INTERLEAVED DIVISION MULTIPLE ACCESS BASED COMMUNICATION

By

SELVAKUMAR.G

Reg. No. 0920107021

of

KUMARAGURU COLLEGE OF TECHNOLOGY

(An Autonomous Institution affiliated to Anna University of Tech, Coimbatore)

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BONAFIDE CERTIFICATE

Certified that this project report entitled "**INTERLEAVED DIVISION MULTIPLE ACCESS BASED COMMUNICATION**" is the bonafide work of Mr.Selvakumar.G [Reg. no. 0920107021] who carried out the research under my supervision. Certified further, that to the best of my knowledge the work reported herein does not form part of any other project or dissertation on the basis of which a degree or award was conferred on an earlier occasion on this or any other candidate.

Project Guide

Ms.M.Shanthi

Head of the Department

Dr. Rajeswari Mariappan

The candidate with university Register no. 0920107021 is examined by us in the project viva-voce examination held on 21.04.2011

Internal Examiner

External Examiner

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LIST OF ABBREVATIONS

APP	A Posteriori Probability
AWGN	Additive White Gaussian Noise
CDMA	Code Division Multiple Access
CSI	Channel State Information
DSSS	Direct Sequence Spread Spectrum
ESE	Elementary Signal Estimator
ETSI	European Telecommunications Standards Institute
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
ICI	Inter Channel Interference
IS	Interim Standard
IDMA	Interleave Division Multiple Access
ISI	Inter Symbol Interference
MAP	Maximum A Posteriori
MAI	Multiple Access Interference
ML	Maximum Likelihood
MMSE	Minimum Mean Square Error
OFDM	Orthogonal Frequency Division Multiplexing
SNR	Signal to Noise Ratio
TDMA	Time Division Multiple Access
UMTS	Universal mobile telecommunications system

CHAPTER 1

INTRODUCTION

Communication is the basic need of every living being. The cry of a just born baby, the sounds produced by birds and animals are always communicating something to the world around. The communication becomes complete and successful only when it is properly received and understood. Therefore from feelings to expression on one hand and the reception to the right understanding on the other, a whole process is involved in the communication systems. Technology based on basic scientific principles has made this task successful. The untiring efforts of scientists starting from Jagadish Chandra Bose (1895) Marconi (1899) to Tern Berners - Lee (1991) has made the communication so easy that the whole world could be communicated and received through a 'Lap Top' and a 'Palm Top'.

1.1 GENERAL COMMUNICATION SYSTEMS

In Transmitter the message which is to be sent is generated. The generated message is sent to encoder. In Encoder actual message is converted in to symbols for transmission. In this a sequence of characters are put in a specialized format for efficient transmission. If a message is transmitted without encoding then there may be a chance to loss of message. Thus encoding is done to increase the security and reliability.

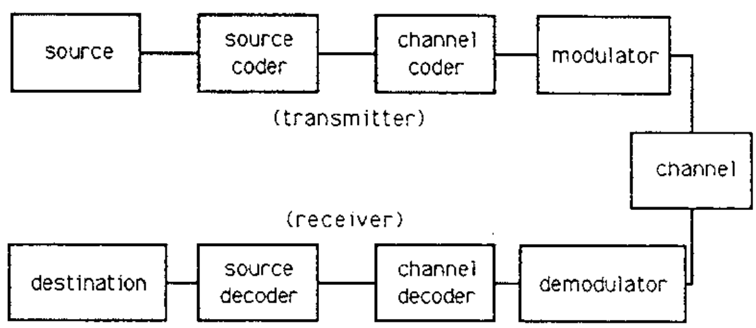


Figure 1.1 Block Diagram of Communication System

Noisy channel is nothing but the medium through which the message is transmitted. Messages are conveyed through this channel. Different channels have different strengths and weaknesses. Each channel has its own frequency and different applications have different operating frequencies.

Decoder is used to decode the encoded message and retrieve the actual message. Decoding must be done correctly. If this part is not performed well then the message which is received might not be correct. So care should be taken while decoding. Receiver is the target to which the information needs to be delivered.

In modern wireless communication systems, the ultimate objective is to enhance the throughput of an overall system. It may be done by employing following process,

- Increasing the Effective Isotropic Radiated power (EIRP)
- Using more Spectrum.
- Reducing system losses.
- Efficient utilization of channel resources.

Of these, increasing EIRP and bandwidth costs high but by using effective multiple access techniques it is possible to gain throughput. Hence many researches focus on various multiple access techniques since the origin of mobile communication. This work also mainly focus on such multiple access technique.

Code Division Multiple Access (CDMA) has made its depth in wireless communication. CDMA offers well-known features: Dynamic channel sharing, soft capacity, frequency reuse, low drop-out rate and large coverage (due to soft handover), ease of cellular planning, robustness to channel impairments and immunity against interference, etc. These advantages are due to spreading the information over a large bandwidth. The performance of conventional CDMA systems is limited by Multiple Access Interference (MAI), as well as Inter Symbol Interference (ISI). Also, the complexity of CDMA multiuser detection has always been a serious concern. The problem can be seen from the angle of computational cost as well complexity of multiuser detection algorithms. The use of signature sequences for user separation is a characteristic feature for a conventional CDMA system.

Interleaving is usually placed between forward error correction (FEC) coding and spreading and is traditionally employed to combat the fading effect. The possibility of employing interleaving for user separation in CDMA systems is possible but the receiver complexity is considered as a main obstacle. A conventional CDMA system (such as IS-95) involves separate coding and spreading operations. Theoretical analysis shows that the optimal multiple access channel (MAC) capacity is achievable only when the entire bandwidth expansion is devoted to coding. This suggests combining the coding and spreading operations using low-rate codes to maximize coding gain. But separation of users without spreading operation is not feasible in CDMA.

One possible solution to this problem is to employ chip-level interleavers for user separation. This work presents an Interleave Division Multiple Access (IDMA) scheme for spread spectrum mobile communication systems, in which users are distinguished by different chip-level interleavers instead of different codes as in a conventional CDMA system.

The scheme considered is a special case of CDMA in which bandwidth expansion is entirely performed by low-rate coding. For convenience, it may be referred as Interleave Division Multiple Access (IDMA). This scheme inherits many advantages from CDMA such as dynamic channel sharing, mitigation of cross-cell interferences, asynchronous transmission, ease of cell planning, and robustness against fading. It also allows a low complexity multiple user detection techniques applicable to systems with large numbers of users in multipath channels. A very low-cost chip-by-chip iterative detection algorithm is explained with complexity independent of user number and increasing linearity with the path number. The advantages of using low-rate coded systems are demonstrated analytically. This will show that the proposed IDMA scheme can achieve better performance than conventional CDMA system.

CHAPTER 2

GENERATION OF MOBILE COMMUNICATION

Over the last decade, the mobile wireless telecommunication industry has undergone tremendous changes and experienced rapid growth. The reason behind this growth is the increasing demand for bandwidth hungry multimedia applications. This demand for even higher data rates at the user's terminal is expected to continue for the coming years as more and more applications are emerging. Therefore, current cellular systems have been designed to provide date rates that range from a few megabits per second for stationary or low mobility users to a few hundred kilobits to high mobility users. This data range is insufficient with respect to the increasing demand for high date rates to high mobility users. Throughout history, the mobile wireless communications has taken on multiple generations depending on the technology used and services provided. This is shown in table 2.1.

Table 2.1 Generation of mobile communication

Technology	Services	Standard	Data Rate
1G	Analog voice	AMPS, NMT, etc.	1.9kbps
2G	Digital voice, short messages	TDMA, CDMA, GSM	14.4 Kbps
2.5G	Higher Capacity packetized data	GPRS, EDGE	384 kbps
3G	Broadband data	WCDMA, CDMA2000	2Mbps
4G	Multimedia data	?	100 Mbps

2.1 FIRST-GENERATION SYSTEMS

First Generation Systems (1G) means the system which used the network of analogue traffic system. AT&T is the first company in North America to introduce first generation system to the customers in during the year of early 1980's. AT&T named the system as Advanced Mobile Phone Service (AMPS). Gradually AMPS technology has been introduced to the countries of South America, Australia and China. 1G constructs the primary architecture of cellular communications and clarifies lots of foundational obstacle, such as adoption of cellular architecture, multiplexing frequency band, roaming across domain, non-interrupted communication etc. First Generation System wasn't able to support lot of services to the customer; primary goal was to support voice chat.

2.2 SECOND-GENERATION SYSTEMS

People started to adopt First Generation AMPS mobile communication in a rapid way. High volume of users started to warn the slower analogue system. Developers started to think a new system which will provide higher quality signals. It's the time to develop Second-generation systems, which will satisfy high volume of customer's needs. 2G systems have been promote to provide higher quality signals, high data rates for support of digital communication, and bigger capacity. 2G systems will use digital technology which will guarantee more accurate signals where as analogue communication was the technology for the First Generation system. Whereas, both of the system use digital signaling to establish connection from radio towers to the telephone subscriber. Second Generation system could be divided in to TDMA or CDMA standard according to the multiplexing they use. The main 2G standards are: GSM, iDEN, IS-136 are the example of TDMA-based second generation standards. CdmaOne which is also called IS-95 is CDMA-based. .

2.3 THIRD-GENERATION SYSTEMS

Once various kinds of 2G systems has been marketed, new people started to show their interest into the cellular communication. Demands of new subscriber for higher data rate increased. It's the time to implement new system which will provide more data rates. The primary goal of the Third-generation (3G) mobile communication is to satisfy more high-speed technology which will higher data rates along with multimedia, data, and video in addition to voice. International telecommunication Union defined their outlook of requirements of the 3G cellular communication in the year of 2000

European Telecommunications Standards Institute (ETSI) to establish a UMTS (Universal mobile telecommunications system) as Europe's 3G cellular standards. One of these is known as Wideband CDMA (W-CDMA). Cdma-2000 is also developed according to the specification of CDMA is the North American version. Because of individual chip and technology of multi-carrier on cdma-2000, W-CDMA differs from cdma-2000. Third generation systems are an extension on the complexity of second generation systems and are already introduced. The system capacity is expected to be increased to over ten times original first generation systems. This is going to be achieved by using complex multiple access techniques like Interleaved Division Multiple Access (IDMA), or an extension of CDMA, and by improving flexibility of services available.

