CHANNEL ESTIMATION IN MULTICARRIER COMMUNICATION SYSTEMS

A THESIS SUBMITTED IN PARTIAL FULFILMENT
OF THE REQUIREMENTS FOR THE DEGREE OF

Bachelor of Technology
In
Electronics and Communication Engineering

By

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Department of Electronics and Communication Engineering
National Institute of Technology Rourkela
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CERTIFICATE

This is to certify that the thesis entitled "**Channel Estimation in Multicarrier Communication Systems**" submitted by **Mr. Sarada Prasanna Dash** and **Mr.Bikash Kumar Dora** in partial fulfilment for the requirements for the award of Bachelor of Technology Degree in **Electronics & Communication Engineering** at National Institute of Technology, Rourkela (Deemed University) is an authentic work carried out by them under my supervision and guidance.

To the best of my knowledge, the matter embodied in the thesis has not been submitted to any other University / Institute for the award of any Degree or Diploma.

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ACKNOWLEDGEMENTS

On the submission of our thesis report of “Channel Estimation in Multicarrier Communication Systems”, we would like to extend our gratitude & sincere thanks to our supervisor **Prof. Poonam Singh**, Department of Electronics and Communication Engineering for her constant motivation and support during the course of our work in the last one year. We truly appreciate and value her esteemed guidance and encouragement from the beginning to the end of this thesis. We are indebted to her for having helped us shape the problem and providing insights towards the solution.

We want to thank all our teachers **Prof. G. Panda**, **Prof. G.S. Rath**, **Prof. S.K. Patra** and **Prof. S.K. Behera** for providing a solid background for our studies and research thereafter. They have been great sources of inspiration to us and we thank them from the bottom of our heart.

Above all, we would like to thank all ours friends whose direct and indirect support helped us completing our project in time. This thesis would have been impossible without their perpetual moral support.

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ABSTRACT

The data rate and spectrum efficiency of wireless mobile communications have been significantly improved over the last decade or so. Recently, the advanced systems such as 3GPP LTE and terrestrial digital TV broadcasting have been sophisticatedly developed using OFDM and CDMA technology. In general, most mobile communication systems transmit bits of information in the radio space to the receiver**.** The radio channels in mobile radio systems are usually multipath fading channels, which cause inter-symbol interference (ISI) in the received signal. To remove ISI from the signal, there is a need of strong equalizer which requires knowledge on the channel impulse response (CIR).This is primarily provided by a separate channel estimator. Usually the channel estimation is based on the known sequence of bits, which is unique for a certain transmitter and which is repeated in every transmission burst. Thus, the channel estimator is able to estimate CIR for each burst separately by exploiting the known transmitted bits and the corresponding received samples.

In this thesis we investigate and compare various efficient channel estimation schemes for OFDM systems which can also be extended to MC DS-CDMA systems. The channel estimation can be performed by either inserting pilot tones into all subcarriers of OFDM symbols with a specific period or inserting pilot tones into each OFDM symbol. Two major types of pilot arrangement such as block type and comb-type pilot have been focused employing Least Square Error (LSE) and Minimum Mean Square Error (MMSE) channel estimators. Block type pilot sub-carriers is especially suitable for slow-fading radio channels whereas comb type pilots provide better resistance to fast fading channels. Also comb type pilot arrangement is sensitive to frequency selectivity when comparing to block type arrangement. However, there is another supervised technique called Implicit Training (IT) based channel estimation which exploits the first order statistics in the received data, induced by superimposing periodic training sequences with good correlation properties, along with the information symbols. Hence, the need for additional time slots for training the equalizer is avoided. The performance of the estimators is presented in terms of the mean square estimation error (MSEE) and bit error rate (BER).

LIST OF ACRONYMS

AWGN	Additive White Gaussian Noise
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
CDMA	Code Division Multiple Access
DFT	Discrete Fourier Transform
FDMA	Frequency Division Multiple Access
FIR	Finite Impulse Response
FFT	Fast Fourier Transform
IDFT	Inverse Discrete Fourier Transform
ICI	Inter Carrier Interference
ISI	Inter Symbol Interference
IT	Implicit Training
LSE	Least Square Estimation
MC DS-CDMA	Multicarrier Direct Sequence Code Division Multiple Access
MCM	Multicarrier Modulation
MMSE	Minimum Mean Square Estimation
OFDM	Orthogonal Frequency Division Multiplexing
PSK	Phase Shift Keying
QPSK	Quadrature Phase Shift Keying
SISO	Single Input Single Output
SNR	Signal to Noise Ratio
TDMA	Time Division Multiple Access
VLSI	Very Large Scale Integration

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CHAPTER-1
INTRODUCTION

1.1 INTRODUCTION

During the past few years, there has been an explosion in wireless technology. This growth has opened a new dimension to future wireless communications whose ultimate goal is to provide universal personal and multimedia communication without regard to mobility or location with high data rates. To achieve such an objective, the next generation personal communication networks will need to be support a wide range of services which will include high quality voice, data, facsimile, still pictures and streaming video. These future services are likely to include applications which require high transmission rates of several Mega bits per seconds (Mbps).

In the current and future mobile communications systems, data transmission at high bit rates is essential for many services such as video, high quality audio and mobile integrated service digital network. When the data is transmitted at high bit rates, over mobile radio channels, the channel impulse response can extend over many symbol periods, which leads to Inter-symbol interference (ISI). Orthogonal Frequency Division Multiplexing (OFDM) is one of the promising candidate to mitigate the ISI. In an OFDM signal the bandwidth is divided into many narrow sub-channels which are transmitted in parallel. Each sub-channel is typically chosen narrow enough to eliminate the effect of delay spread. By combining OFDM with CDMA dispersive-fading limitations of the cellular mobile radio environment can be overcome and the effects of co-channel interference can be reduced.

1.2 Digital Communication Systems

A digital communication system is often divided into several functional units. The task of the source encoder is to represent the digital or analog information by bits in an efficient way. The bits are then fed into the channel encoder, which adds bits in a structured way to enable detection and correction of transmission errors. The bits from the encoder are grouped and transformed to certain symbols, or waveforms by the modulator and waveforms are mixed with a carrier to get a signal suitable to be transmitted through the channel. At the receiver the reverse function takes place. The received signals are demodulated and soft or hard values of the corresponding bits are passed to the decoder. The decoder analyzes the structure of received bit pattern and tries to detect or correct errors. Finally, the corrected bits are fed to the source decoder that is used to reconstruct the analog speech signal or digital data input.. The main question is how to design certain parts of the modulator and demodulator to achieve efficient and robust transmission through a mobile wireless channel. The wireless channel has some properties that make the design especially challenging: it introduces time varying echoes and phase shifts as well as a time varying attenuation of the amplitude (fade).

Orthogonal Frequency Division Multiplexing (OFDM) has proven to be a modulation technique well suited for high data rates on time dispersive channels. There are some specific requirements when designing wireless OFDM systems, for example, how to choose the bandwidth of the sub-channels used for transmission and how to achieve reliable synchronisation. The latter is especially important in packet-based systems since synchronization has to be achieved within a few symbols. In order to achieve good performance the receiver has to know the impact of the channel. The problem is how to extract this information in an efficient way. Conventionally, known symbols are multiplexed into the data sequence in order to estimate the channel. From these symbols, all channel attenuations are estimated with an interpolation filter.

For mobile or wireless applications, the channel is often described as a set of independent multipath components. Among the most important parameters when choosing the modulation scheme are the delay and the expected received power for different delays. Large delays for stronger paths mean that the interference between the different received signal parts can severe, especially when the symbol rate is high so that the delay exceeds several symbols. In that case one has to introduce an equalizer to mitigate the effects of inter-symbol interference (ISI). Another alternative is to use many parallel channels so that the symbol time on each of the channels is long. This means that only a small part of the symbol is affected by ISI and this is the idea behind orthogonal frequency division multiplexing,

1.3 Evolution of Telecommunication Systems

Many mobile radio standards have been developed for wireless systems throughout the world, with more standard likely to emerge. Most first generations systems were introduced in the mid 1980s, and can be characterized by the use of analog transmission techniques, and the use of simple multiple access techniques such as Frequency Division Multiple Access (FDMA). First generation telecommunications systems such as Advanced Mobile Phone Service (AMPS), only provided voice communications. They also suffered from a low user capacity, and security problems due to the simple radio interface used.

Second generation systems were introduced in the early 1990s, and all use digital technology. This provided an increase in the user capacity of around three times. This was achieved by compressing the voice waveforms before transmission.

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 such as Code Division Multiple Access (CDMA), or an extension of TDMA, and by improving flexibility of services available.

1.4 Fourth Generation Wireless Systems

Although carriers are reluctant to discuss 4G, vendors are always mapping future of 4G systems. It is still a decade away (at least), but 4G is already a big topic of discussion behind closed doors. Main advantages of 4G are its spectrum optimization, network capacity and faster data rates, however, carriers are still reluctant to discuss 4G, either because they refuse to take a public position on it when 3G roll-outs still are unfulfilled, or because they are in denial. But carriers soon will find that 4G is not going away. 3G systems are not enough for many services like data transfer between wireless phones or multimedia. Equipment vendors are coming together to speed the adoption of OFDM, which will be part of the 4G set of standards.

Orthogonal Frequency Division Multiplexing OFDM is a multicarrier transmission technique, many carriers, each one being modulated by a low rate data stream share the transmission bandwidth. OFDM is similar to FDMA in that the multiple user access is achieved by subdividing the available bandwidth into multiple channels that are then allocated to users. However, OFDM uses the spectrum much more efficiently by spacing the channels much closer. This is achieved by making all the carriers orthogonal to one another, preventing interference between the closely spaced carriers.

OFDM overcomes most of the problems with both FDMA and TDMA. OFDM splits the available bandwidth into many narrow band channels (typically 100-8000 Hz). The carriers for each channel are made orthogonal to one another, allowing them to be spaced very close together, with no overhead as in the FDMA example. Because of this there is no great need for users to be time multiplexed as in TDMA, thus there is no overhead associated with switching between users.

The orthogonality of the carriers means that each carrier has an integer number of cycles over a symbol period. Due to this, the spectrum of each carrier has a null at the location of each of the other carriers in the system. This results in no interference between the carriers, allowing them to be as close as theoretically possible. This overcomes the problem of overhead carrier spacing required in FDMA. Each carrier in an OFDM signal has a very narrow bandwidth (i.e.1kHz), thus the resulting symbol rate is low. This results in the signal having a high tolerance to multipath delay spread, as the delay spread must be very long to cause significant inter-symbol interference (e.g. $\geq$ 500 µsec).

Multicarrier CDMA (MC-CDMA), a combination of OFDM and CDMA, provides resistance to ISI and multiple access simultaneously. In this modulation scheme, the incoming data stream is first multiplied by a user-specific PN sequence. The length of this spreading code is usually identical to the number of subcarriers. The resulting sequence is then converted into parallel format, and each chip is modulated to an independent subcarrier.

In practical situations, many of the subcarriers are degraded due to frequency-selective fading experienced on the channel. In OFDM systems, subcarriers that experience destructive fades typically produce bit errors. The benefit of MC CDMA is that it experiences frequency diversity because each bit is transmitted over several independently faded subcarriers. If some subcarriers experience destructive fades, diversity combining can be used at the receiver to recover the data. This improves the BER performance over OFDM, and this improvement is more significant as the number of subcarriers is increased. The drawback of MC-CDMA is that it may experience high levels of multiuser access interference (MAI) when the channel is heavily loaded. This occurs because each chip of the PN sequence experiences independent fading, which tends to destroy the orthogonality between spreading sequences. This increases the MAI and degrades the BER performance. As a result, MC-CDMA systems perform best under low channel loads.

Another alternative for 4G wireless systems is MC-DS-CDMA. In this scheme, direct sequence CDMA waveforms are transmitted in parallel over orthogonal subcarriers. To produce this effect, the incoming data stream is first converted into parallel format. Each bit is then multiplied by a user-specific spreading sequence with a higher chip rate. Following this, the IFFT is used to modulate the DS-CDMA signals to the intended subcarriers.

The benefit of MC-DS-CDMA is that it can provide multiple access without the excessive MAI that can occur in MC-CDMA systems. This is possible because all of the PN chips are transmitted on the same subcarrier, which experiences correlated fading provided that the channel slowly varies. The drawback of this approach, however, is that there is no gain from frequency diversity. If a subcarrier experiences a destructive fade, the data may not be recoverable at the receiver. To overcome this problem, adaptive modulation can be used to eliminate transmission or provide additional robustness on poor subcarriers.

1.5 OBJECTIVE AND OUTLINE OF THESIS

The main objectives of this thesis are: (1) Investigate OFDM and MC DS-CDMA as a modulation technique for wireless radio applications. Main factors affecting the performance of an OFDM system are multipath delay spread and channel noise. The performance of both the techniques is assessed using computer simulations performed using Matlab (2) channel estimation, for wireless OFDM transmission using pilot carriers and implicit training sequence and compare the mean squared error performance.

This thesis is organized as follows: In Chapter 2, characteristics of mobile radio channels and the basics of OFDM are presented; In Chapter3, the description of OFDM system; In Chapter 4,the CDMA and MC DS-CDMA method of transmission; In Chapter 5 the different approaches of channel estimation in OFDM systems is presented; Chapter 6 demonstrates Simulations and Results; Chapter 7 concludes the thesis and channel estimation in MC DS-CDMA as a future work is also suggested.

CHAPTER-2

PROPAGATION CHARACTERISTICS OF MOBILE RADIO CHANNELS

For an ideal radio channel, the received signal would consist of only a single direct path signal, which would be a perfect reconstruction of the transmitted signal. However in a real channel, the signal is modified during transmission in the channel. The received signal consists of a combination of attenuated, reflected, refracted, and diffracted replicas of the transmitted signal. On top of all this, the channel adds noise to the signal and can cause a shift in the carrier frequency if the transmitter or receiver is moving (Doppler effect). Understanding of these effects on the signal is important because the performance of a radio system is dependent on the radio channel characteristics.

2.1. Attenuation

Attenuation is the drop in the signal power when transmitting from one point to the another. It can be caused by the transmission path length, obstructions in the signal path and multipath effects. Any objects, which obstruct the line of sight signal from the transmitter to the receiver, can cause attenuation. Shadowing of the signal can occur whenever there is an obstruction between the transmitter and receiver. It is generally caused by buildings and hills, and is the most important environmental attenuation factor.

