DESIGN OF A JAMMER FOR SPREAD SPECTRUM SIGNAL

USING MATLAB SIMULATION

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ABSTRACT

DESIGN OF A JAMMER FOR SPREAD SPECTRUM SIGNAL USING MATLAB SIMULATION

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The term spread spectrum is today one of the most popular terms in the radio engineering and communication community. Since, code division multiple access "CDMA" is a form of spread spectrum, a family of digital communication techniques that have been used in military applications for many years. Originally there were two motivations: either to resist enemy efforts to jam the communications "anti-jam, or AJ", or to hide the fact that communication was even taking place, sometimes called low probability of intercept "LPI". It has a history that goes back to the early days of World War II. In the late 1940's the proposed theoretically use of CDMA for civilian mobile radio applications is novel, but the practical application in the civilian marketplace did not take place until 40 years later.

Nowadays, because of the dramatic growth of using the communication devices in every day's life, and since this usage is sometimes has to be disturbed for some reasons, so to achieve this jammers are used. In practice, many types of jammers such as partial-band noise jammer, pulsed noise jammer, and single-tone jammer are used. While this thesis studies the jammer design differently, by using the same spreading codes which spreads the message signals.

This thesis addresses a study of designing a jammer for spread spectrum signal by analyzing the performance of the system using the MATLAB simulation. To achieve this, after a brief introduction the principles of the spread spectrum systems are explained in details, then the techniques of the detection which are used to retrieve the message signal are presented. Finally, the implementation of the design based on using the same spreading codes, which are used by the jammed system itself. Each of Gold, PN, and Kasami codes is used not only to spread the message signals but also to jam them. The channel is assumed to be an Additive White Gaussian Noise “AWGN”, and the suboptimum detector is used to retrieve the messages. According to the simulation results the designed jammer perform close to the theoretical jammer at a low signal to noise ration “SNR”, and increasing the code length diverge the designed jammer from the theoretical, in contrast the effect of increasing the number of users is opposite.

Keywords: spread spectrum system, CDMA, multiuser detection, spreading codes, jammer design, MATLAB simulation, AWGN.

ÖZET

YAYILI SPEKTRUM SİNYALİ İÇİN MATLAB SİMÜLASYONU İLE SİNYAL BOZUCU TASARIMI

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Ağustos 2012, 66 sayfa

Yayılı spektrum terimi radyo mühendislik ve iletişim toplumunda günümüzün en gözde terimlerindendir. Kod bölmeli çoklu erişim “CDMA” bir yaylı spektrum oluşumu ve yıllarca askeri uygulamalar için kullanılmış olan sayısal iletişim teknikleri takımıdır. Başlangıçta iki tür güdülenim vardı: ya düşman kuvvetlerine karşı koymak amacıyla iletişim sinyalini bozmak “anti jam, veya AJ” ya da gerçekleşen iletişimi gizlemek, bu düşük olasılıklı algılama “LPI” olarak adlandırılabilir. Bunun İkinci Dunya Savaşının ilk günlerine dayanan bir geçmişi vardır. 1940’ların sonlarında CDMA’nın sivil gezgin radyo uygulamaları için önerilen teorik kullanımı alışılmışın dışındaydı ancak sivil markette pratik uygulama 40 yıl sonrasına kadar gerçekleşmedi.

Günümüzde, iletişim aygıtlarının günlük hayatta kullanımının kayda değer artışı nedeniyle ve bu kullanımın bazı nedenlerle ve bazı durumlarda engellenmesi gerektiğinden, bu amaç doğrultusunda sinyal bozucular kullanılmaktadır. Uygulamada, bölümlü-frekans bandı gürültülü karıştırıcı, darbeli gürültülü karıştırıcı, tek tonlu sinyal bozucu gibi çok çeşitli sinyal bozucular kullanılmaktadır. Bu tezde, mesaj sinyallerini gönderen aynı yayılma kodlarının kullanılmasıyla farklı açıdan sinyal bozucu tasarımı araştırılmaktadır.

Bu tez, MATLAB simülasyonu kullanılarak sistem performansının analiz edilmesiyle yayılı spektrum sinyali için sinyal bozucu tasarımı çalışmasını ele alır. Bu amaçla, kısa bir girişin ardından yayılı spektrum sistemlerinin prensipleri detaylı olarak açıklanmış, sonrasında mesaj sinyallerinin alınmasında kullanılan algılama teknikleri sunulmuştur. Sonuç olarak, karışmış sistemin kendisi tarafından kullanılan aynı yayılma kodlarının kullanılmasına dayanan tasarımın uygulaması sunulmustur. Gold, PN, Kasami kodlarının her biri, sadece mesaj sinyallerinin yayılması için değil aynı zamanda bu sinyallerin bozulması için de kullanılmaktadır. Kanal, Toplanır Beyaz Gauss Gürültüsü "AWGN" olarak farz edilir ve yarı optimal detektör mesajların alınması için kullanılır. Simülasyon sonuçlarına göre, tasarlanan sinyal bozucu, düşük sinyal gürültü oranında "SNR", teorik sinyal bozucuya yakın bir performans sergilemiştir ve kod mesafesinin artırılması tasarlanan sinyal bozucuyu teorikten uzaklaştırmıştır, kullanıcıların sayısının artırılmasıyla tam tersi durum gerçekleşir.

Anahtar Kelimeler: yayılı spektrum sistemi, CDMA, çok kullanıcılı algılama, kodların yayılması, sinyal bozucu tasarımı, MATLAB simülasyonu, AWGN.

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There are numerous people that contributed to the completion of this thesis. First and foremost, a sincere thanks is extended to my supervisor Assoc. Prof. Halil Tanyer Eyyuboglu for his guidance, support and invaluable advice.

I would like to express my deepest gratitude to my parents for giving me the opportunity to build a successful life. Gratitude is also extended to my husband for his assistance and support in one way or another.

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LIST OF ABBREVIATION

CDMA:	Code Division Multiple Access.
AJ:	Anti-Jam.
LPI:	Low Probability of Intercept.
AWGN:	Additive White Gaussian Noise.
FDM:	Frequency Division Multiplexing.
FDMA:	Frequency Division Multiple Access.
TDM:	Time Division Multiplexing.
TDMA:	Time Division Multiple Access.
SNR:	Signal to Noise Ratio.
DSSS:	Direct Sequence Spread Spectrum.
FH:	Frequency Hopping.
PN sequence:	Pseudorandom or Pseudo Noise sequence.
PSK:	Phase Shift Keying.
FSK:	Frequency Shift Keying.
P_e:	The Average Probability of Error.
MAI:	Multiple Access Interference.
MUD:	Multiuser Detection.

LIST OF SYMBOLS

R_b:	Information bandwidth or information rate.
R_c:	Transmission bandwidth.
T_b:	Bit interval.
B_c:	The available channel bandwidth.
W:	The expanded bandwidth of the information signal.
b_n:	Represents the transmitted bit of the message signal.
$g(t)$:	Shaping waveform
e_i:	Represents the spreading code, or signature sequence.
T_c:	The chip duration.
N:	The code length.
$p\ (t)$:	Shaping waveform
$i(t)$:	The interference
I_0:	The power-spectral density of the wideband interference.
P_I:	The average power of the interference.
P:	The common average transmitted power.
$R_{k,i}, \hat{R}_{k,i}$:	The continuous-time partial crosscorrelation functions.
$\tau_k, \emptyset_k$:	The time delays.
$\mu_{k,j}(n)$:	The crosscorrelation parameters.
E:	The energy per data symbol.
$\Phi(.)$:	The standard Gaussian cumulative distribution function.
$\eta_0/2$:	The noise density.
K:	The number of users.
τ:	Time offsets.
m:	The shift register length.

$S_w(f)$: The power spectral density of white noise.

J: The average jammer power.

$y(t)$: The received signal.

A_k: The received amplitude for the k^{th} user.

$n(t)$: AWGN.

ρ_{jk}: The cross-correlation of the signature sequences.

$C(y_k, b_k)$: Correlation metrics.

R_s: The correlation matrix.

CHAPTER I

INTRODUCTION

This chapter provides an overview to this thesis, and gives a small entry to the basic background to the main aim of the thesis. Moreover, it clarifies the thesis issues and it constructs the outlines of this work.

1.1 Background

The idea of using a communication channel to enable several transmitters to send information simultaneously dates back to Thomas A. Edison's 1873 invention of the duplex. This revolutionary system enabled the simultaneous transmission of two telegraphic messages in the same direction through the same wire. One message was encoded by changes of polarity; the other by changing absolute values.

Nowadays, there are numerous examples of multi-access communication in which several transmitters share a common channel: mobile telephones transmitting to a base station, ground stations communicating with a satellite, a bus with multiple taps, multidrop telephone lines, local area networks, packet-radio networks, and interactive cable television networks, to name a few. A common feature of those communication channels is that the receiver obtains the superposition of the signals sent by the active transmitters. Oftentimes, the superposition of signals sent by different transmitters occurs unintentionally owing to nonideal effects; for example, crosstalk in telephony and in multitrack magnetic recording or any time the same radio frequency band is used simultaneously by distant transmitters, as in cellular telephony, and radio/ television broadcasting. Although occasionally the terms multiplexing and multi-access are used interchangeably, multi-access usually refers to situations where the message sources are not collocated and/ or operate autonomously. The message sources in a multi-access channel are referred to as users.

Multi-access communication is sometimes referred to as multipoint-to-point communication. The engineering issues in the dual point-to-multipoint channel depend on the commonality of the information transmitted to each destination. At one extreme, the same information is delivered to all recipients, for example, in radio and television broadcasting or in cable television; at the other extreme, the messages transmitted to different recipients are independent, for example, a base station transmitting to mobile units. The latter scenario falls conceptually within the multi-access channel model; the receiver is interested in only one information sources transmitted by the base station [1].

1.2 Multi-access Communication

The advent of radio-frequency modulation in the early 20th century enable several radio transmissions to coexist in time and space without mutual interference by using different carrier frequencies. The same idea was used in long-distance wire telephony. Frequency Division Multiplexing “FDM” or Frequency Division Multiple Access “FDMA” assigns a different carrier frequency to each user so that the resulting spectra do not overlap as shown in Figure 1.1. Band-pass filtering enables separate demodulation of each channel.

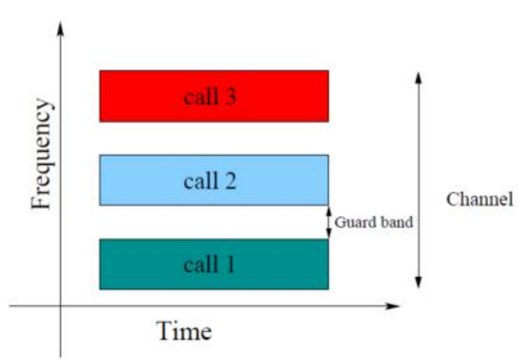


Figure 1.1 Frequency division multiple access.

In Time Division Multiplexing “TDM”, time is partitioned into slots assigned to each incoming digital stream as in Figure 1.2. Demultiplexing is carried out by simply switching on to the received signal at the appropriate epochs. Time division can be used not only to multiplex collocated message sources but also by geographically separated users who have the ability to maintain time-synchronism, in what is commonly referred to as Time Division Multiple Access “TDMA”. Note that FDMA allows completely uncoordinated transmissions in the time domain: no time-synchronization among the users is required. This advantage is not shared by TDMA where all transmitters and receivers must have access to a common clock.

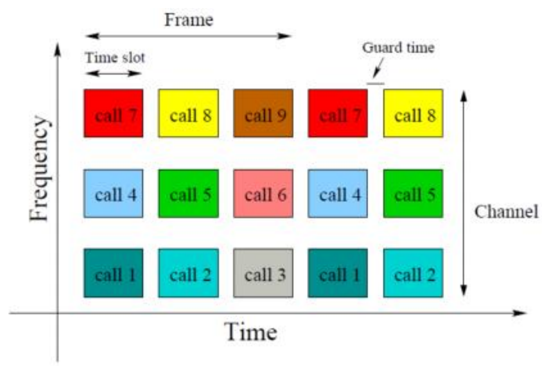


Figure 1.2 Time division multiple access.

