ADDIS ABABA UNIVERSITY

SCHOOL OF GRADUATE STUDIES

INSTITUTE OF TECHNOLOGY

ELECTRICAL AND COMPUTER ENGINEERING DEPARTMENT

Performance Evaluation of Least Mean Square Error (LMS) and Blind Adaptive Algorithm in Suppressing Multiple Access Interference in CDMA Receiver

By

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Declaration

I, the undersigned, declare that this thesis is my original work, has not been presented for a degree in this or any other university, and all sources of materials used for the thesis have been fully acknowledged.

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Dedicated to
Aster Kebede

Abstract

Direct sequence code division multiple access (DS-CDMA) is a high capacity air interface as compared to time division multiple access (TDMA) and frequency division multiple access (FDMA). This system is more resistant to jamming and unintended listener. In this access technique more than one user access the same frequency band width at the same time. The main limitation of this system is that, the signal quality of the intended user degrades as the number of users increases that result in multiple access interferences (MAI).

So, in this thesis adaptive algorithms are used in CDMA receiver for suppressing multiple access interference (MAI). Two families of adaptive algorithms namely, least mean square error (LMS) and constant modulus blind adaptive algorithms are used. The schematic representation of each of the two algorithm receiver is developed, and on which the simulation analysis was based. The LMS adaptive algorithm requires training sequences for its adaptation where as the later does not require any training sequences, instead needs appropriate initialization of tap weights. The performance of these adaptive algorithms in suppressing multiple access interference (MAI) was found to outperform matched filter receiver but they performs less than decorrelating detector receivers. From the simulation result the decorrelating detector achieve 10^{-2} BER around 10.5dB SNR whereas LMS and CMA achieve this BER around 12.5dB and 16dB SNR respectively. The matched filter cannot achieve this BER within 20dB SNR. From the result we can deduce that decorrelating detector is an optimal receiver that almost eliminates multiple access interference. In addition to this it was found that the BER performance of LMS receiver is much better than the CMA receiver.

It was also shown that under perfect synchronization Walsh-Hadamard sequence can completely eliminate MAI due to their orthogonal nature. As the spreading gain of the spreading codes is higher, their ability to reduce MAI also increases which is at the cost of band width.

Key Words: CDMA, LMS algorithm, CMA, decorrelating detector, matched filter receiver.

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Content

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Acronyms

AWGN	Additive white Gaussian noise
BER	Bit error rate
BPSK	Binary phase shift keying
CDMA	Code division multiple access
CM	constant modulus
CMA	Constant modulus Algorithm
DS-CDMA	Direct sequence code division multiple access
DSSS	Direct sequence spread spectrum
FDD	Frequency division duplexing
FDMA	Frequency division multiple access
GSM	Global systems for mobile communications
ISI	Inter symbol interference
LMS	Least mean square error
MAI	Multiple access interference
MMSE	Minimum mean square error
Pdf	probability density function
PN	Pseudo random noise
PSD	Power spectral density
RLS	Recursive least square
S-CDMA	Synchronous CDMA
SDMA	Space division multiple access
SNR	Signal to noise ratio
TDD	Time division duplexing
TDMA	Time division multiple access

Chapter One

1. Introduction

1.1 Motivation and Background

One of the basic concept in data communication is the idea of allowing several transmitters to send information simultaneously over a single communication channel. This allows several users to share a frequency band. This concept is called multiplexing. CDMA employs spread-spectrum technology and a special coding scheme (where each transmitter is assigned a code) to allow multiple users to be multiplexed over the same physical channel. CDMA, in contrast to Frequency Division Multiple Access (FDMA) and Time Division Multiple Access (TDMA), poses no restrictions to the time interval and frequency band to be used for the transmission of different users. All users can transmit simultaneously while occupying the whole available bandwidth. They are separated by uniquely (per user) assigned codes with proper low cross-correlation properties. Thus, while inter-user interference is strictly avoided in TDMA and FDMA systems by assigning different portions of time slots or bandwidth to different users, respectively, inter-user interference, referred to as multiple-access interference (MAI), is inherent in CDMA techniques and is the limiting capacity factor (interference-limited systems) [1].

Some receivers have been proposed that can suppress multiple access interference. For example Verdu has proposed optimal multi-user receiver [2]. This receiver structure and subsequent sub-optimal simplification of it are still generally too complex and /or assume too much knowledge about the channel environment, including accurate user power levels and synchronization information. An alternative approach is to use an adaptive filter receiver structure. The achievable channel capacity is then at least 70% to 80% of the theoretical additive white Gaussian noise (AWGN) channel capacity [3]. Such receiver are often called MMSE receiver, and typically use the least mean square (LMS) tap update algorithm. This algorithm is attractive due to its simplicity and robustness. The main drawback is, slow convergence speed and require long training period. Another commonly used adaptive algorithm in adaptive CDMA receivers is blind adaptive algorithm which works based on the constant modulus criteria and doesn't require training bits.

1.2 Literature Review

Multiple access interference in code division multiple access (CDMA) signal demodulation is caused by different power level of the user's transmitter, due to cross correlation among the spreading sequence of the user signals and due to channel noise that destroy the orthogonal property among the spreading sequences.

The most commonly used techniques to minimize this multiple access interference in CDMA receiver are adaptive power leveling and optimum multi-user detector.

Adaptive algorithms are used to minimize multiple access interference resulting from the power level difference of each user's signal which is called near far effect [3]. This technique uses different adaptive algorithms to compensate for the power level difference between each signal level. But it doesn't take into account the multiple access interference caused due to undesired users' signal. In optimum multi-user detector, [2] verdu has shown that a virterbi type algorithm provides an optimal method of demodulation in additive Gaussian noise channel. In addition to its optimality the verdu type detector is near –far resistant, a problem that causes multiple access interference due to different power levels of the received signals. Although this optimal multi-user detector is more efficient than the conventional single user demodulator in minimizing multiple access interference in near far environment, its complexity increases exponential with the number of active users. An alternative detector is the decorrelating detector which is a linear receiver (in operation and complexity) that retain the near-far resistance of the optimal multi-user detector [4]. A disadvantage of both the decorrelating detector and the optimal multi-user detector is that they require knowledge of the spreading codes of all the active users in the network. This is potentially a significant burden from operational viewpoint. But by introducing an adaptive element into the downlink receiver, performance will be enhanced.

1.4 Problem Statement

One of the major drawbacks of CDMA is the capacity and BER performance limitations inflicted by multiple access interference (MAI). This interference arises due to the non-orthogonality of the spreading sequence of the signal from different mobile users and power level difference among the transmitter.

Optimal receiver has been proposed by Verdu to suppress multiple access interference [2]. This receiver and optimal receivers are too complex and assume too much knowledge about the channel environment including accurate power levels and synchronization information. An alternative approach is to use adaptive filter receiver structure. This receiver exhibits higher performance than the previous one. This receiver uses different adaptive algorithms each with their drawback and good side. The widely used algorithms are least mean square (LMS) and blind adaptive algorithm like constant modulus algorithm (CMA).

LMS algorithm needs training sequences, which means before information bits are transmitted, several training bits should be transmitted for the adaptive filter to converge. When the channel is time-varying, the training sequence must be periodically transmitted according to the variant rate of variation of the channel. However, training sequence decreases the channel bandwidth efficiency. Whereas in the case of blind adaptive algorithm there is no need for training sequence it only needs appropriately initializing update tap weights. The main problem of this algorithm is that if the first response is not good it converges in undesired way.

1.5 Objective

1.5.1 General Objective

The aim of this thesis is to evaluate the performance of least mean square error (LMS) and constant modulus algorithm (CMA) (is one type of blind adaptive algorithm) in suppressing multiple access interference. This limits the maximum number of users one base station can support (in uplink case) and as well degrades the signal quality received by each active user (in down link). This is also true in uplink case.

1.5.2 Specific Objectives

- ✓ Studying LMS adaptive algorithm and CMA algorithm in order to find the good tradeoff between the performance, complexity and bandwidth-extension
- ✓ Comparing the efficiency of receivers that employ adaptive algorithms and those that do not use adaptive algorithm
- ✓ The convergence rate and signal to noise ratio of the two adaptive algorithm will be studied
- ✓ Comparing the good tradeoff between the receiver efficiency and complexity among non adaptive and adaptive receivers.
- ✓ Comparison of BER performance of matched filter CDMA receiver and adaptive receivers
- ✓ Comparing spreading codes in their ability to reduce MAI and their spreading gain effect on MAI reduction.

1.6 Organization of the Thesis

The remainder of the thesis is organized as follow: In chapter two different multiple access systems and spreading code for CDMA are presented. In chapter three mathematical representation of CDMA signal from transmitter to receiver is presented and MAI analysis is done. In chapter four CMA and LMS adaptive algorithms for CDMA receiver is considered. Using these algorithms a block diagram for CDMA is developed on which the simulation is based.

In chapter four simulation results for Walsh-Hadamard sequence and Gold sequence for suppressing MAI is considered. In addition to this their spreading gain effect on MAI reduction and bandwidth extension was considered. Finally the performance of adaptive CDMA receiver, matched filter receiver and decorrelating detector was done.

1.7 Methodology Used in This Thesis

The methodology used in this thesis is Literature survey, system modeling and analysis and design test and performance evaluation. In literature review works closely related to the thesis are studied to get in depth knowledge in the area. In the system analysis and modeling part the effect of the number of users and spreading gain on MAI is mathematically analyzed. Using adaptive algorithms the schematic block diagram for CDMA receiver is developed. Depending on the schematic representation performance evaluation of the receiver is done.

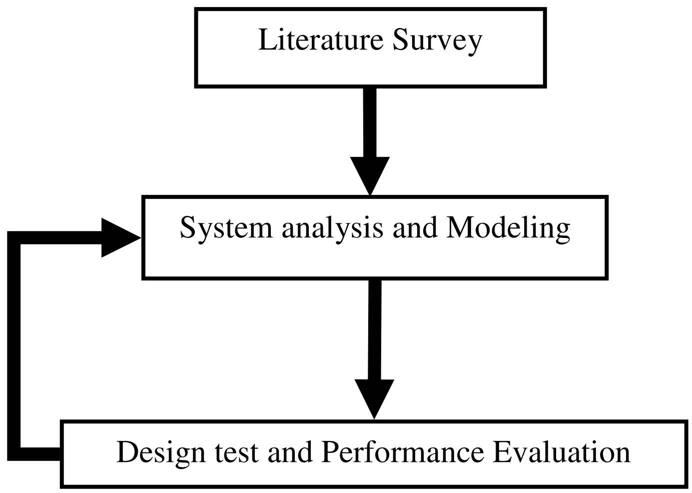


Figure 1.1 Summary of the methodology

Chapter Two

2. Code Division Multiple Access Spread Spectrum System

2.1 Introduction

Wireless cellular telephony has been growing at a faster rate than wired-line telephone networks [5]. This growth is a result of the recent improvements in the capacity of wireless channels due to the use of multiple access techniques, which allow many users to share the same channel for transmission, in association with advanced signal processing algorithms. CDMA is a popular technology for cellular communications, which is based on spread spectrum technology [6]. Unlike other multiple access techniques such as FDMA and TDMA, which are limited in frequency band and time duration respectively, CDMA uses all of the available time-frequency space.

In general, spread spectrum signals are commonly used for [7]:

- Combating or suppressing the detrimental effects of interference due to jamming, interference arising from other users of the channel, and self-interference due to multipath propagation.
- Hiding a signal by transmitting it at low power spectral density and, thus making it difficult for an unintended listener to detect it in the presence of background noise.
- Achieving message privacy in the presence of other listeners.

2.2 Multiple Access Techniques

A limited amount of bandwidth is allocated for wireless services. A wireless system is required to accommodate as many users as possible by effectively sharing the limited bandwidth. Therefore, in the field of communications, the term multiple access could be defined as a means of allowing multiple users to simultaneously share the finite bandwidth with least possible degradation in the performance of the system. There are several techniques through which multiple accessing can be achieved. There are four basic schemes:

1. Frequency Division Multiple Access (FDMA)
2. Time Division Multiple Access (TDMA)
3. Space Division Multiple Access (SDMA)
4. Code Division Multiple Access (CDMA)

2.2.1 Frequency Division Multiple Access (FDMA)

FDMA is one of the earliest multiple-access techniques for cellular systems when continuous transmission is required for analog services. In this technique the bandwidth is divided into a number of channels and distributed among users with a finite portion of bandwidth for permanent use as illustrated in Figure 2.1. The receiver can tune to the specified band and demodulate the information. The vertical axis that represents the code is shown here just to make a clear comparison with CDMA (discussed later in this Chapter). The channels are assigned only when demanded by the users. Therefore when a channel is not in use it becomes a wasted resource. FDMA channels have narrow bandwidth (30 KHz) and therefore they are usually implemented in narrowband systems. Since the user has his portion of the bandwidth all the time, FDMA does not require synchronization or timing control, which makes it algorithmically simple. Even though no two users use the same frequency band at the same time, guard bands are introduced between frequency bands to minimize adjacent channel interference. Guard bands are unused frequency slots that separate neighboring channels. This leads to a waste of bandwidth. When continuous transmission is not required, bandwidth goes wasted since it is not being utilized for a portion of the time. In wireless communications, FDMA achieves simultaneous transmission and reception by using Frequency division duplexing (FDD). In order for both the transmitter and the receiver to operate at the same time, FDD requires duplexers. The requirement of duplexers in the FDMA system makes it expensive [8].

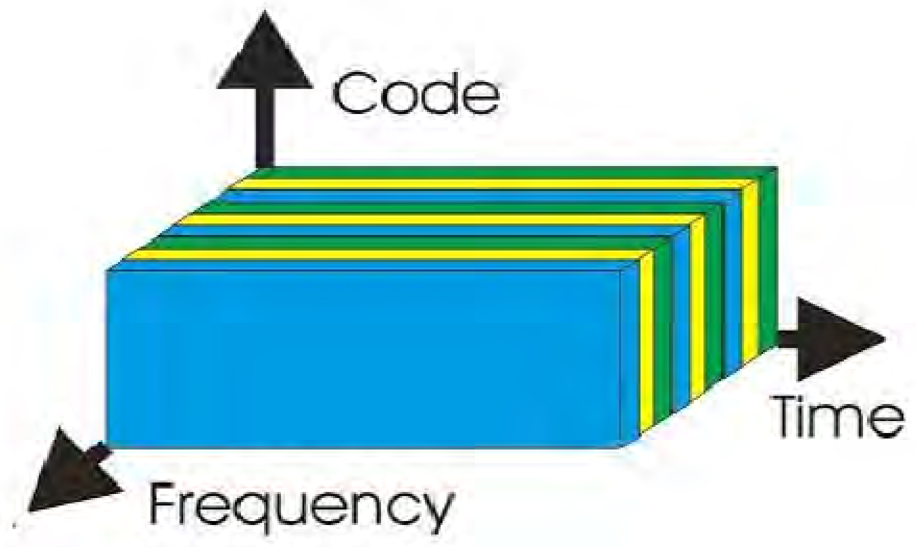


Figure 2.1 channel usage by FDMA

2.2.2 Time Division Multiple Access (TDMA)

In digital systems, continuous transmission is not required because users do not use the allotted bandwidth all the time. In such systems, TDMA is a complimentary access technique to FDMA. Global Systems for Mobile communications (GSM) uses the TDMA technique. In TDMA, the entire bandwidth is available to a user but only for a finite period of time. Transmission of data is only possible during this time-slot, after that the transmitter has to wait until it gets another time-slot. Synchronization of all users is an important issue in this concept. Consequently, there must be a central unit (base-station) that controls the synchronization and the assignment of time-slot. This means that this technique is difficult to apply in random-access systems. In most cases the available bandwidth is divided into fewer channels compared to FDMA and the users are allotted time slots during which they have the entire channel bandwidth at their disposal. This is illustrated in Figure 2.2. TDMA requires careful time synchronization since users share the bandwidth in the frequency domain. Since the number of channels are less, inter channel interference is almost negligible, hence the guard time between the channels is considerably smaller. Guard time is spacing in time between the TDMA bursts. In cellular communications, when a user moves from one cell to another there is a chance that user could experience a call drop if there are no free time slots available. TDMA uses different time slots for transmission and reception. This type of duplexing is referred to as Time division duplexing (TDD). TDD does not require duplexers.

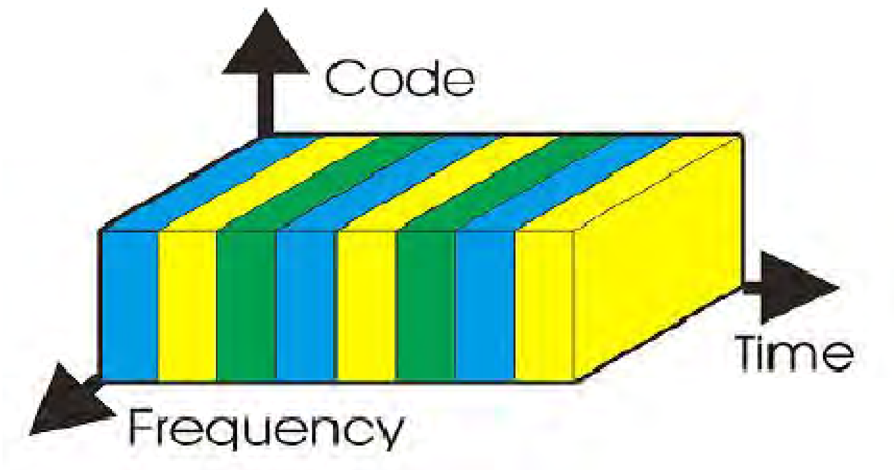


Figure 2.2 channel usage by TDMA

In the previous two systems, each user is essentially orthogonal in either time or frequency, which makes detection and demodulation a relatively easy task. However, these systems cannot take advantage of the properties of the transmitted data. Voice data tends to be very bursty in nature, so much of the time no data is being sent over the channel. This inefficiency tends to limit the capacity of the system.

2.2.3 Space Division Multiple Access (SDMA)

SDMA utilizes the spatial separation of users in order to optimize the use of the frequency spectrum. A primitive form of SDMA is when the same frequency is re-used in different cells in a cellular wireless network. However, for limited co-channel interference it is required that the cells be sufficiently separated. This limits the number of cells a region can be divided into and hence limits the frequency re-use factor. A more advanced approach can further increase the capacity of the network. This technique would enable frequency re-use within the cell. It uses a

Smart antenna technique that employs antenna arrays backed by some intelligent signal processing to steer the antenna pattern in the direction of the desired user and places nulls in the direction of the interfering signals. Since these arrays can produce narrow spot beams, the frequency can be re-used within the cell as long as the spatial separation between the users is sufficient. Figure 2.3 shows three users served by SDMA using the same channel within the cell. In a practical cellular environment it is improbable to have just one transmitter fall within the receiver beam width. Therefore it becomes imperative to use other multiple access techniques in conjunction with SDMA. When different areas are covered by the antenna beam, frequency can be re-used, in which case TDMA or CDMA is employed, for different frequencies FDMA can be used [9].

