Deanship of Graduate Studies

Al- Quds University

Designing MMSE Receivers With Variable Step-Size Adaptive Filters For MC-DS-CDMA Mobile Communication Systems

Hutaf Mohammad Ruweished

M.Sc. Thesis

December, 2007

Designing MMSE Receivers With Variable Step-Size Adaptive Filters
For MC-DS-CDMA Mobile Communication Systems

By

Hutaf Mohammad Ruweished

M.Sc.: Electronics Engineering, Al-Quds University, Palestine

A Thesis Submitted in Partial fulfillment of the requirements for the degree of Master of

Electronic and Computer Engineering, Faculty of Engineering.

Al- Quds University

December, 2007

Electronic and Computer Engineering
Faculty of Engineering
Al-Quds University

Designing MMSE Receivers With Variable Step-Size Adaptive Filters
For MC-DS-CDMA Mobile Communication Systems

By

Student Name: Hutaf Mohammad Ruweished

Registration No: 20320176

Supervisor: Dr. Hanna Abdel Nour

Co- Supervisor: Dr. Ali Jamoos

Master thesis submitted and accepted, Date: 1/12/2007

The names and signatures of the examining committee members are as follows:

1-Dr. Hanna Abdel Nour: Head of Committee **Signature ……………**

2-Dr. Hussein Jaddu: Internal Examiner **Signature ………………..**

3-Dr. Nasser Hamad: External Examiner **Signature ………………..**

Al-Quds University
December,2007

Dedication

To my small family,
My husband and my children,
To my big family,
My parents, brothers and sisters,
To all those interested in this subject.

Declaration:

I Certify that this thesis submitted for the degree of Master is the result of my own research, except where otherwise acknowledged, and that this thesis (or any part of the same) has not been submitted for a higher degree to any other university or institution.

Signed………………………….

Hutaf Mohammad Ruweished

Date: 1-12-2007

ACKNOWLEDGMENTS

I would like to thank my supervisor, Dr. Hanna Abdel Nour, for all the support, encouragement and patience he always guided me with and a special thank to him offering related references (books, scientific magazines, and IEEE papers), reviewing my reports and slides during the last academic year and following up the financial terms to give me the opportunity to participate in the ACIT'06 conference held in Jordan from the 19th to the 21st of December 2006.

Special thanks to Engineer Ali Jamoos – Ph.D. student at Bordeaux 1 University in France-for his help, advice, encouragement, support and suggestions all throughout this work. I would like also to thank him for suggesting the title of this thesis. Also I appreciate the constructive and encouraging comments he offered and the new ideas he proposed. Finally, I greatly appreciate his commenting of the simulations and reviewing this thesis that resulted in publishing a conference paper.

I would like also to thank the Faculty of Engineering at Al-Quds University for the facilities they have offered for the Master students.

Thanks to the Faculty of Higher Education Studies at Al-Quds University for their financial support without which I would not have been able to participate in the ACIT'06 conference in Jordan.

Thanks to Al-Quds University library rich in scientific books. The access to a foreign library in Toronto helped me a lot in viewing published IEEE papers related to my thesis subject.

Special thanks to my colleagues and friends in the Electronics and Computer Engineering Department for sharing their knowledge during the past three years.

My sincere gratitude to the numerous authors listed in the References for their contributions to the state of the art, and without whom this report would not have been materialized.

Last but not least, I express my sincere gratitude to my husband, my parents, my mother-in-law, my sister in-law, my sisters and brothers for their priceless guidance, relentless support and boundless love.

It should be noted that this thesis is the fruit of collaboration with the Signal and Image Group (ESI) at the UMR CNRS 5218 IMS in Bordeaux1 University-France. More particularly, it results from the close cooperation with Mr. A. Jamoos and Dr. E. Grivel who are members of the ESI.

ABSTRACT

The fourth-Generation (4G) of mobile systems aim at providing high data rates interactive multimedia and wireless internet services. However, due to the presence of noise, Inter-Symbol-Interference (ISI) and Multiple Access Interferences (MAI) that contaminate and attenuate the original transmitted signal, these demands give rise to computational complexity in order to retrieve a replica of this signal with the least error.

The conventional correlator based receiver that was first used could not suppress the MAI and the multi-path fading of the mobile channels. An alternative approach used an optimal Minimum Mean Square Error (MMSE) receiver succeeded in reducing MAI and ISI but was considered highly computational.

Later, an adaptive MMSE receiver for asynchronous Multi-Carrier (MC) Direct sequence (DS) Code Division Multiple Access (CDMA) mobile communication systems over frequency selective Rayleigh fading channels was introduced. Two adaptive MMSE design structures were proposed. The Separate Detection (SD) that consists in using a particular filter structure for each carrier and the Joint Detection (JD) defined by the concatenation of the filter weights dedicated to each carrier. Both structures were implemented adaptively using the three traditional well-known adaptive algorithms, mainly, the Recursive Least Square (RLS), the Normalized Least Mean Square (NLMS) and the Affine Projection Algorithm (APA). However, NLMS and APA algorithms have fixed step-sizes. The step-size in these adaptive algorithms should be controlled to meet the conflicting requirements of fast convergence and low steady state excess mean square error.

In this thesis, we propose variable step-size adaptive filter based Minimum Mean Square Error (MMSE) receivers for MC-DS-CDMA mobile systems that tend to reduce the receiver’s computational cost while maintaining good performance in term of fast convergence speed and low mis-adjustment error. Therefore, we investigate the relevance of Variable Step Size (VSS) NLMS and APA adaptive filters.

Finally, varying the step-sizes of the NLMS and the APA algorithms in both the SD and the JD MMSE receivers showed the best Bit Error Rate (BER) and convergence performance, compared to the conventional and the fixed step-size MMSE receivers. In particular, results ensured that the variable step-size joint detection (VSS-JD) MMSE based receiver outer-performed the variable step-size separate detection (VSS-SD) MMSE based receiver and the conventional correlator based receiver by showing the least BER and the fastest convergence speed during the given iterations.

The performances of the proposed receivers are evaluated and compared with other existing receivers by means of Monte Carlo simulations using Matlab.

ملخص

يهدف الجيل الرابع لأجهزة الاتصالات الجوالة الى تزويد المستخدم باتصالات لاسلكية ذات جودة عالية قدر الامكان. ولكن نظرا لوجود الضوضاء والتداخلات بأنواعها عبر قناة الاتصال (the channel)، من المرسل الى المستقبل، فان الاشارة الأصلية المرسلة (صوت وصورة) تتلوث وتضمحل مما يؤدي الى تعقيد الحسابات والمعادلات اللازمة لاسترجاعها أو استرجاع أقرب صورة لها بأقل نسبة خطأ ممكنة.

في البداية تم توظيف المستقبل الذي يستخدم جهاز التنقية التقليدي،لكنه فشل في التخلص من التداخلات الناشئة عبر قنوات الاتصال الجوالة.أما استخدام مستقبل من نوع "أقل معدل مربع خطأ (MMSE)،فقد نجح في تقليل التداخلات،لكنه اعتبر معقد في العمليات الحسابية.

لاحقا،تم استخدام مستقبل MMSE متأقلم في أجهزة الاتصالات الجوالة. اقترح نوعان من هذه المستقبلات:الأول يسمَى (separate detector) ويكون التحقق فيه من الإشارة المستقبلة كل على حدة أما النوع الثاني (joint detector) يكون تدقيق الاشارة فيه إجماليا ،أي بعد تجميع كل الإشارات المستقبلة. يتم في البداية استخدام الخوارزميات الثلاثة المعروفة وهي:أقل معدل مربع(NLMS) ، (RLS)وأقل معدل مربع العام (APA or GNLMS)ويكون حجم الخطوة المستخدم داخل هذه الألغوريثمات ثابت. اذا تم التحكم بحجم الخطوة، فان تصرف المستقبل سيتحسن نتيجة تخفيض نسبة الخطأ والسرعة في الوصول اليه.

في هذا البحث أقترح استخدام المستقبل MMSE ذا جهاز تنقية يوظف الغوريثم متغير ومتلائم لتخفيض أقل نسبة خطأ والسرعة التي يمكن تحقيقها في الوصول اليه مع المحافظة على الاداء الجيد.

في الخطوة التالية تتم المفاضلة بين كل ما ذكر من خلال نتائج البرمجة في توضيح من خلال دراسة أقل نسبة خطا والسرعة في الوصول إلى "صفر الخطأ".

تحديدا يتم تسليط الدراسة والبحث على تصميم أجهزة لاسلكية ذات مستقبل فيه جهاز تصفية ويتكيف مع المتغيرات (Adaptive filter)،بحجم خطوة متغير (variable step-size).

المفاضلة بين نوعين من المستقبلات تأتي تاليا :.أما في المرحلة الاخيرة وهي موضوع الرسالة حيث يتم تغيير "حجم الخطوة" في المستقبلين الاثنين في الخوارزميتين VSS-NLMS و VSS-APA.
نتيجة البحث كانت إيجابية بحيث قل الخطأ وزادت السرعة في الوصول إليه بشكل واضح . زيادة على ذلك أثبت البحث على أن المستقبل VSS-JD فاق المستقبل VSS-SD بتحقيق أقصى سرعة في الوصول الى أقل نسبة خطأ.
ولبيان مدى تفوق المستقبل VSS-APA-JD في بناء جهاز استقبال عال االجودة تم في الجزء الأخير من البحث المفاضلة بينه وبين كل من المستقبل التقليدي(correlator) والVSS-APA-SDمن حيث الزيادة في عدد المستخدمين الفعليين للشبكة والزيادة في نسبة التداخلات على الاشارة.ودلت النتائج مرة أخرى على تحقيق أقل نسبة خطأ لدى المستقبل VSS-APA-JD.الخوارزميات التي تم استخدامها في هذا البحث بنيت باستخدام برنامج Matlab وأهمية وسلوك الطرق المذكورة أعلاه والخصائص المختلفة تم استعراضها من خلال ما يسمى Monte Carlo Simulation .
في الختام، ونتيجة لما سبق فانه يوصى باستخدام VSS-APA-JD في المستقبلات في أجهزة الاتصالات اللاسلكية الجوالة.

TABLE OF CONTENTS

LIST OF TABLES

LIST OF FIGURES

ACRONYMS AND NOTATIONS

ADSL: Asymmetric Digital Subscriber Line
APA: Affine Projection Algorithms
AWGN: Additive White Gaussian Noise
BER: Bit Error Rate
BPSK: Binary Phase Shift Keying
BW: Band Width
BWA: Broadband Wireless Association
CDMA: Code Division Multiple Access
DS: Direct-Sequence
DSSS: Direct Sequence Spread Spectrum
EDGE: Enhanced Data Rates for GSM Evolution
FDM: Frequency-Division Multiplexing
FDMA: Frequency Division Multiple-Access
FH: Frequency-Hop
G: Generation
GNLMS: Generalized Normalized Least Mean Square
GPRS: General Packet Radio Services
GSM: Global System Mobile
ISI: Inter-Symbol Interference
ITU: International Telecommunications Union
ITU-R: International Telecommunications Union Radio Communication Sector
JD: Joint Detection
Kbps: Kilo bits per second
MAI: Multiple Access Interference
MC: Multi-Carrier
MCM: Multi-Carrier Modulation
MF: Matched Filters
MLSE: Maximum Likelihood Sequence Estimation
MMSE: Minimum Mean Square Error
NLMS: Normalized Least Mean Square
NMT: Nordic Mobile Telephone
OFDM: Orthogonal Frequency-Division Multiplexing
PDA: Personal Digital Assistant
RLS: Recursive Mean Square
SC: Single Carrier
SD: Separate Detection
SNR: Signal to Noise Ratio
TACS: Total Access Communication System
TDMA: Time Division Multiple Access
UMTS: Universal Mobile Telecommunications System
UWC: Universal Wireless Communication
VSS: Variable Step Size
WAN: Wide Area Networking
WCDMA: Wideband Code Division Multiple Access
WLAN: Wireless Local Area Networking
WPAN: Wireless Personal Area Networking

WWAN: Wireless Wide area Networking
w: filter weights
$(.)^*$: complex conjugate
$(.)^H$: complex conjugate transpose
i.i.d: independent and identically distributed
(ε):error
$(\hat{.})$ estimate
E[.]: expectation
ω : angular frequency
$(.)^{-1}$: inverse matrix
$(.)^T$: Tranpose
Re: real part
μ :step-size
$\|.\|$: norm
R: covariance matrix
I: identity matrix
P: cross-correlation matrix

INTRODUCTION

Several approaches that combine Direct-Sequence Code Division Multiple Access (DS-CDMA) with Multi-Carrier (MC) transmission have been proposed for the last decade [1] . Indeed, they make it possible to achieve high bandwidth efficiency, fading resilience and interference suppression capability, which are three crucial issues for future broadband mobile wireless communications [2] . In this thesis, we focus our attention on the so-called MC-DS-CDMA transmission scheme [2] [3] , that has been adopted as an option in the CDMA2000 third generation cellular standard [4] .

When dealing with MC-DS-CDMA systems, the data sequence multiplied by a spreading sequence (direct sequence) modulates several carriers [3] . It should be noted that the MC-DS-CDMA scheme deals with time-domain spreading whereas the so-called MC-CDMA is based on frequency-domain spreading. When designing receivers for MC-DS-CDMA mobile wireless systems, two issues should be taken into account. Firstly, the Multiple Access Interference (MAI) caused by other active users concurrently operating in the system, especially when the received signal power of the desired user is less than that of other users (the so-called near-far effect) [7] . Secondly, the multi-path fading of mobile wireless channels which greatly degrades the Bit Error Rate (BER) performance of the system.

The conventional MC-DS-CDMA receiver consists of a correlator for each carrier followed by a maximal ratio combiner to counteract fading and narrow band interference [3] .However, this approach cannot eliminate the MAI and hence is not “near-far resistant”. In addition, the fading processes are assumed to be known at the receiver, which is not the case in practice. An alternative approach consists in designing a Minimum Mean Square Error (MMSE) receiver. This method was previously used with Single-Carrier DS-CDMA systems [5] with adaptive filters such as Least Mean Square (LMS) and Recursive Least Square (RLS) [9] Indeed, the adaptive receivers make it possible to suppress both MAI and Inter Symbol Interference (ISI) and are shown to be near-far resistant [5] . In addition, they offer an attractive trade-off between performance, complexity and the need for side information (spreading codes, timing for all users, channel coefficients, etc.).

MMSE receivers have been recently used for MC-DS-CDMA systems. Thus, two design structures are proposed in [10]: the Separate Detection (SD) structure consists in carrying out a Wiener filter along each carrier, whereas the Joint Detection (JD) structure is defined by the concatenation of the filter weights dedicated to each carrier. In both structures, the Wiener filtering is performed based on the least-square estimated fading processes. However, this approach has very high computational cost. To reduce the computational complexity, adaptive implementations of the SD and JD MMSE receiver structures are proposed in [11]. The authors have carried a comparative study between three adaptive filters: the Normalized LMS (NLMS), the Affine Projection Algorithm (APA) and the RLS algorithms. They showed that the JD receiver structure with the various adaptive implementations outperforms the SD receiver structure in terms of the BER performance. In addition, they confirm that the APA has better convergence features than the NLMS.

They have concluded that the APA in the JD receiver structure corresponds to a trade-off between performance and computational cost. However, the step-size that governs the convergence rate and the steady state mean square error in the NLMS and APA is assumed to be constant.

In this thesis, we will focus on designing adaptive MMSE receivers for asynchronous uplink MC-DS-CDMA systems, over frequency selective Rayleigh fading channels. More particularly, our focus will be on the adaptive SD and JD receiver structures. Our contribution is then twofold. Firstly, to meet the conflicting requirements of fast convergence and low steady-state excess mean square error in the NLMS and APA, we propose to study the relevance of Variable Step-Size (VSS) NLMS and APA, recently elaborated in the framework of signal processing [12], to design the SD and JD receiver structures [13]. Secondly, we carry out a comparative study with the fixed step-size NLMS and APA based receivers by taking into account the convergence features and the BER performance.

The thesis is organized as follows. In the first chapter, an introduction about the mobile wireless systems is provided. Particularly, we point out the evolution of the mobile generations toward the fourth Generation (4G) and the employed multiple access techniques from the first Generation (1G) to the third Generation (3G). In addition, we detail the CDMA technology. In the second chapter, we focus on MC-DS-CDMA systems in Rayleigh fading channels. The transmitter model, the channel model and the conventional receiver model of these systems are described as presented in [3] . In Chapter 3, the optimal MMSE receiver and the various adaptive implementations of the SD and JD receiver structures using the NLMS, RLS and APA [11] are studied. In addition, the VSS-NLMS and VSS-APA based SD and JD receiver structures are developed and their performances with the fixed step-size are compared [13]. Finally, a conclusion and some recommendations for future work are provided in Chapter 4.

CHAPTER 1

FUNDAMENTALS OF MOBILE WIRELESS COMMUNICATION SYSTEMS

During the recent ten years there has been considerable progress in the development of wireless cellular telephony or digital mobile communication systems to compete wired-line telephone networks. This progress is further translated by a large number of mobile multimedia applications starting from wireless Internet applications to satellite transmission passing through wireless telephony. As a result, wireless coverage tremendously increased with no need to the setting up of expensive infrastructure like copper or fiber lines.

1.1 Mobile Communications

Up to these days four main generations of mobile communication systems are universally widely known. These are the first generation (1G), the second generation (2G) followed by the third generation (3G), with the 2.5G in-between, finally emerged the fourth generation (4G) mobile communication system.

The first generation mobile cellular networks, denoted (1G), used analogue modulation and frequency-division multiplexing (FDM) to provide voice transmissions by using frequencies around 900 MHz. Examples such as the system of TIA/EIA-553 in the United States operating around 850 MHz, the Total Access Communication System (TACS), and Nordic Mobile Telephone (NMT) in Europe operating at 450 and 900 MHz bands [14].

Thereafter, an obvious move to wireless technologies started introducing the second-generation of mobile communications (2G) that used Time Division Multiple Access (TDMA) digital technology. The 2G systems designed in the 1980s were still used mainly for voice applications [15] with added new capabilities including facsimile and messaging.

Optimizing the data channels for packet data characterized the 2.5G generation, which introduced access to the Internet from mobile devices. This is formed by the General Packet Radio Services (GPRS), considered as a Global System Mobile (GSM) enhancement.

The third generation (3G) wireless systems were developed in the 1990s as the demand for faster speed, global compatibility and multimedia services became a priority. Moreover, higher quality voice channels, as well as broadband data capabilities, (providing transmission speeds from 125Kbps up to 2 Mbps) made it possible to introduce global roaming with superior voice and data quality.

