Evaluation of Channel Coding in OFDM Systems

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

Master of Technology in 'Telematics and Signal Processing'

By

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CERTIFICATE

This is to certify that the work in this thesis entitled **"*Evaluation of Channel Coding in OFDM Systems*"** by ***Nishar Ahamed Gugudu***, has been carried out under my supervision in partial fulfillment of the requirements for the degree of Master of Technology in **'Electronics and Communication Engineering'** during session 2005-2006 in the Department of Electronics and Communication Engineering, National Institute of Technology, Rourkela and this work has not been submitted elsewhere for a degree.

Place:

Dated:

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ABSTRACT

Channel coding plays a very important role in OFDM systems performance. The structure of OFDM systems makes channel coding more effective in confronting fading channels. Sometimes Coded OFDM is known as COFDM. The role of channel coding in conjunction with frequency and time interleaving is to provide a link between bits transmitted on separated carriers of the signal spectrum, in such a way that the information conveyed by faded carriers can be reconstructed in the receiver. Frequency selectivity, currently known to be a disadvantage, is then turned into an advantage that can be called frequency diversity. Using Channel State Information (CSI), channel coding can yield some additional gain. Channel state information is frequency response of the channel or signal to noise ratio in each carrier.

This thesis analyzes OFDM system and the effect of channel coding in reducing BER. Along with this soft decoding and decoding with CSI is also studied. Besides, performance of convolutional codes Turbo codes in OFDM systems is compared and compared. Besides, we compare the performance of convolution and turbo codes in OFDM systems. The results have been validated through simulations.

This thesis also presents Space-Frequency Coded OFDM system consisting of two transmitters and a single receiver. Simple Alamouti space time code is used. An Mary PSK modulation is used to modulate the symbols across an OFDM channel. We also proposed a variation of the scheme which tries to spread additional symbols across time-frequency attempting to increase the rate of transmission without changing the type of modulation employed or increasing the bandwidth. A Rayleigh frequency selective slow fading channel is assumed through out the analysis. SER performance of the above systems is carried out with emphasis on the modulation scheme and number of carriers.

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ABBREVIATIONS USED

OFDM ------------- Orthogonal Frequency Division Multiplexing

FDM ---------------- Frequency Division Multiplexing

RS coding------------ Reed Solomon coding

Wireless LAN--------wireless Local Area Network

DAB -------------------Digital Audio Broadcasting

DFT --------------------Discrete Fourier Transform

HDSL ------------------High bit-rate digital subscriber line

ADSL ------------------Asymmetric digital subscriber line

VDSL ------------------very high speed digital subscriber line

HDTV ------------------high definition television

SCM --------------------single carrier modulation

RECT pulse ------------Rectangular pulse

FFT ----------------------Fast Fourier Transform

QAM --------------------quadrature division multiplexing

UHF ----------------------Ultra high Frequencies

COFDM ------------------coded OFDM

CDMA ---------------------code division multiple access

TDMA --------------------time division multiple access

BPSK -------------------Binary phase shift keying

QPSK -------------------quadrature phase shift keying

IBI -----------------------Inter block interference

CP -----------------------cyclic prefix

FEC ---------------------forward error correction

RSC --------------------recursive convolutional encoder

WWAN ----------------wireless wide area network

SNR --------------------signal to noise ratio

SER ---------------------symbol error rate

BER ---------------------Bit Error rate

Chapter 1

INTRODUCTION

1.1 Introduction:

The telecommunications' industry is in the midst of a veritable explosion in wireless technologies. Once exclusively military, satellite and cellular technologies are now commercially driven by ever more demanding consumers, who are ready for seamless communication from their home to their car, to their office, or even for outdoor activities. With this increased demand comes a growing need to transmit information wirelessly, quickly and accurately. To address this need, communications engineer have combined technologies suitable for high rate transmission with forward error correction (FEC) techniques. This is particularly important as wireless communications channels are far more hostile as opposed to wire alternatives, and the need for mobility proves especially challenging for reliable communications.

1.2 Motivation:

For the most part, OFDM is the standard being used throughout the world to achieve the high data rates necessary for data intensive applications that must now become routine.
Different coding schemes like Hamming coding, Reed Solomon (RS) coding, Convolutional coding, Turbo coding have been studied and compared. A combination of the coding schemes is also implemented and simulated like RS-Convolutional coding, RS-Turbo coding.
At last Space Frequency coding is studied which performs well in a frequency selective channel.

1.3 Background literature survey:

The concept of using parallel data transmission by means of frequency division multiplexing (FDM) was published in mid 60's [2]. Some early development with this can be traced back to the 50s [1]. A U.S. patent was filled and issued in January

1970 [3]. 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. Weinstein and Ebert [5] applied the discrete Fourier transform (DFT) to parallel data transmission system as part of the modulation and demodulation process. In the 1980s, OFDM has been studied for high-speed modems [6], digital mobile communications [8] and high-density recording [7]. Various fast modems were developed for telephone networks [12]. In 1990s, OFDM has been exploited for wideband data communications over mobile radio FM channels [14], wireless LAN [11] wireless multimedia communication [15], high-bit-rate digital subscriber lines (HDSL) [16], asymmetric digital subscriber lines (ADSL) [9], very high-speed digital subscriber lines (VHDSL) [13], [17], digital audio broadcasting (DAB) [18] and HDTV terrestrial broadcasting [10].

In a classical parallel data system, the total signal frequency band is divided into N nonoverlapping frequency subchannels. Each subchannel is modulated with a separate symbol and then the N subchannels are frequency-multiplexed. It seems good to avoid spectral overlap of channels to eliminate interchannel interference. However, this leads to inefficient use of the available spectrum. To cope with the inefficiency, the ideas proposed from the mid-1960s were to use parallel data and FDM with overlapping subchannels, in which, each carrying a signaling rate b is spaced b apart in frequency 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.