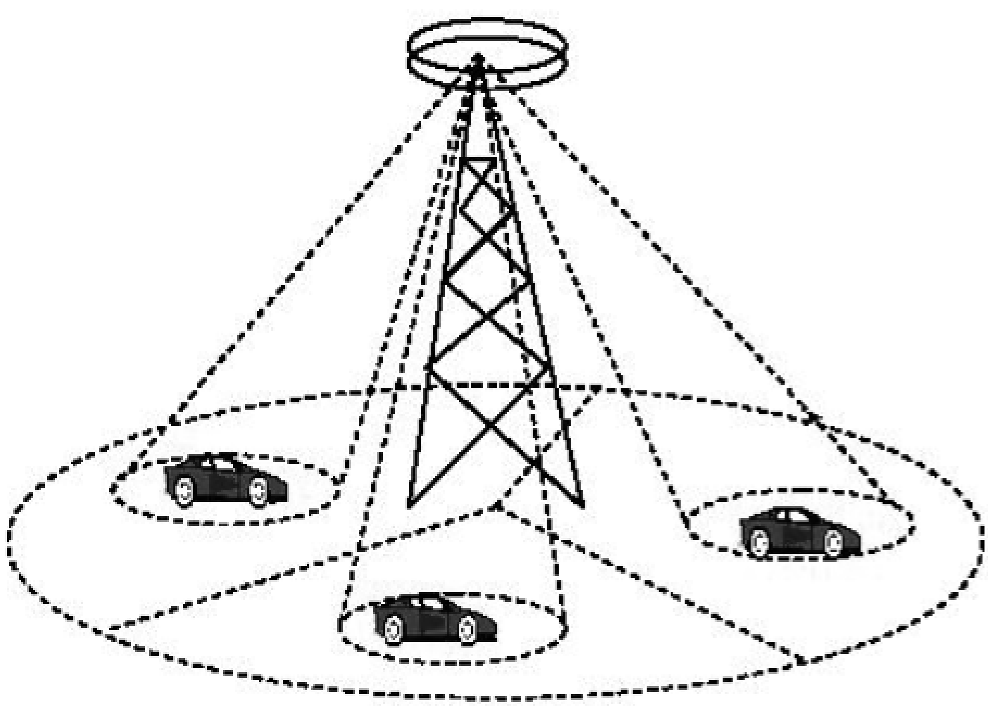


Figure 2.3 Intra-cells SDMA

2.2.4 Code Division Multiple Access

CDMA is a system which is based on spread spectrum technology. It was initially developed for military antijamming. Since 1940's spread spectrum systems were developed for antijam and low probability of intercept (LPI) applications [10][6][11]. A spread spectrum signal is characterized by the bandwidth which is used to decrease the transmitting power and eliminate interference from other users as well as interference from the same user. In a CDMA system the users are spread across both frequency and time in the same channel as shown in Figure 2.4. The capacity of this system depends on the amount of interference from the other users since the fundamental problem of CDMA is that each user causes multiple access interference (MAI) affecting all the other users.

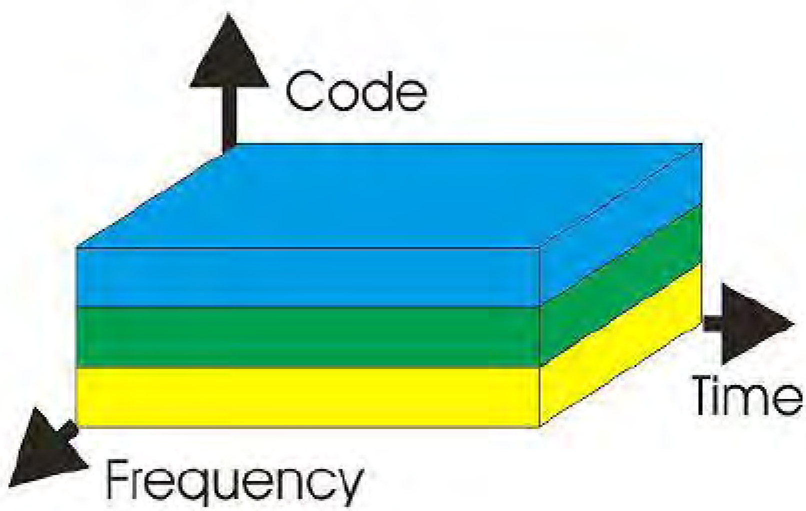


Figure 2.4 channel usage by CDMA

The pseudo-random codes or PN sequences used to increase the bandwidth and are essential for providing security as in the situation of antijaming. In the antijam systems spectral spreading secures the signal against narrowband interferers [12] to make the detection as difficult as possible for an unwanted interceptor by hiding the signal in the noise as shown in Figure 2.5.

The PN codes spread the baseband data before transmission. The signal is transmitted in a channel, which is below noise level. The receiver then uses a correlator to “despread” the wanted signal, which is passed through a narrow band pass filter. Unwanted signals will not be “despread” and will not pass through the filter. Codes take the form of a carefully designed one/zero sequence produced at a much higher rate than that of the baseband data. The rate of a spreading code is referred to as chip rate rather than bit rate.

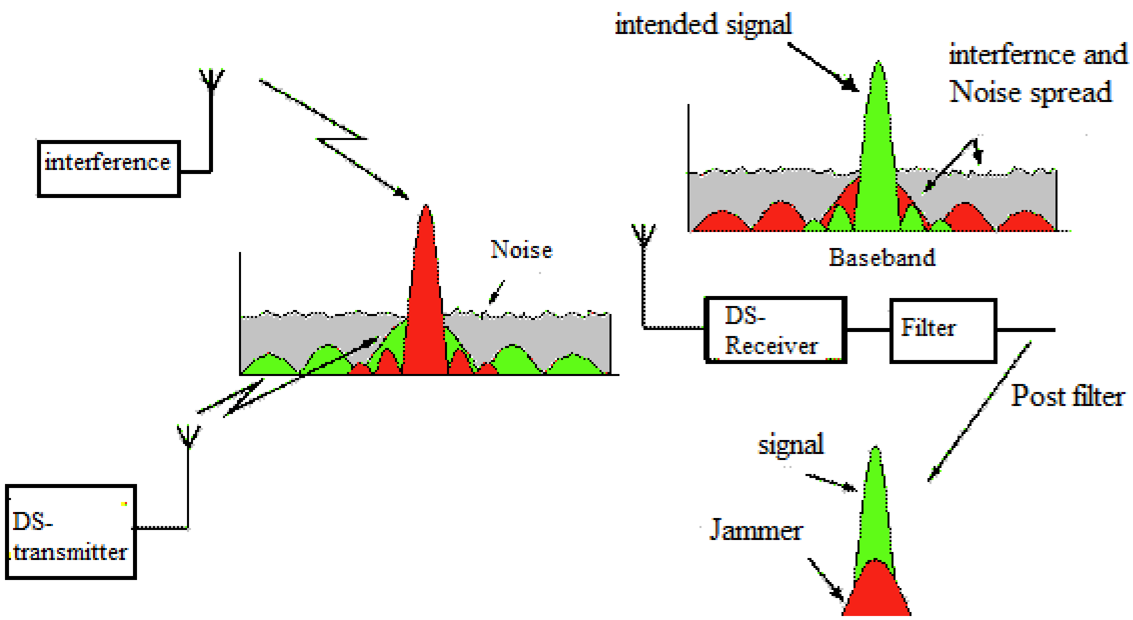


Figure 2.5 Interference rejections in CDMA

2.5 Spreading Codes

In CDMA system, a user signal is multiplied by pseudo random sequence at the transmitter. This sequence must be known to the receiver in order to recover the information bearing signal. A PN code is a sequence of chips valued -1 and 1 (polar) or 0 and 1 (non-polar) and has noise-like properties.

There are many PN sequences such as *m*–sequences, Gold sequences, Walsh-Hadamard codes and Kasami sequences.

2.5.1 Maximal Length Sequence (M-Sequence)

The maximum-length shift-register sequence, or shortly m-sequence, is probably the most widely known PN sequence. It can be generated by n-stage shift register with linear feedback. The sequence length generated is $N = 2^n - 1$ chips long, meaning that they are as long as a shift register can produce. To construct m sequences of length N, (N is also the period of the sequence or spreading factor), one needs a primitive polynomial $h(x)$ of degree n, (where n is the number of shift register)

$$h(x) = h_0 x^n + h_1 x^{n-1} + h_2 x^{n-2} + \cdots + h_{n-1} x + h_n \qquad (2.1)$$

Where $h_0 = h_n = 1$. A primitive generator polynomial always yields an m-sequence [13]. This polynomial specifies a linear feedback shift register. If the register contains $S_{n-1}, S_{n-2}, \ldots\ldots S_1, S_0$ at time $l = n - 1$, then the output at time $l = n$ is:

$$S_l = \sum_{i=1}^{n} h_i S_{l-i}\,, \qquad l \leq n, \qquad h_i \in [0,1] \qquad (2.2)$$

The m sequences have good autocorrelation property and are being used in many applications including IS-95. As the cross-correlation property of these sequences is relatively poor compared to Gold codes, the same sequence with different offset are usually used for different users or for different base stations. With this method, the discrimination property between different spreading codes depends only on partial autocorrelation property.

2.5.2 Gold Sequence

Gold codes have worse autocorrelation properties than maximal-length codes, but better cross-correlation properties if properly designed. The chip sequences associated with a Gold code are produced by the binary addition of two m-sequences each of length $2^n - 1$, and they inherit the balanced, run length, and shift properties of these component codes, hence are referred to as pseudorandom sequences. Gold codes take advantage of the fact that if two distinct m-sequences with time shifts τ_1 and τ_2 are modulo-2 added together, the resulting sequence is unique for every unique value of τ_1 or τ_2. Thus, a very large number of unique Gold codes can be generated, which allows for a large number of users in a multiuser system. However, if the m-sequences that are modulo-2 added to produce a Gold code are chosen at random, the cross-correlation of the resulting code may be quite poor. Thus, Gold codes are generated by the chip sequences associated with the modulo-2 addition of preferred pairs of m-sequences. These preferred pairs are chosen to obtain good cross-correlation in the resulting Gold code. However, the preferred pairs of m-sequences have different autocorrelation properties than general m-sequences. A method for choosing the preferred pairs such that the cross-correlation and autocorrelation functions of the resulting Gold code are bounded was given by Gold in [14], and can also be found in [15]

The preferred sequences are chosen so that Gold codes have a three-valued cross-correlation with values

$$\rho_{ij}(\tau) = \begin{cases} -1/N \\ -t(n)/N \\ \\ \dfrac{1}{N}[t(n) - 2] \end{cases} \qquad (2.3)$$

where
$$t(n) = \begin{cases} 2^{(n+1)/2} + 1 & for\ n\ odd \\ 2^{(n+2)/2} + 1 & for\ n\ even \end{cases} \qquad (2.4)$$

and n is the number of shift register per preferred m-sequence

The autocorrelation takes on the same three values.

2.5.3 Kasami Code

Kasami chip sequences have similar properties as the preferred sequences used to generate Gold codes, and are also derived from m-sequences. However, the Kasami codes have better cross-correlation properties than Gold codes. There are two different sets of Kasami chip sequences that are used to generate Kasami codes, the large set and the small set. To generate the small set, we begin with an m-sequence $\boldsymbol{a}$ of length $2^n - 1$ for n even and form a new shorter sequence $\boldsymbol{a'}$ by sampling every $2^{n/2} + 1$elements of $\boldsymbol{a}$. The resulting sequence $\boldsymbol{a'}$ will have period $2^{n/2} - 1$. We then generate a small set of Kasami sequences by taking the modulo-2 sum of $\boldsymbol{a}$ with all cyclic shifts of the $\boldsymbol{a'}$ sequence. There are $2^{n/2} - 2$ such cyclic shifts, and by also including the original sequence $\boldsymbol{a}$ in the set, we obtain a set of $2^{n/2}$ binary sequences of length $2^n - 1$. As with the Gold codes, the autocorrelation and cross-correlation of the Kasami spreading codes obtained from the Kasami chip sequences are three-valued, taking on the values

$$\rho_{ij}(\tau) = \begin{cases} -1/N \\ -s(n)/N \\ \frac{1}{N}[s(n) - 2] \end{cases} \qquad (2.5)$$

where: $s(n) = 2^{n/2} + 1$. Since $|s(n)| < |t(n)|$, Kasami codes have better autocorrelation and cross-correlation than Gold codes. In fact, the Kasami codes achieve theWelch lower bound for the autocorrelation and cross-correlation for any set of $2^{n/2}$ sequences of length $2^n - 1$, and hence are optimal in terms of minimizing the autocorrelation and cross-correlation for any such code [15][16].

The large set of Kasami sequences is formed in a similar way as the small set. It has a larger number of sequences than the smaller set, and hence can support more users in a multiuser system, but the autocorrelation and cross-correlation properties across the spreading codes generated from this larger set are inferior to those generated from the smaller set. To obtain the large set, we take an m-sequence **a** of length $N = 2^n - 1$ for n even and form two new sequences $\boldsymbol{a'}$ and $\boldsymbol{a''}$ by sampling the original sequence every $2^{n/2} + 1$ elements for $\boldsymbol{a'}$ and every $2^{(n+2)/2} + 1$ elements for $\boldsymbol{a''}$. The set is then comprised by adding $\boldsymbol{a}$, $\boldsymbol{a'}$, and $\boldsymbol{a''}$ for all cyclic shifts of $\boldsymbol{a'}$ and $\boldsymbol{a''}$. The number of such sequences is $2^{3n/2}$ if n is a multiple of 4 and $2^{3n/2} +$

$2^{n/2}$ if $mod_4(n) = 2$. The autocorrelation and cross-correlation of the spreading codes generated from this set can take on one of five values:

$$\rho(\tau) = \begin{cases} \frac{-1}{N} \\ \frac{1}{N}(-1 \pm 2^{n/2}) \\ \frac{1}{N}(-1 \pm (2^{n/2} + 1)) \end{cases} \quad (2.6)$$

Since these values exceed those for codes generated from the small Kasami set, we see that the Kasami codes generated from the large Kasami set have inferior cross-correlation and autocorrelation properties to those generated from the small Kasami set.

2.5.4 Walsh-Hadamrd Code

Walsh-Hadamard codes of length $N = {T_s}/{T_c}$, (where T_s is the symbol period and T_c is the chip period) that are synchronized in time are orthogonal over a symbol time, so that the cross-correlation of any two sequences is zero. Thus, synchronous users spreaded withWalsh-Hadamard codes can be separated out at the receiver with no interference among them, as long as the channel does not destroy the orthogonality of the codes. While it is possible to synchronize users on the downlink, where all signals originate from the same transmitter, it is more challenging to synchronize users in the uplink, since they are not co-located. Hence, Walsh-Hadamard codes are rarely used for DSSS uplink channels.

Walsh-Hadamard sequences of length N are obtained from the rows of an $N \times N$ Hadamard matrix H_N. For N = 2 the Hadamard matrix is

$$H_2 = \begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix} \quad (2.7)$$

Larger Hadamard matrices are obtained using H_2 and the recursion

$$H_{2N} = \begin{bmatrix} H_N & H_N \\ H_N & H_N \end{bmatrix} \tag{2.8}$$

Each row of H_N specifies the chip sequence associated with a different sequence, so the number of spreading codes in a Walsh-Hadamard code is N. Thus, DSSS with Walsh-Hadamard sequences can support at most $N = T_s/T_c$ users. Since DSSS uses roughly N times more bandwidth than required for the information signal, approximately the same number of users could be supported by dividing up the total system bandwidth into N non overlapping channels (frequency-division). Similarly, the same number of users can be supported by dividing time up into N orthogonal timeslots (time-division) where each user operates over the entire system bandwidth during his time slot. Hence, any multiuser technique that assigns orthogonal channels to the users such that they do not interfere with each other accommodates approximately the same number of users.

Chapter Three

3. CDMA Signal Model and Multiple Access Inference Analysis (MAI)

3.1 Introductions

In this chapter, CDMA mathematical model is represented from transmitter to receiver. And a mathematical analysis of the effect of number of interfering users and spreading factor on the BER of the desired user signal is shown.

3.2 CDMA Transmitter

In the transmitter part of the DS-CDMA system, each user data symbol is modulated using a unique signature waveform $a_i(t)$, having a normalized energy over a data bit interval T, i.e.,

$$\int_0^T ||a_i(t)||^2 dt = 1, \qquad (3.1)$$

given by [17]:

$$a_i(t) = \sum_{j=1}^{N} a_i(j) P_c(t - jT_c), \ \ j = 1 \ldots N, \qquad (3.2)$$

Where the $a_i(j)$ represents the j^{th} chip of the i^{th} user's code sequence and are assumed to be elements of {-1, +1}, and $P_c(t)$ is the chip pulse waveform defined over the interval $[0 \quad T_c]$ with T_c as the chip duration which is related to the bit duration through the processing gain N, with $T_c = {}^{T_s}/_{N}$. In the following analysis we consider Binary Phase Shift Keying (BPSK) modulation for signal transmission. Then, the i^{th} user transmitted signal is

$$S_i(t) = \sqrt{2P_i} b_i(t) a_i(t), \qquad i = 1, \ldots K, \qquad (3.3)$$

where P_i is the i^{th} user bit power and

$$b_i(t) = \sum_{m=1}^{N_b} b_i(m)P(t - mT), \quad b_i(m) \epsilon \{-1, +1\} \tag{3.4}$$

where: $b_i(m)$ is the binary data sequence for i^{th} user,

N_b is the number of received data bits,

Figure 3.1 depicts the base band CDMA modulation. The information-bearing baseband signal of the i^{th} user $b_i$is modulated by the corresponding spreading code $a_i(t)$. A pulse shaping filter, e.g., the raised cosine pulse shape [7], can be used to limit the bandwidth of the spreading code. For an illustrative purpose, however, the rectangular pulse shape filter is assumed. The bandwidth of the modulated signal $S_i(t)$ is greater than that of $b_i(t)$ by the processing gain or spreading factor which is defined by the number of chips per bit or the ratio of the symbol interval T_s to the chip interval T_c [18].

In Figure 3.1, the spreading gain is designed to be five.

