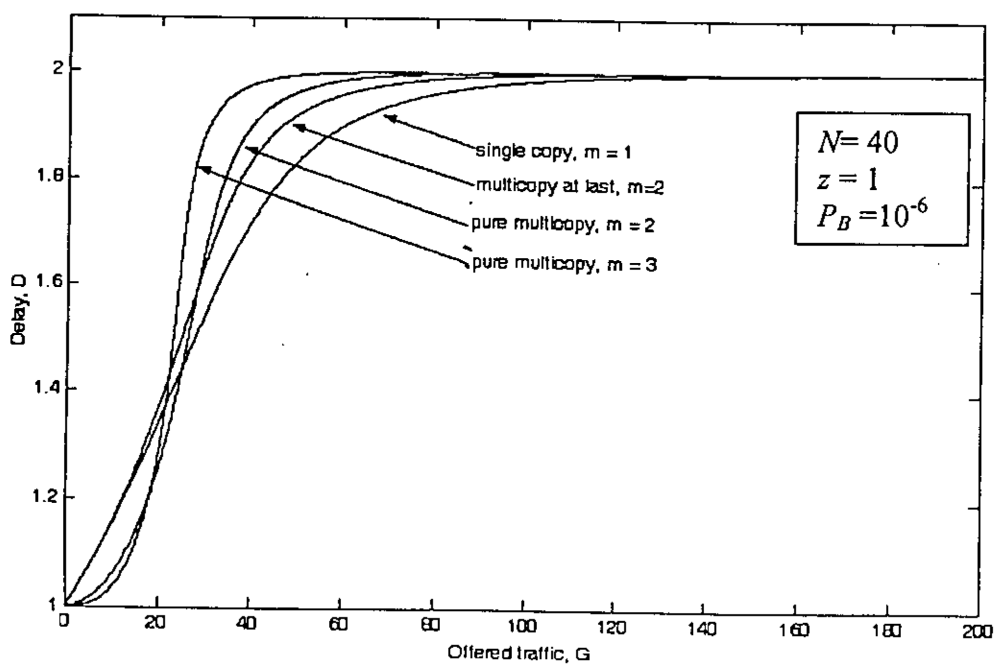

Figure 5.28: Delay vs. offered traffic in uncoded captured system for large number of channels and low bit error rate.

When G=60, delay is 1.9 for single copy and is 2 for m=3. Upto traffic G=18.5, m=3 gives the least delay. From G=18.5 to G=25, m=2 gives the least delay and after G=25, single copy gives the best delay performance. Delay saturates to 2, after channel traffic of G=180.

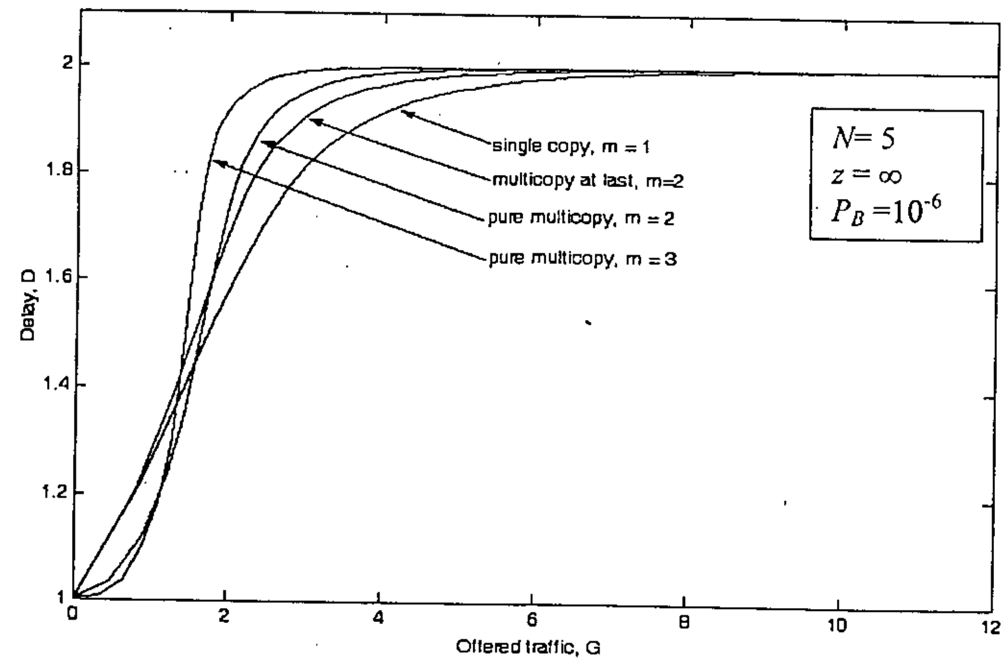

Figure 5.29: Delay vs. offered traffic in coded uncaptured system for small number of channels and low bit error rate.

