Interference Suppression in WCDMA with Adaptive Thresholding based Decision Feedback Equaliser.

A thesis submitted in partial fulfilment of the requirements of the degree of

Master of Technology

in

Electronics and Communication Engineering

Specialisation: Communication and Signal Processing

by

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Department of Electronics & Communication Engineering

National Institute of Technology, Rourkela

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Under the Guidance

Of

Prof.S.K.PATRA

Department of Electronics & Communication Engineering

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Rourkela

2013

Dedicated to my parents

And

My brother

Department of Electronics and Communication Engineering

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Rourkela-769008, Orissa, India.

CERTIFICATE

This is to certify that the work in this thesis entitled " **Interference suppression in WCDMA with Adaptive Thresholding based Decision Feedback Equaliser**" by Ms. Shefalirani Patel has been carried out under my supervision in partial fulfilment of the requirements for the degree of M.Tech in Electronics and Communication Engineering (Specialization- Communication and Signal Processing) during session 2011-2013 in the Department of Electronics and Communication Engineering, National Institute of Technology, Rourkela, and this work has not been submitted elsewhere for a degree.

Dr. S. K. Patra
Professor,
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Shefalirani Patel

Roll No.-211EC4108

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Nomenclature

$*$	:	Convolution
$\otimes$	:	Knocker Product
ϕ	:	Channel matrix
σ	:	Covariance matrix
γ_k	:	Power of the k^{th} user
$w_{k,m}(t)$	:	convolution of the transmitted spreading waveform with the channel impulse response
A	:	Product between code matrix T and channel matrix H
A^H	:	Hermitian of matrix A
D_k	:	Delay of the k^{th} user
e_i	:	i^{th} error symbol
G_k	:	Spreading factor for k^{th} user
h_e	:	Equalised channel weight
h_k	:	Multi-path fading channel of k^{th} user
I	:	Identity Matrix
I_{c_i}	:	Index of the correct symbol at i^{th} iteration
k	:	Number of users
p	:	Cross correlation matrix
R	:	Residual matrix
ro	:	Level of sparity

R^{-1}	:	Inverse of the autocorrelation matrix
s_{min}	:	Minimum distance among symbols of the used constellation
T	:	Code Matrix
T_d	:	Symbol duration
th_i	:	Threshold value at i^{th} iteration
T^l	:	l^{th} column of code matrix T
T_{km}	:	Spreading code of k^{th} user and m^{th} symbol
w	:	Additive white Gaussian noise
x	:	Transmitted symbol
$\hat{x}$	:	Estimated front end output
x_{mmse}	:	Estimated symbol using MMSE
y_k	:	k^{th} received symbol
$\widehat{y_k}$	:	k^{th} Equalised output signal

Abbreviation

1G	:	First Generation
2G	:	Second Generation
3G	:	Third Generation
3GPP	:	Third Generation Partnership Project
AWGN	:	Additive White Gaussian Noise
BER	:	Bit Error Rate
BPSK	:	Binary Phase Shift Keying
CDMA	:	Code Division Multiple Access
CGP	:	Conjugate Gradient Pursuit
CoSaMP	:	Compressive Sampling Matching Pursuit
DFE	:	Decision Feedback Equaliser
DPCH	:	Dedicated Physical Channels
DPDCH	:	Dedicated Physical Data Channel
DPDCH	:	Dedicated Physical Control Channel
EGC	:	Equal Gain Combining
EU	:	Enhanced Uplink
FDD	:	Frequency Division Duplexing
FDMA	:	Frequency Division Multiple Access
GP	:	Gradient Pursuit
HSDPA	:	High Speed Downlink Packet Access
HSUPA	:	High Speed Uplink Packet Access

I	:	In phase
IS	:	Interim Standard
ISI	:	Inter Symbol Interference
LTE	:	Linear Transversal Equaliser
MAI	:	Multiple Access Interference
MSE	:	Mean Square Error
MMSE	:	Minimum Mean Square Error
MRC	:	Maximal Ratio Combining
MS	:	Mobile Station
MUI	:	Multi User Interference
OMP	:	Orthogonal Matching Pursuit
OVSF	:	Orthogonal Variable Spreading Factor
PDC	:	Personal Digital Cellular
Q	:	Quadrature phase
QPSK	:	Quadrature Phase Shift Keying
RR	:	Rake Receiver
SC-FDMA	:	Single Carrier Frequency Division Multiple Access
SF	:	Spreading Factor
SNR	:	Signal to Noise Ratio
StOMP	:	Stage-wise Orthogonal Matching Pursuit
TDD	:	Time Division Duplexing
TDMA	:	Time Division Multiple Access
TFCI	:	Transport Format Combination Indicator
UMTS	:	Universal Mobile Telecommunication Standard

UTRA	:	UMTS Terrestrial Radio Access
WCDMA	:	Wideband Code Division Multiple Access
WH	:	Walsh Hadamard
ZF	:	Zero Forcing

Abstract

WCDMA is considered as one of the 3G wireless standards by 3GPP. Capacity calculation shows that WCDMA systems have more capacity compared to any other multiple access technique such as time division multiple access (TDMA) or frequency division multiple access (FDMA). So it is widely used.

Rake receivers are used for the detection of transmitted data in case of WCDMA communication systems due to its resistance to multipath fading. But rake receiver treat multiuser interference (MUI) as AWGN and have limitation in overcoming the effect of multiple access interference (MAI) when the SNR is high. A de-correlating matched filter has been used in this thesis, which eliminates and improves system performance. But the given receiver works well only in the noise free environment.

A DFE, compared to linear equaliser, gives better performance at severe ISI condition. The only problem in this equalisation technique is to select the number of symbols that are to be fed back. This thesis gives an idea on multiple symbol selection, based on sparity where an adaptive thresholding algorithm is used that computes the number of symbols to feedback.

Simulated results show a significant performance improvement for Regularised Rake receiver along with thresholding in terms of BER compared to a rake receiver, de-correlating rake receiver and regularised rake receiver. The performance of the receiver in different channels is also analysed.

1

Introduction

1.1 Background

Wireless communication has a very rapid growth as seen in history because of the introduction of new technologies introduced. In 1946, AT&T first introduced the American communication mobile radio telephone service for private customers [1]. This has a drawback of having limited power transmission so was not in much use. In 1969 Bell system developed a commercial cellular radio operating on frequency reuse concept. The first modern cellular mobile system was developed by Advance Mobile Phone Service (AMPS) in the year 1980. It was an analogue communication and used 666 numbers of channels. Roaming within the city of United States was easy as it used analogue system, but roaming in Europe was tough. Analog systems do not cope with the increasing demand so, digital cellular concept came into existence where new frequency bands were used. Analog systems do not fully utilise the signal between the phone and cellular system because analog signal cannot be compressed and manipulated as easily as compared to the digital signal. GSM system was considered as the first digital cellular system.

1.1.1 Advantages and Disadvantages of Wireless Communication

Advantages

There are a number of advantages of wireless communication over wired communication. Some of them include mobility, increased reliability, rapid disaster recovery, ease of installation and low cost.

Mobility: The primary advantage of wireless communication is the freedom of moving that it provides to the user while communicating with someone. This means the user is not bound to be fixed at a place as in wired communication.

Increased reliability: The main problem with wired communication is failure of network or damage to wire because of environment. Thus the wireless communication not only eliminates the above said problem but also increases reliability.

Rapid Disaster Recovery: Natural disasters are never predictable and they harm the wires that are not in case of wireless communication.

Ease of Installation: The time to install a network cable may take some days to week while using wireless LAN eliminates this trouble.

Cost: The cost required in the cables and other hardware equipment is reduced compared to wireless system.

Disadvantage

As all fields have two sides i.e. a brighter and a darker side, the darker side of wireless communication include radio signal interference, security problems and health hazards.

Signal Interference: Signals from different other wireless devices can interrupt its transmission or a wireless device may itself be a source of interference for other wireless devices.

Security: Here in wireless communication device, it transmits signal over a wide range and thus security becomes major concern.

Health Hazards: The frequency range in which the wireless communication operates is very high that may cause damage to the sensitive organs.

1.1.2 Mobile Communication 1G to 3G

First Generation Analog Cellular System

The first generation cellular system comprises of analog transmission technology. Mostly AMPS was developed in this generation in most parts of US, South America, Australia and China. It uses frequency modulation (FM) for transmitting the signal. Here the entire service area is divided into number of cells and each cell is allocated with a unique frequency band. For frequency reuse concept the frequency band is divided among seven cells that improves the voice quality as each subscriber uses larger bandwidth. It uses a channel bandwidth of 30 KHz in 800MHz spectrum. This system uses two separate bands for uplink and downlink.

Second Generation Digital Cellular System

The second generation deals with digital transmission technology. The digital cellular technology compared to analog system supports larger subscribers within the given frequency band thus supports higher user capacity, superior voice quality as well as security. To have efficient use of frequency spectrum, it uses time division and code division multiple access technique that transmits low data rate as well as voice. A 2G system mainly comprises of four standards i.e. North American Interim Standard (IS-54), GSM, the pan-European digital cellular; Personal Digital Cellular (PDC) and IS-95 in North America. Among the four, first three use TDMA technique and last one uses CDMA. As in 1G, 2G system also uses FDD technique i.e. different bands for forward and reverse link and use the frequency band that ranges 800 MHz - 900 MHz. Here the signals are transmitted in the form of packets or frames. The length of the packets should not be short enough, so that the channels do not change significantly during transmission. The length of packet should not be long enough so that the time interval between packets should not be smaller than the length of the packet. GSM supports eight users in 200 KHz band and IS-54 uses three users in 30 KHz band.

Evolution from 2G to 3G

GSM is a digital cellular technology that supports voice as well as data transmission at a speed of 9.6 kbps along with Short Message Service (SMS). It operates at 800 MHz and 1.8 GHz band in Europe and 850 MHz and 1.9 GHz band in US. It provides international roaming and GPRS. Besides GSM the technology also has CDMA communication standard that supports data as well as multimedia services. It was started in 1991 as IS-95A. It supports a variable number of users in 1.25 MHz wide channel. It can operate at high interference environment because of its interference resistance property.

Third Generation

Here the system aims to combine telephony, internet and multimedia to a single device i.e. it supports high quality images and videos and also supports high data rates. WCDMA is the most widely used 3G air interfaces. It was developed by Third Generation Partnership Project (3GPP), which is the joint standardisation project of the standardisation bodies from Europe, Japan Korea China and USA. WCDMA was selected as an air interface for Universal Mobile Telecommunication Services (UMTS).

Code Division Multiple Access communication networks have been developed by a number of companies over the years, but development of cell-phone networks based on CDMA (prior to W-CDMA) was dominated by Qualcomm [1]. Qualcomm was the first company to develop a practical and cost-effective CDMA implementation for consumer cell phones. Its air interface standard IS-95 has since evolved into the current WCDMA and CDMA2000 standard [2].

1.2 Literature Survey

CDMA was first commercially used in the mid of nineties. After the tremendous success of IS-95, WCDMA was adopted as the3GPP air interface with initial released in 99 and 4 [2]. Since then WCDMA was used as wireless internet, video telephony and voice over IP. High speed downlink packet access (HSDPA) is used for packet switch connectivity in downlink direction as in release 5 in 2002. High speed uplink packet Access (HSUPA) also known as enhanced uplink (EU) is used to support the packet in uplink in release 6 in 2004. The HSPA for WCDMA was standardised in 3GPP release 7 known as HSPA+ in June 2007 [2]. A rake receiver that is used in the receiver section was used to overcome the multiple access interference. It was first proposed by Prince and Green and patent in 1958 [3]. The main problem with rake receiver is that it treats multiuser interference (MUI) as Additive white Gaussian noise (AWGN) and is unable to overcome the effect of multiple access interference (MAI). So, blind two dimensional rake receivers was first proposed by Zoltowski et al in his paper [4] and further developed in his further papers [5, 6]. In his paper he used the conventional matched filter in the first stage followed by post processing to mitigate the effect of multiuser interference. There he used channel estimation via matrix pencil technique base on second order statistic. De-correlating rake receiver was proposed by Lang Tong et al [7]. Here he used the rake concept where de-correlating matched filter was used in the front end to separate users and perform single-user optimal rake combining as the second step. Here least square estimation was used which has an advantage of requiring small number of samples. Here the channel estimation as well symbol detection was proposed that not only gives better performance but also reduces the computational complexity. The technique works well but at high interference condition its performance degrades. For that one should go for non-linear equaliser. In the paper [8, 9] DFE was used to overcome the effect of interference in case of Single Carrier Frequency Division Multiple Access (SC-FDMA). The paper gives an adaptive thresholding technique that computes the sparsity solutions and thus decides the number of symbols used to be fed back.

1.3 Motivation

In case of spread spectrum communication, rake receivers (RR) are used in the receiver section for the detection of the transmitted symbols. As in case of WCDMA, unique orthogonal codes are used for each user and the same code is used in the receiver section to get back the transmitted symbol. The RR works on the principle of multipath diversity and decodes the received signal. When the signal is transmitted through the channel, then it suffers from multipath propagation and the orthogonality of the code is lost. As the code orthogonality is lost RR is unable to work properly. So we go for different equalisation technique to overcome the above effect.

Equalisation technique is used to overcome the effect of inter symbol interference [10]. Linear equalisation is preferred as they are simple and easy to build. But at severe ISI condition they are unable to work properly so a non-linear equaliser is preferred at high interference environment. A DFE non-linear equaliser is used that lowers the interference level and improves the performance in terms of BER. The DFE comprises of a feed forward and a feedback path along with a decision device. An adaptive thresholding is used in the decision devise of DFE that helps in improving the performance.