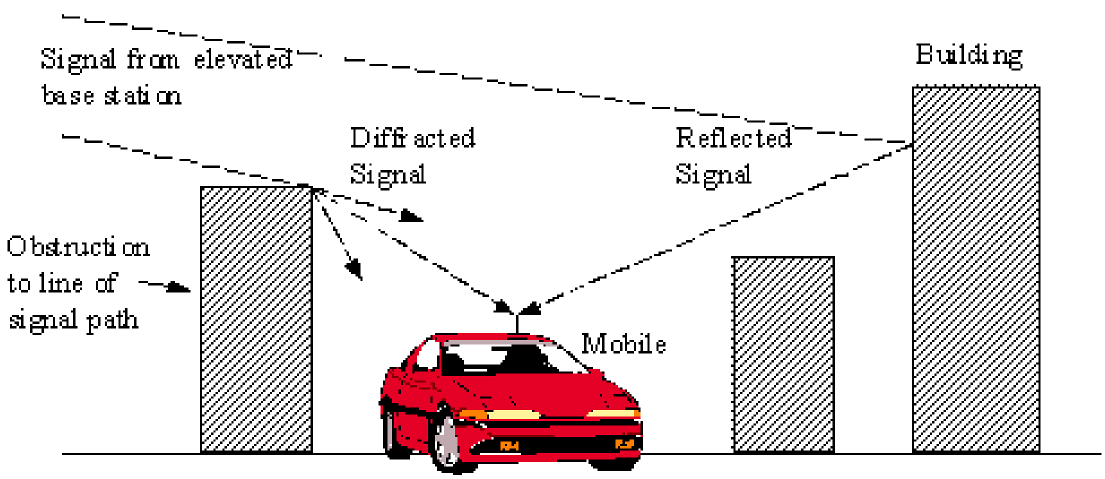

Figure-2.1

Shadowing is most severe in heavily built up areas, due to the shadowing from buildings. However, hills can cause a large problem due to the large shadow they produce. Radio signals diffract off the boundaries of obstructions, thus preventing total shadowing of the signals behind hills and buildings. However, the amount of diffraction is dependent on the radio frequency used, with low frequencies diffracting more than high frequency signals. Thus, high frequency signals, especially, Ultra High Frequencies (UHF), and microwave signals require line of sight for adequate signal strength. To overcome the problem of shadowing, transmitters are usually elevated as high as possible to minimise the number of obstructions.

2.2. Multipath Effects

2.2.1. Rayleigh fading

In a radio link, the RF signal from the transmitter may be reflected from objects such as hills, buildings, or vehicles. This gives rise to multiple transmission paths at the receiver. The relative phase of multiple reflected signals can cause constructive or destructive interference at the receiver. This is experienced over very short distances (typically at half wavelength distances), thus is given the term *fast fading*. These variations can vary from10-30dB over a short distance. Figure 6 shows the level of attenuation that can occur due to the fading.

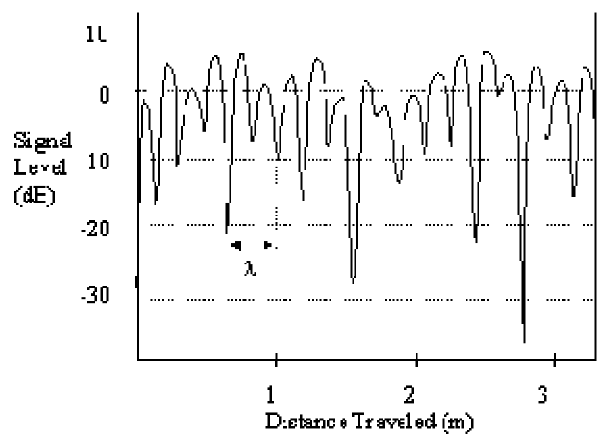

Figure-2.2

The Rayleigh distribution is commonly used to describe the statistical time varying nature of the received signal power. It describes the probability of the signal level being received due to fading. Table 2 shows the probability of the signal level for the Rayleigh distribution.

Signal Level (dB about median)	% Probability of Signal Level being less than the value given
10	99
0	50
-10	5
-20	0.5
-30	0.05

Table-2.1

2.2.2. Frequency Selective Fading

In any radio transmission, the channel spectral response is not flat. It has dips or fades in the response due to reflections causing cancellation of certain frequencies at the receiver. Reflections off near-by objects (e.g. ground, buildings, trees, etc) can lead to multipath signals of similar signal power as the direct signal. This can result in deep nulls in the received signal power due to destructive interference.

For narrow bandwidth transmissions if the null in the frequency response occurs at the transmission frequency then the entire signal can be lost. This can be partly overcome in two ways. By transmitting a wide bandwidth signal or spread spectrum as CDMA, any dips in the spectrum only result in a small loss of signal power, rather than a complete loss. Another method is to split the transmission up into many small bandwidth carriers, as is done in a COFDM/OFDM transmission. The original signal is spread over a wide bandwidth and thus, any nulls in the spectrum are unlikely to occur at all of the carrier frequencies. This will result in only some of the carriers being lost, rather than the entire signal. The information in the lost carriers can be recovered provided enough forward error corrections is sent.

2.3. Delay Spread

The received radio signal from a transmitter consists of typically a direct signal, plus reflections of object such as buildings, mountings, and other structures. The reflected signals arrive at a later time than the direct signal because of the extra path length, giving rise to a slightly different arrival time of the transmitted pulse, thus spreading the received energy. Delay spread is the time spread between the arrival of the first and last multipath signal seen by the receiver. In a digital system, the delay spread can lead to inter-symbol interference. This is due to the delayed multipath signal overlapping with the following symbols. This can cause significant errors in high bit rate systems, especially when using time division multiplexing (TDMA). As the transmitted bit rate is increased the amount of inter symbol interference also increases. The effect starts to become very significant when the delay spread is greater than ~50% of the bit time.

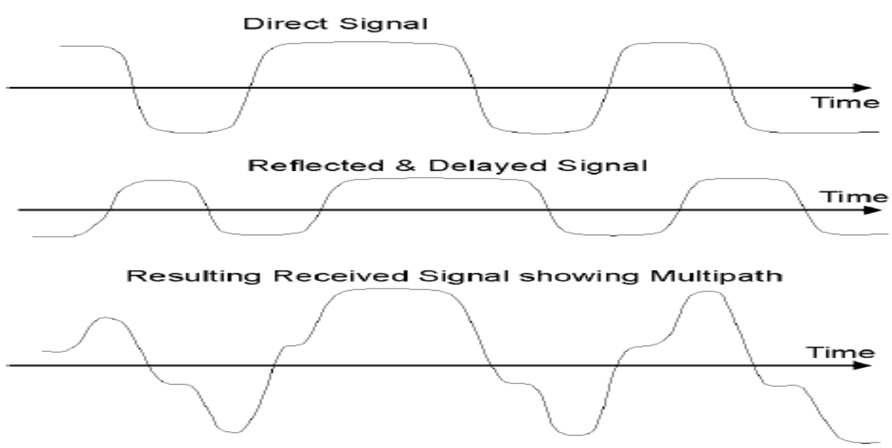

Figure-2.3

Inter-symbol interference can be minimized in several ways. One method is to reduce the symbol rate by reducing the data rate for each channel (i.e. split the bandwidth into more channels using frequency division multiplexing). Another is to use a coding scheme, which is tolerant to inter symbol interference such as CDMA.

2.4. Doppler Shift

When a wave source and a receiver are moving relative to one another the frequency of the received signal will not be the same as the source. When they are moving toward each other the frequency of the received signal is higher than the source, and when they are approaching each other the frequency decreases. This is called the *Doppler's effect*. An example of this is the change of pitch in a car's horn as it approaches then passes by. This effect becomes important when developing mobile radio systems.

The amount the frequency changes due to the Doppler effect depends on the relative motion between the source and receiver and on the speed of propagation of the wave. The Doppler shift in frequency can be written

$$\Delta f \approx \pm f_0 \frac{v}{c} \tag{2.1}$$

where Δf is the change in frequency of the source seen at the receiver, fo is the frequency of the source, v is the speed difference between the source and transmitter, and c is the speed of light.

Doppler shift can cause significant problems if the transmission technique is sensitive to carrier frequency offsets or the relative speed is higher, which is the case for OFDM. If we consider now a link between to cars moving in opposite directions, each one with a speed of 80 km/hr, the Doppler shift will be double.

CHAPTER-3

ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING

3.1 OFDM history

The concept of using parallel data transmission by means of frequency division multiplexing (FDM) was published in mid 60s. Some early development can be traced back in the 50s. A U.S. patent was filled and issued in January, 1970. The idea was to use parallel data streams and FDM with overlapping sub channels to avoid the use of high speed equalization and to combat impulsive noise, and multipath distortion as well as to fully use the available bandwidth. The initial applications were in the military communications. In the telecommunications field, the terms of discrete multi-tone (DMT), multichannel modulation and multicarrier modulation (MCM) are widely used and sometimes they are interchangeable with OFDM. In OFDM, each carrier is orthogonal to all other carriers. However, this condition is not always maintained in MCM. OFDM is an optimal version of multicarrier transmission schemes.

For a large number of sub channels, the arrays of sinusoidal generators and coherent demodulators required in a parallel system become unreasonably expensive and complex. The receiver needs precise phasing of the demodulating carriers and sampling times in order to keep crosstalk between sub channels acceptable. Weinstein and Ebert applied the discrete Fourier transform (DFT) to parallel data transmission system as part of the modulation and Demodulation process. In addition to eliminating the banks of subcarrier oscillators and coherent demodulators required by FDM, a completely digital implementation could be built around special-purpose hardware performing the fast Fourier transform (FFT). Recent advances in VLSI technology enable making of high-speed chips that can perform large size FFT at affordable price.

In the 1980s, OFDM has been studied for high-speed modems, digital mobile communications and high-density recording. One of the systems used a pilot tone for stabilizing carrier and clock frequency control and trellis coding was implemented. Various fast modems were developed for telephone networks. In 1990s, OFDM has been exploited for wideband data communications over mobile radio FM channels, high-bit-rate digital subscriber lines (HDSL, 1.6 Mb/s), asymmetric digital subscriber lines (ADSL, 1,536 Mb/s), very high-speed digital subscriber lines (VHDSL, 100 Mb/s), digital audio broadcasting (DAB) and HDTV terrestrial broadcasting.

3.2 Qualitative description of OFDM

In multimedia communication, a demand emerges for high-speed, high-quality digital mobile portable reception and transmission. A receiver has to cope with a signal that is often weaker than desirable and that contains many echoes. Simple digital systems do not work well in the multipath environment.

In a conventional serial data system, the symbols are transmitted sequentially, with the frequency spectrum of each data symbol allowed to occupy the entire available bandwidth. In a parallel data transmission system several symbols are transmitted at the same time, what offers possibilities for alleviating many of the problems encountered with serial systems.

In OFDM, the data is divided among large number of closely spaced carriers. This accounts for the “frequency division multiplex” part of the name. This is *not* a multiple access technique, since there is no common medium to be shared. The entire bandwidth is filled from a single source of data. Instead of transmitting in serial way, data is transferred in a parallel way. Only a small amount of the data is carried on each carrier, and by this lowering of the bit rate per carrier (not the total bit rate), the influence of inter symbol interference is significantly reduced. In principle, many modulation schemes could be used to modulate the data at a low bit rate onto each carrier.

It is an important part of the OFDM system design that the bandwidth occupied is greater than the correlation bandwidth of the fading channel. A good understanding of the propagation statistics is needed to ensure that this condition is met. Then, although some of the carriers are degraded by multipath fading, the majority of the carriers should still be adequately received. OFDM can effectively randomize burst errors caused by Rayleigh fading, which comes from interleaving due to parallelisation. So, instead of several adjacent symbols being completely destroyed, many symbols are only slightly distorted. Because of dividing an entire channel bandwidth into many narrow sub bands, the frequency response over each individual sub band is relatively flat. Since each sub channel covers only a small fraction of the original bandwidth, equalization is potentially simpler than in a serial data system. A simple equalization algorithm can minimize mean-square distortion on each sub channel, and the implementation of differential encoding may make it possible to avoid equalization altogether.

In addition, by using a guard interval the sensitivity of the system to delay spread can be reduced. In a classical parallel data system, the total signal frequency band is divided into N non-overlapping frequency sub channels. Each sub channel is modulated with a separate symbol and then, the N sub channels are frequency multiplexed. There are three schemes that can be used to separate the sub bands:

1. Use filters to completely separate the sub bands. This method was borrowed from the conventional FDM technology. The limitation of filter implementation forces the bandwidth of each sub band to be equal to $(1+a)f_m$, where a is the roll-off factor and fm is the Nyquist bandwidth. Another disadvantage is that it is difficult to assemble a set of matched filter when the number of carriers is large.

2. Use staggered QAM to increase the efficiency of band usage. In this way the individual spectra of the modulated carriers still use an excess bandwidth, but the are overlapped at the 3 dB frequency. The advantage is that the composite spectrum is flat. The separability or orthogonality is achieved by staggering the data (offset the data by half a symbol). The requirement for filter design is less critical than that for the first scheme.

3. Use discrete Fourier transform (DFT) to modulate and demodulate parallel data. The individual spectra are now *sinc* functions and are not band limited. The FDM is achieved, not by band-pass filtering, but by baseband processing. Using this method, both transmitter and receiver can be implemented using efficient FFT techniques that reduce the number of operations from N2 in DFT, down to NlogN.

OFDM can be simply defined as a form of multicarrier modulation where its carrier spacing is carefully selected so that each subcarrier is orthogonal to the other subcarriers. As is well known, orthogonal signals can be separated at the receiver by correlation techniques; hence, inter symbol interference among channels can be eliminated. Orthogonality can be achieved by carefully selecting carrier spacing, such as letting the carrier spacing be equal to the reciprocal of the useful symbol period. In order to occupy sufficient bandwidth to gain advantages of the OFDM system, it would be good to group a number of users together to form a wideband system, in order to interleave data in time and frequency (depends how broad one user signal is).

3.3 The importance of orthogonality

The “orthogonal” part of the OFDM name indicates that there is a precise mathematical relationship between the frequencies of the carriers in the system. In a normal FDM system, the many carriers are spaced apart in such way that the signals can be received using conventional filters and demodulators. In such receivers, guard bands have to be introduced between the different carriers and the introduction of these guard bands in the frequency domain results in a lowering of the spectrum efficiency.

It is possible, however, to arrange the carriers in an OFDM signal so that the sidebands of the individual carriers overlap and the signals can still be received without adjacent carrier interference. In order to do this the carriers must be mathematically orthogonal. The receiver acts as a bank of demodulators, translating each carrier down to DC, the resulting signal then being integrated over a symbol period to recover the raw data. If the other carriers all beat down to frequencies which, in the time domain, have a whole number of cycles in the symbol period (t), then the integration process results in zero contribution from all these carriers. Thus the carriers are linearly independent (i.e. orthogonal) if the carrier spacing is a multiple of 1/t.

Mathematically, signals are orthogonal if

$$\int_a^b \Psi_p(t)\Psi_q^*(t)dt = \begin{cases} K & for\ p = q \\ 0 & for\ p \neq q \end{cases}$$

<3.1>

where the * indicates the complex conjugate and interval [a b] is a symbol period. A fairly simple mathematical proof exists, that the series *sin* (*mx*) for *m*=1,2,… is orthogonal over the interval -p to p. Much of transform theory makes the use of orthogonal series, although they are by no means the only example.