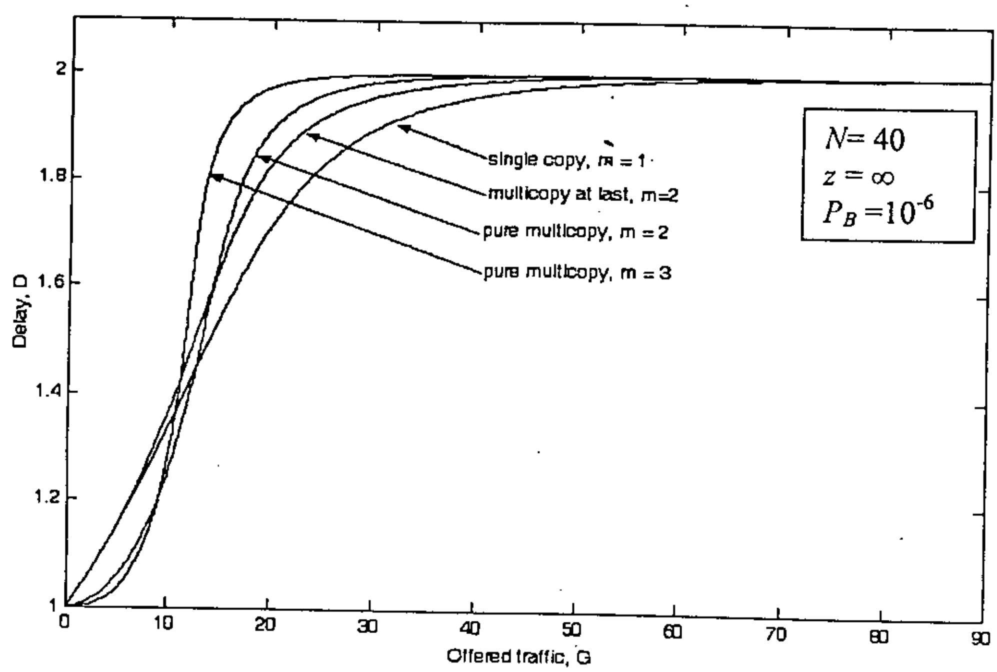

Figure 5.30: Delay vs. offered traffic in coded uncaptured system for large number of channels and low bit error rate.

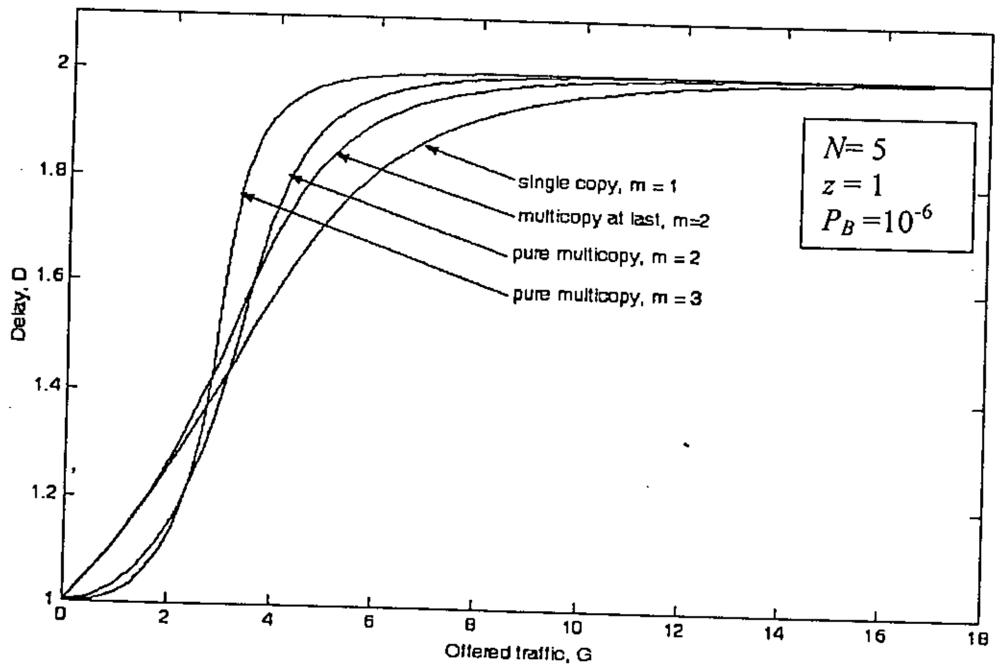

Figure 5.31: Delay vs. offered traffic in coded captured system for small number of channels and low bit error rate.