1.4 Thesis Outline

This thesis is organised into four chapters. The first chapter deals with the introduction to wireless and mobile communication and different technology used in different generations. The chapter ends with the outline of the thesis.

The second chapter deals with the basics related to WCDMA. There the multipath propagation and the receiver that is used are given in details. Here a notation is used to describe the received signal. The working of rake receiver is briefly explained in this chapter.

Next chapter starts with a brief introduction to different equalisation technique. It was found that non-linear equalisation technique gives better performance so they are discussed briefly in this part. The simulation results are also shown in this chapter that give a comparative study between different equalisation techniques.

The last chapter is the summary and the discussion on the work presented in this thesis. It deals with the limitations and also outlines the future work.

2
WCDMA

2.1 Introduction

Wideband Code Division Multiple Access (WCDMA) also known as UMTS Terrestrial radio Access (UTRA). It was first developed by Third Generation Partnership Project (3GPP) [2] which is the joint standardisation project of the standardisation bodies from Europe, Japan, Korea, the USA and China**.** It has a band width of about 5MHz. Because of this large bandwidth it's named as Wideband CDMA or Wideband Code Division Multiple Access**.** Here we can use two modes for communication i.e. Frequency Division Duplexing (FDD) and Time Division Duplexing (TDD).

In TDD mode, uplink and downlink transmission is performed in the same frequency band but they are differentiated by allotting separate time slots. It is a transmission scheme that allows asymmetric flow for uplink and downlink data transmission. Here users are allocated time slots for uplink and downlink transmission.

In FDD two separate frequency bands are allotted for the uplink and downlink transmission. A pair of frequency bands with specified separation is assigned for a connection.

Uplink is the connection from mobile to base station and downlink is the reverse one i.e. connection from base station to mobile.

2.1.1 Specification

The specification of WCDMA [11] is given in the Table 2-1. WCDMA has a chip rate of 3.84Mcps. It has a frame length of 10 mili-seconds and each frame is divided into 15

Table 2-1 Features of WCDMA

Channel bandwidth	5 MHz
Duplex mode	FDD and TDD
Chip rate	3.84Mbps
Frame length	10 mili-seconds
Data modulation	BPSK(uplink)
	QPSK(downlink)
Channel Coding	Convolutional, Turbo code
Spreading factors	4–256 (uplink), 4–512 (uplink)
Spreading (downlink)	OVSF sequences for channel separation Gold sequences 218-1 for cell and user separation (truncated cycle 10 mili-seconds)
Spreading (uplink)	OVSF sequences, Gold sequence 241 for user separation (different time shifts in I and Q channel,(truncated cycle 10 mili-seconds)
Power control	Open or closed loop.

slots. Here the spreading factor can be different for uplink and downlink. The ratio of the chip rate to the data rate is called the spreading factor. The spreading factor for uplink ranges from 256 to 4 while for downlink it is 512 to 4.

2.1.2 Features of WCDMA

Some of the features of the WCDMA are given as below:

- It supports high data rate transmission i.e. 384 Kbps with wide area coverage and 2 Mbps of local coverage.
- It has high service flexibility i.e. it supports multiple parallel variable rate services on each connection.
- It operates on both Frequency Division Duplex (FDD) and Time Division Duplex (TDD)
- It has an ability to support for future capacity and coverage enhancing technologies like adaptive antennas, advanced receiver structures and transmitter diversity.
- It supports inter-frequency hand over and the hand over among other systems, including handover to GSM.
- Has efficient packet access capability.
- It employs coherent detection on uplink and downlink based on the use of pilot symbols.
- One of the important features of WCDMA is soft handover, which is supported by simultaneously delivering data to mobile from different base station.
- It provides multipath diversity for small cells.

Advantage of WCDMA

Some of the advantages of WCDMA include:

- Service flexibility-WCDMA has a capability to allow carrier of 5MHz to process mixed service from 8Kbps to 2Mbps. In also supports circuit switched service and packet switched service in the same channel, and a single terminal is used for carrying out multiple circuits and packets switched services, and thus realize genuine multimedia service. It supports services with different requirements such as voice and packet data and also ensures high quality and perfect coverage.
- Spectrum efficiency- WCDMA has the ability of making the efficient use of available radio spectrums. Here no frequency planning is required as a single cell multiplexing is adopted. Network capacity can be improved drastically by using technologies of hierarchical cell structure, coherent de-modulation and adaptive antenna array.
- Capacity and Coverage- This has an increased number of voice users which is about eight times when compared with a typical narrowband transceiver. Here at a time, the RF carriers can handle about 80 voice calling users or about fifty internet data users.
- Every connection can provide multiple services: Circuit and packet switched services can be mixed freely in different band width and can be provided to the same user in the same time instance. Every WCDMA terminal can access up to six different types of services, which can be the combination of various data services such as video, e-mails, voice, fax, etc.

- Network scale economics – If a digital cellular network, such as GSM, is working in two systems and WCDMA wireless access is added to it then also the same multiplexed core network and the same base stations can be used. The latest ATM mode i.e. micro-cell transmission procedure is used for the links between WCDMA access network and GSM core network. Using this method high data packets can be processed effectively, which can further enhance the capacity of standard E1/T1 lines to 300 voice calls compared to 30 voice calls for that of existing network. It is found that about 50% transmission cost has been saved using this multiple access technique.
- Outstanding voice capability - Although the main purpose for the next generation mobile access is to transmit high bit rate data, along with the voice communication, it can also support other specs.

2.2 Spreading Codes

The codes that are used for spreading the transmitted sequence can be long codes or short codes. Some of the well-known codes are Walsh-Hadamard codes (WH codes), m-sequences, Gold codes and Kasami codes. Walsh codes are orthogonal on zero code delay whereas the m-sequences, Gold codes and Kasami codes are non-orthogonal codes with varying cross-correlation properties. Gold codes and Walsh-Hadamard codes are used in uplink communication of WCDMA. The signals are spread using the orthogonal variable spreading factor (OVSF) codes and are then scrambled using the p-n codes that helps in differentiating different base stations. Scrambling is done on top of spreading which is shown in Figure 2-1

Short code also known as **Channelization code**, as the name states, is used to separate different channels that are transmitted. These codes are generally orthogonal to each other that help in minimizing interference between different users. Orthogonal Codes are used to distinguish between data channels from the same mobile. These codes have an effect on the bandwidth of the signal i.e. the signal bandwidth increases.

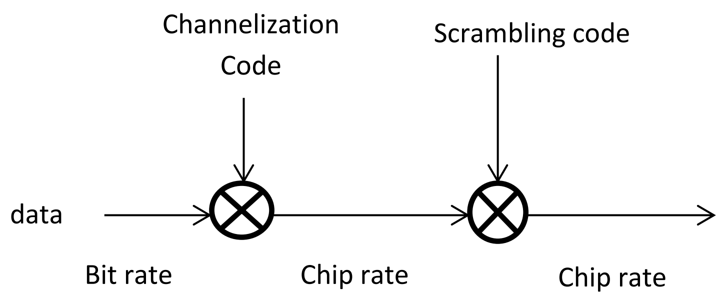

Figure 2-1 Spreading and Scrambling

Generally **Walsh-Hadamard** codes are used as channelization code as they are orthogonal to each other. Figure 2-2 gives the structure of the Walsh code that is an orthogonal code.

Long code also known as **scrambling code** are used to distinguish between all base transceiver station and all mobile stations. Good long codes should have low out of phase auto-correlation peaks to maximize the probability of correct synchronization. The long codes should have low cross-correlation peaks in order to minimise the interference between different base transceiver station and mobile station. These codes have no effect on the signal bandwidth i.e. the bandwidth remains same. Gold code sequences are used as long codes.

2.2.1 Walsh-Hadamard code

Walsh-Hadamard codes [12], also known as OVSF (Orthogonal Variable Spreading Factor) codes, are used as uplink spreading codes. OVSF codes can be understood using the code tree as shown in Figure 2-2. The subscript here gives the spreading factor and the argument within the brackets provides the code number for that particular spreading factor.

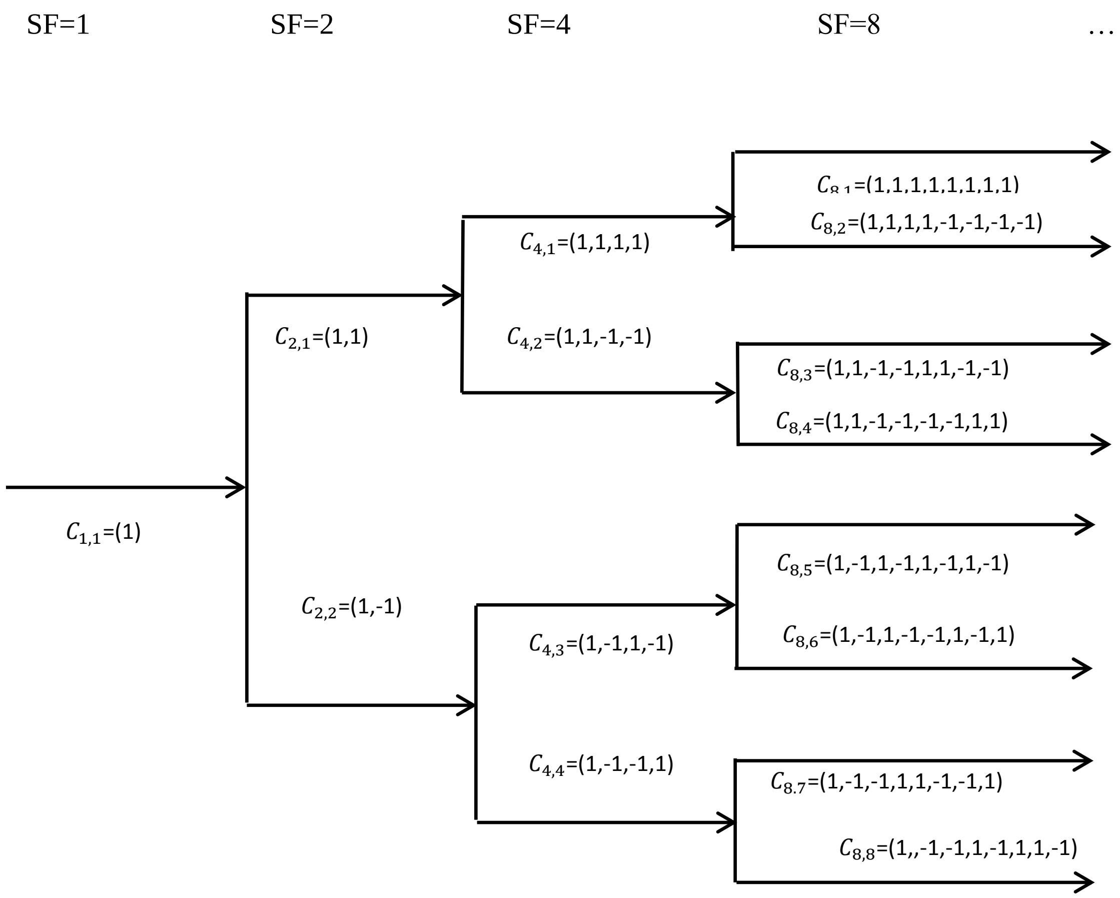

Figure 2-2 Walsh Hadamard tree

The above Figure 2-2, defines the spreading codes of length SF, which represents a particular spreading factor. The spreading factor in uplink can be computed using the equation as

$$SF = \frac{256}{2^k} \tag{2-1}$$

Where the parameter k represents the number of bits in each slot .The spreading factor may thus range from 256 down to 4.

The number of codes for a particular spreading factor is equal to the spreading factor itself. All the codes of the same level are orthogonal to each other as they constitute a set. If we consider any two codes of different levels then they are orthogonal to each other only if one of them is not the mother code of the other code [12]. For example the codes $C_{16,2}$, $C_{8,1}$ and $C_{4,1}$ are all mother codes of $C_{32,3}$ and hence are not orthogonal to $C_{32,3}$. Thus all the codes within the code tree cannot be used simultaneously by a mobile station. A code can be used by an Mobile station (MS) if and only if no other code on the path from the specific code to the root of the tree or in the sub-tree below the specific code is used by the same mobile station. The OVSF can be generated with the help of the following matrix equations known as Walsh Hadamard matrix as

$$H_2 = \begin{bmatrix} +1 & +1 \\ +1 & -1 \end{bmatrix} \tag{2-2}$$

The above equation gives the code for a spreading factor of 2 and if M is power of 2 and greater than 2 then the matrix is given by

$$H_M = \begin{bmatrix} +H_{\frac{M}{2}} & +H_{\frac{M}{2}} \\ +H_{\frac{M}{2}} & -H_{\frac{M}{2}} \end{bmatrix} \quad \textbf{(2-3)}$$

Any two rows of any $\boldsymbol{H_M}$ are mutually orthogonal to each other. These sequences have zero cross-correlation when the codes are synchronous. But when all the users are not synchronised to a single time base or when significant multipath is present in that case the advantage of using OVSF codes is lost.