3.4 Mathematical treatment of OFDM

After the qualitative description of the system, it is valuable to discuss the mathematical definition of the modulation system. This allows us to see how the signal is generated and how receiver must operate, and it gives us a tool to understand the effects of imperfections in the transmission channel. As noted above, OFDM transmits a large number of narrowband carriers, closely spaced in the frequency domain. In order to avoid a large number of modulators and filters at the transmitter and complementary filters and demodulators at the receiver, it is desirable to be able to use modern digital signal processing techniques, such as fast Fourier transform (FFT).

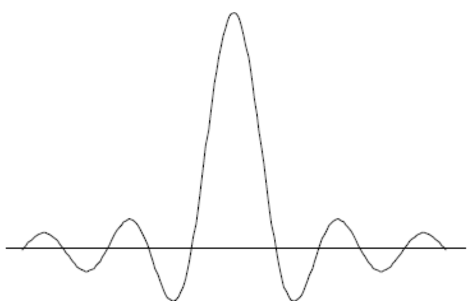

Figure-3.1 (a) a single sub-channel OFDM spectrum

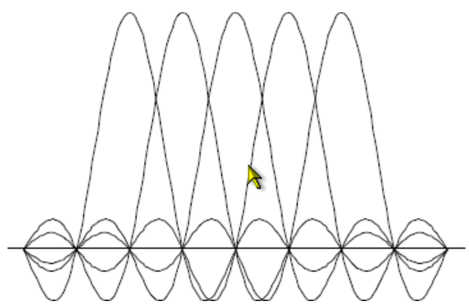

Figure-3.1 (b) 5 carriers based OFDM spectrum

Mathematically, each carrier can be described as a complex wave:

$$s_c(t) = A_c(t)e^{j\left[\omega_c t+\phi_c(t)\right]}$$

. <3.2>

The real signal is the real part of $s_c(t)$. Both *Ac(t)* and *fc(t),* the amplitude and phase of the carrier, can vary on a symbol by symbol basis. The values of the parameters are constant over the symbol duration period .OFDM consists of many carriers. Thus the complex signals $s_s(t)$ (Fig. 4.1) is represented by:

$$s_s(t) = \frac{1}{N}\sum_{n=0}^{N-1} A_N(t)e^{j\left[\omega_n t+\phi_n(t)\right]}$$

<3.3>

Where

$$\omega_n = \omega_0 + n\Delta\omega$$

This is of course a continuous signal. If we consider the waveforms of each component of the signal over one symbol period, then the variables *Ac(t)* and *fc(t)* take on fixed values, which depend on the frequency of that particular carrier, and so can be rewritten:

$$\phi_n(t) \Rightarrow \phi_n$$
$$A_n(t) \Rightarrow A_n$$

If the signal is sampled using a sampling frequency of 1/T, then the resulting signal is represented by:

$$s_s(kT) = \frac{1}{N}\sum_{n=0}^{N-1} A_n e^{j\left[(\omega_0 + n\Delta\omega)kT + \phi_n\right]} \qquad \text{<3.4>}$$

At this point, we have restricted the time over which we analyse the signal to N samples. It is convenient to sample over the period of one data symbol. Thus we have a relationship:

$$\tau = NT$$

If we now simplify eqn. 4.4, without a loss of generality by letting w0=0, then the signal becomes:

$$s_s(kT) = \frac{1}{N}\sum_{n=0}^{N-1} A_n e^{j\phi_n} e^{j(n\Delta\omega)kT} \qquad \text{<3.5>}$$

Now Eq. 4.5 can be compared with the general form of the inverse Fourier transform:

$$g(kT) = \frac{1}{N}\sum_{n=0}^{N-1} G\left(\frac{n}{NT}\right) e^{j2\pi nk/N} \qquad \text{<3.6>}$$

Eqns. 3.5 and 3.6 are equivalent if:

$$\Delta f = \frac{\Delta\omega}{2\pi} = \frac{1}{NT} = \frac{1}{\tau} \qquad \text{<3.7>}$$

This is the same condition that was required for orthogonality. Thus, one consequence of maintaining orthogonality is that the OFDM signal can be defined by using Fourier transform procedures.

3.5 FFT implementation in OFDM

3.5.1 The Fourier transform

The Fourier transform allows us to relate events in time domain to events in frequency domain. There are several version of the Fourier transform, and the choice of which one to use depends on the particular circumstances of the work.

The conventional transform relates to continuous signals which are not limited to in either time or frequency domains. However, signal processing is made easier if the signals are sampled. Sampling of signals with an infinite spectrum leads to aliasing, and the processing of signals which are not time limited can lead to problems with storage space.

To avoid this, the majority of signal processing uses a version of the discrete Fourier transform (DFT). The DFT is a variant on the normal transform in which the signals are sampled in both time and the frequency domains. By definition, the time waveform must repeat continually, and this leads to a frequency spectrum that repeats continually in the frequency domain.

The fast Fourier transform (FFT) is merely a rapid mathematical method for computer applications of DFT. It is the availability of this technique, and the technology that allows it to be implemented on integrated circuits at a reasonable price, that has permitted OFDM to be developed as far as it has. The process of transforming from the time domain representation to the frequency domain representation uses the Fourier transform itself, whereas the reverse process uses the inverse Fourier transform.

3.5.2 The use of the FFT in OFDM

The main reason that the OFDM technique has taken a long time to become a prominence has been practical. It has been difficult to generate such a signal, and even harder to receive and demodulate the signal. The hardware solution, which makes use of multiple modulators and demodulators, was somewhat impractical for use in the civil systems.

The ability to define the signal in the frequency domain, in software on VLSI processors, and to generate the signal using the inverse Fourier transform is the key to its current popularity. The use of the reverse process in the receiver is essential if cheap and reliable receivers are to be readily available. Although the original proposals were made a long time ago, it has taken some time for technology to catch up.

At the transmitter, the signal is defined in the frequency domain. It is a sampled digital signal, and it is defined such that the discrete Fourier spectrum exists only at discrete frequencies. Each OFDM carrier corresponds to one element of this discrete Fourier spectrum. The amplitudes and phases of the carriers depend on the data to be transmitted. The data transitions are synchronised at the carriers, and can be processed together, symbol by symbol.

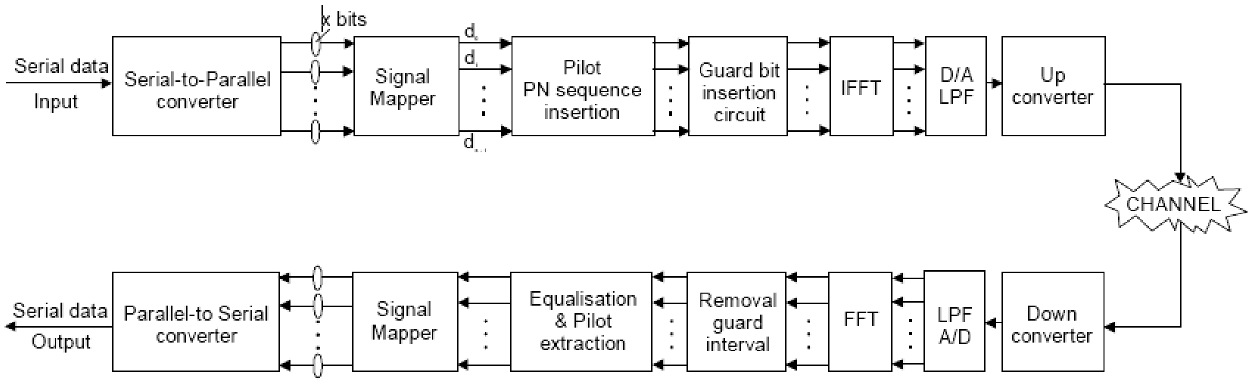

Figure-3.2 Block diagram of an OFDM system using FFT, pilot PN sequence and a guard bit insertion

The definition of the (N-point) discrete Fourier transform (DFT) is

$$X_p[k] = \sum_{n=0}^{N-1} x_p[n] e^{-j(2\pi/N)kn}$$

<3.8>

and the (N-point) inverse discrete Fourier transform (IDFT):

$$x_p[n] = \frac{1}{N} \sum_{k=0}^{N-1} X_p[k] e^{j(2\pi/N)kn}$$

<3.9>

A natural consequence of this method is that it allows us to generate carriers that are orthogonal. The members of an orthogonal set are linearly independent.

3.6 Guard interval and its implementation

The orthogonality of sub-channels in OFDM can be maintained and individual sub-channels can be completely separated by the FFT at the receiver when there are no inter symbol interference (ISI) and inter carrier interference (ICI) introduced by transmission channel distortion. In practice these conditions can't be obtained. Since the spectra of an OFDM signal is not strictly band limited (*sinc(f)* function),linear distortion such as multipath cause each sub-channel to spread energy into the adjacent channels and consequently cause ISI. A simple solution is to increase symbol duration or the number of carriers so that distortion becomes insignificant. However, this method may be difficult to implement in terms of carrier stability, Doppler shift, FFT size and latency.

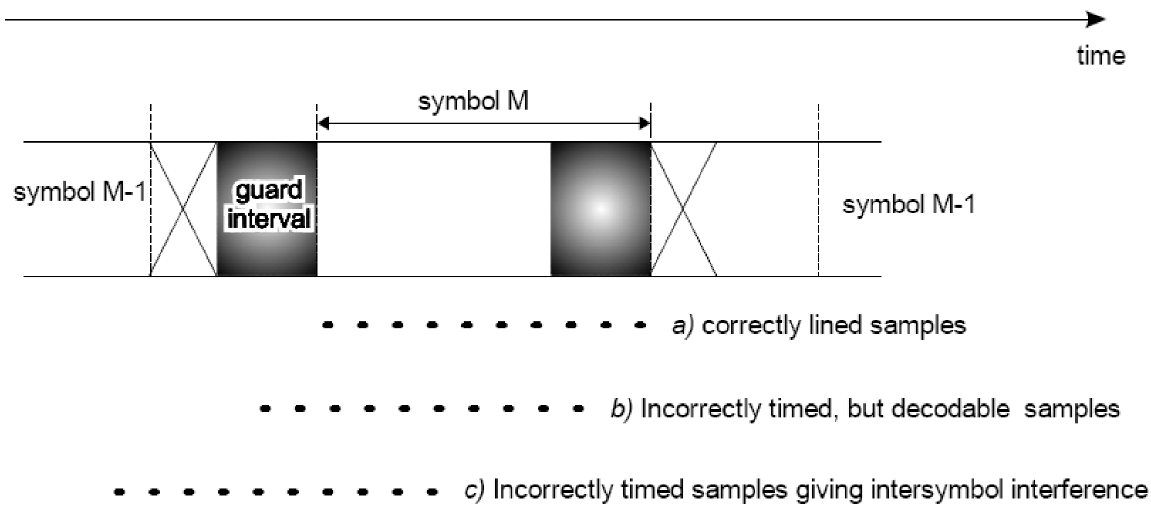

Figure 3.3 (Effect on the timing tolerance of adding a guard interval. With a guard interval included in the signal, the tolerance on timing the samples is considerably more relaxed).

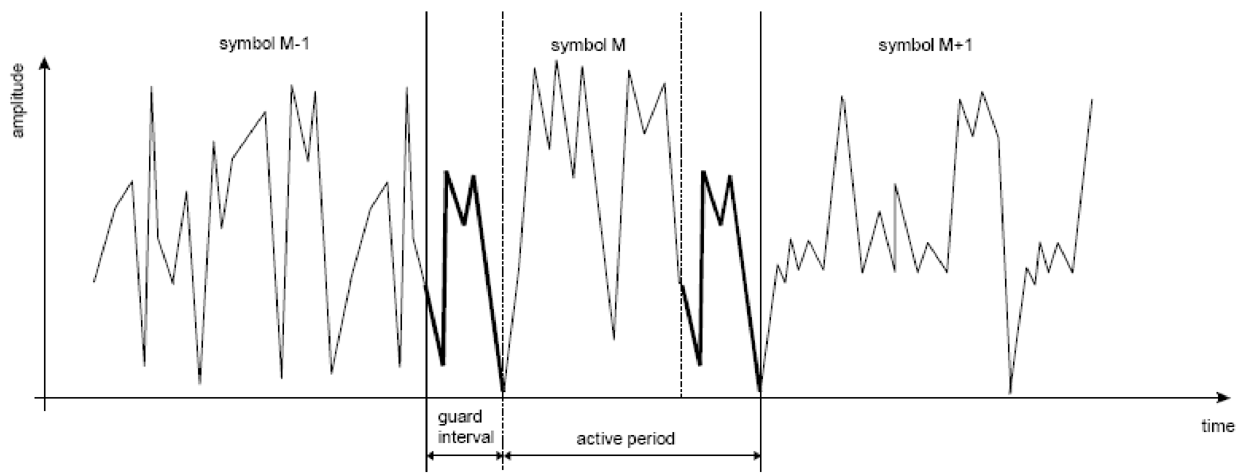

Fig 3.4 Example of the guard interval. Each symbol is made up of two parts. The whole signal is contained in the active symbol

One way to prevent ISI is to create a cyclically extended guard interval, where each OFDM symbol is preceded by a periodic extension of the signal itself. The total symbol duration is $T_{total}=T_g+T$, where T_g is the guard interval and T is the useful symbol duration. When the guard interval is longer than the channel impulse response or the multipath delay, the ISI can be eliminated. However, the ICI, or in-band fading, still exists. The ratio of the guard interval to useful symbol duration is application-dependent. Since the insertion of guard interval will reduce data throughput, *T*g is usually less than *T*/4.

The reasons to use a cyclic prefix for the guard interval are:

1· To maintain the receiver carrier synchronization; some signals instead of a long silence must always be transmitted;

2· Cyclic convolution can still be applied between the OFDM signal and the channel response to model the transmission system.

3.7 Windowing

Essentially, an OFDM signal consists of a number of unfiltered QAM sub-carriers. This means that the out-of-band spectrum decreases rather slowly, following a sinc function. For larger number of subcarriers, the spectrum goes down rapidly in the beginning, which is caused by the fact that the side lobes are closer together.