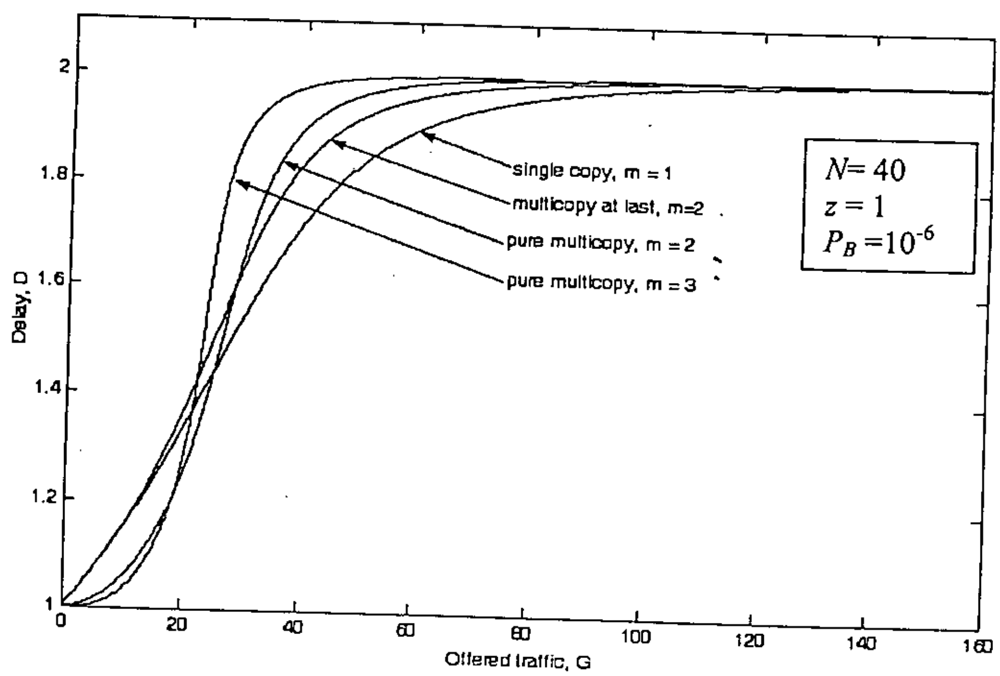

Figure 5.32: Delay vs. offered traffic in coded captured system for large number of channels and low bit error rate.

Effect of Reed Solomon coding on the delay of system of figure 5.25, figure 5.26, figure 5.27 and figure 5.28 is shown in figure 5.29, figure 5.30, figure 5.31 and figure 5.32

respectively. It is seen that coding can contribute no significant improvement in the system delay.

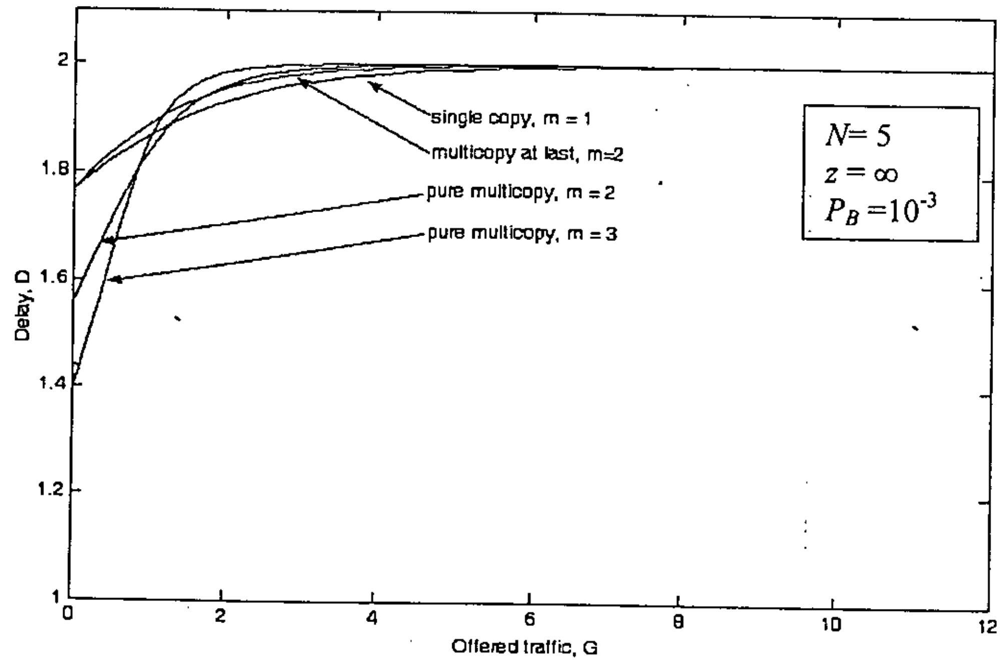

Figure 5.33: Delay vs. offered traffic in uncoded uncaptured system for small number of channels and high bit error rate.