2.2.1 Frame Structure

The frame structure of the uplink dedicated physical channel is shown in Figure 2-3. As shown, here each frame of 10 mili-seconds is split into 15 slots. Each slot has a length of 0.667 mili-seconds with a 2560 chips [11]. A super frame has a length of 720 mili-seconds duration which means it has 72 frames. Pilot bits assist coherent demodulation and channel estimation. Transport Format Combination Indicator (TFCI) is used to indicate and identify several simultaneous services. Feedback Information (FBI) bits are to be used to support techniques requiring feedback. Transmit Power Control (TPC) is used for power control purposes. The Spreading Factor (SF) may range from 256 down to 4. The spreading factor is selected according to the data rate.

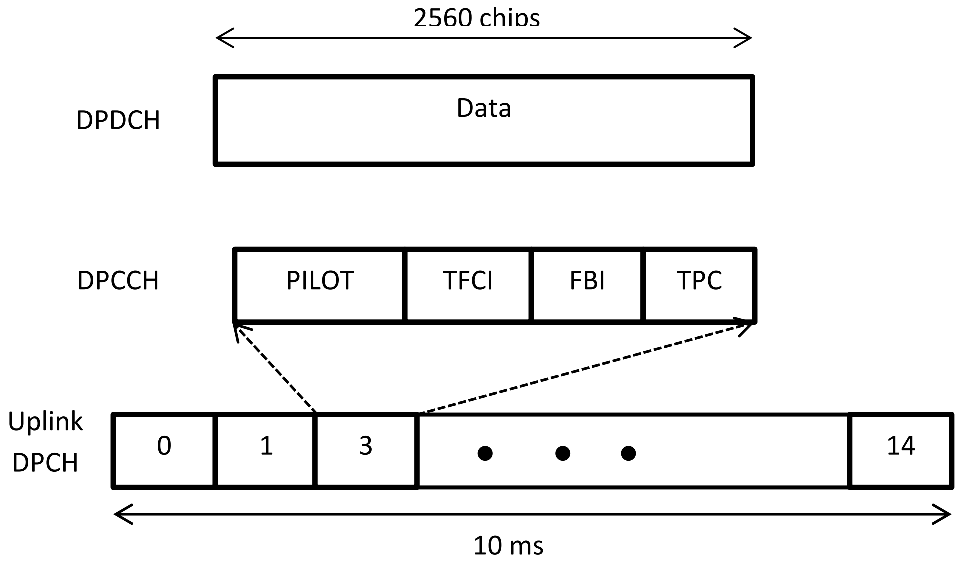

Figure 2-3 Frame and Slot structure for uplink

Figure 2-4 shows the frame structure of downlink transmission. As in uplink, the downlink structure also has one frame that has duration of 10 mili-seconds and consists of 15 slots; each slot contains 2560 chips where each chip is an RF signal symbol.

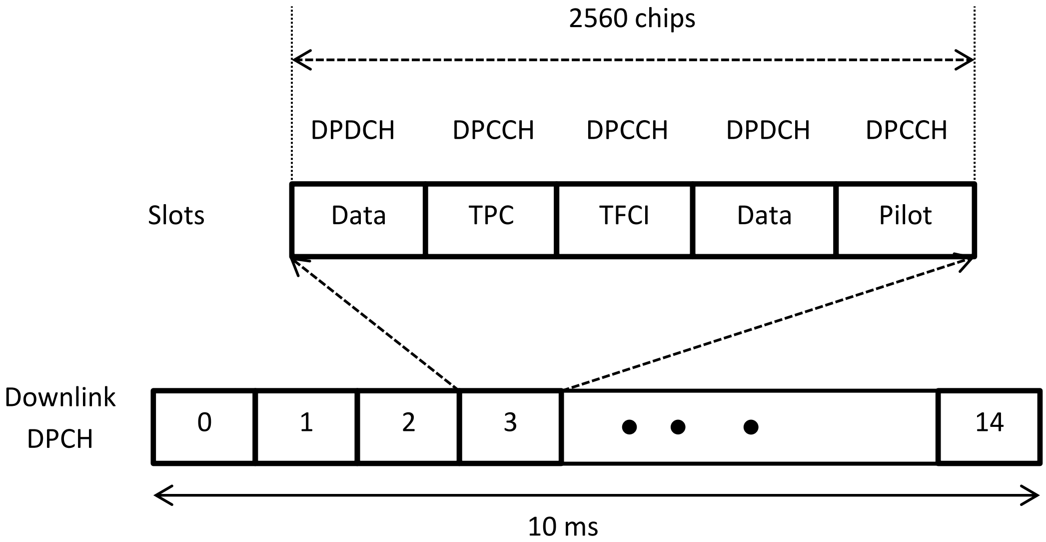

Figure 2-4 Frame and Slot Structure for downlink

2.3 Modulation

In the uplink communication of WCDMA, WH codes are used to spread the transmitted data, and gold codes are used to scramble the spread data, so known as scrambling code. There are two types of uplink dedicated physical channels (DPCH) i.e. uplink Dedicated Physical Data Channel (uplink DPDCH) and uplink Dedicated Physical Control Channel (uplink DPCCH) [2]. The uplink DPDCH is used to carry dedicated data generated at Layer 2 and above while uplink DPCCH is used to carry control information generated at Layer 1.

In the uplink, data modulation of both the DPDCH and the DPCCH is Binary Phase Shift Keying (BPSK). The modulated DPCCH is mapped to the quadrature phase channel (Q-channel), while the first DPDCH is mapped to the in-phase channel (I-channel) and then DPDCHs are mapped alternatively to the Q or the I-channel. After data modulation and before pulse shaping the spreading modulation is applied to the symbols. The spreading modulation used in the uplink is dual channel QPSK. Spreading modulation comprises of two different operations where the first one is spreading where each data symbol is spread to a number of chips given by the spreading factor (SF). This increases the bandwidth of the transmitted signal. The second operation is scrambling where a complex valued scrambling code is applied to the spread signal.

Figure 2-5 shows the spreading and modulation for an uplink data transmission. The uplink user has a DPDCH and a DPCCH.

Quadrature Phase Shift Keying (QPSK) data modulation is performed in case of downlink transmission. Each pair of two bits is serial-to-parallel converted and mapped to the In-phase and Quadrature phase respectively. The data in the I and Q branches are then spread to the chip rate by using the same channelization code. This spread signal is then scrambled by a cell specific scrambling code.

Figure 2-6 shows the spreading and modulation for the downlink transmission. Modulation in case of uplink and downlink transmission is the same except in downlink an extra serial to parallel converter is used.

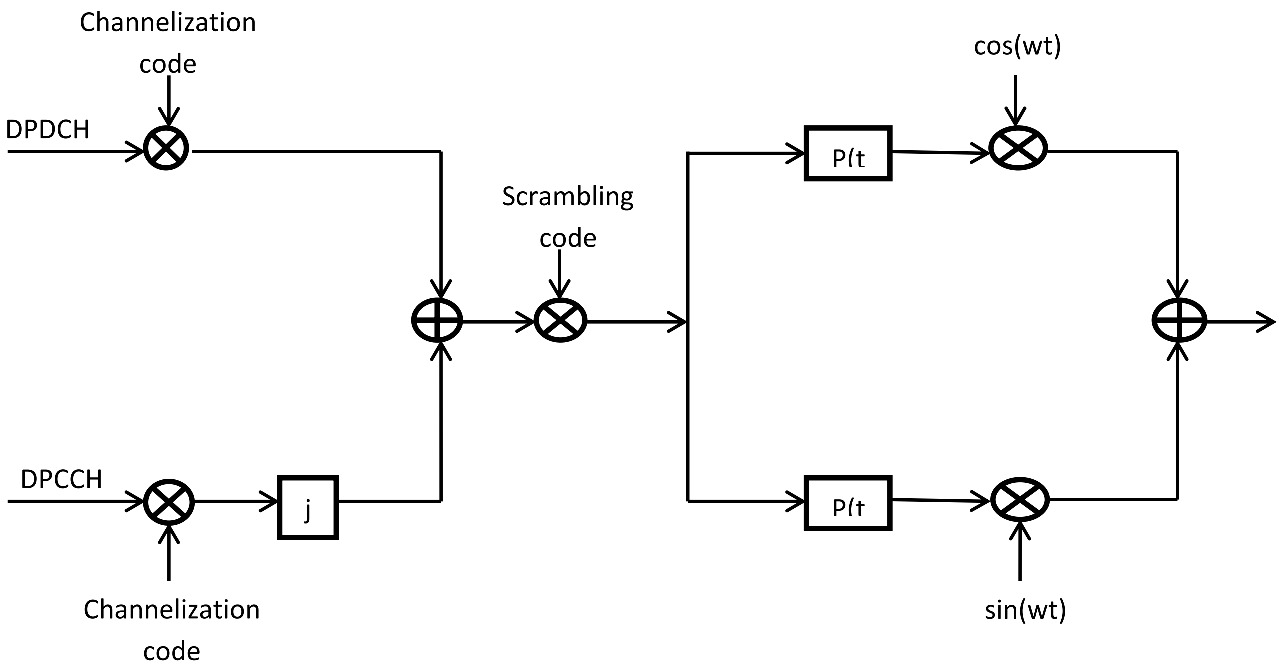

Figure 2-5 Uplink Spreading and Modulation

The bipolar data symbols on I and Q branches are independently multiplied by different channelization codes. As discussed above the channelization codes are known as Orthogonal Variable Spreading Factor (OVSF) codes or Walsh-Hadamard codes.

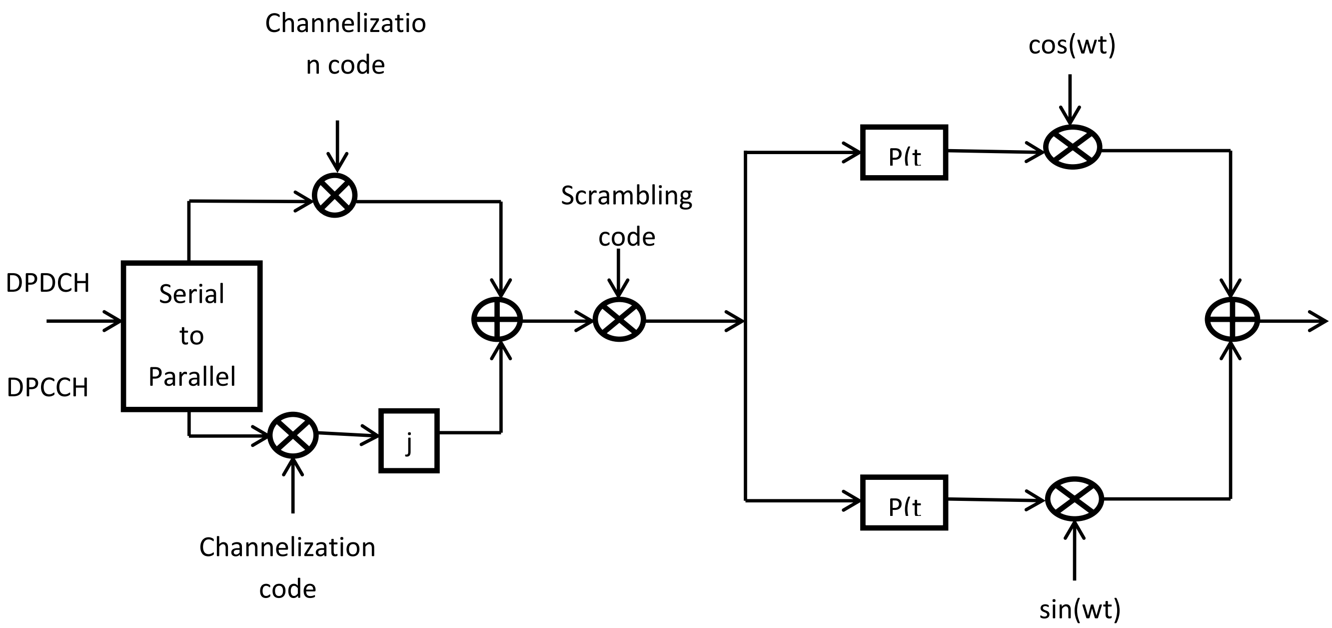

Figure 2-6 Downlink Spreading Modulation

2.4 Multipath Channel

The signal that is transmitted through a channel suffers from multipath propagation as shown in Figure 2-7 **Error! Reference source not found.** . The basic propagation mechanisms that are related in the transmission are reflection, scattering, diffraction or shadowing.

2.4.1 Reflection

Reflection occurs when the signal transmitted strikes an object that has greater dimension compared to the wavelength of the signal. The reflected wave may interfere constructively or destructively with the received signal. This occurs from the surface of earth, building walls, etc.

2.4.2 Diffraction

Diffraction occurs when the signal is being obstructed by some sharp irregularities present between the transmitter and receiver section. At high frequencies it depends upon the nature of the object it strikes, as well as the amplitude, phase and polarisation of the incident signal at the point of diffraction.