2.4 FOURTH-GENERATION SYSTEMS

This latest standard of telecommunication is focused to aggregate and replace the 3G standard. Connect information anywhere, anytime, with a seamless communication to a broad range of information and services, and receiving a high structure of information, data, pictures, video, and so on, are the keys of the 4G communication. The future 4G basis will consist of a set of broad communication using Internet protocol as a common protocol so that subscribers are in command because subscriber will be able to select every application and environment

According to the progressive features of cellular system, 4G will have higher bandwidth, higher data rate, and easier and quicker handoff and will focus on seamless applicability across a multitude of mobile systems and networks. The main focus is integrating the 4G capabilities with all of the existing mobile technologies through advanced technologies. Application adaptability and being highly dynamic are the main features of 4G services of concern to subscribers. These features mean services can be delivered and be available to various subscribers and assist the subscriber in moving traffic, air interfaces, radio environment, and supreme perform of service. Linking to the cellular communication can be transform into multiple forms and layers correctly and easily. The commanding method of access to this pool of information will be the cellular telephone, Personal Digital Assistant, and laptop to seamlessly access the voice communication, high-speed information services, and multimedia broadcast services. The 4G will support most systems from different networks, public to private; company based broadband connection to private areas; and ad hoc networks. The 4G systems will run with cooperation of with 2G and 3G systems, as well as with broadband transmission systems. Further more, 4G systems will provide

Internet Protocol based wireless communication. This entire aspect shows the various range of systems that the 4G defines to satisfy, from satellite broadband to high distance platform to cellular 3G and 3G systems to wireless local loop and fixed wireless access to wireless local area network and personal area network, all with Internet Protocol as the adapting technique .

The details about technology, standards, services and bandwidth of various generations of communications are listed in the Table 2.2.

Table 2.2 Short history of communications evolution.

Technology	**1G**	**2G**	**2.5G**	**3G**	**4G**
Design Began	1970	1980	1985	1990	2000
Implementation	1984	1991	1999	2002	2010
Services	Analog voice, synchronous data to 9.6 kbps	Digital voice, Short messages	Higher capacity, packetized data	Higher capacity, Broadband data up to 2Mbps	Higher capacity, completely IP oriented, multimedia data
Standards	AMPS, TACS, NMT, etc.	TDMA, CDMA, GSM, PDC	GPRS, EDGE, 1xRTT	WCDMA, cmda2000	OFDM, UWB
Data Bandwidth	1.9 kbps	14.4 kbps	384 kbps	2 Mbps	10 Mbps - 20 Mbps
Multiplexing	FDMA	TDMA, CDMA	TDMA, CDMA	CDMA	CDMA, IDMA
Core Network	PSTN	PSTN	PSTN, Packet network	Packet Network	All-IP Networks

CHAPTER 3

MULTIPLE ACCESS TECHNIQUES

Multiple access technique is a technique where various coexisted subscriber use the same bandwidth of radio spectrum. To accommodate more number of subscribers, sharing of the spectrum is needed. There are three main technologies that have been designed so far to share usable bandwidth in radio communication. These are FDMA, TDMA and CDMA.

3.1 FREQUENCY DIVISION MULTIPLE ACCESS

Frequency Division Multiple Access or FDMA is a connection technology that is used by radio systems to divide the radio spectrum among multiple users. The term "multiple access" defines the adapting of the spectrum between subscribers, and the "frequency division" defines the way of sharing, by slicing the radio spectrum between the subscribers by various carrier frequencies. The objective of the Frequency Division Multiple Access (FDMA) is to subdivide bandwidth into a number of narrower band channels. Particular user is appropriate to use particular frequency band in order to receive and transmit data. The time one user is using the particular call no other user could use the same frequency band. The FDMA technique is shown in Figure 3.1.

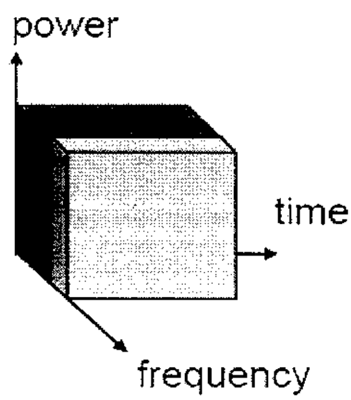


Figure 3.1 Frequency Division Multiple Access

Particular subscriber uses a particular channel from the base station to the mobile phone, which is called forward link. The way back channel from mobile phone to the base station is called reverse link. Either way it's called single way link. Both of the forward link and reverse link are narrower band channels of continuous signal granting analog transmission. The use of the bank channel are different between US and Europe. US use 30 kHz where as Europe uses 25 kHz band channels of bandwidth. Transmission of two channels mobile to base (up) and base to mobile (down) uses true full duplex voice communication link for their transmission.

3.2 TIME DIVISION MULTIPLE ACCESS

Time Division Multiple Access (TDMA) is very recognized and suitable for digital communication. In the TDMA access technique each station lets all the other station to use the whole applicable bandwidth rather than shortened them into smaller bandwidth. But in this technique TDMA defines the time period of the applicable bandwidth so that they don't overlap. Each station has the same opportunity to access to the network, defines a short period of time slot and returns to use for each station as a round robin way. The size of the time slot or frame is as much as the station in a network who can communicate each other either way and simultaneously. In this way every subscriber is allocated to one time slots per frame. Whole data communication is break down in to frames and each frame is break down into time slots where each subscriber is given one time slot. Data stream is used as a guard period to to synchronize.

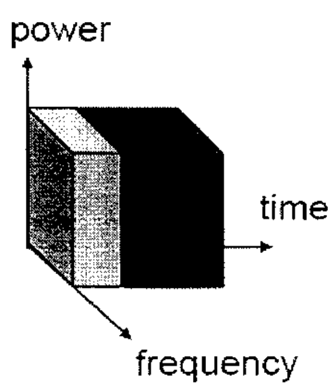


Figure 3.2 Time Division Multiple Access

TDMA technique shown in Figure 3.2 is to associate with the FDMA, in order to split the whole applicable bandwidth into various bandwidths. In order to clip the number subscriber in each channel which concludes lower data rates in the channel, this method is to be used. This method minimizes the response of delay spread on the transmission. Every channel is using FDMA technique with that split into smaller time slot by using TDMA allows multiple subscribers to communicate of the each channel. Most the Second Generation digital communication system used this kind of transmission system.

3.3 CODE DIVISION MULTIPLE ACCESS

CDMA is the most commanding technology of 3G systems. The Code Divisional Multiple Access (CDMA) is the multiple access technique in which separate code (signature) is assigned for each user. This spread spectrum technique does not use frequency channel or time slots. Subscribers of the CDMA system utilize the same frequency band and transmit or receive data at the same time. This CDMA uses a technology called Direct Sequence Spread Spectrum (DSSS) which provides better security.

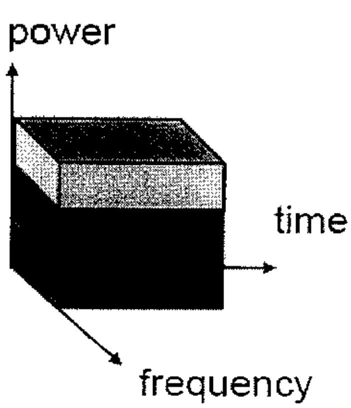


Figure 3.3 Code Division Multiple Access.

Spread spectrum systems tries to utilize as much of available bandwidth as possible, in its efforts to be more resistant to interference. It utilizes a so-called Pseudo Noise sequence (PN) and other special features to accomplish this. There are different types of spread spectrum systems. The two commonly used types are Frequency Hopped (FH) spread spectrums and Direct Sequence (DS) systems. CDMA is a type of DS spread spectrum system, so there will be a concentration on this and no discussion on FH systems will be made.

3.3.1 DIRECT SEQUENCE (DS) SPREAD SPECTRUM SYSTEM

A DS system utilizes the PN-sequence as a kind of code, which makes sure that a user with this special code doesn't suffer from interference disturbance in a high degree as a normal system do. This also enables several different users to use all the bandwidth of the channel at the same time, and thus have a better quality of transmission and higher bit rate among other things. This method of allowing several users with different codes to use the same frequency interval is widely known as a Code Division Multiple Access system. Even though only one transmitter and one receiver is to be used, from now on the system will be called a CDMA system, since the principle is the same.

Pseudo Noise (PN) Sequences

Spread spectrum systems are constructed in same way as conventional systems but there are differences. The main difference is that the system includes two identical pseudorandom sequence generators. One pseudorandom sequence generators interfaces with the modulator at the transmitting end and the second one is located in the receiver. These two generators produce a pseudorandom or pseudonoise (PN) binary-valued sequence that is used to spread the transmitted signal in frequency at the modulator and to despread the received signal in the receiver. This sequence is in our system in fact a sequence of random 1's and -1's that is white with good correlation properties. This sequence is still the same in all transmissions though, i.e. the PN-sequence of length 8 is always the same and so on.

A critical factor of the system is time synchronization of the PN sequence generated at the receiver, and the PN sequence contained in the received spread-spectrum. The PN sequence generated at the modulator is used in conjunction with the PSK modulation to shift the phase of the PSK signal pseudo randomly, at a rate that is an even multiple of the bit rate.