Transition from analog to digital cellular network has almost completed towards the end of 1995. Thereafter, the International Telecommunications Union defined future wireless systems beyond 3G wireless as 4 G mobile [15]. Fourth Generation (4G) mobile communications are qualified by the ability to support advanced and wideband multimedia services that can be symmetrical and asymmetrical, real-time and non-real-time 4 G will basically focus on the open wireless architecture, cost-effective and spectrum-efficient high-speed wireless mobile transmission**.**

The competitive rush to design and implement digital systems led to a variety of different and incompatible standards [15]. Mobile systems all over the world have either GSM (global system mobile), mainly in Europe; TDMA (time division multiple access) (IS-54/IS-136) and CDMA (code division multiple access) (IS-95), in the U.S., or PDC (personal digital cellular) in Japan.

Two second generation cellular systems (IS-54, GSM) use time/frequency multiple-access where by the available spectrum is divided into frequency slots (e.g., 30 kHz bands) but then each frequency slot is divided into time slots. Each user is then given a pair of frequencies (uplink and downlink) and a time slot during a frame. Different users can use the same frequency in the same cell except that they must transmit at different times. This technique is also being used in 2.5 generation wireless systems (e.g. Enhanced Data Rates for GSM Evolution, EDGE). It allows considerably higher transmission speeds of between 150 Kbps and up to 200 Kbps [15] [16].

Third Generation Standards, associated to Wide area networking coverage (WAN)s, are dominated by WCDMA access methods (Wideband Code Division Multiple Access)with a 5Mhz channel bandwidth),are considered to be four times the bandwidth of cdmaOne and 25 times that of GSM [15] [17].

3G WAN standards operate on licensed bands whereas WLAN/WPAN (Wide local area networking/ Wide personal area networking respectively) operates in unlicensed bands such as the Industrial, Scientific, and Medical (ISM) band used by Bluetooth [17] .

The major step from the second generation to 3G wireless or 4G mobile was the ability to support advanced and wideband multimedia services .

The cellular network evolution from the 1G to the 3G is summarized in Fig.(1.1).

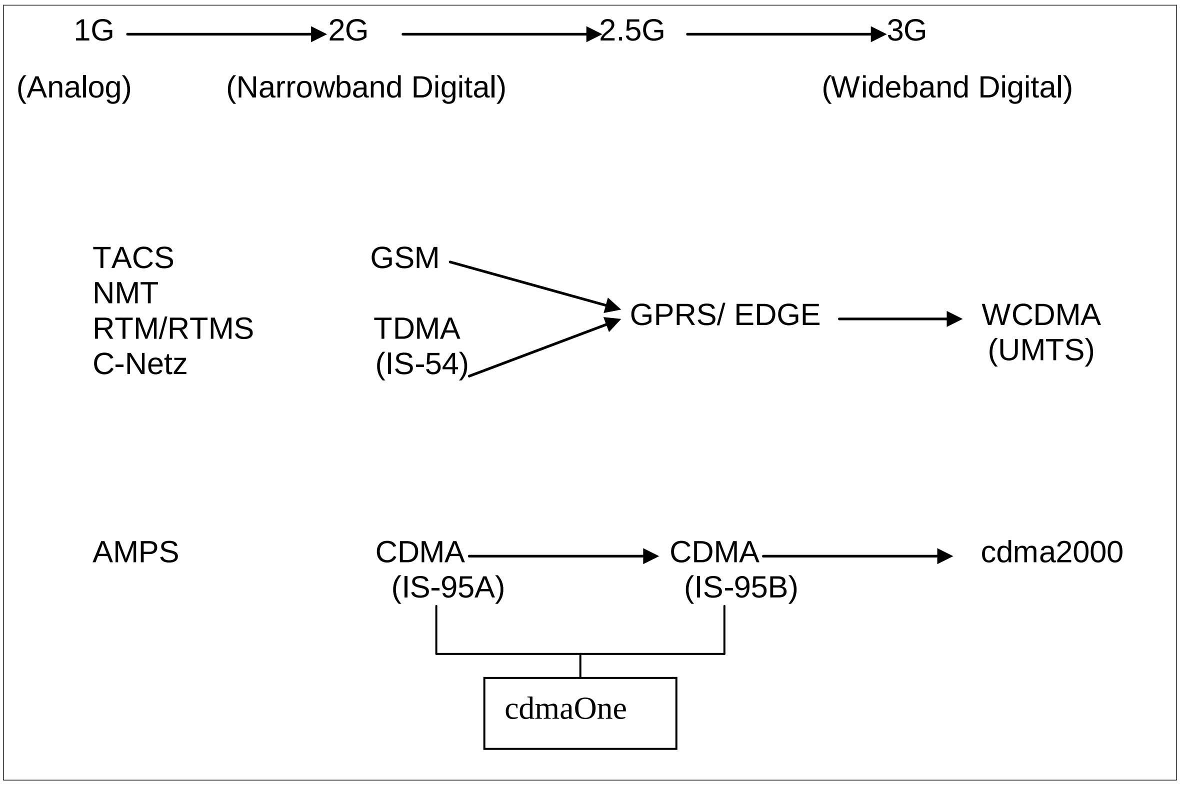

Fig.1.1: Cellular network evolution from the 1G to the 3G

GPRS is an extension of the GSM standard providing packet data capabilities with connection speeds up to 115 Kbps.

The International Telecommunications Union Radio Communication Sector (ITU-R) [15] [17] published in 1999 issued a set of standards for third-generation (3G) wireless systems. These systems include cdma2000, Universal Mobile Telecommunications System (UMTS) Wideband CDMA (W-CDMA) FDD, UMTS WCDMA TDD, and Time Division Multiple Access (TDMA) system known as Universal Wireless Communication-136 (UWC-136).

UMTS includes both voice and data capabilities in circuit and packet modes with connection speeds of 144, 384 Kbps & 2Mbps (availability dependent on mobility conditions).

Later on, came out the bluetooth wireless technology which is considered as an open specification for a low-cost, low-power, short-range radio technology for ad-hoc wireless communication of voice and data anywhere in the world. Distance covered by this technology ranges from 10-100 meters only operating in 2.4 GHz band at a data rate of 720 Kbps [17].

Table (1.1) below shows the data transmission-rates of the mentioned standards of the different technologies.

Table1.1: data transmission of different technologies

Cellular technology	Generation	Data transmission capacity
GSM	2G	14.4 Kbps
CDMA (IS-95B)	2.5G	64Kbps to 115 Kbps
GPRS	2.5G	115Kbps
CDMA 2000	3G	125Kbps to 2 Mbps
Bluetooth	3G	720Kbps
UMTS	3G	144, 384 Kbps & 2Mbps

It is noticed that the third generation CDMA 2000 cellular technology attains the highest data transmission capacity (up to 2Mbps) over the other mentioned technologies.

In terms of speed of transmission, the theoretical data rate that GSM mobile networks could support is 9.6 Kbps, while the CDMA networks could support from 64Kbps up to 2Mbps again with the CDMA2000.

1.2 Cellular Fundamentals

In mobile technology, the geographical area used for transmission of data is divided into small areas called cells after which the name cellular came. The transmission from the base station to a mobile is referred to as the downlink and the transmission from a mobile to a base station is the uplink. The uplink- downlink system is shown in Fig.(1.2).

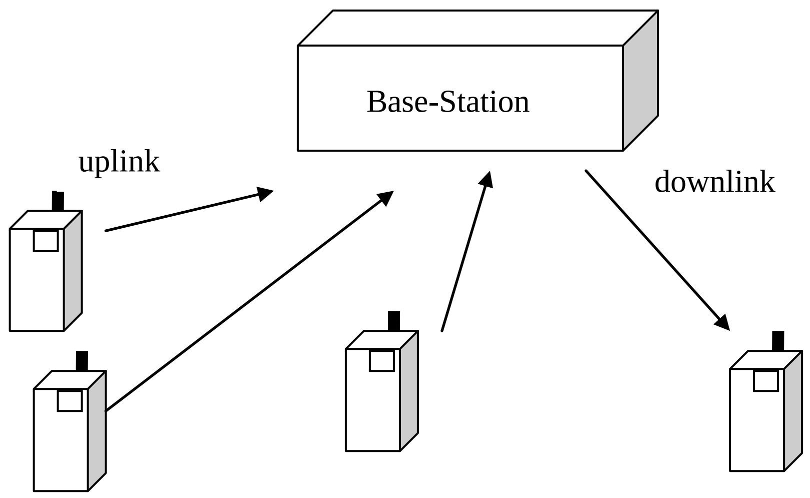

Fig.1.2: Uplink and downlink in mobile wireless systems

A mobile is ready to initiate and receive a call after it scans the control channels and tunes to a channel with the strongest signal [18]. It then exchanges identification information with the base station and is then authorized to use the network.

The cellular wireless system consists of transmitter, channel and receiver. At the transmitter the bandwidth required for bearing data such as voice and video that have redundant information in them, can be reduced (or compressed) by a process known as source coding. Due to interference from other users, thermal noise and fading (time-varying amplitude response of the channel) that affect the data transmitted through the channel, the signal is said to be susceptible to errors. However, it is possible to detect and correct some of these errors at the channel by applying channel coding to the bits being transmitted.

The transmitted signal passes through the channel on its way to the destination i.e. receiver, base station…etc. Further, the composite signal from the channel, results from multiple signals bouncing off obstacles, suffering varying delays and attenuations and getting superimposed. Both decoding and filtering are the major processes held at the receiver to retrieve the original transmitted data signal with the least minimum error.

As the number of users of a cellular communication system is in increase, there should emerge a technology to compensate this tremendous growth. This led to introducing multiple access techniques that will be discussed in the next section.

1.3 Multiple Access Techniques

The goal in the design of cellular systems is to be able to handle as many calls as possible (this is called capacity in cellular terminology) in a given bandwidth with some reliability. There are several different ways to allow multiple access to the channel. These include the following:

- Frequency Division Multiple-Access (FDMA)
- Time Division Multiple-Access (TDMA)
- Time/Frequency Multiple-Access
- Code Division Multiple-Access (CDMA)
    - Frequency-Hop (FH) CDMA
    - Direct-Sequence (DS) CDMA
    - Multi-Carrier (MC) CDMA
- Orthogonal Frequency-Division Multiplexing (OFDM)

These schemes are known as the Multiple Access Techniques [19].

Multiple access systems designed for mobile communications have traditionally employed TDMA and FDMA techniques. Orthogonal frequency-division multiplexing (OFDM) is chosen over a single carrier solution due to lower complexity of equalizers for high delay spread channels or high data rates.

Initially multiple-access techniques for cellular systems used Frequency Division Multiple Access (FDMA). Here the allocated spectrum is divided into frequency slots (frequency division duplexing). Each user is assigned a pair of frequencies when placing or receiving a call. One frequency is used for downlink (base station to mobile) and one for uplink (mobile to base station). A disadvantage appears in preventing a reassignment of the spectrum even though the user may not be talking.

In Time Division Multiple Access (TDMA), on the other hand, the time-domain transmission frame is periodically divided into time slots. Thus each user transmitting in TDMA or FDMA, accesses its own slot in either the time-domain or frequency-domain and hence they are thus orthogonal to each other either in time or frequency, respectively.

Orthogonal Frequency-Division Multiplexing**, (**OFDM) is a modulation scheme that has recently gained immense popularity in the design of wireless communication systems. It is a digital multi-carrier modulation scheme, which uses a large number of closely-spaced orthogonal sub-carriers. Each sub-carrier is modulated with a conventional modulation scheme at a low symbol rate, maintaining data rates similar to conventional single-carrier modulation schemes in the same bandwidth. The orthogonality of the sub-carriers results in zero cross-talk, even though they are so close that their spectra overlap. It has high bandwidth efficiency and is scalable to high data rates. It solves the problem of Inter Symbol Interference (ISI). An example of this application is the Asymmetric Digital Subscriber Line (ADSL).

Code division multiple-access or CDMA technique is becoming a popular technology for cellular communications since they allow many users to simultaneously access a given frequency allocation [19].The following sub-section introduces this technique with a brief historical introduction.

1.3.1 Code Division Multiple Access Technique

CDMA was initially used during World War II by English allies to foil German attempts at jamming transmissions. The allies decided to transmit over several frequencies, instead of one, making it difficult for the Germans to pick up the complete signal [20].

Moreover, CDMA is defined as a digital cellular technology that uses spread-spectrum techniques [2] [19]. CDMA provides better capacity for voice and data communications since individual conversations are encoded with a pseudo-random digital sequence [3] thus allowing more subscribers to connect at any given time (asynchronous).

Unlike the above mentioned multiple access techniques (namely the TDMA and the FDMA, which are limited in time duration and frequency band respectively), a Code-Division Multiple Access (CDMA) uses all of the available time-frequency space asynchronously [7] [8] and thereby posses a very high spectral capacity that enables it to accommodate more users per MHz of bandwidth [4] [8] . This appears obviously in Fig.(1.3) below.

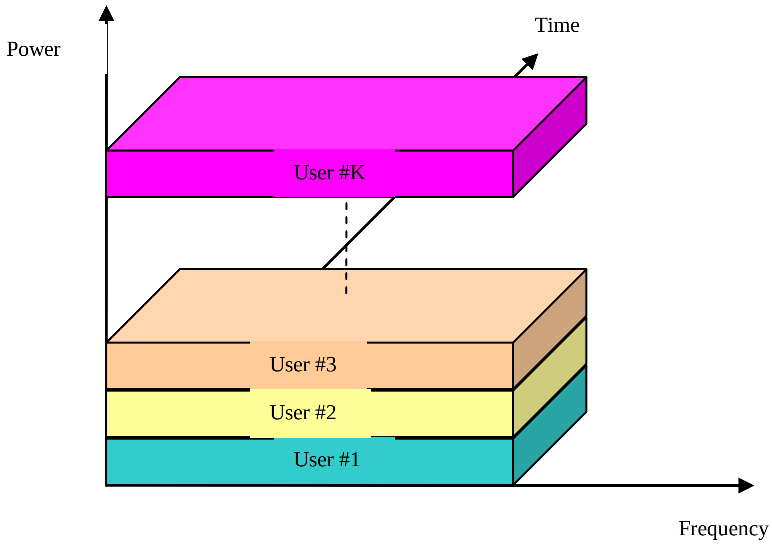

Fig.1.3: CDMA technique

CDMA also have better security and higher data and voice transmission quality because of the spread spectrum technology it uses, which has increased resistance to multipath distortion.

In a Single Carrier (SC) transmission, the data sequence multiplied by a spreading sequence modulates a single carrier [2] . Due to the presence of severe (ISI), the Bit Error Rate (BER) performance of a SC transmission is degraded in frequency selective fading channels. As a result and in a step aiming at improving the BER performance, DS CDMA has been applied and Rake combining is employed.

1.4 Direct-Sequence CDMA (DS-CDMA)

DS-CDMA uses a set of unique spreading sequences (signature waveforms or codes) to modulate the data bits of different users. The signature waveform is a signal which has a much larger bandwidth than the information bearing signal from the user [6] [7] . In particular, it spreads the signal directly by multiplying the data waveform with a user-unique high bandwidth pseudo-noise binary sequence. It follows that CDMA is called a spread spectrum technique [3] . Standards such as IS-95 and W-CDMA are based on CDMA technology). The signature sequences or codes also allow the receiver to demodulate the message transmitted by multiple users of the channel, who transmit simultaneously and generally, asynchronously.

Direct-Sequence Code Division Multiple Access (DS-CDMA) is classified into a synchronous system when time of transmission is known and an asynchronous system without its knowledge. In the asynchronous system, however, each base station has a different and distinct spread code and the mobile station needs to designate that code in the initial cell search.

1.4.1 Spreading Codes

Recall that in a CDMA transmitter the information signal is modulated by the spreading code and in the receiver it is correlated with a replica of the same code. Thus, low cross-correlation between the desired and the interfering users is important to reduce the MAI.

For example, random codes exhibit good autocorrelation properties but worse cross-correlation than the deterministic ones. Spreading codes can be divided into pseudo-noise codes and orthogonal codes.

I) Pseudo-noise sequences

They are generated with a linear feedback shift register generator. The outputs of the shift register cells are connected through a linear function formed by exclusive-or (XOR) logicgates into the input of the shift register as appears in Fig. (1.4). Bits 1, 5, 6, and 7 of the shift-register are XORed together and the result is shifted into the highest bit of the register. The lowest bit, which is shifted out, is the output of the PN generator.

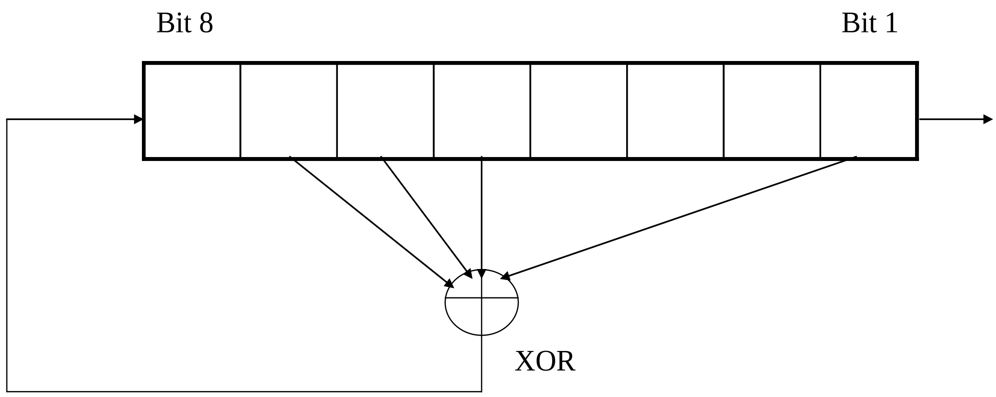

Fig.1.4: PN generator

Three of the basic classes of pseudo-noise sequences are:
_ Maximum length sequences (M-sequences)
_ Gold sequences
_ Kasami sequences

Gold codes and M-sequences are a set of spreading sequences which have the property of very low cross-correlation known binary sequences The Gold sequence having relatively good correlation values is to be used in the simulation protocols of this thesis [21] [22].

II) Orthogonal sequences:
These are completely orthogonal for zero delay but for other delays they have very bad cross-correlation properties [22] [23] . Moreover, they lose their properties when passed through different multi-path channels and thus are suitable only for synchronous applications. The Walsh-Hadamard sequence is an example of orthogonal sequences.

However, the use of orthogonal codes and multiple spreading techniques provide flexible code allocation to the base station and mobile user [24] . As a result of using unique code sequences in the CDMA system it is possible to achieve user separation at the receiver.

1.4.2 General Characteristics of Spread Spectrum (SS) Signals

Spread spectrum signals are characterized by [25]
-Combating or suppressing the determinal effects of interference due to jamming, interference arising from other users of the channel, and self-interference due to multi-path propagation

-Hiding a signal by transmitting it at low power and, thus, making it difficult for an unintended listener to detect in the presence of background noise.