1.3 Random Multi-access

Random multi-access communication is one of the approaches to dynamic channel sharing. When a user has a message to transmit it goes ahead and transmits it as if it were the sole user of the channel. If nobody else is transmitting simultaneously, then the message is received successfully. However, the users are uncoordinated and the possibility always exists that the message will interfere “in time and frequency” with another transmission. In such case, it is typically assumed in random multi-access communication that the receiver cannot demodulate reliably several simultaneous messages. The only alternative left is to notify the transmitters that a collision has happened and, thus, their messages have to be retransmitted. Under those circumstances, the channel sharing approaches which discussed so far are violated in random access communication, since the receiver is unable to recover any of the colliding transmissions. In fact, reception free from interchannel interference is a consequence of the use of orthogonal signaling. It is important to realize that this can be accomplished by signals that overlap both in time and in frequency. Hence, at any given time a subset of users may transmit information simultaneously over the common channel to corresponding receivers. Assuming that all the users employ the same code for the encoding and decoding of their respective information sequences, the transmitted signals in this common spectrum may be distinguished from one another by superimposing a different pseudorandom pattern "code" in each transmitted signal. Thus a particular receiver can recover the transmitted information intended for it by knowing the pseudorandom pattern. This type of communication technique which allows multiple users simultaneously to use a common channel for transmission of information is called Code Division Multiple Access "CDMA", as illustrated in Figure 1.3.

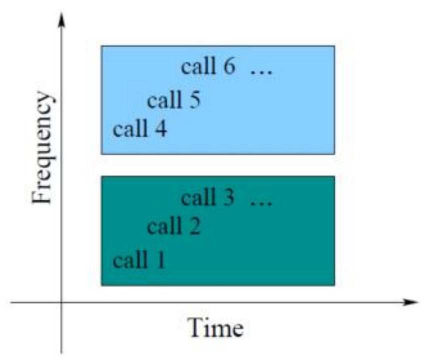


Figure 1.3 Code division multiple access.

1.4 Motivation of the Work

At the beginning of the spread spectrum systems development, they were originally used for military communications either to hide the signal by transmitting it at low power or to provide resistance to jamming. However, spread spectrum systems are now used to provide reliable communications in a variety of civilian applications, including digital cellular communication. This reliability "anti-jamming facility" has to be broken for some issues such as the military issues, normally the system is jammed by transmitting a signal that disturbs it by decreasing the signal to noise ratio “SNR”. Nevertheless, this thesis will jam the spread spectrum system by using the codes, which are in use in the system itself. Since, this kind of study are for special issues, for instant the security matters in the government, or in the war business. Thereupon, a jammer will be designed by spreading a random message signals with the same spreading sequences in the aimed system and transmit them to disturb the system, then the probability of error of the system will be examined under different circumstances.

1.5 Thesis Outline

This thesis is organized in five chapters; the introduction chapter of this thesis gives a background of the multi-access communications, and explains its techniques, moreover demonstrate the random multi-access communications.

Chapter two develops a framework for understanding the direct sequence spread spectrum “DS-SS” communication system. It begins presenting the fundamentals of spread spectrum techniques which represent the basis of CDMA technologies. Since DS-CDMA systems are the main concern in this work, direct sequence spread spectrum technique is only discussed. Also, represent the signature sequences, which are used in the DS-CDMA systems.

Chapter three addresses the detection techniques for the multiuser systems, this chapter will explain this issue from two different approaches; the first approach is based on the concept of single user detection, the second approach is called multiuser detection.

The existing results from the thesis work are presented in chapter four, where the effect of each number of users, the length of the spreading codes, the code type are all examined when the jammer designed.

Finally, chapter five concludes this thesis. It reviews and highlights the contributions that have been presented in this work. Furthermore, the suggestions for the future work are discussed too.

CHAPTER II

SPREAD SPECTRUM AND CODE-DIVISION MULTIPLE ACCESS

First, the spread spectrum system is introduced and modeled. Then, each of the direct sequence spread spectrum principles, the codes sequences, and the probability of error of the system is presented. Finally, a glimpse from the noise and the jamming is explained in this chapter.

2.1 Introduction

In many systems, due to a limited spectrum, multiple users must share the same band of frequencies. Some examples are cellular telephones, radio dispatch offices for taxi companies, and air-traffic control communications. There are many ways of sharing the spectrum efficiently as described in the previous chapter; this chapter will focus on the approach of sharing the radio spectrum by using CDMA. CDMA refers to a multiple access technique in which the individual terminals use spread spectrum techniques and occupy the entire spectrum whenever they transmit. It is the interference attenuation property of spread spectrum that allows multiple users to occupy the same spectrum at the same time.

Spread spectrum systems encompass modulation techniques in which the signal, with an information bandwidth R_b, is spread to occupy a much larger transmission bandwidth R_c. Spread spectrum techniques were originally developed for military applications, but commercial interest in such techniques has increased recently, due to their promise of greater tolerance for interference. The impetus for this interest has been twofold, the growth of cellular telephony systems; and regulatory rulings that have allowed the unlicensed use of some frequency bands for low power transmitters as long as the signal is spread to reduce interference.

There are two basis spread spectrum techniques: direct sequence “DS” and frequency hopping “FH”. Also a variety of hybrid techniques use different combinations of these two basic techniques. These techniques have increased the tolerance to interference, increased the tolerance to multipath, and increased the ranging capability. With direct sequence spreading the original signal is multiplied by a known signal of much larger bandwidth. With frequency hopped spreading the center frequency of the transmitted signal is varied in a pseudorandom pattern.

2.2 Model of a Spread Spectrum Digital Communication System

The basic elements of a spread spectrum digital communication system are illustrated in Figure 2.1. According to the diagram the channel encoder and decoder and the modulator and the demodulator are the basic elements of a conventional digital communication system. In addition to these elements, two identical pseudorandom sequence generators are employed, one interfaces with the modulator, and the other with the demodulator at the transmitting and receiving ends respectively. The generators produce a pseudorandom or pseudo noise “PN” binary valued sequence, which is used to spread the transmitted signal and despread the received one [4].

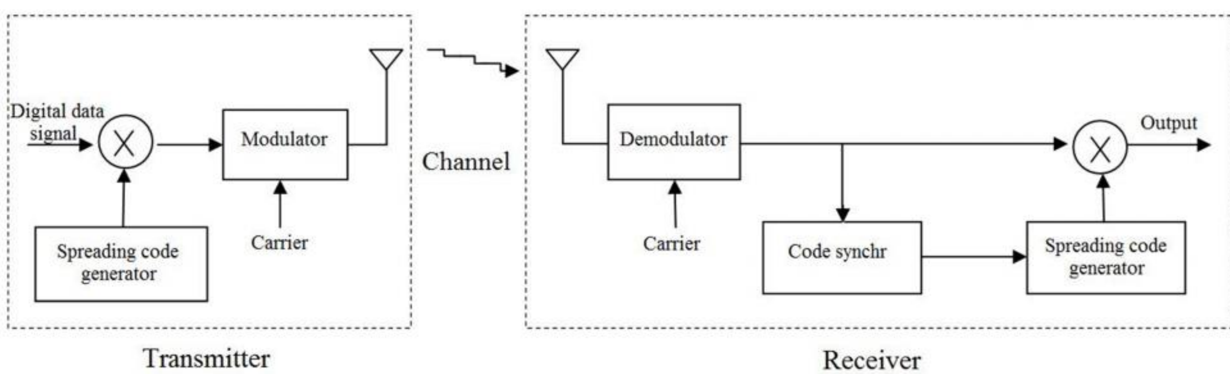


Figure 2.1 Model of spread spectrum digital communication system [4].

Time synchronization of the PN sequence generated at the receiver with the PN sequence contained in the received signal is required in order to properly despread the received signal. Initially, prior to the transmission of information, synchronization may be achieved by transmitting a fixed pseudorandom bit pattern that the receiver will recognize in the presence of interference with a high probability [2].

Both Phase Shift Keying “PSK” and Frequency Shift Keying “FSK” modulation types are employed in spread spectrum communication systems. PSK is appropriate in applications where phase coherence between the transmitted signal and the received one can be maintained over a time interval that is relatively long compared to the reciprocal of the transmitted signal bandwidth. On the other hand, FSK modulation is appropriate in applications where such phase coherence cannot be maintained due to time variant effects on the communications link.

The PN sequence generated at the modulator is used in conjunction with the PSK modulation to shift the phase of the PSK signal pseudorandomly at a rate that is an integer multiple of the bit rate, the modulated signal is called a direct sequence spread spectrum “DSSS” signal. When used in conjunction with FSK, the PN sequence is used to select the frequency of the transmitted signal pseudorandomly, then the modulated signal is called a frequency hopped spread spectrum signal. Only the DSSS will be studied in this thesis.

2.3 Direct Sequence Spread Spectrum

Since spread spectrum signaling formats feature large duration bandwidth products. Direct sequence refers to a specific approach to construct spread spectrum waveforms; to illustrate this let us consider the transmission of a binary information sequence by means of binary PSK. The information rate is R_b bits/sec and the bit interval is $T_b = 1/R_b$ sec. the available channel bandwidth is B_c Hz, where $B_c \gg R$. At the modulator, the bandwidth of the information signal is expanded to $W = B_c$ Hz by shifting the phase of the carrier pseudorandomly at a rate of W times/sec according to the pattern of the PN generator. To demonstrate the spreading process let us first start with the information baseband signal which can be expressed as:

$$m(t) = \sum_{n=-\infty}^{\infty} b_n g(t - nT_b) \tag{2.1}$$

Where;

$b_n \in \{1, -1\}$, represents the transmitted bit of the message signal.

T_b is the duration of one symbol.

$g(t)$ Shaping waveform, $g(t) = \begin{cases} 1 & 0 < t < T_b \\ 0 & otherwise \end{cases}$

This signal is multiplied by the signal from the PN sequence generator, which can be expressed as:

$$e(t) = \sum_{i=0}^{N-1} e_i\, p(t - iT_c) \tag{2.2}$$

Where;

$e_i \in \{1, -1\}$, represents the spreading code, or signature sequence.

T_c is the chip duration.

N is the code length "$N = \frac{T_b}{T_c}$; $T_b \gg T_c$".

$p\,(t)$ is the shaping waveform, $p(t) = \begin{cases} 1 & 0 < t < T_c \\ 0 & otherwise \end{cases}$.

This multiplication operation, as shown in Figure 2.2, serves to spread the bandwidth of the information signal, whose bandwidth is R_b Hz, into the wider bandwidth occupied by PN generator signal $e(t)$, whose bandwidth is $1/T_c$. The spectrum spreading is illustrate in Figure 2.3, which shows the convolution of the two spectra, the narrow spectrum corresponding to the information signal and the wide spectrum corresponding to the PN sequence. The product signal $m(t)e(t)$, is used to amplitude modulate the carrier $A_c \cos(2\pi f_c t)$ and to generate the DSB-SC signal

$$s(t) = A_c m(t) e(t) \cos(2\pi f_c t) \tag{2.3}$$

Since $m(t)e(t) = \mp 1$ for any t, it follows that the carrier-modulated transmitted signal can also expressed as:

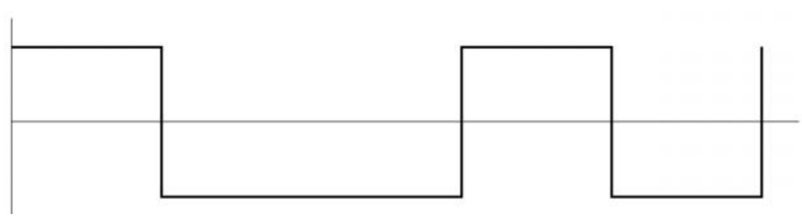
$$s(t) = A_c \cos\big(2\pi f_c t + \theta(t)\big) \tag{2.4}$$

Where $\theta(t) = 0$ when $m(t)e(t) = 1$, and $\theta(t) = \pi$ when $m(t)e(t) = -1$. Therefore, the transmitted signal is a binary PSK signal.