1.4 Thesis contribution:

OFDM has received increased attention due to its capability of supporting high-data-rate communication in frequency selective fading environments which cause inter-symbol interference (ISI) [18]. Instead of using a complicated equalizer as in the conventional single carrier systems, the ISI in OFDM can be eliminated by adding a guard interval which significantly simplifies the receiver structure. However, in order to take advantage of the diversity provided by the multi-path fading, appropriate frequency interleaving and coding is necessary. Therefore, coding becomes an inseparable part in most OFDM applications and a considerable amount of research has focused on optimum encoder, decoder, and interleaver design for information transmission via OFDM over fading environments, e.g. [19]-[22].

Different coding schemes like Hamming coding, RS coding, Convolutional coding, Turbo coding have been studied and compared. A combination of the coding schemes is also implemented and simulated like RS-Convolutional coding, RS-Turbo coding.

At last Space Frequency coding is studied which performs well in a frequency selective channel.

1.5 Thesis outline:

Following this introduction the remaining part of the thesis is organized as under;

Chapter 2 provides the fundamental concepts of OFDM and principles behind OFDM. Propagation characteristics of mobile radio channels are also discussed along with advantages and disadvantages of OFDM compared with single carrier modulation. Chapter 3 provides discusses the implementation of OFDM that also discusses the simulation model used here. Chapter 4 discusses the different coding techniques implemented with OFDM. Chapter 5 discusses the Space Frequency coded OFDM and its implementation. Chapter 6 summarizes the work undertaken in this thesis and points to possible directions for future works.

Chapter 2

OFDM

2.1 Introduction

This thesis discusses about the evaluation of channel coding in OFDM systems. In order to establish the context and need for the work undertaken, it is necessary to discuss the fundamental concepts behind the work. This chapter brings out the need for channel coding in OFDM systems. The chapter also discusses the propagation characteristics of a mobile communication channel

This chapter is organized as follows. Following this introduction, section 2.2 discusses Fundamental concepts behind OFDM, section 2.3 discusses the propagation characteristics of a mobile communication channel, section 2.4 discusses about the principles of OFDM Section 2.5 presents the advantages and disadvantages of OFDM compared with single carrier scheme and finally section 2.8 provides the concluding remarks.

2.2 Fundamental concepts behind OFDM:

2.2.1 What is Orthogonal Frequency Division Multiplexing?

OFDM is simply defined as a form of multi-carrier modulation where the carrier spacing is carefully selected so that each sub carrier is orthogonal to the other sub carriers. Two signals are orthogonal if their dot product is zero

That is, if you take two signals multiply them together and if their integral over an interval is zero, then two signals are orthogonal in that interval. 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. As the sub carriers are orthogonal, the spectrum of each carrier has a null at the center frequency of each of the other carriers in the system. This results in no interference between the carriers, allowing them to be spaced as close as theoretically possible. Mathematically, suppose we have a set of signals Ψ , where Ψp is the p-th element in the set.

$$\int_a^b \psi_p(t)\, \psi_q^*(t)\, dt = \begin{cases} k & \text{for } p=q \\ 0 & \text{for } p \neq q \end{cases} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots(1)$$

Where, Ψp and ψ_q are p^{th} and q^{th} elements in the set.

The signals are orthogonal if the integral value is zero. Interval [a, b] is a symbol period. Since the carriers are orthogonal to each other the nulls of one carrier coincides with the peak of another sub carrier. As a result it is possible to extract the sub carrier of interest

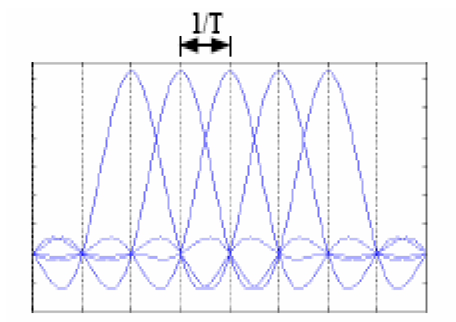


Figure: 2.2.1

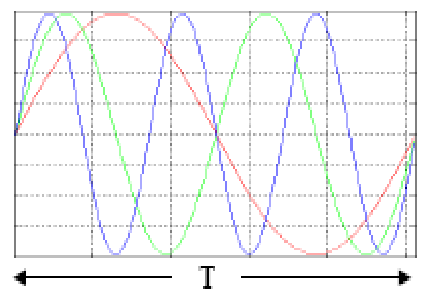
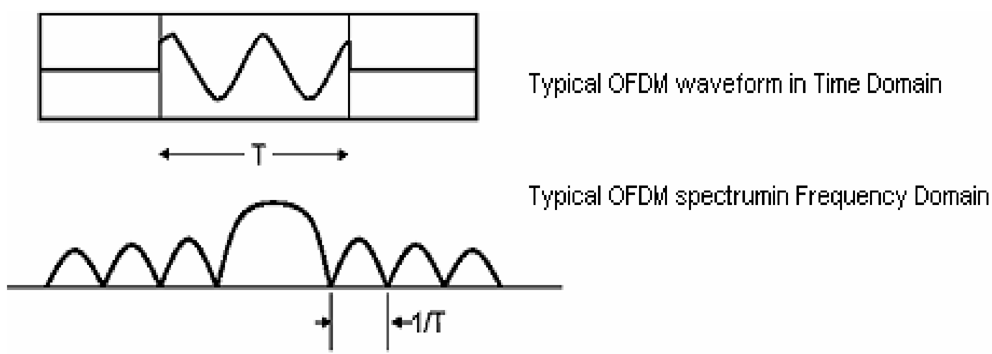