To make the spectrum decrease faster, windowing is applied to the OFDM signal. The standard doesn't specify the kind of window to be used but an example is included using the following function

$$w_T(t) = \begin{cases} \sin^2\left(\frac{\pi}{2}\left(0.5 + \frac{t}{T_{TR}}\right)\right) & for(-\frac{T_{TR}}{2} < t < \frac{T_{TR}}{2}) \\ 1 & for(\frac{T_{TR}}{2} \le t < T - \frac{T_{TR}}{2}) \\ \sin^2\left(\frac{\pi}{2}\left(0.5 - \frac{t-T}{T_{TR}}\right)\right) & for(T - \frac{T_{TR}}{2} \le t < T + \frac{T_{TR}}{2}) \end{cases}$$

<3.10>

T_{TR} is the transition time between two consecutive periods of FFT, as it can be seen in Figure

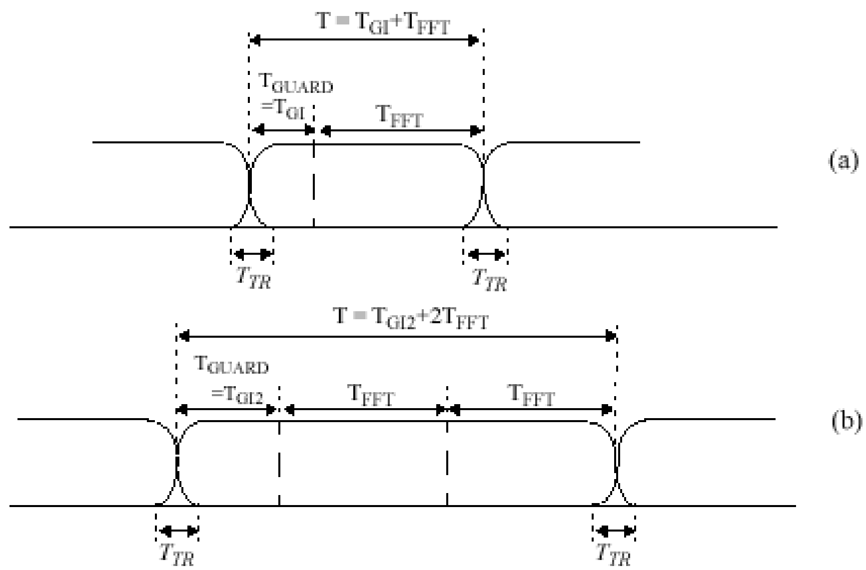

Figure 3.5 OFDM frame with cyclic extension and windowing for (a) single

reception or (b) two receptions of the FFT period

Figure 3.5 also illustrates the possibility of extending the windowing function over more than one period, T_{FFT} , and additionally shows smoothed transitions by application of a windowing function, as exemplified in Equation (3.10). In particular, window functions that extend over multiple periods of the FFT are utilized in the definition of the preamble. Several other conventional windows were simulated including raised cosine, Hann, Hamming, Blackman and Kaiser. The best performance was obtained for the Blackman window, considering the resulting stop band attenuation and the transition bandwidth

3.8 Choice of the key elements

Useful symbol duration

The useful symbol duration T affects the carrier spacing and coding latency. To maintain the data throughput, longer useful symbol duration results in increase of the number of carriers and the size of FFT (assuming the constellation is fixed). In practice, carrier offset and phase stability may affect how close two carriers can be placed. If the application is for the mobile reception, the carrier spacing must be large enough to make the Doppler shift negligible. Generally, the useful symbol duration should be chosen so that the channel is stable for the duration of a symbol.

Number of carriers

The number of subcarriers can be determined based on the channel bandwidth, data throughput and useful symbol duration. The carriers are spaced by the reciprocal of the useful symbol duration. The number of carriers corresponds to the number of complex points being processed in FFT. For HDTV applications, the number of sub-carriers are in the range of several thousands, so as to accommodate the data rate and guard interval requirement.

Modulation scheme

The modulation scheme in an OFDM system can be selected based on the requirement of power or spectrum efficiency. The type of modulation can be specified by the complex number $d_n=a_n+jb_n$, defined in section *The use of FFT in OFDM*. The symbols a_n and b_n can be selected to (±1, ±3) for 16QAM and ±1 for QPSK. In general, the selection of the modulation scheme applying to each sub-channel depends solely on the compromise between the data rate requirement and transmission robustness. Another advantage of OFDM is that different modulation schemes can be used on different sub-channels for layered services.

3.9 OFDM APPLICATIONS:

OFDM is digital transmission technique developed into a popular scheme for wideband digital communication systems. It is well suited for wideband, high data rate transmissions. The main advantage is that less equalization is necessary. The OFDM use has increased greatly in the last 10 years. Nowadays, OFDM is mainly used for one to many (broadcast) communications like radio or television broadcasting. Examples are digital broadcasting systems such as DAB and DVB. It is now proposed for Digital audio broadcasting such as in Eureka 147 standard and Digital Radio Mondiale (DRM). Digital Audio Broadcasting (DAB) is an international, standardized digital broadcasting system developed by the European EUREKA-147 Project. OFDM is used for modem/ADSL application where it coexists with phone line. For ADSL use, the channel, the phone line, is filtered to provide a high SNR. OFDM here is called Discrete Multi Tone (DMT.) HDSL: High bit rate Digital Subscriber Line is another implementation for symmetric speeds (uplink rate = downlink rate). HiperLAN2 is the all new high performance radio technology, specifically suited for operating in LAN environments. HiperLAN2 is a technology being developed within the European Telecommunications Standardisation Institute (ETSI). OFDM is the modulation used in the physical layer of HiperLAN2.OFDM is also in use in wireless internet modem and this usage is called 802.1la.

CHAPTER-4

CODE DIVISION MULTIPLE ACCESS

4.1 INTRODUCTION TO MULTIPLE ACCESS TECHNOLOGY

In communication network systems multiplexing is a term used to refer to a process where multiple analog message signals or digital data streams are combined into one signal over a shared medium. For example, in telecommunications, several phone calls may be transferred using one wire. The multiplexed signal is transmitted over a link, which may be a physical transmission medium. In a multiuser system, capacity is certainly a sensitive issue. The efficiency with which the available resources are used determines the number of users the system can support. Higher efficiency is achieved if the resources are made available to all users and they are assigned on a demand basis. This characterizes what is known as *demand-assigned multiple-access* (DAMA) or, simply, multiple access. The assignment of the resources on a demand basis suggests that some initial protocol be established before the resource is actually assigned. Handshaking protocols usually typify burst traffic, whereas payload information can be characterized by either burst or steady-flow traffic.

Certainly, the ultimate aim of any communication network is provision of resources to convey payload information. Overhead traffic, although necessary, should be kept to a minimum to increase efficiency. The initial access protocols may use the same physical resources as those used to convey payload information. However, splitting the resources into two distinct groups—one for initial access purposes and another for payload information purposes is a practice common to networks where a great amount of signalling is required. And this is true of wireless systems. The proportion of channels dedicated to each function varies according to the needs. The two most basic forms of technologies used by the wireless networks are frequency division multiple access (FDMA), time division multiple access (TDMA).

1. FDMA

FDMA is certainly the most conventional method of multiple access and was the first technique to be employed in modern wireless applications. In FDMA, the available bandwidth is split into a number of equal sub-bands, each of which constitutes a physical channel. The channel bandwidth is a function of the services to be provided and of the available technology and is identified by its center frequency, known as a carrier. In single channel per carrier FDMA technology, the channels, once assigned, are used on a non-time-sharing basis. Thus, a channel allocated to a given user remains allocated until the end of the task for which that specific assignment was made.

2. TDMA

TDMA is another widely known multiple-access technique and succeeded FDMA in modern wireless applications. In TDMA, the entire bandwidth is made available to all signals but on a time-sharing basis. In such a case, the communication is carried out on a buffer-and-burst scheme so that the source information is first stored and then transmitted. Prior to transmission, the information remains stored during a period of time referred to as a frame. Transmission then occurs within a time interval known as a (time) slot. The time slot constitutes the physical channel.

Besides TDMA & FDMA there exists another efficient multiple access technology called CDMA (Code Division Multiple Access) which increases the spectrum capacity by allowing all the users to occupy all the channels at the same time. It makes use of spread spectrum modulation techniques to achieve this property. Spread spectrum and CDMA are cutting-edge technologies widely used in operational radar, navigation and telecommunication systems and play a pivotal role in the development of the forthcoming generations of systems and networks.

4.2 General Principle of CDMA

Code division multiple access (CDMA) is a multiple access technique where different users share the same physical medium, that is, the same frequency band, at the same time. The main ingredient of CDMA is the *spread spectrum* technique, which uses high rate signature pulses to enhance the signal bandwidth far beyond what is necessary for a given data rate. In a CDMA system, the different users can be identified and, hopefully, separated at the receiver by means of their characteristic individual *signature pulses* (sometimes called the *signature waveforms*), that is, by their individual *codes*. Nowadays, the most prominent applications of CDMA are mobile communication systems like CDMA One (IS-95), UMTS or CDMA 2000. To apply CDMA in a mobile radio environment, specific additional methods are required to be implemented in all these systems. Methods such as power control and soft handover have to be applied to control the interference by other users and to be able to separate the users by their respective codes.

4.2.1 The concept of spreading

Spread spectrum means enhancing the signal bandwidth far beyond what is necessary for a given data rate and thereby reducing the power spectral density (PSD) of the useful signal so that it may even sink below the noise level. One can imagine that this is a desirable property for military communications because it helps to hide the signal and it makes the signal more robust against intended interference (*jamming*). Spreading is achieved – loosely speaking – by a multiplication of the data symbols by a *spreading sequence* of pseudo random signs. These sequences are called *pseudo noise* (PN) sequences or code signals.

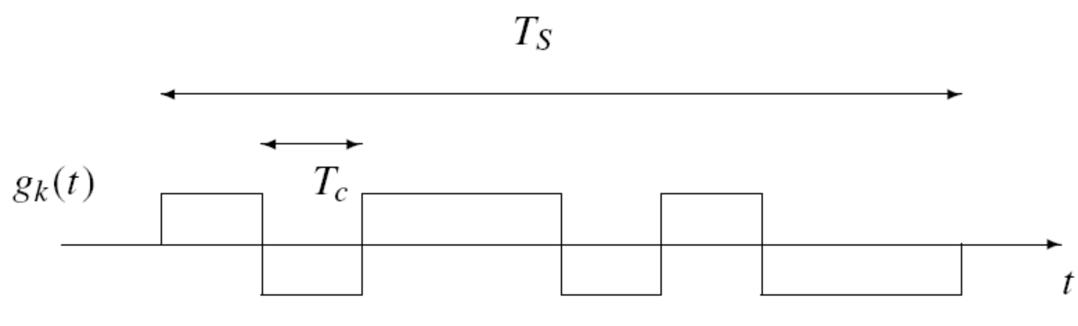

Figure 4.1 N=8 rectangular chips

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Consider a rectangular transmit pulse

$$g(t) = \frac{1}{\sqrt{T_S}} \Pi \left(\frac{t}{T_S} - \frac{1}{2} \right)$$

<4.1>

of length *TS*. We divide the pulse into *N* sub-rectangles, referred to as *chips*, of length *Tc* = *TS/N* and change the sign of the sub-rectangles according to the sign of the pseudorandom spreading sequence. Figure 5.1 shows the resulting transmit pulse $g_k(t)$ of user number *k* for *N* = 8. Here, the spreading sequence for user *k* is given by *(+, −, +, + ,−, +, −, −)*. When it is convenient (e.g. for the performance analysis) the sign factors shall be appropriately normalized. We note that in practice smooth pulse shapes (e.g. raised cosine pulses) will be used rather than rectangular ones.

The increase of the signalling clock by a factor *N* from f_s to f_c leads to an increase of bandwidth by a factor of T_S/T_c. For this reason, $N = T_S/T_c$ is called the *spreading factor* or, more precisely, the *spreading factor of the signature pulse*. This spreading is due to multiplication by the code sequence. While within the specification documents for CDMA mobile communication systems the spreading factor is often denoted by *SF*, formulas are kept simpler by using the symbol *N*.

It is often not uniquely defined where channel coding ends and where modulation starts and thus it may be ambiguous to speak of a bit rate after channel coding. We regard it as convenient to define the effective spreading factor by

$$SF_{\text{eff}} = \frac{R_{\text{chip}}}{R_b}$$

<4.2>

where R_b is the useful bit rate and $R_{\text{chip}} = 1/Tc$ the chip rate. Obviously, this spreading factor is approximately the inverse of the spectral efficiency for a single user. The objective of spreading is – loosely speaking – a waste of bandwidth for the single user to achieve more robustness against multiple access interference (MAI). It would thus be a contradiction to this objective to use bandwidth-efficient higher-level modulation schemes. Any modulation scheme that is more efficient than BPSK would reduce the spreading factor.

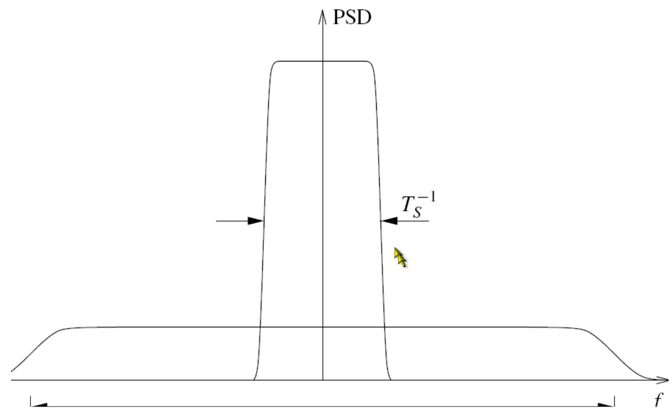

Figure-4.2 Power spectral density (PSD) for CDMA.

Therefore, BPSK and QPSK are used as the basic modulation schemes in most practical communication systems. Nevertheless, higher-order modulation techniques like 8-PSK and 16-QAM also are applied as additional transmission options to offer a high speed packet transfer at good propagation conditions.

4.3 P-N Sequences

In CDMA networks there are a number of channels each of which supports a very large number of users. For each channel the base station generates a unique code that changes for every user. The base station adds together all the coded transmissions for every subscriber. The subscriber unit correctly generates its own matching code and uses it to extract the appropriate signals.

In order for all this to occur, the pseudo-random code must have the following properties:

1. It must be deterministic. The subscriber station must be able to independently generate the code that matches the base station code.

2. It must appear random to a listener without prior knowledge of the code (i.e. it has the statistical properties of sampled white noise).