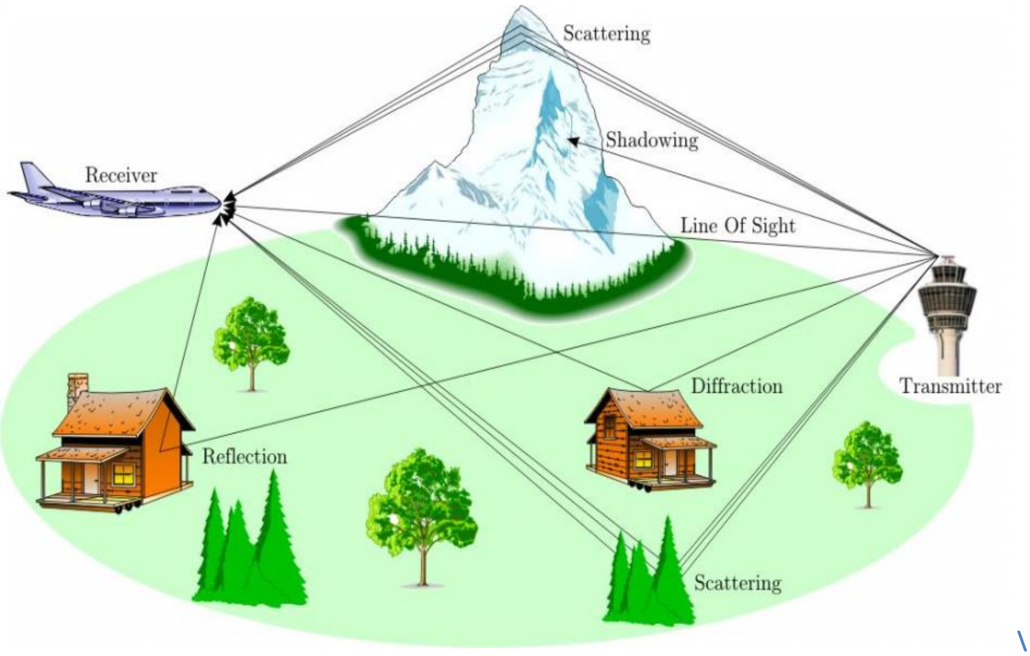

Figure 2-7 Multipath channel

2.4.3 Scattering

Scattering occurs when the medium through which the signal travels consists of the objects that have dimensions smaller compared to the signal wavelength, and where the numbers of obstacles per unit volume are large. This generally occurs because of the tree foliage, street marks and lamp posts etc.

2.4.4 Shadowing

Shadowing occurs when large object block the path of propagation of the signal. Here the signal is unable to propagate and thus it is transmitted from the transmitter section but never reaches the destination i.e. the receiver section.

2.5 Rake Receiver

The conventional Rake receiver was first given by R Price and P E Green and patent in the year 1958 [10, 13]. Sometimes it's also known as diversity combiner. As the name states, it has the similar feature with that of the garden rake. A rake receiver is used that counters the effect of multipath fading. Rake receiver does this by using several sub-receivers, also known as fingers, each delayed slightly in order to tune with the delayed multipath component. Each component is decoded but at the later stage combined coherently in order to make use of the different transmission characteristic of each transmission path.

2.5.1 Different Combining Techniques

Different combining techniques that are used in the reception side are

2.5.1.1 Selective diversity

Here the receiver finds the signal having maximum signal to noise ratio (SNR) from different signals and use that signal for detection.

$$w_i = \begin{cases} 1, if\ |y_i| = \max\{|y_1|, |y_2| \ldots . |y_M|\} \\ 0, otherwise \end{cases} \quad \textbf{(2-4)}$$

2.5.1.2 Maximal Ratio Combining (MRC): Here different weights are used corresponding to each finger so that the output SNR is maximised. The envelope of the resulting combined signal $\boldsymbol{y_j}$

is given as

$$\boldsymbol{y_j} = \sum_{i=1}^{M} \boldsymbol{w_i y_i} \qquad \textbf{(2-5)}$$

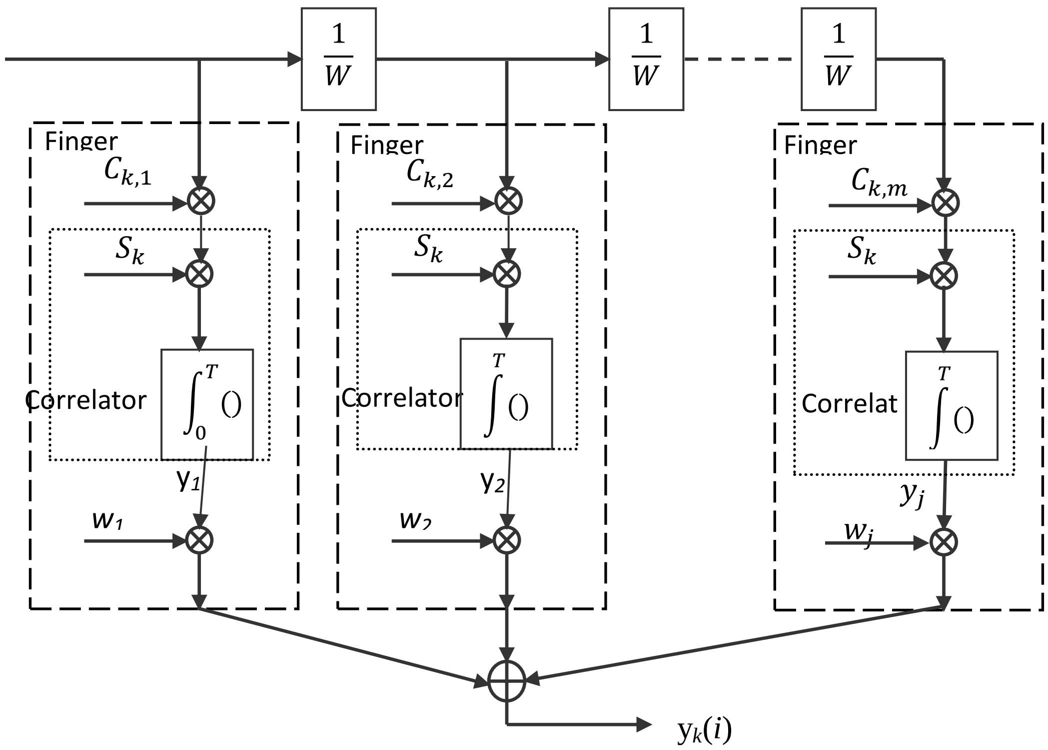

Figure 2-8 Conventional Rake Receiver

2.5.1.3 Equal Gain Combining (EGC): Here this method maximises the SNR of the received combined signal. The main drawback of maximal gain combining is that the receiver complexity increases. So, this combining method is used that reduces complexity as all the weights are same i.e.

$$\boldsymbol{w_1 = w_2 = w_3 = \cdots = w_{M-1}} \tag{2-6}$$

The weight is considered to be 'one' and output signal is given by

$$\boldsymbol{y_j = \sum_{i=1}^{M} y_i} \tag{2-7}$$

Out of three, maximal ratio combining is generally used because of its advantage of producing output with an acceptable SNR even when none of the signals are acceptable. This combining technique gives the best reduction in fading compared to other techniques so is mostly used.

A rake receiver collects the signal energy of different multipath components [13]. The optimal RAKE receiver actually implements a channel matched filer which maximizes the received signal to noise ratio. This means that the identified multipath components are weighted proportionally to the amplitude of the component. A typical rake receiver for WCDMA can contain 3 to 6 number of fingers.

A typical rake receiver consists of a delay line; taps with complex multipliers and integrators which in a spread-spectrum system are implemented as correlators with user's spreading code. As in the Figure 2-8 the received symbols are being delayed which corresponds to the channel delay. Then the symbols are decoded by multiplying the received symbol with that of spread code. The corresponding symbols are combined with the weights that correspond to the channel coefficients and finally all the outputs are combined constructively.

The weighting coefficients *h* are normalised to the output signal power of the correlator.

The main advantage of using a rake receiver is that it improves the SNR. As each coin has two sides, its drawback is that it treats multi-user interference (MUI) as AWGN and is thus unable to overcome the effect of multiple access interference (MAI). It also has a disadvantage of having high cost. That is when we insert more receiver we need more space and thus complexity increases and hence the cost. And in reality, the amount of multipath components that arrive at the receiver is large and the number is not fixed. That is everything depends on the environment.

3

Decision Feedback Equaliser

A wireless channel is generally dispersive i.e. after the signal being transmitted; a system will receive multiple copies of the signal, with different channel gain, at different instance of time. This dispersion in time, in the channel causes inter-symbol interference (ISI) and thus the performance of the system degrades. Thus to overcome the effect of ISI, a number of equalisation techniques are used. The channel equaliser can be broadly classified as linear equaliser and non-linear equaliser. A linear equaliser like a zero forcing equaliser (ZFE) forces the ISI to reduce to zero when the channel is noiseless. But when the channel is noisy, it enhances the noise. This causes degradation of the performance, hence rarely used. Another type of equaliser i.e. minimum mean square equaliser (MMSE) is used that performs better than the ZFE as it takes noise into account. But its performance is not enough for channels with severe ISI. So a non-linear equaliser i.e. a decision feedback equaliser is used.

A DFE is a non-linear equaliser that uses previous detected decisions to eliminate ISI on the current received symbol. The basic idea behind the DFE is to subtract from our observation at the receiver (feed back to the receiver) correctly detected symbols in order to reduce the interference for the currently equalized symbols. A standard DFE has two filters i.e. feed forward filter and feedback filter.

3.1 Introduction

Equalisers are used to mitigate the effect of ISI [14]. The main advantage of this approach is that a digital filter is easy to build and is easy to alter for different equalization schemes, and also to fit different channel conditions. So equaliser can be defined as any signal processing technique that reduces ISI. As shown in Figure 3-1, an equalisation can be broadly classified as linear equaliser and non-linear equaliser. If the received information is not feedback to adapt the equaliser then it's known as linear equaliser while if the signal is being feedback then its non-linear equaliser.

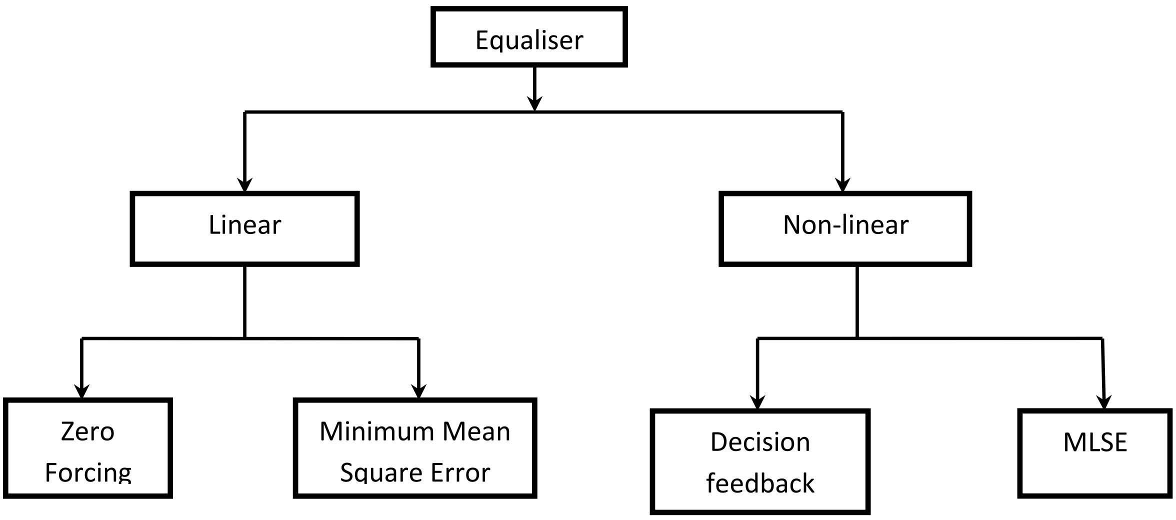

Figure 3-1 Different Type of Equaliser

Linear equaliser's are generally used in the receiver section because of their simplicity, stability and fast convergence. They are also termed as transversal filters or finite impulse response (FIR) filter. Some of the linear equaliser that is commonly used is Zero Forcing equaliser (ZF) and Minimum Mean Square Error equaliser (MMSE).

3.1.1 Zero Forcing

One of the simplest ways to remove ISI is by choosing the transfer function of the equaliser such that the output of equaliser gives back the transmitted information, only if noise is not present, which mean ZF equaliser applies the inverse of the channel frequency response to the conventional detected output in order to restore the received information from the channel $H(z) = \frac{1}{G(z)}$.

This technique is called zero forcing equalisation because ISI component at the equaliser output is forced to zero. Thus it has an advantage of eliminating multiple access interference (MAI) and has less computational complexity compared to maximum likelihood sequence estimator (MLSE). In this type of equalisation, the effect of noise is neglected, but practically noise exists in environment. Although ISI is forced to zero, there is a chance to enhance the noise power by the equaliser. Hence the error performance of the receiver still gets poorer. One more problem of using this technique is the computation needed to inverse the correlation matrix is difficult to perform in real time.

3.1.2 Minimum mean Square Error

Although zero forcing equaliser removes ISI, but it may not give best error performance in communication environment as it does not take noises into account in the system. So we go for minimum mean square error equalisation (MMSE) [7, 15] which is based on mean square error criteria. It was patent by Lucky in the year 1965 [10]. A linear equalizer is being used that minimizes the mean square error between the received signaly_k and the output of the equaliser $\widehat{y_k}$. The mean square error is given as

$$\boldsymbol{MSE} = \boldsymbol{E}[\boldsymbol{e^2}_{\boldsymbol{k}}] = \boldsymbol{E}[(\boldsymbol{y_k} - \widehat{\boldsymbol{y_k}})^2] \qquad \textbf{(3-1)}$$

To compute minimum mean square error the derivative of the above equation with respect to h_E is taken and is equated to 0.