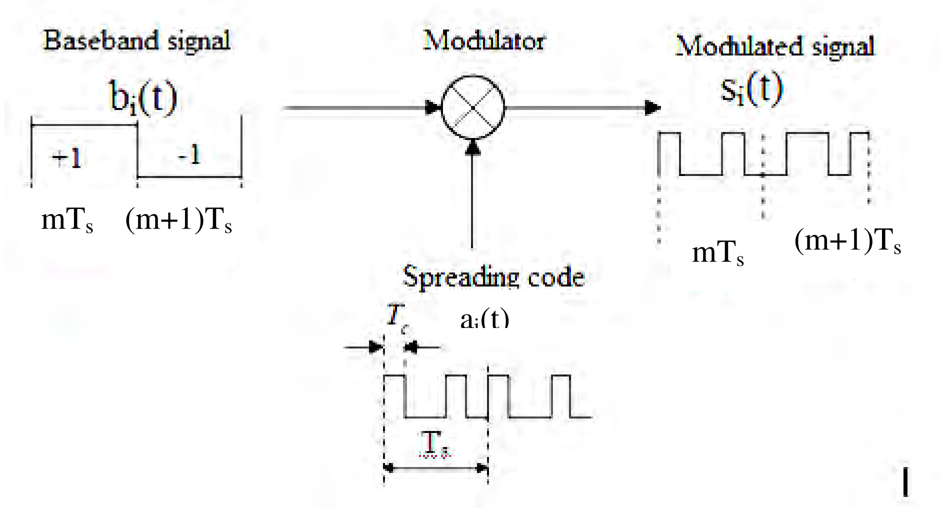


Figure 3.1: Baseband model of the modulation of information bits of the k^{th} user

The spreading code $a_i(t)$ with spreading gain $N \triangleq \frac{T_s}{T_c} = 5$ is used to modulate $b_i(t)$.

3.3 Received CDMA Signal

Signal model and performance of multiuser spread spectrum also depends on whether the multiuser system is a **downlink** channel (one transmitter to many receivers) or an **uplink** channel (many transmitters to one receiver). These channel models are illustrated in Figure 3.2 and Figure 3.3. The downlink channel is also called a broadcast channel or forward link, and the uplink channel is also called reverse link. The performance differences of direct sequence spread spectrum (DSSS) in uplink and downlink channels result from the fact that in the downlink, all transmitted signals are typically synchronous, since they originate from the same transmitter. Moreover, both the desired signal and interference signals pass through the same channel before reaching the desired receiver. In contrast, users in the uplink channel are typically asynchronous, since they originate from transmitters at different locations, and the transmitted signals of the users travel through different channels before reaching the receiver [19].

3.3.1 Received Signal in Uplink Model

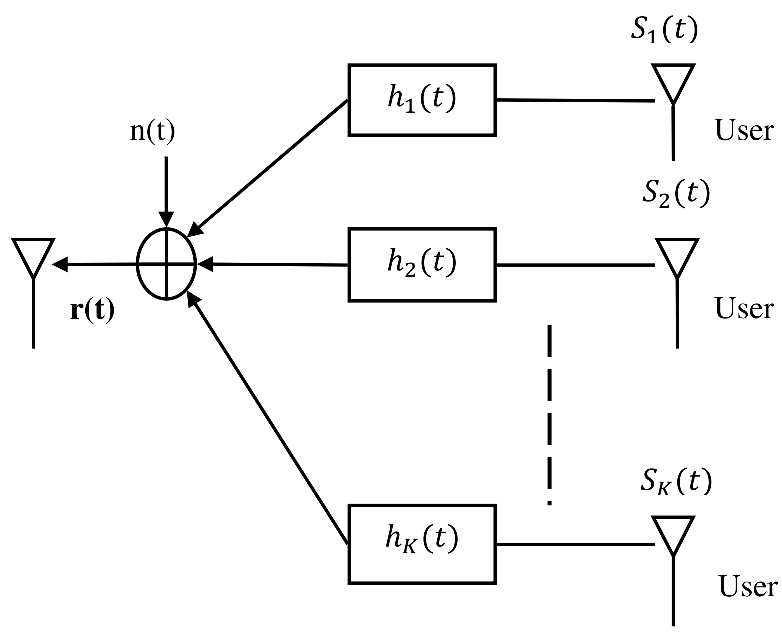


Figure 3.2 Signal model for uplink DS-CDMA

3.3.2 Received Signal in the Downlink Model

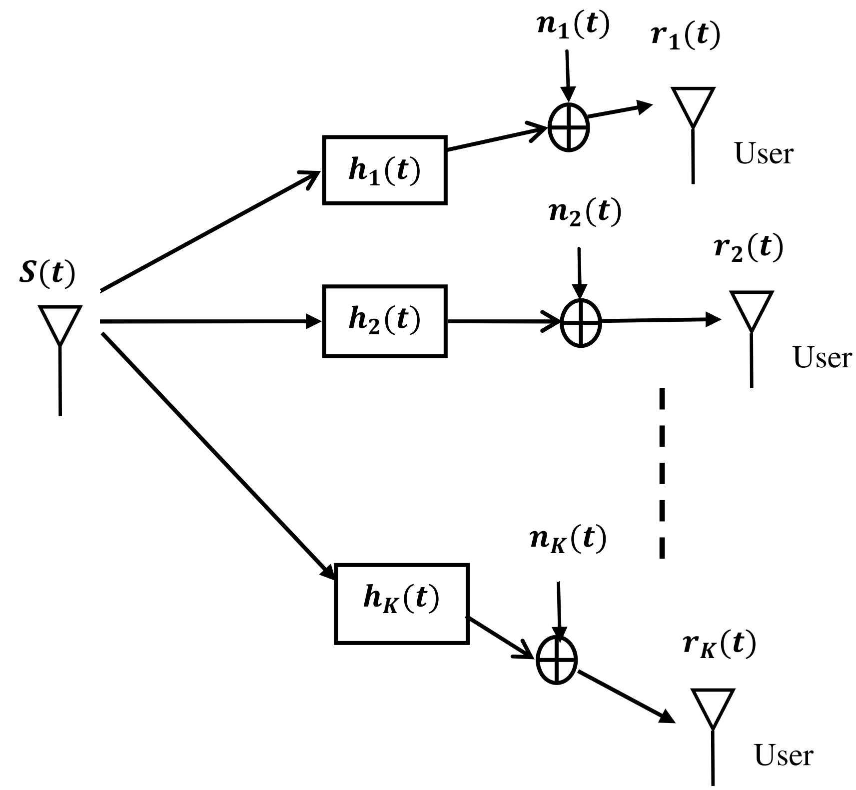


Figure 3.3 Signal model for down link DS-CDMA

$$r_i(t) = \sum_{l=1}^{L-1}\sum_{i=1}^{K} \alpha_{li}\sqrt{2P_i}b_i(t-\tau_{li})a_i(t-\tau_{li}) + n(t), \qquad (3.5)$$

where: $n(t)$is the two-sided power spectral density ${}^{N_0}/_{2}$ additive white Gaussian noise. α_{li} represents the l^{th} multipath gain of the i^{th} user signal, and τ_{li} represent the l^{th} multipath delay of the i^{th} user signal, for L resolvable multipath, for all users $i = 1, \dots K$.

Assuming perfect chip timing at the receiver, the received signal in the above equation is passed through a chip-matched filter followed by sampling at the end of each chip interval to give the m^{th} data bit interval:

$$y_{m,l} = \int_{mT+lT_c}^{mT+(l+1)T_c} r(t)P_c(t-lT_c)dt, \qquad l = 0,1,\dots,N-1, \tag{3.6}$$

where $P_c(t)$ is the chip pulse shape, which is taken to be a rectangular pulse with amplitude $1/\sqrt{N}$. Using the above equation and taking the k^{th}user as the desired one, the output of the chip matched filter after sampling for the m^{th}data bit is given by(for simplicity of mathematical analysis we assumed that $0 \le \tau < T_c$)

$$y_{m,l} = \sqrt{2P_k}\alpha_{lk}b_k(m)a_k(l) + \sum_{\substack{i=1 \\ i\neq k}}^{K} \alpha_{li}\sqrt{2P_i}b_i(m)a_k(l)a_i(l) + n(m,l), \tag{3.7}$$

where the MAI is

$$I(m,l) = \sum_{\substack{i=1 \\ i\neq k}}^{K} \alpha_{li}\sqrt{2P_i}b_i(m)a_k(l)a_i(l) \tag{3.8}$$

and the components of the noise $n(m,l)$ vector consists of independent Zero-mean Gaussian random variables with variance $N_0/_{2N}$.

3.4 Channel Model

Channel is defined as the frequency band in the frequency domain, or its digital time slot equivalent in the time domain, established to provide a communication path between the transmitter and receiver. Also, a channel can be described as everything from source to the destination of a radio signal which may include the physical medium such as free space, fiber optics link or waveguide structure [20]. The term fading refers to the time-varying channel conditions such as amplitude [7]. When the transmitted signal travels over multiple reflective paths, multipath propagation phenomenon rises. This is a common phenomenon in a wireless mobile communication system which can cause fluctuations in the received signal's amplitude, phase and angle of arrival giving rise to multipath fading [5] [21]. The performance of mobile communication system is primarily affected by the dynamics of multipath fading [22].

3.4.1 AWGN Channel

For this model of the channel, the modulated signal $S(t)$ is passed through a channel which corrupts the signal with additive white Gaussian (AWGN). This channel model is of interest since it applies for various classes of physical channels and due to its mathematical tractability. For an AWGN channel, the received signal $r(t)$ seen at the receiver represents the summation of the modulated signals and noise. In the following subtopic i.e. in multiple access interference analysis we use this channel as a model for the simplicity of mathematical analysis since our attention is analyzing the effect of multiple access interference rather than ISI that results due to multipath effect.

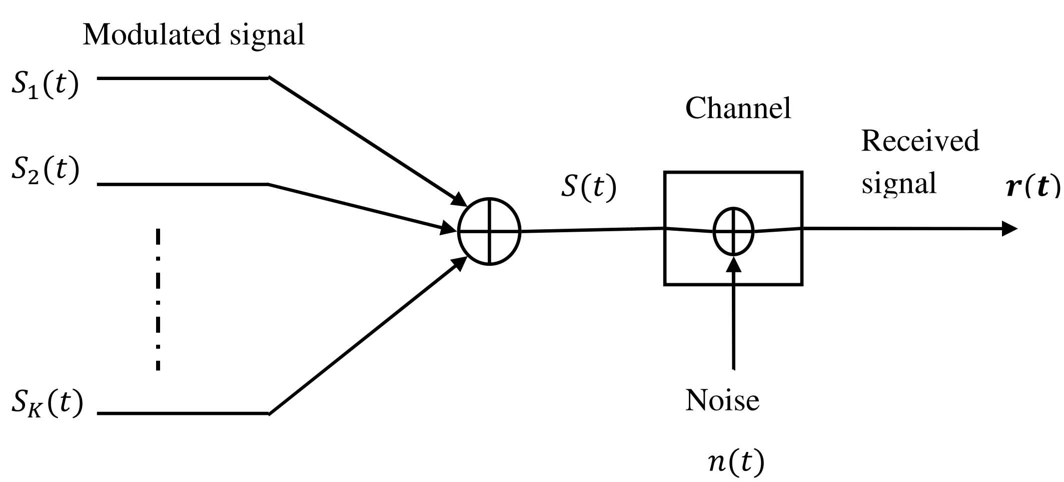


Figure 3.4 signal model for $\boldsymbol{r(t)}$ propagating through an AWGN channel

Where $n(t)$ is a zero mean additive white Gaussian noise. Figure 3.4 shows the signal model of the AWGN channel. The additive Gaussian noise includes receiver thermal noise introduced by electronic components, background electromagnetic noise and also man-made interference [18].

3.4.2 Rayleigh Fading Channel

DS-CDMA systems involve a spreading process which results in a transmitted signal whose bandwidth is much wider than the channel coherence bandwidth, and therefore undergoes frequency-selective fading. This type of fading is typically modeled by a linear filter which for the k^{th} user is characterized by a complex valued low pass equivalent impulse response [23–25]

$$h_k(t) = \sum_{l=1}^{L_p} \alpha_{k,l} e^{-j\theta_{k,l}} \delta(t - \tau_{k,l}) \tag{3.9}$$

where δ(.) is the Dirac delta function, l the propagation path index, and $\{\alpha_{k,l}\}_{l=1}^{L_p}$, $\{\theta_{k,l}\}_{l=1}^{L_p}$, and $\{\tau_{k,l}\}_{l=1}^{L_p}$ are the random path amplitudes, phases, and delays, respectively. We assume that the sets $\{\alpha_{k,l}\}_{l=1}^{L_p}$, $\{\theta_{k,l}\}_{l=1}^{L_p}$, and $\{\tau_{k,l}\}_{l=1}^{L_p}$ are mutually independent.

In (3.10), L_p is the number of resolvable paths (the first path being the reference path whose delay $\tau = 0$) and is related to the ratio of the maximum delay spread τ_{max} and to the chip time duration T_c. We assume slow fading, so that L_p is constant over time, and $\{\alpha_{k,l}\}_{l=1}^{L_p}$, $\{\theta_{k,l}\}_{l=1}^{L_p}$, and $\{\tau_{k,l}\}_{l=1}^{L_p}$ are all constant over a symbol interval. If the different paths of a given impulse response are generated by different scatterers, they tend to exhibit negligible correlations [26, 27]. In this case it is reasonable to assume that the $\{\alpha_{k,l}\}_{l=1}^{L_p}$ are statistically independent random variables (RV's). We denote the fading amplitude of the k^{th}-user l^{th} resolved path by $\alpha_{k,l}$, which is a RV whose mean-square value $\alpha_{k,l}^2$ is assumed to be independent of k and is denoted by Ω_l.

After passing through the fading channel, the signal is perturbed by additive white Gaussian noise (AWGN) with a one-sided power spectral density which is denoted by $N_0\left(\frac{W}{Hz}\right)$. The AWGN is assumed to be independent of the fading amplitudes $\{\alpha_{k,l}\}_{l=1}^{L_p}$. Hence, the instantaneous SNR per bit of the l^{th} channel is given by $\gamma_{k,l} = {\alpha_{k,l}^2 E_b}/{N_0}$, where E_b is the energy per bit, and the average SNR per bit of the l^{th} channel is given by $\bar{\gamma}_l = {\Omega_l E_b}/{N_0}$

Another factor influencing fading is the Doppler Shift. The time variations in the channel are evidenced as Doppler broadening which is Doppler shift of a spectral line. Doppler spread is used to quantify the signal fading due to Doppler shift.

Fading in the cellular environment which results from multipath propagation of the transmitted signal can be divided in two types, large-scale fading and small-scale fading.

Large-scale fading which represents the average signal power attenuation or path loss due to motion over large areas. On the other hand *small-scale fading* refers to the dramatic changes in signal amplitude and phase that can be experienced as a result of small changes in spatial separation between receiver and transmitter [5] [21]. Small-scale fading also known as Rayleigh fading because if there are multiple reflective paths with no line-of-sight signal component, the envelope of the received signal is statistically described by Rayleigh probability density function (pdf) which can be expressed as [21]:

$$f_G(g) = \frac{g}{\sigma^2} e^{\left(-g^2/2\sigma^2\right)} \qquad for\ g \geq 0 \qquad (3.10)$$

where $\sigma^2 = E[GG^*]$ is the variance of the Gaussian process. The phase pdf has a uniform distribution [7][5]:

$$f_{\theta(\theta)} = \frac{1}{2\pi}, \qquad 0 \leq \theta \leq 2\pi \tag{3.11}$$

Propagation in free space is the ideal case. However, when propagation takes place close to obstacles, the following propagation mechanisms occur [5]:

- Reflection
- Diffraction and
- Scattering

3.5 MAI Analysis

Let us focus our attention on the detection of a DSSS signal in the case of a multiuser CDMA system in which K users are concurrently active. We will start considering the case a CDMA multiuser communication in which all of the spreading sequences of the different users are synchronous. We will refer to this arrangement as Synchronous CDMA (S-CDMA). This is the case of a CDMA signal originating from a single transmitter, i.e., from a base station (or satellite) to a group of mobile receivers. We will therefore address such a scenario as single-cell. The received signal, after baseband conversion and under the hypothesis of perfect carrier recovery, can be written as

$$r(t) = \sum_{l=0}^{L-1} \sum_{i=1}^{K} \alpha_{li} \sqrt{2P_i}\, b_i(t-\tau_{li}) a_i(t-\tau_{li}) + n(t), \tag{3.12}$$

For simplicity of analysis we have assumed that the channel is additive white Gaussian noise. And the above equation reduces to:

$$r(t) = \sum_{i=1}^{K} \sqrt{2P_i}\, b_i(t) a_i(t) + n(t) \tag{3.15}$$

Notice that, in order to take the multiple users accessing the radio frequency spectrum into account we have introduced the subscript i which identifies the power, data, and code chips of the generic i^{th} user.