3. The cross-correlation between any two codes must be small

4. The code must have a long period (i.e. a long time before the code repeats itself).

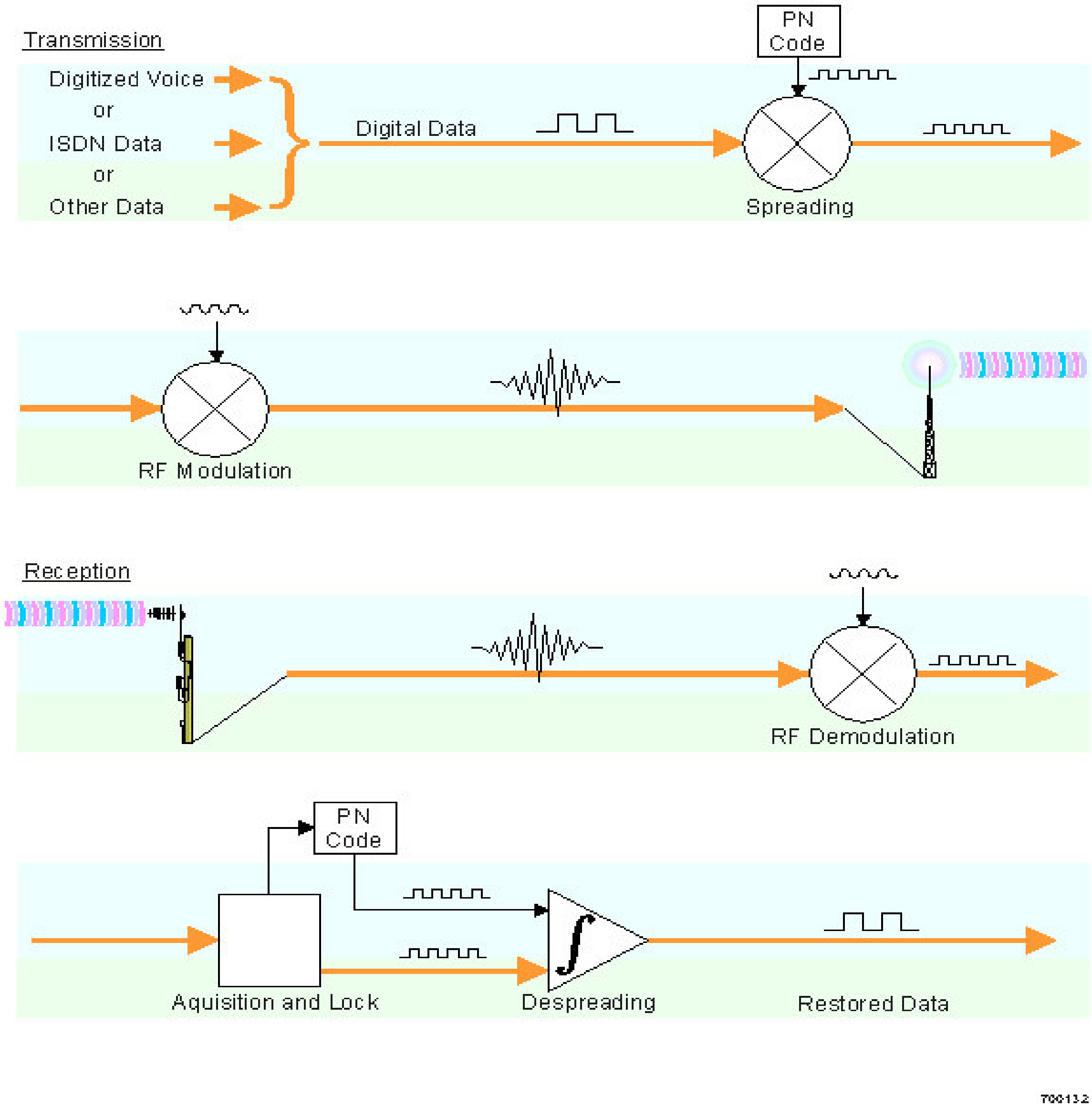

Figure 5.3 Signal transmission and reception in CDMA

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4.3.1 **Generation of PN sequence**:

The p-n sequence is usually generated using a shift register with feedback-taps. Binary sequences are shifted through the shift registers in response to clock pulses and the output of the various stages are logically combined and fed back as input to the stage. When the feedback logic consists of exclusive –OR gates, the shift register is called a linear PN sequence generator.

The initial contents of the memory stages and the feedback logic circuit determine the successive contents of the memory. If a linear shift register reaches zero state at some time, it would always remain in the zero state and the output would subsequently be all zeros. Since there are exactly 2^m-1 nonzero states for an m-stage feedback shift register, the period of a PN sequence produced by a linear m-stage shift register cannot exceed 2^m-1 symbols which is called maximal length sequence.

4.4 **Code Correlation**

In general the correlation function has these properties:

- It equals 1 if the two codes are identical
- It equals 0 of the two codes have nothing in common

Intermediate values indicate how much the codes have in common. The more they have in common, the harder it is for the receiver to extract the appropriate signal.

There are two correlation functions:

- Cross-Correlation: The correlation of two different codes. As we've said, this should be as small as possible.
- Auto-Correlation: The correlation of a code with a time-delayed version of itself. In order to reject multi-path interference, this function should equal 0 for any time delay other than zero.

The receiver uses cross-correlation to separate the appropriate signal from signals meant for other receivers, and auto-correlation to reject multi-path interference.

4.5 Walsh Codes

The PN Sequence codes have a disadvantage that they are not perfectly orthogonal to each other. Hence, non-zero from undesired user may affect the performance of the receiver. So Walsh function can be used in place of PN sequence codes.

Walsh codes are otherwise known as Walsh Hadamard codes. These are obtained by selecting as code words the rows of a Hadamard matrix . A Hadamard matrix is a N*N matrix of 1's and -1's, such that each row differs from any other row in exactly N/2 locations. One row contains all -1's with the remainder containing N/2 1's and -1's.

A 4 by 4 walsh code is given by

$$\begin{bmatrix} C_1 \\ C_2 \\ C_3 \\ C_4 \end{bmatrix} = \begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & -1 & 1 & -1 \\ 1 & 1 & -1 & -1 \\ 1 & -1 & -1 & 1 \end{bmatrix} \rightarrow$$

4-Ary Walsh Codes

Walsh codes are mathematically orthogonal codes. As such, if two Walsh codes are correlated, the result is intelligible only if these two codes are the same. As a result, a Walsh-encoded signal appears as random noise to a CDMA capable mobile terminal, unless that terminal uses the same code as the one used to encode the incoming signal.

4.6 Processing Gain

An important concept relating to the bandwidth is the processing gain (G_p). This is a theoretical system gain that reflects the relative advantage that frequency spreading provides. The processing gain is equal to the ratio of the chipping frequency to the data frequency: Gp = Fc / Fi

There are two major benefits from high processing gain:

- Interference rejection: the ability of the system to reject interference is directly proportional to G_p.
- System capacity: the capacity of the system is directly proportional to G_p.
- So the higher the PN code bit rate (the wider the CDMA bandwidth), the better the system performance.

4.7 INTRODUCTION TO MULTI CARRIER DIRECT SEQUENCE CDMA

A multicarrier DS-CDMA (MC-DS-CDMA) system is an orthogonal frequency-division multiplexing (OFDM) system in which a direct sequence code-division multiple access (DS-CDMA) signal is transmitted on each OFDM subcarrier. So, on each subcarrier, the signal of several users is multiplexed on the basis of the CDMA codes assigned to each user. In a conventional MC-DS-CDMA system, each user has a specific spreading code and the user employs this code to spread the data on each subcarrier

Transmitter

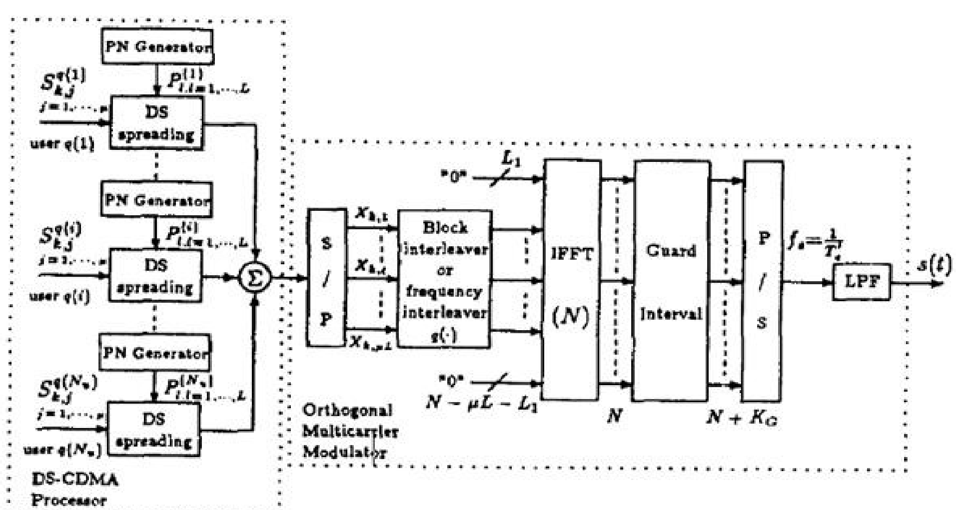

Figure-4.4

1. There are N users each transmitting a QPSK symbol every T_s seconds.

2. Every user is assigned a specific PN-sequence code. The codes corresponding to different users bear very small cross-correlation with each other. The code consists of L chips of duration $T_C=T_S/L$.

3. The kth symbol of each user is spreaded in accordance with the corresponding PN-code of the user.

4. We then perform a sum on the *N* users during a block duration T_S. After serial to parallel conversion, we obtain the L point IFFT of the parallel stream and then, again parallel to serial conversion is carried out and transmitted over the channel.

5. The same operation is carried out for every symbol transmitted per user.

RECEIVER:-

1. After a serial to parallel conversion the FFT is carried out and then immediately parallel to serial conversion is employed.

2. Then the dispreading is made by multiplying the serial data stream at the output of the converter with the corresponding codes of the different users, which are known in advance, to get the QPSK symbol transmitted by the corresponding users.

3. This process is repeated for every incoming L chips of duration T

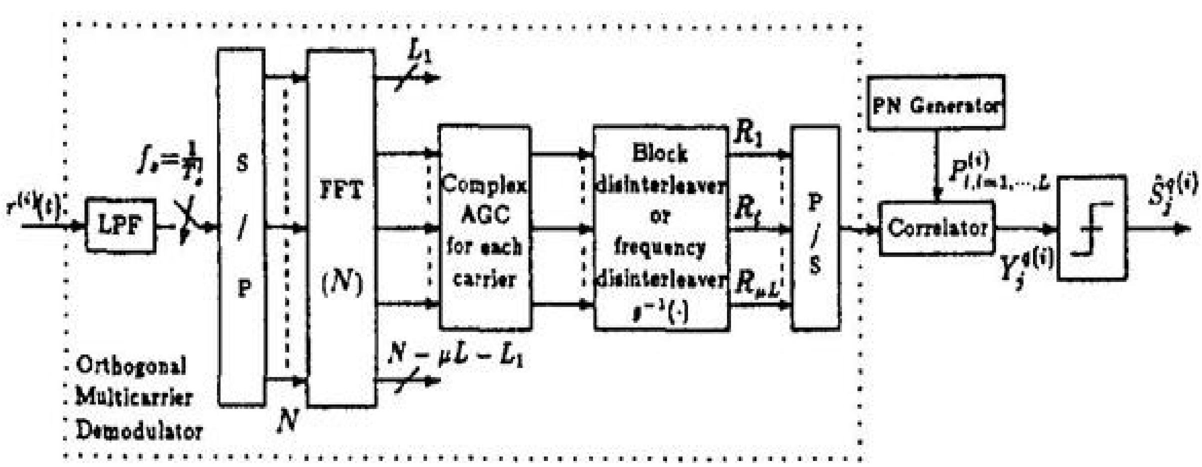

Figure 4.5

CHAPTER-5

CHANNEL ESTIMATION TECHNIQUES

5.1 INTRODUCTION

A wideband radio channel is normally frequency selective and time variant. For an OFDM mobile communication system, the channel transfer function at different subcarriers appears unequal in both frequency and time domains. Therefore, a dynamic estimation of the channel is necessary. Pilot-based approaches are widely used to estimate the channel properties and correct the received signal. In this chapter we have investigated two types of pilot arrangements.

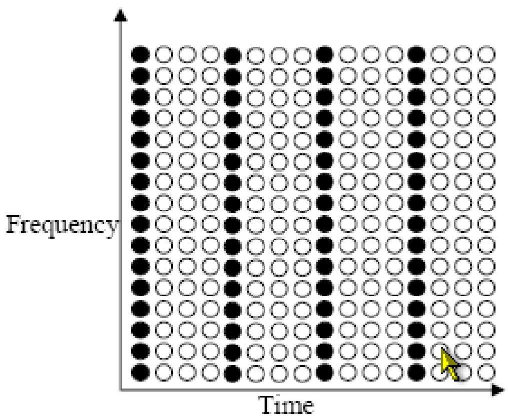

Figure-5.1 Block type pilot arrangement

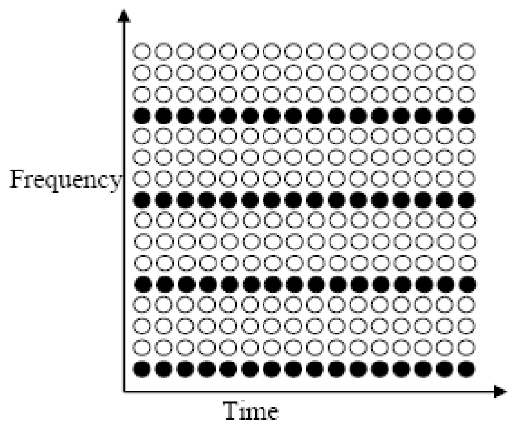

Figure-5.2 Comb type pilot arrangement

The first kind of pilot arrangement shown in Figure 5.1 is denoted as block-type pilot arrangement. The pilot signal assigned to a particular OFDM block, which is sent periodically in time-domain. This type of pilot arrangement is especially suitable for slow-fading radio channels. Because the training block contains all pilots, channel interpolation in frequency domain is not required. Therefore, this type of pilot arrangement is relatively insensitive to frequency selectivity. The second kind of pilot arrangement shown in Figure 5.2 is denoted as comb-type pilot arrangement. The pilot arrangements are uniformly distributed within each OFDM block. Assuming that the payloads of pilot arrangements are the same, the comb-type pilot arrangement has a higher re-transmission rate. Thus the comb-type pilot arrangement system is provides better resistance to fast-fading channels. Since only some sub-carriers contain the pilot signal, the channel response of non-pilot sub-carriers will be estimated by interpolating neighboring pilot sub-channels. Thus the comb-type pilot arrangement is sensitive to frequency selectivity when comparing to the block-type pilot arrangement system.

5.2 CHANNEL ESTIMATION BASED ON BLOCK-TYPE PILOT ARRANGEMENT

In block-type pilot based channel estimation, OFDM channel estimation symbols are transmitted periodically, in which all sub-carriers are used as pilots. If the channel is constant during the block, there will be no channel estimation error since the pilots are sent at all carriers. The estimation can be performed by using either LSE or

MMSE .If inter symbol interference is eliminated by the guard interval, we write in matrix notation:

$$Y = XFh + W$$
$$= XH + W$$

<5.1>

Where

$$X = diag\{X(0), X(1), \ldots\ldots X(N-1)\}$$
$$Y = [Y(0), Y(1), \ldots\ldots Y(N-1)]^T$$
$$W = [W(0), W(1), \ldots\ldots W(N-1)]^T$$
$$H = [H(0), H(1), \ldots. H(N-1)]^T = DFT_N\{h\}$$
$$F = \begin{bmatrix} W_N^{00} & \cdots & W_N^{0(N-1)} \\ \vdots & \ddots & \vdots \\ W_N^{(N-1)0} & \cdots & W_N^{(N-1)(N-1)} \end{bmatrix}$$
$$W_N^{nk} = \frac{1}{N} e^{-j2\pi(n/N)k}$$

〈5.2〉

5.2.1 Minimum Mean Square Error (MMSE) Estimation:

MSE (mean square error) is expresses as

$$J(e) = E[(H - \hat{H})^2]$$
$$= E[(H - \hat{H})^{\mathcal{H}}(H - \hat{H})]$$

〈5. 3〉

Invoking the well-known orthogonality principle in order to minimize the mean square error vector e =H-H^ has to be set orthogonal by the MMSE equalizer to the estimators input vector Y.

$$E[(H - \hat{H})Y^{\mathcal{H}}] = 0$$

$$\Rightarrow E[HY^{\mathcal{H}}] - ME[YY^{\mathcal{H}}] = 0$$
$$\Rightarrow E[FhY^{\mathcal{H}}] - ME[YY^{\mathcal{H}}] = 0$$

<5.4>

If the time domain channel vector h is Gaussian and uncorrelated with the channel noise W, then

$$FR_{hY} = MR_{YY}$$ <5.5>

Where $R_{hY} = E[hY^H]$ and $R_{YY} = E[YY^H]$

$$R_{hY} = E[hY^H] = E[h(XFh + w)^H]$$
$$R_{hY} = R_{hh}F^H X^H$$

because of $E[hw^H] = 0$ i.e. h is uncorrelated with w.