Spreading of The Signal

The spreading of the signal is done by using the pseudo noise sequence. It may be considered as binary information sequence with an information rate of R bits per second is to be spread. The bit interval of the signal is $T_b = 1/R$ seconds and the available channel bandwidth is B_c Hz, where $B_c >> R$. The results for various B_c and N are shown in the Table 3.1.

At the modulator the bandwidth of the information signal is expanded to $W = B_c$ Hz by shifting the phase of the transmitted signal pseudo randomly at a rate of W times per second according to the pattern of the PN generator. The effective bit rate will therefore depend on the available bandwidth B_c, and the length of the PN-sequence N as

$$R_b = B_c * \left(\frac{1}{(1+N)} \right)$$

Table 3.1 Effective bit rates

[illegible]		
B_C (kHz)	N	R_b (kHz)
2	8	0.22
	16	0.12
	32	0.06
5	8	0.56
	16	0.29
	32	0.15
10	8	1.11
	16	0.59
	32	0.30

The information-bearing signal can be expressed as

$$v(t) = \sum_{n=-\infty}^{\infty} a_n g_T(t - nT_b)$$

where $\left\{a_n = \pm 1, \quad \infty < n < \infty\right\}$ and $g_t(t)$ is a rectangular pulse of duration T_b. This signal is multiplied by the signal from the PN sequence generator.

This sequence can be expressed as

$$c(t) = \sum_{n=-\infty}^{\infty} c_n p(t - nT_c)$$

Where $\{c_n\}$ represents the binary PN code sequence of ±1 and p(t) is a rectangular pulse of duration T_c. This multiplication is the actual spreading of the information-bearing signal. It is shown in the Figure 3.4.

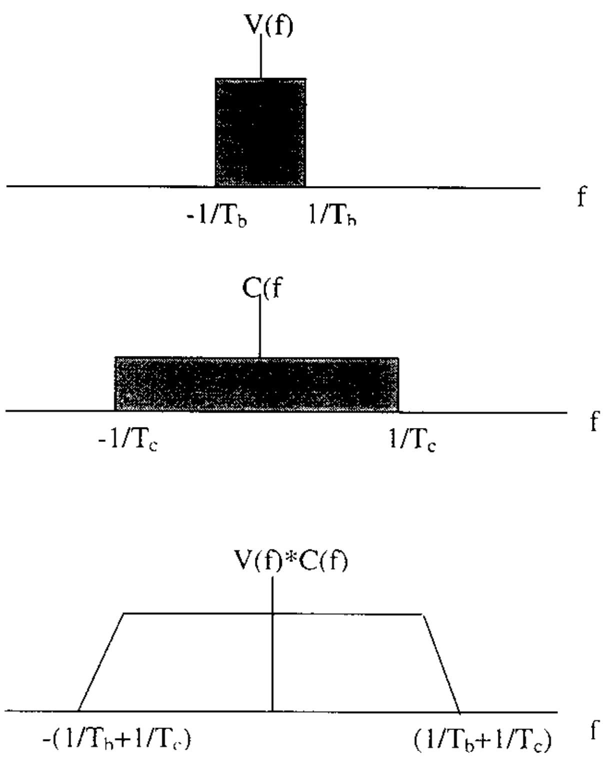


Figure 3.4 Spreading of Signal

The product signal v(t)c(t) is then used to amplitude modulate the carrier $A\cos(2\pi f_c t)$ and, thus generate the DSB-SC signal .

$$u(t) = A_c v(t)c(t)\cos(2\pi f_c t)$$

Because v(t)c(t) = ±1 for any t, the carrier-modulated signal may also be expressed as

$$U(t)=A_c cos(2\pi f_c t + \theta(t))$$

where θ(t) = 0 when v(t)c(t) = 1 and θ(t) = π when v(t)c(t) = -1. Therefore it can easily be seen that the transmitted signal is actually a BPSK-modulated signal. This procedure shown in the Figure 3.5.

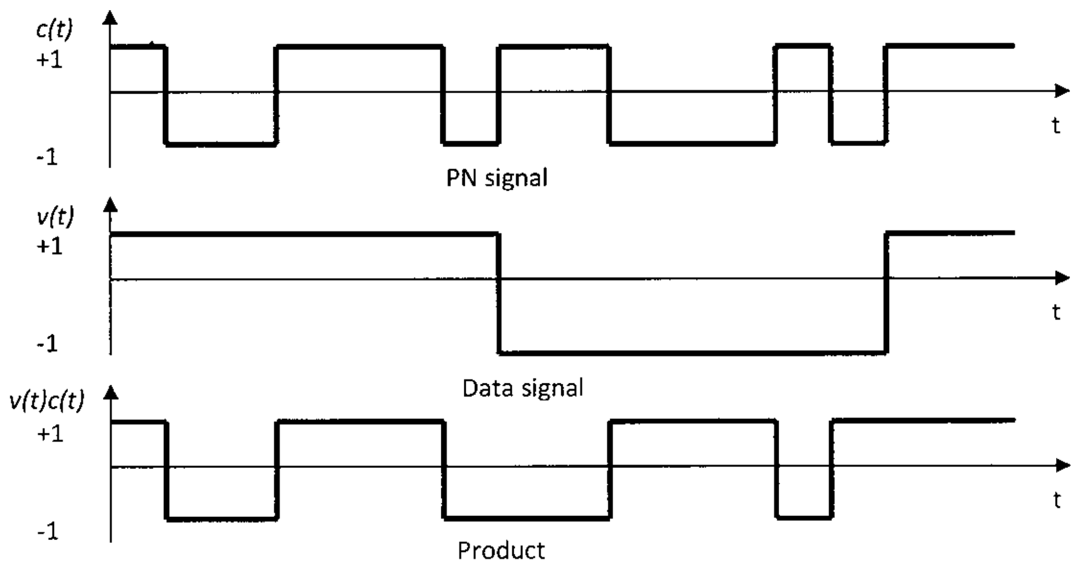


Figure 3.5 Generation of a DS spread-spectrum signal

Chip Rate

The rectangular pulse p(t) is commonly known as a chip, and its time duration T_c is called the chip interval. The expression $1/T_c$ is called the chip rate and it is approximately the same as the bandwidth W of the transmitted signal. The ratio of the bit interval T_b to the chip interval T_c is expressed as

$$L_c = \frac{T_b}{T_c}$$

where L_c is usually chosen to be an integer. In other words, L_c is the number of chips of the PN code sequence per information bit. In the system L_c can take on three different values, namely 8, 16 and 32. A lesser value will be too small to be effective against interference and a larger value means that there will be too few information bits per second transmitted, i.e. the system will be too slow.

Despreading of the Signal

The despreading of the signal is performed in the receiver by first multiplying it with a replica of the waveform c(t) generated by the PN code sequence generator. This generator is synchronized to the PN code in the received signal. Thus we have

$$A_c v(t) c^2(t) \cos(2\pi f_c t)$$

since $c^2(t) = 1$ for all t. The resulting signal occupies a bandwidth of approximately R Hz, which is the bandwidth of the information-bearing signal.

Interference in CDMA Systems

The resistance towards interfering signals is one of the CDMA based communication system biggest benefits. This work deals with the resistance against interference . A sinusoidal signal is to be used as a disturbance when the transmission is performed.

Suppose that the received signal is

$$r(t) = A_c v(t) c(t) \cos(2\pi f_c t) + i(t)$$

where i(t) denotes the interference. The despreading operation at the receiver yields

$$r(t)c(t) = A_c v(t)\cos(2\pi f_c t) + i(t)c(t)$$

The effect to multiplying the interference i(t) with c(t), is to spread the bandwidth of i(t) to W Hz. For example, consider the sinusoidal interfering signal

$$i(t) = A_J \cos(2\pi f_J t)$$

where f_J is a frequency within the bandwidth of the transmitted signal. The multiplication with c(t) results in a wideband interference with a power spectral density $J_0 = P_J/W$, where $P_J = A^2J/2$ is the average power of the interference. The desired signal is demodulated by a matched filter or a correlator with a bandwidth of R. This means that the total power in the interference at the output of the demodulator is,

$$J_0 R = \frac{P_J R}{W} = \frac{P_J}{W/R} = \frac{P_J}{T_b/T_c} = \frac{P_J}{L_c}$$

Therefore, the power in the interfering signal is reduced by an amount equal to the bandwidth expansion factor, W/R. This factor, $W/R = T_b/T_c = L_c$ is called the processing gain of the spread-spectrum system. Thus the effect of the interfering signal on the transmission is significantly reduced.

CDMA Process Gain

The process gain of a system is as same as the ratio of the spread spectrum bandwidth used, to the original data bit rate. It can can be written as:

$$G_P = \frac{BW_{RF}}{BW_{info}}$$

Where BW_{RF} is the transmitted bandwidth after the data is expanded, and BW_{info} is the bandwidth of the information data being sent.

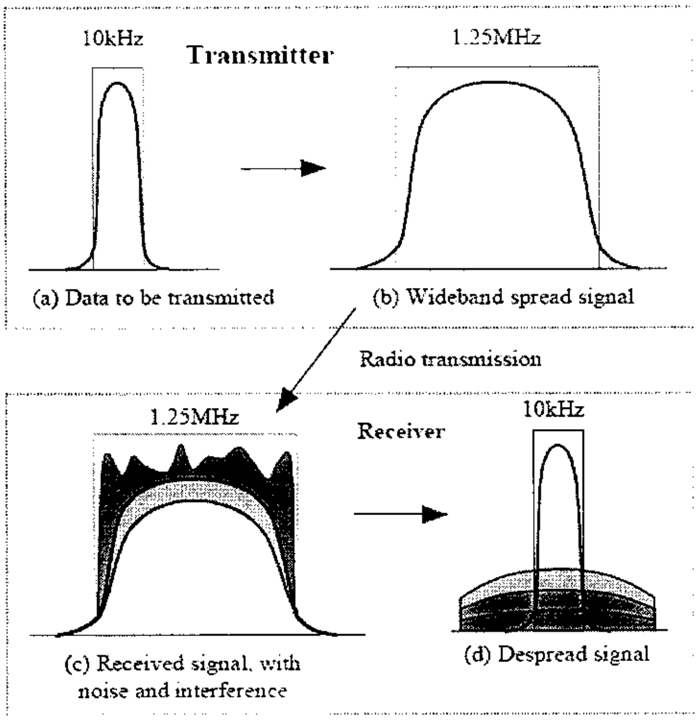


Figure 3.6 Process of a CDMA transmission.