Assuming now, without loss of generality, that the receiver intends to detect the data transmitted by user k, we can re-write the above equation as

$$\boldsymbol{r}(t) = \sqrt{2P_k}b(t)a_k(t) + \sum_{\substack{i=1 \\ i\neq k}}^{K} \sqrt{2P_i}b_i(t)a_i(t) + n(t) \tag{3.14}$$

where the first term on the right hand side is the useful signal to be detected, while the second one, denoted in a more compact form as $I(t)$, which is given by equation (3.16), is an additional component due to multiple access interference.

$$I(t) = \sum_{\substack{i=1 \\ i\neq k}}^{K} \sqrt{2P_i}b_i(t)a_i(t) \tag{3.15}$$

In a conventional correlation receiver, the received signal in (3.15), is passed through the chip matched filter and sampled at chip rate yielding the samples $y_{m,l}$.

$$y_{m,l} = \sqrt{2P_k}b_k(m)a_k(l)a_k(l) + \sum_{\substack{i=1 \\ i\neq k}}^{K} \sqrt{2P_i}b_i(m)a_i(l)a_k(l) + V_{m,l} \tag{3.16}$$

where $\quad I_{m,l} = \sum_{\substack{i=1 \\ i\neq k}}^{K} \sqrt{2P_i}b_i(m)a_i(l)\, a_k(l) \qquad (3.17)$

$$and \qquad V_{m,l} = a_k(l)n_{m,l}$$

the variance of $I_{m,l}$ is

$$\sigma_{I_{m,l}} = E\{I_{m,l}^2\}. \tag{3.18}$$

The sampled signal in (3.16) averaged over N to give the m^{th} bit of user k is given as follow:

$$z_{k,m} = \frac{1}{N}\sum_{l=1}^{N} y_{m,l} \tag{3.19}$$

After some algebra we find

$$z_{k,m} = \frac{\sqrt{2P_k}b_k(m)}{N}\sum_{l=1}^{N} a_k(l)a_k(l) + \sum_{\substack{i=1\\ i\neq k}}^{K}\frac{\sqrt{2P_i}b_k(m)}{N}\sum_{l=1}^{N} a_i(l)a_k(l) + V_m \tag{3.20}$$

We define the following partial auto-correlation and cross-correlations as:

$$x_{a_k a_k} = \frac{1}{N}\sum_{l=1}^{N} a_k(l)a_k(l) = 1, \tag{3.21}$$

$$x_{a_k a_i} = \frac{1}{N}\sum_{l=1}^{N} a_k(l)a_i(l), \tag{3.22}$$

The despreaded signal is passed to the final detector which regenerates the transmitted digital data stream of the user k (the desired user signal). From (3.20) it is apparent that the despreaded data $z_{k,m}$ is composed of three terms: i) the useful datum (1^{st} term); ii) Gaussian noise (third term) and iii) an additional term arising from the concurrent presence of multiple users and called Multiple Access Interference (MAI). In particular, the MAI term can be expressed as [9]:

$$I_m = \sum_{\substack{i=1\\ i\neq k}}^{K} \sqrt{2P_i}b_i(m)x_{a_k a_i}, \tag{3.23}$$

$$\sigma_{I_m}^2 = E\{I_m^2\} = \frac{I_0}{T_s} \tag{3.24}$$

We introduce an equivalent PSD I_0 for the MAI term, assuming implicitly that it can be considered flat (white) over the whole signal spectrum. Now rewriting (3.20) into the form

$$z_k(m) = \sqrt{2P_k} b_k(m) + I_m + V_m \qquad (3.25)$$

Under certain hypotheses, which we will discuss in a little while, the MAI contribution can be modeled as an additional (white) Gaussian noise (independent of V_m). This assumption holds true only when the number of interfering users or the spreading gain is high. Therefore, the BER performance of the DSSS signal can be analytically derived simply by assuming an equivalent noise term $V'_m = I_m + V_m$ with a total, equivalent PSD given by

$$N'_0 = N_0 + I_0, \qquad (3.26)$$

and the decision strobe becomes equivalent to a single user within an AWGN channel. Then equation becomes

$$z_k(m) = \sqrt{2P_k} b_k(m) + V'_m \qquad (3.27)$$

Consequently the BER for BPSK modulation in the presence of Gaussian MAI, can be obtained by [5]

$$P(e) = Q\left(\sqrt{\frac{2E_b}{N'_0}}\right) = Q\left(\sqrt{\frac{2E_b}{N_0 + I_0}}\right) \qquad (3.28)$$

If very long pseudo-random (i.e., noise-like) spreading sequences are used then the chips $a_i(l)$ of each user code can be approximately modeled as independent random variables belonging to the alphabet $\{-1, +1\}$. Also, the chips of different users can be modeled as uncorrelated random variables. It follows that if $K \gg 1$ and if all of the signal powers are (almost) equal $(i.e., P_i = P$, $\forall i)$, then the power of the MAI is $P_{MAI} = (K - 1)P$ and the variance of the MAI is [9]

$$\sigma_I^2 = \frac{I_0}{T_s} = \frac{P_{MAI} / \left(1/T_c \right)}{NT_c} = \frac{(K-1)P}{N} \tag{3.29}$$

where P_{MAI} is the power of the interfering signals.

This situation is actually experienced, for instance, in a CDMA system with accurate power control, so that all the users' signals are received at (almost) the same power level. Under this hypothesis the PSD of the MAI is

$$I_0 = T_s \frac{(K-1)P}{N} = (K-1)PT_c = (K-1)E_c = \frac{(K-1)}{N} E_b, \tag{3.30}$$

Where $E_c = PT_c$ represents the average energy of the signal per chip, and as given in [9] we have set $E_c = {E_b}/{N}$. Then Equation (3.28) becomes

$$P(e) = Q\left(\sqrt{\frac{2E_b}{N_0 + (K-1)E_c}}\right) \tag{3.31}$$

After some manipulation we got for BPSK

$$P(e) = Q\left(\sqrt{\frac{2E_b}{N_0}} \cdot \frac{1}{\sqrt{1 + \frac{(K-1)}{N}\frac{E_b}{N_0}}}\right) \tag{3.32}$$

From the expressions above it turns out that the MAI degrades the BER performance. In particular, the degradation increases with the number of interfering users and decreases for large processing gains. Note also that, in the particular case $K = 1$ equation (3.32) reduces to the conventional BER expression relevant to single user BPSK modulation over AWGN channel and matched filter detection.

3.6 Conventional Receiver

The detection in the 2G system, namely, IS-95, is based on correlators (matched filtering). This detector is also known as the conventional detector. At the base station it consists of a bank of K correlators, as shown in Figure 3.5. Each user is treated separately in such a detector. The received signal is correlated with the desired user's PN code and the output is sampled at the bit rate, which yields soft estimates of the transmitted data. The final hard data (±1) decisions are made according to the signs of the soft output [28] [29].

It is clear from Figure 3.5 that the conventional detector follows a single user detector strategy. Each branch detects one user without considering the other existing users. This means that the conventional detector treats MAI as noise. Thus, the performance and capacity of the system is highly dependent on the number of the active users in the system since there is no sharing of multiuser information.

Assume there are K users in a synchronous CDMA system. The baseband received signal was given in Equation (3.13).

The received signal will be correlated to the corresponding users' PN codes as shown in Figure 3.5 If N is the processing gain defined in Chapter 2, then the output of the k^{th} correlator is given by:

$$z_k = \sum_{l=1}^{N} r(l) a_k(l) \tag{3.33}$$

$$z_k = \sum_{i=1}^{K} \sqrt{2P_i} b_i \left(\sum_{l=1}^{N} a_i(l) a_k(l) \right) + \sum_{l=1}^{N} n(l) a_k(l) \tag{3.34}$$

$$= \sum_{i=1}^{K} \sqrt{2P_i} b_i \rho_{ik} + n_k \tag{3.35}$$

where, $n_k = \sum_{l=1}^{N} n(l) a_k(l)$ (3.36)

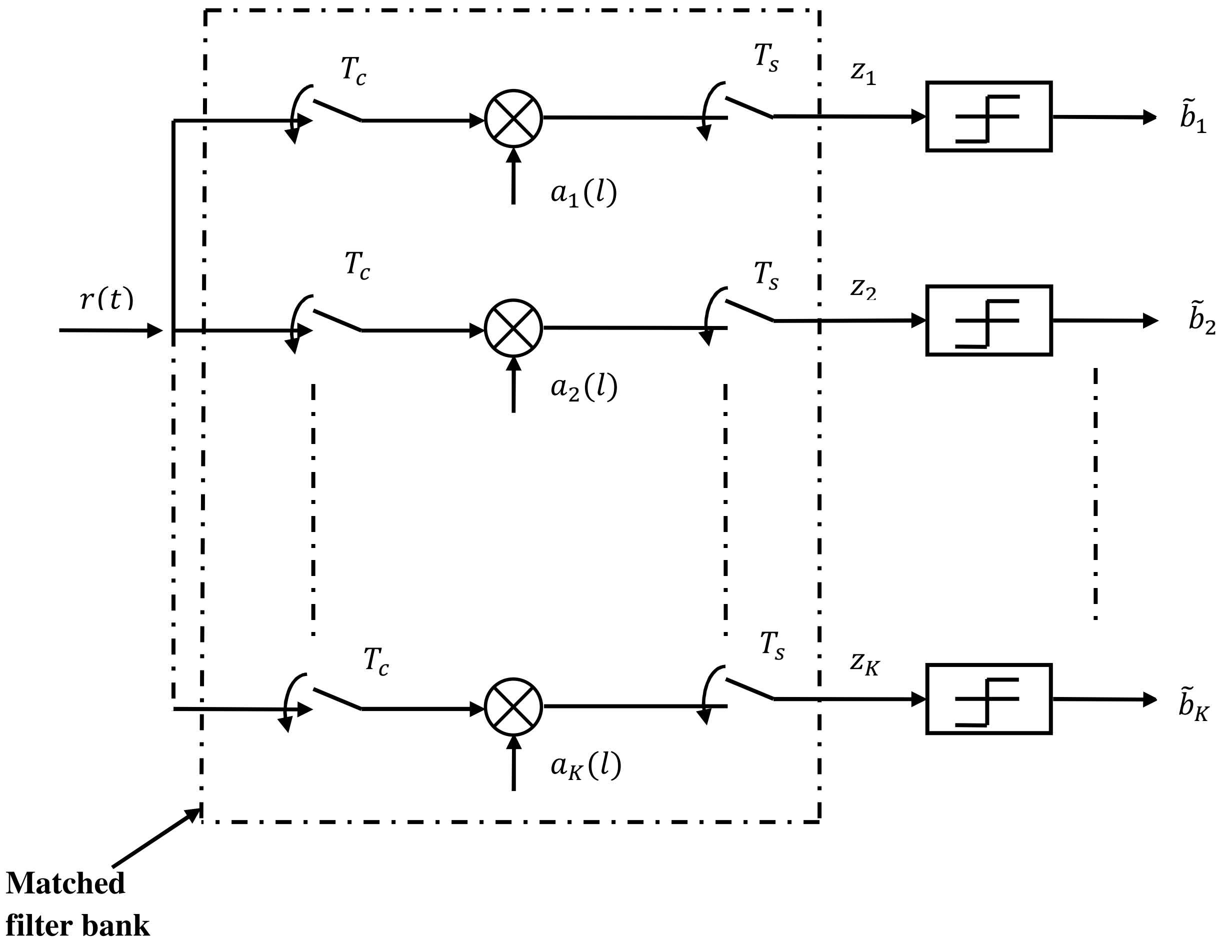


Figure 3.5 the conventional detector

and ρ_{ik} is the normalized cross-correlation of the signature waveforms defined as:

$$\rho_{ik} = \langle a_i, a_k \rangle = \frac{1}{N} \sum_{l=1}^{N} a_i(l) a_k(l) \tag{3.37}$$

From (3.35) the output of the k^{th}user's correlator for a particular bit interval is:

$$z_k = \sqrt{2P_k}b_k + \sum_{\substack{i=1 \\ i \neq k}}^{K} \sqrt{2P_i}b_i\rho_{ik} + n_k \qquad (3.38)$$

$$\sqrt{2P_k}b_k + MAI_k + n_k \qquad (3.39)$$

It is clear form (3.38) that the correlation with the k^{th} user itself gives rise to the recovered data (1st term), correlation with all other active users gives rise to MAI (2nd term) and the correlation with the thermal noise yields to the noise term (3rd term).

In order to reduce the effect of MAI on user k, usually PN codes are generally designed to have very low cross correlation relative to autocorrelation (i.e. ρ_{ik}<< 1) [30]. As mentioned earlier, the conventional detector depends on the number of active users in which MAI has a significant impact on the capacity and performance of conventional DS-CDMA system. As the number of interfering users increases, the amount of MAI increases.

Another problem that causes the degradation in the performance of the conventional detector is the near-far effect. This is the case where the users' signals arrive at the receiver at different power levels due to the different geographical locations of the transmitters relative to the receiver. In such a case, weaker users may be overwhelmed by stronger users. As illustrated in Figure 3.6, although the cross-correlation between the codes of 1st and 2nd users is low, the interfering user is close to the receiver in which its signal is received at higher power level compared to the intended users. This might affect the proper data detection of the intended user. Since the signal power of the intended user overwhelmed by the nearby user it causes problem during detection of the desired user signal.

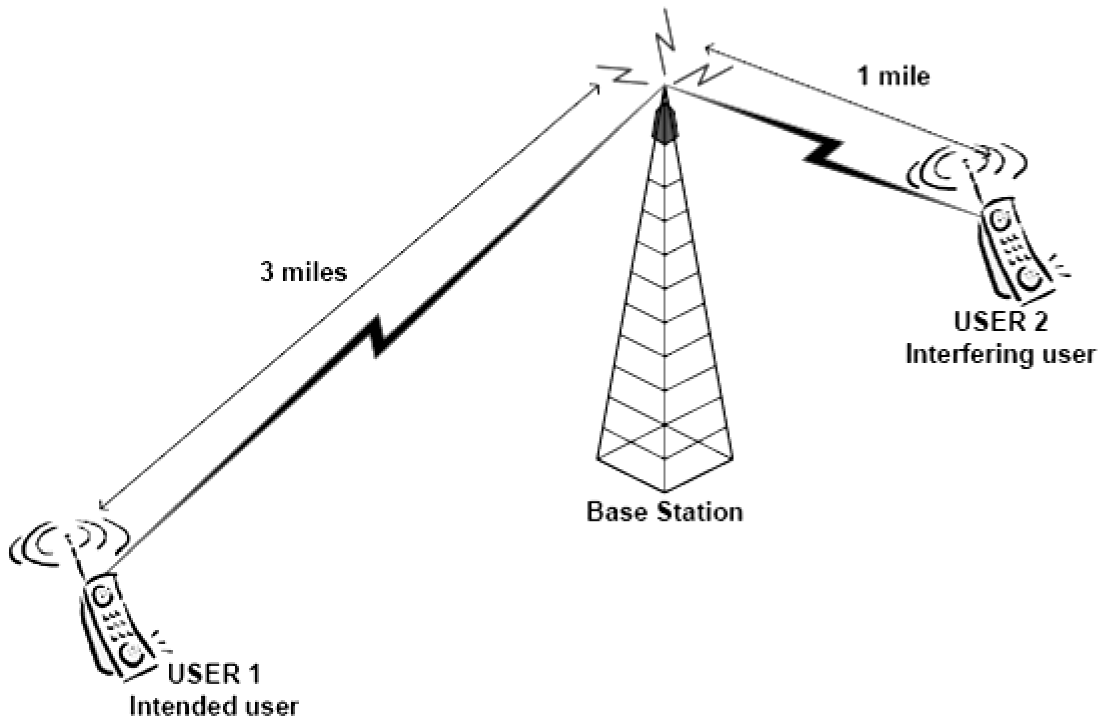


Figure 3.6 the near-far scenario

To mitigate the near-far problem, power control is employed in the 2G CDMA systems, e.g. IS-95 system. Ideal power control ensures all users have identical signal power level when they arrive at the receiver.

3.7 Decorrelating Detector Receiver

Although the conventional detector is easy and simple to implement, it suffers from the MAI and near-far problem which affect the performance of the detector. As discussed in the previous Section, the conventional detector has no information of multiple users which can be jointly used to better detect each individual user. Due to this lack of multiuser information, there has been great interest in enhancing DS-CDMA detection through the use of multiuser detectors.

Verdu shows in that the optimal MU detection can be achieved by the maximum likelihood sequence detector that uses the Viterbi algorithm [31]. The optimal detector shows a huge performance and capacity enhancement over the conventional detector. However, its complexity grows exponentially with the number of active users. Due to this problem, the optimal detector is too complex for practical DS-CDMA systems. Therefore, most of the research has focused on finding suboptimal low complexity detectors solutions that are more feasible to implement.

The decorrelator detector is a linear multiuser detector where a linear transformation is applied to the soft output of the conventional detector in order to produce a new set of decisions with MAI completely decoupled. It offers many desirable features, such as: it yields an optimal value of near-far resistance performance and does not need to estimate the received signal amplitude.

We can define a cross-correlation matrix **R** as,

$$\boldsymbol{R} = \begin{bmatrix} \rho_{11} & \rho_{12} & \cdots & \rho_{1K} \\ \rho_{21} & \rho_{22} & \cdots & \rho_{2K} \\ \vdots & \vdots & \ddots & \vdots \\ \rho_{K1} & \rho_{K2} & \cdots & \rho_{KK} \end{bmatrix} \tag{3.40}$$

the matrix **R** has the following properties

- It is symmetric,
- The diagonal elements are equal to 1 (normalized),
- In general, the matrix is non-negative definite.

The receiver in case of DS-CDMA consists of a bank of matched filters.

Equation (3.36) can be set into a matrix form,

$$z_k = \boldsymbol{r}_k \sqrt{\boldsymbol{2Pb}} + n_k \tag{3.41}$$

Where:

- $\boldsymbol{r_k} = [\rho_{k1},\ \ \rho_{k2},\ \cdots,\ \rho_{kK}]$, the cross-correlation vector of the k^{th} user with all other users.
- $\boldsymbol{P} = diag(P_1,\ \ \cdots,\ \ \ P_K)$, the matrix of the received signal power
- $\boldsymbol{b} = [b_1,\ \ \cdots,\ \ b_K]^T$, the vector of the transmitted bits

If the outputs of all users are considered, the above equation can be written as,

$$\begin{bmatrix} z_1 \\ z_2 \\ \vdots \\ z_K \end{bmatrix} = \begin{bmatrix} \rho_{11} & \rho_{12} & \cdots & \rho_{1K} \\ \rho_{21} & \rho_{22} & \cdots & \rho_{2K} \\ \vdots & \vdots & \ddots & \vdots \\ \rho_{K1} & \rho_{K2} & \cdots & \rho_{KK} \end{bmatrix} \begin{bmatrix} \sqrt{2P_1} & 0 & \cdots & 0 \\ 0 & \sqrt{2P_2} & \cdots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \cdots & \sqrt{2P_K} \end{bmatrix} \begin{bmatrix} b_1 \\ b_2 \\ \vdots \\ b_K \end{bmatrix} + \begin{bmatrix} n_1 \\ n_2 \\ \vdots \\ n \end{bmatrix} \tag{3.42}$$

In a compact matrix notation, the previous equation can be expressed as,

$$\boldsymbol{z} = \boldsymbol{R}\sqrt{\boldsymbol{2P}}\boldsymbol{b} + \boldsymbol{n} \tag{3.43}$$

The decorrelator detector (shown in Figure 3.7) applies the inverse of the correlation matrix **R,** that is $\boldsymbol{R}^{-1}$**,** to the output of the conventional detector in order to decouple the data. From (the above equation), the soft estimate of the detector is:

$$\widehat{\boldsymbol{b}} = \boldsymbol{R}^{-1}\boldsymbol{z} = \sqrt{\boldsymbol{2P}}\boldsymbol{b} + \boldsymbol{R}^{-\boldsymbol{1}}n \tag{3.44}$$

Thus, the decorrelator detector completely eliminates the MAI at the expense of noise enhancement at the output of decorrelator detector as it can be seen in (3.43).

Another disadvantage of the decorrelator detector is that the computations needed to compute the inverse of the matrix **R** are difficult to perform in real time, especially for asynchronous CDMA.