And
$$R_{YY} = E[YY^H] = E[(XFh + w)(XFh + w)^H]$$
$$R_{YY} = XFR_{hh}F^H X^H + \sigma^2 I_N$$

where σ^2 is the variance of noise.

$$M = FR_{hY} R_{YY}^{-1}$$
$$\hat{H} = FR_{hY} R_{YY}^{-1} Y$$

The time domain MMSE estimate of h is given by

$$\hat{h}_{MMSE} = R_{hY} R_{YY}^{-1} Y$$

<5.6>

5.2.2 Least Square Error (LSE) Estimation:

We have to minimize

$$J = (Y - XH)^H (Y - XH)$$
$$= (Y^T - H^H X^H)(Y - XH)$$
$$= Y^H Y - Y^H XH - H^H X^H Y + H^H X^H XH$$ <5.7>

For minimization of J we have to differentiate J with respect to H

$$\left.\frac{\partial J}{\partial H}\right|_{\hat{H}} = 0$$

$$\Rightarrow \hat{H} = X^{-1}Y$$

<5.8>

The time domain LS estimate of h is given by

$$\hat{h} = F^{H} X^{-1} Y$$

<5.9>

5.3 CHANNEL ESTIMATION BASED ON COMB-TYPE PILOT ARRANGEMENT

In comb-type based channel estimation, the N_p pilot signals are uniformly inserted into X(k) according to following equation:

$$\begin{aligned} X(k) &= X(mL + l) \\ &= \{X_p(k), \qquad l = 0 \\ &= \{\inf.data \qquad l = 1, \ldots\ldots. L-1 \end{aligned}$$

<5.10>

L = number of carriers/*Np*
$x_p(m)$ is the *mth* pilot carrier value.

We define $\{Hp(k)\ k = 0, 1, \ldots Np\}$ as the frequency response of the channel at pilot sub-carriers. The estimate of the channel at pilot sub-carriers based on LS estimation
is given by:

$$H_e = \frac{Y_p}{X_p} \qquad k = 0, 1, \ldots N_p - 1$$

<5.11>

Yp(*k*) and *Xp*(*k*) are output and input at the *kth* pilot sub-carrier respectively.

Since LS estimate is susceptible to noise and ICI, MMSE is proposed while compromising complexity. Since MMSE includes the matrix inversion in each iteration, the simplified linear MMSE estimator is suggested in which the inverse is only needed to be calculated once. The complexity is further reduced with a low-rank approximation by using singular value decomposition.

5.4. IT BASED CHANNEL ESTIMATION

5.4.1 SYSTEM MODEL

Consider a Single Input Single Output (SISO) discrete-time baseband, complex envelope equivalent, communications link, where *x(n)* is the sequence of elements drawn from a finite alphabet representing the information symbols. Let the statistical mean of x(n) be zero. The Implicit Training sequence c(n)to be defined later, is added to *x(n)* to produce the transmitted signal *s(n)*

$$s(n) = x(n) + c(n)$$ <5.12>

The sequence c(n) is periodic with period *P* and the average power in c(n) is

$$\sigma_c^2 = (1/P)\sum_{i=0}^{P-1} | c(i) |^2 .$$ <5.13>

Under these conditions *s(k)* will be a cyclo-stationary sequence possessing a periodically time varying mean μ(n)=c(n).s(n) travels through a channel with discrete time finite impulse response h(n) and after the output is combined with an AWGN, w(n) with zero mean and variance ϭ^2 we get the received signal

$$y(n) = \sum_{l=0}^{L-1} h(l)s(n-l) + w(n)$$ <5.14>

Here the channel is assumed to have a finite impulse response (FIR) of order *L*. The order *L* is determined by dividing the maximum path delay by the sampling period. First order statistics is used for IT based channel estimation. Now define

$$z(j) = E\left[y\left(iP + j\right)\right], \quad j = 0,1,...P-1$$ <5.15>

Expanding the above equation and keeping in mind that *x(n)* and w(n) are drawn from zero mean random processes, we can write

$$z(j) = \sum_{l=0}^{L-1} h(l)c(iP + j - l)$$

$$= \sum_{l=0}^{L-1} h(l)c(j-l)_P, \quad j = 0,1,...,P-1.$$ <5.16>

In which (.)*P* indicates the arithmetic modulo-P added version of (.) defined as

$$c(n)_P = c(n+P),$$ <5.17>

Here (5.16) represents a set of P unique linear equations one for each value j, as c(n) is periodic in P. In matrix notation this can be represented as

$$\mathbf{z} = \mathbf{Ch}$$ <5.18>

$\mathbf{h} = \left[h(P-1), h(P-1), \ldots, h(0)\right]^T$, $\mathbf{z} = \left[z(P-1), z(P-2), \ldots, z(0)\right]^T$ and $\mathbf{C} = circ(c(0), c(1), \ldots, c(P-1))$.

5.4.2. Channel Estimation

The basic problem is to estimate the elements of **z** from a finite, say N number of received samples. When $Q=N/P$ (assuming that N is a multiple of P), then from (5.15) under the assumptions of stationarity and ergodicity, we may use the following expression for estimating $z(j)$,

$$\hat{z}(j) = \frac{1}{Q}\sum_{i=0}^{Q-1} y(iP+j), \quad j = 0,1,\ldots,P-1$$ <5.19>

The above expression is asymptotically unbiased. From (5.18) and (5.19) the channel estimate can be obtained as,

$$\hat{\mathbf{h}} = \mathbf{C}^{-1}\hat{\mathbf{z}}$$ <5.20>

The properties of the IT sequence play a critical role in channel identification. The conditions to be satisfied for channel identification are the following. i) $P \geq L$. The periodicity P of the implicit training sequences puts an upper bound on the model order, L of the estimated channel. ii) The circulant matrix **C** of the IT sequences should be of full rank. Since first order statistics is used to estimate the channel, the effect of the unknown dc offset has to be considered. The offset is introduced into the system at the RF front end of the receiver. This introduces a constant ambiguity factor in the channel estimate. Here we assume a zero dc offset in the receiver. The training sequences are independent of the channel characteristics, with unity Peak to Average Power Ratio (PAPR).

$$|c(n)| = \sigma_c, \qquad n = 0,1,\ldots,P-1 \quad \text{and}$$
$$|C(l)| = \sqrt{P}\sigma_c, \quad l = 0,1,\ldots,P-1$$

The sequence satisfying the above properties can be a chirp sequence given by c(n)

$$c(n) = \sigma_c \exp(j2\pi n(n+i)/P)$$ <5.21>

For even value of P, $i=2$ and when P is odd, $i=1$,

CHAPTER-6

SIMULATIONS AND RESULTS

6.1 OFDM MODEL USED IN SIMULATIONS

An OFDM system is modelled using Matlab to allow various parameters of the system to be varied and tested. The aim of doing the simulations is to measure the performance of OFDM system under different channel conditions, and to allow for different OFDM configurations to be tested. An OFDM system model used is shown in Figure 6.1.

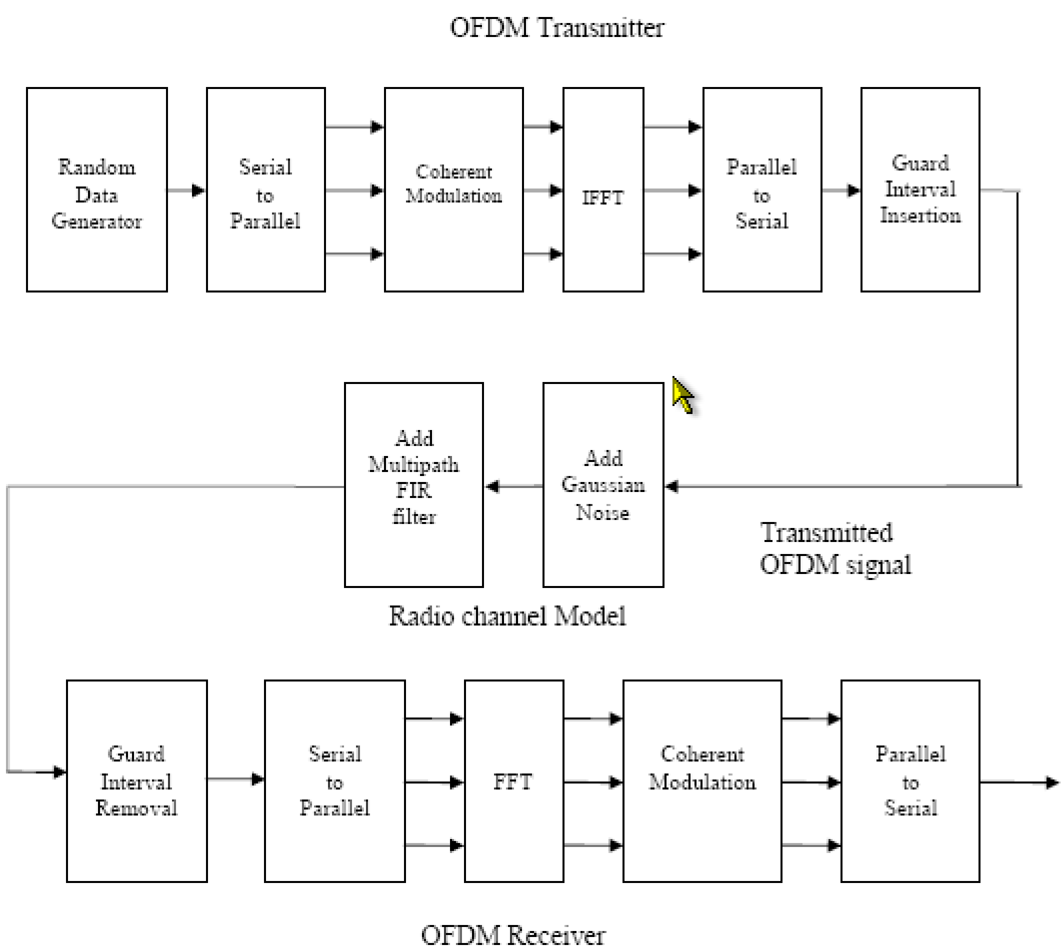

Figure-6.1

Random data generator:
The input signal which we used is the random data generated by randn() function of the matlab,
.

SERIAL TO PARALLEL CONVERSION:
The input serial data stream is applied to a serial to parallel converter and shifted into a parallel format. The data is then transmitted in parallel by assigning each data word to one carrier in the transmission.

MODULATION OF DATA
The data to be transmitted on each carrier is modulated into a QAM and M-ary PSK format. The data on each symbol is thus mapped. In the simulations we used BPSK and QPSK type digital modulation.

INVERSE FAST FOURIER TRANSFORM
After the required spectrum is worked out, an inverse fourier transform is used to find the corresponding time domain waveform.

GUARD INTERVAL
The guard period is then added to the start of each symbol .The length of the guard interval should be at least more than the multipath delay spread of the fading channel.

PARALLEL TO SERIAL CONVERTER
The parallel data is converted to the serial form to be passed through the channel.

AWGN NOISE
The serially transmitted data is corrupted by additive white Gaussian noise (zero mean) at a particular signal to noise ratio. As the SNR value increases, the chances of the bits being corrupted decreases.

CHANNEL MODEL USED
A Channel model is then applied to the noise-corrupted transmitted signal. Standard channels are that of Rayleigh fading channel model and Rician channel are mostly used ones. In our simulation we have used a standard multipath FIR filter as the channel.

RECEIVER
The receiver basically does the reverse operation to the transmitter. The serially received data is converted to parallel form and then guard period is removed. The FFT of each symbol is then taken to find the original transmitted spectrum. The OFDM symbols are then subjected to demodulation by using a bank of sub-carriers as those used in the transmitter. The words are combined back to produce the original data stream.