Figure 5.33 shows delay plotted against channel traffic for number of channel 5 and bit error rate 10^{-3} with no capture. Delay is higher compared to low BER ($=10^{-6}$) system. Around channel traffic G=1, pure multicopy gives better delay. When G>1, single copy transmission gives the best delay performance. Multicopy at last attempt is worse than single copy but better than pure multicopy after G=1. After G=10, all transmission scheme gives the constant delay of 2.

In figure 5.34, effect of increase of number of channel is shown. Increase of channel allows higher traffic. Delay is better upto traffic G=6.5 for m=3. From G=6.5 to G=10, m=2 gives better delay. Beyond G=10, single copy transmission is the best policy with respect to delay performance. After G=85, delay for all transmission scheme saturates to 2.

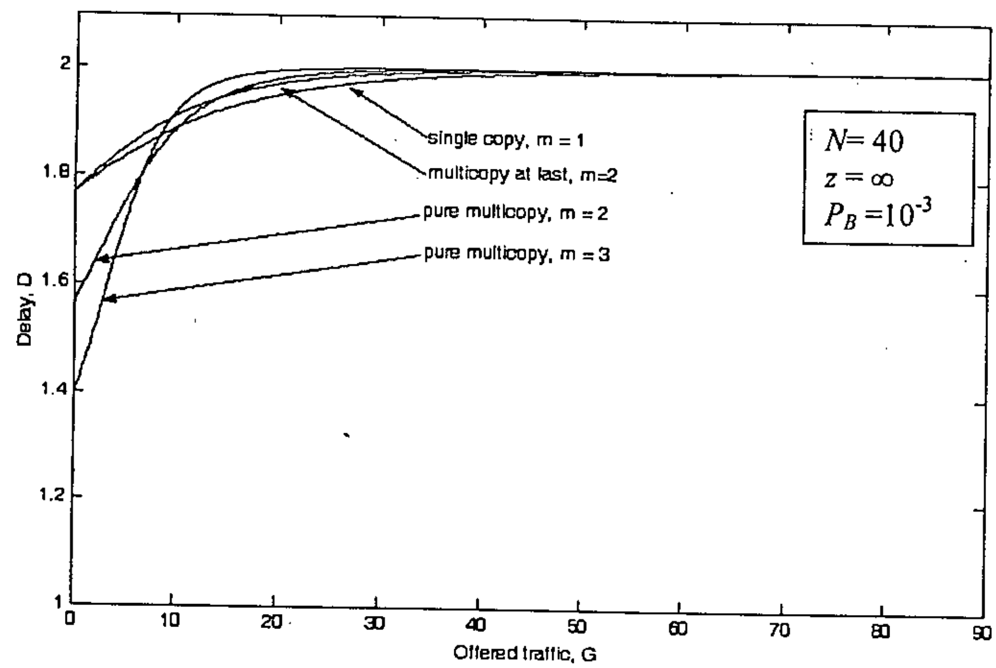

Figure 5.34: Delay vs. offered traffic in uncoded uncaptured system for large number of channels and high bit error rate.