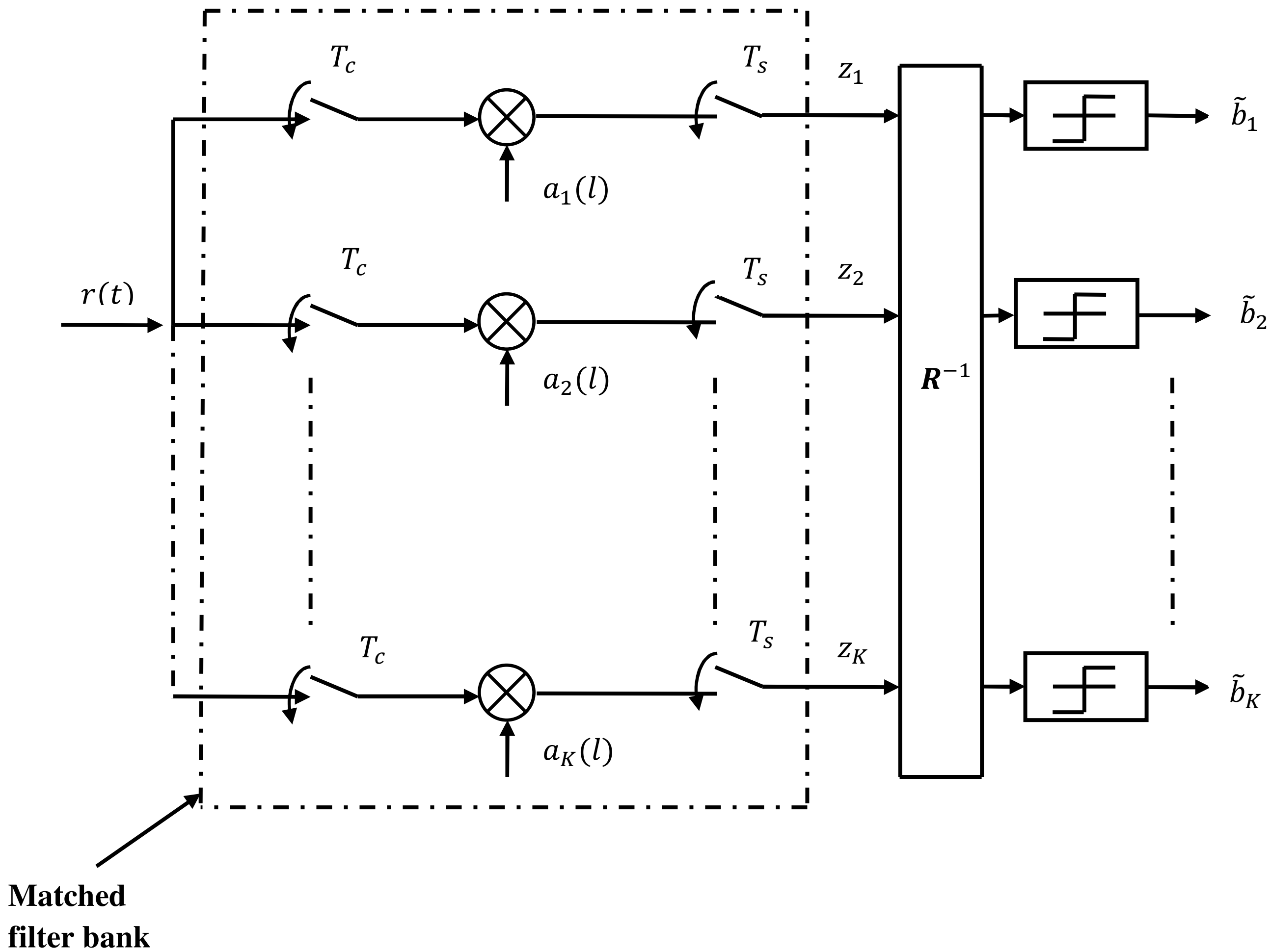


Figure 3.7 Decorrelting detector

Chapter Four

4. Adaptive Algorithms for CDMA receiver

4.1 Introductions

Adaptive algorithm receivers for MAI suppression have a role analogous to equalizers in single-user applications used to mitigate ISI. The underlying difference is that for adaptive equalization applications, the equalizer can only have one tap per symbol whereas for interference suppression applications, the receiver must have multiple taps per symbol, but can span only one symbol [1]. Multiple-symbol spanning gives the MAI suppressing receiver an ability to suppress ISI. Adaptive algorithm receivers do not require any side information other than the information required by the conventional matched filter receiver. One of the most important parts of the adaptive receiver scheme is the adaptive algorithm, mainly because it is responsible with the convergence rate of the system. The adaptive algorithms can be divided into two major categories. The first one contains the algorithms based on training sequences for adaptation, whose representative member is the Least Mean Square (LMS) algorithm. The second category is blind adaptive algorithms that do not require any training sequence for its adaptation but needs appropriate initialization of tap weights, whose representative member is constant modulus algorithm (CMA).

4.2 LMS Adaptive Algorithm

The LMS algorithm was presented by Widrow and Hoff in 1960 [32]. The LMS algorithm is used in suppressing MAI so that the tap weight coefficients are obtained to minimize the MSE.

In our analysis of LMS algorithm we assumed that $r(n)$ is the received signal, $b(n)$ is the desired signal, $y_{est}(n)$ is the estimated signal and $w(m)$ is the update tap weight. The error between the desired signal and the estimated signal is given as [32]:

$$e(n) = b(n) - y_{est}(n), \qquad (4.1)$$

Now, our aim is to achieve the MSE criterion, which is the minimization of the output mean square error and is defined as:

$$J(n) = E[e(n)^2] \tag{4.2}$$

Where E [ ] is the statistical expectation.

The gradient of the MSE $J(n)$ with respect to the k^{th} tap weights w_k is [32].

$$\frac{\partial J(n)}{\partial w_k} = 2E\left[e(n)\frac{\partial e(n)}{\partial w_k}\right] \tag{4.3}$$

$$= -2E\left[e(n)\frac{\partial y_{est}(n)}{\partial w_k}\right] \tag{4.4}$$

$$= -2E[e(n)r(n-k)] \tag{4.5}$$

The expectation $E[e(n)\, r(n-k)]$ is the cross-correlation between the error signal $e(m)$ and the input signal $r(n)$ for a lag of k samples. Therefore,

$$R_{er}(k) = E[e(n)r(n-k)] \tag{4.6}$$

Now, to have a minimum MSE, the following optimum condition should be satisfied:

$$\frac{\partial J(n)}{\partial w_k} \approx 0 \qquad for\ k = 0,1,2,\cdots,N-1 \tag{4.7}$$

$$or$$

$$R_{er}(k) \approx 0 \tag{4.8}$$

The result in Equation (4.8) is known as the principle of orthogonality [8].

The MSE performance can be visualized as a parabolic surface like bowl shape. The surface is a function of the tap weights. The process of adjusting these taps is like seeking the bottom of the bowl where the MSE attains its minimum value. This is the basic idea of the steepest decent algorithm [34] which can be described mathematically by the following recursive formula:

$$w_k(n+1) = w_k(n) - \frac{1}{2}\mu\frac{\partial J(n)}{\partial w_k},\ for\ k = 0, 1, 2, \cdots, N-1 \qquad (4.9)$$

Where μ is a small positive constant called the step-size parameter.

Since the use of the steepest decent algorithm requires exact knowledge of the cross correlation function $R_{er}(k)$, (between the desired signal and the input data vector) and as this parameter is not available, we can use instead the instantaneous estimate for $R_{er}(k)$. The following estimate may be used [8]:

$$\hat{R}_{er}(k) = e(n)r(n-k),\quad for\ k = 0, 1, 2, \cdots, N-1 \qquad (4.10)$$

Using Equations (4.6), (4.9) and (4.10) we get the following recursive formula for updating the tap weights of the equalizer:

$$w_k(n+1) = w_k(n) + \mu e(n)r(n-k),\ for\ k = 0, 1, 2, \cdots, N-1 \quad (4.11)$$

This algorithm is known as the least mean-square (LMS) algorithm where $w_k(n+1)$ is the current value of the k^{th} tap weights, $w_k(n)$ is the previous value of the k^{th} tap weights and $\mu e(n)r(n-k)$ is the correction applied to compute the updated value (current value $w_k(n+1)$).

4.2 LMS Adaptive Algorithm for CDMA Receiver

The LMS algorithm with its simple implementation suffers from slow convergence, which implies long training overhead with low system throughput. Nevertheless, the LMS algorithm is still preferred in practical implementations of adaptive algorithm receivers. A solution for increasing its convergence rate is to adjust the filter tap weights iteratively several times every transmitted bit interval. A drawback of these approaches is that they are time consumers. During a bit interval, every iteration has to "wait" the result from the previous one, which is the natural function mode of every iterative process. Anyway, from the time processing reason, we have used a single formula instead of those multiple iterations and use more number of training sequences instead.

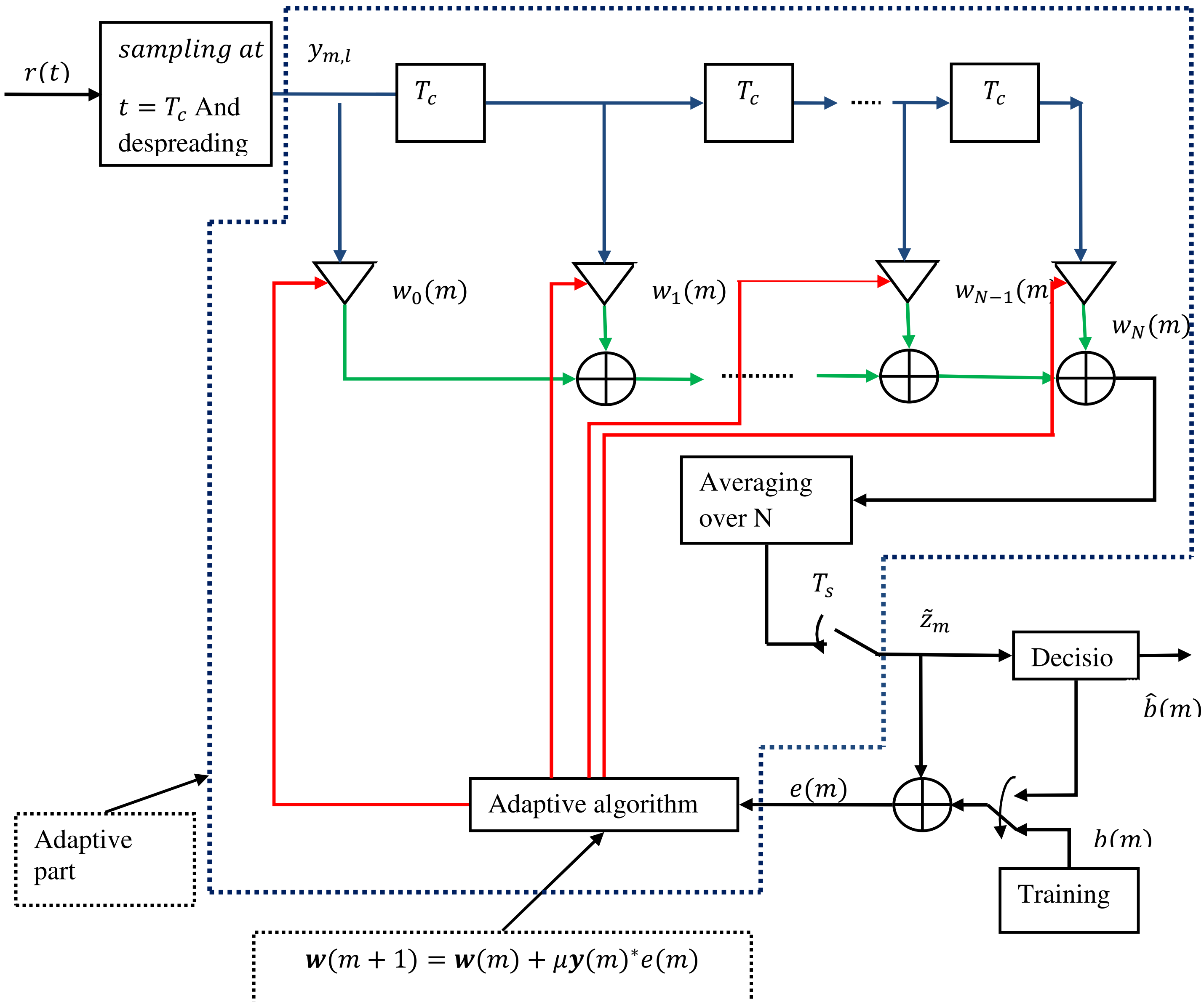


Figure 4.1 Adaptive LMS receiver for DS-CDMA

A block diagram of the adaptive least mean square (LMS) receiver structure is depicted in Fig. 4.1. After converting the received signal to its baseband form using a down converter, the received signal is given by Equation (3.12).

Since the LMS algorithm assumes knowledge of the transmission bits $b_k[m]$, it must be implemented as a training mode or decision-directed mode where by the estimate of the transmitted data bit is obtained from the decision device as shown in Figure 4.1. Let us consider the following vectors:

$$\boldsymbol{w}(m) = [w_0(m), w_1(m), \cdots, w_{N-1}(m)]^T \qquad (4.12)$$

$$\boldsymbol{y}(m) = \left[y_{m,0,}\ y_{m,1}, \cdots, y_{m,N-1}\right]^T \qquad (4.13)$$

with $y_{m,l}$ given by Equation (3.17) and N is the number of spreading gain. The output signal $\tilde{z}(m)$ (given by Equation (3.26) but here in the estimated version) will be an estimate of $b(m)$. For the estimation of $\boldsymbol{w}(m)$ a stochastic gradient approach based on LMS adaptive algorithm is used.

The output signal is $\tilde{z}(m) = \frac{\boldsymbol{1}}{\boldsymbol{N}}\boldsymbol{w}^T(m)\boldsymbol{y}(m)$ (4.14)

The receiver forms an error signal

$$e(m) = b(m) - \tilde{z}(m) \qquad (4.15)$$ and

A new filter tap weight vector is estimated according to

$$\boldsymbol{w}(m+1) = \boldsymbol{w}(m) + \mu\boldsymbol{y}(m)e(m) \qquad (4.16).$$

The parameter μ represents the adaptation step size of the algorithm, chosen to optimize both the convergence rate and the mean squared error.

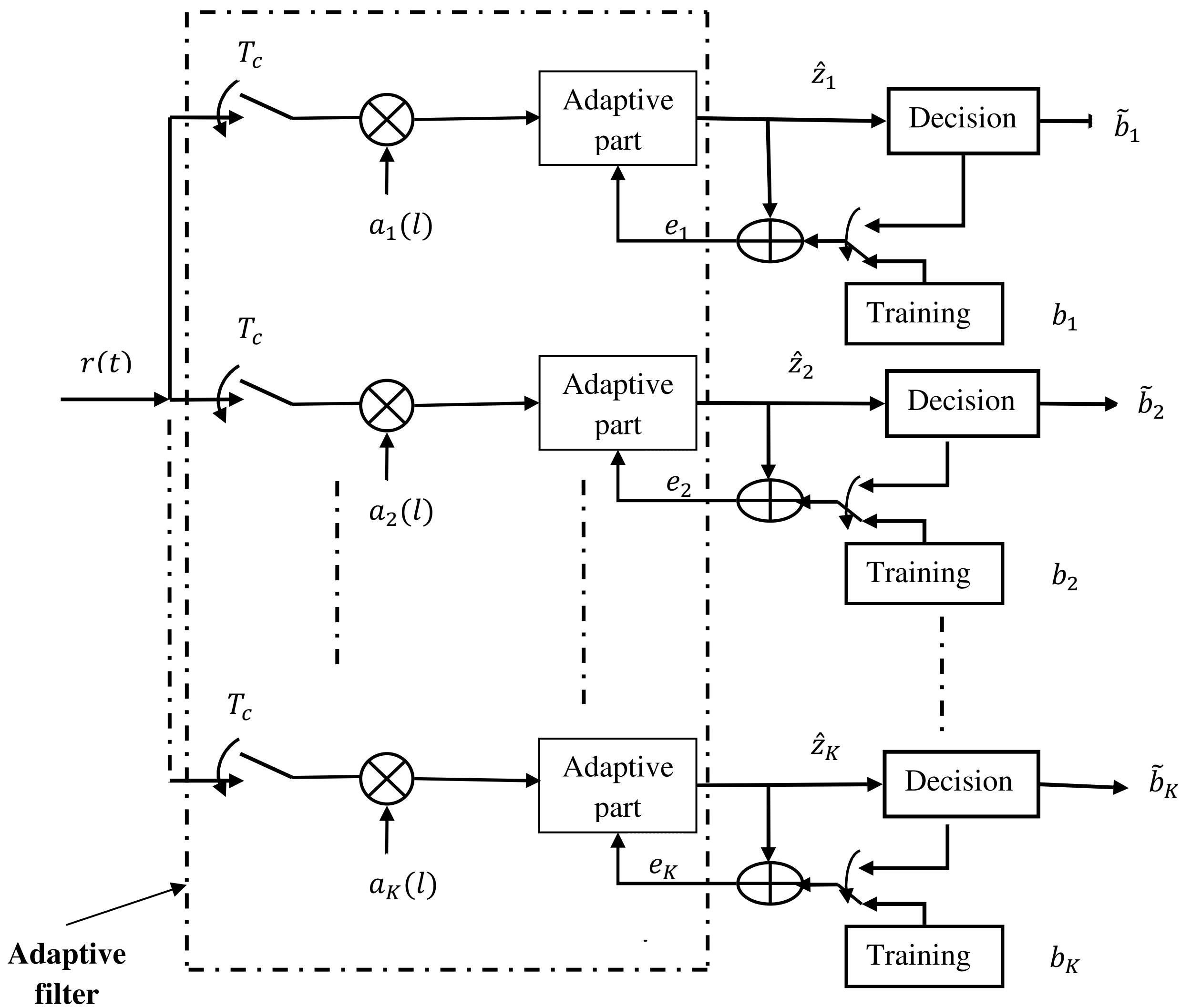


Figure 4.2 adaptive LMS multi-user receiver in DS-CDMA

In the above Figure it shows the schematic of multiuser CDMA receiver where the block i.e., *Adaptive part* is given in Figure 4.1.

While simplicity in implementation, low computational complexity and robustness to noise are major advantages of the LMS algorithm, there are some aspects one must be aware of regarding convergence speed of the algorithm. Large μ corresponds to faster convergence speed but is associated with greater excess MSE due to an increased amount of random coefficient fluctuations about the mean [32]. The choice of initialization also affects the convergence speed of the LMS algorithm as discussed in [1] and [33]. In the training mode, the receiver attempts to cancel the MAI and adapts its coefficients using a short training sequence employing an adaptive algorithm. After training is acquired, the receiver switches to the decision directed mode and continues to adapt and track channel variations.

4.2.1 Convergence and Stability of LMS Algorithms

The mean square error $\mathcal{E}(n)$ converges to a steady-state value denoted by $\mathcal{E}(\infty)$, if and only if the step size parameter μ satisfies the following two conditions [32].