-Achieving message privacy in the presence of other listeners.

1.5 Conclusion

In spread spectrum CDMA all users use the same bandwidth, but each transmitter is assigned a distinct code. Cellular CDMA systems offer the potential of high spectrum efficiency and are proved to be multipath resistant if the receiver structure is appropriately chosen.

As a result, this technology is proposed for second- and especially third-generation cellular mobile systems.

CHAPTER 2

MULTI-CARRIER DS-CDMA SYSTEMS

Recent studies have been shifted from DS-CDMA to Multi-Carrier (MC) transmission techniques to overcome the frequency-selectivity of the fading channel. In this chapter DS-MC-CDMA will be introduced.

However, it is possible to combine Direct-Sequence Code Division Multiple Access with Multi-Carrier transmission [5] to improve the performance of the communication system .The following sections of this chapter will explain this. The system model (transmitter, channel and receiver) will be described and discussed.

2.1 Multi-Carrier Modulation (MCM)

Multi-carrier modulation is a method of transmitting data by splitting it into several components, and sending each of these components over separate carrier signals. The individual carriers have narrow bandwidth, but the composite signal can have broad bandwidth.

When dealing with MC-DS-CDMA systems, the data sequence multiplied by a spreading sequence (direct sequence) modulates several carriers [3] .Thus, a severe frequency-selective fading channel is transformed into several frequency-non-selective flat fading channels, hence avoiding ISI [26] [27].

The advantages of MC modulation are robustness in the presence of multipath fading, a very much reduced system complexity due to equalization in the frequency domain and the capability of narrow-band interference rejection.

2.2 MC-DS-CDMA System Model

Similar to all communication systems, a multi-carrier direct sequence code division multiple access (MC-DS-CDMA) mobile communication system is made up of transmitter, channel (the atmosphere) and receiver.

The system model of a MC-DS-CDMA is illustrated in Fig. (2.1). The data sequences for the $"K"$ users are transmitted through a Rayleigh fading channel. At the receiver, the original transmitted signal at the m th carrier disturbed by Additive White Gaussian Noise (AWGN) proccess $\eta_m(t)$, is passed through the receiver block.

The conventional MC-DS-CDMA based receiver provides a correlator for each carrier and the outputs of the correlators are combined by a maximal ratio combiner. A significant advantage is attained mainly in suppressing the narrow band interference effect and counteracting fading. Thus, there is no need for using the Rake structure [10] or even an interference suppression filter.

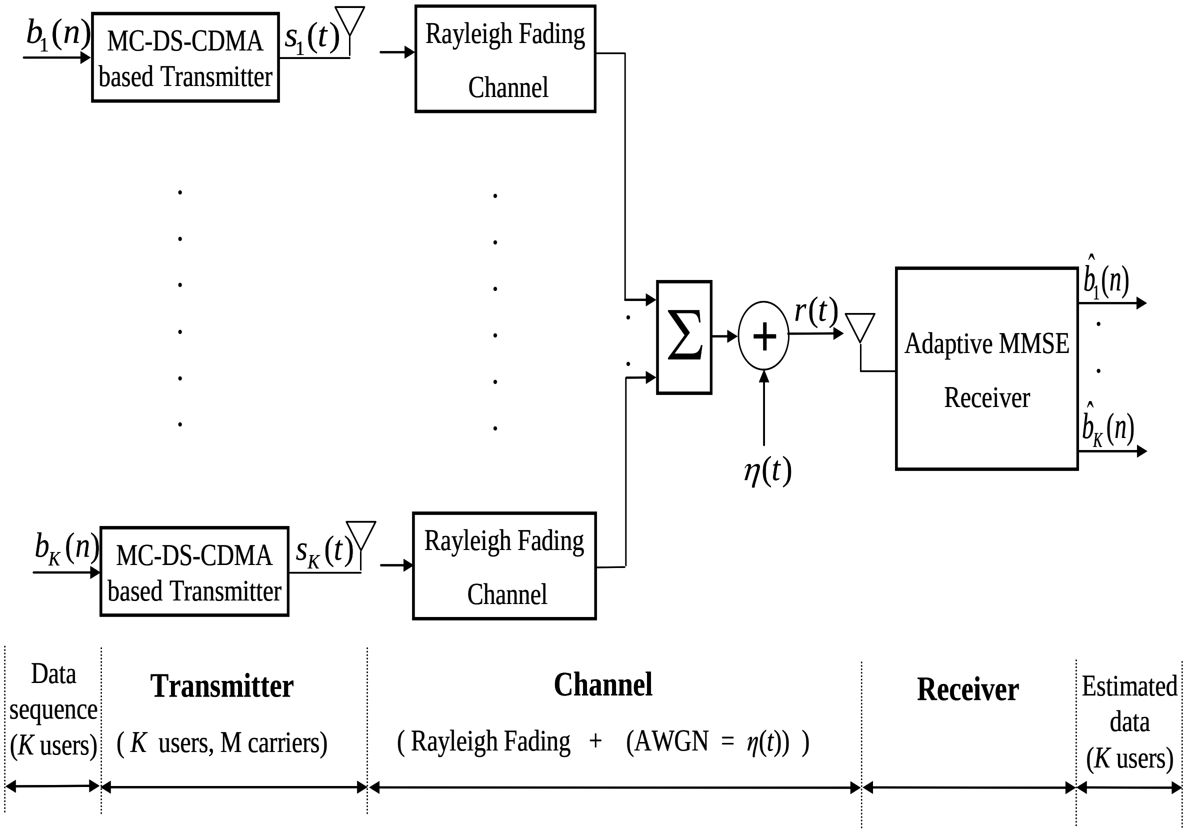

Fig.2.1: MC-DS-CDMA system model.

On the other hand, an obvious disadvantage arises as the Multiple Access Interference (MAI) cannot be suppressed and hence the system is said to be “not near-far resistant”. An alternative approach consists in designing a Minimum Mean Square Error (MMSE) receiver [6] [7] . It is noticed that the data sequences for the *"K"* users are adaptively detected using a minimum mean square error (MMSE) receiver to yield the closest estimates of that data.

2.3 Transmitter Model

In this section the spectrum of a single carrier DS-CDMA is first introduced Fig. (2.2a), followed by that of a multi-carrier DS-CDMA appear in Fig. (2.2b).
The bandwidth is given by

$$BW_1 = (1+\alpha)\frac{1}{T_C}$$

Where $0 < \alpha < 1$ and T_C is the single carrier chip duration. Now, if this spectrum is equally divided into M disjoint frequency bands, a spectrum of a multi-carrier system is obtained.

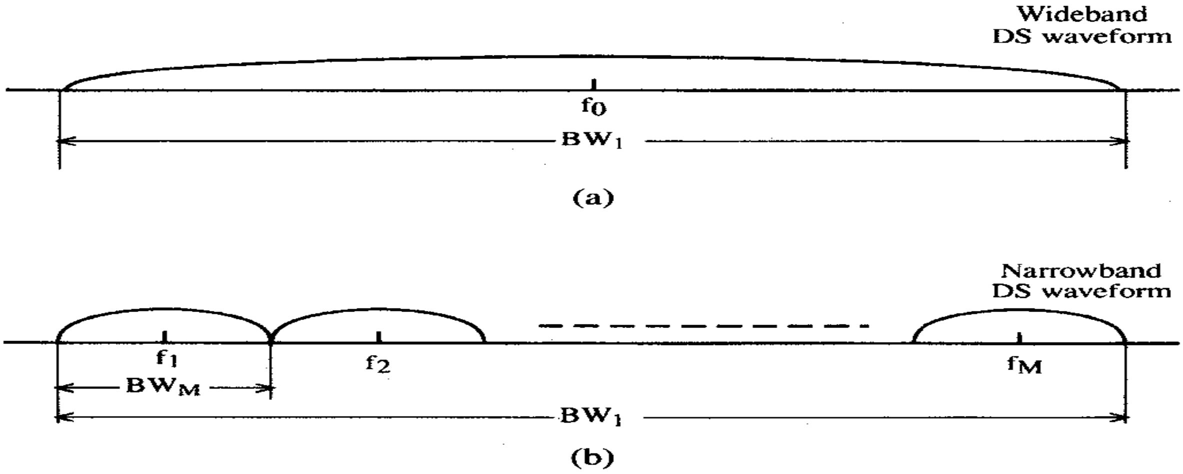

Fig.2.2 :(a)Power spectral density of SC DS waveform,(b) Power spectral density of MC DS waveform

Then the bandwidth of each frequency band can be given by

$$BW_M = (1+\alpha)\frac{1}{MT_C}$$

where MT_C is the multi-carrier chip duration and M is the number of carriers.

An uplink asynchronous MC-DS-CDMA system with Binary Phase Shift Keying (BPSK) modulation is considered based on M carriers and involving K users is shown in Fig. (2.3).

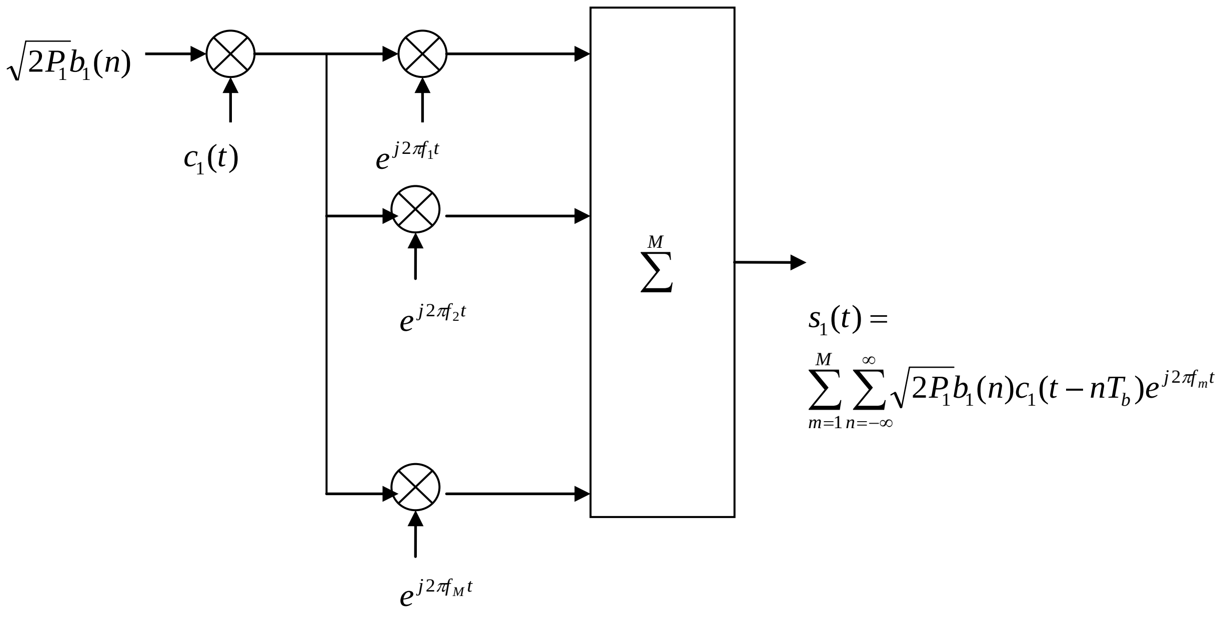

Fig. 2.3: Transmitter model

where: P_1 is the transmitted power of the *1*st user's signal

f_m the m^{th} carrier frequency

$b_1(n)$ is the *1*st user information symbol sequence, chosen independently and equally from $\{-1,+1\}$

T_b is the bit duration.

and c_1 the spreading waveform of the *1*st user.

At the transmitter the information-bearing signal is modulated with a user's unique spreading code. Here, the gold sequence being the considered sequence and the type of modulation considered here is the(BPSK) which is appropriate in applications where phase coherence between the transmitted signal and the received signal can be maintained over a time interval that is relatively long compared to the reciprocal of the transmitted signal bandwidth.

The resulting signal modulates several carrier frequencies and is then transmitted. The transmitted MC-DS-CDMA signal of the k^{th} user can be expressed as follows

$$s_k(t) = \mathrm{Re}\left[\sum_{m=1}^{M}\sum_{n=-\infty}^{+\infty}\sqrt{2P_k}\,b_k(n)c_k(t-nT_b)e^{j2\pi f_m t}\right] \tag{2.1}$$

where:

P_k is the transmitted power for each carrier of the k^{th} user signal

f_m the m^{th} carrier frequency

$b_k(n)$ the k^{th} user information symbol sequence, chosen independently and equally from $\{-1,+1\}$

T_b is the bit duration.

The spreading waveform of the k^{th} user is given by:

$$c_k(t) = \sum_{i=0}^{N-1} a_k(i)\psi(t-iT_c) \tag{2.2}$$

Where:

T_c is the chip duration

$N = T_b / T_c$ is the processing gain,

$a_k(i) \in \{\pm 1/\sqrt{N}\}$ with $i = 0,1,\ldots,N-1$ is the normalized spreading sequence

$\psi(t)$ is the chip pulse shape, assigned to one over the interval $[0,T_c]$ and 0 otherwise.

2.4 Channel Model

The transmitted MC-DS-CDMA signal goes through a frequency selective fading channel. Frequency-selective fading causes different frequencies of an input signal to be attenuated and phase shifted differently in a channel. Frequently, channels experiencing frequency-selective fading may require an equalizer to achieve the desired performance. The fading channel impulse response is given by

$$h(t)=\sum_{l=1}^{L}|h_l(t)|e^{j\phi_l(t)}\delta(t-lT_c) \tag{2.3}$$

where L is the number of resolvable multi-path components. In addition, the l^{th} resolvable path is characterized by its time-varying envelope $h_l(t)$ and its time-varying phase $\phi_l(t)$.
By suitably choosing the number M of carriers, the carrier spacing and the bandwidth of the chip pulse shape, $\psi(t)$, (assigned to one over the interval $[0,T_C]$ and zero otherwise), each carrier can be assumed to undergo independent frequency non-selective flat Rayleigh fading. Therefore, the system will have a frequency diversity gain equal to the number of carriers.
Then the narrowband complex Gaussian channel co-efficient over the m^{th} carrier is given by

$$h_m(t)=|h_m(t)|e^{\phi_m(t)} \tag{2.4}$$

where

- $h_m(t)$ is the random attenuation factor, assumed to be Rayleigh distributed.
- $\phi_m(t)$ is the random phase shift, assumed to be uniformly distributed over $[0,2\pi]$.

In addition to the fading, the transmitted signal is also corrupted by an independent Additive White Gaussian Noise (AWGN) process $\eta(t)$ with variance σ_η^2.

2.5 Conventional Receiver Model

Conventional asynchronous DS-CDMA systems allow each user to transmit and receive independently. Each receiver performs a simple correlation between the received baseband signal and the corresponding user's spreading sequence.

With the knowledge of the unique spreading codes, the receiver can isolate the data corresponding to each user by the process of channel estimation and detection. A conventional correlator based MC-DS-CDMA receiver for user number one proposed by Kondo *et al* [3] appears in Fig.(2.4). The received signal demodulated over M carriers, passes on to a filter having MT_C durations into a sampler that collects N samples and finally into a correlator that correlates it with the spreading code of the first user, c_1.

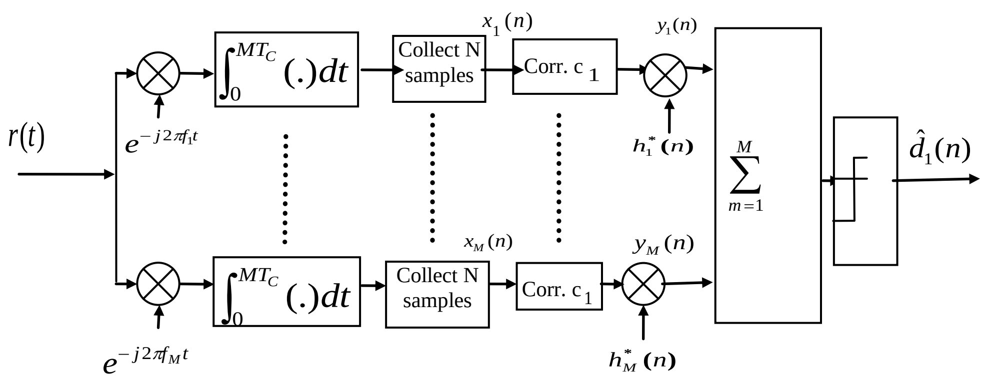

Fig.2.4: Conventional MC-DS-CDMA based receiver for the first user

The resulting vector at the correlator output is then given by

$$y_m(n) = c_1^T x_m(n) = \sqrt{P_1} d_1(n) h_m(n) + \sum_{k=2}^{K} \sqrt{P_k} d_k(n) h_m(n) c_1^T c_k + c_1^T \eta_m(n) \qquad (2.5)$$

where

c_1 is the spreading code of user #1.

P_1 is the transmitted power for user #1.

$d_1(n)$ is the desired output for user #1

$h_m(n) = |h_m(n)| e^{j\phi_m(n)}$, is a fading process with $h_m(n)$ having zero mean, a uniformly distributed phase $\phi_m(n)$ and a Rayleigh distributed envelope $|h_m(n)|$.

$\eta_m(n)$ is an independent Additive White Gaussian Noise (AWGN) process with variance σ^2.

The receiver then re-multiplies with the binary {± 1} pseudo-noise sequence. This effectively, assuming perfect synchronization, removes the pseudo-noise signal and what remains (of the desired signal) is just the transmitted data waveform.

Because other users do not use completely orthogonal spreading codes, there is residual multiple-access interference present at the chip-matched filter output [7] [8] [18] [28]. Due to the non-orthogonality of practical spreading sequences, the conventional correlator receiver suffers from the near–far problem. Figure (2.5) illustrates one simple scenario of the near-far problem. Let user 1 be the desired user, then user 2 is the interfering user. Suppose that both users transmit at the same power level.