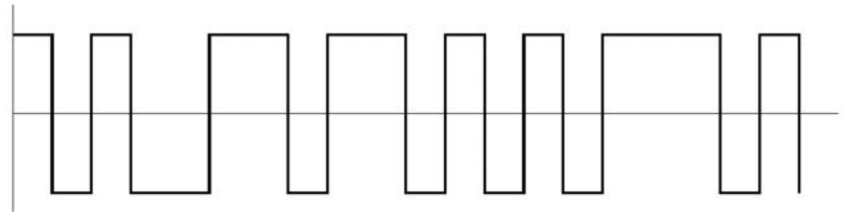
In other words, since the rectangular pulse $p(t)$ is called a chip, and its time duration T_c is called the chip interval, also the reciprocal $1/T_c$ is called the chip rate, and corresponding to the bandwidth W of the transmitted signal. The ratio of the bit interval T_b to the chip interval T_c is usually selected to be an integer, which denoted as:

$$N = \frac{T_b}{T_c} \tag{2.5}$$

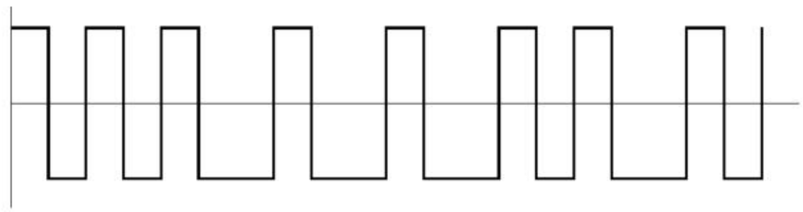
Hence, T is the number of chips of the PN code sequence/information bit. Another interpretation is that T represents the number of possible 180° phase transitions in the transmitted signal during the bit interval T_b.



(a) Message signal



(b) PN sequence.



(c) Spread signal

Figure 2.2 Generation of a DS spread spectrum signal.

The demodulation of the signal is performed as illustrated in Figure 2.3. The received signal is first multiplied by a replica of the waveform $e(t)$ generator at the receiver, which is synchronized to the PN code in the received signal. This operation is called despreading, since the effect of multiplication by $e(t)$ at the receiver is to undo the spreading operation at the transmitter. Thus, it will be:

$$A_c m(t) e^2(t) \cos(2\pi f_c t) = A_c m(t) \cos(2\pi f_c t) \tag{2.6}$$

Since, $e^2(t) = 1$ for all t. The resulting signal $A_c m(t) \cos(2\pi f_c t)$ occupies a bandwidth R_b Hz, which is the bandwidth of the information signal. Since, the demodulator has a bandwidth that is identical to the bandwidth of the despread signal; the only additive noise that corrupts the signal at the demodulator is the noise that falls within the information-bandwidth of the received signal.

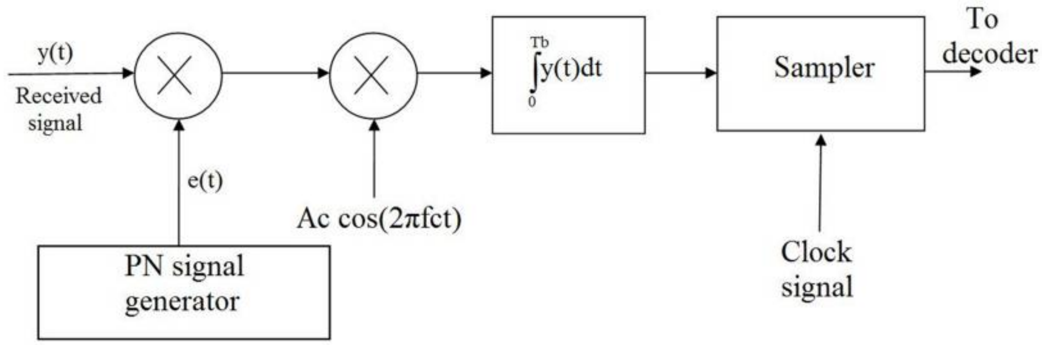


Figure 2.3 Demodulation of DS spread spectrum signal.

2.3.1 Effect of Despreading on Narrowband Interference

To investigate the effect of an interfering signal on the demodulation of the desired information signal let us suppose that the received signal is:

$$y(t) = A_c m(t) e(t) \cos(2\pi f_c t) + i(t) \tag{2.7}$$

Where $i(t)$ denotes the interference. The despreading operation at the receiver yields:

$$y(t)e(t) = A_c m(t) \cos(2\pi f_c t) + i(t)e(t) \tag{2.8}$$

The effect of multiplying the interference $i(t)$ with $e(t)$, is to spread the bandwidth of $i(t)$ to W Hz. Consider a sinusoidal interfering signal of the form:

$$i(t) = A_I \cos(2\pi f_I t) \tag{2.9}$$

Where f_I is a frequency within the bandwidth of the transmitted signal. Its multiplication with $e(t)$ results in a wideband interference with power-spectral density $I_0 = P_I/W$, where $P_I = A_I^2/2$ is the average power of the interference. Since the desired signal is demodulated by a matched filter that has a bandwidth R, the total power in the interference at the output of the demodulator is:

$$I_0 R_b = \frac{P_I R_b}{W} = \frac{P_I}{W/R} = \frac{P_I}{T_b/T_c} = \frac{P_I}{N} \tag{2.10}$$

Therefore, the power in the interfering signal is reduced by an amount equal to the bandwidth expansion factor W/R. The factor $W/R = T_b/T_c = N$ is called the processing gain of the spread-spectrum system. The reduction in interference power is the basic reason for using spread-spectrum signals to transmit digital information over channels with interference.

Overall, the PN code sequence is used at the transmitter to spread the information signal into a wideband to be transmitted over the channel. The desired signal is despread back by multiplying the received signal with a synchronized replica of the PN code, while any interference signals are a spread over a wide bandwidth. The net effect is a reduction in the interference power by the factor W/R, which is the processing gain of the spread-spectrum system [4].

2.4 Probability of Error for DS-CDMA

In the first place, studying the performance of the DS-CDMA has to start by considering the received signal for the synchronous system, which has *K* number of users as:

$$y(t) = \sum_{k=1}^{K} \sqrt{2P}\, b_k e_k(t-\tau_k) \cos(2\pi f_c t + \emptyset_k) + n(t) \tag{2.11}$$

Where *P* is the common average transmitted power, and $\emptyset_k = \theta_k + 2\pi f_c \tau_k$, also $n(t)$ is the additive white Gaussian noise process “AWGN” with two-sided spectral density $\eta_0/2$.

If the received signal $y(t)$ is the input to a correlation receiver matched to $s_k(t)$, the output is:

$$z_k = \int_0^{T_b} y\,(t) e_k(t) \cos(2\pi f_c t)\, dt \tag{2.12}$$

At the next step, the output of the correlation receiver at $t = T$ is given by

$$z_k = \sqrt{P/2}\left\{ b_{j,0}T \sum_{\substack{k=1 \\ k\neq j}}^{K} \left[b_{k,-1} R_{k,j}(\tau_k) \right.\right.$$

$$\left.\left. + \, b_{k,0} \hat{R}_{k,j}(\tau_k) \right] \cos \emptyset_k \right\}$$

$$+ \int_0^{T_b} n(t) e_k(t) \cos(2\pi f_c t)\, dt \qquad (2.13)$$

Where $R_{k,i}, \hat{R}_{k,i}$ are the continuous-time partial crosscorrelation functions, which defined by:

$$R_{k,j}(\tau) = \int_0^{T_b} e_k\,(t-\tau) e_j(t) dt,$$

$$\hat{R}_{k,j}(\tau) = \int_\tau^{T_b} e_k\,(t-\tau) e_j(t) dt \qquad (2.14)$$

For $0 \le \tau \le T_b$. It is easy to see that for $0 \le lT_c \le \tau \le (l+1)T_c \le T_b$, these two crosscorrelation functions can be written as:

$$R_{k,j}(\tau) = e_{k,j}(l-N)T_c + \left[e_{k,j}(l+1-N) - e_{k,j}(l-N) \right](\tau - lT_c) \qquad (2.15)$$

$$\hat{R}_{k,j}(\tau) = e_{k,j}(l)T_c + \left[e_{k,j}(l+1) - e_{k,j}(l) \right](\tau - lT_c) \qquad (2.16)$$

Since, in this case the system is synchronized, then the time delays can be ignored "$\tau_k = 0$ and $\emptyset_k$", also because of the symmetry only $b_{j,0}$ can be considered to be +1. The desired signal component of z_k is then $\sqrt{P/2}\,T_b$ while the variance of the noise component of z_k is:

$$Var\{z_k\} = \left(\frac{P}{4T_b}\right)\sum_{\substack{k=1\\k\neq j}}^{K}\int_0^{T_b} R_{k,j}^2(\tau) + \hat{R}_{k,j}^2(\tau)d\tau + \frac{1}{4}\eta_0 T_b$$

$$= \left(\frac{P}{4T_b}\right)\sum_{\substack{k=1\\k\neq j}}^{K}\sum_{l=0}^{N-1}\int_{lT_c}^{(l+1)T_c} R_{k,j}^2(\tau) + \hat{R}_{k,j}^2(\tau)d\tau$$

$$+ \frac{1}{4}\eta_0 T_b \quad (2.17)$$

Thereafter, substitute for $R_{k,j}(\tau)$ and $\hat{R}_{k,j}(\tau)$ from (2.15) and (2.16) into (2.17). Then, the variance is given by:

$$Var\{z_k\} = \frac{PT^2}{12N^3}\left(\sum_{\substack{k=1\\k\neq j}}^{K} r_{k,j}\right) + \frac{1}{4}\eta_0 T_b \quad (2.18)$$

Where,

$$r_{k,j} = \sum_{l=0}^{N-1}\{e_{k,j}^2(l-N) + e_{k,j}(l-N)e_{k,j}(l-N+1) + e_{k,j}^2(l-N+1) + e_{k,j}^2(l)$$

$$+ e_{k,j}(l)e_{k,j}(l+1) + e_{k,j}^2(l+1)\} \quad (2.19)$$

In other words, equation (2.19) can be written in terms of the crosscorrelation parameters $\mu_{k,j}(n)$ which defined as:

$$\mu_{k,j}(n) = \sum_{l=1-N}^{N-1} e_{k,j}(l)e_{k,j}(l+n) \quad (2.20)$$

Consequently,

$$\mu_{k,j}(0) = \sum_{l=1-N}^{N-1} e_{k,j}^2(l) = \sum_{l=0}^{N-1} e_{k,j}^2(l-N) + e_{k,j}^2(l)$$

$$= \sum_{l=0}^{N-1} e_{k,j}^2(l-N+1) + e_{k,j}^2(l+1) \quad (2.21)$$

And,

$$\mu_{k,j}(1) = \sum_{l=1-N}^{N-1} e_{k,j}(l)e_{k,j}(l+1)$$

$$= \sum_{l=0}^{N-1} e_{k,j}(l-N)e_{k,j}(l-N+1) + e_{k,j}(l)e_{k,j}(l+1) \quad (2.21)$$

Therefore,

$$r_{k,j} = 2\mu_{k,j}(0) + \mu_{k,j}(1) \quad (2.22)$$

The signal-to-noise ratio is $\sqrt{1/_2\, PT_b}$ divided by the rms noise $\sqrt{Varz_k}$, which is:

$$SNR_k = \left\{(6N^3)^{-1}\sum_{\substack{k=1\\k\neq j}}^{K}\left[2\mu_{k,j}(0) + \mu_{k,j}(1)\right] + \frac{\eta_0}{2E}\right\}^{-1/_2} \quad (2.23)$$

Where $E = PT_b$ is the energy per data symbol and the average probability of error "P_e" can be approximated by:

$$P_e = 1 - \Phi(SNR_k) \quad (2.24)$$

Where, $\Phi(.)$ is the standard Gaussian cumulative distribution function.