Figure 2.2.2

OFDM transmits a large number of narrowband subchannels. The frequency range between carriers is carefully chosen in order to make them orthogonal one another. In fact, the carriers are separated by an interval of 1/T, where T represents the duration of an OFDM symbol. The frequency spectrum of an OFDM transmission is illustrated in figure 2.2.1. Each sinc of the frequency spectrum in the Fig 2.2.1 corresponds to a sinusoidal carrier modulated by a rectangular waveform representing the information symbol. One could easily notice that the frequency spectrum of one carrier exhibits zero-crossing at central frequencies corresponding to all other carriers. At these frequencies, the intercarrier interference is eliminated, although the individual spectra of subcarriers overlap. It is well known, orthogonal signals can be separated at the receiver by correlation techniques [ ]. The receiver acts as a bank of demodulators, translating each carrier down to baseband, the resulting signal then being integrated over a symbol period to recover the data. If the other carriers all beat down to frequencies which, in the time domain means an integer number of cycles per symbol period (T), then the integration process results in a zero contribution from all these carriers. The waveform of some carriers in a OFDM transmission is illustrated in Fig 2.2.2. The figure indicates the spectrum of carriers significantly over laps over the other carrier. This is contrary to the traditional FDM technique in which a guard band is provided between each carrier.

From the Figures illustrated, it is clear that OFDM is a highly efficient system and hence is often regarded as the optimal version of multi-carrier transmission schemes. The number of sub channels transmitted is fairly arbitrary with certain broad constraints, but in practical systems, sub channels tend to be extremely numerous and close to each other. For example the number of carriers in 802.11 wireless LAN is 48 while for Digital Video Broadcast (DVB) it is as high as 6000 sub carriers. If we consider a single OFDM carrier, we can model the transmitted pulse as a sinusoid multiplied by a RECT function. In the frequency domain, the resulting spectrum has a $\frac{\sin(x)}{x}$ shape centered at the carrier frequency as shown in the Figure 2.2.3.



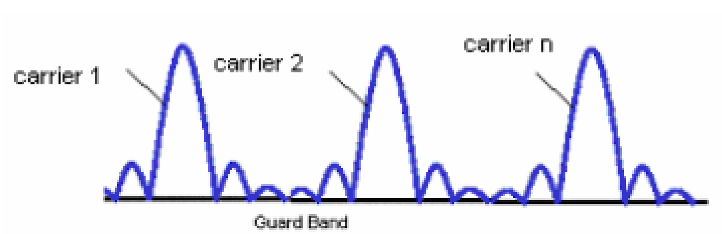


Figure 2.2.3: A Single carrier of OFDM

It is worth mentioning here that relative to single carrier Modulation technique (SCM), the OFDM carriers occupy a significant amount of bandwidth of the spectrum relative to the symbol rate. This characteristic is not a problem given that the carriers overlap significantly. The slow sin (x)/x roll off, which implies a wider carrier bandwidth is only an issue at the edge of the channel spectrum. Standards like 802.11a, allow the RECT pulse to be modified such that the rising and falling edges are softer (Raised cosine) at the edge of their assigned spectrum. This helps constrain the spectrum without affecting data transmissions OFDM offers several advantages over single carrier system like better multipath effect immunity, simpler channel equalization and relaxed timing acquisition constraints. But it is more susceptible to local frequency offset and radio front-end non-linearities. The frequencies used in OFDM system are orthogonal. Neighboring frequencies with overlapping spectrum can therefore be used. This results in efficient usage of BW. The OFDM is therefore able to provide higher data rate for the same BW introduced between the different carriers and in the frequency domain, which results in a lowering of spectrum efficiency.

Moreover, to eliminate the banks of subcarrier oscillators and coherent demodulators required by frequency-division multiplex, completely digital implementations could be built around special-purpose hardware performing the fast Fourier transform (FFT), which is an efficient implementation of the DFT. Recent advances in very-large-scale integration (VLSI) technology make high-speed, large-size FFT chips commercially affordable. Using this method, both transmitter and receiver are implemented using efficient FFT techniques that reduce the number of operations from N^2 in DFT down to N log N. In the 1980s, OFDM was studied for high-speed modems, digital mobile communications, and high-density recording. One of the systems realized the OFDM techniques for multiplexed QAM using DFT and by using pilot tone, stabilizing carrier and clock frequency control and implementing trellis coding are also implemented. Moreover, various-speed modems were developed for telephone networks.

In the 1990s, OFDM was exploited for wideband data communications over mobile radio FM channels [14], high-bit-rate digital subscriber lines (HDSL; 1.6 Mbps) [16], asymmetric digital subscriber lines (ADSL; up to 6 Mbps) [9], very-high-speed digital subscriber lines (VDSL; 100 Mbps) [17], digital audio broadcasting (DAB) [18], and high-definition television (HDTV) terrestrial broadcasting [10].

2.3. Propagation Characteristics of mobile radio channel

In an ideal radio channel, the received signal would consist of only a single direct path signal, which would provide perfect reconstruction of the transmitted signal at receiver.

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 and it is termed as 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.

2.3.1 Attenuation:

Attenuation is the drop in the signal power when transmitting signal from one point to 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. 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 then high frequency signals. Thus high frequency signals, especially, Ultra High Frequencies (UHF), and microwave signals require line of sight for adequate signal strength. To over come the problem of shadowing, transmitters are usually elevated as high as possible to minimize the number of obstructions. Typical amounts of variation in attenuation due to shadowing are shown in Table 2.1. Here typical attenuation due to shadowing in heavy built up urban center, sub-urban area, open rural area and terrain irregularities are shown.