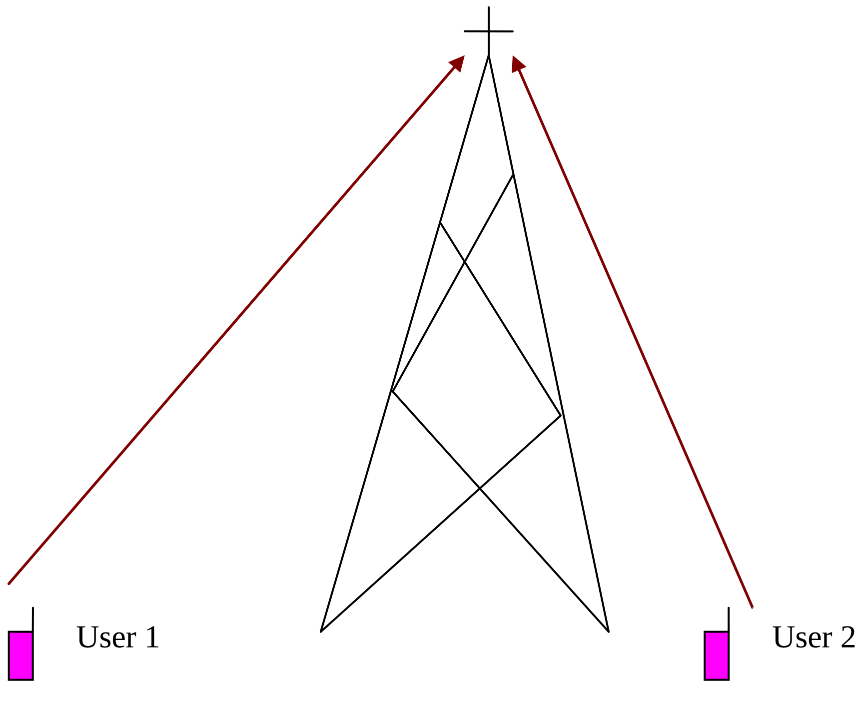

Fig.2.5: Simplest near-far problem

It is well known that the signal power attenuates with respect to distance it travels. Since user 2 is much closer to the base station, its signal power at the base station is stronger than that of user 1. Moreover, the interference caused by user 2 may overwhelm the desired signal of user number 1. As a result, the performance of user 1 is deteriorated.

On the other hand, suppose that user 2 is the desired user. Due to multi-path channel at the base station from the replicas of both users, the signal power of user 2 may be less although it is closer to base station. This multiple-access interference problem causes attenuation on the users' signal power and can be defined as

$$ISR = 10\log_{10}\left(\frac{P_k}{P_1}\right), \qquad 2 \le \mathrm{k} \le \mathrm{K} \tag{2.6}$$

This implies that the cross correlation between the spreading sequence of the user of interest and the signal from a strong interferer can be larger than the correlation with the signal from the desired user. Detection then is rendered unreliable.
Results of simulation in the next section clarify the effect of the near far problem on the BER performance of the conventional match-filter based receiver structure in a MC-DS-CDMA communication system.

The classical way to deal with this is power control, whereby all users' transmittted powers are controlled so that the powers received from all users are equal. However this adds complexity to the system.

Researches thereafter proved that, adaptive Minimum Mean Square Error (MMSE) receivers for Multiple Access (MC-DS-CDMA) mobile communication systems, over frequency selective fading channels, still outer-perform the conventional ones [2] [3] [5] [6] [10] .
The time signal on the systems uplink in its complex analytic form is given by:

$$r(t) = \sum_{k=1}^{K}\sum_{m=1}^{M}\sum_{n=-\infty}^{+\infty}\sqrt{2P_k}\,h_{k,m}b_k(n)c_k(t-nT_b-\tau_k)e^{j2\pi f_m t} + \eta(t) \tag{2.7}$$

where
$\tau_k \in [0, T_b]$ is the delay of the k^{th} user signal
$\{h_{k,m}\}_{k=1,2,\ldots,K;\ m=1,2,\ldots,M}$ are independent and identically distributed (i.i.d.) zero-mean complex Gaussian random variables which model the overall effects of phase shift and fading of the m^{th} carrier for the k^{th} user.

2.6 Simulation results

In this section, the effect of MAI and the near far problem on the BER performance of the conventional receiver of the MC-DS-CDMA communication system will be illustrated. It is assumed that 6 users are using the system.

Three near-far scenarios for six users of the system will be discussed. First, the case in which the desired user (user #1) signal has a power equal to that of all other interfering user signals is studied. Second, the near far scenario1 problem, in which the other interfering users have a 10dB power advantage over the desired user (ISR=10dB) is studied. Finally, the near-far scenario 2 problem in which the interfering users have 1000 power advantage over the desired user is taken into account.

Fig.(2.6) shows the BER performance of the three scenarios mentioned above. It is obvious that the BER of the near-far scenario 2, attains a value of 0.5 (considered the worst behavior) when the interference to signal ratio (ISR) is equal to 12 dB, compared to its value (0.001) in the case of the near-far scenario 1. The best performance, however, is achieved in the case of equal users' power, since the least BER value (0.0001) is attained at ISR=12dB.

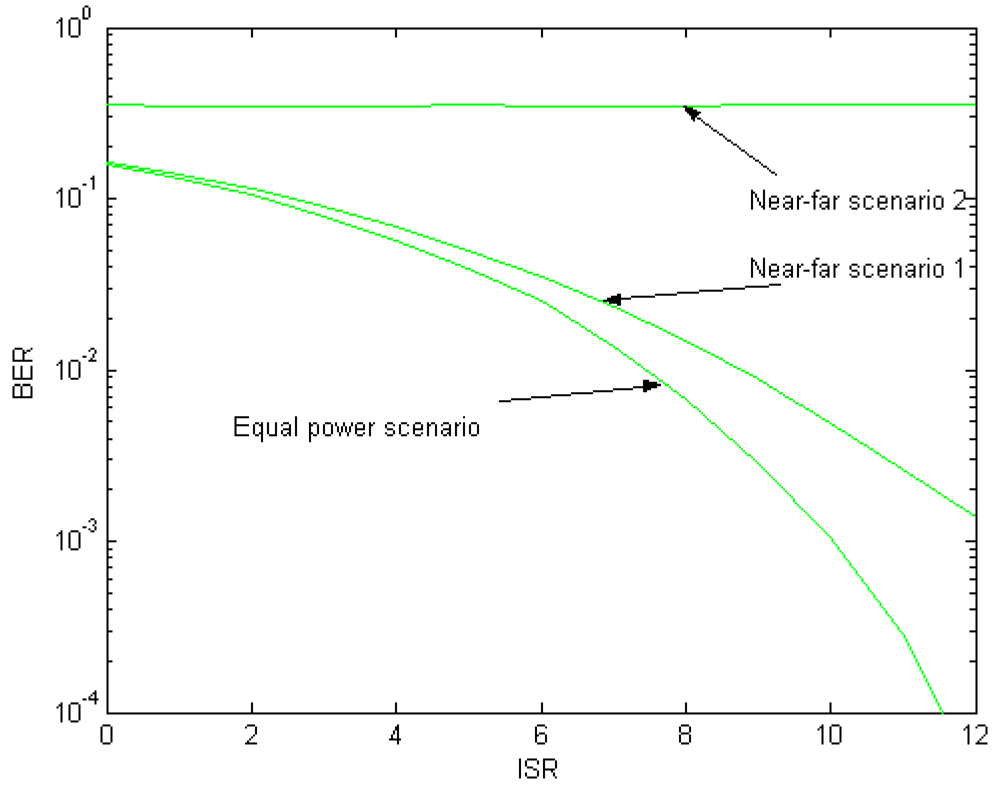

Fig.2.6: BER vs ISR for the three near-far scenarios

2.7 Conclusion

MC-DS-CDMA is being utilized in mobile communication systems over Rayleigh fading channels through the last decades. However, the received signal at the input of the receiver has MAI due fading. Moreover, the signal's power of the desired user is attenuated in strength due to distance away from the base station, a phenomena known as the near-far problem. The conventional matched-filter based receiver structure was proposed to suppress interfering signals and improve BER performance. However, it is considered severely limited by MAI and the near-far problem. An alternative approach using MMSE filter based receivers will be introduced in the following chapter.

CHAPTER 3

DESIGNING ADAPTIVE MINIMUM MEAN SQUARE ERROR (MMSE) RECEIVERS

In this chapter, to suppress the MAI and to mitigate the effect of fading, we propose to study variable step size adaptive filter based MMSE receivers for asynchronous MC-DS-CDMA systems in Rayleigh fading channels.

This chapter is organized as follows. In Section 3.1, the state of the art on receiver design for single and multicarrier DS-CDMA systems is provided. In section 3.2, the optimal MMSE receivers for MC-DS-CDMA systems are reviewed. In section 3.3, two adaptive MMSE receiver structures for MC-DS-CDMA systems are introduced. In section 3.4, two Variable Step-Size adaptive algorithms are implemented to the system. The results are discussed in Section 3.5 followed finally by the conclusion in the last section.

3.1 State of the art on Receiver Design

Adaptive receivers performing single-user detection have many advantages over the conventional receiver [29] , as they suppress interference and perform de-spreading.

Efforts have been made to develop multi-user detection receivers to suppress MAI [5] [7] [8] [18][19][26] and mitigate the near-far problem. The Rake receiver implemented first, could not eliminate MAI and was considered limited by the near-far problem. Later, an optimum multiuser receiver proposed by Verdu consisted of a matched filter for each user solved the near-far problem and completely eliminated MAI but was considered highly computational as the number of users is in increase. A solution to this problem, proposed introducing the de-correlating receiver [29] [31] [32] that completely eliminate MAI and achieve optimal near far resistance with less computational cost, due to processing linear complexity in the number of users. However, it was not suitable for adaptive implementations, a reason that pushed for using an MMSE receiver proven to be as near far resistant as the de-correlating receiver

3.2 Minimum Mean Square Error Receiver

The use of a minimum mean square error receiver allows a tradeoff between complexity of signal selection and receiver complexity in a multi-user environment. Recently, MMSE receivers have been used for MC-DS-CDMA [10] [11].

Here, our purpose is to retrieve the first user signal $b_1(n)$ from the received signal $r(t)$ contaminated by fading, additive white noise and multiple access interference. For that purpose, we assume that synchronization has been achieved with the first user. Therefore, the delay τ_1 of the first user is equal to zero.

Before introducing the MMSE receiver, let us have a look at the mean square error criterion. In the MSE criterion, the tap coefficients (weights) over the m^{th} carrier denoted by $w_m(n)$ of the filter of the receiver are adjusted to minimize the mean square value of the error

$$\varepsilon_m(n) = b_1(n) - w_m^H(n) y_m(n) \qquad (3.1)$$

where $b_1(n)$ is the data information symbol, $\hat{b}_1(n)$ is the estimate of that symbol at the output of the filter and $y_m(n)$ is the received vector. The MSE criterion is defined as the expectation of the error squared

$$MSE_m = E\left[|\varepsilon_m|^2\right] = E\left[\,|b_1(n) - w_m^H(n) y_m(n)|^2\,\right] \qquad (3.2)$$

Then the block diagram of the MMSE receiver is shown in Fig. (3.1).

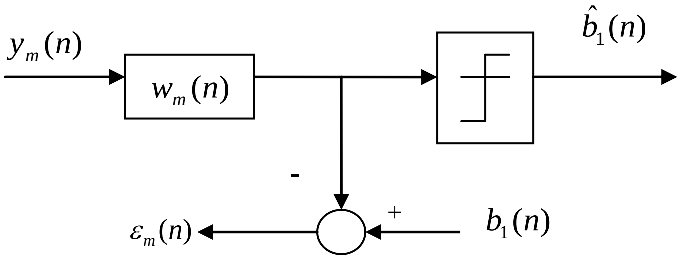

Fig.3.1: MMSE receiver

The optimal receiver comprises a bank of matched filters (MF), one for each user, followed by a Viterbi algorithm for maximum likelihood sequence estimation (MLSE). See Fig. (3.2) below

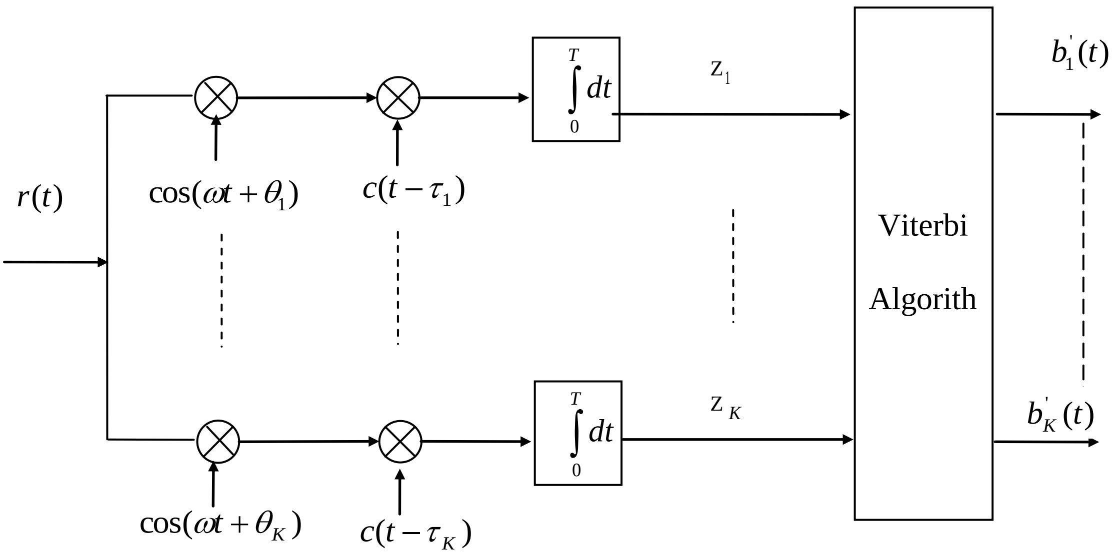

Fig.3.2: Optimum multi-user receiver.

Now, if the squared error signal in equation (3.2) exceeds a certain threshold, a training period or sequence is initiated by the MMSE based receiver. Since it cannot be ruled out that two or more receivers are trained simultaneously, the training sequences have to be user-specific to prevent these receivers from adapting to the same transmitter. The training sequence is applied iteratively or adaptively.

Consequently, this receiver proved better performance when adjusting the filter weights adaptively. As a result, the least minimum error is attained [5] [8] [31] [33][34] [36], without the need to undergo complex computations to solve the resulting equations. Moreover, the linear MMSE [37] receiver has the property that a single user can be detected without having to detect all other users [19][31] .

It follows that the adaptive receiver has the following block diagram Fig.(3.3).

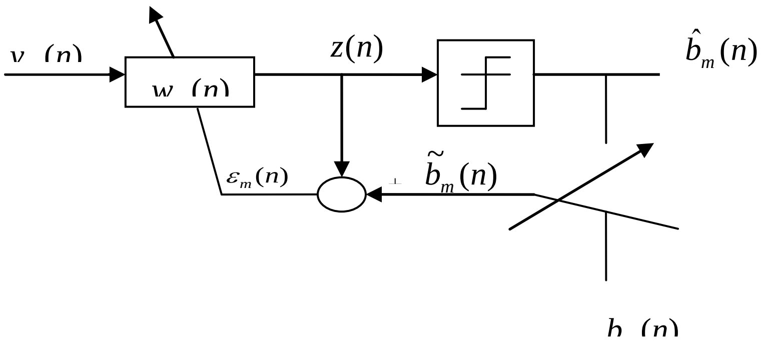

Fig.3.3: Adaptive MMSE receiver.

where

$y_m(n)$ is the received filter input signal

$w_m(n)$ is the filter weight

$b_m(n)$ is the original information data signal

$$\tilde{b}_m(n) = \begin{cases} b_m(n) = \text{training sequence}, & \text{in training mode} \\ \hat{b}_m(n) = \operatorname{sgn}(\operatorname{Re}(z(n))), & \text{in decision mode} \end{cases}$$

$\hat{b}_m(n)$ is the estimated data signal at the output of the filter

and $\varepsilon_m(n)$ is the resultant error.

In the following section two design strategies of the MMSE receivers will be discussed.

3.3 Adaptive MMSE Receivers

The adaptive nature makes it possible to handle time-varying system parameters [34] such as varying channel characteristics and changing number of users to achieve better bit error rates (BER's) and a larger system capacity [38].

Two main adaptive MMSE receivers for detecting asynchronous multi-carrier direct sequence code division multiple access are illustrated [11].

3.3.1 The Separate Detection Structure (SD-MMSE)

Separate detection MMSE design structure consists in considering a particular filter for each carrier Fig. (3.4). $\mathbf{y_1}(n)$

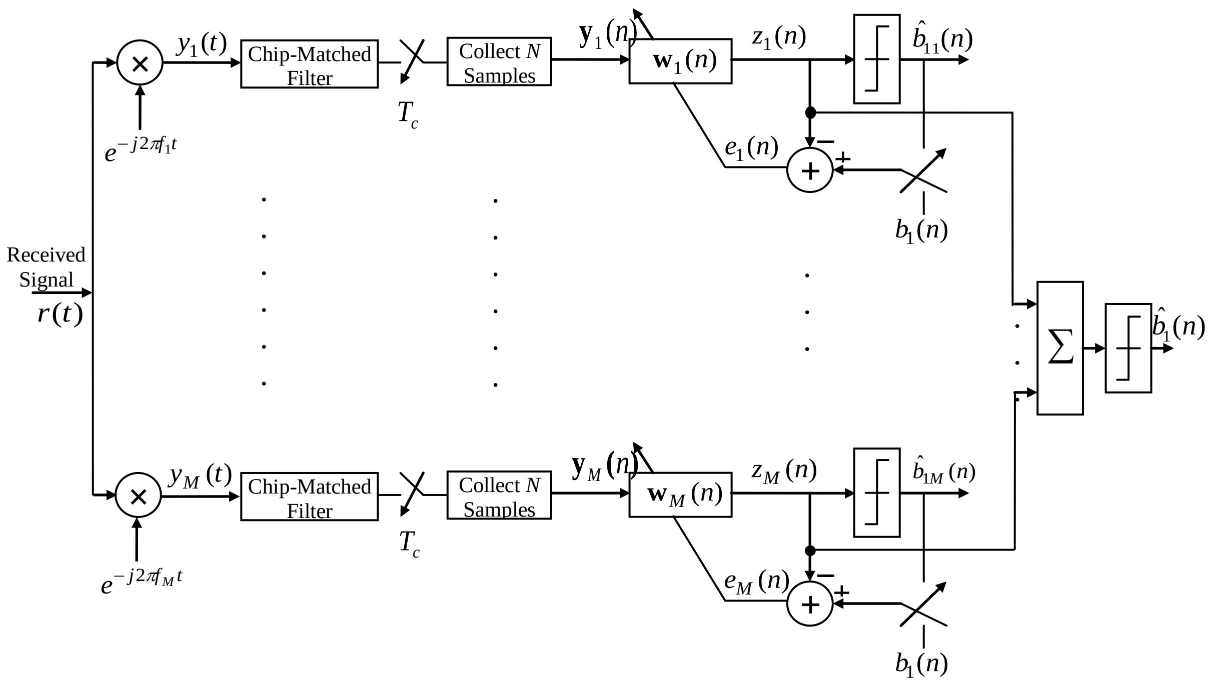

Fig.3.4: Adaptive MMSE Receiver with Separate Detection (SD)

The received signal $r(t)$ is firstly demodulated along each carrier to provide M outputs $\{y_m(t)\}_{m=1,2,\ldots,M}$. Each output is processed by a chip-matched filter and a chip rate sampler. Hence, the discrete time output of the m^{th} carrier during the n^{th} bit interval can be modeled as the following $N\times 1$ vector

$$\mathbf{y}_m(n) = [y_m(1) \quad y_m(2) \quad \cdots \quad y_m(N)]^T \tag{3.3}$$

The MMSE receiver consists in defining the impulse response { $\mathbf{w}_m(n)$ } of the filter:

$$\mathbf{w}_m(n) = [w_m(1) \quad w_m(2) \quad \cdots \quad w_m(N)]^T \tag{3.4}$$

that minimizes the following Mean Square Error (MSE) criterion:

$$MSE_m = E\left[\left|b_1(n) - \mathrm{w}_m^H(n)\mathrm{y}_m(n)\right|^2\right] = E\left[\left|b_1(n) - z_m(n)\right|^2\right] \tag{3.5}$$

This leads to the well known Wiener-Hopf solution

$$\mathbf{w}_m(n) = \mathbf{R}_m^{-1}(n)\mathbf{p}_m(n) \tag{3.6}$$

Where:
$\mathbf{R_m}(n) = E[\mathbf{y_m}(n)\mathbf{y_m^H}(n)]$ denotes the auto-correlation matrix of the m^{th} carrier output vector, and
$\mathbf{p}_m(n) = E\left[\mathbf{b}_1^*(n)\mathbf{y}_m(n)\right]$ is the cross-correlation vector between desired symbol and the m^{th} carrier output vector.