6.1.1 Simulation Results for OFDM with BPSK:

FFT size=64;
Number of Sub-carries=64;
Bits transmitted=10^5;
Cyclic Prefix length=16;
Channel parameters= [.23 .96 .23]

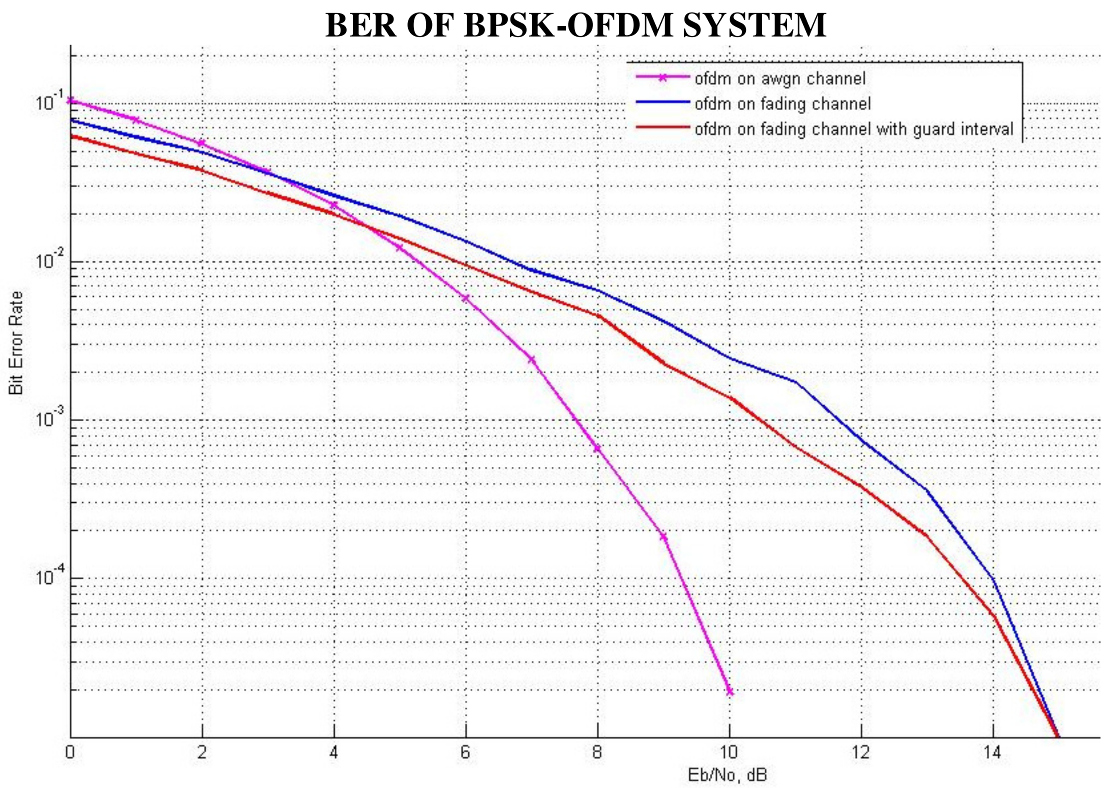

Figure-6.2

6.1.2 Simulation Results for OFDM with QPSK:

FFT size=64;
Number of Sub-carries=64;
Bits transmitted=10^5;
Cyclic Prefix length=16;
Channel parameters= [.23 .96 .23]

BER OF QPSK-OFDM SYSTEM

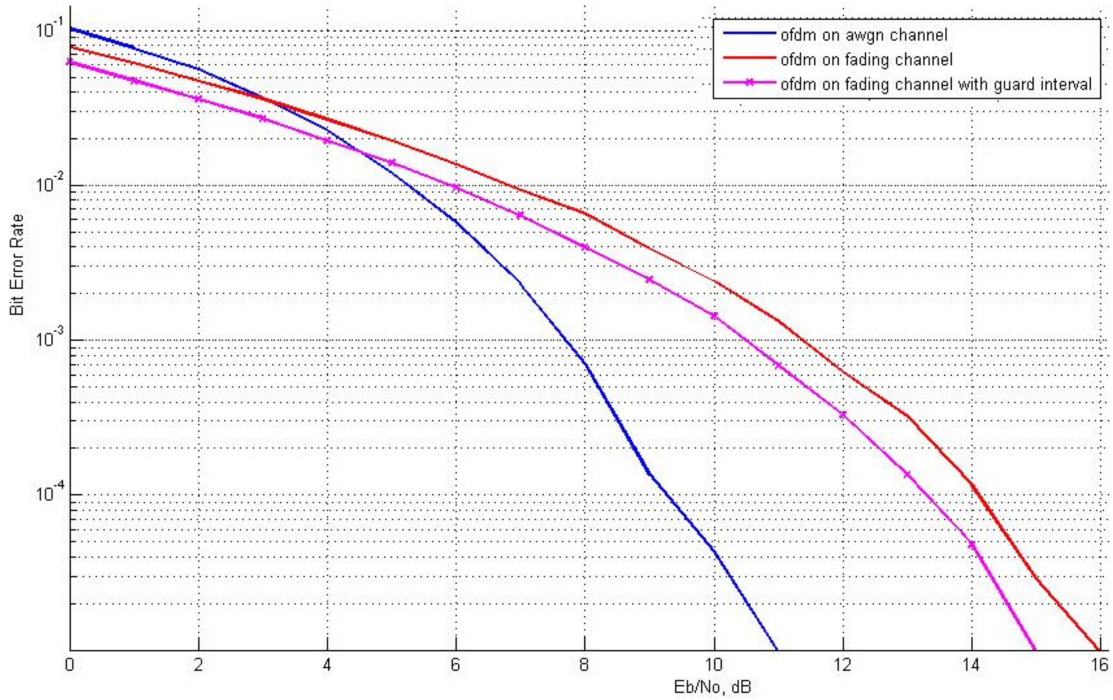

Figure 6.3

6.1.3 COMPARISON BETWEEN BPSK AND QPSK BASED OFDM

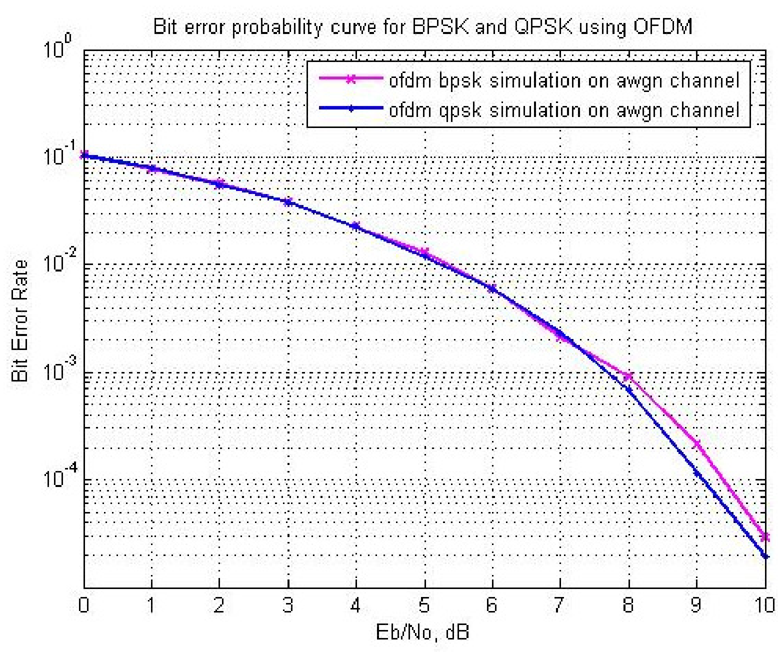

Figure 6.4

6.2 CDMA MODEL USED IN SIMULATIONS

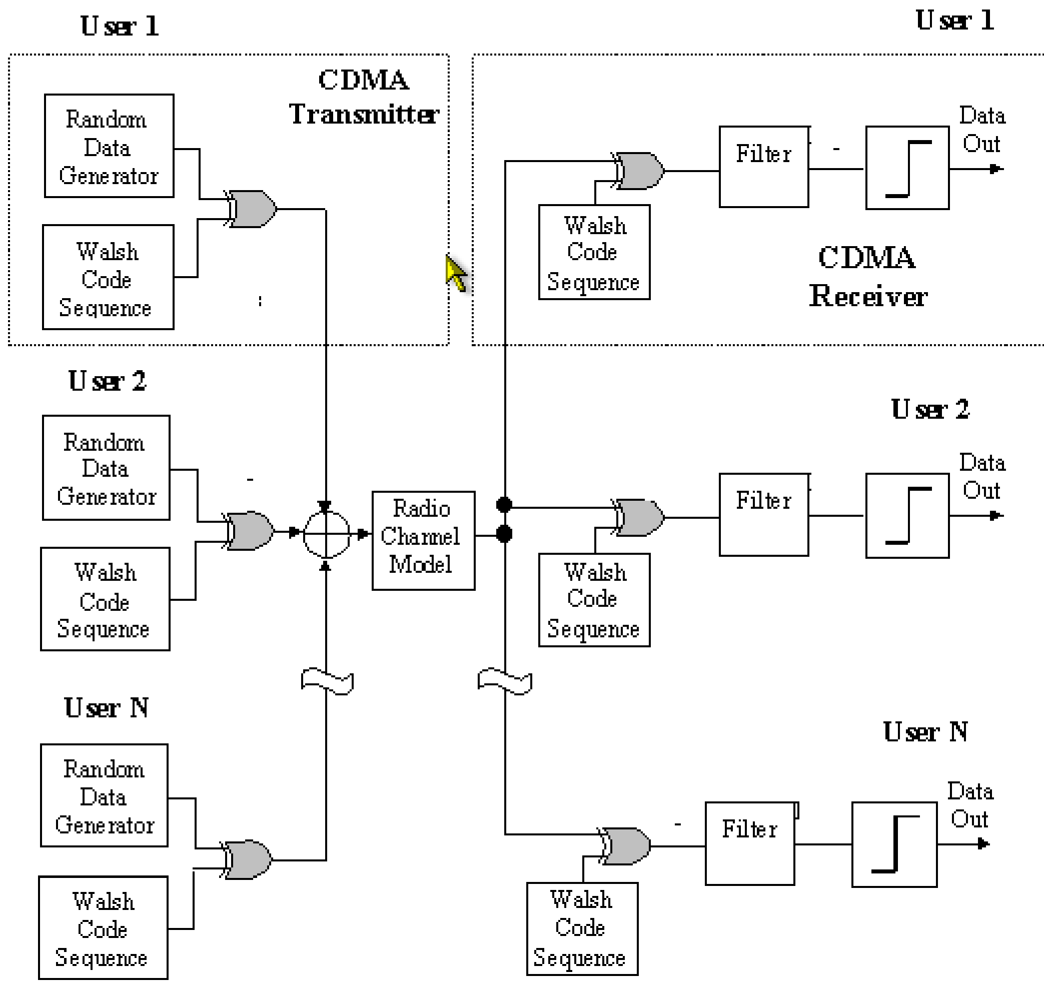

Figure-6.5

The forward link of the CDMA system modelled uses orthogonal Walsh codes to separate the users. Each user is randomly allocated a Walsh code to spread the data to be transmitted. The transmitted signals from all the users are combined together, then passed through a radio channel model. This allows for clipping of the signal, adding multipath interference, and adding white gaussian noise to the signal. The receiver uses the same Walsh code that was used by the transmitter to demodulate the signal and recover the data. After the received signal has been despread using the Walsh code, it is sub-sampled back down to the original data rate. This is done by using an integrate-and-dump filter, followed by a comparator to decide whether the data was a 1 or a 0. The received data is then compared with the original data transmitted to calculate the bit error rate (BER).

6.2.1 Simulation Results for CDMA:

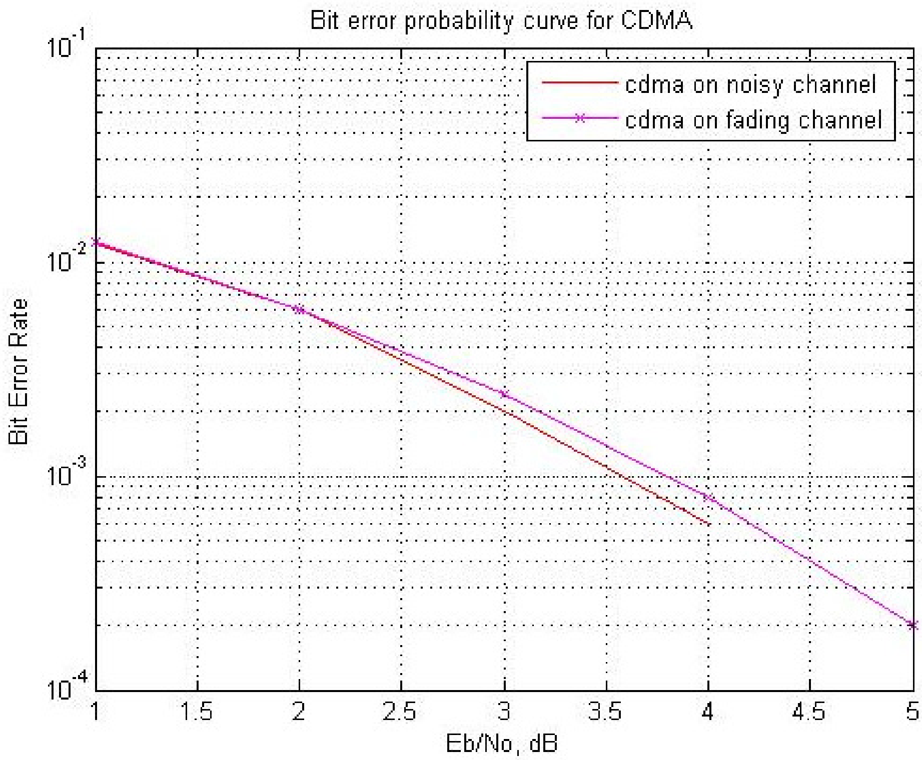

Figure 6.6

6.3 Simulation Results for MC DS-CDMA:

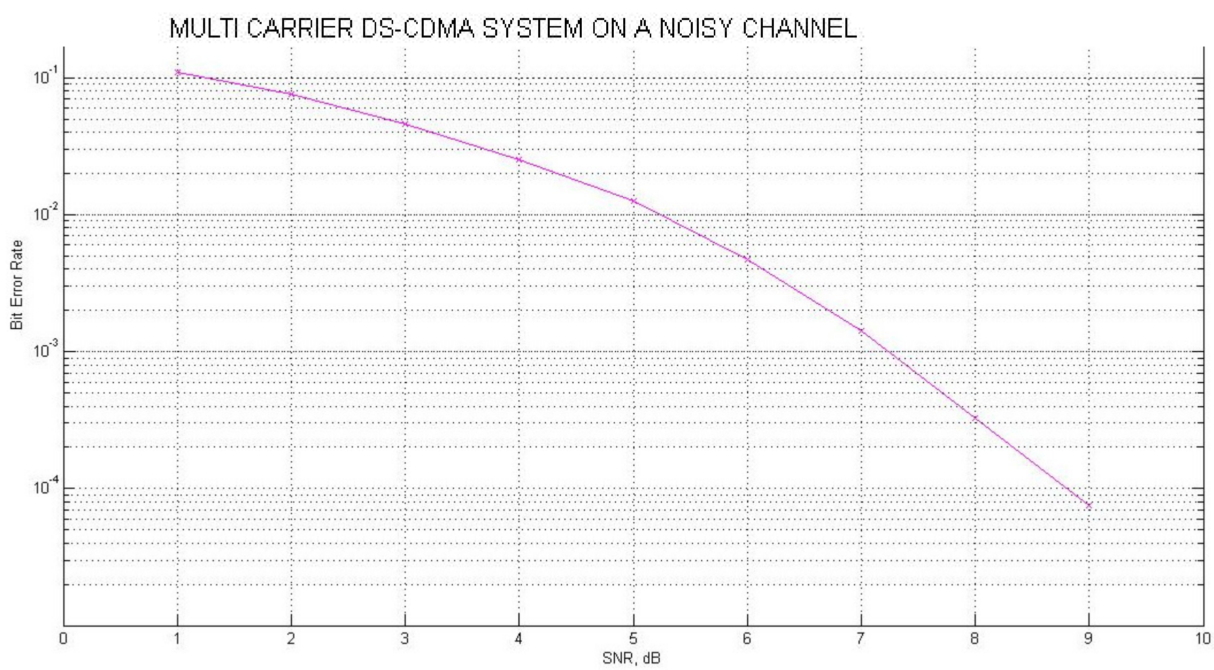

Figure 6.7

6.4 SIMULATION RESULT FOR CHANNEL ESTIMATION

6.4.1 SIMULATION RESULT FOR BLOCK TYPE PILOT ARRANGEMENT

In the simulation we consider a system operating with a bandwidth of 500 kHz, divided into 64 tones with total symbol period of 138 µs, of which 10 µs is a cyclic prefix. Sampling is performed with a 500 kHz rate. A symbol thus consists of 69 samples, five of which are contained in the cyclic prefix. 10,000 channels are randomized per average SNR. We consider the two ray channel as h(t)= [0.5 3.5]. Figure 5.3 demonstrates Mean square error of channel estimation at different SNRs in dB. As SNR increases mean square error decreases for both LSE and MMSE. Figure 5.4 shows Average SNR versus Symbol Error Rate (SER). As SNR increases Symbol Error Rate decreases for both cases. For a given SNR, MMSE estimator shows better performance than LSE estimator. The complexity of MMSE estimators will be larger than LSE estimators but give better performance in comparison to LSE. It should be noticed that MMSE estimators have been derived under assumption of known channel correlation and noise variance. In practice these quantities R_{hh} and σ^2 are either taken as fixed or estimated, possibly in an adaptive way. This will increase the estimator complexity but improve performance over LSE estimators.