Solving the above equation we get

$$\boldsymbol{h_E} = \boldsymbol{R^{-1}p} \qquad \textbf{(3-2)}$$

Where

$$\boldsymbol{R} = \boldsymbol{E}[\widehat{\boldsymbol{y_k}}\widehat{\boldsymbol{y_k}}] \qquad \textbf{(3-3)}$$

$$\boldsymbol{p} = \boldsymbol{E}[\boldsymbol{y_k}\widehat{\boldsymbol{y_k}}] \qquad \textbf{(3-4)}$$

If R and p are known, then the MMSE equalizer can be found by solving the linear matrix Equation (3-2). It can be shown that the signal-to-noise ratio at the output of the MMSE equalizer is better than that of the zero-forcing equalizer. This type of equaliser is most popularly known as Linear Transversal Equaliser (LTE).

The linear MMSE equalizer can also be found iteratively. The gradient of the MSE with respect to h_E gives the direction to change h_E for the largest increase of the MSE. To decrease MSE one can update h_E in the opposite direction to the gradient. This is known as steepest descent algorithm.

3.1.3 Decision Feedback Equaliser

Although MMSE equalizer gives better performance compared to zero forcing equaliser, but its performance is not enough for channels having high ISI. For that case non-linear equaliser such as Decision Feedback Equalisation is used. Austin first published a report on DFE in the year 1967 and further optimization of DFE received for minimum mean square error was analysed and accomplished by Monsen in the year 1971 [15].

As shown in Figure 3-2 a DFE mainly comprises of a feed forward section and a feedback section. The feed forward section corresponds to a linear transversal filter whose taps are spaced by reciprocal of the symbol rate. Similarly feedback filter also comprises of transversal filter whose taps are spaced by reciprocal of symbol rate. The input to the feedback filter is fed after the symbol is passed through a decision devise that operates on the previously detected symbols. The function of feedback section is to subtract that portion of ISI produced by previously detected symbol from the estimation of the future detected symbols. Thus DFE reduces the effect of interference. But in the feedback section if the correct symbols are not fed back then the interference level is further increased. So it’s very crucial to decide the correct symbols that should be fed so as to reduce interference.

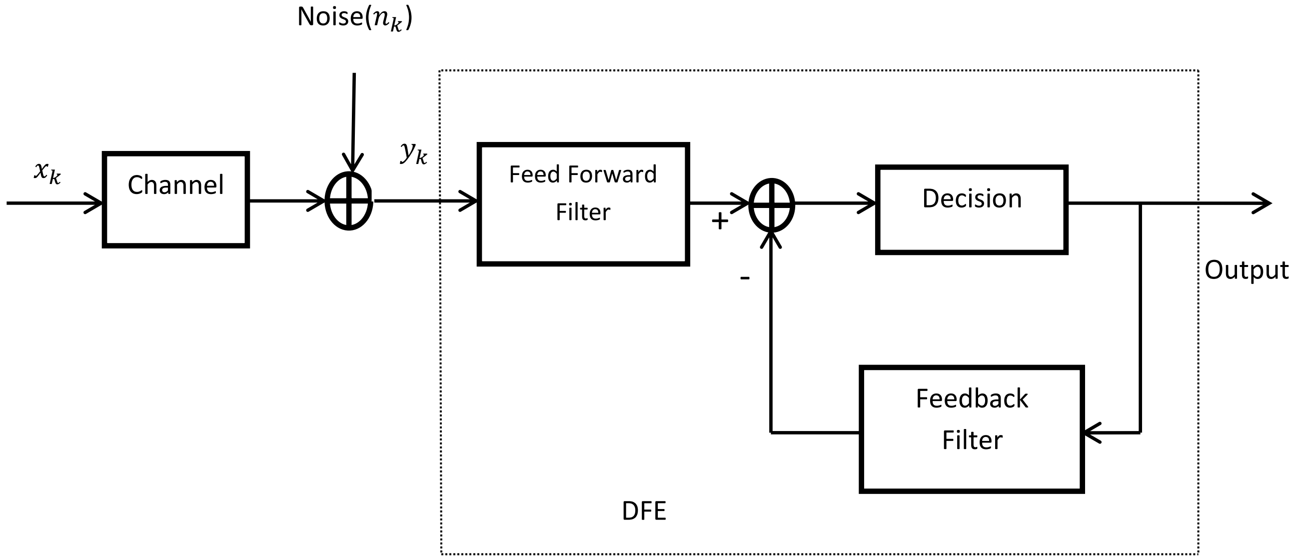

Figure 3-2 Decision Feedback Equaliser

There is a problem to find the number of symbols that should be fed back in DFE. If a single symbol is fed back at a time then the performance of the system increases but it consumes lot of time. So an adaptive threshold technique is used that decides the number of symbols that should be feedback so that we get good performance as well as the complexity is also not high.

For example if the l^{th} symbol of vector x is to be equalised using the above said technique then the output y can be given as

$$\boldsymbol{y} = \boldsymbol{T}^{\boldsymbol{l}}\boldsymbol{x}[\boldsymbol{l}] + \sum_{\boldsymbol{i \epsilon L}} \boldsymbol{T}^{\boldsymbol{i}}\boldsymbol{x}[\boldsymbol{i}] + \boldsymbol{w} \quad \textbf{(3-5)}$$

where T^l is the l^{th} column of the channel code matrix. First term of the equation is the symbol that is to be equalised which is scaled by the channel. Second term represents interference that affects the symbol $x[l]$ and w is additive white Gaussian noise. If some of the symbols $x[i]$ where i ϵ P such that P is subset of L, have been correctly equalised and detected then using that one can compute the summation part i.e. $\sum_{i \epsilon L} T^i x[i]$ and thus interference part is being subtracted from the detected symbol y and interference is reduced. Thus the process is continued iteratively and in each step the system is reduced because in each step the column of the system T is reduced that corresponds to the index of the correctly equalised symbol. Thus, in each iteration the code matrix size T is reduced and we move nearer towards the correct symbol.

The main problem with the above method is to determine which symbols are correctly equalised and are to be fed back. While feeding back if a wrong symbol is used instead of correct one, then interference further increases and error propagation occurs. Another problem with this technique is that, it is difficult to judge the number of symbols that are used to feedback in each iteration. Safest case is to feedback one symbol at a time. But the computational time increases for a system having larger blocks of symbols. In a good SNR condition, most of the symbols would be correct, so in that case feeding back one symbol at a time would be wastage of resource. While if more number of symbols are fed back at a time then there is a chance of feeding back the wrong symbol which results in increasing interference. Thus if one considers from performance point of view, then fewer number of symbols are to be feedback that are guaranteed to be correct while if one considers from computational point of view then more number of symbols are fed so that the number of iteration is reduced.

So we go for an adaptive thresholding technique. Here, we use a ZF or MMSE to compute the estimate of the transmitted symbol from received symbol y. ZF gives the exact channel values but only in the absence of noise but at high noise environment its performance degrades. So to overcome the above said we go for MMSE which is given by

$$\boldsymbol{x_{mmse} = A^H(AA^H + \sigma^2 I)^{-1}} \tag{3-6}$$

where A^H is the Hermitian matrix of A which is the product of code matrix T and channel matrix H.

Next part of DFE comprises of a decision devise i.e. we have to decide which symbols are to be used as output and which are to be feedback for further processing. For that we have to find the sparse solution. Sparsity means that there is a relatively small proportion of relatively large entry or it may also means that there are relatively a small proportion of non-zero entries. There are a number of existing greedy algorithms to find the sparse solution.

3.2 Greedy Algorithm

The greedy algorithms [17] as shown in Figure 3-3 are broadly classified as:-

- Matching Pursuit
    - Matching Pursuit (MP):- It was first proposed by Mallat and Zhang in the year 1993. It finds the best matching projection on multi-dimensional data.

- Orthogonal Matching Pursuit (OMP):- The main difference between MP and OMP is that here the symbols are orthogonally projected.
- Stage wise Orthogonal Matching Pursuit (StOMP):- Compared to OMP these have faster computational time as here more than one symbols are fed inside the model at a time. It also has an advantage over OMP of taking fixed number of stages

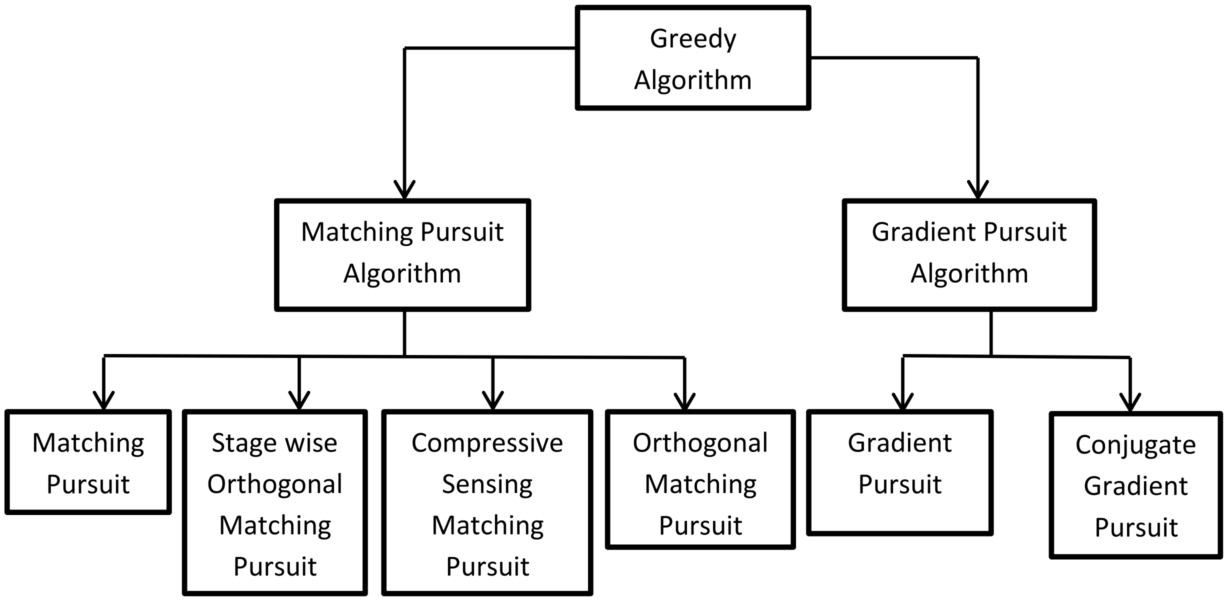

Figure 3-3 Different types of Greedy Algorithm

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- Compressive Sampling Matching Pursuit (CoSaMP):- Here the best matching solution is computed by sorting the first largest 2K entries of the inner product, K being the non-zero coefficients.

- Gradient Pursuit
  - Gradient Pursuit (GP): This algorithm computes the new estimate by choosing gradient direction and step size while solving the least square by MP. The main disadvantage of using this algorithm is the extra computational cost of computing the step size.
  - Conjugate Gradient Pursuit (CGP): This algorithm computes the conjugate of gradient direction and conjugate of step size.

From paper [17] that deals with the greedy algorithm, it is found that the reconstruction error of StOMP and CoSaMP is significantly better than other algorithm such as MP, OMP, GP and CGP algorithm in case of small sparity and more measurements. But if we go for faster computation then gradient pursuit are better compared to matching pursuit algorithm. Here we have used StOMP algorithm for finding the sparse solution.

3.3 Stage wise Orthogonal Matching Pursuit

StOMP [18] is an iterative thresholding algorithm that is used to find the sparse solution. Its schematic diagram is shown in Figure 3-4. This algorithm starts with an initial solution x_0 to be 0 and initial residual r_0 which is considered to be the output *y*. A counter is set to one and after each step its value is increased. Here the algorithm also finds the locations of the non-zero values of the transmitted symbol x_0.

The residual correlation of the i^{th} stage is given by

$$\boldsymbol{c_i = \phi^T r_{i-1}} \qquad \textbf{(3-7)}$$

$\emptyset$ is the channel matrix and r_{i-1} is the residual of the previous stage.

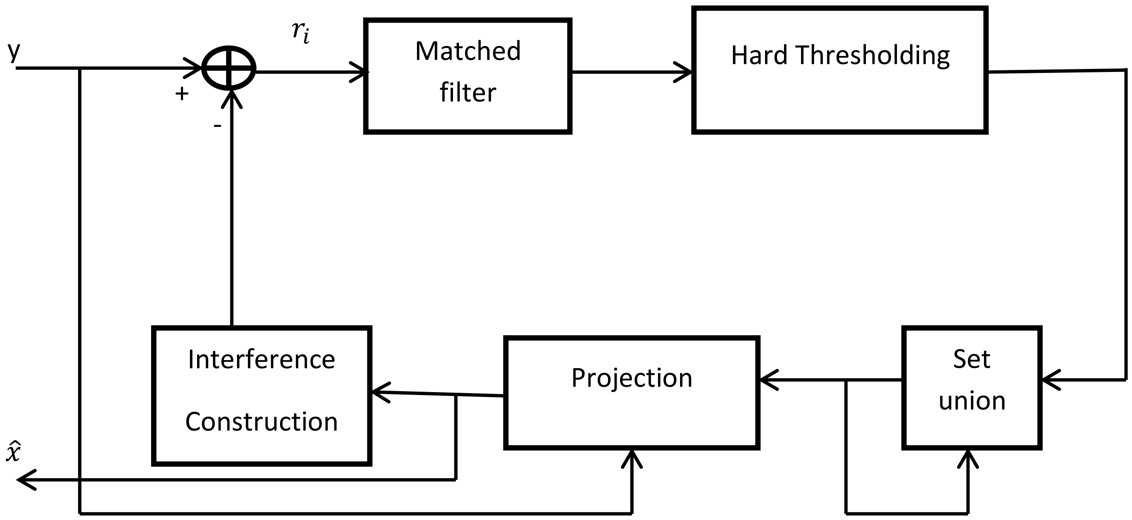

Figure 3-4 Schematic diagram of StOMP Algorithm

After computation of the residue, hard thresholding is performed to find the non-zero location of the solution x_0. The thresholding is based on gaussianity. After that the subset of the newly selected estimate are merged with the previously estimated solution. Next the output is projected to find the estimate coefficient which is given by

$$\hat{\boldsymbol{x}} = (\boldsymbol{\phi}^T \boldsymbol{\phi})^{-1} \boldsymbol{\phi}^T \boldsymbol{y} \quad \textbf{(3-8)}$$

Next the residue is updated using the equation

$$\boldsymbol{r_i} = \boldsymbol{y} - \boldsymbol{\phi} \boldsymbol{x_i} \quad \textbf{(3-9)}$$

A stopping condition is checked, and if we do not reach the stopping condition then the counter is incremented but if we reach to end then $\hat{x}_i = x_i$.