Finally, for preliminary system design it is useful to be able to carry out a tradeoff between the parameters *K*, *N*, and E/η_0. Such a tradeoff can be based on the approximation:

$$(6N^3)^{-1}\sum_{\substack{k=1\\k\neq j}}^{K}\left[2\mu_{k,j}(0) + \mu_{k,j}(1)\right] \approx \frac{(K-1)}{3N} \quad (2.25)$$

Which yields,

$$SNR_k \approx \left\{\frac{K-1}{3N} + \frac{\eta_0}{2E}\right\}^{-1/_2} \quad (2.26)$$

The main use of (2.26) is to determine roughly what code sequence length *N*, bit energy *E*, and noise density $\eta_0/2$ are required to achieve a given SNR for a given number of users *K* [6], [7].

2.5 Spreading Codes

The objective of every multiple-access strategy is to allow multiple users to access the radio resource in a manner that maximize the use of the resource and minimize the interference among users. These users must be approximately orthogonal. With TDMA, the users are time orthogonal; with FDMA, the users are approximately frequency orthogonal. With CDMA, since all the users occupy the same transmission bandwidth at the same time, and each user has a different code, these codes are approximately orthogonal.

To achieve this orthogonality, a different symbol-shaping function (2.2) is assigned to each user. The approximate orthogonality of $e_k(t)$ and $e_j(t)$ for different time offsets τ can be represented as in (2.14), which says that the crosscorrelation of two different codes is approximately zero for all time offsets τ. An additional self-orthogonality that minimizes a number of practical channel and receiver effects is:

$$R_{kk}(\tau) \approx 0 \qquad\qquad for\ \tau > 0 \qquad\qquad (2.27)$$

Consequently, for all time offsets greater than zero, each spreading code is orthogonal to itself. The orthogonality conditions mean that different users can be separated at the receiver, even though they occupy the same frequency channel and the same period [3], [5].

2.5.1 Pseudo-Noise Sequences

A PN sequence is a coded binary sequence with a noiselike waveform that is usually generated by means of a feedback shift register. A feedback shift register consists of an ordinary shift register made up of *m* flip-flops “two state memory stages” and a logic circuit that are interconnected to form a multiloop feedback circuit. The most widely known binary PN code sequence, which is used in spread spectrum communication, is the maximum-length shift-register sequence, or m-sequence for short. This type is usually periodic with a period of $N = 2^m - 1$ bits and is generated by an m-stage shift register with linear feedback; the feedback function is obtained using modulo-2 addition if the outputs of the various flip-flops. This operation is illustrated in Figure 2.4 and table 2.1 [3].

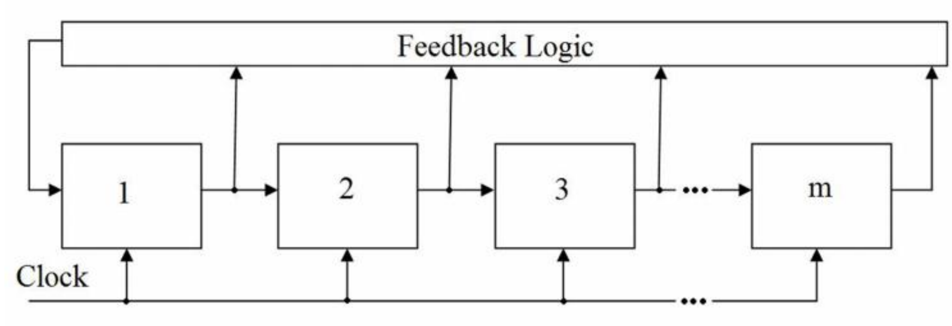


Figure 2.4 Feedback shift register with m-stages.

Table 2.1 Maximal-length sequences of lengths 2-8 [8].

Shift Register Length "m"	PN Sequence Length "N"	Feedback Taps
2	3	[2, 1]
3	7	[3, 1]
4	15	[4, 1]
5	31	[5, 2], [5, 4, 3, 2], [5, 4, 2, 1]
6	63	[6, 1], [6, 5, 2, 1], [6, 5, 3, 2]
7	127	[7, 1], [7, 3], [7, 3, 2, 1], [7, 4, 3, 2], [7, 6, 4, 2], [7, 6, 3, 1], [7, 6, 5, 2], [7, 6, 5, 4, 2, 1], [7, 5, 4, 3, 2, 1]
8	255	[8, 4, 3, 2], [8, 6, 5, 3], [8, 6, 5, 2], [8, 5, 3, 1], [8, 6, 5, 1], [8, 7, 6, 1], [8, 7, 6, 5, 2, 1], [8, 6, 4, 3, 2, 1]

As the shift register length m, or in other words the period N of the maximal-length sequence is increased, the sequence becomes increasingly similar to the random binary sequence.

Maximal-length sequences have many of the properties possessed by a truly random binary sequence. A random binary sequence is a sequence in which the presence of binary symbol 1 or -1 is equally probable. Some properties of maximal-length sequences are:

- In each period of a maximal-length sequence, the number of 1s is always one more than the number of 0s. This property is called the *balance property*.
- Among the runs of 1s and 0s in each period of a maximal-length sequence, one-half the runs of each kind are of length one, one-fourth are of length two, one-

eighth are of length three, and so on as long as these fractions represent meaningful number of runs. This property is called the *run property*. For a maximal-length sequence generated by a linear feedback shift register of length m, the total number of runs is $(N+1)/2$.

- The autocorrelation function of a maximal-length sequence is periodic and binary valued. This property is called the correlation property. This property is so important and is usually defined in terms of the bipolar sequences as:

$$R_c(m) = \sum_{n=1}^{N} e_n e_{n+m} \qquad ; 0 \le m \le N-1 \tag{2.28}$$

Ideally, the maximal-length sequence should have an autocorrelation function that has correlation properties similar to white noise. That is:

$$R_c(m) = \begin{cases} L, & m = 0 \\ -1, & 1 \le m \le N-1 \end{cases} \tag{2.29}$$

Therefore, the maximal-length sequences are very close to ideal PN sequences when viewed in terms of their autocorrelation function. Moreover, in some applications such as CDMA, the crosscorrelation properties of PN sequences are as important as the autocorrelation [4], [8].

2.5.2 Gold Sequences

Methods for generating PN sequences with better periodic crosscorrelation properties than the maximal-length “m-sequences” have been developed by Gold and by Kasami. Gold sequences can be constructed by taking a pair of specially selected m-sequences, called preferred m-sequence, and forming the modulo-2 sum of the two sequences, for each of N cyclicly shifted versions of one sequence relative to the other sequences, as shown in Figure 2.5.

The crosscorrelation function of any pair of Gold sequences was proven by Gold to be three valued with possible values $\{-1, -t(m),\ t(m)-2\}$, where $t(m)$ is defined [2]:

$$t(m) = \begin{cases} 2^{\frac{(m+1)}{2}} + 1 & ; m: odd \\ 2^{\frac{(m+2)}{2}} + 1 & ; m: even \end{cases} \tag{2.30}$$

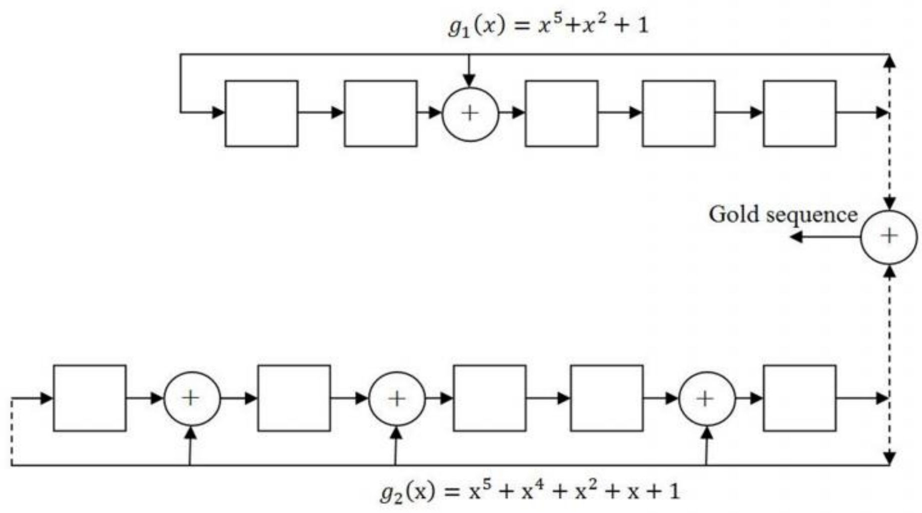


Figure 2.5 Generation of Gold sequence with length 31.

2.5.3 Kasami Sequences

In 1966 Kasami described a method for constructing PN sequences by decimating an m-sequence. Generation of a small set of Kasami sequences is similar to the generation of Gold sequence with $2^{m/2}$ binary sequences of period $2^{m} - 1$, where m is even. To generate a Kasami sequence every $2^{m/2} + 1$ bit of an m-sequence is selected.

Kasami sequences sets are one of the important types of binary sequences because of their very low crosscorrelation. The crosscorrelation function of these sequences take on values form the set $\{-1, -t(m),\ t(m) - 2\}$.

2.6 Noise

The term noise is used customarily to designate not only unwanted but also uncontrolled signals that tend to disturb the transmission and processing of signals in communication systems. In practice, there are many potential sources of noise in a communication system. The sources of noise may be external to the system such as atmospheric noise, galactic noise, man-made noise, or internal to the system. The second category includes an important type of noise that arises from spontaneous fluctuations of current or voltage in electrical circuits. This type of noise represents a basic limitation on the transmission or detection of signals in communication systems involving the use of electronic devices, and the two most of this type are shot noise and thermal noise. Since our study is far from these types, then just the external noise will be under concentration [8].

2.6.1 White Noise

The noise analysis of communication systems is customarily based on an idealized form of noise called *white noise*, the power spectral density of which is independent of the operating frequency. The adjective *white* is used in the sense that white light contains equal amounts of all frequencies within the visible band of electromagnetic radiation. The power spectral density of white noise can be expressed as:

$$S_w(f) = \frac{\eta_0}{2} \tag{2.30}$$

White noise has infinite average power and it is not physically realizable. Nevertheless, white noise has simple mathematical properties which make it useful in statistical system analysis [11].

2.7 Jamming

The jamming processer main goal is to disturb the communication, and it is enough to deny an effective transfer of information by approximately jam 30% of the transmission. The jammers encountered in practice include the following types:

- The barrage noise jammer, which consists of band-limited white Gaussian noise of high average power. The barrage noise jammer is a brute-force jammer that does not exploit any knowledge of the antijam communication system except for its spread bandwidth.
- The partial-band noise jammer, which consists of noise whose total power, is evenly spread over some frequency band that is a subset of the total spread bandwidth. Owing to the smaller bandwidth, the partial-band noise jammer is easier to generate than the barrage noise jammer.
- The pulsed noise jammer, which involves transmitting wideband noise of power:

$$J_{peak} = \frac{J}{p} \tag{2.31}$$

For a fraction p of the time, and nothing for the remaining fraction 1-p of the time. The average noise power equals J.

- The single-tone jammer, which consists of a sinusoidal wave whose frequency lies inside the spread bandwidth.
- The multitone jammer, which is the tone equivalent of the partial-band noise jammer.

In addition to these types many other kinds of jamming waveform occur in practice, there is no single jamming waveform that is worst for all spread spectrum systems and there is no single spread spectrum system that is best against all possible jamming waveforms [8], [9].