Nevertheless, solving the Wiener-Hopf equation in (3.6), by finding the inverse of R, is considered too cumbersome. Instead, an iterative or adaptive algorithm is used. This adaptive algorithm needs a certain number of iterations to converge to the optimum solution [39], and has the advantage to reduce the computational cost.

3.3.2 The Joint detection Structure (MMSE-JD)

The second design structure is based on a joint structure defined by the concatenation of the adaptive filter weights dedicated to each carrier Fig.(3.5).

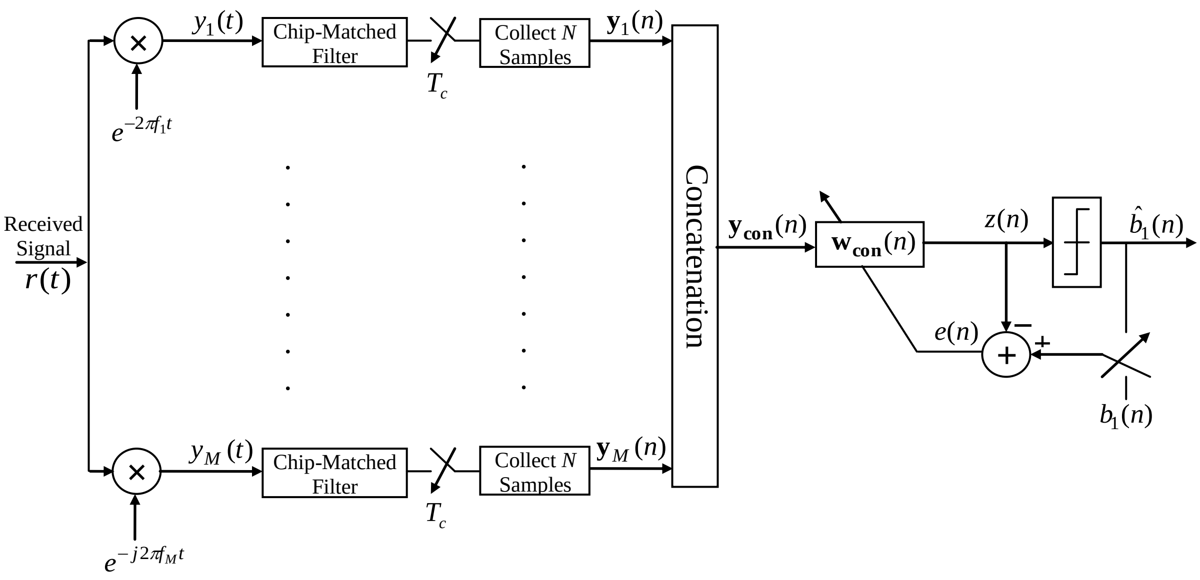

Fig.3.5: Adaptive MMSE Receiver with Joint Detection (JD)

It can be derived by replacing in the above approach:

- $\mathbf{y}_m(n)$ by a vector $\mathbf{y_{con}}(n)$ concatenating the M discrete time outputs

$$\mathbf{y}_{con}(n) = [\ \mathbf{y}_1^T(n) \ \ \mathbf{y}_2^T(n) \ \ \cdots \ \ \mathbf{y}_M^T(n)\ ]^T \tag{3.7}$$

and

- $\mathbf{w}_m(n)$ by a vector $\mathbf{w_{con}}(n)$ concatenating the M receiver filter weights

$$\mathbf{w_{con}}(n) = [\mathbf{w}_1^T(n) \ \ \mathbf{w}_2^T(n) \ \ \cdots \ \ \mathbf{w}_M^T(n)]^T \tag{3.8}$$

Therefore, the criterion can be directly written as follows

$$\hat{b}_1(n) = \mathrm{sgn}\left(\mathrm{Re}\left(\mathbf{w_{con}}^H(n)\mathbf{y_{con}}(n)\right)\right) \tag{3.9}$$

In addition, if a suitable training period is used, the code sequence, the carrier phase and the channel parameters are not necessary at the receiver. Once the training period is over, the filter weights can either be locked for a use in a stationary environment or track channel variations in a decision directed manner.

At that stage, the training sequence $b_1(n)$ is replaced by the estimated data symbol of the desired user along each carrier, obtained as follows:

$$\hat{b}_{1m}(n) = sgn\ \left(Re\ \left(\mathbf{w}_m^T(n)\mathbf{y}_m(n)\right)\right) \tag{3.10}$$

Then, equation (3.5) becomes:

$$MSE_m = E\left[\left|\hat{b}_{1m}(n) - \mathbf{w}_m^T(n)\mathbf{y}_m(n)\right|^2\right] \tag{3.11}$$

The data symbol of the desired user is finally obtained as follows

$$b_1(n) = \mathbf{sgn}\left(\ \mathbf{Re}\left(\sum_{m=1}^{M} w^T(n)y_m(n)\right)\right) \tag{3.12}$$

Adaptive implementation of the MMSE receiver will be discussed in the next section.

3.4 Adaptive Implementations of the MMSE Receivers

The adaptive receivers make it possible to suppress both MAI and Inter-Symbol Interference (ISI) [41][42] and are shown to be near-far resistant [43], since their built in adaptive filters are known to effectively cope with both multiuser interference and multipath propagation in case of static channels [9] [34] .They also offer an attractive trade-off between performance, complexity and the need for side information (spreading codes, timing for all users, channel parameters, etc.).

One way to overcome the problem with the expectation operator that appears in eq. (3.11) is to replace it by a time average leading to Least-Squares Algorithms described in the following sub-sections. It is well known that the MMSE filter can be approximated adaptively by many of these algorithms [40] .

Two fixed step-size algorithms mainly, the Normalized Least Mean Square (NLMS) and the Affine Projection Algorithms (APA), in addition to the Recursive Mean Square (RLS) algorithm are implemented to adaptively adjust the filter coefficients to optimize its performance and to adaptively compensate for time variations in the channel characteristics. The performance characteristics of each algorithm are analyzed including their rate of convergence and computational complexity.

3.4.1 The Normalized Least Mean Square (NLMS) Algorithm

The LMS algorithm is basically a (stochastic) steepest descent method [25] . The major advantage of the steepest descent algorithm lies in its computational simplicity. Thus the main benefits of the LMS algorithm are its simplicity in-addition to its robustness to noise [44] .A significant disadvantage, on the other hand, appears obviously in its slow convergence speed [45] .

The principle of the LMS algorithm is to minimize the MSE by adjusting the vector of filter co-efficients in direction of the negative gradient of the MSE with respect to $w_m(n)$.
Then the weight update equation is given by

$$\mathbf{w}(n+1)=\mathbf{w}(n)+\mu_{LMS}\,\mathbf{y}(n)\boldsymbol{\varepsilon}(n)^{*} \tag{3.13}$$

where

$$\boldsymbol{\varepsilon}(n)=\mathbf{b}(n)-\hat{\mathbf{b}}(n)=\left(\mathbf{b}(n)-\mathbf{w}^{T}(n)\mathbf{y}(n)\right) \tag{3.14}$$

$\mathbf{y}(n)$ is the received filter input signal
$\mathbf{b}(n)$ is the original information data signal
$\hat{\mathbf{b}}\ (n)$ is the estimated data signal at the output of the filter
and μ_{LMS} a small positive constant, called step size, governing the convergence speed.

If μ_{LMS} is selected to be a large value, then the filter will converge faster than when selecting it to be small.
It is well known that when the input signal is highly correlated, i.e., when the co-variance matrix of *y(n)* has a large range of eigen-values, the LMS algorithm can suffer from slow convergence.

Yet, the convergence rate of the filter depends on the power of the received input signal $y(n)$ which varies considerably in the case of fading channels. This led to normalizing the LMS. The normalized LMS (NLMS) utilizes the effective step-size μ, which is the LMS step-size normalized by the input energy [28] to overcome this problem. The update of the filter coefficients is normalized by an estimate of the received input signal power {**y**(n)}. Then, equation (3.13) is given by

$$\mathbf{w}_m(n+1)=\mathbf{w}(n)+\frac{\mu_{LMS}}{\gamma+\left\|\mathbf{y}(n)\right\|^{2}}\mathbf{y}(n)(\,\mathbf{b}_1(n)-\mathbf{w}^{H}(n)\mathbf{y}(n))^{*} \tag{3.15}$$

where:
the effective step size μ is chosen to be $0<\mu_{NLMS}\le 1$, to ensure that the error converges regardless of the energy input signal.
γ is a small number that prevents the denominator from becoming zero if the signal power becomes small and thus increasing the stability of the filter .
In matrix form equation (3.15) can be written as follows

$$\mathrm{w}_m(n+1)=\mathrm{w}(n)+\mu_{LMS}\,\mathrm{y}(n)\left[\mathrm{y}^{T}(n)\mathrm{y}(n)+\gamma\right]^{-1}\mathrm{e}(n)^{*} \tag{3.16}$$

for μ =1, the NLMS algorithm can be viewed as the solution to a least-squares problem that determines $\mathbf{w}(n+1)$ as the vector closest to ***w (n)*** such that

$$\mathbf{w}(n+1)\mathbf{y}(n)=\mathbf{b}(n) \tag{3.17}$$

and can be interpreted as a projection of ***w(n)*** that solves an underdetermined least-squares problem. As a result, the basic form of the Generalized Normalized Least Mean Square (GNLMS) algorithm [46] or the affine projection algorithms (APA) can be simply derived.

These algorithms update the weights on the basis of multiple input signal vectors [47] , while the NLMS algorithm updates the weights on the basis of a single input vector [48] .

3.4.2 The Affine Projection Algorithm (APA)

A generalized form of the NLMS algorithm (GNLMS), known also as the affine projection algorithm will be introduced here. The APA is a better alternative than NLMS in applications where the input signal is highly correlated. APA algorithms offer the capability to trade off performance with computational complexity [49] whereas NLMS updates the weights $\mathbf{w}(n)$ by using the current vector $\mathbf{y}(n)$, APA updates the weights by using the current and the L-1 delayed input vectors $\mathbf{y}(n),\ldots,\mathbf{y}(n-L+1)$, (where L is the number of observers or the block size). By changing algorithm parameters and not fundamental structure, it is possible to improve performance [12][53].

The signal vectors {y(n)} have zero mean and are independent and identically distributed (i.i.d.) with covariance matrix

$$\mathbf{R_m}(n) = E[\mathbf{y_m}(n)\mathbf{y_m^H}(n)] \tag{3.18}$$

If we define a matrix:

$$\mathbf{Y}^H(n) = [\mathbf{y}(n) \;\; \mathbf{y}(n-1) \;\; \ldots\ldots \;\; \mathbf{y}(n-L+1)], \tag{3.19}$$

$$\mathbf{b}(n) = [b_1(n) \;\; b_1(n-1) \;\; \ldots\ldots \; b_1(n-L+1)]^T \tag{3.20}$$

and

$$\mathbf{e}(n) = \mathbf{b}(n) - \mathbf{Y}(n)\mathbf{w}(n) \tag{3.21}$$

the weights $\mathbf{w}(n)$ can be updated as follows:

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu_{APA}\mathbf{Y}^H(n)\left(\gamma_{APA}\mathbf{I}_L + \mathbf{Y}(n)\mathbf{Y}^H(n)\right)^{-1}\mathbf{e}(n) \tag{3.22}$$

where:

- μ_{APA} is the step size
- γ_{APA} a small positive constant used for regularization.
- L is the block size as mentioned in [50].

It should be noted that APA also approximates the RLS algorithm when the block size L approaches the filter length.

3.4.3 The Recursive Least Squares Algorithm

Due to the quicker convergence of the RLS algorithm, it would be attractive to believe that it could also be more effective in tracking non-stationary channel conditions [45] . However, it requires a significantly greater amount of memory and is much more computationally expensive.

The Recursive Least Squares algorithm updates the coefficients to attain the minimization of the squared error signal instead of the expected value of an error signal from the time the filter initiated operation up to the current time.

The coefficients at time n are chosen to minimize the following cost function:

$$J_{LS} = \sum_{k=0}^{n} \lambda^{n-k} |e(n)|^2 = \sum_{k=0}^{n} \lambda^{n-k} \left| b_1(n) - \mathbf{w}^H(n)\mathbf{y}(n) \right|^2 \tag{3.23}$$

The forgetting factor λ is chosen to be $0 < \lambda \leq 1$.

The RLS algorithm operates in three steps at each recursion [25]

Step(1):
$$\mathbf{k}(n+1) = \frac{\mathbf{P}(n)\mathbf{y}(n+1)}{\lambda + \mathbf{y}^H(n+1)\mathbf{P}(n)\mathbf{y}(n+1)} \tag{3.24}$$

Step(2):
$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mathbf{k}(n+1)\left(b_1(n+1) - \mathbf{w}^H(n)\mathbf{y}(n+1)\right) \tag{3.25}$$

Step(3):
$$\mathbf{P}(n+1) = \frac{1}{\lambda}\left(\mathbf{P}(n) - \mathbf{k}(n+1)\mathbf{y}^H(n+1)\mathbf{P}(n)\right) \tag{3.26}$$

where $\mathbf{P}(n+1)$ is the error covariance matrix, $\mathbf{k}(n+1)$ is the filter gain and $\mathbf{P}(0) = \delta^{-1}\mathbf{I}_N$ (δ is a small positive constant).

3.5 Computational Cost of the Various Implementations

In this section, a comparative study is carried out between NLMS, APA and RLS when using SD or JD schemes taking into account the computational cost, the convergence speed and the BER performance. Also different values for the block size (L) are studied. JD structure makes it possible to reduce the computational cost when using NLMS and APA, but this is no longer the case with RLS (See Table 3.1).

Table 3.1: Computational cost for various adaptive filters [11]

Adaptive filters	Separate Detection		Joint Detection	
	Add./Sub.	Mult./Div.	Add./Sub.	Mult./Div.
NLMS	$3NM$	$3NM+M$	$3NM$	$3NM+1$
APA	$NM(L^2+2L+1)$ $+L^2M$	$NM(L^2+2L)$ $+M(L^2+2L+O(L^2))$	$NM(L^2+2L+1)$ $+L^2$	$NM(L^2+2L)$ $+L^2+2L+O(L^2)$
RLS	$3N^2M+3NM+M$	$3N^2M+4NM$	$3N^2M^2+3NM+1$	$3N^2M^2+4NM$

where N are gold code spreading sequences , M is the number of carriers and L is the block size.

3.6 Simulation Results

Simulation Protocol in this test are based on 300 Monte Carlo simulations. The spreading sequences are gold codes. An asynchronous uplink transmission scenario is assumed with 5 interferers having each 10 dB power advantage over the desired user (user k=1). The delays τ_k of the interferers are chosen to satisfy $\tau_k = 3(k-1)T_c, 2 \leq k \leq 6$. We use a moderate value for the number of carriers, i.e. M=6. As a Rayleigh fading is considered, the channel parameters $\{\alpha_{k,m}\}_{k=1,2,...K:m=1,2,...M}$ are generated according to the complex Gaussian distribution with zero-mean and unit-variance.

Two important and essential characteristics of the adaptive algorithms that will be studied are the convergence speed and the bit error rate performance.

In Fig.(3.6) below the three algorithms, the NLMS, APA and RLS are compared according to their convergence characteristics when the JD receiver structure is implemented. The SNR of the desired user is assigned to 20 dB. It is evident that the RLS minimizes the weighted error, therefore achieving optimum learning .It also has a faster rate of convergence compared to LMS-based algorithms. However, it is much more computationally expensive.

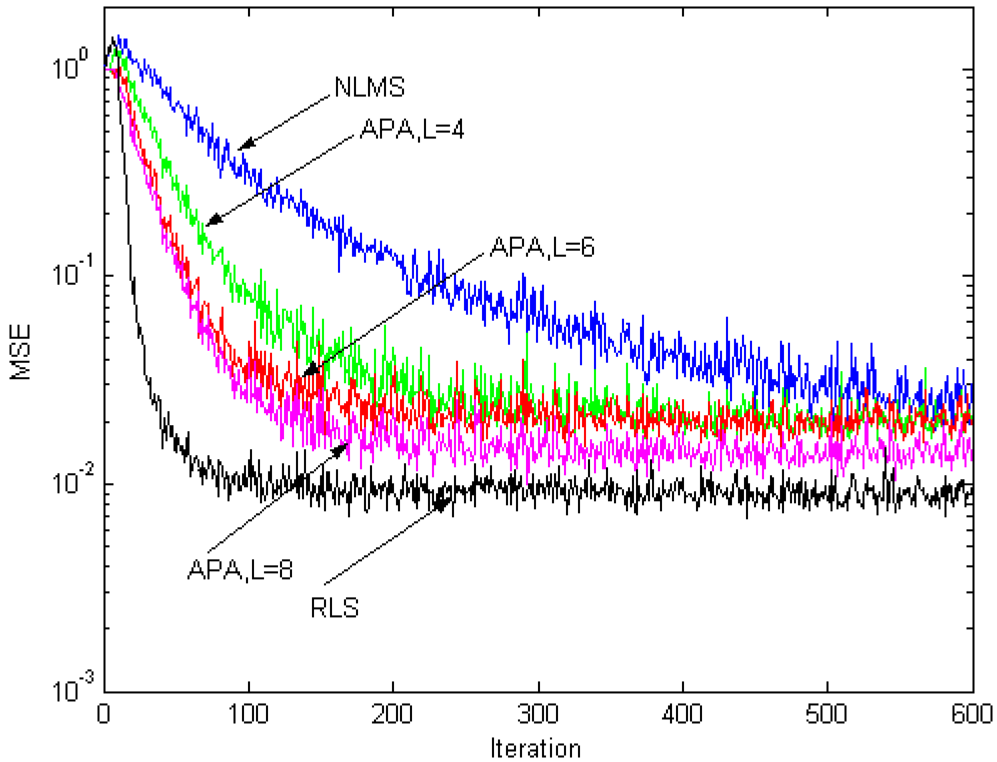

Fig.3.6: Convergence characteristics of NLMS, APA and RLS for the JD receiver structure.