The signal to be transmitted as in Figure 3.6(a) is expanded by modulating the data with its PN code. This makes the signal expanded as like in Figure 3.6(b). Here process gain is 125 as the spread spectrum bandwidth is 125 times bigger than the data bandwidth. Figure 3.6(c) shows the signal after that have been received. The receiving signal might have included of the background noise and any obstruction from other CDMA subscriber or radio signals and with that our required signal. The received signal is discovered by multiplying the signal by the original spreading code. Multiplying the signal by the original spreading code performs the real transmitted signal. However, all other signals which are uncorrelated to the PN spreading code used become more spread. Our required signal in Figure 3.6(d) is then filtered taking off the wide spread obstruction and other unwanted noises.

CHAPTER 4

CDMA VS IDMA

IDMA is special form of CDMA in which different interleaver are used for user separation. In CDMA encoding and spreading are done separately whereas in IDMA, encoding and spreading are combined as shown in Figure 4.1. The advantages of IDMA over CDMA at least focus on two aspects:

Firstly, DS-CDMA system can be regarded as a system with a channel coding concatenated with a repeated coding (repeated coding does not have any coding gain). Both the FEC and repeated coding will lead to spectrum spreading. While for an IDMA system, all spread spectrums can be used to FEC coding for achieving larger coding gain.

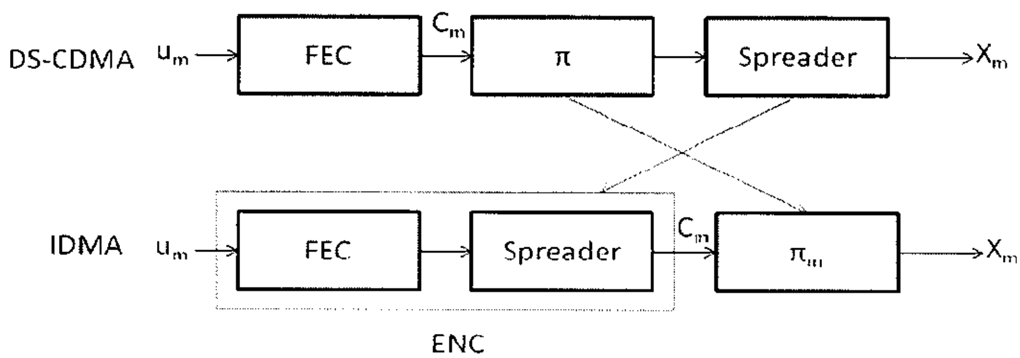
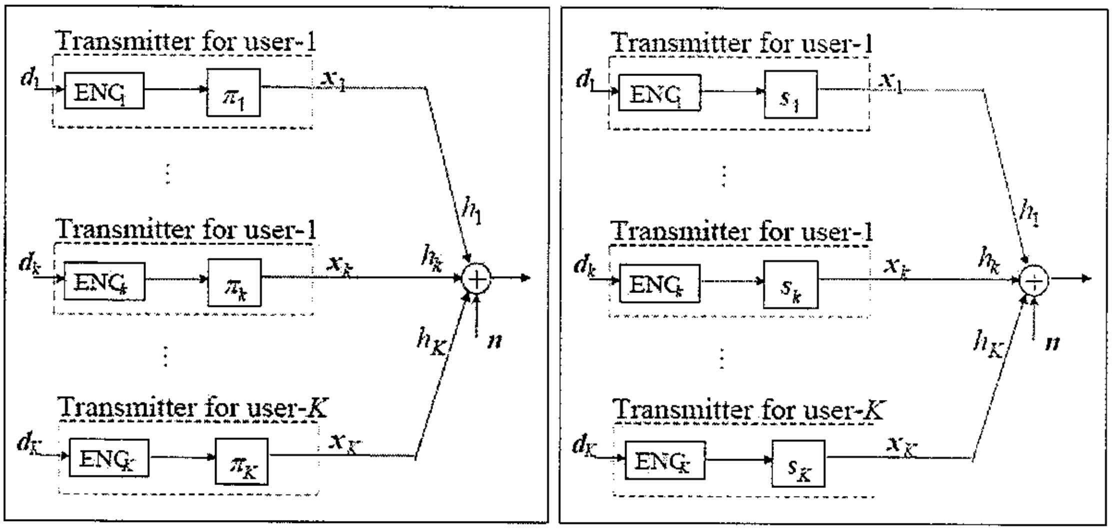


Figure 4.1 CDMA Vs IDMA

On the other hand, IDMA system can achieve single-user performance with iterative multi-user decoding. Take a two-user system for instance. After deinterleaving, the interference to N adjacent symbols of the desired data sequence come from N random-distributed symbols of the interfering sequence. Therefore the decoding of the two users are fully uncorrelated, thus the iterative reception can achieve a significant gain. However, in a system using scrambling codes to distinguish users, a desired data segments (consists of N adjacent symbols) coincides with the interfering segment. Thus a severe positive feedback will take place in the iterative decoding.

The main difference between CDMA and IDMA is simple as shown Figure 4.2 in which user data is interleaved and then spread in CDMA whereas in IDMA the data is spread first and then it is interleaved with different interleaver for different user.



IDMA CDMA

Figure 4.2 Transmitter parts of CDMA and IDMA

4.1 FACTOR-GRAPH REPRESENTATION OF CDMA_AND IDMA

To visualize the difference between IDMA and CDMA, factor-graph representations are plotted for CDMA and IDMA schemes. The factor graph representation is done in chip level. Figure 4.2 and 4.3 shows the factor graphs representation of CDMA and IDMA respectively. For an example, consider a system with 2 users and spreading length of 3. The (coded) bits and chips are represented by white circles, and channel observation by black circles. FEC coding, spreading (repetition) and channel constraints are represented by squares. The chip connections in CDMA are regular. In consequence, a mass of short circles, which are unfavorable to the receiver, appear in the factor graphs of CDMA. This situation is meliorated in IDMA because the chip interleavers break the fixed and regular connections and enlarge the length of the shortest circles. From these graphs, it is observed that IDMA can possibly outperform CDMA.

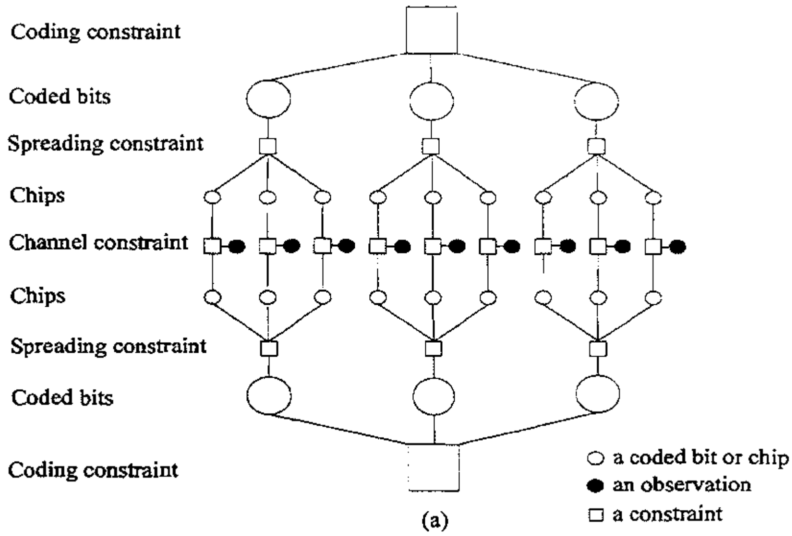


Figure 4.3 Factor graph representation of CDMA

The CDMA technique use different signature (sequence) for user separation in a multiuser environment whereas in Interleaved Division multiple access (IDMA) different interleavers are used for user separation.

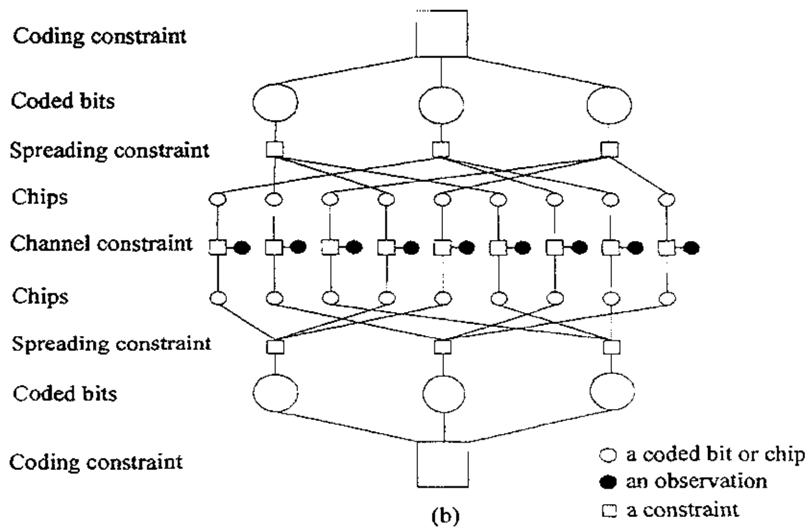


Figure 4.4 Factor graph for IDMA systems

CHAPTER 5

INTERLEAVE DIVISION MULTIPLE ACCESS TECHNIQUE

Although the existing CDMA systems have advantages in multi- cell cellular environments, it has relatively low throughput in single cell environments compared with IDMA and TDMA. This is mainly due to the effect of the Multiple Access Interference (MAI). A conventional CDMA scheme, such as IS 95, employs a forward error control (FEC) coding rate at around 1/2 —1/3. Quite long spreading sequences are required to support a large number of users. This is not an optimized approach from the bandwidth efficiency point of view. The theoretical analysis shows that the optimal Multiple Access Channel (MAC) capacity is achievable when the entire bandwidth expansion is devoted to coding.