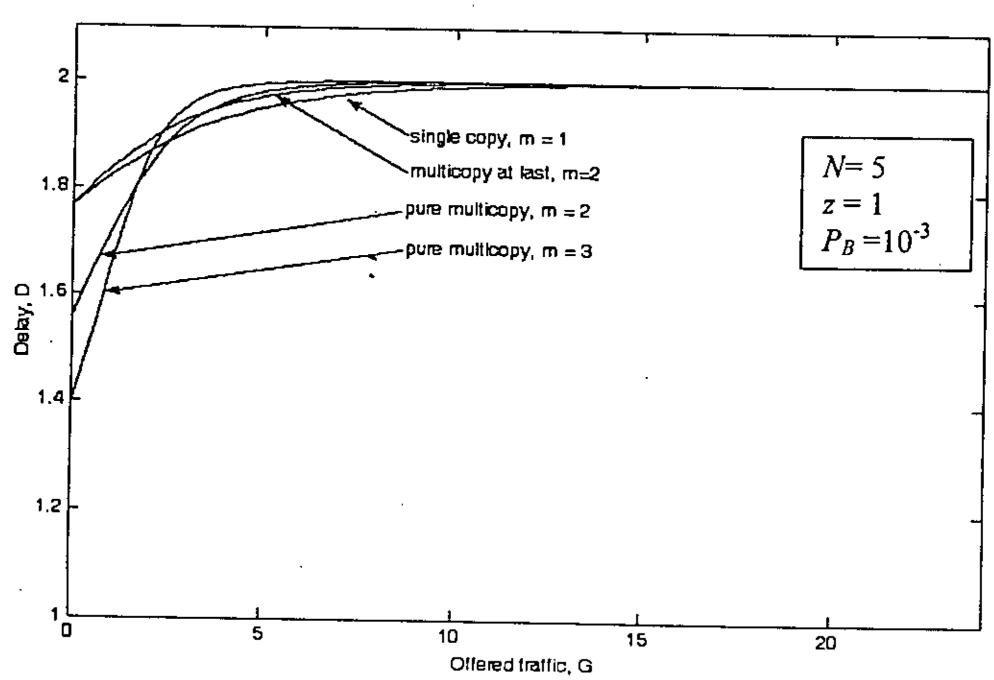

Figure 5.35: Delay vs. offered traffic in uncoded captured system for small number of channels and high bit error rate.

Figure 5.35 shows effect of capture on the delay with other conditions being same as in figure 5.33. Upto G=1.5, m=3 gives better delay and from G=1.5 to G=2.5, m=2 gives better delay. After G=2.5, single copy transmission gives the best delay performance. Multicopy at last attempt gives lower delay than pure multicopy transmission but higher than single copy after G=2.5. Delay becomes constant at 2 after G=17 for all transmission schemes.

Effect of increase of number of channel on delay in the captured uncoded system is shown in figure 5.36. It is seen that system now allows more traffic load. Compared to system in figure 5.34, the same delay can be obtained at almost double traffic. Delay is better upto traffic G=13 for m=3. From G=13 to G=20, m=2 gives better delay. Beyond G=20, single copy transmission is the best policy with respect to delay performance. After G=166, delay for all transmission scheme saturates to 2.

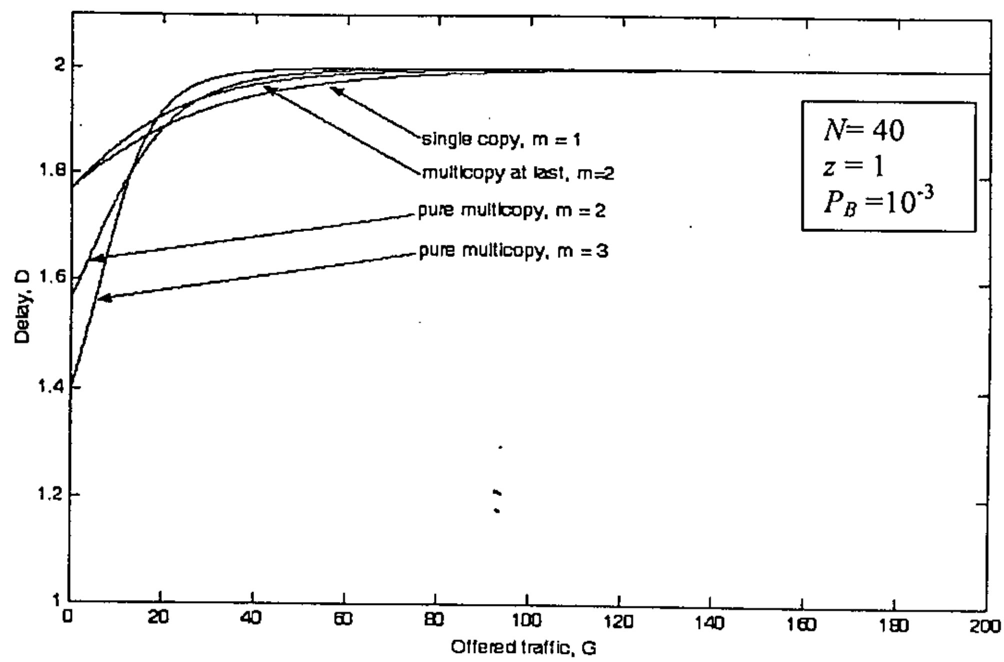

Figure 5.36: Delay vs. offered traffic in uncoded captured system for large number of channels and high bit error rate.