Simulations in this figure confirm that the higher the block-size (L), the higher the convergence speed of the APA algorithm. Moreover, APA converges faster than the NLMS. Satisfactory MSE convergence is observed in around 200 iterations, compared to about 550 for the NLMS.

Next in Fig.(3.7), the BER performance for both the SD and the JD receiver structures (in decision directed mode) is studied when applying the fixed step-size NLMS, the fixed step-size APA and the RLS adaptive algorithms with various Signal to Noise Ratios.

Notice that an increase in the SNR leads to a better BER performance. Again the RLS shows the least BER when applied to the JD receiver structure at a SNR approximately equal to 12 dBs, rather than when applied to the SD structure. Therefore, it is recommended to implement the adaptive algorithms to the JD structure.

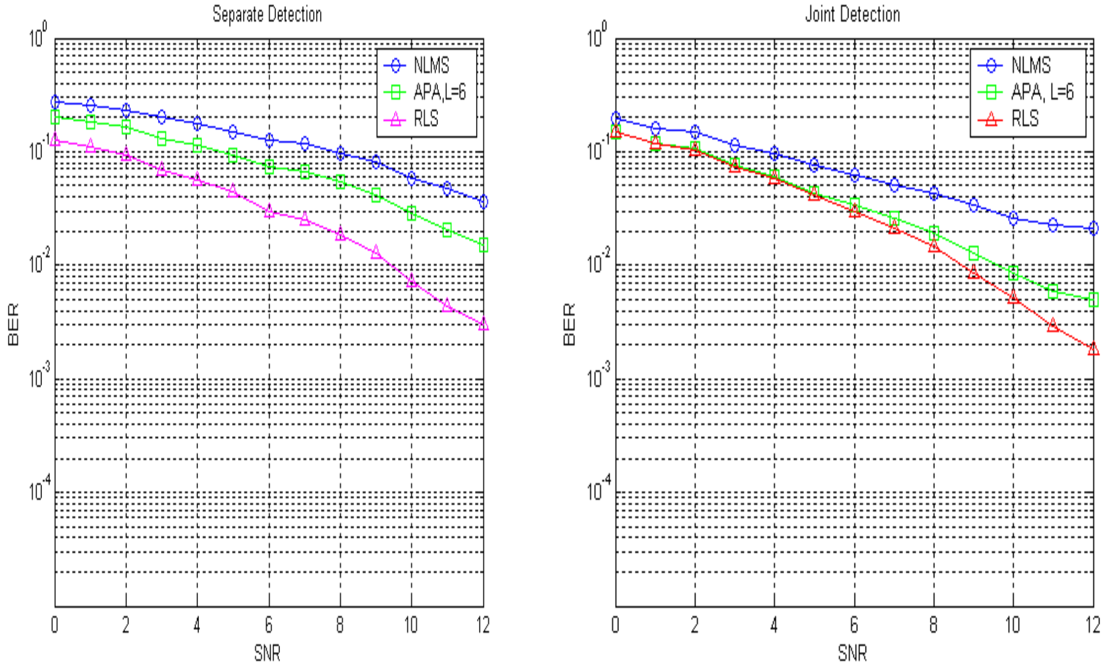

Fig.3.7: BER vs. SNR for the separate detection and joint detection receiver structures with the various adaptive algorithms.

From the above, we conclude that the JD receiver structure with the APA-L=6 applied, makes it possible to achieve a lower BER performance than when the NLMS is applied.

In all the cases studied above, the normalized least mean square (NLMS) and the affine projection (APA), the fixed step-size, μ, governs the rate of convergence and the steady-state excess mean-square error. To meet the conflicting requirements of fast convergence [51] and low steady-state excess mean square error in the NLMS and APA, the concept of Variable Step Size (VSS)[12][13] where the step-size needs to be controlled is proposed.

3.7 Variable Step-Size Adaptive Implementation

According to the recent theoretical and practical work[6] proposed in the previous sections, the step-size (μ) that governs the rate of convergence and the steady-state excess mean-square error needs to be adjusted.

3.7.1 Variable Step-Size Algorithms

Several research papers [9] [13] [34] for proposing different schemes for controlling the step-size where published all of which tend to obtain the optimum value at which the adaptive filter reaches its steady state faster. Two novel algorithms are presented and analyzed. They outperform significantly the existing ones, while maintaining a reduced computational cost and realization simplicity [13].

Some authors worked on varying the step-size in an NLMS based filter design [51][52] , while others achieved progress when varying it in an APA-based one [12] [13].

The relevance of both the VSS-NLMS and VSS-APA is investigated to adaptively implement the SD and JD receiver structures by again taking into account the convergence features and the BER performances.

3.7.2 Optimal Variable Step-Size

To meet the conflicting requirements of fast convergence and low misadjustment error, a variable step-size algorithm [12][13] will be introduced in this section.

Mentioned above the APA updates the weights $\mathrm{w}(n)$ by using the current and the L-1 delayed input vectors $\mathbf{y}(n),\ldots,\mathbf{y}(n-L+1)$.
If we again define a matrix

$$\mathbf{Y}^H(n)=\left[\mathbf{y}(n)\quad \mathbf{y}(n-1)\quad .....\quad \mathbf{y}(n-L+1)\right] \tag{3.27}$$

it follows:

$$\mathbf{b}(n)=\left[b_1(n)\quad b_1(n-1)\ .....\ b_1(n-L+1)\right]^T \tag{3.28}$$

$$\mathbf{e}(n)=\mathbf{b}(n)-\mathbf{Y}(n)\mathbf{w}(n) \tag{3.29}$$

$$\mathbf{w}(n+1)=\mathbf{w}(n)+\mu_{APA}\mathbf{Y}^H(n)\left(\mathbf{Y}(n)\mathbf{Y}^H(n)\right)^{-1}\mathbf{e}(n) \tag{3.30}$$

The update recursion equation (3.30) can be written in terms of the weight error vector

$$\mathbf{w}_e(n+1)=\mathbf{w}^{\circ}(n)-\mathbf{w}(n+1) \tag{3.31}$$

where $\mathbf{w}^{\circ}(n)$ is the unknown weight vector and $\mathbf{w}(n+1)$ is the estimated weight vector.
Then equation (3.30) becomes

$$\mathbf{w}_e(n+1)=\mathbf{w}^{\circ}(n)-\mathbf{w}(n)-\mu_{APA}\mathbf{Y}^H(n)\left(\mathbf{Y}(n)\mathbf{Y}^H(n)\right)^{-1}\mathbf{e}(n) \tag{3.32}$$

But $\mathbf{w}^{\circ}(n)-\mathbf{w}(n)=\mathbf{w}_e(n)$, so equation (3.32) is reduced to

$$\mathbf{w}_e(n+1)=\mathbf{w}_e(n)-\mu_{APA}\mathbf{Y}^H(n)\left(\mathbf{Y}(n)\mathbf{Y}^H(n)\right)^{-1}\mathbf{e}(n) \tag{3.33}$$

Squaring both sides and taking expectations of eq (3.33) yields,

$$E\left[\ \mathbf{w_e}(n+1)\ \right]^2=E\left[\ \mathbf{w_e}(n)-\mu_{APA}\mathbf{Y}^H(n)\left(\mathbf{Y}(n)\mathbf{Y}^H(n)\right)^{-1}\mathbf{e}(n)\ \right]^2 \tag{3.34}$$

$$\begin{aligned}&\mathrm{E}\|\ \mathbf{w_e}(n+1)\ \|^2=\\&E\|\ \mathbf{w_e}(n)\ \|^2+2\,\mu_{APA}\mathrm{Re}\left(\mathrm{E}\left[\mathbf{e}(n)^*\left(\mathbf{Y}(n)\mathbf{Y}(n)^*\right)^{-1}\mathbf{Y}(n)\mathbf{w}_e(n)\right]\right)+\mu_{APA}{}^2\mathrm{E}\left[\mathbf{e}(\mathrm{n})^*\left(\mathbf{Y}(n)\mathbf{Y}^H(n)\right)^{-1}\mathbf{e}(n)\right]\end{aligned} \tag{3.35}$$

Since $E\|w_e(n)\|^2$ is initially zero and if we suppose that $E\|w_e(n+1)\|^2=\Delta(\mu)$, then

$$\Delta(\mu) = 2\,\mu_{APA} Re\left(E\left[\mathrm{e}(n)^*\left(\mathrm{Y}(n)\mathrm{Y}(n)^*\right)^{-1}\mathrm{Y}(n)\mathrm{w}_e(n)\right]\right) + \mu_{APA}{}^2 E\left[\mathbf{e}(n)^*\left(\mathbf{Y}(n)\mathbf{Y}^H(n)\right)^{-1}\mathrm{e}(n)\right] \quad (3.36)$$

maximizing the above equation with respect to "μ" , minimizes the weight error vector, then

$$\mu_{opt}(n) = \frac{ReE\left(\left[\mathrm{e}(n)^*\left(\mathrm{Y}(n)\mathrm{Y}(n)^*\right)^{-1}\mathrm{Y}(n)\mathrm{w}_e(n)\right]\right)}{E\left[\mathrm{e}(n)^*\left(\mathrm{Y}(n)\mathrm{Y}(n)\right)^{-1}\mathrm{e}(n)\right]} \quad (3.37)$$

Assuming the noise $\eta(t)$ is identically and independently distributed and statistically independent of the data vector, and neglecting the dependency of filter coefficient $w_e(n)$ on past noises, $\mu_{opt}(n)$ can be written as follows:

$$\mu_{opt}(n) = \frac{E\left\|\mathrm{w}_e(n)\right\|^2_\Sigma}{E\left\|\mathrm{w}_e(n)\right\|^2_\Sigma + \sigma_\eta^2 Tr\{E[(\mathrm{Y}(n)\mathrm{Y}(n)^*)^{-1}]\}} \quad (3.38)$$

Where:

$E\left\|\mathrm{w}_e(n)\right\|^2_\Sigma = E[\mathrm{w}_e(n)^*\,\mathrm{Y}(n)^*(\mathrm{Y}(n)\mathrm{Y}(n)^*)^{-1}\mathrm{Y}(n)\mathrm{w}_e(n)]$ and *Tr.* is the trace matrix.

Observe that the range space of $\mathbf{Y}(\mathrm{n})^*$ is $R(\mathbf{Y}(\mathrm{n})^*)$, then $\mathbf{Y}(n)^*(\mathbf{Y}(n)\mathbf{Y}(n)^*)^{-1}\mathbf{Y}(n)$ is a projection matrix onto $R(\mathbf{Y}(\mathrm{n})^*)$.

Consequently, if we define

$\mathrm{p}(n) \overset{\Delta}{=} \mathrm{Y}(n)^*(\mathrm{Y}(n)\mathrm{Y}(n)^*)^{-1}\mathrm{Y}(n)\mathrm{w}_e(n)$, then $\boldsymbol{p}(n)$ will be the projection of $w_e(n)$ onto $R(\mathbf{Y}(\mathrm{n})^*)$ and $\left\|\mathbf{p}(n)\right\|^2$ can be written as

$$\left\|\mathrm{p}(n)\right\|^2 = \mathrm{w}_e(n)^*\,\mathrm{Y}(n)^*(\mathrm{Y}(n)\mathrm{Y}(n)^*)^{-1}\mathrm{Y}(n)\mathrm{w}_e(n) \quad (3.39)$$

And the optimum step-size is given by the following equation

$$\mu_{opt}(n) = \frac{E\left\|\mathbf{p}(n)\right\|^2}{E\left\|\mathbf{p}(n)\right\|^2 + \sigma_\eta^2 Tr\{E[(\mathbf{Y}(n)\mathbf{Y}(n)^*)^{-1}]\}} \quad (3.40)$$

Let a positive constant,

$C = \sigma_\eta^2 Tr\{E[(\mathbf{Y}(n)\mathbf{Y}(n)^*)^{-1}]\}$, thus (3.40) will be reduced to:

$$\mu_{opt}(n) = \frac{E\left\|\mathbf{p}(n)\right\|^2}{E\left\|\mathbf{p}(n)\right\|^2 + C} \quad (3.41)$$

Note that, C can be approximated as the block-size *L* over the signal to noise ratio that is used in simulations. Then

$$C = L / SNR$$

Equation (3.41) is the general equation used to calculate an optimum value of the step-size after a number of iterations *n*. It will be used to find both the optimal NLMS and APA step-sizes as will be seen in the following two sub-sections.

3.7.3 Variable Step-Size APA

Recalling equation (3.41), we notice that it is indirectly dependant on the unknown filter coefficients ***w(n)*** or directly depending on the value of *p(n)* which is not available during iterating .However, when $\eta(n) = 0, \mathbf{p}(n) = \mathbf{Y}(n)^*(\mathbf{Y}(n)\mathbf{Y}(n)^*)^{-1}\mathbf{e}(n)$ and the expectation even with the presence of noise is given by

$$E[\ \mathbf{p}(n)] = E[\ \mathbf{Y}(n)^{*}(\mathbf{Y}(n)\mathbf{Y}(n)^{*})^{-1}\,\mathbf{e}(n)] \tag{3.42}$$

In addition, $\hat{\mathbf{p}}(n)$ is updated as follows:

$$\hat{\mathbf{p}}(n) = \alpha\hat{\mathbf{p}}(n-1) + (1-\alpha)\mathbf{Y}^{H}(n)\left(\gamma\mathbf{I}_{L} + \mathbf{Y}(n)\mathbf{Y}^{H}(n)\right)^{-1}\mathbf{e}(n) \tag{3.43}$$

where

$0 \le \alpha < 1$ is a smoothing factor

and

γ is a small positive constant used for regularization.

Substituting $E\|\mathbf{p}(n)\|^{2}$ by $\|\hat{\mathbf{p}}(n)\|^{2}$, equation (3.41) becomes

$$\mu_{opt}(n) = \frac{\|\hat{\mathbf{p}}(n)\|^{2}}{\|\hat{\mathbf{p}}(n)\|^{2} + C} \tag{3.44}$$

Then the following weight-update equation in the case of affine projection algorithm

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu_{APA}\,\mathbf{Y}^{H}(n)\left(\gamma\mathbf{I}_{L} + \mathbf{Y}(n)\mathbf{Y}^{H}(n)\right)^{-1}\mathbf{e}(n) \tag{3.45}$$

yields

$$\mu_{APA}(n) = \mu_{max}\,\frac{\|\hat{\mathbf{p}}(n)\|^{2}}{\|\hat{\mathbf{p}}(n)\|^{2} + C} \tag{3.46}$$

Where $\mu_{\max} < 2$ is the maximum allowable step size that guarantee filter stability and C is a positive constant that can be approximated by $C \cong L/\mathrm{SNR}$.

It should be noted that, when $\|\hat{\mathbf{p}}(n)\|^{2}$ is large $\mu_{APA}(n)$ tends to $\mu_{\max}$. On the other hand, when $\|\hat{\mathbf{p}}(n)\|^{2}$ is small the step size will be small. Therefore, depending on $\|\hat{\mathbf{p}}(n)\|^{2}$, the step size $\mu_{APA}(n)$ varies between 0 and $\mu_{\max}$.

3.7.4 Variable Step-Size NLMS (VSS-NLMS)

As a special case of the VSS-APA, the VSS-NLMS algorithm can be obtained by setting L=1[9]. Thus, the weight update equation of the VSS-NLMS can be written as follows

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \frac{\mu_{NLMS}(n)}{\gamma + \|\mathbf{y}(n)\|^{2}}\,\mathbf{y}(n)\,e^{*}(n) \tag{3.47}$$

Where

$$\mathbf{e}(n) = \mathbf{b}_{1}(n) - \mathbf{w}^{H}(n)\mathbf{y}(n) \tag{3.48}$$

The variable step size $\mu_{NLMS}(n)$ is now updated as follows:

$$\mu_{NLMS}(n) = \mu_{max}\,\frac{\|\hat{\mathbf{p}}(n)\|^{2}}{\|\hat{\mathbf{p}}(n)\|^{2} + C} \tag{3.49}$$

$$\hat{\mathbf{p}}(n) = \alpha\,\hat{\mathbf{p}}(n-1) + (1-\alpha)\,\frac{\mathbf{y}(n)\mathbf{e}^{*}(n)}{\gamma + \|\mathbf{y}(n)\|^{2}} \tag{3.50}$$

again, $C \cong 1/SNR$

3.8 Simulation Results

In this section we carry out a comparative simulation study between the various adaptive filters (NLMS, APA, VSS-NLMS, VSS-APA) when used to implement the SD and JD receiver structures. In addition, these adaptive implementations are compared with the conventional correlator based receiver [3] .These receivers are compared in terms of the convergence features and the BER.

The advantages and the performance of the proposed techniques, along with a variety of characteristics are demonstrated by means of Monte Carlo simulations. The simulation held here is done by the use of the mathematical program Matlab. More particularly, we carry out a comparative study between the various adaptive filters (NLMS, APA, VSS-NLMS and VSS-APA) taking into account the convergence features and the bit error rate performance. Again, the effect of the active number of users and inter-symbol interference on the BER performance will be studied.

This test is based on 300 Monte Carlo simulations. The spreading sequences are gold codes of length *N*=31. An asynchronous uplink MC-DS-CDMA system is considered, with *K*=6 multiple-access active users (five interfering users $2 \le k \le 6$ plus the desired user $k = 1$).

The interfering users have each a 10 dB power advantage over the desired user. The delays $\{\tau_k\}_{k=1,2,\ldots,K}$ of the users are chosen to satisfy $\tau_k = 2(k-1)T_c$, $1 \le k \le 6$. Here, a moderate value for the number of carriers (i.e., *M*=3) is used as adopted by the third generation CDMA2000 standard. As a Rayleigh fading is considered, the channel coefficients $\{\beta_{k,m}\}_{k=1,2,\ldots,K;\ m=1,2,\ldots,M}$ are generated according to the complex Gaussian distribution with zero-mean and unit-variance.

In Fig. (3.8) the convergence characteristic of NLMS, VSS-NLMS, APA and the VSS-APA for the joint detection receiver structure are shown with the SNR of the desired user assigned to 20dB. Note that when implementing the NLMS algorithm, the mean square error converges to 0.08 only after 600 iterations. However, varying the step-size of this algorithm, i.e., implementing the VSS-NLMS, faster convergence to the 0.08 MSE is reached after 300 iterations only, thus reducing the time of convergence to the half.