3.4 System model

Here we consider the WCDMA based system model where K numbers of asynchronous users are transmitting the modulated symbol [19]. The received noiseless signal is given as

$$\boldsymbol{y(t)} = \sum_{k=1}^{K}\sum_{m=1}^{M} \gamma_k \boldsymbol{s_k(m)}\, \boldsymbol{a_{k,m}(t - mT_d)} \tag{3-10}$$

where $s_k(m)$ is the m^{th} data transmitted by the k^{th} user, γ_k being the power associated by the k^{th} user and T_d being the symbol duration. The symbol $a_{k,m}(t)$ represents the convolution of the transmitted spreading waveform with the channel impulse response which is given by

$$\boldsymbol{w_{k,m}(t)} = \sum_{j=1}^{L_k} \boldsymbol{T_{k,m}(j)\, h(t - jT_c)} \tag{3-11}$$

Similarly if we consider the matrix model for the noise free channel then the equation is given as

$$\begin{aligned}\boldsymbol{y_{km}} &= \sum_{m=1}^{M_k} \boldsymbol{T_{km} h_{kj} s_{km}} \\ &= \boldsymbol{T_k (I_{M_k} \otimes h_k) s_k}\end{aligned} \tag{3-12}$$

Here T_{km} corresponds to the spreading code of k^{th} user and m^{th} symbol and h_k is the multipath fading channel of k^{th} user. It is assumed that the k^{th} user has a relative delay of D_k chip with respect to the reference of the received symbol. As shown in Figure 3-5, $D_k + (m-1)G_k$ values of the first column represents zeros followed by the spreading code value and then the rest values are again filled by zeros so that it fills the entire chips of the slot. Similarly the second column of the code matrix represents the second multipath component with corresponding delay and spreading code and so on.

Considering noise, above equation becomes

$$\boldsymbol{y} = \boldsymbol{THs} + \boldsymbol{w} \tag{3-13}$$

$$\boldsymbol{T} = [\boldsymbol{T_1}, \boldsymbol{T_2}, \boldsymbol{T_{3,}} \dots, \boldsymbol{T_M}] \tag{3-14}$$

$$\boldsymbol{H} = \boldsymbol{diagonal}(\boldsymbol{I_{M_1}} \otimes \boldsymbol{h_1}, \boldsymbol{I_{M_2}} \otimes \boldsymbol{h_2}, \dots, \boldsymbol{I_{M_k}} \otimes \boldsymbol{h_k}) \tag{3-15}$$

H being the knocker diagonal matrix between identity matrix I and the channel matrix h , s being the symbols transmitted and w is the additive white Gaussian noise.

3.4.1 Procedure

In decision feedback algorithm first we have to compute the correctly equalized symbols. A solution that is close to the actual transmitted vector will give accurate information for our decision feedback rule. The simplest way to compute the equalized

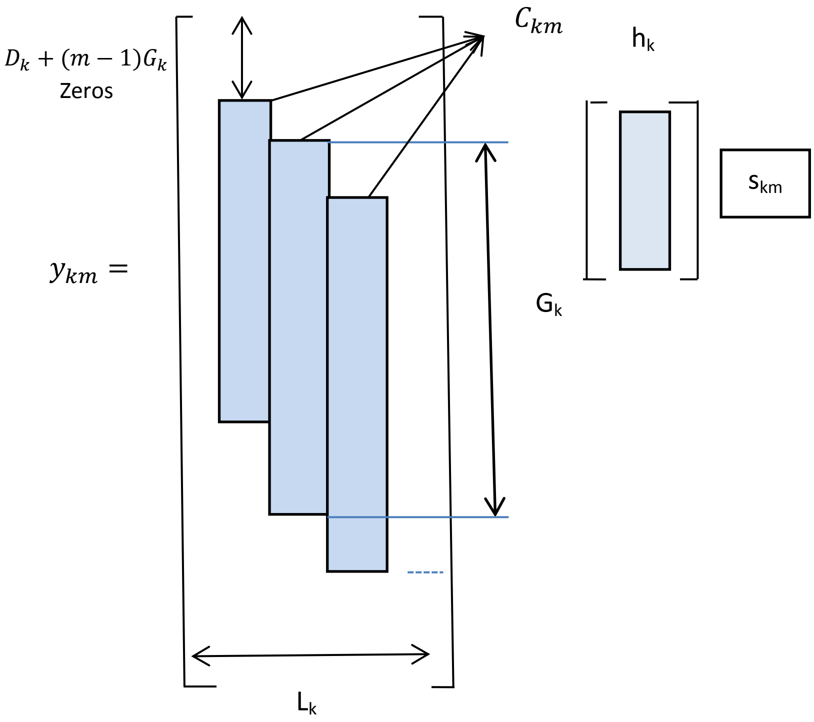

Figure 3-5 Spreading Code Matrix

solution is using MMSE solution i.e.

$$\boldsymbol{x_{mmse} = A^H(AA^H + \sigma^2 I)^{-1} y} \tag{3-16}$$

Adaptive thresholding

A number of algorithms exist to find the sparsest solution for large, underdetermined systems. As discussed above here we have used Stage wise Orthogonal Matching Pursuit (StOMP), which is an iterative thresholding algorithm for finding the sparse solution. StOMP is used to compute the correct symbols from the current solution so that the correct symbols should be feedback in order to reduce interference for the next iteration. This is generally preferred as it has the advantage of having better computational error.

First we compute the residual i.e. for i^{th} iteration, if the initial solution is x_i, that was not correctly equalised in previous iteration, and then the residual can be given as

$$\boldsymbol{r_i} = \boldsymbol{y_i} - \boldsymbol{A_i x_i} \quad \textbf{(3-17)}$$

where A_i is the matrix that is obtained by leaving the column, from the matrix $\boldsymbol{A}$ that corresponds to the correct symbols in previous iteration. y_i is the observation in the i^{th} iteration, obtained by subtracting correctly equalised symbols in the previous iteration. Initially i.e. in the first iteration, the dimension of solution x_0 and error estimate e_0 are both M dimensional, but as the iteration increases their size decreases.

Then we compute the estimate of error. The error estimate for i^{th} iteration is

$$\boldsymbol{\hat{e}_i} = \boldsymbol{A_i^H r_i} \quad \textbf{(3-18)}$$

Here e is a sparse, spiky signal added with noise and is given as

$$\boldsymbol{\hat{e}_i} = \boldsymbol{e_i} + \boldsymbol{w_i} \quad \textbf{(3-19)}$$

As we have obtained e, now we have to compute a threshold which will help us to determine which entries in $\boldsymbol{e}$ are small enough to be considered just noise and thus should be fed back.

It's known that the maximum of the random Gaussian sequence is bounded by the given equation as:

$$\mathbf{max}(|\boldsymbol{g}[\boldsymbol{i}]|) < \sqrt{\boldsymbol{2\sigma lnM}} \quad \textbf{(3-20)}$$

with high probability. So if we had an unknown spike function embedded in AWGN, the right hand side of the above Equation (3-20 would be threshold that would distinguish between the spike and the noise.

The level of sparsity is determined by the number of errors we make in our solution. This number will be different in each iteration, so our threshold has to adapt appropriately. We cannot know the number of error, ro that occurred in our current solution but we need to know the level of sparsity of the actual error vector e . We obtain this estimate in the i^{th} iteration as

$$\boldsymbol{ro} = \frac{||\boldsymbol{r_i}||^2}{\boldsymbol{s}_{\boldsymbol{min}}^2} \quad \textbf{(3-21)}$$

$\boldsymbol{s_{min}}$ is the minimum distance among symbols for the used constellation. The Equation (3-21 is used because the most likely errors are caused by mapping symbols into its nearest neighbour symbols so the nonzero entries in $\boldsymbol{e_i}$ is typically of size $\boldsymbol{s_{min}}$. As we don't know $\|\boldsymbol{e_i}\|$ we consider $\|\boldsymbol{r_i}\| = \|\boldsymbol{e_i}\|$ and obtain

$$\boldsymbol{ro} = \frac{||\boldsymbol{r}||^2}{\boldsymbol{s}_{\boldsymbol{min}}^2} \quad \textbf{(3-22)}$$

as the estimate of sparity of e_i.

The threshold for i^{th} iteration is given as

$$\boldsymbol{th_i} = \sqrt{\boldsymbol{2ln}\left(\frac{\boldsymbol{m_i}}{\boldsymbol{ro_i}}\right)}\frac{||\boldsymbol{r_i}||}{\sqrt{\boldsymbol{m_i}}} \tag{3-23}$$

Using the above threshold location of the correct symbols is computed so that they can be subtracted in order to reduce interference. The symbols having estimated error less than the threshold value are considered to be the correct values and represented by I_i. Thus each iteration the estimated error, residual value and threshold is computed and the corresponding position of correct symbols is subtracted. As the number of iteration increases the size of A_i is reduced. The iteration process is repeated till all the values of I_i is over.

3.4.2 Algorithm

- *Step 1.*

Estimate of the transmitted symbol is computed using the received symbol.

- *Step 2.*

Residual is computed using the formulae

$$\boldsymbol{r_i} = \boldsymbol{y_i} - \boldsymbol{A_i}\widehat{\boldsymbol{x}}_{\boldsymbol{i}} \tag{3-24}$$

- *Step 3.*

Estimated error is computed using

$$\hat{\boldsymbol{e}} = \boldsymbol{A^T}\boldsymbol{r_i} \tag{3-25}$$

- *Step 4.*

Threshold value is computed using

$$\boldsymbol{th_i} = \sqrt{\boldsymbol{2ln}\left(\frac{\boldsymbol{m_i}}{\boldsymbol{ro_i}}\right)}\frac{||\boldsymbol{r_i}||}{\sqrt{\boldsymbol{m_i}}} \tag{3-26}$$

- *Step 5.*

Location of the symbols is found those have the estimated error value less than the threshold value i.e. the position of correct symbols are computed.

- *Step 6.*

Interference caused by the correct symbol is removed by using the equation

$$\boldsymbol{y_{i+1}} = \boldsymbol{y_i} - \boldsymbol{A_i}\left(:, \boldsymbol{I_{c_i}}\right)\boldsymbol{\hat{x}_i}(\boldsymbol{I_{c_i}}) \tag{3-27}$$

Here $A_i\left(:, I_{c_i}\right)$ corresponds to the matrix having all rows but only the columns having index I_{c_i} i.e. the correct symbol index.

- *Step 7.*

The code matrix A_{i+1} is updated by leaving the column that corresponds to the index set I_{c_i}.

- *Step 8.*

If the index set I_{c_i} is not exhausted then we move to step 1 else move to end.

- *Step 9.*

End.

3.5 Results and Discussion

The BER performance of WCDMA is evaluated for multipath channel. The BER performance is evaluated by using Monte-Carlo simulation. This simulation is used to compute the probability of error in the receiver. Here the symbols that are transmitted is compared with the symbols that are received at the receiver section. If both are equal means no error has occurred while they are not equal then the symbol is corrupted. Thus BER is calculated using the formulae

$$BER = \frac{Number\ of\ errors}{Total\ symbols\ transmitted} \tag{3-28}$$

Here we have used BPSK modulation. The simulation was performed by varying the load in the system and also by varying the spreading factor. The results for different channel were considered i.e.

- Minimum Phase channel: This is the causal and stable channel that has all the zeros lying within the unit circle. Zeros are the values in the system for which the system tends to zero. This means all the zeros have a value less than 1. Here we considered the channel to be $1 + 0.5z^{-1} + 0.2\ z^{-2}$.
- Maximum phase channel: In this channel all the zeros lie outside the unit circle. These are named so because they have maximum group delay of the set of systems that have the same magnitude response. Here we have considered the maximum channel to be $0.33 + 0.667z^{-1} + z^{-2}$.
- Mixed phase channel: This type of channel has some of its zeros inside the unit circle while others lying outside the unit circle. Thus, its group delay is neither minimum nor maximum but somewhere between the group delay of the

minimum phase and the maximum phase equivalent system. The channel values were $0.407 + 0.815\,z^{-1} + 0.407z^{-2}$.