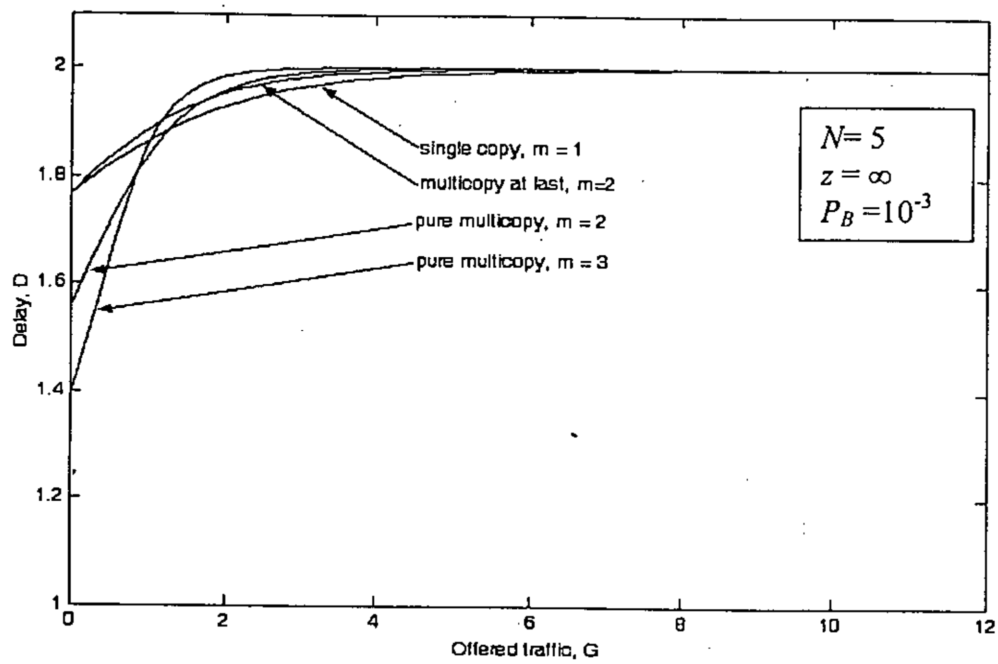

Figure 5.37: Delay vs. offered traffic in coded uncaptured system for small number of channels and high bit error rate.

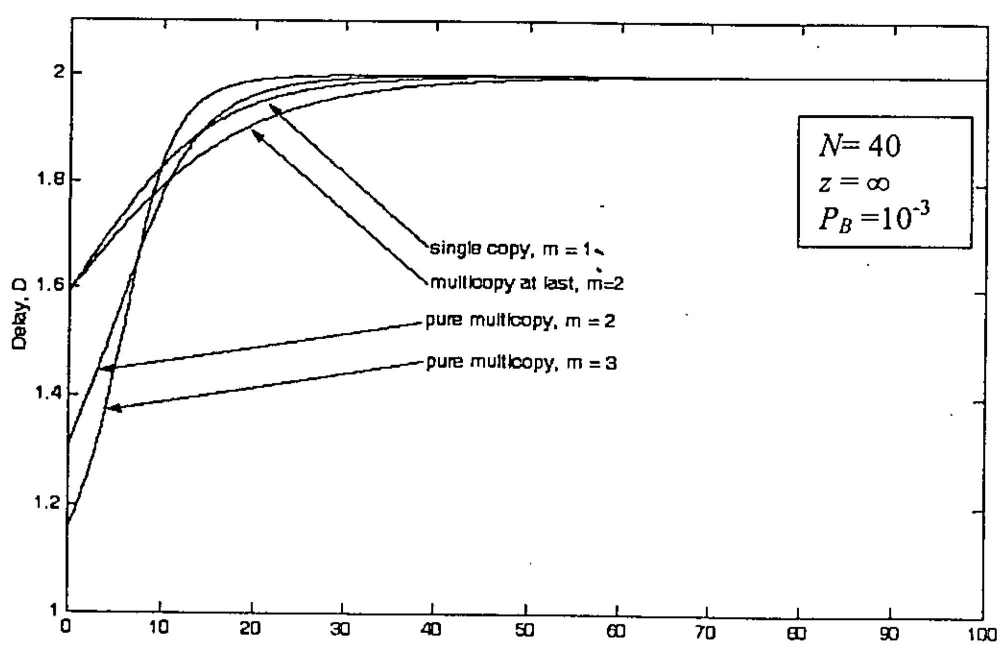

Figure 5.38: Delay vs. offered traffic in coded uncaptured system for large number of channels and high bit error rate.

Figure 5.37 shows the effect of coding on delay with other conditions being as same as in the system of figure 5.33. It is seen that in both cases curves are of the same shape and same delay occurs at the same traffic. So coding results no improvement here.