Observe that the APA with the block-size L=6, converges to 0.03 after 200 iterations while VSS-APA with L=6 also, converges to the same value of the MSE after only 150 iterations.

Moreover, the MSE converges to 0.02 after 100 iterations and to 0.01 when iterating 180 times thus recording the fastest convergence speed and the least MSE. Indeed, satisfactory MSE is observed in around 150 iterations for VSS-APA, compared to about 500 for the VSS-NLMS.

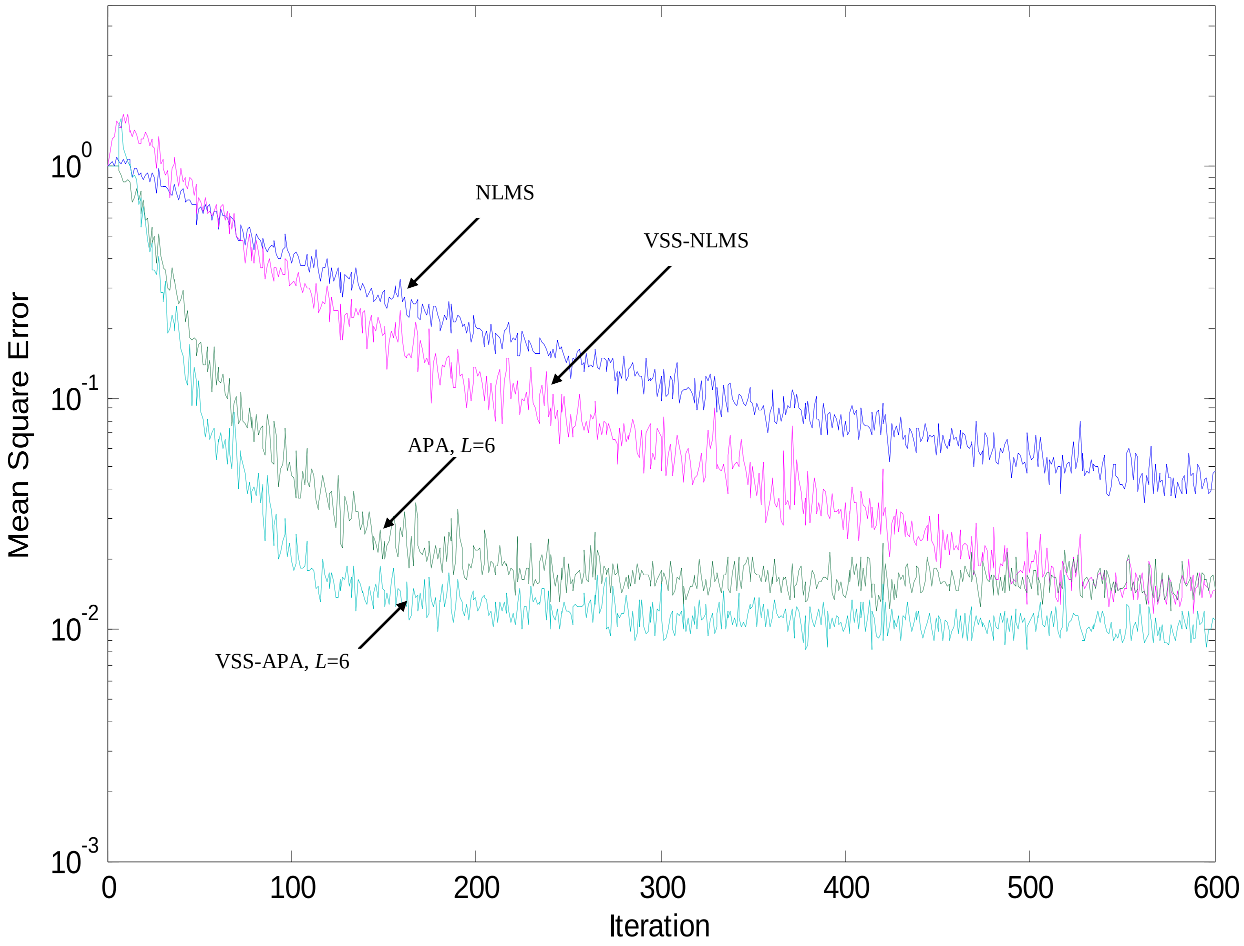

Fig. 3.8: Convergence characteristics of NLMS, VSS-NLMS, APA and VSS-APA for the JD receiver structure.

This means retrieving the closest replica of the original transmitted signal with the least error in the least time or in the highest speed.

The convergence characteristic is handled again in Fig. (3.9) below, but this time when different block sizes of the VSS-APA, using the JD structure. The SNR is assigned to 20 dB.

It is noticed that the VSS-NLMS convergence speed is the slowest, since it needed 600 iterations to reach approximately a 0.02 mean square error value. Regarding the VSS-APA, specifically when only four observers are there,(L=4), converging to the 0.02 occurred after 200 iterations only, thus achieving a higher convergence speed , thereby reducing the time of convergence to the third (200/600).

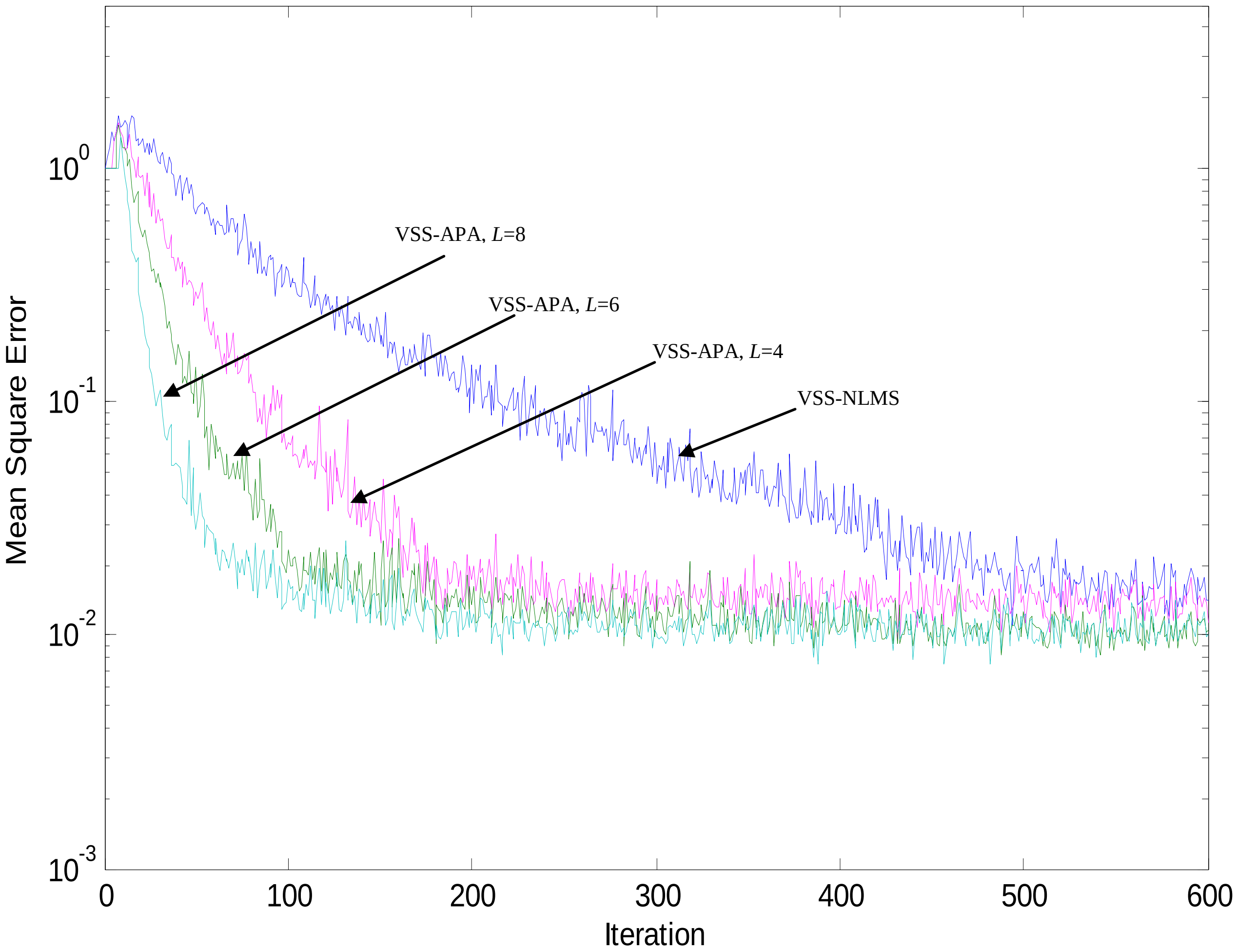

Fig.3.9: Convergence features of the VSS-APA with different values of L as compared with the VSS-NLMS.

However, when L is chosen to be six observers, the VSS-APA converges to 0.02 after 150 iterations, while when considering L=8, this value is reached after 80 iterations. Moreover, the VSS-APA (L=8), converges to 0.01(least square error) value after being iterated for only 180 iterations. Therefore, from this figure, it is quite clear that, the higher *L*, the higher the convergence speed of the VSS-APA algorithm.

Fig.(3.10) below, illustrates the BER performance of the single detection (SD) receiver structure with the various adaptive algorithms. The SNR is studied during the period 0 to 20 dBs. Through the period 0 to 20 dBs, the VSS-NLMS, showed better BER performance compared to the fixed NLMS. Whereas, during the same period, the VSS-APA(L=6) algorithm assured the best and least BER performance compared to the APA(L=6), VSS-NLMS and the NLMS algorithms.

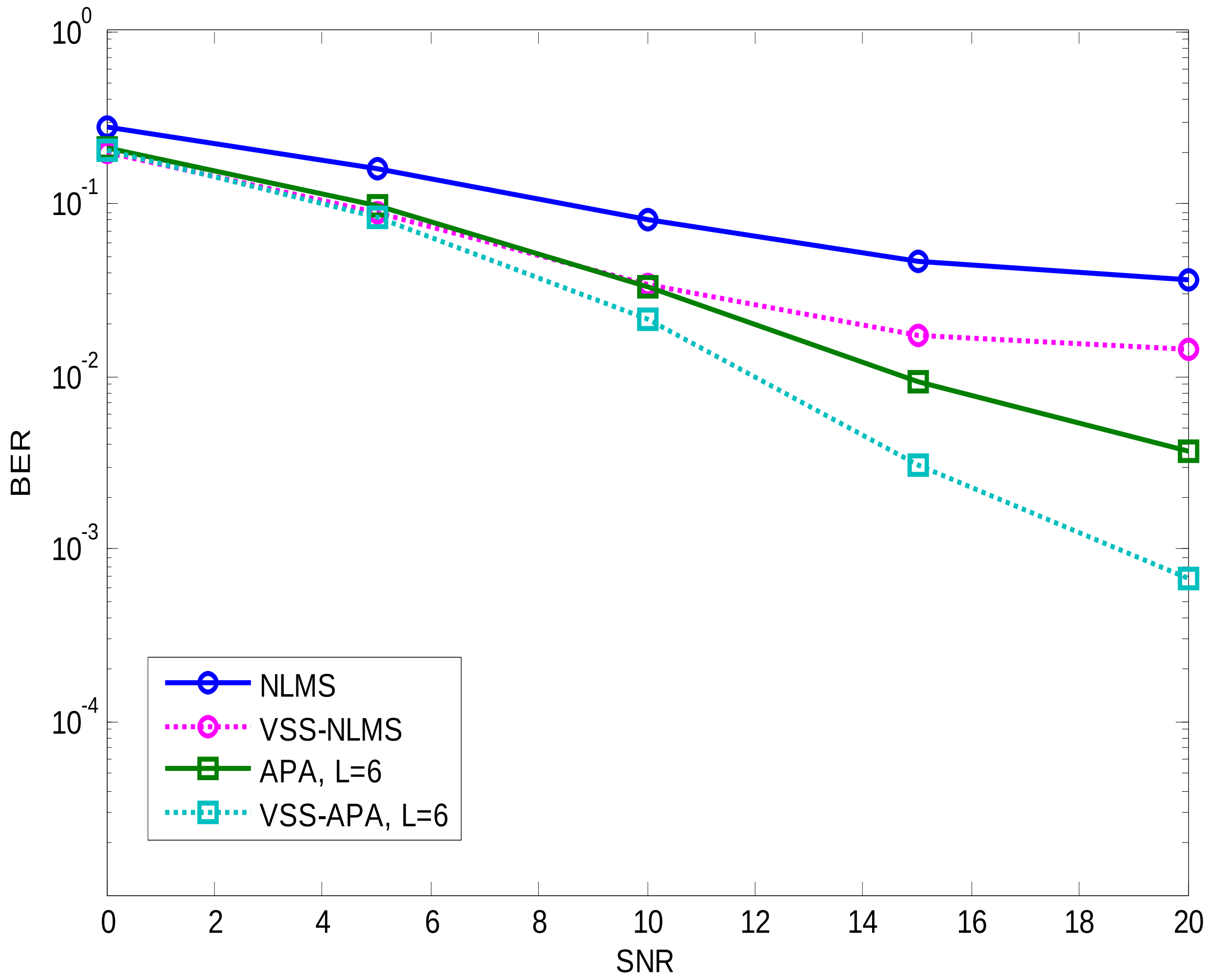

Fig.3.10: BER performance of the SD receiver structure with the various adaptive filters.

Now let us apply the four adaptive algorithms (the NLMS, the VSS-NLMS, the APA(L=6) and the VSS-APA(L=6)) to the JD receiver structure and observe the results in Fig. (3.11).

Notice that again the VSS-APA with L=6, shows the least BER (10^{-5}) at SNR=20, while when the fixed step-size APA with L=6 is applied the BER is 0.001. Moreover, the VSS-NLMS outer-performs the fixed step-size NLMS.

Next, the near-far problem and the performance of an MMSE adaptive MC-DS-CDMA communication system, with the presence of Interference to signal ratio (ISR), will be examined. The conventional correlator based receiver, the VSS-APA-SD receiver structure and finally, the VSS-APA-JD receiver structure are the three cases for which the BER performance is to be studied. Fig. (3.12) shows the interference to signal ratio ranges from 0 to 20dBs, the number of users is chosen to be six and the SNR of the desired user is assigned to 15 dB.

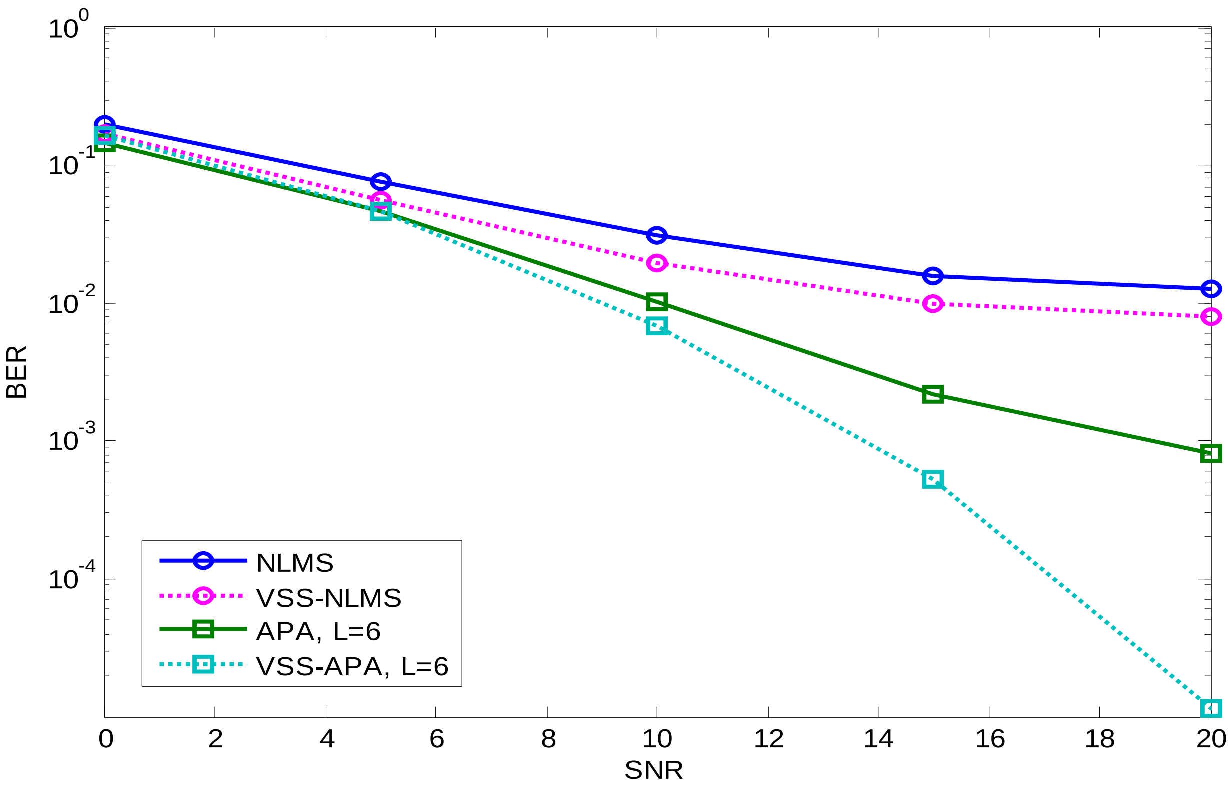

Fig.3.11: BER performance of the JD receiver structure with the various adaptive filters.