Figure 3-6, Figure 3-7 and Figure 3-8 shows the simulation result for rake receiver, de-correlating rake receiver, regularised receiver and regularised receiver with thresholding in minimum phase channel at different active load. Here the SF used was 64. It was found that when there are 2 active users in that case the regularised receivers with thresholding gives much better performance compared to the other three receivers. It is found that we get a gain of about 5dB when compared with rake receiver and about 4 dB when compared with de-correlating rake receiver and regularised rake receiver at BER of $\mathbf{10^{-3}}$. In further figures it was found that even if the load is increased then also we get better performance in regularised rake receiver with thresholding when compared with the other receivers.

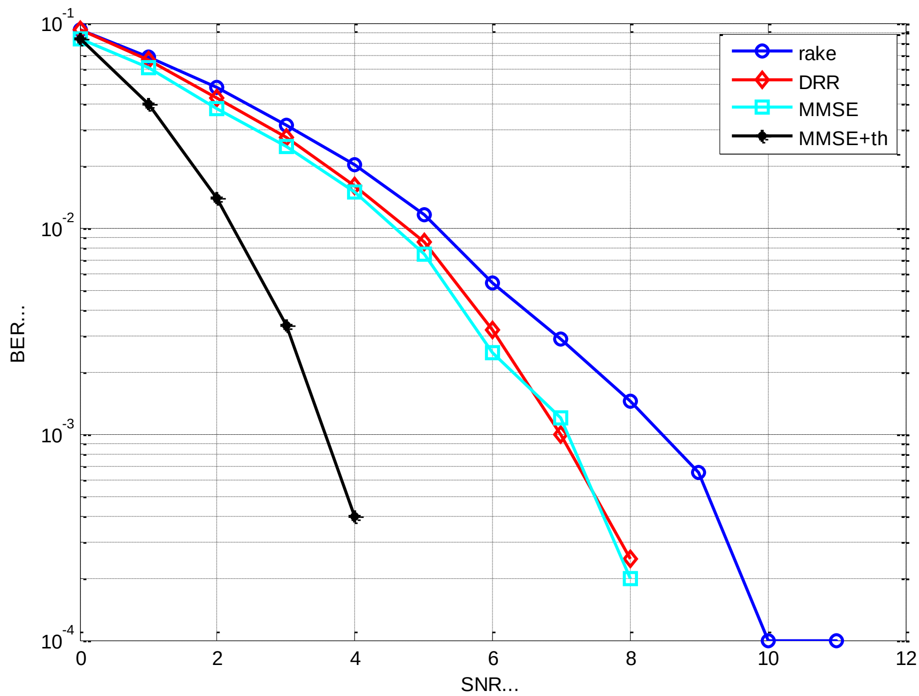

Figure 3-6 BER vs. SNR for 2 user in minimum phase channel of$\mathbf{1 + 0.5z^{-1} + 0.2z^{-2}}$ with a SF of 64

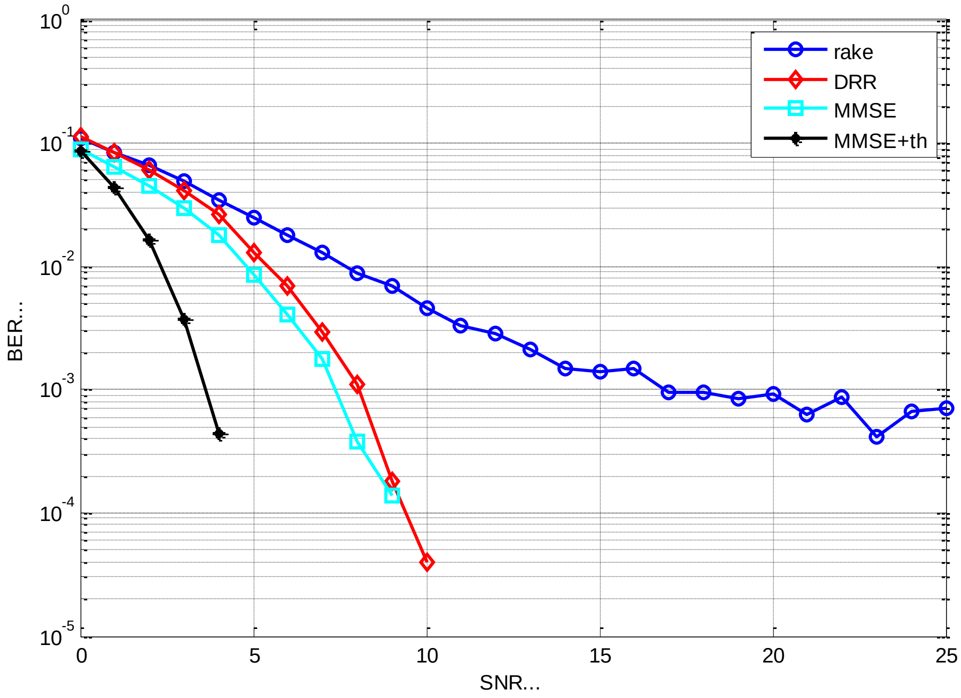

Figure 3-7 BER vs. SNR for 5 users at a SF of 64 in min phase channel of $1 + 0.5z^{-1} + 0.2z^{-2}$

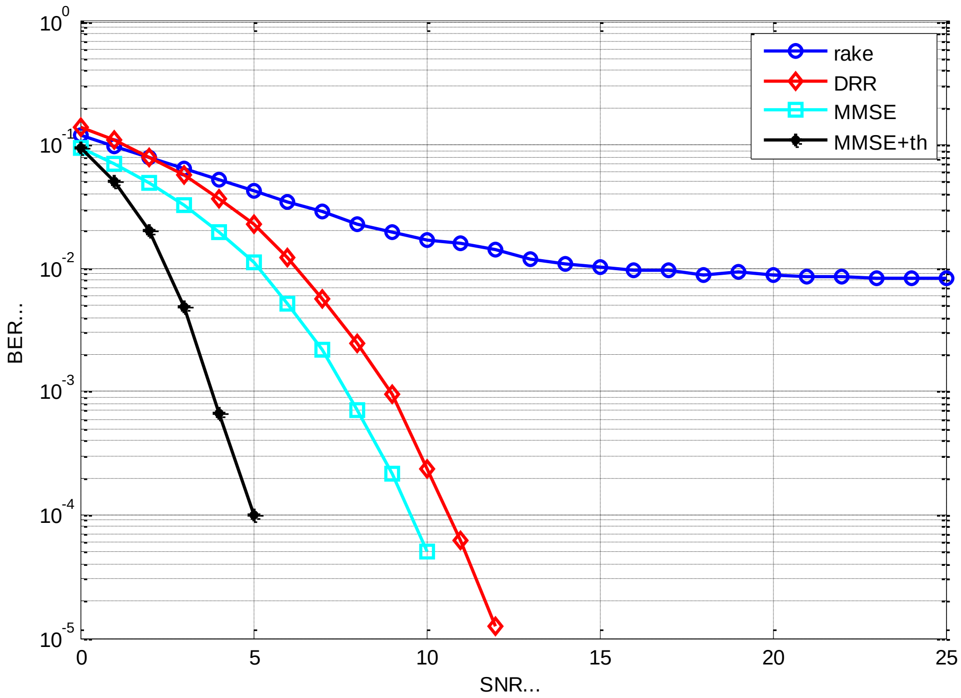

Figure 3-8 BER vs. SNR for 8 users at a SF of 64 in min phase channel of $1 + 0.5z^{-1} + 0.2z^{-2}$

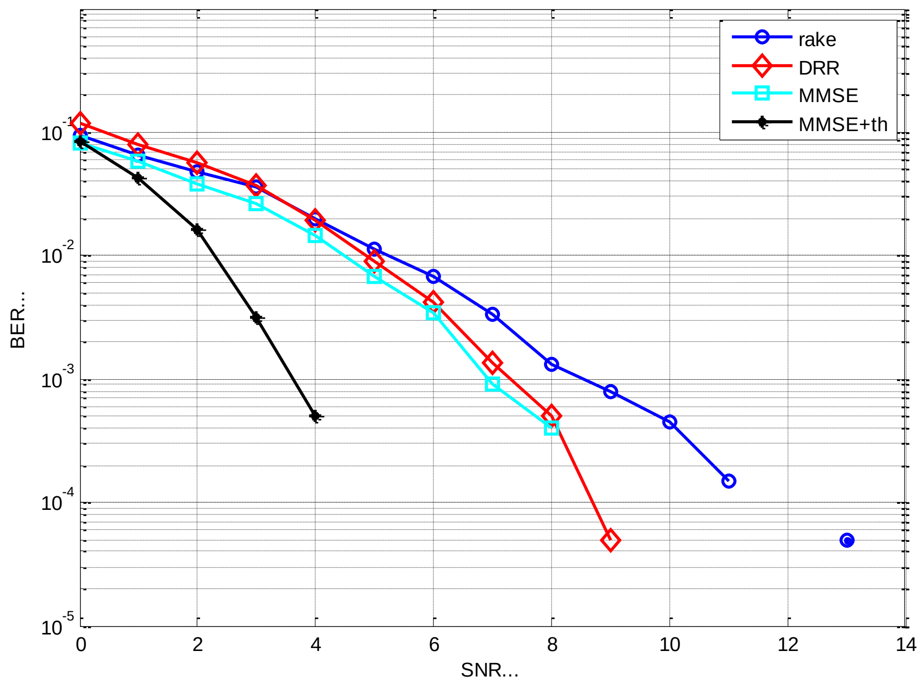

Figure 3-9 BER vs. SNR in maximum phase channel of $0.33 + 0.667z^{-1} + z^{-2}$ at a spreading factor of 64 for 2 users

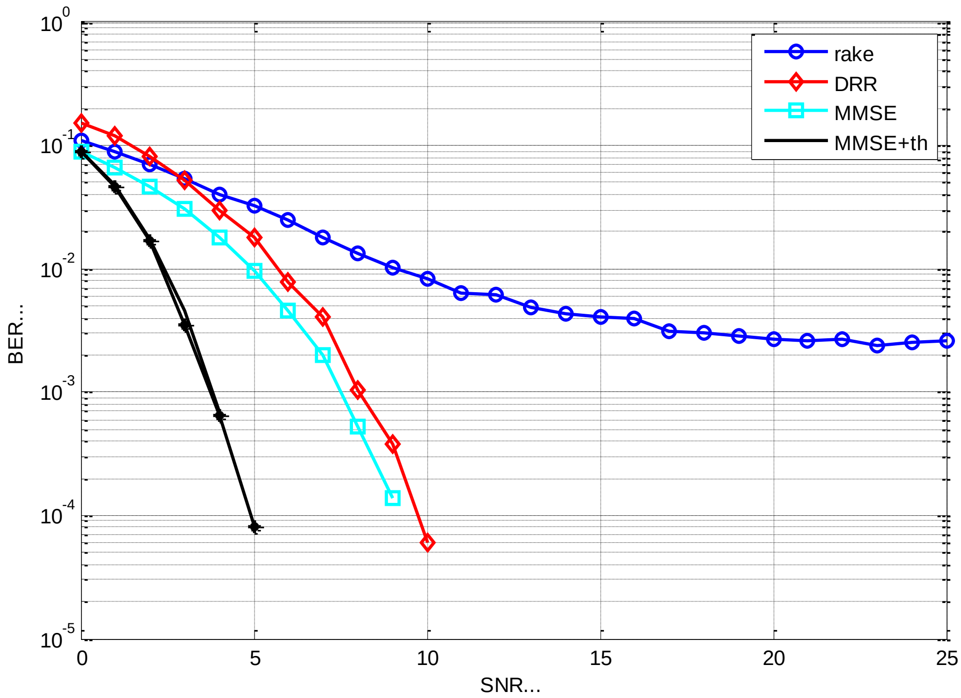

Figure 3-10 BER vs. SNR for 5 users in maximum phase channel of $0.33 + 0.667z^{-1} + z^{-2}$ at a spreading factor of 64

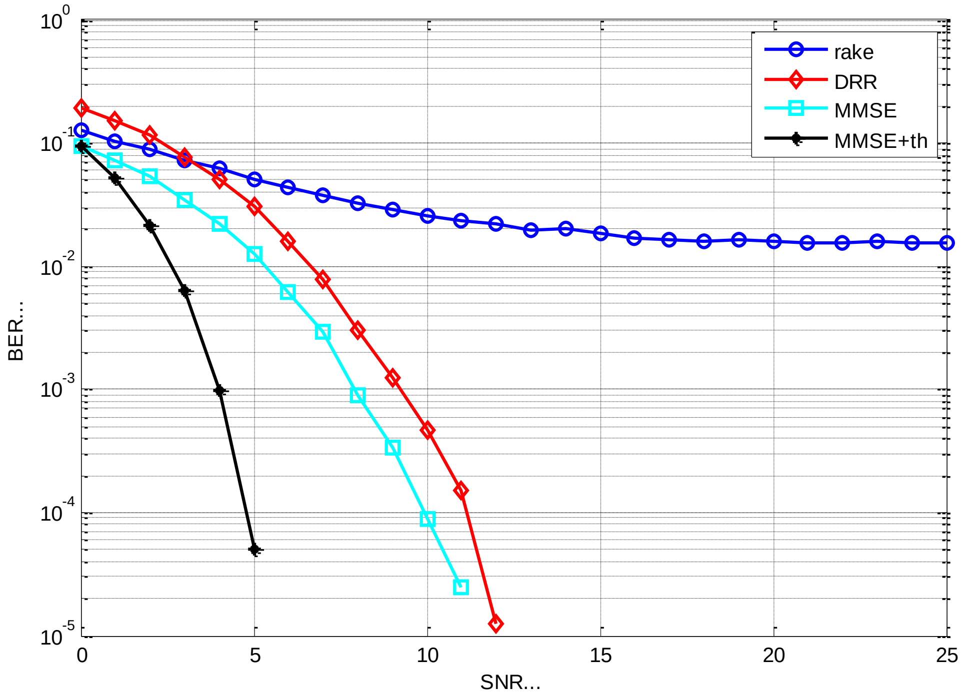

Figure 3-11 BER vs. SNR for 8 users in maximum phase channel of value $0.33 + 0.667z^{-1} + z^{-2}$ at a Spreading factor of 64