Condition 1: $0 < \mu < \frac{2}{\lambda_{max}}$

Condition 2: $\sum_{i=1}^{2M+1} \frac{\mu\lambda_i}{2-\mu\lambda_i} < 1$

Where λ_i are the eigenvalues of the input signal correlation matrix **R** and $2M + 1$ is the number of filter coefficients; λ_{max} is the largest eigenvalue of $\boldsymbol{R}$. Where these two conditions are satisfied LMS is convergent in the mean square. If condition 1 is satisfied, then condition 2 can be replaced by a weaker form:

$$\sum_{i=1}^{2M+1} \frac{\mu\lambda_i}{2} < \sum_{i=1}^{2M+1} \frac{\mu\lambda_i}{2-\mu\lambda_i} < 1 \; \dots\dots\dots\dots\dots \textit{this can be written as}$$

$\mu < \frac{2}{\sum_{i=1}^{2M+1} \lambda_i} = \frac{2}{P_t}$, where P_t is the total input power

This shows that the convergence speed of the adaptive algorithm is limited by the total received signal power, including MAI. An additional convergence speed reduction occurs when the number of users becomes large, due to increased correlation between users signatures. This is even true in the systems with strict power control.

The steady-state mean-squared error in the LMS algorithm is given by:

$\mathcal{E}(\infty) = \frac{\mathcal{E}_{opt}}{1-\sum_{i=1}^{2M+1}\frac{\mu\lambda_i}{(2-\mu\lambda_i)}}$, this shows that the steady state error is proportional to the number of interferers.

4.3 Constant Modulus Algorithm (CMA)

To rid the receiver from the cumbersome reliance on training, blind adaptive receivers based on CMA are developed here. The CM criterion penalizes deviations in the modulus (i.e., magnitude) of the recovered signal away from a fixed value; the dispersion constant.

In searching for a self-recovering equalization algorithm for a multipoint network, Godard [34] developed the carrier-phase independent cost function based only on the output signal modulus $|z_m|$. In multiuser applications, the Godard cost function for the desired user (which is assumed to be the desired k^{th} user) is defined as [34]

$$J_p\left(w^{(k)}\right) \triangleq \frac{1}{2p} E\left\{\left(\left|z_n^{(k)}\right|^p - \gamma_p\right)^2\right\}, \tag{4.17}$$

where: $\gamma_p = \frac{E\{|b_k[n]|^{2p}\}}{E\{|b_k[n]|^p\}}$ represents the dispersion constant for the k^{th} user.

For ideal situations, minimizing the Godard cost function results in perfect equalization, i.e., the desired user signal perturbed by noise and MAI can be recovered. The corresponding stochastic-gradient LMS-like adaptation of the tap-weight vector that approximately minimizes the Godard cost is given by

$$w_{n+1}^{(k)} = w_n^{(k)} - \mu r_n^* \phi\left(z_n^{(k)}\right), \tag{4.18}$$

Where $(.)^*$ denotes complex conjugate, μ is the small positive step-size, r_n is the input signal and $\phi\left(z_n^{(k)}\right)$ denotes the prediction error function [35] of the k^{th} user and is given by

$$\phi\left(z_n^{(k)}\right) \triangleq \left(\left|z_n^{(k)}\right|^p - \gamma_p\right)\left|z_n^{(k)}\right|^{p-2} z_n^{(k)}. \qquad (4.19)$$

In the special case of p=2, the algorithm which attempts to minimize the cost function (4.17) is well known as the constant modulus algorithm (CMA) as was developed by Treichler and Agee [36]. The constant modulus (CM) criterion penalizes the deviation of the squared output modulus $\left|z_n^{(k)}\right|^2$ from the dispersion constant $\gamma_2 = \frac{E\{|b_k[n]|^4\}}{\sigma_b^2}$ and is defined as

$$J_{CM}\left(w^{(k)}\right) = \frac{1}{4} E\left\{\left(\left|z_n^{(k)}\right|^2 - \gamma_2\right)^2\right\}. \qquad (4.20)$$

And the error function will be $\phi\left(z_n^{(k)}\right) \triangleq \left(\left|z_n^{(k)}\right|^2 - \gamma_2\right) z_n^{(k)}$

The application of the constant modulus algorithm (CMA) to blind equalization has been widely studied to its low complexity yet robust and near wiener receiver performance [37, 38]. In the multiuser applications, however, without control of its convergence behavior, the CMA receiver may lock onto an interfering user rather than the desired user. This mis-convergence problem is aggravated when the interference is stronger than the desired user. Such a power disparity situation occurs due to the near-far problem.

4.2.1 Weight Initialization Strategy in CMA

Initialization is regarded as one of the most important aspects affecting the convergence of a CMA equalizer [37 39 40]. Unlike the MSE cost function (the cost function in LMS algorithm), the CM cost function is multimodal. Therefore apart from global minima, other stationary points of the CM cost may include local minima and saddle points. The trajectories of the CMA algorithm are generally attracted to a stationary point nearest to the initialization point. If the nearest stationary point is a local minimum, the receiver still retrieves the transmitted data but with potentially large steady-state MSE [37]. If the nearest stationary point is a saddle point, it has the effect of attracting the trajectories, resulting in spending a long time before converging to a global or a local minimum [41].

Single spike (double-spike in a fractionally-spaced (FS) CMA receiver) or centre-tap initialization is probably the most employed initialization strategy in single-user system [37]. A problem arises in using CMA for a multiuser application such as CDMA. Since the statistical properties of all users are identical, there is no control over the user to which the receiver will converge. In CDMA applications, however, the information contained in the spreading code of the desired user is available to the receiver and can be used as partial information for the detection algorithms. Although code –constrained methods have been applied within CMA receivers [42, 43, 44], it is revealed that the performance is still largely affected by the choice of initialization. The two mostly widely used initialization strategies are:

1. kurtosis based tap weight initialization strategy and
2. second-order signal-to-interference plus noise ratio(SINR) method

In this thesis we use kurtosis based initialization method for which the detailed analysis is given in [45].

This employs the second – and fourth-order moments of the input and output probability distributions to form a kurtosis measure which is maximized in the optimization process.

Assume that the data bit of any user $b[n]$ is a zero-mean independent identically distributed (i.i.d) random variable, the kurtosis of which can be defined as

$$kurt(b[n]) \triangleq E\{b^4[n]\} - 3E^2\{b^2[n]\} \qquad (4.21)$$

$$J_{SW}(\boldsymbol{w}_{init}) = -\frac{kurt(z[n])}{kurt(b[n])}(3m_2^2 - m_4) \qquad (4.22)$$

Where $m_4 \triangleq E\{b^4[n]\}$ and $m_2 \triangleq E\{b^2[n]\}$

4.2.2 Constant Modulus for CDMA Receiver

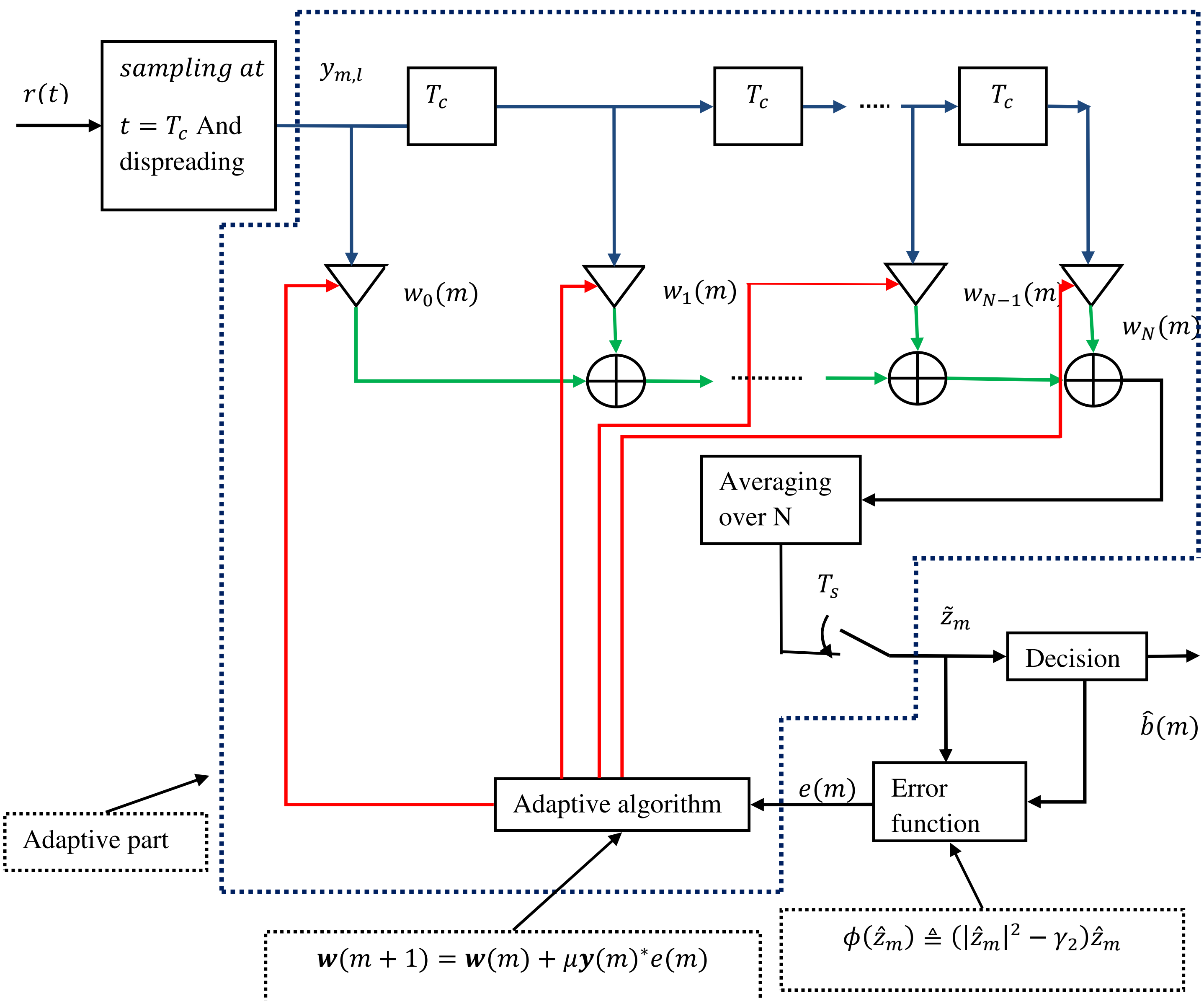


Figure 4.3 Adaptive CMA receiver for DS-CDMA

Where $\gamma_2 = \frac{E\{|\hat{b}_m[n]|^4\}}{\sigma_b^2}$ and $e(m) = \phi(\hat{z}_m)$ the other thing is similar to LMS receiver.

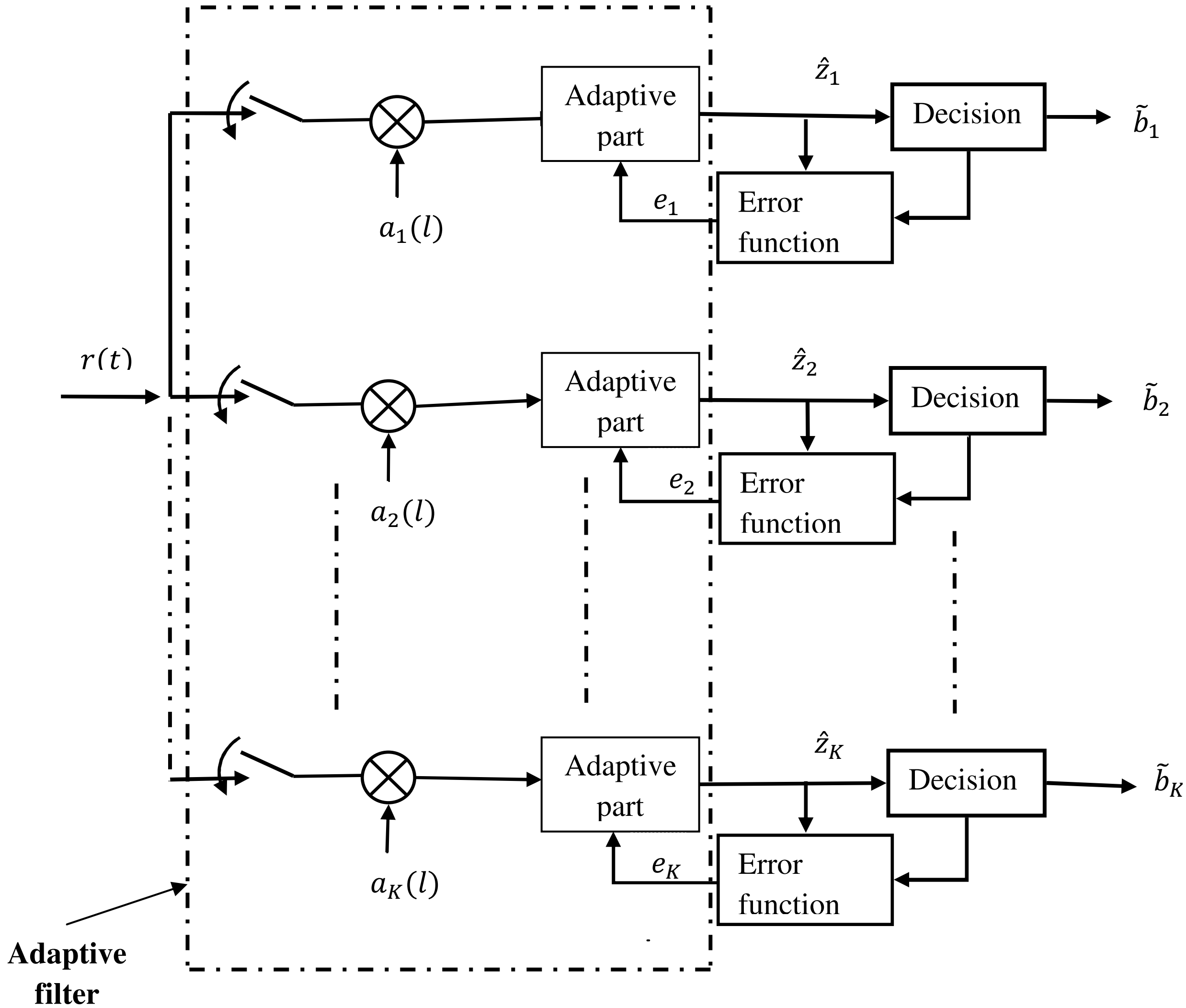


Figure 4.2 adaptive CMA multi-user receiver in DS-CDMA

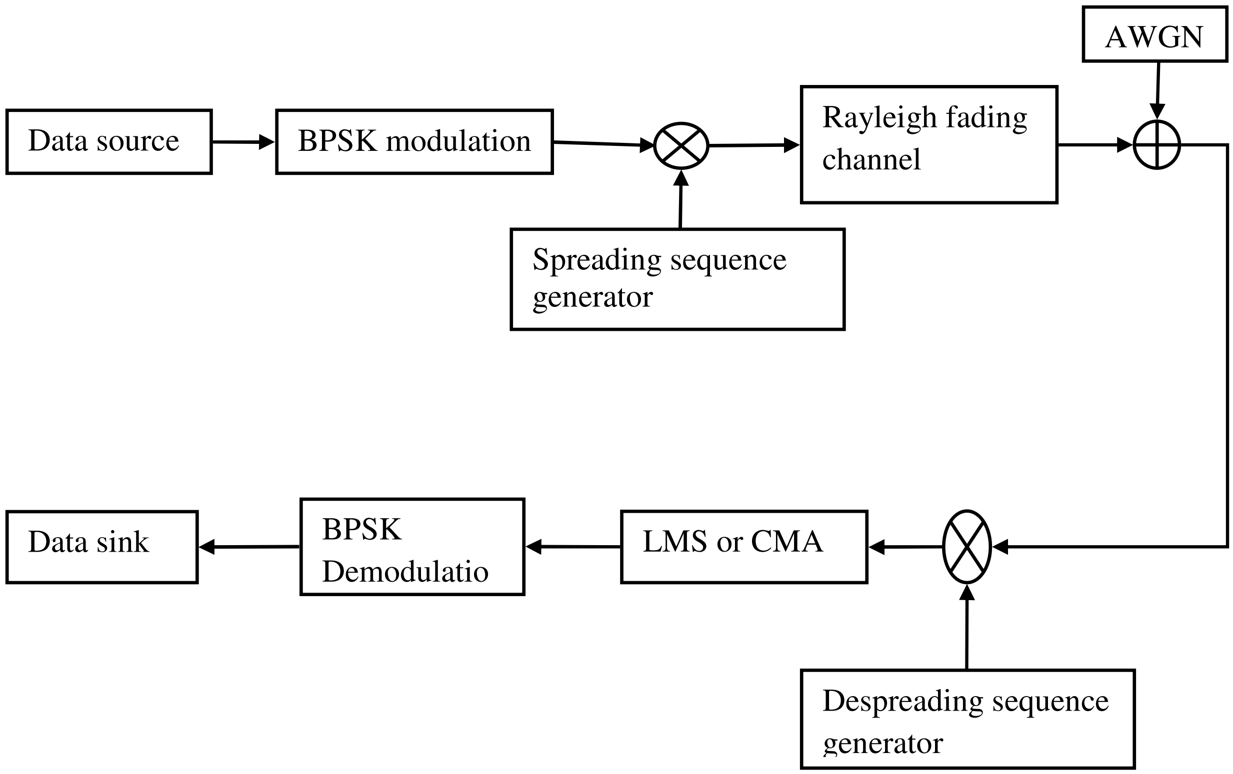
Chapter 5

5. Simulation and Analysis

5.1 Introduction

In this Chapter the simulation of different spreading code, conventional matched filtered CDMA receiver, decorrelating detector and adaptive algorithm receiver that are LMS algorithm and CMA are simulated and analyzed based on the result obtained. The performance of each spreading code, receiver type, and adaptive algorithm will be seen in accordance with their ability to reduce MAI being prime criteria for their comparison where as there complexity, bandwidth extension and practical applicability of each will be considered as well.

As we tried to mention in the first paragraph the aim of the thesis is performance evaluation of the algorithms in reducing MAI interference. Therefore all factors affecting their performance are kept constant throughout the simulation even if they are not completely ignored. One of such factor is ISI resulted due to multipath effect. This is because CDMA technique is a wide band signal in which the signal bandwidth is greater than the coherence band width of the channel in which case different frequency spectrum of the signal are affected differently. Throughout the simulation we assumed that there are three resolvable multipaths. To reduce the effect of ISI two finger rake with maximum ratio combining is used in all matched filter receiver. And another assumption made in these simulations is that there is perfect power control for each mobile subscriber that is, signal of different active user arriving at the receiver are equal. This assumption is necessary due to near far effect arising from power level difference of each users signal arriving at the receiver.



5.1 Simulation Block Diagram

5.2 Simulation Parameters

The simulation parameter used in this thesis is based on the practical property of CDMA system. The sampling frequency is chosen in microseconds due to CDMA bandwidth, which around 3.25 MHz. We considered pedestrian subscriber, so a Doppler frequency of 5Hz at 2GHz carrier frequency is used. For LMS adaptive algorithm the number of training symbols used are chosen considering convergence and bandwidth wastage in to consideration.