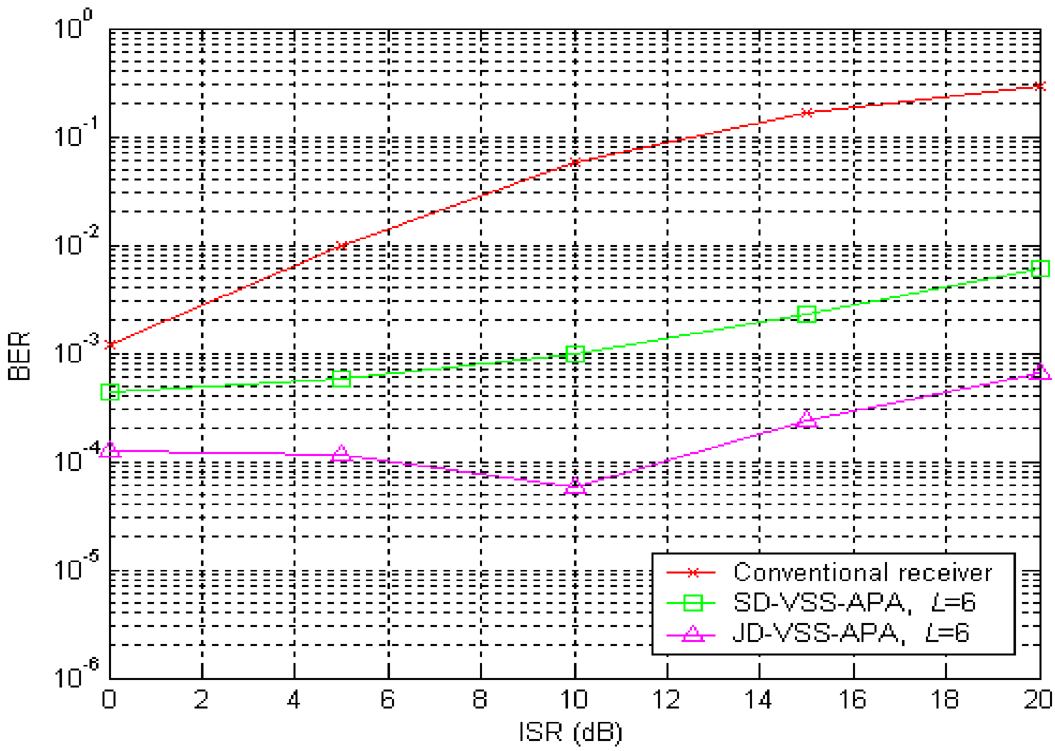

Fig.3.12: BER vs. ISR for the conventional correlator based receiver, APA with Separate Detection, APA with Joint Detection.

Again the VSS-APA-JD based receiver structure showed the least BER in the decision directed mode after training with 500 symbols. It attained a 0.0008 error rate value compared to 0.008 to that of the SD-VSS-APA and a 0.3 to the conventional correlator based receiver.

In order to observe the effect of the number of active users of the MC-DS-CDMA communication system on the BER performance when the VSS-APA algorithm is applied to both the VSS-APA-SD and the VSS-APA-JD, in addition to the conventional correlator based receiver, simulation has been run after training with 500 symbols and 10000 Monte Carlo loops, Fig. (3.13) was obtained.

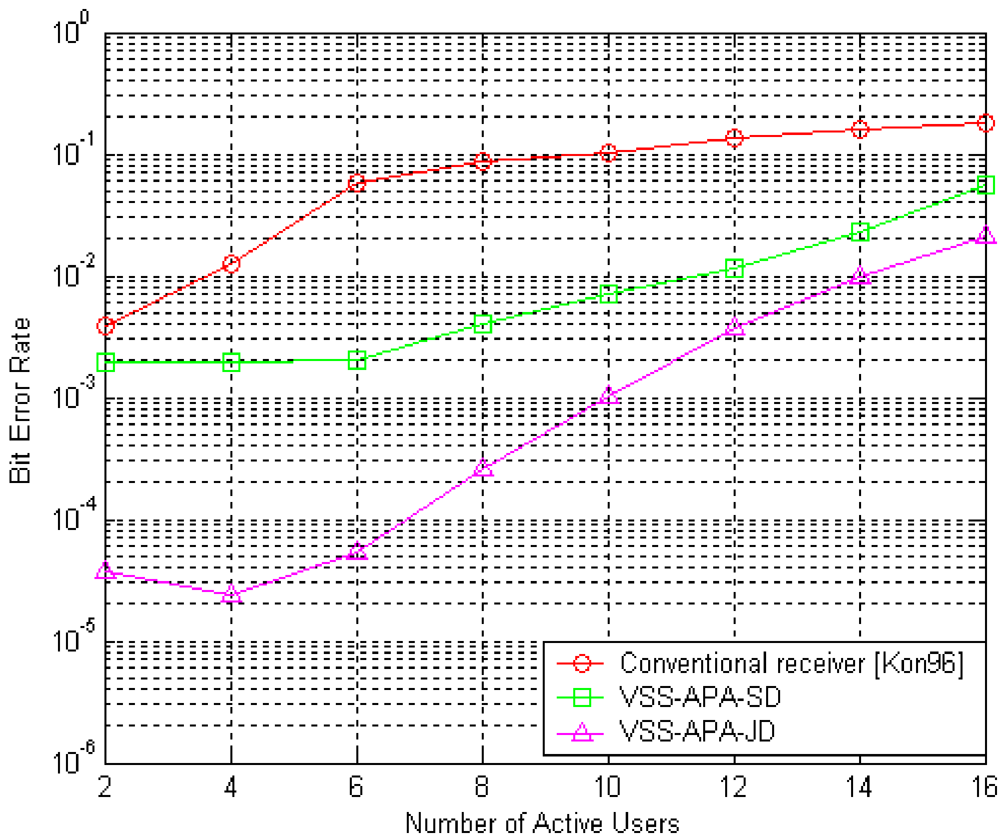

Fig.3.13: BER vs. Number of active users for the conventional correlator based receiver, APA with Separate Detection, APA with Joint Detection

It is noticed that the conventional correlator based receiver proposed by Kondo [3] attained a high bit error rate (0.2) while the VSS-APA-SD based receiver ended at 0.05 BER value with 16 active users of the system but still the VSS-APA-JD recorded the least BER (0.02) value among the mentioned receivers.

3.9 Conclusion

Simulation results have proved that the VSS-APA implemented to the JD receiver structure outer-performed the VSS-NLMS algorithm by achieving better BER performance. Therefore, it is recommended to use it when designing receiver structures in MC-DS-CDMA mobile communication systems. Nevertheless, the computational complexity is considered high.

CHAPTER 4

CONCLUSION AND FUTURE WORK

In this chapter, the conclusions of the work introduced and the work proposed in the previous chapters will be verified in the first section. Moreover, a comparison among the various receiver structures is held. In the second section the relevance to the future work will be introduced.

4.1 Conclusion

Over the last decades, efforts have been made to improve the performance of MC-DS-CDMA mobile communication systems. Since the transmission is wireless, the original signal is exposed to multi-path fading and MAI while passing through a Rayleigh fading channel in its way to the receiver.

In particular, researches on receiver-design have been elaborating methods to retrieve a replica of the transmitted signal with the least MSE by suppressing MAI, mitigating the near-far problem and compensating for multi-path fading.

The so-called Rake receiver cannot eliminate the MAI and, hence, it is limited by the near-far effect. The optimal multi-user detector that consisted of a matched filter for each user, followed. It solved the near-far problem and completely eliminated MAI. However, it was considered highly computational as the number of users of the system is increasing. A solution to reduce this complexity led to use a de-correlating receiver. However, it is not suitable for adaptive implementations. As an alternative, MMSE receivers can be considered and can be implemented adaptively.

Optimal adaptive MMSE receivers succeeded in reducing MAI and compensating for the effect of fading. However, they have high computational cost. To reduce the computational cost, two adaptive MMSE receiver structures are considered. The so-called separate detection structure uses an a separate adaptive filter along each carrier while the so-called joint detection structure uses one adaptive filter to jointly detect the transmitted symbol over all carriers.

Using adaptive filtering algorithms to implement the MMSE receivers is appealing to trade performance with complexity. A number of algorithms, mainly, the NLMS, APA and the RLS were implemented. It is confirmed that the JD receiver based structure outperforms the SD receiver based one in terms of BER performance and convergence features. However, the step-size in these adaptive algorithms should be controlled to meet the conflicting requirements of fast convergence and low steady state excess mean square error. Therefore, we propose to investigate the relevance of Variable Step Size (VSS) NLMS and APA adaptive filters. Simulation results show that the VSS-NLMS and VSS-APA based receivers have, respectively, better performance than those based on fixed step size NLMS and APA, in terms of convergence features and BER.

4.2 Future Work

Throughout the chapters of this thesis, various receiver structures were investigated. The aim accompanied by the implementation of each of these structures was to improve the performance of the MC-DS-CDMA communication system in Rayleigh fading channels. All approaches faced the following problems:

1-Reducing the near-far problem
2-Suppressing the MAI
3-Reducing computational complexity

Receiver designers solved part of these problems and made it possible to trade-off performance with complexity.

Affine projection algorithm is proven to achieve the best performance compared to the traditional LMS and RLS algorithms. In particular, it has been shown through our work that applying variable step size to the NLMS and the APA algorithms without a change in the main structure, led to better performance in terms of convergence features and BER.

It is confirmed that the APA can trade performance with complexity by changing the scalable parameters L (the block size) and μ (the step-size). To reduce farther the computational cost, our future work will focus on investigating the use of low-complexity adaptive algorithms as in [53] in our algorithms to update filter co-efficient.

With the fact that APA algorithms are row projection methods with the advantage of reducing complex computational processes to linear computations, it is possible to combine them with data-reusing strategies [47] to derive a new class of data-reusing adaptive algorithms for updating filter coefficients. Moreover data reusing strategies are easily implemented. By repeatedly reusing projection matrices and reusing the output data for a number of times, we postulate that it is possible to improve performance.

REFERENCES

[1] Hara S. and Prasad R., "Overview of multicarrier CDMA", *IEEE Communication Magazine*, pp. 126-133, Dec. 1997.

[2] Hanzo L., Yang L., Kuan E. and Yen K., *Single and Multi-carrier DS-CDMA*, John Wiley and Sons, 2003.

[3] Kondo S. and Milstein L., "Performance of multicarrier DS-CDMA systems", *IEEE Trans. Communications*, vol.44, n^{o}2, pp. 238-246, Feb. 1996.

[4] *CDMA2000, Physical layer standard for CDMA2000 spread spectrum systems*.Version1, 2001.

[5] Woodward G. and Vucetic B., "Adaptive detection for DS-CDMA", *Proceedings of the IEEE*, vol. 86, n^{o}7, pp. 1413-1434, July 1998.

[6] Miller S., Honig M. and Milstein L., "Performance analysis of MMSE receivers for DS-CDMA in frequency-selective fading channels", *IEEE Trans. Communications*, vol. 48, n^{o}11, pp. 1919-1929, Nov. 2000.

[7] Madhow U. and Honig M., "MMSE Interference suppression for direct-sequence spread-spectrum CDMA", *IEEE Trans. Communications*, vol. 42, n^{o}12, pp. 3178-3188, Dec. 1994.

[8] Verdu S., *Multiuser Detection*. Cambridge, U.K., Cambridge University press, 1998.

[9] Haykin S., *Adaptive Filter Theory*, Prentice-Hall, 2002.

[10] Miller S. and Rainbolt B., "MMSE detection of multicarrier CDMA", *IEEE Journal on Selected Areas in Communications*, vol. 18, pp. 2356-2362, Nov. 2000.

[11] Jamoos A., Grivel E. and Najim M., "Designing adaptive filters-based MMSE receivers for asynchronous multicarrier DS-CDMA systems", *in Proc. IEEE PIMRC'04*, pp. 2930-2934, Sept. 2004.

[12] Shin H., Sayed A. H. and Song W., "Variable step-size NLMS and affine projection algorithms", *IEEE Signal Processing Letters*, Vol. 11, n°2, Feb. 2004.

[13] Ruweished H., Jamoos A. and Abdel Nour H., "Variable Step-Size Adaptive Filter Based MMSE Receivers for MC-DS-CDMA Mobile Systems", in *Proceedings of the international Arab Conference on Information Technology (ACIT2006)*, Yarmouk University, Jordan Dec. 19-21, 2006.

[14] cdmaOne BTS Measurement Solutions, www.tektronix.com/commtest.

[15] Ibrahim J.,"4G Features", Bechtel Telecommunications Technical Journal, vol. 1, pp. 11- 14, 2002**.**

[16] Swisscom Mobile, "Technologies–Today and Tomorrow", research of Swisscom Mobile.

[17] Díaz L., "A Guide to Important Topics", *Wireless Communications Education*, Olavarrieta Nov. 2005.

[18] Georgoulis S., "Transmitter Based Techniques for ISI and MAI Mitigation in CDMA-TDD Downlink".

[19] Rappaport T., *Wireless Communication: Principles and Practice*, second edition,2002.

[20] http://en.wikipedia.org/wiki/W-CDMA

[21] Sundaramurthy V.," A Software Simulation Test-bed for CDMA Wireless Communication Systems", published at Rice University.

[22] Gold R., "Optimal Binary Sequences for Spread Spectrum Multiplexing," *IEEE Trans. Info. Theory*, vol. IT-B, pp. 619-21, Oct. 1967.

[23] Lindholm H., "An Analysis of the Pseudo-Randomness Properties of Subsequences of Long m-Sequences", *IEEE Trans. Info. Theory*, vol. IT-14, pp. 569-76, Jul. 1968.

[24] Dinan E. and Jabbari B., "Spreading Codes for Direct Sequence CDMA and Wideband CDMA Cellular Networks", George Mason University.

[25] Proakis J., "Digital Communications", McGraw-Hill, New York, 1995.

[26] Garcia I., and Marciano J., "Electromagnetic Interference Cancellation in Audio Systems using Adaptive Filters", *Digital Signal Processing Laboratory, Communications Engineering Laboratory* University of the Philippines.

[27] Barbosa A. and Miller S., "Adaptive detection of DS-CDMA signals in fading channels", *IEEE Trans. Commun.*, vol. 46, pp. 115–124, Jan. 1998.

[28] Milstein L. B, "Interference rejection techniques in spread spectrum communications", *IEEE Proc.*, vol. 76, pp. 657–671, June 1988.

[29] Majmundar M., Sandhu N and Reed J., "Adaptive Single-User Receivers for Direct-Sequence Spread-Spectrum CDMA Systems", *IEEE Trans. on Vehicular Technology*, vol. 49, Issue 2, pp. 379 – 389, Mar 2000.

[30] Rupp M., "A family of adaptive filter algorithms with decorrelating properties", *IEEE Trans. Signal Processing*, vol. 46, pp. 771–775, Mar. 1998.

[31] Elders-Boll H., Herper M., and Busboom A., "Adaptive Receivers for Mobile DS-CDMA Communication Terms", Institut fur Elektrische Nachrichten technik, Aachen University of Technology (RWTH), Melatener Strafie 23, D-52056 Aachen, Germany

[32] Pateros C. N. and. Saulnier G. J, "An adaptive correlator receiver for direct sequence spread spectrum communication", *IEEE Trans. Commun.*, vol. 44, pp. 1543–1552, Nov. 1996.

[33] Varanasi M. K and Aazhang B, "An iterative detector for asynchronous spread-spectrum multiple-access systems", in *Proc. IEEE Global Telecommunications Conf.*, pp. 556–560 1988.

[34] Sayed A. H., *Fundamentals of Adaptive Filtering*., New York: Wiley, 2003.

[35] Grivel E., Jamoos A. and Najim M., "Blind Adaptive Multi-user detection for MC-DS-CDMA Systems in frequency Selective Fading Channels", Equipe Signal & Image, UMR5131 Laps, France.

[36] Kalofonos D., Stojanovic M., and Proakis J., "Performance of Adaptive MC-CDMA Detectors in Rapidly Fading Rayleigh Channels", *IEEE Trans. on Wireless Communications*, vol. 2, Issue 2, pp. 229 – 239, Mar 2003.

[37] Alexander P, Rasmussen L., and Schlegel C., "A linear receiver for coded multi-user CDMA", *IEEE Trans. Commun.*, vol. 45, pp. 605, May 1997.

[38] Rapajic P and Vucetic B., "Adaptive single-user receiver for asynchronous CDMA system in the narrowband interference environment", in *Proc. Mobile and Personal Communications Systems,* Adelaide, Australia, pp. 143–148, Apr. 12–13 1992.

[39] Clark G. A, Mitra S. K., and Parker S. R "Block implementation of adaptive digital filters" , *IEEE Trans. Acoust. Speech, Signal Processing*, vol. ASSP-29, no. 3, pp. 744-752, Jun. 1981.

[40] Honig M. and Tsatsanis M., "Adaptive techniques for Multi-user CDMA receivers", Enhanced signal Processing with short spreading codes.

[41] Latva-aho M, "Advanced receivers for wideband CDMA systems", Faculty of Technol., Univ. of Oulu, 1998

[42] Verdu S., "Minimum probability of error for asynchronous Gaussian multiple-access channels", *IEEE Trans. Inform. Theory,* vol. IT-32, pp. 85–96, Jan. 1986.

[43] Honig M., Madhow U., and Verdu S., "Blind adaptive multiuser detection", *IEEE Trans. Inform. Theory*, vol. 41, pp. 944–960, July 1995.

[44] Benesty J. and Duhamel P., "A fast exact LMS adaptive algorithm", *IEEE Trans Signal Processing,* vol. 40, no. 12, Dec.1992

[45] Montazeri M. and Duhamel P., "A Set of Algorithms Linking NLMS and Block RLS Algorithms",*IEEE International Conference on Acoustics, Speech, and Signal Processing*, ICASSP-93., pp.412 – 415, vol.3, 27-30 Apr 1993.

[46] Morgan D. R. and Kratzer S. G., "On a class of computationally efficient, rapidly converging, generalized NLMS algorithms", *IEEE Signal Processing Lett.*, vol. 8, pp. 245–247, Aug. 1996.

[47] Apolinario J., Campos M., and. Diniz P, "Convergence analysis of the binormalized data-reusing lms algorithm", *IEEE Trans. Signal Processing*, vol. 48, pp. 3235–3242, Nov. 2000.

[48] Carayannis G., Manolakis D, and Kalouptsidis N., "A fast sequential algorithm for least square filtering and prediction", *IEEE Trans. Acoust. Speech, Signal Processing,* vol. ASSP-31, no. 6, pp. 1394-1402, Dec.1983.

[49] Sankaran S. G. and Beex A., "Convergence Behavior of Affine Projection Algorithms", *Student Member, IEEE Trans. on Signal Processing,* vol. 48, Issue 4, pp. 1086 – 1096, Apr 2000.

[50] Ozeki K. and Umeda T., "An adaptive filtering algorithm using an orthogonal projection to an affine subspace and its properties", *Electron. Commun. Jpn.*, vol. 67-A, no. 5, pp. 19–27, 1984.

[51] Tanaka M., Kaneda Y., Makino S., and Kojima J., "Fast projection algorithm and its step size control", in *Proc. IEEE Int. Conf Acoust., Speech, Signal Processing,* adaptive filtering algorithm in *Proc. IEEE Workshop,* pp. 945-948. 1995.

[52] Kwong R. H. and Johnston, "A variable step size LMS algorithm", *IEEE Trans. Signal Processing,* vol. E. W. 40, no. 7, pp. 1633-1642, July 1992.

[53] Soni R., Gallivan K. and Jenkins K., "Low-Complexity Data Reusing Methods in Adaptive Filtering", *IEEE Trans. on Signal Processing,* vol. 52, Issue 2, pp. 394 – 405, Feb. 2004.

[54] Gay S., "A fast converging, low complexity adaptive filtering algorithm", *IEEE Workshop on Applications of Signal Processing to Audio and Acoustics,* pp. 4–7, 17-20 Oct 1993.