CHAPTER III

MULTIUSER DETECTION

The detection techniques in the spread spectrum systems are studied in this chapter. Initially, the conventional detectors are explained briefly, then the multiuser detection is presented.

3.1 Introduction

In multi-access communication systems, multiple access interference "MAI" is the major factor limiting the performance and hence the capacity of the system. MAI occurs in multi-access communication systems where simultaneously occurring digital streams of information interfere with each other. Therefore, analyses the effect of MAI on the system performance as well as ways to suppress MAI have been the major focus of research.

Multiuser Detection "MUD" is the intelligent demodulation of transmitted bits in the presence of MAI. There are two different approaches to the problem; the first approach is based on the concept of *single user detection*. In this approach, just one of the users is identified in the system as the desired user, and just treats the MAI as AWGN. However, unlike AWGN, MAI has a nice correlative structure that is quantified by the cross-correlation matrix of the signature sequences. The second approach is called *multiuser detection*, in which all signals from all users are detected jointly and simultaneously by the receiver [1].

3.2 DS-CDMA System Model

An idealized setting suffices to illustrate the basic concepts, hence a K-users, discrete time, real baseband, synchronous system is considered, so the received signal is given by [1]:

$$y(t) = \sum_{k=1}^{K} A_k\, b_k e_k(t) + n(t) \tag{3.1}$$

Where;

A_k is the received amplitude for the kth user.

$b_k \in \{-1,1\}$ is the transmitted bit for the kth user.

e_k is the signature waveform of the kth user.

$n(t)$ is an AWGN.

Since synchronous CDMA is considered, it is assumed that the receiver has some means of achieving perfect chip synchronization. Then the cross-correlation of the signature sequences is defined as:

$$\rho_{jk} = \int_{0}^{T_b} e_j\,(t) e_k(t)\, dt \qquad ; j \neq k \tag{3.2}$$

3.3 Conventional Single User Detector

In conventional single-user digital communication systems, the matched filter is used to generate sufficient statistics for signal detection. This is the simplest way to demodulate the received signal, a bank of matched filters, one filter to each user's waveform, is applied to the received signal as shown in Figure 3.1. Thus, it demodulates each user independently from the others. The statistics of the MAI term are different from the noise term and hence it should be treated differently. Since the conventional matched filter detector is designed for the case of orthogonal spreading waveforms, it does not take MAI into account. On applying the same detector to the non-orthogonal case, MAI is treated as a noise term. The output of the kth matched filter is given by [1], [10], [11]:

$$y_k = \int_0^{T_b} y(t)\, e_k(t)dt$$

$$= A_k b_k + \sum_{j \neq k} A_j\, b_j \rho_{jk} + \sigma \int_0^{T_b} n(t) e_k(t)dt$$

$$= A_k b_k + MAI_k + n_k \tag{3.3}$$

In other words, correlation with the kth user itself gives rise to the recovered data term, correlation with all the other users gives rise to MAI, and correlation with the noise yields the term n_k [11]. Where n_k is a Gaussian random variable with zero mean and σ^2 variance. Since the codes are generally designed to have very low crosscorrelation relative to autocorrelation "$\rho_{jk} \ll 1$", the interfering effect on the user k of the other users is greatly reduced. If the signature waveform of the kth user is orthogonal to the other signature waveforms, then $\rho_{jk} = 0,\ j \neq k$ and the matched filter output (3.3) reduces to that obtained in the single user problem [1]:

$$y_k = A_k b_k + n_k$$

The optimum detector should take into account the information available in all the received signals to estimate the bit of a particular user. This is known as Multi-user Detection and was proposed by Sergio Verdu in the early 1980's. Any multi-user detector will utilize the information available in the MAI term to demodulate the signal and will not treat it like a noise term.

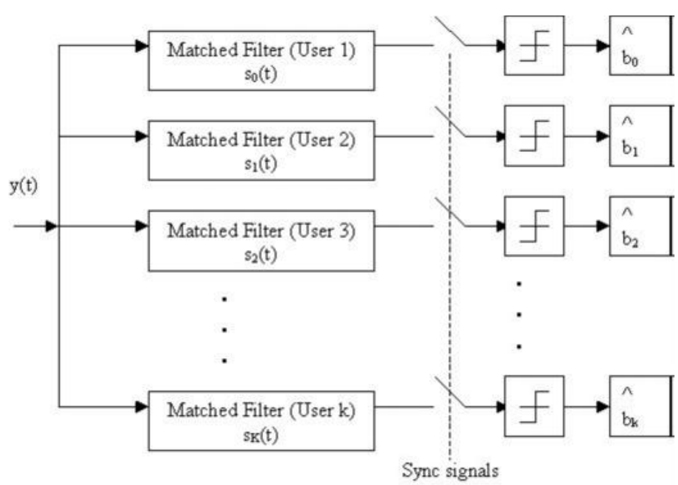


Figure 3.1 Bank of single user matched filter [1].

3.4 Multiuser Detection

Multiuser detection refers to reception techniques that exploit the structure of the MAI in receiver design, rather than ignoring it as in conventional reception. In multiuser detection code and timing information of multiple users are jointly used to better detect each individual user. The important assumption is that the codes of the multiple users are known to the receiver a priori [12], [13].

The multiuser detectors can be classified in one of two categories: linear multiuser detectors, and subtractive interference cancellation detectors. In linear multiuser detection, a linear mapping is applied to the soft outputs of the conventional detector to produce a new set of outputs, which hopefully provide better performance. In subtractive interference cancellation detection, estimates of the interference are generated and subtracted out. Only the linear detectors are under focus in our work.

3.4.1 Optimum Multiuser Detection

The conventional single user matched filter receiver requires no knowledge beyond the code and timing of the users it wants to demodulate. In the derivation of an optimum receiver, in this case the receiver not only knows the code and timing of every active user, but it also knows the received amplitudes of all users and the noise level.

The optimum detector is defined as the receiver that selects the most probable sequence of bits given the received signal $y(t)$. In synchronous transmission, each user produces exactly one symbol which interferes with the desired symbol. In AWGN, it is sufficient to consider the signal received in one signal interval "$0 \leq t \leq T_b$", according to equation (3.1), then determine the optimum receiver.

The optimum maximum-likelihood receiver computes the log-likelihood function as:

$$\Lambda(b) = \int_0^{T_b} \left[y(t) - \sum_{k=1}^{K} A_k b_k e_k(t) \right]^2 dt \tag{3.4}$$

In the next step, select the information sequence "b_k" that minimize $\Lambda(b)$. By expanding the integral in (3.4), then $\Lambda(b)$ will be:

$$\Lambda(b) = \int_0^{T_b} y^2(t)dt - 2\sum_{k=1}^{K} A_k b_k \int_0^{T_b} y(t)e_k(t)dt$$
$$+ \sum_{j=1}^{K}\sum_{k=1}^{K} A_j A_k b_j b_k \int_0^{T_b} e_k(t)e_j(t)dt \qquad (3.5)$$

By analyzing (3.5), it is obvious that the first term "$\int_0^{T_b} y^2(t)dt$" is common to all possible sequences "b_k" and is of no relevance in determining the transmitted sequence. Hence, it may be neglected. From the second term:

$$y_k = \int_0^{T_b} y(t)e_k(t)dt \qquad ; 1 \le k \le K \qquad (3.6)$$

According to the previous explanation (3.6) represents the crosscorrelation of the received signal with each of the K signature sequences. Instead of the cross correlators, matched filters may be employed. Finally, notably that the integral in the last term is:

$$\rho_{jk} = \int_0^{T_b} e_j(t)e_k(t)\, dt \qquad (3.7)$$

As a result of the analysis equation (3.5) can be expressed in a form of correlation metrics as:

$$C(y_k, b_k) = 2\sum_{k=1}^{K} A_k b_k y_k + \sum_{j=1}^{K}\sum_{k=1}^{K} A_j A_k b_j b_k \rho_{jk} \qquad (3.8)$$

By expressing equation (3.8) in vector inner product form it became:

$$C(y_k, b_k) = 2b_k' y_k - b_k' R_s b_k \qquad (3.9)$$

Where the correlation matrix is R_s, with elements ρ_{jk}.

There are 2^K possible choices of the bits in the information sequence of the K users. The optimum detector computes the correlation metrics for each sequence and selects the sequence that yields the largest correlation metric. Consequently, the optimum detector has a complexity that grows exponentially with the number of users.

To sum up, the optimum detector for synchronous transmission consists of K banks "correlators or matched filters" followed by a detector that computes 2^K

correlation metrics as given by (3.9) corresponding to the 2^K possible transmitted information sequences. Then, the detector selects the sequence corresponding to the largest correlation metric [2].

3.4.2 Suboptimum Multiuser Detection

The large gaps between the conventional single user matched filter and the optimum detector in both performance and complexity yields to the detectors that exhibit good performance complexity tradeoffs. The decorrelating detector is not only a simple and natural strategy, but also optimal according to three criteria “least squares, near-far resistance and maximum likelihood” when the received amplitudes are unknown.

Multiuser detectors commonly have a front-end whose objective is to obtain a discrete-time process from the received continuous-time waveform $y(t)$. Continuous to discrete time conversion can be realized by conventional sampling, or by correlation of $y(t)$ with deterministic signals. One way of converting the received signal into a discrete-time process is to pass it through a bank of matched filters, each one to the signature sequence of a different user. In the synchronous case, the output of the bank of matched filter is [1]:

$$y_k = \int_0^{T_b} y(t) e_k(t) dt$$

The output of the k^th matched filter can be express as:

$$y_k = A_k b_k + \sum_{j \neq k} A_j b_j \rho_{jk} + n_k \tag{3.10}$$

It is convenient to express (3.10) in vector form:

$$y = RAb + n \tag{3.11}$$

Where, R is the normalized crosscorrelation matrix.

As known that the conventional receiver may make errors even in the absence of noise “$\sigma = 0$”, that is, it may happen that:

$$\hat{b}_k = sgn((RAb)_k) \neq b_k$$

In contrast, the optimum receiver demodulates the data error-free in the absence of noise. Of course, this is a desirable feature for any multiuser detector, even though the situation of absence of background noise is completely hypothetical. A simple receiver suffices to recover the transmitted bits error-free in the absence of noise, without requiring knowledge of the received amplitudes. Let us assume that the normalized crosscorrelation matrix R is invertible. By premultiplying the vector of matched filter outputs by R^{-1}, then:

$$R^{-1}y = R^{-1}RAb = Ab \tag{3.12}$$

First, let us consider the case of the absence of noise "$\sigma = 0$". Taking the sign of each of the components in (3.12) to recover the transmitted data:

$$\hat{b}_k = sgn((R^{-1}y)_k) = sgn((Ab)_k) = b_k \tag{3.13}$$

If the signature sequences are linearly independent the detector in (3.13) achieves perfect demodulation for every active user.

Second, in the existing of noise, where the processing of the matched filter bank outputs (3.11) with R^{-1} results in:

$$R^{-1}y = Ab + R^{-1}n \tag{3.12}$$

Notice that the kth component of (3.12) is still free from interference caused by any of the other users, that is, it is independent of all $b_j, j \neq k$. The only source of interference is the background noise. This is why the detector that performs (3.12) is called the decorrelating detector, and it is shown in Figure 3.2.

As can be seen, in the absence of background noise the decorrelating detector achieves perfect demodulation unlike the matched filter bank. Moreover, the decorrelating detector does not require knowledge of the received, and also it can readily be decentralized, in the sense that the demodulation of each user can be implemented completely independent [1].