Table 5.1 Simulation Parameters

Sampling time	1e-6
Channel	Rayleigh
Delay spread	[0 1e-6 2e-6]
Average path gain(dB)	[-2 , -18,-20]
Number of randomly generated bit source	100,000
Number of update tap weight	32
Modulation	BPSK
Carrier frequency	2Ghz
LMS and CMA step size	0.01
Number of realization per SNR	200
Doppler frequency (Jakes' model)	5Hz
Number of training sequence for LMS	400

5.3 The Performance of Different Code in Reducing MAI

Different spreading codes have different capabilities in reducing MAI due to their difference in their cross correlation property.

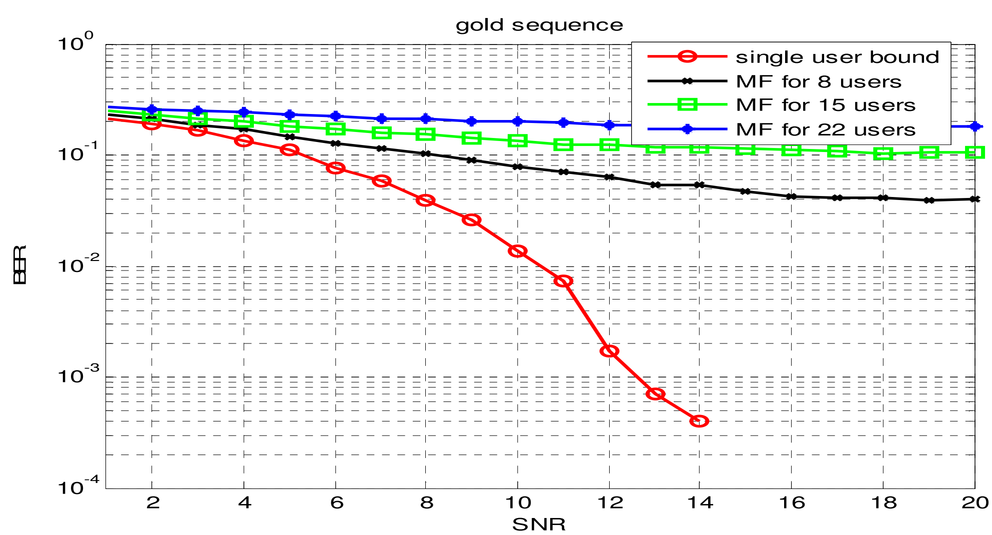


Figure 5.2 MAI reduction capability of Gold code with N=31

As can be seen in Figure 5.2 the BER performance for the desired user signal degrades rapidly as the number of interfering user varies from 8 to 22 for gold sequence of spreading gain 31 (N=31). But in Figure 5.3 the BER of the desired user signal with interference is almost as equal as the BER performance for user without any inference for which the spreading code is Walsh-Hadamard code of spreading gain 32 (N=32). This is because the interfering users' signals are completely eliminated due of the orthogonal nature of the spreading sequences.

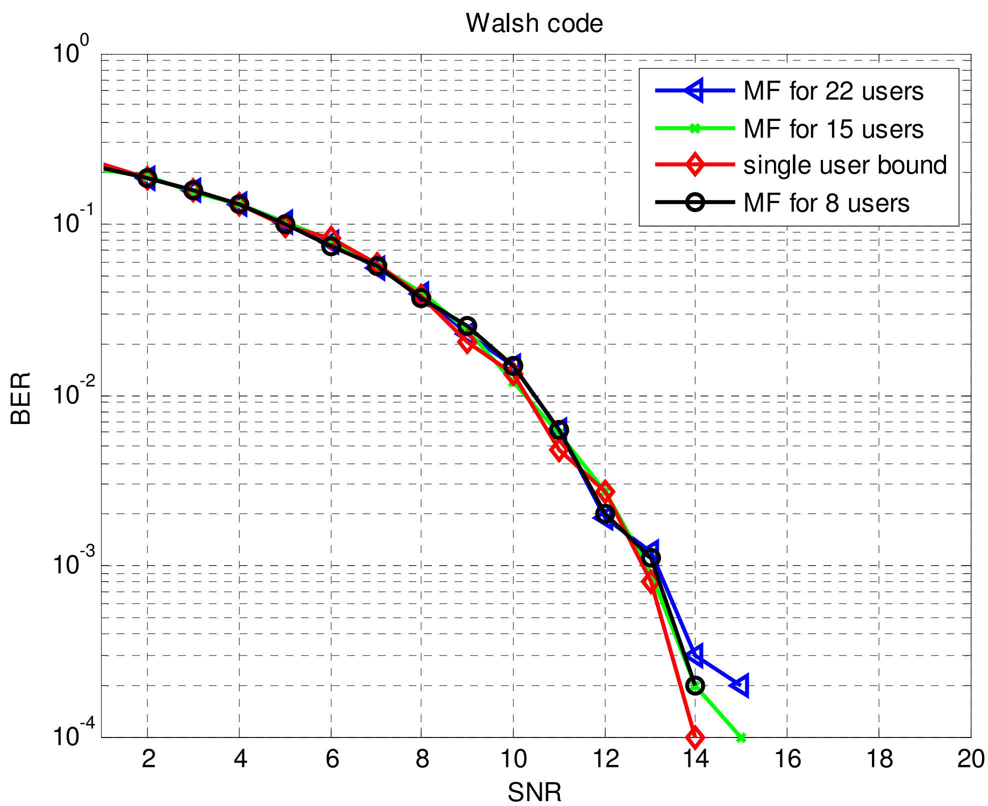


Figure 5.3 MAI reduction capability of Walsh-Hadmard code with N=32

5.3.1 Effect of Spreading Gain in Reducing MAI Vs Bandwidth Extension

As the spreading gain of each spreading code increases their ability to reduce MAI increases, this is obtained actually at the cost of bandwidth. As we seen in the Figure 5.4 and 5.5 the signal quality of the desired user signal degrades as the number of interfering user increases from 8 to 22 for both cases.

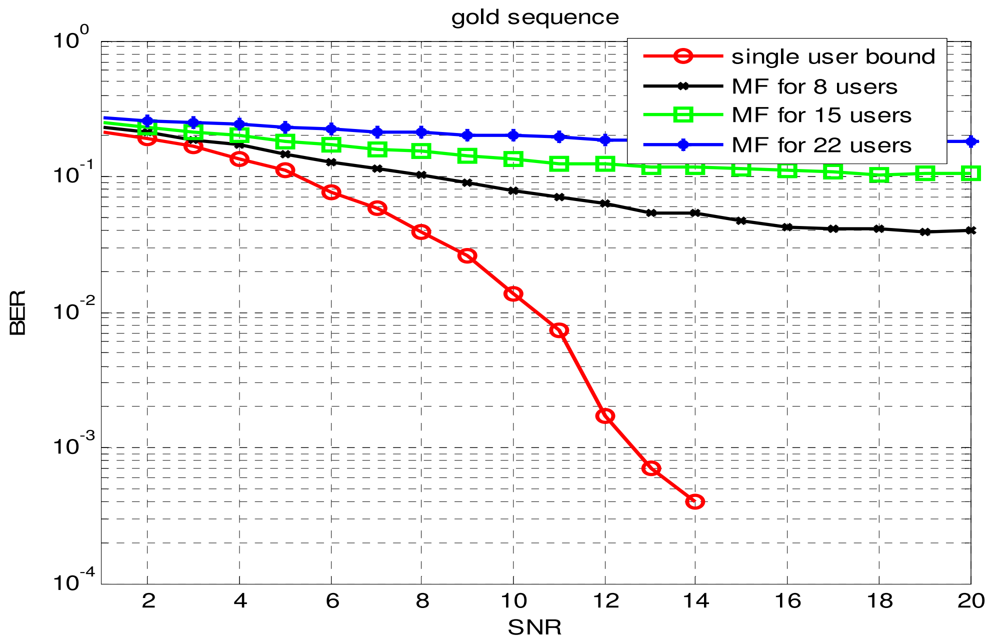


Figure 5.4 MAI in Gold code with spreading gain N=31

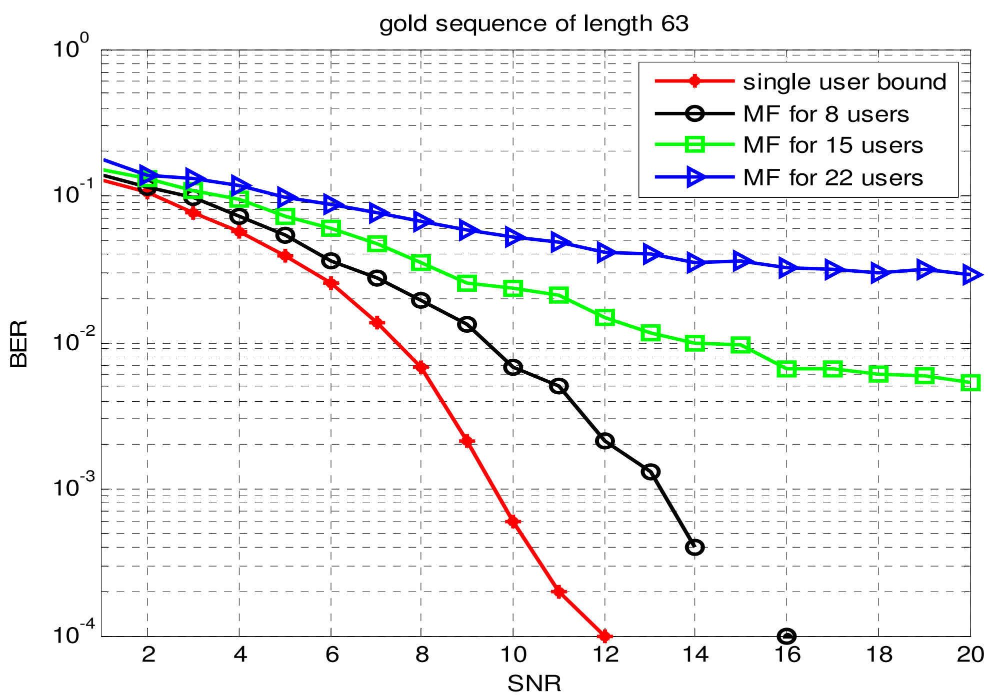


Figure 5.5 MAI in Gold code with spreading gain N=63

But as the spreading gain increases from N= 31(in Figure 5.4) to N= 63 (in Figure 5.5) for gold code there is BER improvement. In addition to MAI reduction signal spreaded with higher spreading gain has better BER performance. As can be seen in Figure 5.5 there is a 3dB improvement in SNR than the single user in Figure 5.4 at the BER of 10^{-3}. Also the BER of the desired user signals with 8 interferences show much improvement as seen in Figure 5.5 than signal interfered with 8 users in Figure 5.4. But bandwidth required for signal spreaded with N=63 (in Figure 5.5) is almost twice that of signal spreaded with N=31(in Figure 5.4).

For the case of Walsh-Hadamard code of spreading gain N=33 (in Figure 5.6) and N=64 (in Figure 5.7) there is complete elimination of MAI in both cases. This is because of the orthogonality nature of the spreading sequences. But one thing we can see in these two figures (Figure 5.6 and Figure 5.7), there is almost 3dB improvement in SNR for signal spread with spreading gain of N = 64 than signal spreaded with N=32 at 10^{-3} BER.

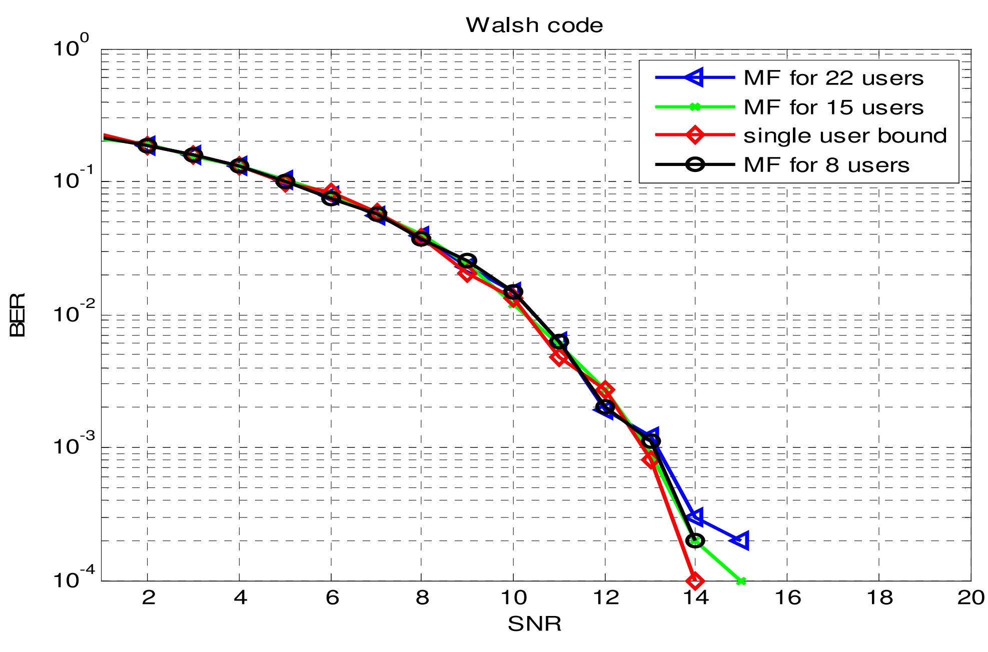


Figure 5.6 MAI in Walsh-Hadamard with spreading gain N=32

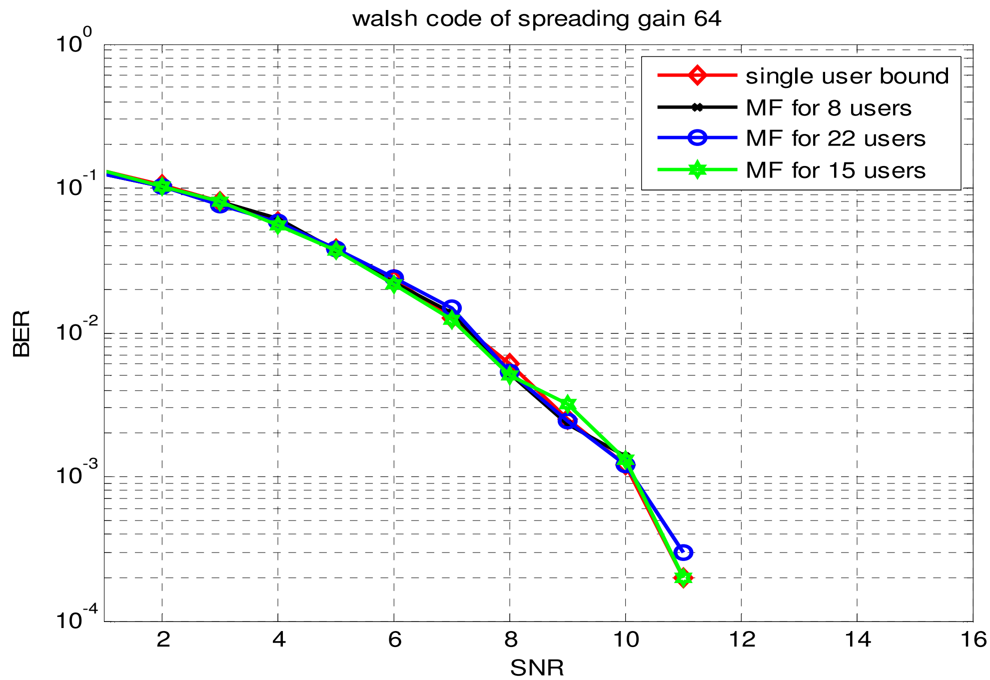


Figure 5.7 MAI in Walsh-Hadamard with spreading gain N=64

The 3dB improvement shown in Figure 5.7 above is at expense of bandwidth for the same number of interfering users as in Figure 5.5. The band width required in the Figure 5.7 is twice the bandwidth required in Figure 5.6.

5.4 Performance Evaluation of Adaptive Algorithm in Reducing MAI

The performance of adaptive LMS and CMA depends on the selected step size and number of tap weights as discussed below.

5.4.1 Effect of Step Size on the Convergence of the Adaptive Algorithm

The mean square error (MSE) was estimated over 2000 number of iterations for different step size. As we can see in Figure 5.8 the algorithm with higher step size (μ= 0.1) value converges to steady state value more quickly than algorithm with small step size (μ= 0.01 andμ= 0.001). But the main disadvantage of using algorithm with large step size is that it causes instability about the mean as discussed in Chapter four, Section 4.2.1.

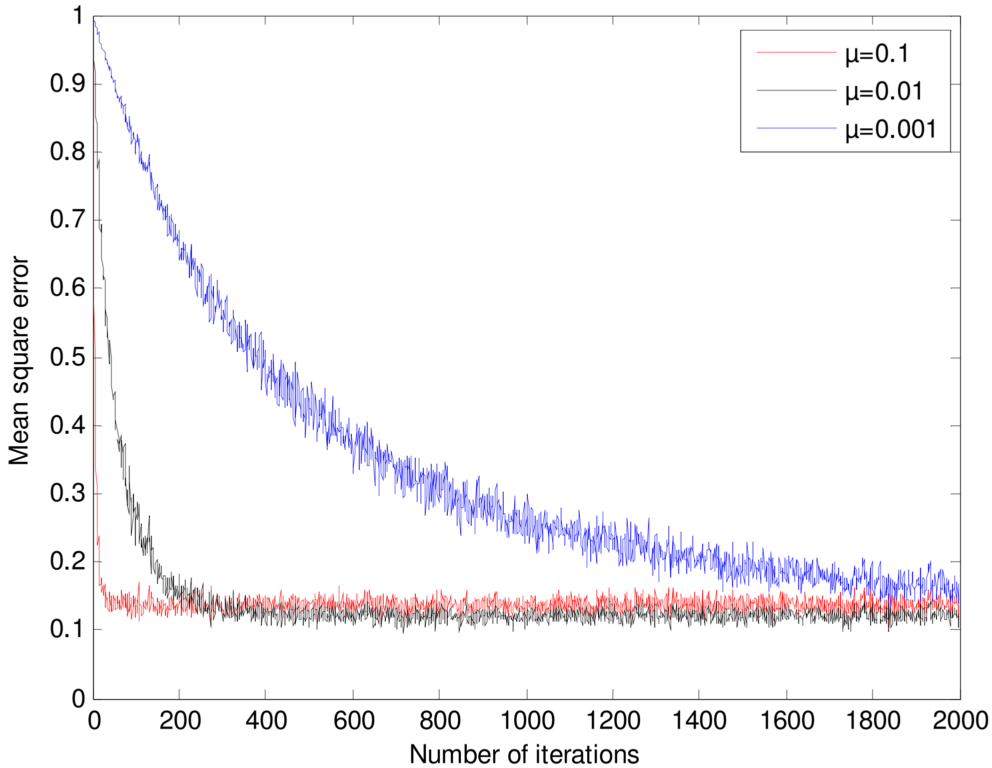


Figure 5.8 effect of step size on convergence of the algorithms

5.4.2 Effect of Number of Tap Weight on the Adaptive Algorithm Performance

Figure 5.9 shows the effect of the number of tap weights in the convergence of algorithm. As can be seen in the figure algorithm with high number of tap weights will converges to steady state quickly than algorithm with small number of tap weights. But the main drawback of using large number of tap weights results in higher complexity (i.e., more number of addition and multiplication). So in selecting the number of tap weights, convergence versus complexity must be taken into consideration.