Effect of coding on the system of figure 5.34 is shown in figure 5.38. Now the same delay occurs at higher traffic. Delay is better upto traffic G=7.5 for m=3. From G=7.5 to G=11, m=2 gives better delay. Beyond G=11, single copy transmission is the best policy with respect to delay performance. After G=90, delay for all transmission scheme saturates to 2.

Figure 5.39 shows effect of Reed Solomon coding on the system of figure 5.35. Coding results improvement of delay performance here. For example, when G=2 and m=3, delay in coded system is 1.7 whereas in uncoded system it is 1.86. Upto G=2, m=3 gives better delay and from G=2 to G=2.7, m=2 gives better delay. After G=2.7, single copy transmission gives the best delay performance. Multicopy at last attempt gives lower delay than pure multicopy transmission but higher than single copy after G=2.7. Delay becomes constant at 2 after G=20 for all transmission schemes.

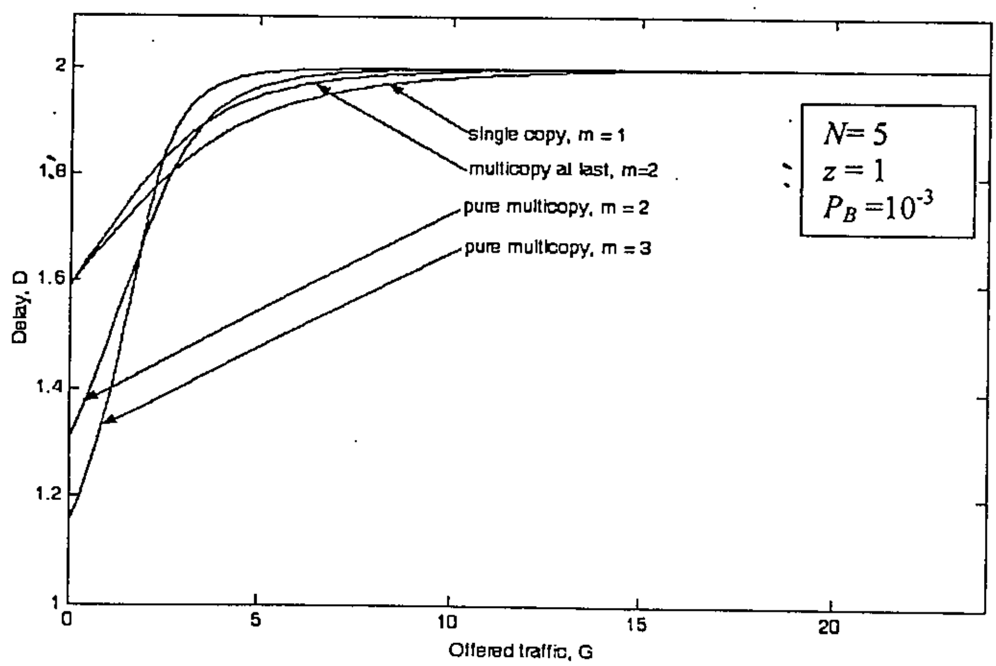

Figure 5.39: Delay vs. offered traffic in coded captured system for small number of channels and high bit error rate.

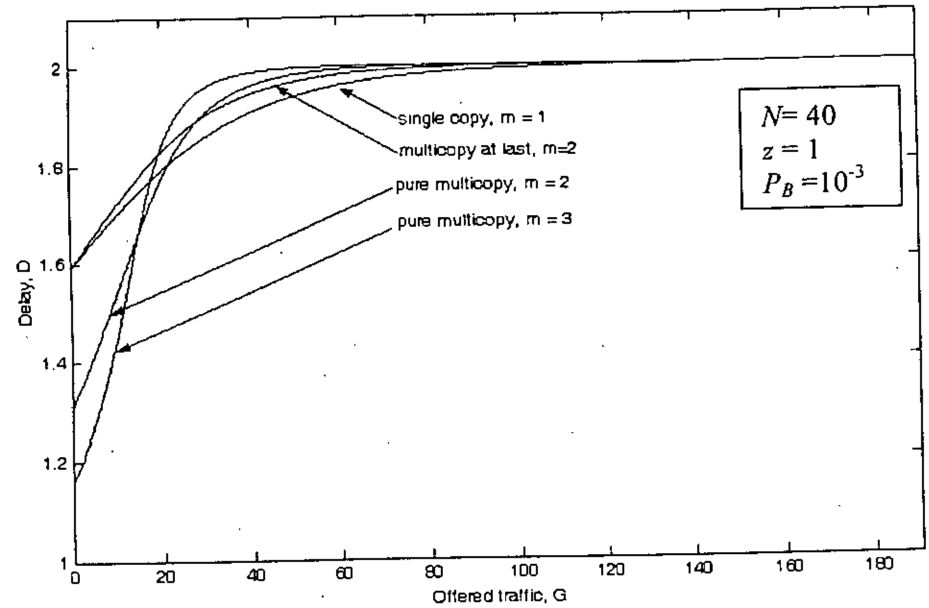

Figure 5.40: Delay vs. offered traffic in coded captured system for large number of channels and high bit error rate.