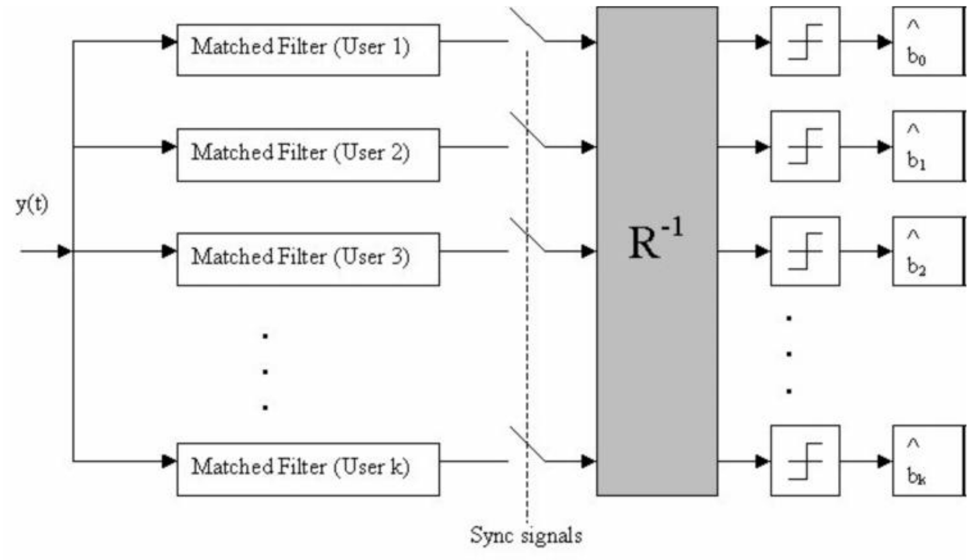


Figure 3.2 Decorrelating detector for the synchronous channel [1].

CHAPTER IV

JAMMER DESIGN

At first, the theoretical bases are explained in this chapter, and then the MATLAB simulation bases are provided. At the end, the results are illustrated and discussed under different aspects.

4.1 Theoretical Bases

The MATLAB program is based on the direct sequence spread spectrum principles as illustrated in chapter two. First, the message signal is based on equation (2.1), and the spreading sequences used in this operation are based on equation (2.2).

As can be seen from equation (2.4) the transmitted signal is a binary PSK signal, since the carrier has no effect on the system performance, the carrier is ignored from now on. As a result of this note, the DSSS is obtained by the time multiplication as:

$$s(t) = m(t)e(t) \tag{4.1}$$

Assuming that the system have K users, $K = 1, \ldots, k$, then the signal transmitted by the k^{th} user is:

$$s_k(t) = \sum_{n=-\infty}^{\infty} b_{n_k} g(t - nT_b) \sum_{i=0}^{N-1} e_{i_k} \, p(t - iT_c)$$

In this work it is assumed that the system is disturb by AWGN, which was explained is chapter two, according to this assumption the transmitted signal will be:

$$s_k(t) = \sum_{n=-\infty}^{\infty} b_{n_k} g(t - nT_b) \sum_{i=0}^{N-1} e_{i_k} \, p(t - iT_c) + n_k(t) \tag{4.2}$$

The jammer is designed by disturbing the system using the same spreading codes, which used in the original system. Jamming the transmitted signals occur through the following steps:

- Generating random messages:

$$m_j(t) = \sum_{m=-\infty}^{\infty} j_m q(t - mT) \tag{4.3}$$

Where:

$j_m \in \{1, -1\}$

T: is the duration of one symbol.

$q(t)$: Shaping waveform, $q(t) = \begin{cases} 1 & 0 < t < T \\ 0 & otherwise \end{cases}$

- Generating the same spreading sequences “codes”, which is based on equation (2.2):

$$e(t) = \sum_{i=0}^{N-1} e_i \, p(t - iT_c)$$

- Spreading $m_j(t)$:

$$jam(t) = m_j(t)e(t) = \sum_{m=-\infty}^{\infty} j_m q(t - mT) \sum_{i=0}^{N-1} e_i \, p(t - iT_c) \tag{4.4}$$

As a result the received signal after jamming is:

$$r(t) = \sum_{n=-\infty}^{\infty} b_{n_k} g(t - nT_b) \sum_{i=0}^{N-1} e_{i_k}\, p(t - iT_c) + n_k(t)$$

$$+ \sum_{m=-\infty}^{\infty} j_{m_k} q(t - mT) \sum_{i=0}^{N-1} e_{i_k}\, p \tag{4.5}$$

Suboptimum detector “decorrelating detector” is used to retrieve the transmitted signals. Despreading the received signal is the first step as:

- Assuming synchronous operations, and for k^{th} user the symbol a_{0k} is the transmitted symbol, then:

$$r(t) = \sum_{k=1}^{K} b_{0_k} e_k(t) + n(t) + \sum_{k=1}^{K} j_{0_k}\; e_k(t)$$

$$r_k = \int_0^{T_b} r(t) e_k(t)\, dt$$

$$= a_{0_k} \int_0^{T_b} {e_k}^2(t) dt + \sum_{\substack{j=1 \\ j \neq k}}^{K} \int_0^{T_b} e_j(t) e_k(t) dt + \int_0^{T_b} n(t) e_k(t) dt$$

$$+ j_{0_k} \int_0^{T_b} {e_k}^2(t) dt + \sum_{\substack{j=1 \\ j \neq k}}^{K} \int_0^{T_b} e_j(t) e_k(t) dt$$

$$= a_{0_k} + \frac{N_0}{2} T_b + j_{0_k}$$

4.2 MATLAB Simulation

Our CDMA system, which is the aimed system to be jammed, is designed as in Figure 4.1 and illustrated in the following:

- Creating each of the massage signals $m(t)$, the noise signal $n(t)$, and the jamming signals $m_j(t)$, this issue is to be done by using *randian, wgn, and randn* functions in *MATLAB* respectively.
- The *mdl* file utility of *MATLAB* is used to generate the *GOLD* spreading sequences, those sequences used to spread the message signals not only at the transmitter "$m(t)$", but also at the jammer "$m_j(t)$". Finally, they are used to retrieve the transmítted signals at the receiver.
- The receiver in our system is based on the decorrelating detector, the received bits are resolved by equation (3.13).
- The experimental probability of error "P_e or BER" for this jammed system is to be calculated by dividing the accumulated number of errors by the total number of received bits, which is expressed as:

$$P_{exp} = \sum_{i=1}^{n} \frac{\delta_i}{n} \qquad 4.6$$

- On the other hand, the theoretical probability of error is to be calculated according to equation (2.24).

4.3 Results and Discussion

The results for this work are examined under different aspects, by examining the theoretical performance of the system without jamming, and by testing the effect of different parameters as in the following sections.

4.3.1 System Performance without Jamming

In the construction of our simulation model for the DS-CDMA system, the *mdl* file utility of MATLAB is used to generate the GOLD spreading sequences. To the same file, a scope and signal from workspace blocks were added to visualize waveforms of message signal, gold sequences, transmitted signal as the sum of signals belonging to all active users. A glimpse from these waveforms is shown in Figure 4.1.

The second plot shows the theoretical performance of the system without jamming, the performance presented by calculating the probability of error of the system as shown in Figure 4.2.

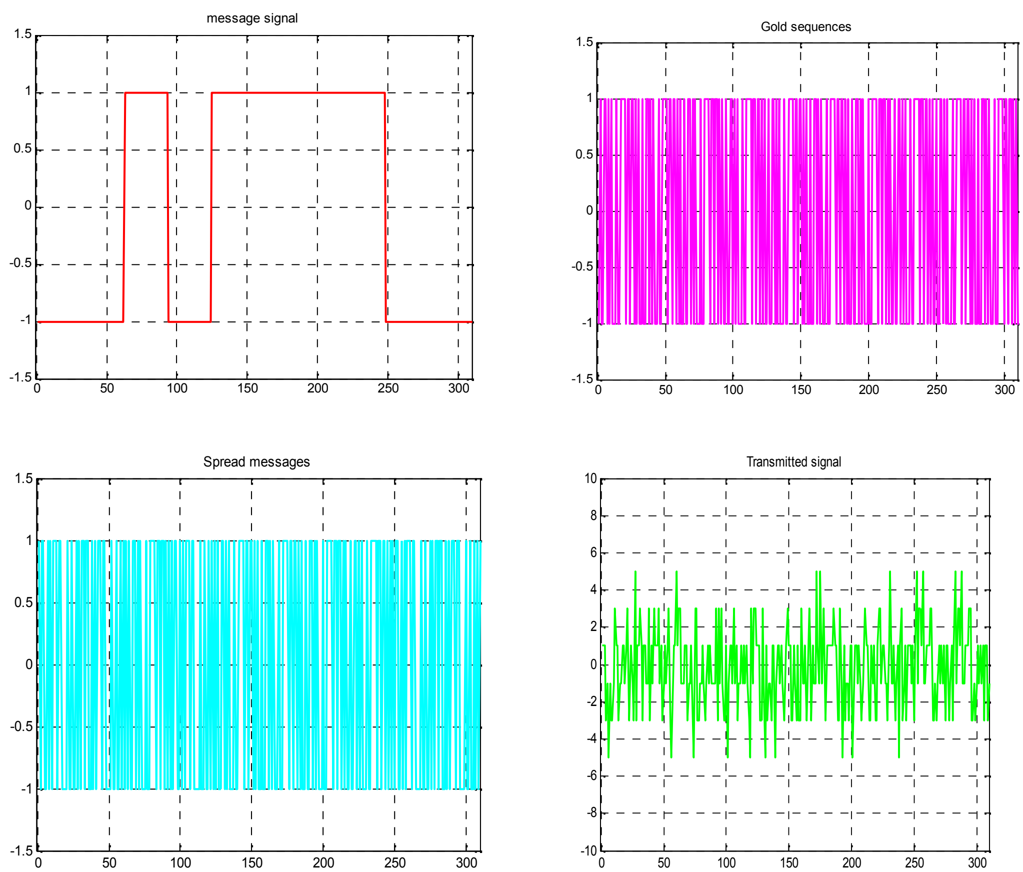


Figure 4.1 Typical DS-CDMA waveforms at transmitter.

4.3.2 Performance of the Jammed System

Continuing with the Gold Code as the spreading sequences the system is jammed in the following, besides that the effect of the number of users “K”, the code length “N” are examined, and the messages length “n” is not, since the theoretical performance of the system is independent from “n”.

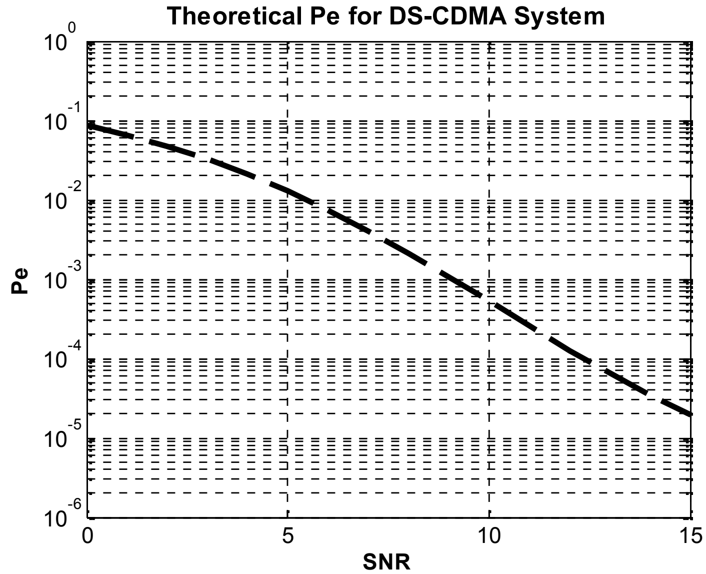


Figure 4.2 Theoretical probability of error for DS-CDMA system.