Shadowed areas tend to be large, resulting in the rate of change of the signal power being slow. For this reason, it is termed slow-fading or log-normal shadowing.

Table 2.1 Typical Attenuation in a radio channel

Description	Typical Attenuation due to Shadowing
Heavy built-up urban center	20dB variation from street to street
Sub-Urban area(fewer large buildings)	10dB greater signal power than built up urban center
Open rural area	20dB greater signal power than sub-urban areas
Terrain irregularities and tree foliage	3-12dB signal power variation

2.3.2 Multipath Effects:

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 from 10-30dB over a short distance. Figure 2.3.1 shows the level of attenuation that can occur due to the fading. Here the probability of signal level being less than the value given is shown for different signal levels.

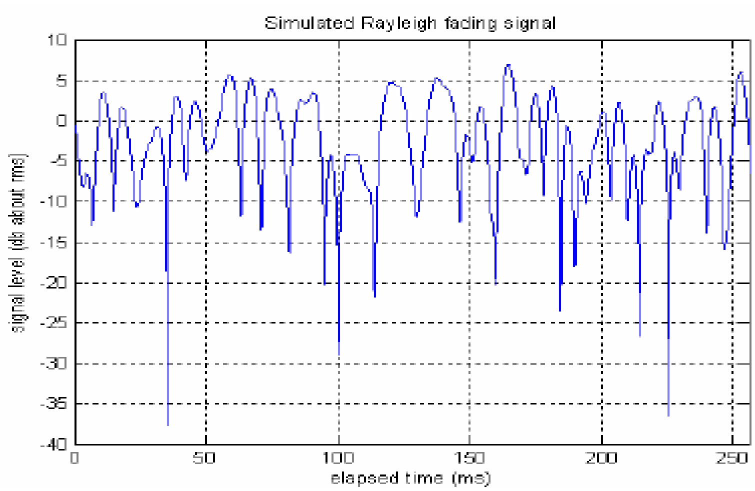


Fig: 2.3.1 A typical Rayleigh Fading Envelope

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 2.2 shows the probability of the signal level for the Rayleigh distribution

Table 2.2: Cumulative Distibution for 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

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 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 then the entire signal. The information in the lost carriers can be recovered provided enough FEC is sent.

2.3.3 Delay Spread:

The received radio signal from a transmitter consists of typically a direct signal, plus reflections of object such as buildings, mountains 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). Figure 2.3.2 shows the effect of inter-symbol interference due to delay spread on the received signal. As the transmitted bit rate is increased the amount of intersymbol interference also increases. The effect starts to become very significant when the delay spread is greater then ~50% of the bit time.

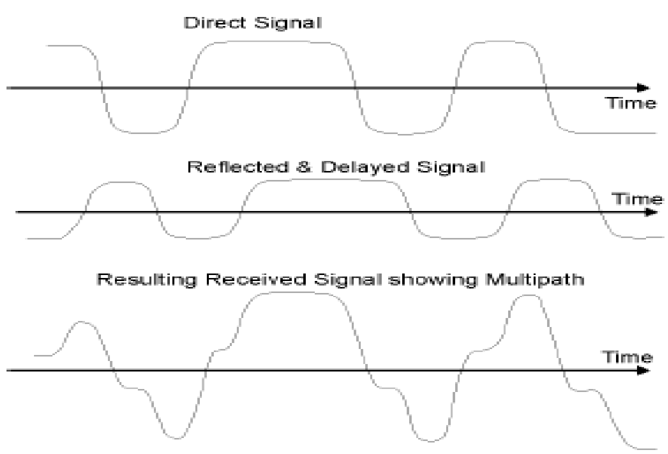


Fig 2.3.2 Multipath Delay Spread

Table 2.3 shows delay spread that can occur in various environments. The maximum delay spread in an outdoor environment is approximately 20µsec, thus significant intersymbol interference can occur at bit rates as low as 25kbps

Table 2.3: Delay Spread in various environments:

Environment or Cause	Delay Spread	Maximum Path l Length Difference
Indoor (room)	40nsec-200nsec	1 2m - 60m
Outdoor	1usec-20usec	300m-6km

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 intersymbol interference such as CDMA.

2.3.4. Doppler Shift:

When a wave source and a receiver are moving relative to one [1]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 then the source, and when they are approaching each other the frequency decreases. This is called the Doppler 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:

$$\Delta f \approx \pm f_0 v/c$$

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.

For example: Let f_o = 5.9 GHz, and v = 80km/hr = 22.22m/s (6.116 kms/hr aprox) then the Doppler shift will be 437 Hz. This shift of 437Hz in the carrier will generally not effect the transmission However, 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 al link between to cars moving in opposite directions, each one with a speed of 80 km/hr, the Doppler shift will be double.