5.1 MOTIVATION

Multiuser detectors, such as the optimal (MAP) detector, decorrelator, linear MMSE detector, are able to provide good performance in mitigating MAI. But their high computational complexities restrict practical implementations. With the advent of turbo codes iterative multiuser detections has been proposed and studied for the purpose of lowering the heavily computational load. From both capacity and complexity points of view, Low-rate coded CDMA is a natural choice. Code-spread CDMA systems are designed under this guideline especially, low-rate convolutional codes with maximum free distance are proposed for such a purpose. However, how to separate users when the spreading sequences are removed? This work deals with the use interleavers for user separation as described below, which forms the so-called interleave-division multiple-access (IDMA). In this scheme, interleavers are employed to distinguish signals from different users. The proposed IDMA scheme is aimed at achieving high spectral efficiency and improving performance with low receiver complexity. This scheme relies on interleaving as the only means to differentiate signals from different users. IDMA inherits many advantages from CDMA. in particular, dynamic channel sharing, graceful degradation, high flexibility in transmission rate, soft handoff, diversity against fading and mitigation of the worst-case other-cell user interference problem, furthermore, it allows a very simple chip-by-chip (CBC) iterative multiuser detection strategy. The normalized multiuser detector cost (per user) is independent of the number of users K.

5.2 IDMA TRANSMITER

The transmitter structure of an IDMA system with K simultaneous users. The input data sequence d_k of user-k is encoded based on a low-rate code C. The coded sequence is then interleaved by a chip-level interleaver π_k, producing $x_k \equiv [x_k(1), \ldots, x_k(j), \ldots, x_k(J)]T$. Following the convention, the elements x_k is called as "chips". The key principle of IDMA is that the interleavers$\{\pi_k\}$ should be different for different users. It is assume that the interleavers are generated independently and randomly. These interleavers disperse the coded sequences so that the adjacent chips are approximately uncorrelated, which facilitates the simple chip-by-chip. Assume quasi-static single-path channels.

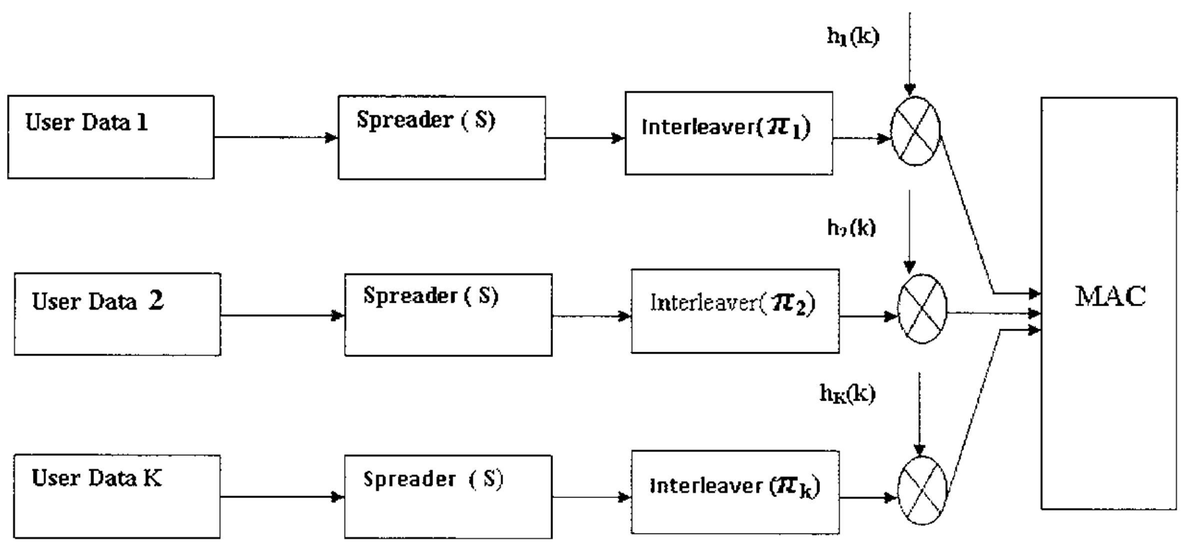


Figure 5.1 IDMA Transmitter

After chip matched filtering, the received signal from K users can be written as,

$$r(j) = \sum_{k=1}^{K} h_k x_k(j) + n(j), \qquad j = 1, 2, \ldots, J$$

Where h_k is the channel coefficient for user-k $\{n(j)\}$ are samples of an AWGN with variance $\sigma^2 = N_0/2$. We assume that the channel coefficients $\{h_k\}$ are known a priori at the receiver side.

5.3 DATA INTERLEAVER

Interleaving is a process of rearranging the ordering of a data sequence in a one to one deterministic format. The inverse of this process is called deinterleaving which restores the received sequence to its original order. Interleaving is a practical technique to enhance the error correcting capability of coding. In turbo coding, interleaving is used before the information data is encoded by the second component encoder. The basic role of an interleaver is to construct a long block code from small memory convolutional codes, as long codes can approach the Shannon capacity limit. Secondly, it spreads out burst errors.

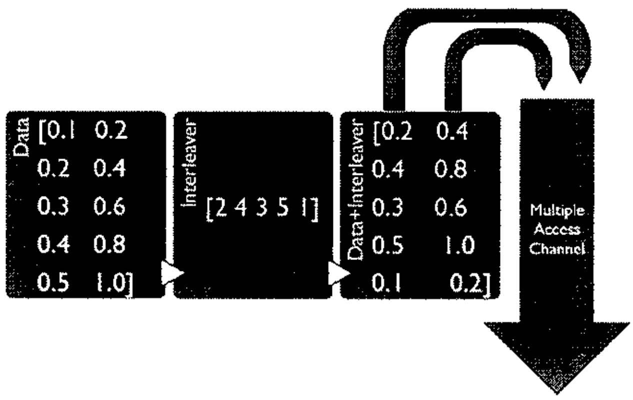


Figure 5.2 Data Interleaver

The interleaver provides scrambled information data to the second component encoder and decorrelates inputs to the two component decoders so that an iterative suboptimum-decoding algorithm based on uncorrelated information exchange between the two component decoders can be applied. The final role of the interleaver is to break low weight input sequences, and hence increase the code free Hamming distance or reduce the number of code words with small distances in the code distance spectrum. The size and structure of interleavers play a major role in the performance of turbo codes. There are a number of interleavers, which can be implemented.

5.3.1 RANDOM INTERLEAVER:

The Random Interleaver rearranges the elements of its input vector using a random permutation. The incoming data is rearranged using a series of generated permuter indices. A permuter is essentially a device that generates pseudo-random permutation of given memory addresses. The data is arranged according to the pseudo-random order of memory addresses.

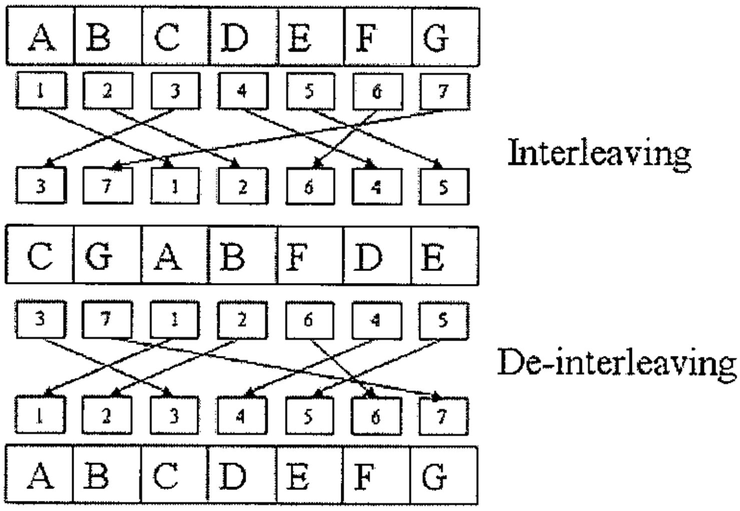


Figure 5.3 Random Interleaver

5.3.2 BLOCK INTERLEAVER

The block interleaver is the most commonly used interleaver in communication system. It writes in column wise from top to bottom and left to right and reads out row wise from left to right and top to bottom.

5.3.3 DIAGONAL INTERLEAVER

Diagonal interleaver writes in column wise from top to bottom and left to right and reads out diagonally from left to right and top to bottom.

5.4 IDMA RECEIVER

The chip-level interleavers $\{\pi_k\}$ allow us to adopt a chip-by-chip estimation technique as detailed below. The receiver in Fig.6.3 consists of an elementary signal estimator (ESE) and K single-user a posteriori probability (APP) decoders (DECs). For simplicity, we first consider BPSK signaling, i.e., $x_k(j)\{+1, -1\}$, and real channel coefficients. . Denote by $e(x_k(j))$ the extrinsic information about $x_k(j)$.