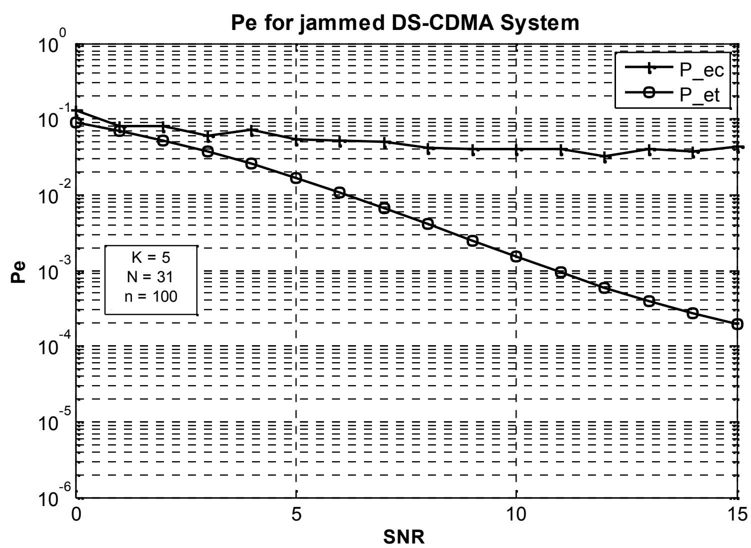
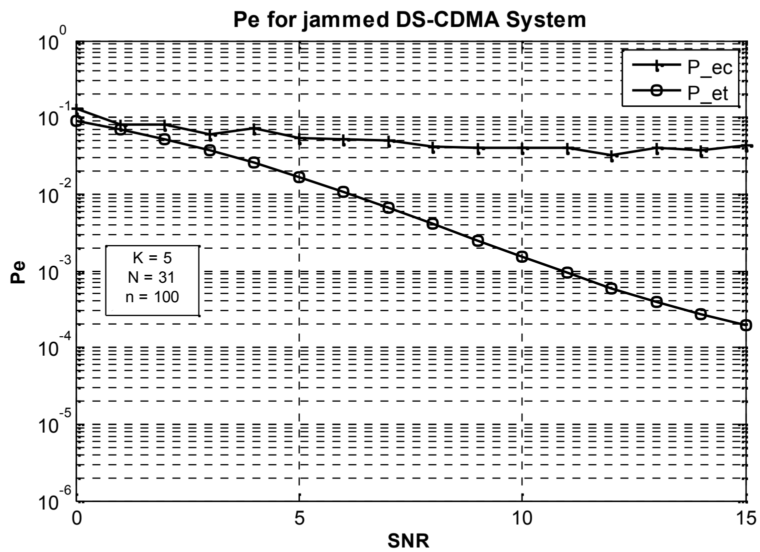


Figure 4.3 The P_e for the jammed system.

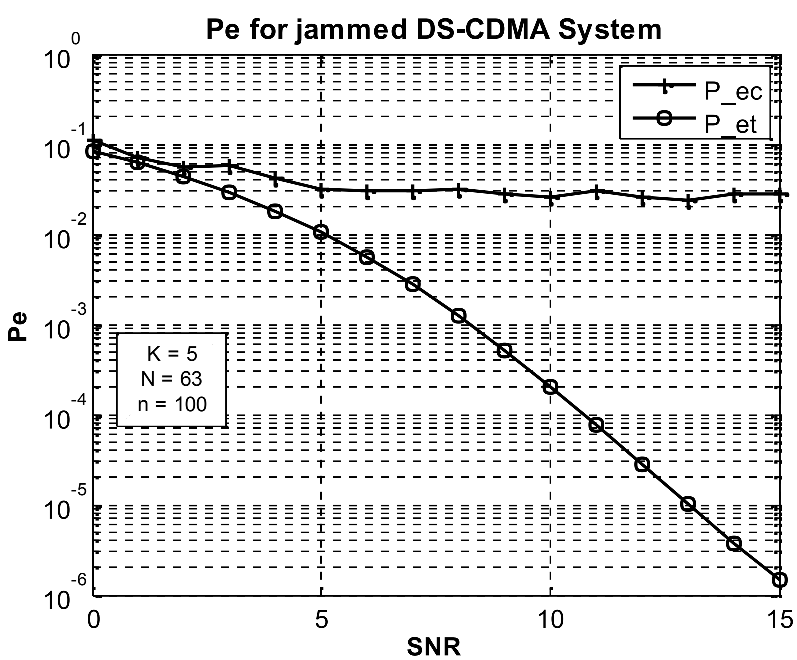
According to Figure 4.3 it is obvious that at low SNR a close agreement has been achieved between the theoretical and the experimental curves, afterwards the curves start to diverge.

- The code length impacts:

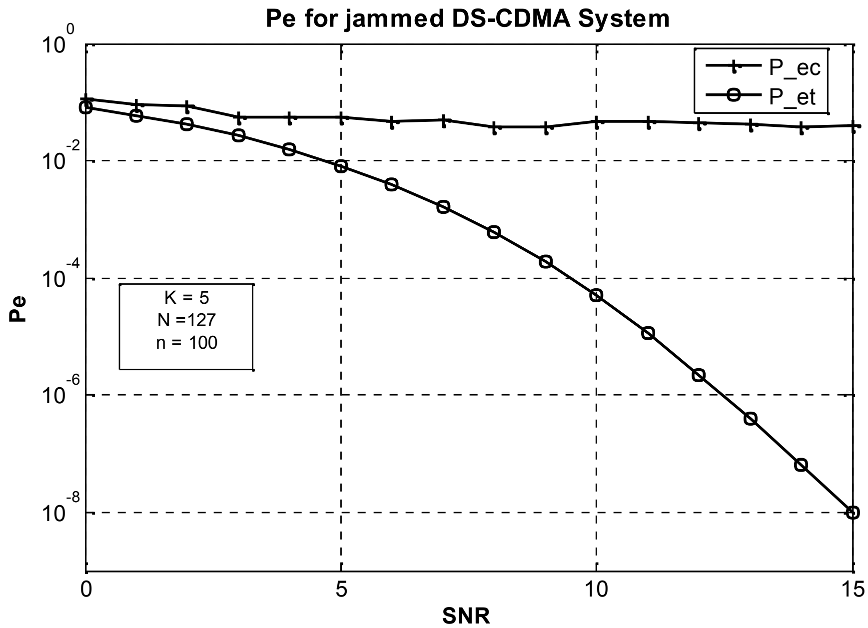
This impact as shown in Figure 4.4 is studied under fixed circumstances "n = 100, K = 5".



(a)



(b)



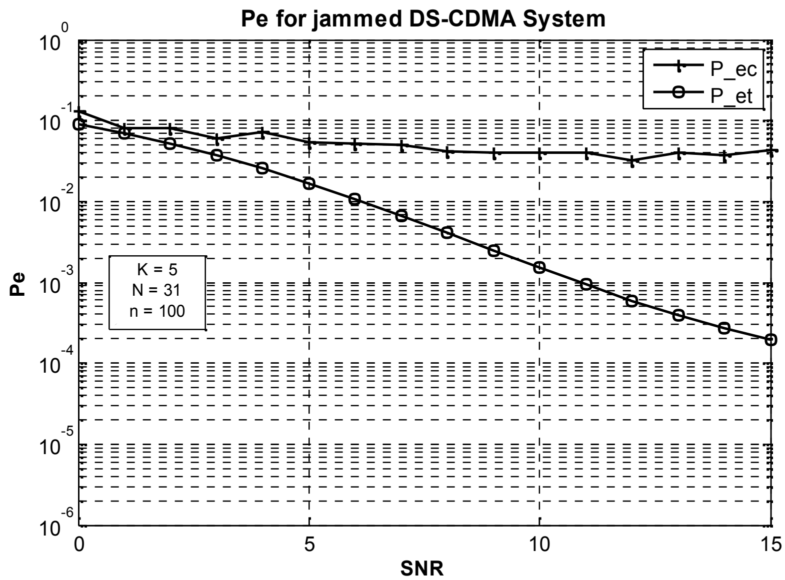
(c)

Figure 4.4 The impact of the code length, (a) N = 31, (b) N = 63, (c) N = 127.

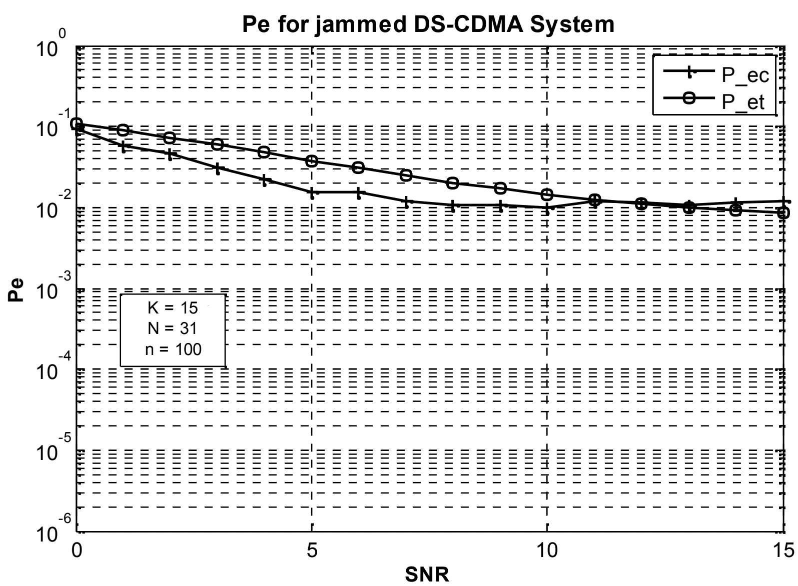
As can be seen from Figure 4.4 there is a little converge between the two curves is achieved at low SNR, while at high SNR the divergence between the curves increase by increasing the code length.

- Number of users impacts:

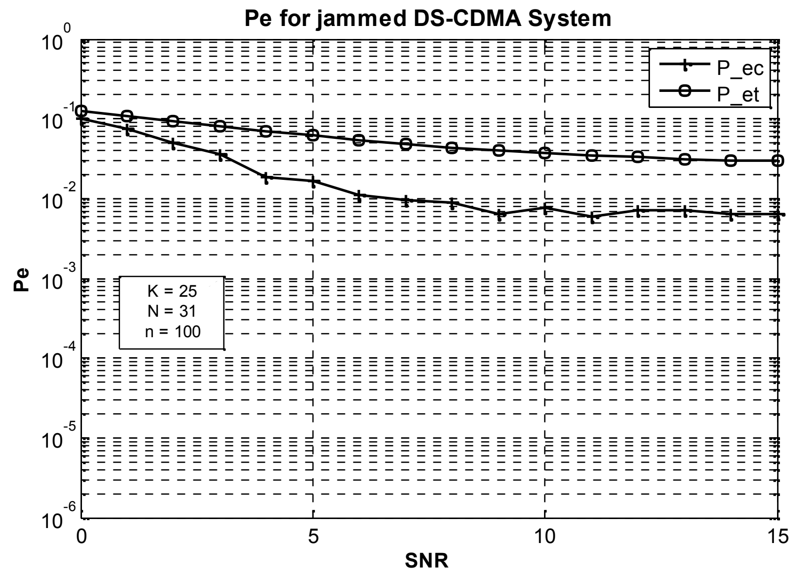
This impact as shown in Figure 4.5 is studied under fixed circumstances “n = 100, N = 31”.