2.4 Principles of OFDM:

In a conventional serial data system, the symbols are transmitted sequentially, one by one, with the frequency spectrum of each data symbol allowed to occupy the entire available bandwidth. A high rate data transmission supposes a very short symbol duration, conducing at a large spectrum of the modulation symbol. There are good chances that the frequency selective channel response affects in a very distinctive manner the different spectral components of the data symbol, hence introducing the ISI [23]. The same phenomenon, regarded in the time domain consists in smearing and spreading of information symbols such, the energy from one symbol interfering with the energy of the next ones, in such a way that the received signal has a high probability of being incorrectly interpreted. Intuitively, one can assume that the frequency selectivity of the channel can be mitigated if, instead of transmitting a single high rate data stream, we transmit the data

Simultaneously, on several narrow-band subchannels (with a different carrier corresponding to each subchannel), on which the frequency response of the channel looks "flat" (Reference Figure 2.4.1). Hence, for a given overall data rate, increasing the number of carriers reduces the data rate that each individual carrier must convey, therefore lengthening the symbol duration on each subcarrier. Slow data rate (and long symbol duration) on each subchannel merely means that the effects of ISI are severely reduced. This is in fact the basic idea that lies behind OFDM. Transmitting the data among a large

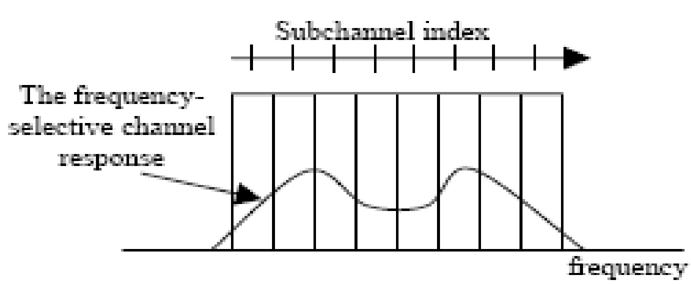


Figure 2.5.1 The frequency-selective channel response and the relatively flat response on each subchannel

number of closely spaced subcarriers accounts for the “frequency division multiplexing” part of the name. Unlike the classical frequency division multiplexing technique, OFDM will provide much higher bandwidth efficiency. This is due to the fact that in OFDM the spectra of individual subcarriers are allowed to overlap. In fact, the carriers are carefully chosen to be orthogonal one another. As it is well known, the orthogonal signals do not interfere, and they can be separated at the receiver by correlation techniques. The orthogonality of the subcarriers accounts for the first part of the OFDM name.

The block diagram of an OFDM system:

In figure 2.5.2, a classical OFDM transmission scheme using FFT (Fast Fourier Transform) is presented:

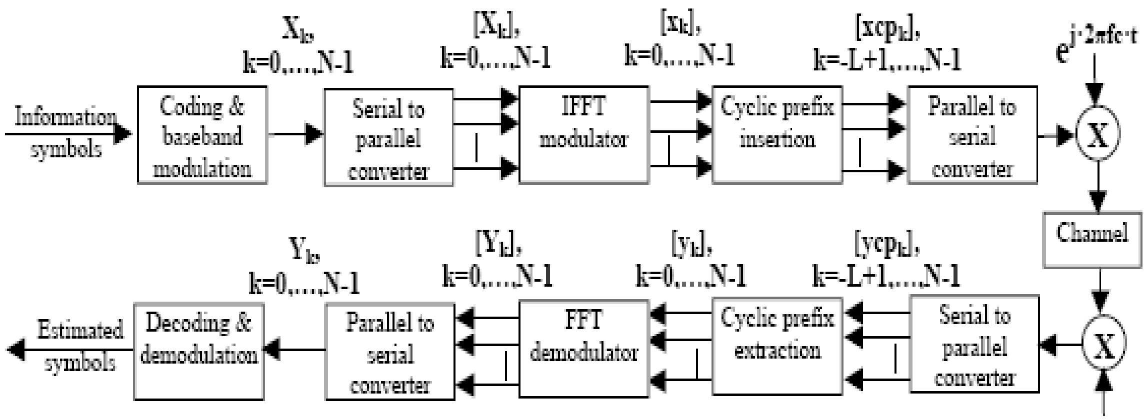


Fig.2: The block diagram of an OFDM System

The input data sequence is baseband modulated, using a digital modulation scheme. Various modulation schemes could be employed such as BPSK, QPSK (also with their differential form) and QAM with several different signal constellations. There are also forms of OFDM where a distinct modulation on each subchannel is performed (e.g. transmitting more bits using an adequate modulation method on the carriers that are more „confident”, like in ADSL systems). The modulation is performed on each parallel substream, that is on the symbols belonging to adjacent DFT frames. The data symbols are parallelized in N different substreams. Each substream will modulate a separate carrier through the IFFT modulation block, which is in fact the key element of an OFDM scheme, as we will see later. A cyclic prefix is inserted in order to eliminate the inter-symbol and inter-block interference (IBI). This cyclic prefix of length L is a circular extension of the IFFT-modulated symbol, obtained by copying the last L samples of the symbol in front of it. The data are back-serial converted, forming an OFDM symbol that will modulate a high-frequency carrier before its transmission through the channel. The radio channel is generally referred as a linear time-variant system. To the receiver, the inverse operations are performed: the data are down-converted to the baseband and the cyclic prefix is removed. The coherent FFT demodulator will ideally retrieve the exact form of transmitted symbols. The data are serial converted and the appropriated demodulation scheme will be used to estimate the transmitted symbols.

In this section, the key points of OFDM are presented: the principles of a multicarrier (parallel) transmission, the usage of FFT and the cyclic prefix.