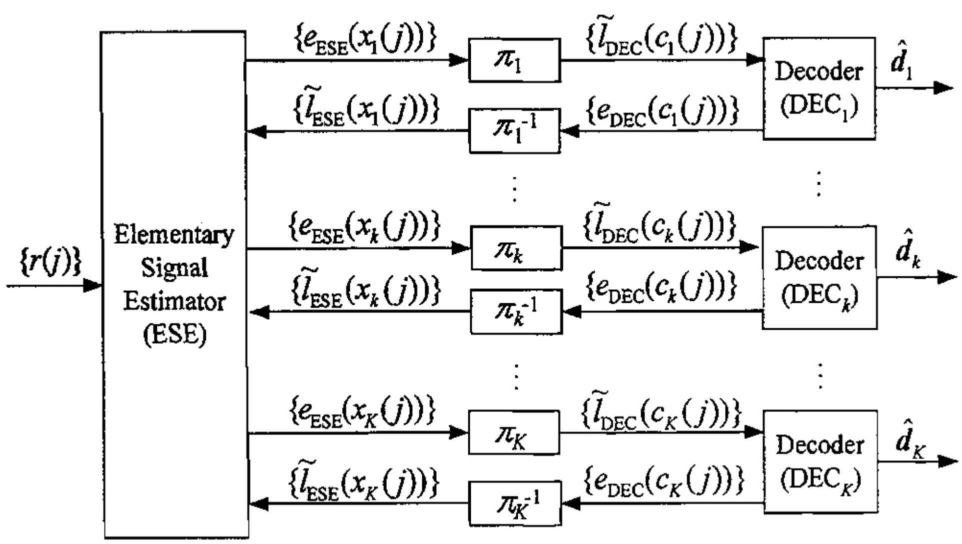


Figure 5.3 IDMA Receiver

It is further distinguished by subscripts, i.e., $(e_{ESE}(x_k(j))$ and $(e_{DEC}(x_k(j))$, depending on whether it is generated by the ESE or DECs. During the turbo-type iterative process, the extrinsic information generated by the ESE is used (after de-interleaving) as the a priori information in the DECs, and vice versa. The ESE uses $\{r(j)\}$ and $(e_{DEC}(x_k(j))$ as its inputs and only constraint (ii) is considered here. The output of the ESE is the extrinsic information $(e_{ESE}(x_k(j))$ about $\{xk(j)\}$. The de-interleaved version of $(e_{ESE}(x_k(j))$ is used as the a priori information i the kth DEC for user-k. The output of the kth DEC (after interleaving) is the updated extrinsic information $(e_{DEC}(x_k(j))$ based on the code constraint of C. The overall procedure is repeated a pre-set number of times. During the final iteration, the DECs produce hard decisions on information bits $\{k_d\}$.

5.5 IDMA TRANSMITTER AND RECEIVER PRINCIPLES

The Figure 5.1 shows the transmitter structure of the multiple access scheme under consideration with K simultaneous users. The input data sequence d_k of user-k is encoded based on a low-rate code C, generating a coded sequence $c_k \equiv [c_k(1), \ldots, c_k(j), \ldots, c_k(J)]^T$, where J is the frame length. The elements in c_k are referred to as coded bits. Then c_k is permutated by an interleaver π_k, producing $x_k \equiv [x_k(1), \ldots, x_k(j), \ldots, x_k(J)]^T$. Following the CDMA convention, refer the elements in x_k "chips". Users are solely distinguished by their interleavers, hence the name interleave-division multiple-access (IDMA).

The key principle of IDMA is that the interleavers $\{\pi_k\}$ should be different for different users. We assume that the interleavers are generated independently and randomly. These interleavers disperse the coded sequences so that the adjacent chips are approximately uncorrelated, which facilitates the simple chip-by-chip detection scheme discussed below.

This work adopt an iterative sub-optimal receiver structure, as illustrated in Figure 5.3 which consists of an elementary signal estimator (ESE) and K single-user *a posteriori* probability (APP) decoders (DECs). The multiple access and coding constraints are considered separately in the ESE and DECs. The outputs of the ESE and DECs are extrinsic log-likelihood ratios (LLRs) about $\{x_k(j)\}$ defined below :

$$e(x_k(j)) \equiv \log\left(\frac{\Pr(x_k(j)=+1)}{\Pr(x_k(j)=-1)}\right), \qquad \forall k, j. \qquad (1)$$

These LLRs are further distinguished by subscripts, i.e., $e_{ESE}(x_k(j))$ and $e_{DEC}(x_k(j))$, depending on whether they are generated by the ESE or DECs. A global turbo-type iterative process is then applied to process the LLRs generated by the ESE and DECs , as detailed below.

5.5.1 The Basic ESE Function

It is assumed that the channel has no memory. After chip-matched filtering, the received signal from K users can be written as

$$r(j) = \sum_{k=1}^{K} h_k x_k(j) + n(j), \qquad j = 1, 2, \ldots, J \qquad (2)$$

where h_k is the channel coefficient for user-k and $\{n(j)\}$ are samples of an AWGN process with variance $\sigma^2 = N_0/2$. It is assumed that the channel coefficients $\{h_k\}$ are known *a priori* at receiver. Due to the use of random interleavers $\{\pi_k\}$, the ESE operation can be carried out in a chip-by-chip manner, with only one sample $r(j)$ used at a time. Rewrite (2) as

$$r(j) = h_k x_k(j) + \zeta_k(j) \tag{3a}$$

where

$$\zeta_k(j) \equiv r(j) - h_k x_k(j) = \sum_{k' \neq k} h_{k'} x_{k'}(j) + n(j) \tag{3b}$$

is the distortion (including interference-plus-noise) in $r(j)$ with respect to user-k. From the central limit theorem, $\zeta_k(j)$ can be approximated as a Gaussian variable, and $r(j)$ can be characterized by a conditional Gaussian probability density function

$$p(r(j) \mid x_k(j) = \pm 1) = \frac{1}{\sqrt{2\pi \mathrm{Var}(\zeta_k(j))}} \exp\left(-\frac{(r(j) - (\pm h_k + \mathrm{E}(\zeta_k(j))))^2}{2\mathrm{Var}(\zeta_k(j))} \right) \tag{4}$$

where $\mathrm{E}(\cdot)$ and $\mathrm{Var}(\cdot)$ are the mean and variance functions, respectively.

The following is a list of the ESE detection algorithm based on (2)-(4) , assuming that the *a priori* statistics $\{\mathrm{E}(x_k(j))\}$ and $\{\mathrm{Var}(x_k(j))\}$ are available.

Algorithm 1. Chip-by-Chip Detection in a Single-Path Channel

Step (i): Estimation of Interference Mean and Variance

$$\mathrm{E}(r(j)) = \sum_k h_k \mathrm{E}(x_k(j)). \tag{5a}$$

$$\mathrm{Var}(r(j)) = \sum_k |h_k|^2 \mathrm{Var}(x_k(j)) + \sigma^2. \tag{5b}$$

$$\mathrm{E}(\zeta_k(j)) = \mathrm{E}(r(j)) - h_k \mathrm{E}(x_k(j)). \tag{5c}$$

$$\mathrm{Var}(\zeta_k(j)) = \mathrm{Var}(r(j)) - |h_k|^2 \mathrm{Var}(x_k(j)). \tag{5d}$$

Step (ii): LLR Generation

$$e_{ESE}(x_k(j)) = 2h_k \cdot \frac{r(j) - \mathrm{E}(\zeta_k(j))}{\mathrm{Var}(\zeta_k(j))}. \tag{6}$$

5.5.2 The ESE Function for Multi-Path Channels

We now consider the ESE function in a quasi-static multi-path fading channel with memory length L-1. Let $\{h_{k,0}, \cdots, h_{k,L-1}\}$ be the fading coefficients related to user-k. After chip-matched filtering, the received signal can be represented by

$$r(j) = \sum_{k=1}^{K} \sum_{l=0}^{L-1} h_{k,l} x_k(j-l) + n(j), \qquad j = 1, \cdots, J + L - 1 \tag{7}$$

We write

$$r(j+l) = h_{k,l}x_k(j) + \zeta_{k,l}(j) \quad (8a)$$

where

$$\zeta_{k,l}(j) = r(j+l) - h_{k,l}x_k(j)\,. \quad (8b)$$

The similarity between (8a) and (3a) is clearly seen. Assume again BPSK signaling and real channel coefficients. Algorithm 2 below is a straightforward extension of Algorithm 1.

Algorithm 2. Chip-by-Chip Detection in a Multi-Path Channel

Step (i): Estimation of Interference Mean and Variance

$$\mathrm{E}(r(j)) = \sum_{k,l} h_{k,l}\mathrm{E}(x_k(j-l))\,, \quad (9a)$$

$$\mathrm{Var}(r(j)) = \sum_{k,l} |h_{k,l}|^2 \mathrm{Var}(x_k(j-l)) + \sigma^2\,, \quad (9b)$$

$$\mathrm{E}(\zeta_{k,l}(j)) = \mathrm{E}(r(j+l)) - h_{k,l}\mathrm{E}(x_k(j))\,, \quad (9c)$$

$$\mathrm{Var}(\zeta_{k,l}(j)) = \mathrm{Var}(r(j+l)) - |h_{k,l}|^2 \mathrm{Var}(x_k(j))\,. \quad (9d)$$

Step (ii): LLR Generation and Combining

$$e_{ESE}(x_k(j))_l = 2h_{k,l} \cdot \frac{r(j+l) - \mathrm{E}(\zeta_{k,l}(j))}{\mathrm{Var}(\zeta_{k,l}(j))}\,. \quad (10a)$$

$$e_{ESE}(x_k(j)) = \sum_{l=0}^{L-1} e_{ESE}(x_k(j))_l\,. \quad (10b)$$

5.5.3 The ESE Function for More Complex Channels

Extending the discussion to more complex situations. It will use either superscripts "Re" and "Im" or function notations Re(.) and Im(.) to indicate real and imaginary parts, respectively. Consider quadrature-phase-shift-keying (QPSK) signaling.