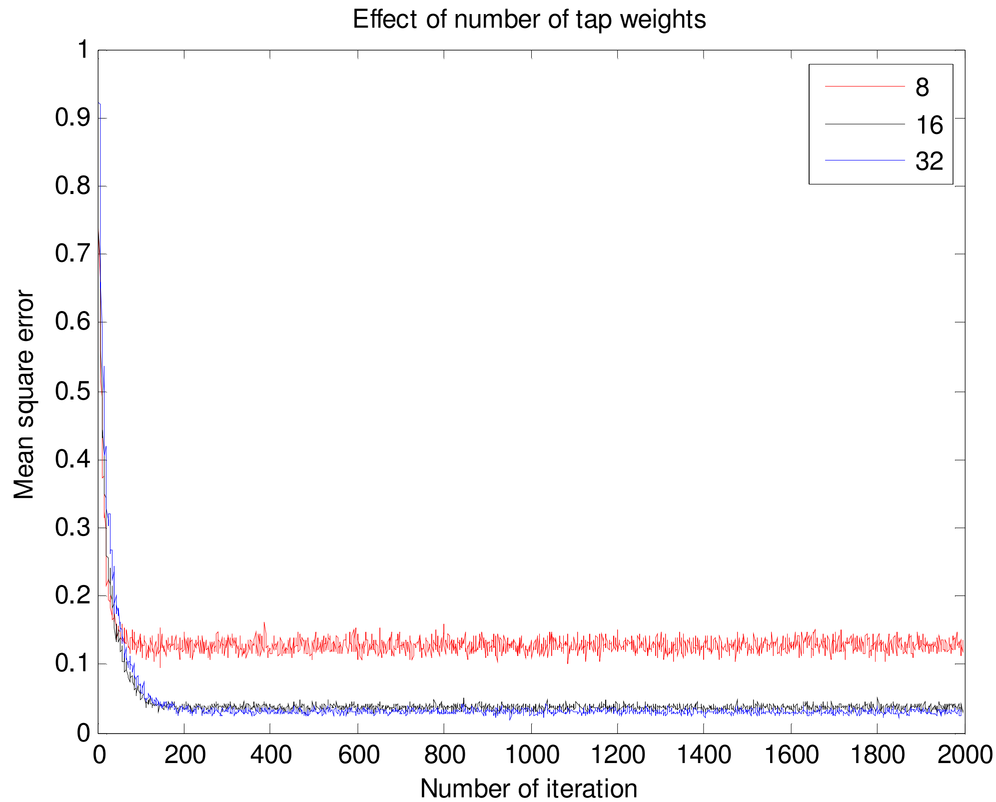


Figure 5.9 effect of number of weights on convergence of the algorithms

5.4.3 MAI Reduction Comparison of Matched Filter Receiver, Decorrelating Receiver and Adaptive Receivers.

There are three ways in which we can suppress MAI in CDMA receiver. Among this appropriate selection of spreading code that is those with small cross correlation or orthogonal spreading code is one option. We have already discussed this in previous Section. The second option is perfect power control of each user's received signal, so there will not be any problem of near far effect. The third mechanism is using appropriate detection mechanism in which complexity versus performance should be given due attention.

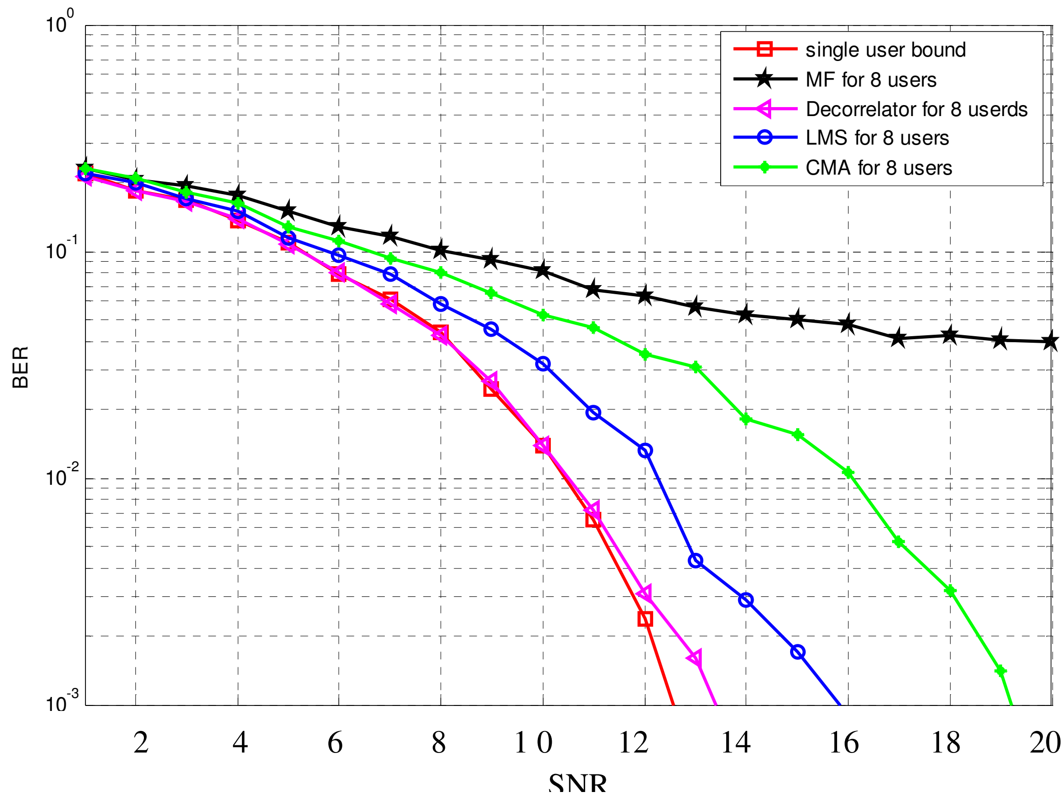


Figure 5.10 performance comparisons for LMS, CMA, decorrelating detector with gold code N=31.

In the Figure above (Fig. 5.9) we have considered three detection mechanisms. Those are matched filter receiver, decorrelating detector and adaptive algorithm receivers. As we can see from the Figure 5.10 the BER performance of the matched filter for single user with no interference is optimal but it performance degrades as the number of interfering users increases to 8. The decorrelating detector is optimal in achieving the error probability that is near the single user scenario. The main limitation of this detection is that it requires the knowledge of the spreading codes of all active users in the network. This is potentially a significant burden for some types of networks from an operational viewpoint. It also poses a security problem for certain communication applications. While it is reasonable to assume that the base station in a cellular access to the spreading codes of all of the users present in the base station's cell, each of the individual, mobile users in the cell may not have access to this information. Since such information will also be dynamic, it may not be practical for the base station to continually transmit such information.

By introducing an adaptive element into the CDMA receiver, performance can be enhanced. As shown in Figure 5.9 the error probability of adaptive algorithm is much better than matched filter receiver where it performs less compared to decorrelating detector. But these receivers (adaptive) do not require any information about the interfering users' spreading code hence preserves CDMA security system. The other thing that we can observe in the BER performance of the receivers shown in Figure 5.10 is the performance difference between LMS and CMA receiver. It is shown that the error probability for LMS outperforms that of CMA. This is due to the algorithm prior adaptation to the desired user signal with training sequences where it converges more rapidly than the blind CMA. For a fast varying channel the transmitter should have to send a long training sequence repeatedly which results in waste of bandwidth. The CMA even if it does not require training sequence for its adaptation it does require appropriate tap weight initialization.

Chapter 6

6. Conclusion and Recommendation

6.1 Conclusion

DS-CDMA is a high capacity air interface as compared to TDMA and FDMA. However, the problems specific to DS-CDMA systems are due to multiple users sharing the same bandwidth generally at the same time. A DS-CDMA receiver is therefore required to operate as a multiple access interference canceller in multi user receiver. The ultimate goal in the design of a multiuser CDMA receiver is to achieve minimum error probability approaching that which would be achieved in single-user scenarios, i.e., in the absence of interfering users. This task is further complicated by the requirement to handle high-speed data transmission while maintaining low computational complexity.

Conventional receivers employed in commercial CDMA systems are based on matched filtering in the form of RAKE receivers. The performance can degrade dramatically if strong interference is present in the channel. Practically, CDMA is therefore an interference-limited air interface for multiple-access communications systems.

A linear detector called the decorrelating detector is a multiuser detector which can attain the almost the same error probability as single user bound while the complexity is linear in the number of interfering users. Nevertheless, a major disadvantage of the decorrelating detector is the problem of noise enhancement. Adaptive algorithm receiver doesn't require knowledge of the spreading code of the desired user and other information of the interfering users; but does require training sequence. Unlike the training based adaptive receiver, a blind adaptive receiver does not require a training sequence. Knowledge of such information can be cumbersome in some applications that require high data rate transmission. It was found that the performance of adaptive algorithm is better than that of matched filter receiver but with more complexity (i.e., more addition and multiplication). LMS receiver which has better performance than CMA receiver wastes bandwidth during its training secessions.

6.2 Future Work

Future generation wireless communications systems require high-performance receivers to cope with high-speed data transmission. Capability in handling a large number of users may be necessary in operating in high population areas. Flexibility in accommodating different services with different rates of transmission, e.g., data transferring and voice-grade services, is also required. The theory and design of the least mean square error (LMS) and constant modulus algorithm (CMA) adaptive receivers used in this thesis may be employed to support these requirements. Future work related to the presented receiver structures can be suggested as follows:

- Combing adaptive algorithm with rake receiver which mitigates ISI and hence enhance the capability of the algorithms in reducing MAI.
- Least mean square error (LMS) algorithm suffers from slow convergence therefore it may be advisable to use fast convergent algorithm like RLS until it converges and then switching to LMS algorithm.
- The CMA converges in undesired way if the first response or tap weight initialization is not appropriate therefore it is better to initialize the tap weight of this algorithm with training based algorithms using small number of training sequences and then switching to CMA.

References

[1] Honig, M. L., and Poor, H. V., "Adaptive interference suppression in wireless communications" Signal processing perspectives, H. V. Poor and G. W. Wornell, Eds. Prentice-Hall, Upper saddle River, NJ, 1998, pp. 64-128.

[2] S. Verdu, "Minimum probability of error for asynchronous Gaussianmultiple-access channels," *IEEE Trans. Inform. Theory,*vol. 32, no. 1, pp. 85-96, January 1986.

[3] Urbashi Mitra and H. Vincent poor, "Adaptive receiver algorithm for near-far resistant CDMA", *IEEE Trans on Comms*, VOL. 43, NO. 2/3/4. February/March/April 1995

[4] Urbashi Mitra and H. Vincent poor, "Adaptive receiver algorithm for near-far resistant CDMA", *IEEE Trans on Comms*, VOL. 43, NO. 2/3/4. February/March/April 1995

[5] T. S. Rappaport, B D Woerner, and J H Reed, *Wireless Personal Communications*, Kluwer Academic Publishers, 1996.

[6] R. L. Pickholtz, , D. L. Schilling and L. B. Milstein,"Theory of Spread-Spectrum Communications—A Tutorial" IEEE Trans. Commun., vol. 30, no. 5, pp 855-884, May 1982

[7] J. Proakis, *Digital Communications*, 3rd edition, McGraw-Hall, 1995.

[8] Haykin, S., *Communication Systems*, John Wiley & Sons, Inc, 3rd edition. 1994.

[9] Luca Fanucci et al., *An Experimental Approach to CDMA and Interference Mitigation: From System Architecture to Hardware Testing through VLSI Design*, Kluwer Academic Publishers, 2004.

[10] H. Liu., *Signal Processing Applications in CDMA Communications*, Artech House, Inc. 2000.

[11] Robert A. Scholtz, "The Origins of Spread-Spectrum Communications", IEEE Transactions on Communications, Vol.Com.-30, No. 5, May 1982.

[12] Robert C. Dixon, *Spread Spectrum Systems with Commercial Applications*, 3rd edition, 1994, John Wiley & Sons.

[13] R. L. Peterson, R. E. Ziemer, D. E. Borth, *Introduction to Spread Spectrum Communications*, Prentice-Hall, 1995.

[14] R. Gold, “Optimum binary sequences for spread-spectrum multiplexing,” *IEEE Trans. Inform. Theory*, pp. 619-621, Oct. 1967

[15] E.H. Dinan and B. Jabbari, “Spreading codes for direct sequence CDMA and wideband CDMA cellular networks,” *IEEE Commun. Mag.,* pp. 48–54, Sept. 1998.

[16] M. K. Simon, J.K. Omura, R.A. Scholtz, B.K. Levitt, *Spread Spectrum Communications Handbook*. New York: McGraw Hill, 1994.

[17] S. Glisic, B. Vucetic, *CDMA Systems for Wireless Communications*, Artech House, 1997.

[18] Verdu, S. *Multiuser Detection*. Cambridge University Press, New York, 1998.

[19] Andrea Goldsmith, *Wireless Communications*. Cambridge University Press,2005.

[20] Michel C. Jeruchim, Philip Balaban, K. Sam Shanmugan, *Simulation of Communication Systems*. Publisher: Plenum, 1992.

[21] Bernard Sklar. “Rayleigh Fading Channels in Mobile Digital communication Systems; Part I: Characterization.” IEEE communications Magazine, July 1997.

[22] Akbar M. Sayeed and Behnaam Aazhang. “Communcation over Multipath Fading Cahnnels: A Time-Frequencey Perspective.” PIMRC’97, Helsinki, Finland, September 1997.

[23] E. Biglieri, C. Caire, G. Taricco, and J. Ventura-Traveset, “Simple method for evaluating error probabilities,” *IEE Electron. Lett.*, vol. 32, February 1996, pp. 191–192.

[24] E. Biglieri, C. Caire, G. Taricco, and J. Ventura-Traveset, “Computing error probabilities over fading channels: a unified approach,” *Eur. Trans. Telecommun.*, vol. 9, February 1998, pp. 15–25

[25] D. Divsalar, “Performance of mismatched receivers on bandlimited channels,” Ph.D. dissertation, University of California at Los Angeles, 1978.

[26] M. Abramowitz and I. A. Stegun, *Handbook of Mathematical Functions with Formulas, Graphs, and Mathematical Tables*, 9th ed. New York: Dover Press, 1972.

[27] D. A. Johnston and S. K. Jones, “Spectrally efficient communication via fading channels using coded multilevel DPSK,” *IEEE Trans. Commun.*, vol. COM-29, no. 3, March 1981, pp. 276–284.

[28] S. Moshavi, “Multiuser Detection for DS-CDMA Communications," IEEE Communications magazine, pp. 124-136, Oct. 1996.

[29] A. Duel-Hallen, J. Holtzman, and Z. Zvonar, "Multiuser Detection for CDMA Sysems" IEEE Personal Communications. April 1995.

[30] J. B. Groe and L. E. Larson, *CDMA Mobile Radio Design*, Artech House, Inc.2000.

[31] S. Verdu,"Minmum Probability of Error for Asynchronous Multiuser Channels" IEEE Trans. Info. Theory, vol 32, no.1, pp.85-96. Jan. 1986.

[32] Haykin, S. *Adaptive Filter theor*y. Prentice-Hall, Englewood Cliffs,NJ, 1986.

[33] Lim, T. J., Gong, Y., and Farhang-Boroujeny, B. "Convergence analysis of chip- and fractionally spaced LMS adaptive multiuser CDMA detectors", IEEE Trans. Signal processing 48 (2000), pp. 2219-2228.

[34] Takeo Ohgane, "Characteristics of CMA adaptive array for selective fading compensation in digital land mobile radio communications," Electronics and Communications in Japan, Part 1, vol. 74, no. 9, pp. 43-53, 1991.

[35] Li, Y., and Liu, K .J. R., "static and dynamic convergence behavior of adaptive blind equalizers", IEEE Trans. Signal processing 44 (1996), pp. 2736-2745.

[36] Treichel, J. R., and Agee, B. G. A, "new approach to multipath correlation of constant modulas signals", IEEE Trans. Acoust, Speech, Signal, Processing 31 (1983), pp. 459-472.

[37] Johnson Jr., C. R., Schniter, P., Endres, T. J., Behm, J. D., Brown, D. R., and Casas, R.A. "Blind equalization using the constant modulus criterion", A review. Proc. IEEE 86 (1998), pp. 1927-1950.

[38] Zeng, H. H., Tong, L., and Johnson Jr., C. R. "Relationships between the constant modulus and wiener receivers." IEEE Trans. Inform. Theory 44 (1998), pp. 1523-1538.

[39] Johnson Jr., C. R., Schniter, P., Fijalkow, I., Tong, L., Behm, J. D., Larimore, M. G., Brown, D. R., Casas, R. A., Endres, T. J., Lambotharan, S., Touzni, A., Zeng, H. H., Green, M., and Treich Ler, J. R., "The core of FSE-CMA behavior theory. In unsupervised Adaptive Filtering", S. Haykin, Ed., vol.2.John wiley and sons, New York, 2000, pp.13-112.

[40] Li, Y., and Ding, Z., "Convergence analysis of finite length blind adaptive equalizers." IEEE Trans. Signal processing 43 (1995), pp. 2120-2129.

[41] Lambotharan, S., Chambers, J., and Johnson Jr., C. R. ,"Attraction of saddles and slow convergence in CMA adaptation", Signal processing 59 (1997), pp. 335-340.

[42] Miguez, J., and Castedo, L.," A linearly constrained constant modulus approach to blind adaptive multiuser interference suppression", IEEE Commun. Lett. 2 (1998), pp. 217-219.

[43] Li. L., and Fan, H. H., "Blind CDMA detection and equalization using linearly constrained CMA", In Proc. Int. Conf. Acoustics, speech and signal processing (Istanbul, Turkey, 2000), vol. 5. Pp. 2905-2908.

[44] Tugnait, J. K., and Li, T. "Blind asynchronous multiuser CDMA receivers for ISI channels using code-aided CMA", IEEE J. Select. Areas Commun. 19 (2001), pp. 1520-1530.

[45] Peerapol Yuvapoositanon, "Blind Adaptive Techniques for Direct-sequence Code Division Multiple Access Receivers", Ph.D dissertation, university of London, 2002.