2.4.1. The concept of multicarrier (parallel) transmission:

In a mobile radio environment, the signal is carried by a large number of paths with different strength and delays. Such multipath dispersion of the signal is commonly referred as, "channel-induced ISI" and yields the same kind of ISI distortion caused by an electronic filter [24]. In fact, the multipath dispersion leads to an upper limitation of the transmission rate in order to avoid the frequency selectivity of the channel or the need of a complex adaptive equalization in the receiver. In order to mitigate the time-dispersive nature of the channel, the finding of the multicarrier technique was to replace a single-carrier serial transmission at a high data rate with a number of slower parallel data streams. Each parallel stream will be then used to sequentially modulate a different carrier. By creating N parallel substreams, we will be able to decrease the bandwidth of the modulation symbol by the factor of N, or, in other words, the duration of a modulation symbol is increased by the same factor. The summation of all of the individual subchannel data rates will result in total desired symbol rate, with the drastic reduction of the ISI distortion. The price to pay is of course very important, since the multicarrier transmission seems to act as a frequency multiplexation, which will generate problems in terms of bandwidth efficiency usage. The things go however better than seemed, because in OFDM the carriers are orthogonal to each-other and they are separated by a frequency interval of $\Delta f = 1/T$. The frequency spectrum of the adjacent subchannels will overlap one another, but the carrier's orthogonality will eliminate in principle the interchannel interference that we feared of.

2.4.2 The Discrete Fourier Transform (DFT):

Though Multi carrier technique was introduced in the 1950 [1], the main reason that hindered the OFDM expansion for a very long time was practical. As it seemed difficult to generate such a signal, and even harder to receive and appropriately demodulate such a signal. Also this technique required a very large array of sinusoidal generators and also a large array of coherent demodulators to make it work. Therefore, the hardware solution cant be practical. As a consequence of the explosive development of digital signal processors (DSP), which can be used for generating and demodulating an OFDM signal, the magic idea was to use Fast Fourier Transform (FFT), a modern DSP technique. FFT merely represents a rapid mathematical method for computer applications of Discrete Fourier Transform (DFT). The ability to generate and to demodulate the signal using a software implementation of FFT algorithm is the key of OFDM current popularity. In fact, the signal is generated using the Inverse Fast Fourier Transform (IFFT), the fast implementation of Inverse Discrete Fourier Transform (IDFT). There is a mysterious connection between this transform and the concept of multicarrier modulation. According to its mathematical distribution, IDFT summarizes all sine and cosine waves of amplitudes stored in X[k] array, forming a time domain signal:

$$x[n] = \sum_{k=0}^{N-1} X[k].\, e^{j.k\pi \frac{2}{N} n} = \sum_{k=0}^{N-1} X[k].\,(\cos (k\pi \frac{2}{N} n) + j.\sin(k\pi \frac{2}{N} n)),\ n=0,1,\ldots,N-1 \quad (2)$$

From (2), we can simply observe that IDFT takes a series of complex exponential carriers, modulate each of them with a different symbol from the information array X[k], and multiplexes all this to generate N samples of a time domain signal (Figure 2.5.3). And what is really important, the complex exponential carriers are orthogonal to each other, as we know from the Fourier decomposition. These carriers are frequency spaced with $\Delta\Omega = 2\pi/N$. If we consider that the N data symbols X[k] come from sampling an analog information with a frequency of f_s, an easy to make discrete to analog frequency conversion indicates a $\Delta f = 1/T$ spacing between the subcarriers of the transmitted signal. The schema presented in figure 6, relies on a classical signal synthesis algorithm. The N samples of the time domain signal are synthesized from sinusoids and cosinusoids of frequencies $k \cdot 2\pi / N$. The „weight" with which each complex exponential contributes to the time domain signal. waveform is given by the modulation symbol X[k]. Therefore, the information X[k] to be transmitted could be regarded as being defined in the frequency domain. In its most simplest form, when X[k] stores a binary information („0" and „1"), each symbol to the IDFT entry will simply indicate the presence (an „1") or the absence (a „0") of a certain carrier in the composition of the time domain signal.

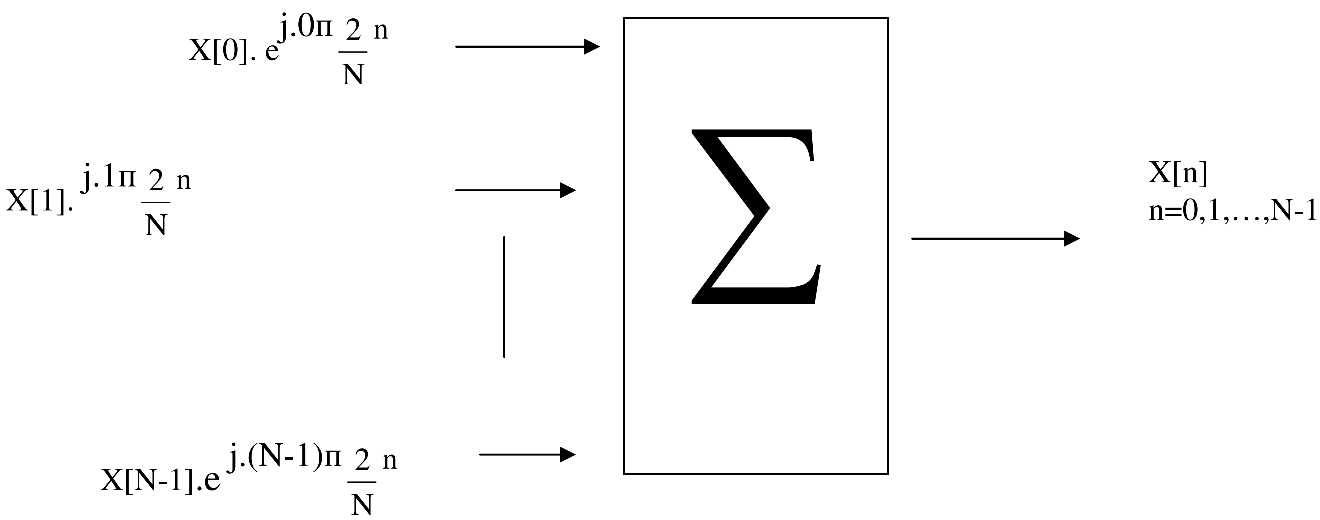


Figure: 2.4.3 Multi Carrier Modulation using IDFT

At the receiver, the inverse process is realized, the time domain signal constitutes the input to a DFT "signal analyser", implemented of course using the FFT algorithm. The FFT demodulator takes the N time domain transmitted samples and determines the amplitudes and phases of sine and cosine waves forming the received signal, according to the equation below:

$$X[K] = \sum_{n=0}^{N-1} x[n]\, e^{j.k\pi \frac{2}{N} n} = \sum_{n=0}^{N-1} x[n]\ \left(\cos\left(k\pi \frac{2}{N} n\right) + j.\sin\left(k\pi \frac{2}{N} n\right)\right),\ k=0,1,..N-1 \ldots..(3)$$