$$x_k(j) = x_k^{\mathrm{Re}}(j) + i x_k^{\mathrm{Im}}(j)\,, \quad (11)$$

where $i = \sqrt{-1}$, $x_k^{\mathrm{Re}}(j)$ and $x_k^{\mathrm{Im}}(j)$ are two coded bits from c_k. For convenience, we still call the elements in x_k "chips". Note that in this case, each chip contains two coded bits. We adopt channel model (7) and expand it using complex channel coefficients $\{ h_{k,l} = h_{k,l}^{\mathrm{Re}} + i\, h_{k,l}^{\mathrm{Im}} \}$ as

$$r(j) = \sum_{k,l} \left(h_{k,l}^{\mathrm{Re}} x_k^{\mathrm{Re}}(j-l) - h_{k,l}^{\mathrm{Im}} x_k^{\mathrm{Im}}(j-l)\right) + i\sum_{k,l} \left(h_{k,l}^{\mathrm{Re}} x_k^{\mathrm{Im}}(j-l) + h_{k,l}^{\mathrm{Im}} x_k^{\mathrm{Re}}(j-l)\right) + n(j) \quad (12)$$

where $\{n(j)\}$ are samples of a complex AWGN process with variance σ^2 per dimension. Denote by $\overline{h_{k,l}}$ the conjugate of $h_{k,l}$. Recall (8): $r(j+l) = h_{k,l}x_k(j) + \zeta_{k,l}(j)$. The phase shift due to $h_{k,l}$ is cancelled out in $\overline{h_{k,l}}r(j+l)$, which means that $\mathrm{Im}(\overline{h_{k,l}}r(j+l))$ is not a function of $x_k^{\mathrm{Re}}(j)$. Therefore the detection of $x_k^{\mathrm{Re}}(j)$ only requires

$$\mathrm{Re}(\overline{h_{k,l}}r(j+l)) = |h_{k,l}|^2 x_k^{\mathrm{Re}}(j) + \mathrm{Re}(\overline{h_{k,l}}\zeta_{k,l}(j)). \tag{13}$$

Algorithm 3 below outlines the procedure to estimate $x_k^{\mathrm{Re}}(j)$ based on (13).

Algorithm 3. Chip-by-Chip Detection in a Complex Multi-Path Channel

Step (i): Estimation of Interference Mean and Variance

$$\mathrm{E}(r^{\mathrm{Re}}(j)) = \sum_{k,l}\left(h_{k,l}^{\mathrm{Re}}\mathrm{E}(x_k^{\mathrm{Re}}(j-l)) - h_{k,l}^{\mathrm{Im}}\mathrm{E}(x_k^{\mathrm{Im}}(j-l))\right), \tag{14a}$$

$$\mathrm{E}(r^{\mathrm{Im}}(j)) = \sum_{k,l}\left(h_{k,l}^{\mathrm{Re}}\mathrm{E}(x_k^{\mathrm{Im}}(j-l)) + h_{k,l}^{\mathrm{Im}}\mathrm{E}(x_k^{\mathrm{Re}}(j-l))\right), \tag{14b}$$

$$\mathrm{Var}(r^{\mathrm{Re}}(j)) = \sum_{k,l}\left((h_{k,l}^{\mathrm{Re}})^2\mathrm{Var}(x_k^{\mathrm{Re}}(j-l)) + (h_{k,l}^{\mathrm{Im}})^2\mathrm{Var}(x_k^{\mathrm{Im}}(j-l))\right) + \sigma^2, \tag{14c}$$

$$\mathrm{Var}(r^{\mathrm{Im}}(j)) = \sum_{k,l}\left((h_{k,l}^{\mathrm{Im}})^2\mathrm{Var}(x_k^{\mathrm{Re}}(j-l)) + (h_{k,l}^{\mathrm{Re}})^2\mathrm{Var}(x_k^{\mathrm{Im}}(j-l))\right) + \sigma^2. \tag{14d}$$

$$\Psi(j) = \sum_{k,l} h_{k,l}^{\mathrm{Re}}h_{k,l}^{\mathrm{Im}}\left(\mathrm{Var}(x_k^{\mathrm{Re}}(j-l)) - \mathrm{Var}(x_k^{\mathrm{Im}}(j-l))\right), \tag{15}$$

$$\mathrm{E}\left(\mathrm{Re}(\overline{h_{k,l}}\zeta_{k,l}(j))\right) = h_{k,l}^{\mathrm{Re}}\mathrm{E}(r^{\mathrm{Re}}(j+l)) + h_{k,l}^{\mathrm{Im}}\mathrm{E}(r^{\mathrm{Im}}(j+l)) - |h_{k,l}|^2\mathrm{E}(x_k^{\mathrm{Re}}(j)). \tag{16a}$$

$$\begin{aligned}\mathrm{Var}\left(\mathrm{Re}(\overline{h_{k,l}}\zeta_{k,l}(j))\right) &= (h_{k,l}^{\mathrm{Re}})^2\mathrm{Var}(r^{\mathrm{Re}}(j+l)) + (h_{k,l}^{\mathrm{Im}})^2\mathrm{Var}(r^{\mathrm{Im}}(j+l)) \\ &\quad + 2h_{k,l}^{\mathrm{Re}}h_{k,l}^{\mathrm{Im}}\Psi(j+l) - |h_{k,l}|^4\mathrm{Var}(x_k^{\mathrm{Re}}(j)).\end{aligned} \tag{16b}$$

Step (ii): LLR Generation and Combining

$$e_{ESE}(x_k^{\mathrm{Re}}(j))_l = 2|h_{k,l}|^2 \cdot \frac{\mathrm{Re}(\overline{h_{k,l}}r(j+l)) - \mathrm{E}(\mathrm{Re}(\overline{h_{k,l}}\zeta_{k,l}(j)))}{\mathrm{Var}(\mathrm{Re}(\overline{h_{k,l}}\zeta_{k,l}(j)))}, \tag{17a}$$

$$e_{ESE}(x_k^{\mathrm{Re}}(j)) = \sum_{l=0}^{L-1} e_{ESE}(x_k^{\mathrm{Re}}(j))_l . \tag{17b}$$

5.5.4 The DEC Function

The DECs in Figure. 5.3 carry out APP decoding using the output of the ESE as the input. With BPSK signaling, their output is the extrinsic LLRs $\{e_{DEC}(x_k(j))\}$ of $\{x_k(j)\}$ defined in (1), which are used to generate the following statistics

$$E(x_k(j)) = \tanh(e_{DEC}(x_k(j))/2), \quad (19a)$$

$$\mathrm{Var}(x_k(j)) = 1-(E(x_k(j)))^2. \quad (19b)$$

The Chip by Chip (CBC) detection algorithm is summarized as follows,

1. Initialize the extrinsic information of decoded output as zero

$$e_{DEC}(x_k(j)) = 0$$

2. Next assign the mean value of decoded output $(x_k(j)$ as follows,

$$E(x_k(j)) = \tanh(e_{DEC}(x_k(j))/2)$$

3. Find the variance of $(x_k(j)$ as

$$\mathrm{Var}(x_k(j)) = 1-(E(x(j)))^2$$

4. Now find the mean of received signal r(j) from decoded signal and channel coefficients

$$E(r(j)= \sum h_{k'} E(x_{k'}(j))$$

5. Next find the variance of r(j) as,

$$\mathrm{Var}(r(j)) \Leftarrow \sum_{k'=1}^{K} |h_{k'}|^2 \mathrm{Var}(x_{k'}(j)) + \sigma^2$$

6. Finally find Elementary Signal Estimation function as,

$$e_{ESE}(x_k(j)) \Leftarrow 2h_k \cdot \frac{r(j) - E(r(j)) + h_k E(x_k(j))}{\mathrm{Var}(r(j)) - |h_k|^2 \mathrm{Var}(x_k(j))}$$

The function is iterated for user defined number. During the turbo-type iterative process, the extrinsic information generated by the ESE is used (after de-interleaving) as the a priori information in the DECs, and vice versa.

CHAPTER 6

RESULTS AND DISCUSSION

The simulation of this project has been done using MATLAB R2009b.

MATLAB (meaning "**MAT**rix **LAB**oratory") was invented in the late 1970s by Cleve Moler, then chairman of the computer science department at the University of New Mexico. MATLAB was first adopted by control design engineers, Little's specialty, but quickly spread to many other domains. It is now also used in education, in particular the teaching of linear algebra and numerical analysis, and is popular amongst scientists involved with image processing. It provides many convenient ways for creating vectors, matrices, and multi-dimensional arrays.

In this project, various parameters have been taken. They are number of users, PN sequence, channel type, type of interleaver, repetition code (FEC), BER, and SNR ratio. It is also simulated for coded IDMA with random PN sequence and results have been discussed.