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In figure 5.40, effect of coding is shown on the system of figure 5.36. There is some improvement in system delay now. For instance, for single copy transmission and G=20, uncoded system shows delay of 1.89 whereas coded system exhibits delay of 1.79. Delay is better upto traffic G=15 for m=3. From G=15 to G=22, m=2 gives better delay. Beyond G=22, single copy transmission is the best policy with respect to delay performance. Delay for all transmission scheme saturates to 2 after G=180.

5.6 Comparison with Other ALOHA Protocols

The maximum throughput of OFDM based ALOHA system can be compared with other existing system to evaluate performance improvement obtained from the proposed system in this thesis. Following table shows the comparison.

Table 5.2: Comparison of Different ALOHA Protocols

System	Maximum Throughput (per slot)
Pure ALOHA (TDMA) [12]	18.4%
Slotted ALOHA (TDMA) [12]	36.8%
Slotted ALOHA (CDMA) [16]	46%
Slotted ALOHA (OFDM)	74%

From the above table, it is evident that OFDM based ALOHA provides much better throughput than the other systems.

Chapter 6

Conclusions and Future Work

6.1 Conclusions of This Study

From the system model, system analysis and simulation numerous conclusions can be drawn. Though ALOHA is one of the oldest random access schemes, but this work reveals that it can still be integrated with the promising high speed future technology like OFDM. It has been shown that OFDM based ALOHA can be fruitful in different channel BER conditions. Here performance of the system in two different channel BER condition has been thoroughly analyzed. It is found that in worse channel BER condition, OFDM based ALOHA can enhance system performance significantly. As seen from simulation results, OFDM based ALOHA exhibits more improvement in packet success rate, normalized throughput and delay performance in high channel BER condition than in low channel BER condition. OFDM technology commonly applies Reed Solomon forward error correction coding. So this work also compares system performance under coded and uncoded condition. Simulation results indicate that coding can be much more effective when channel condition becomes worse. Capture effect has been implemented in most of the previous ALOHA schemes and all past researches showed that capture improves the performance. So this work also investigates effect of capture and like previous research this research also has proved the claim that capture can contribute to system performance. In a general sense it seems that if number of channels is increased, system performance will automatically increase. The simulation results obtained here also supports this inference. Another important aspect this study uncovers is that with the channel loading condition being informed by the base station to the users, the users can decide on which transmission scheme is to be followed. For example, in peak time when channels are heavily loaded with traffic, users can implement single copy transmission scheme whereas in off peak or lightly loaded condition they may switch to pure multicopy transmission. And for moderately loaded channels, users may follow multicopy at last attempt approach. Thus users can adopt appropriate scheme to achieve the optimum performance from the system. The performance of OFDM based ALOHA protocol is compared with other ALOHA protocols and it is found that significant improvement is obtained from the system proposed in this thesis. Therefore this study proves OFDM based ALOHA to be a potential random access scheme to meet the future huge demand of high speed performance with greater flexibility and effectiveness.

6.2 Suggestions for Future Works

Though this research unveils performance of OFDM based ALOHA in different conditions and with numerous variable parameters, still there are scopes for future improvement of this work. Some possible future research works related to this work may be as follows:

1. This work focuses only on coding effect. But with coding, bandwidth requirement is varied whose effect has not been investigated here.
2. The work can be extended to other channel BER conditions also.
3. Code efficiency or code rate r is defined as ratio of k and n, i.e, $r = k/n$. Error correction capability can be varied by varying code rate. This work is carried out only at one code rate (r =24/32=3/4). So performance can be evaluated with other code rates.
4. Slotted ALOHA has been shown to be bistable protocol [31]. Bistability exhibits in heavily loaded channels with high retransmission deadline, L. However in this work only a fixed retransmission deadline (L=3) has been chosen to avoid bistable condition. Therefore, stability of the system yet to be analyzed.
5. For implementation of capture, z=1 has been used. However in [32], it has been shown that z should be varied with channel traffic condition to achieve higher throughput which is not considered here. So performance can be investigated under varying capture condition.

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