Pe for jammed DS-CDMA System
P_ec
P_et
K = 5
N = 31
n = 100
Pe
SNR

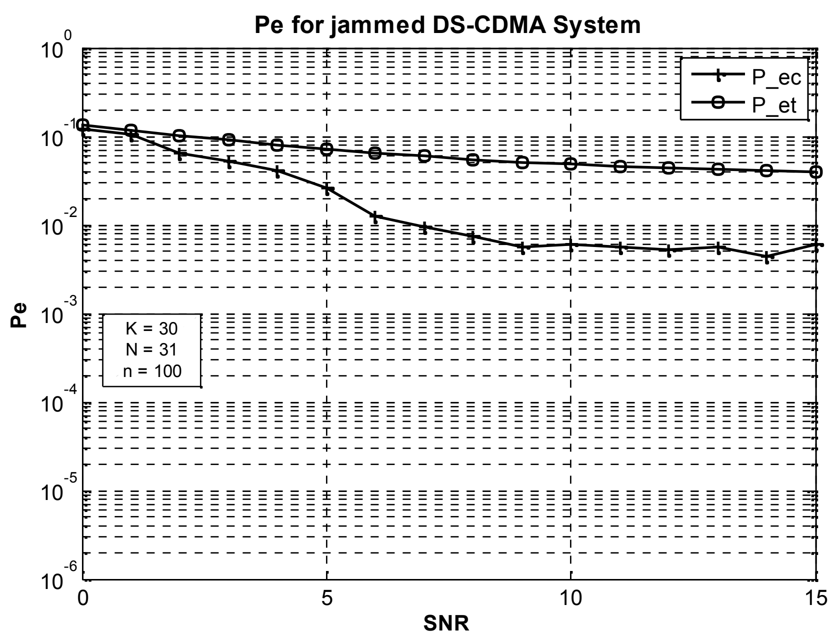
(a)


Pe for jammed DS-CDMA System
P_ec
P_et
K = 15
N = 31
n = 100
Pe
SNR

(b)



(c)



(d)

Figure 4.5 The impact of the number of users, (a) K = 5, (b) K = 15, (c) K = 25, (d) K = 30.

Increasing the number of users decreases the divergence between the experimental and the theoretical performance.

4.3.3 Performance of the Jammed System Using Different Codes

In the first place, the theoretical performance of the system is examined as plotted in Figure 4.6, for each of Gold sequences, PN sequences and finally Kasami sequences, under the same setting "n = 100, K = 5, N = 31".

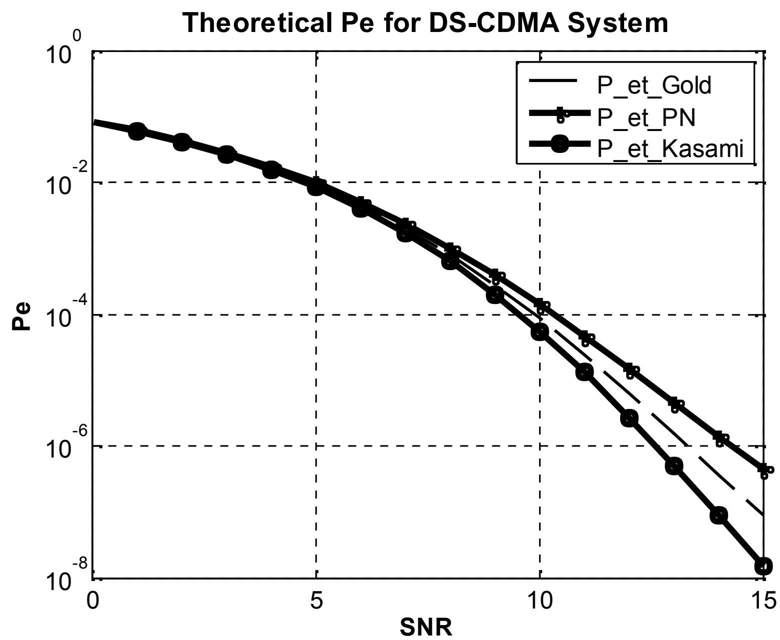
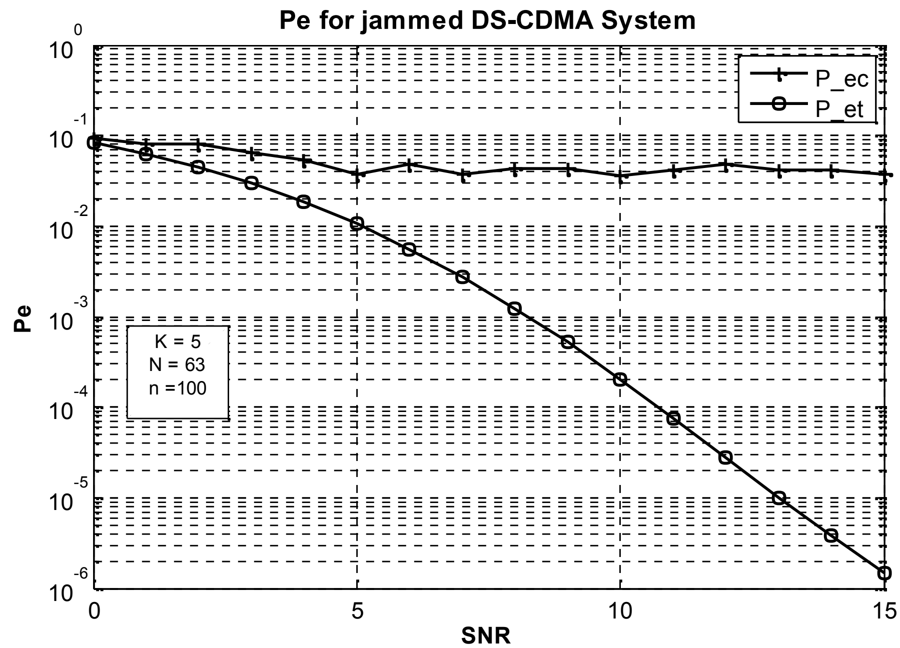


Figure 4.6 Theoretical P_e for the system using different codes.

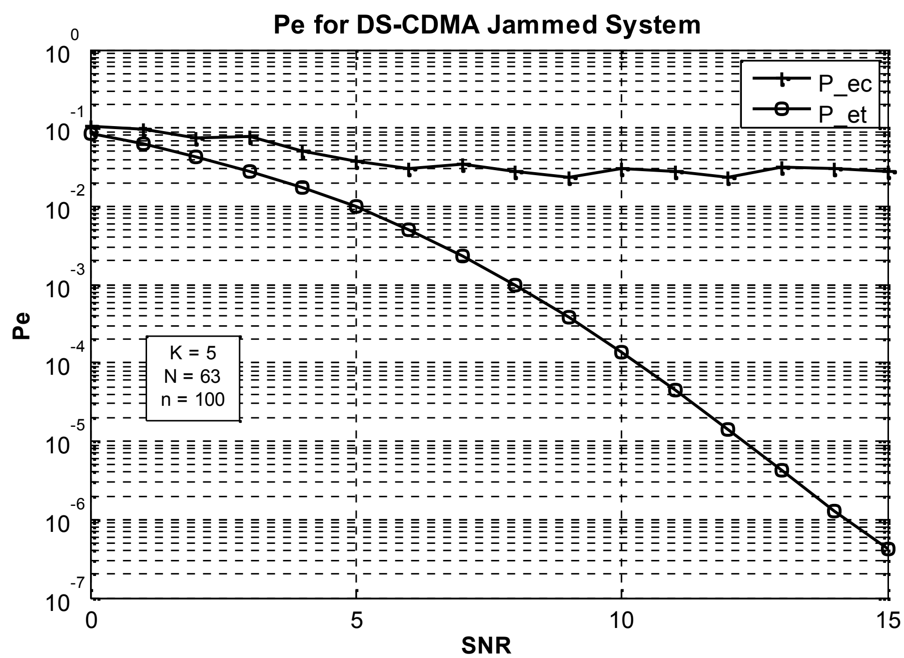
As can be seen from the plots the three sets of sequences have the same performance at low SNR. On the other hand, at higher SNR Kasami sequences have the best P_e, then the Gold sequences, and for the PN sequences have lowest P_e. This is of course because the crosscorrelation properties of these sequences.

Furthermore, another important comparison has to be discussed, which the compared experimental and theoretical P_e, for the same previous setting among the codes as shown in Figure 4.7.



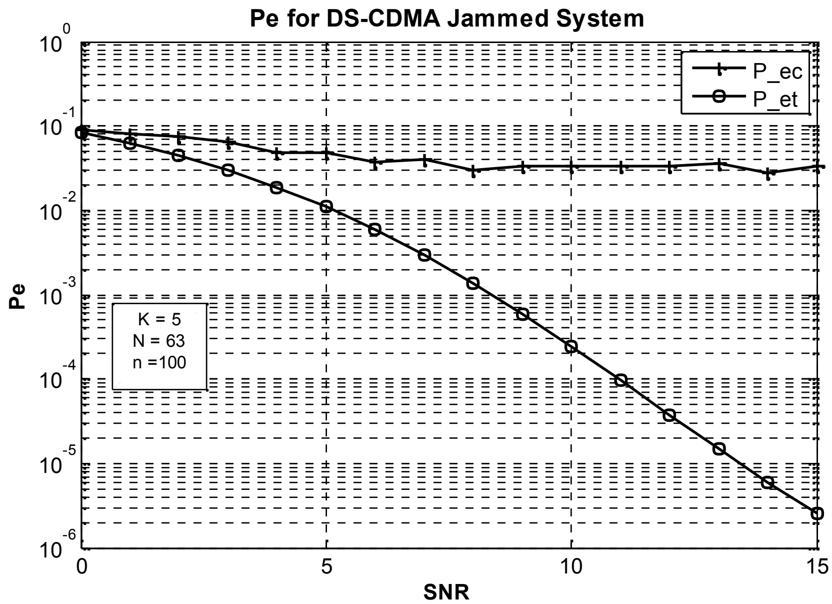
Pe for jammed DS-CDMA System
P_ec
P_et
K = 5
N = 63
n =100
Pe
SNR

(a)



Pe for DS-CDMA Jammed System
P_ec
P_et
K = 5
N = 63
n = 100
Pe
SNR

(b)



(c)

Figure 4.7 P_e for the jammed DS-CDMA system, (a) Gold sequences, (b) Kasami sequences, (c) PN sequences.

It can be easily understood that the three codes has similar performance at low SNR, in which the experimental and the theoretical curves are close. On the other hand, at higher SNR the divergence of the curves is different, since its smallest value by using PN sequences, then it increases for the Gold sequences to reach the highest value for the Kasami sequences.

CHAPTER V

CONCLUSION

The main purpose of the work presented in this thesis was to design a jammer of DS-CDMA systems by using a new technique which different from the jammer types used in practice. This thesis implemented this work by considering the suboptimum MUD, also by testing different codes. In this final chapter the main conclusions from the novel findings and future research directions on this work are presented.

5.1 Summary

Since for some applications such as the security issues disturbing the system is a necessary need this thesis has studied the design of a jammer for DS-CDMA systems. To achieve this aim each of the direct sequence spread spectrum systems and the spreading codes and the jamming principle were covered in chapter two. Additionally, the principles of the multiuser detection technique and its approaches were explained in chapter three.

Finally, in chapter four the system was designed and its performance was examined, and according to this design the plots which presented the principle of DS-CDMA system and its performance "P_e" are illustrated. As a next step, the jammed performance is illustrated, and it can be concluded that the theoretical and the experimental performance are converge just at lower SNR. Then, by studying the impact of both the length of the code and the number of users the results are reverse, since by increasing the code length the divergence between the curves increases, and the opposite is true for the other impact. The effect of the message length has not been studied, since the theoretical performance of the system does not depend on it.

5.2 Future Work

In the communication system the protection of the system from the external disturbers such as the jammers is the most important issue to be designed. On the other hand, in some cases disturbing the system is extremely important, so the jammer design has to be studied. In this thesis this study is different, by using the codes which operate the jammed system. According to this just the main point in the design has been considered "the performance of the jammer". In what follows, certain points are presented that can be considered of great interest for further research work in this topic.

- The system is assumed to be synchronous so the asynchronous system is worth to be assessed.
- For simplicity, the spreading codes were just applied without a technique to choose them and since in practice the codes have to be chosen to give the most appropriate performance, then these techniques would be of a great importance as a future work.
- At the receiver just the suboptimum detector was applied in the work, so applying different types of MUDs and studying whether they would perform better would be an important work.

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APPENDIX A
CURRICULUM VITAE

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