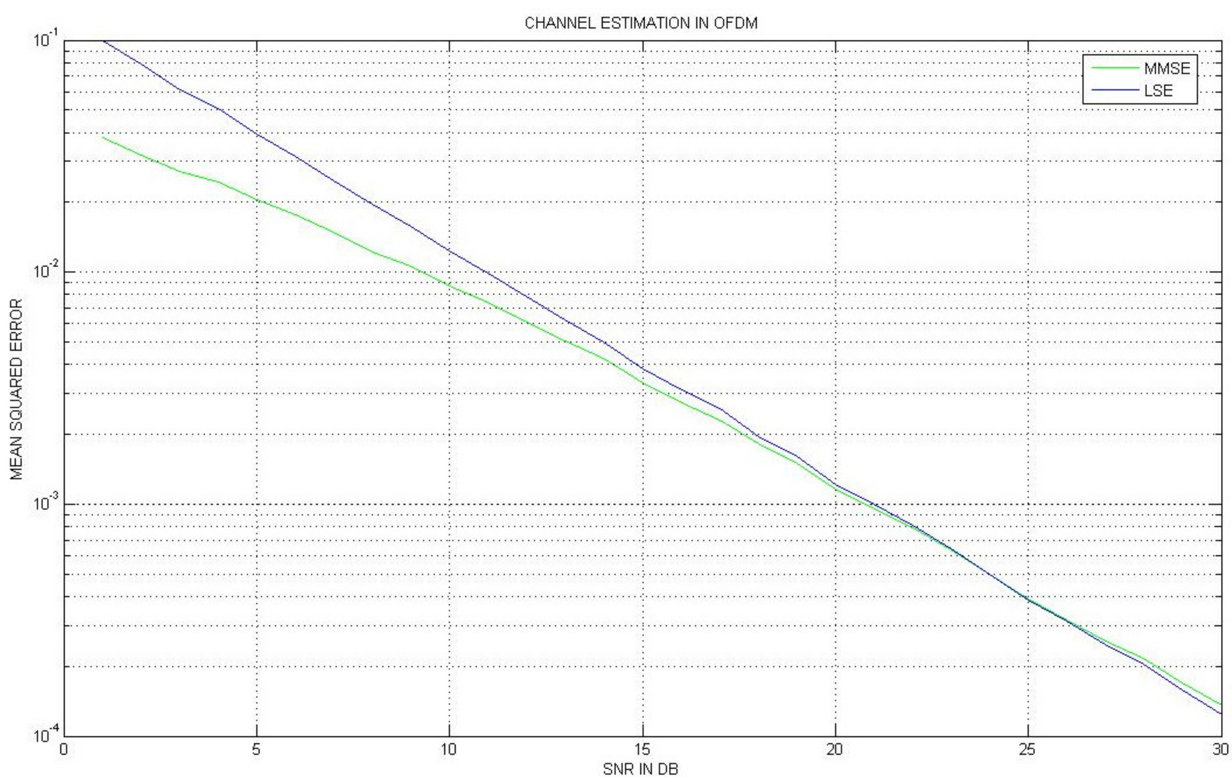

Figure 6.8

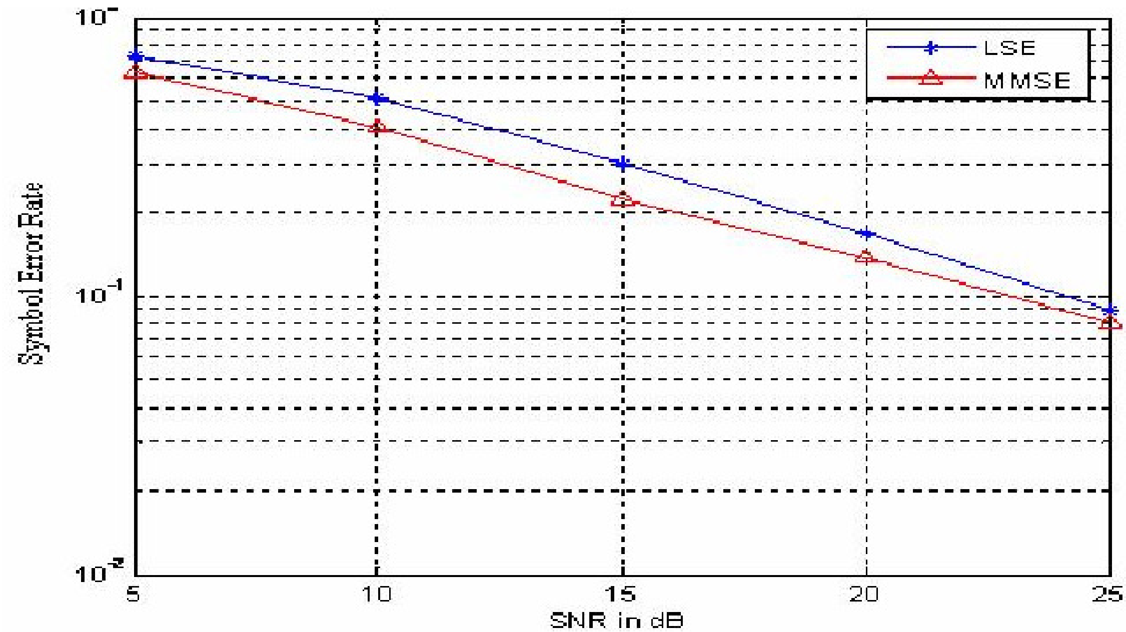

Figure 6.9

6.4.2 SIMULATION RESULT FOR COMB-TYPE PILOT ARRANGEMENT

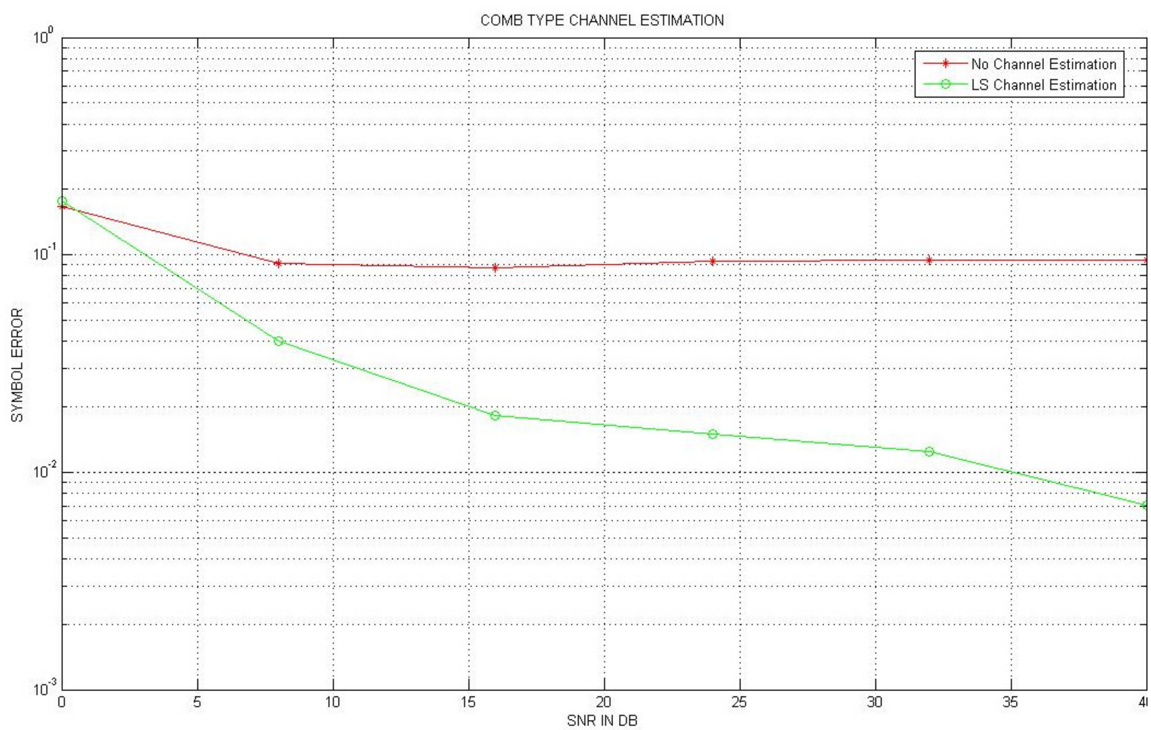

Figure 6.10

6.4.3 SIMULATION RESULT FOR IMPLICIT TRAINING METHOD

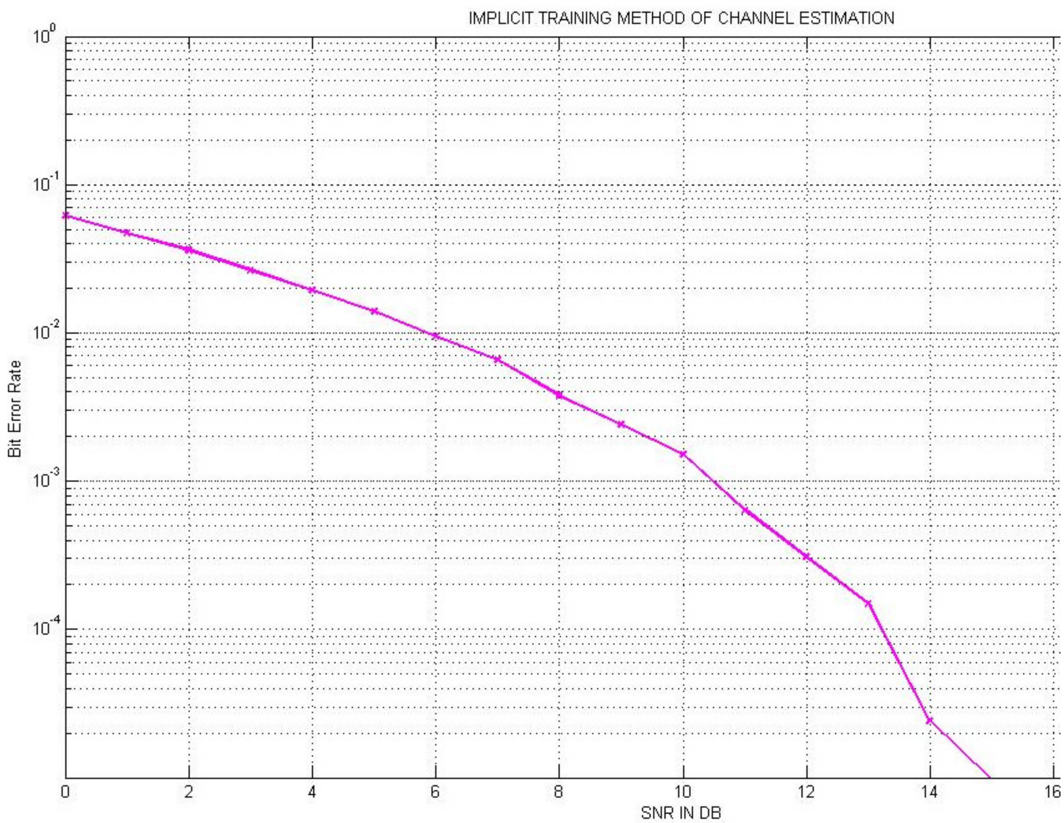

Figure 6.11

CHAPTER-7

CONCLUSION AND FUTURE WORK

7.1 CONCLUSION

In this work, we have studied LSE and MMSE estimators for both block type and comb type pilot arrangement. The estimators in this study can be used to efficiently estimate the channel in an OFDM system given certain knowledge about channel statistics. The MMSE estimators assume a priori knowledge of noise variance and channel covariance. Moreover, its complexity is large compare to the LSE estimator. For high SNRs the LSE estimator is both simple and adequate. The MMSE estimator has good performance but high complexity. The LSE estimator has low complexity, but its performance is not as good as that MMSE estimator basically at low SNRs.

In comparison between block and comb type pilot arrangement, block type of pilot arrangement is suitable to use for slow fading channel where channel impulse response is not changing very fast. So that the channel estimated, in one block of OFDM symbols through pilot carriers can be used in next block for recovery the data which are degraded by the channel. Comb type pilot arrangement is suitable to use for fast fading channel where the channel impulse response is changing very fast even if one OFDM block. So comb type of pilot arrangement can't be used in this case. We used both data and pilot carriers in one block of OFDM symbols. Pilot carriers are used to estimate the channel impulse response. The estimated channel can be used to get back the data sent by transmitter certainly with some error.

In the Implicit Training (IT) method for channel estimation, periodic pilots are superimposed on to the data at a low power, thus avoiding the need for additional time slots for training. This induces first order cyclo-stationary statistics in the received data which is used to perform channel estimation. The first order statistics is estimated from a finite number of received samples. Hence the deterministic mean of the data affects the channel estimate. The deterministic mean of the data is subtracted from the data before transmission and iterative detectors are employed for symbol recovery. This scheme requires a good estimate of the channel model order. Here, we avoid transmitting data on those subcarriers which are influenced by the IT sequence. The number of such subcarriers in one OFDM symbol is equal to the period of the IT sequence. This improves the accuracy of the channel estimate. Also, this method works with an upper bound on the channel model order. The loss in bandwidth makes the system comparable with channel estimation schemes in OFDM in which pilot tones are multiplexed with data tones in each OFDM symbol, in terms of spectral efficiency. As compared to the channel estimation using comb type pilots, IT based method gives an improved performance in frequency selective fading channels as more accurate channel estimates are available at all the subcarriers.

7.2 Future Work

With the increasing requirements for future wireless applications, MC-DS-CDMA may be considered as the most important for 4G wireless systems. This system has the ability to incorporate very large bandwidths without sacrificing equalization complexity. The long symbol duration is effective at mitigating ISI, and adaptive modulation or frequency diversity can be used to provide protection against destructive fades.

In this paper though we have studied about the principles of CDMA and MC DS-CDMA we haven't implemented the channel estimation techniques as in the case of OFDM. But the same can be carried out using pilot and implicit training sequence techniques to optimize the equalizer performance.

Also lies the feasibility study of Multiple Input Multiple Output (MIMO) OFDM systems. In this study we have discussed about Single Input Single Output (SISO) OFDM systems. MIMO OFDM can be implemented using multiple transmitting and receiving antennas which is an interesting work of future.

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