2.4.3 The cyclic prefix:

Since OFDM transmits data in blocks (usually a block is referred to as an „OFDM symbol") any type of a non-ideal transmission channel (such as a multipath channel, in mobile communications system, or a classical dispersive channel as in wired transmissions) will „spread" the OFDM symbol, causing the blocks of signal to interfere one another. This type of interference is called Inter-Block Interference. IBI will eventually lead to ISI, since two adjacent blocks will overlap, causing the distortion of the symbol affected by overlapping (Figure 2.4.4).

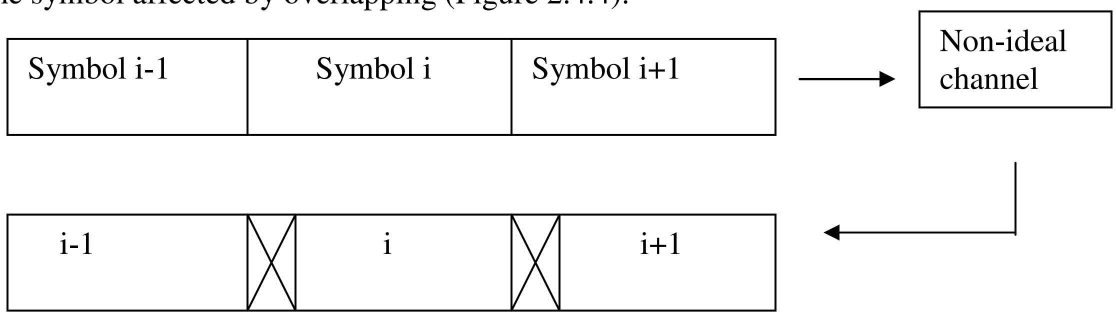


Figure 2.4.4 The transmitted symbols frame structure and the received symbols are Overlapping during the "interference interval"

In order to combat this interference, one of the possible approaches was to introduce „a silence period" between the transmitted frames. Known as "zero prefix", this silence period consists in a number of zero samples, added to the front of each symbol. The residual effect of the previous transmitted frame will affect only the „zero padding" portion (if the channel is considered to be linear). These altered samples are discarded to the receiver, and useful samples, unaffected by the IBI are used in order to demodulate the signal. However, the zero-padding doesn't seem the ideal solution, because the zero prefix will destroy the periodicity of the carriers. The demodulation process that uses FFT will be facilitated by keeping this periodicity, as we will see next.

Instead of this "quiet period" we could use a cyclic prefix (CP) at the beginning of each symbol. The cyclic prefix consists of the last L samples of the OFDM symbol that are copied in front of the data block [18]. If CP duration spans more than the channel impulse response or than the multipath delay, the residual contribution from the previous block is entirely absorbed by the cyclic prefix samples, that are thrown up to the receiver [18, 25, 26]. The same thing happened if a zero prefix is used, but the CP facilitates the receiver carrier synchronization, since some signals instead of a long silence period are transmitted. Furthermore, using a circular extension maintains the carrier's periodicity, which is important in order to simplify the proper reconstruction of the signal using DFT. The beneficent effect of the CP is illustrated in the figure 2.5.5. The transmitted OFDM symbols are two cosinusoides, the second being phase shifted with 180^o. If we consider a radio channel, multipath attenuated replica of the first symbol will arrive to the receiver with a certain delay. In figure 2.5.5 a) the delayed replica of the symbol i-1 will affect the next received symbol for a duration marked with the "interference" label. If a cyclic extension of duration Tg=T/4 is inserted in front of the useful data, the delayed replica of the first symbol, will affect only the CP portion of the second symbol, that is actually discarded to the receiver. If the guard

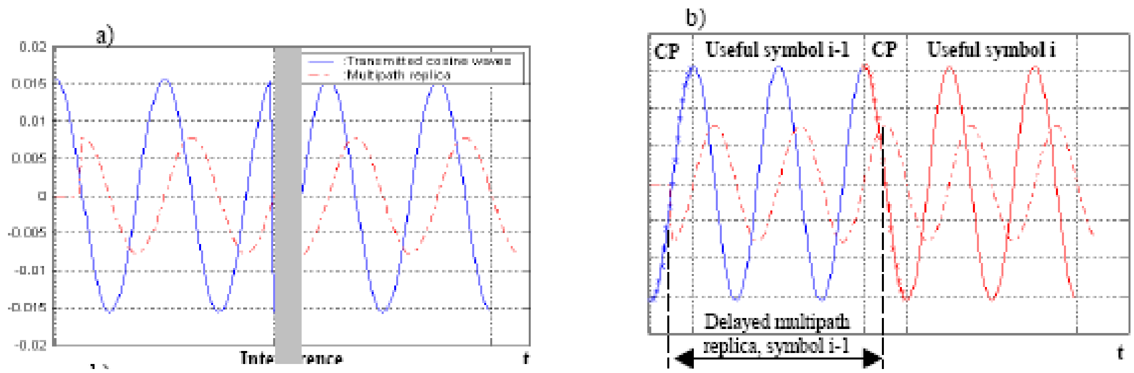


Fig: 2.5.5 IBI-two adjacent OFDM symbols interfere (a), the cyclic prefix eliminates this interference (b)

2.5 Conclusion:

This chapter provides a brief introduction to different aspects dealt in this thesis. The chapter discusses OFDM communication system in general and the propagation channel characteristics of a mobile radio channel. Principles of OFDM are dealt in detail. These concepts will be used in subsequent chapters of this thesis.