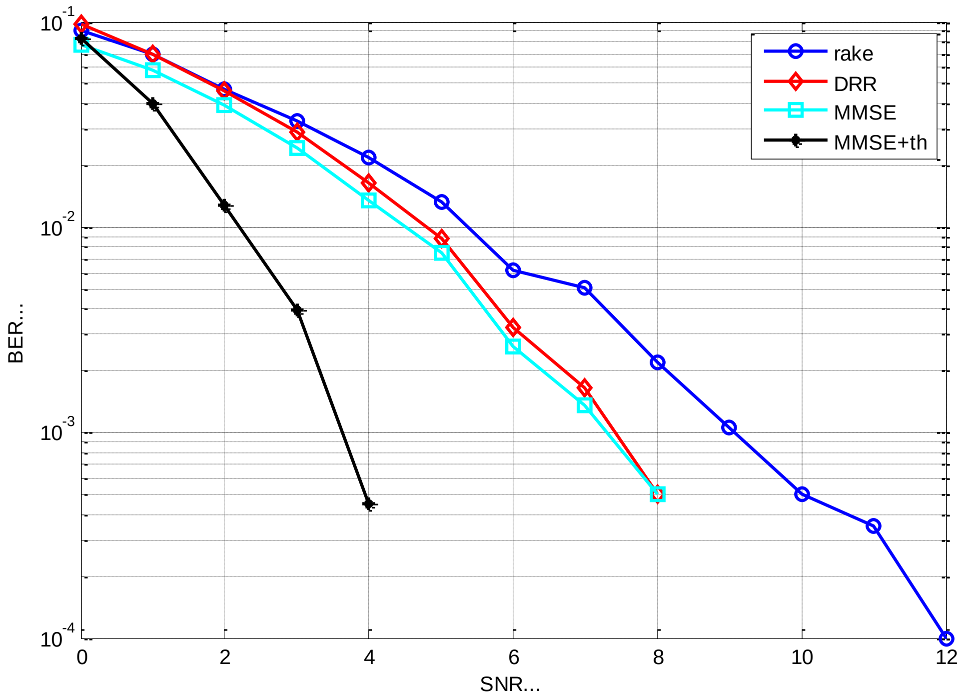

Figure 3-12 BER vs. SNR for 2 users in mixed phase channel having the value of $0.407 + 0.815z^{-1} + 0.407z^{-2}$ at a spreading factor of 64

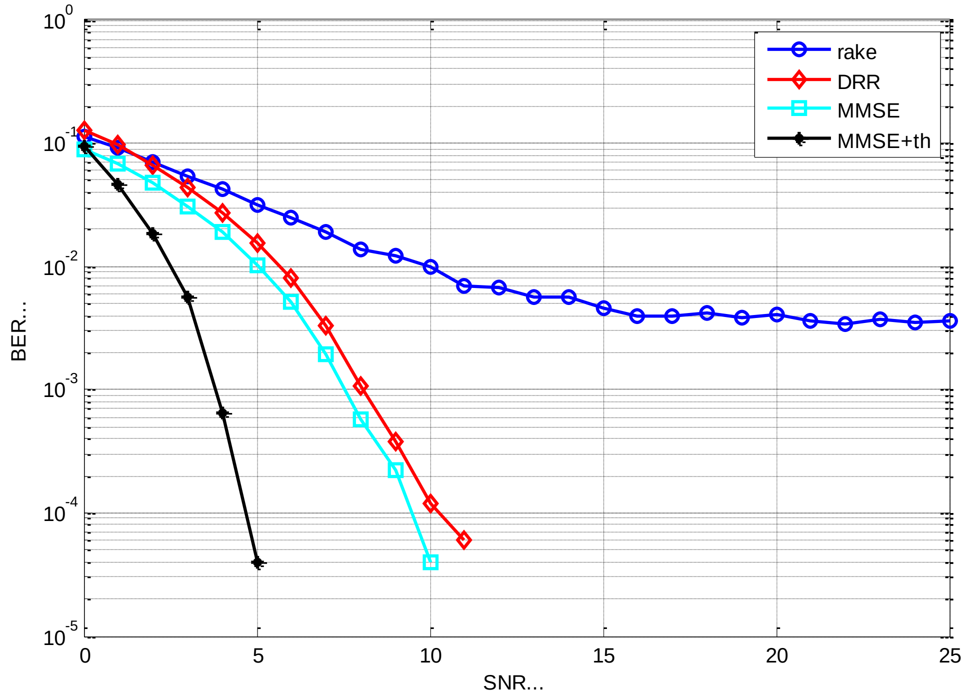

Figure 3-13 BER vs. SNR curve for 5 user in mixed phase channel of value $0.407 + 0.815z^{-1} + 0.407z^{-2}$ with a spreading factor of 64

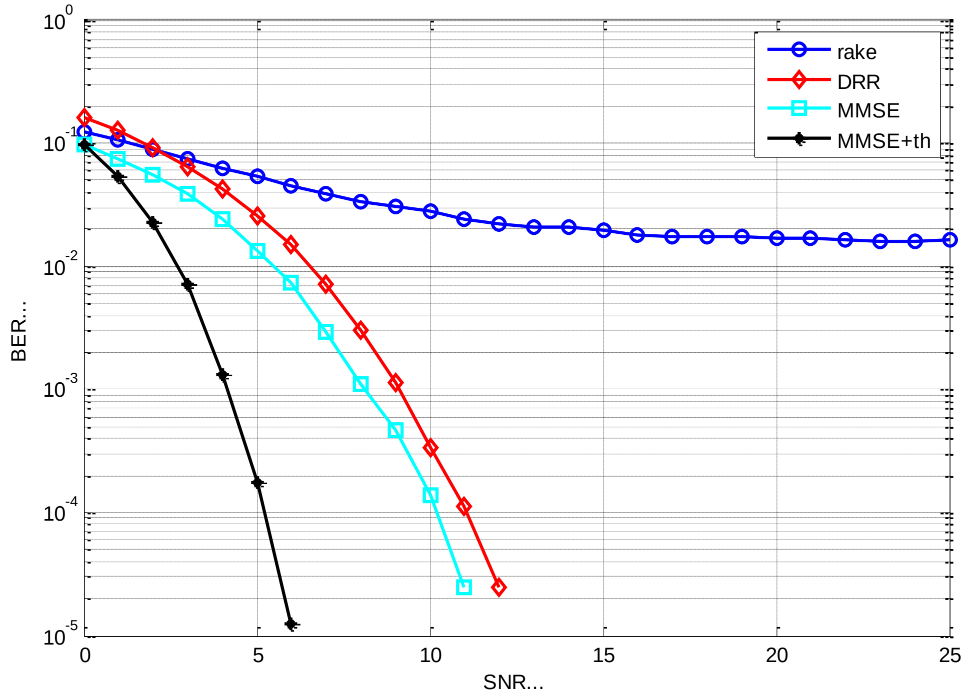

Figure 3-14 BER vs. SNR for 8 users at mixed phase channel having the value of $0.407 + 0.815z^{-1} + 0.407z^{-2}$ with a spreading factor of 64

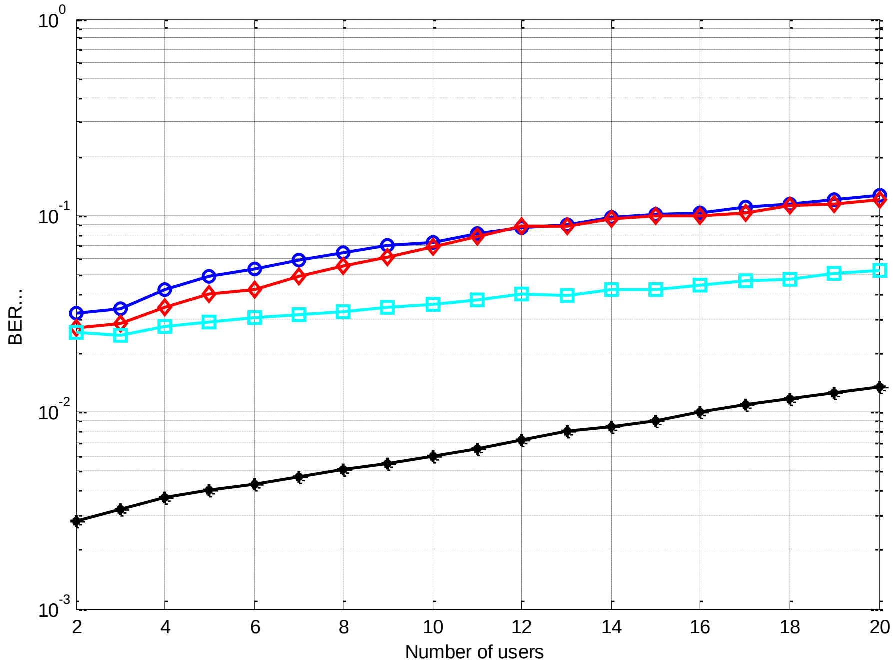

Figure 3-15 BER vs. number of users in minimum phase channel at SNR of 3 dB

Similarly, Figure **3-9**, Figure 3-10 and Figure 3-11 gives a comparative simulation result of different receiver in maximum phase channel and Figure 3-12, Figure 3-13 and Figure 3-14 gives the simulation result in mixed phase channel for 2 users, 5 users and 8 users respectively.

Figure 3-15 shows the BER verses number of user curve for minimum phase channel at SNR of 3 dB. It was found that the given technique has minimum error than other technique and thus gives best performance.

4

Conclusion

This thesis begins with the introduction to wireless communication and WCDMA, reviewed the previously established receiver structures and introduced a new adaptive thresholding technique. A nonlinear receiver was analysed in this thesis.

4.1 Summary

The first chapter deals with the introduction to wireless communication. It also deals with the pro and cons of the communication. Then it gave a brief idea about the different techniques used in different generation of wireless communication. This chapter ends with the outline of the thesis.

Chapter 2 begins with introduction to WCDMA. It gives the different modulation technique used in the uplink and downlink and the type of spreading code used for transmission is shown. Then it gives a basic idea to the working of RR, its advantage and its shortcoming.

Chapter 3 reviewed equalisers and different type of receiver structure. Initially the linear and non-linear equaliser was discussed. A novel non-linear receiver was presented for WCDMA. A DFE with adaptive thresholding technique was used for channel equalisation application. It used the StOMP algorithm for computing the sparity error solution that gives an idea of feeding back symbols in case of DFE. Monte Carlo simulation compared the receiver structure with the established linear and non-linear receivers. The simulation results showed that the proposed receiver outperforms the MMSE and other receiver techniques.

4.2 Conclusion

The thesis gives a brief idea about different equalisation algorithms for WCDMA system. A de-correlating rake receiver was used in the receiver front end. This works well only in the noise-free environment. A MMSE equaliser works better than the de-correlating receiver in the noisy situation. At severe interference condition non-linear equaliser is preferred as they lower the interference.

Here a decision feedback equalisation algorithm is used that works on adaptive thresholding technique. The adaptive thresholding is used to find the sparsity of the estimated error signal. This gives a theoretical frame work for the multiple feedback symbol selection in each iteration which leads to a very fast convergence and also has low computational complexity.

The simulation results shows that the work in this thesis gives much better performance when compared with the linear equaliser such MMSE. We get a drastically improved result when compared with the rake receiver, de-correlating rake receiver and regularised rake receiver.

4.3 Limitation of Work

In this DFE technique, we are feeding back those symbols whose values are less than the threshold value i.e. the location of the correct symbols are feedback. There is a problem to find the error symbols. Here we have assumed the error and the residual value to be equal. A threshold value is used that helps in separating the error value and the correct symbols is computed by using some of the estimation. Finding the threshold is very crucial as, if the error symbol is fed back then the interference is increased. Here lot of assumption are taken to find the different values. Here the simulation work is done which is limited to software. There is a chance of variation of result in the real world.

4.4 Future Work

The basic problem in developing a WCDMA receiver having better performance than any linear receiver is that it should have some approximation technique. The techniques should have the following constraints i.e. it should have simple receiver structure with as low complexity as possible.

The given work in this thesis is a simple receiver and also gives good performance when compared with the linear receiver. But it has a scope of having further work. It has a chance of further improvement at low SNR region.

Now a day, a lot of compressed sensing and adaptive message passing (AMP) algorithms have been used to improve the performance of iterative threshold algorithm for computing the sparse solutions. There a chance of improving the performance at low SNR by using other algorithms.

Publication

- S.Patel, S.K.Patra, ”Interference Suppression using De-correlating rake receiver in WCDMA”, *International Conference on Emerging Trends in Computing, Communication and Nanotechnology*, Tamil Nadu, India,25-26 March ,2013.(Accepted and Presented)

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Online resources

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