Figure 6.1 shows the Simulation result for performance of CDMA systems with following considerations

Number of user = 32

PN-Sequence = 11000

SNR ratio = -25:5:25

Channel type = AWGN

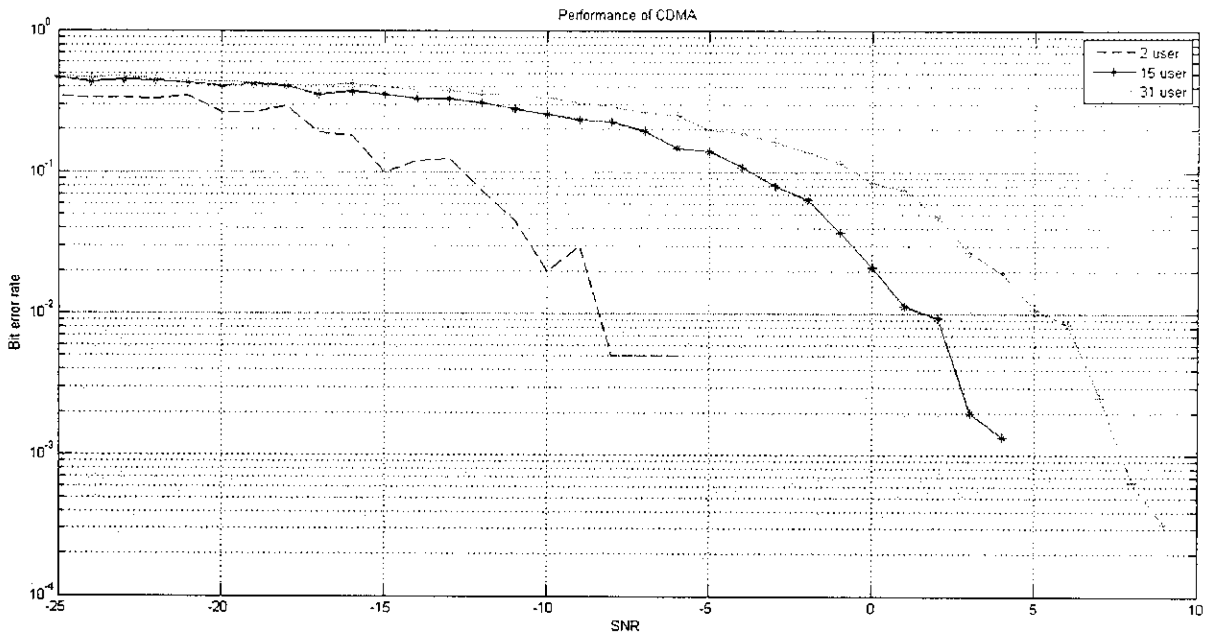


Figure 6.1 Simulation result for performance of CDMA systems

From the Figure 6.1 it is observed that increasing the number of user in a multi user CDMA environment leads to the increase of error.

When SNR = -10,

BER for 2 user=$10^{-1.8}$=0.0316227

BER for 15 user=$10^{-0.6}$=0.158484893

BER for 31 user=$10^{-0.5}$=0.316227766

Figure 6.2 shows the Simulation result for performance of IDMA systems with following parameters.

Number of user = 2

Spreading length = 16

SNR ratio = -20:5:20

Number of bits = 100

Chip length = spread length * number of bits

Number of iteration = 5

Channel type = AWGN

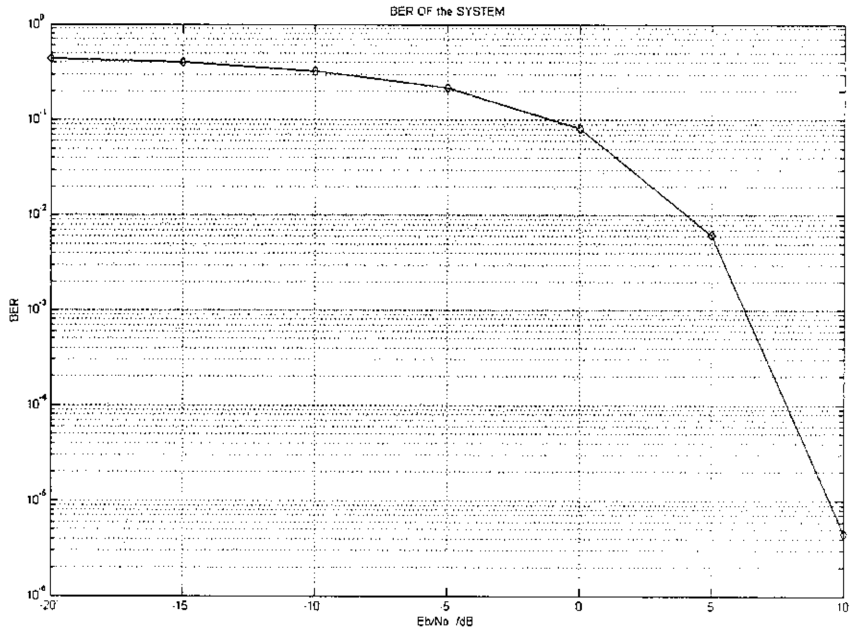


Figure 6.2 Simulation result for Performance of IDMA system

From the figure 6.2 the performance of IDMA system is observed as follows

When SNR=5db, BER=$10^{-2.7}$=0.0019

When SNR=10db, BER=$10^{-5.5}$=$3.6*10^{-6}$

Figure 6.3 shows the Simulation result for comparison of CDMA and IDMA systems with following considerations

Length of the data = 4

Bit rate = 6 bps

Number of iteration = 3

Channel type = AWGN

Chip length = length of the data*bit rate

=24 bps

Channel type = AWGN

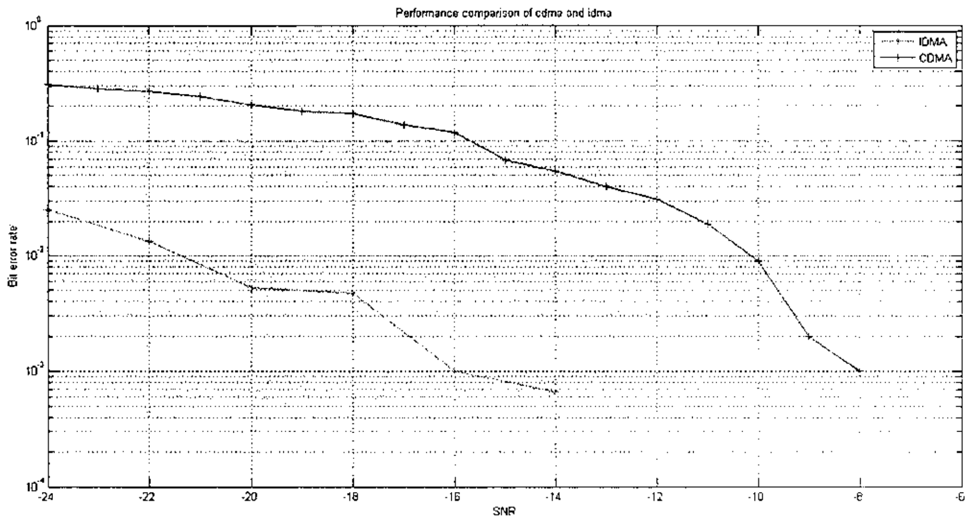


Figure 6.3 Simulation result for comparison of CDMA and IDMA systems

From the figure 6.3 the performance of CDMA and IDMA is compared as follows, When SNR= -20db,

BER of CDMA=$10^{-0.7}$=0.1995

BER of IDMA=$10^{-2.5}$=0.00316

Figure 6.4 shows the Simulation result for comparison of coded and uncoded IDMA systems with following considerations

Length of the data = 6

Bit rate = 8 bps

PN-Sequence generation = random

Number of iteration = 5

Channel type = AWGN

Chip length = length of the data*bit rate

=4 bps

Channel type = AWGN

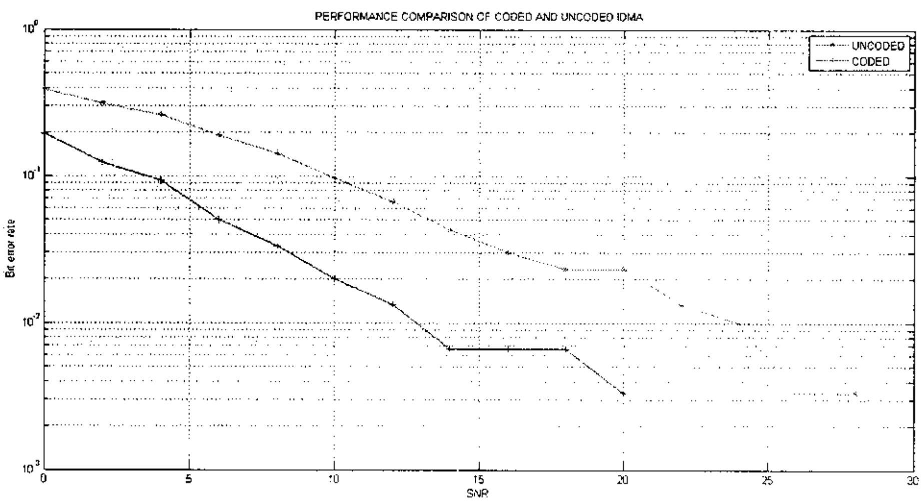


Figure 6.4 Simulation result for the comparison of coded and uncoded IDMA systems

From the figure 6.4 the performance of uncoded and coded IDMA is compared as follows,

When SNR= 10db,

BER of coded IDMA=$10^{-1.8}$=0.0158

BER of uncoded IDMA=10^{-1}=0.1

CHAPTER 7

CONCLUSION AND FUTURE SCOPE

CONCLUSION

To reduce the complexity of multiuser detection and improve system performance, IDMA scheme is used by introducing the chip-level interleavers to conventional CDMA. From the simulation results, it is observed that increasing the number of users leads to poor bit error rate in a CDMA system. The simple iterative CBC detection algorithm is used to reduce system complexity. It is also observed that error rate can be further controlled by using coded IDMA scheme. This improved multiple access technique is best suited for large network with better bit error performance. The experimental results of CDMA and IDMA schemes show that the IDMA technique is effective multiple access technique. In short, the IDMA system provides an efficient and effective solution to high-rate multiuser wireless communications. The low-complexity and high- performance properties make the IDMA scheme a competitive candidate for next generation wireless systems.

FUTURE SCOPE

Recently, the frequency-domain transmission and detection scheme has been proven to be an efficient approach. In particular, OFDM is a promising technique to treat ISI and has almost been deemed as the major transmission technique for 4G systems. It is worthy of investigating the IDMA system over OFDM. In this case, the basic CBC algorithm is also applicable. In this work, it is assumed that the channel state information (CSI) is known a prior at the receiver side. For practical applications, CSI has to be estimated. The channel estimation can be incorporated into a turbo receiver.

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Co-author(s);

MARIE JOHNSON
DIRECTOR

MARIAZEENA JOHNSON
DIRECTOR