Chapter 3

IMPLEMENTATION OF OFDM

3.1 INTRODUCTION:

This chapter discusses the implementation and simulation of OFDM system in Matlab. In order to know the performance of OFDM in AWGN and Frequency selective channels, simulations are performed. We can study the performance of OFDM system under different noise conditions and different environments. BER plots are plotted as the SNR is varied. The chapter organizes as follows Section 3.2 discusses the OFDM model used for simulation. Section 3.3 presents some simulation results with discussion

.

3.1 OFDM Model Used:

An OFDM system was modeled using Matlab (version 6.5) to allow various parameters of the system to be varied and tested. The aim of doing the simulations was to measure the performance of OFDM under different channel conditions, and to allow for different OFDM configurations to be tested. Four main criteria were used to assess the performance of the OFDM system. They are tolerance to multipath delay spread, peak power clipping, channel noise and time synchronization errors. The OFDM system was modeled using Matlab and is shown in Figure 3.1. A brief description of the model is provided below.

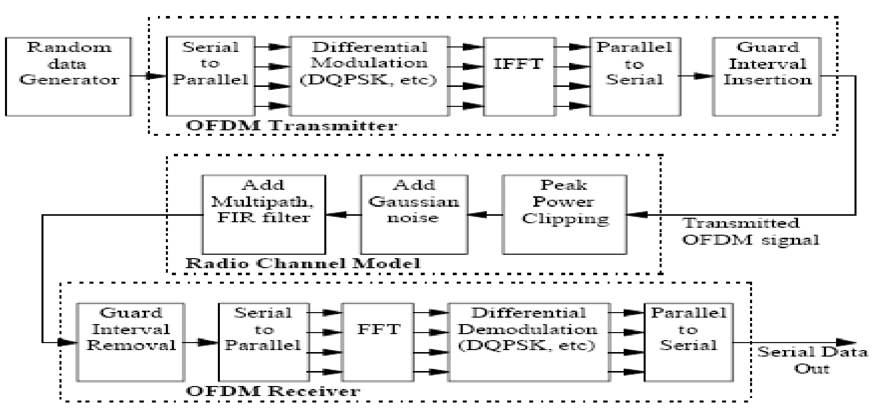


Figure: 3.1 OFDM Model used for Simulations

3.1.1 Serial to Parallel Conversion:

The input serial data stream is formatted into the word size required for transmission, e.g. 2 bits/word for QPSK, and shifted into a parallel format. The data is then transmitted in parallel by assigning each data word to one carrier in the transmission.

3.1.2 Modulation of Data:

The data to be transmitted on each carrier is then differential encoded with previous symbols, then mapped into a Phase Shift Keying (PSK) format. Since differential encoding requires an initial phase reference an extra symbol is added at the start for this purpose. The data on each symbol is then mapped to a phase angle based on the modulation method. For example, for QPSK the phase angles used are 0, 90, 180, and 270 degrees. The use of phase shift keying produces a constant amplitude signal and was chosen for its simplicity and to reduce problems with amplitude fluctuations due to fading.

3.1.3 Inverse Fourier Transform:

After the required spectrum is worked out, an inverse Fourier transform is used to find the corresponding time waveform. The guard period is then added to the start of each symbol.

3.1.4 Guard Period:

The guard period used was made up of two sections. Half of the guard period time is a zero amplitude transmission. The other half of the guard period is a cyclic extension of the symbol to be transmitted. This was to allow for symbol timing to be easily recovered by envelope detection. However it was found that it was not required in any of the simulations as the timing could be accurately determined position of the samples. After the guard has been added, the symbols are then converted back to a serial time waveform. This is then the base band signal for the OFDM transmission.

3.1.5 Channel:

A channel model is then applied to the transmitted signal. The model allows for the signal to noise ratio, multipath, and peak power clipping to be controlled. The signal to noise ratio is set by adding a known amount of white noise to the transmitted signal. Multipath delay spread then added by simulating the delay spread using an FIR filter. The length of the FIR filter represents the maximum delay spread, while the coefficient amplitude represents the reflected signal magnitude.

3.1.6 Receiver:

The receiver basically does the reverse operation to the transmitter. The guard period is removed. The FFT of each symbol is then taken to find the original transmitted spectrum. The phase angle of each transmission carrier is then evaluated and converted back to the data word by demodulating the received phase. The data words are then combined back to the same word size as the original data.

3.1.7 OFDM simulation parameters:

Table 3.1 shows the configuration used for most of the simulations performed on the OFDM signal. An 800-carrier system was used, as it would allow for up to 100 users if each were allocated 8 carriers. The aim was that each user has multiple carriers so that if several carriers are lost due to frequency selective fading that the remaining carriers till allow the lost data to be recovered using forward error correction. For this reason any less then 8 carriers per user would make this method unusable. Thus 400 carriers or less was considered too small. However more carriers were not used due to the sensitivity of OFDM to frequency stability errors. The greater the number of carriers a system uses, the greater it required frequency stability. For most of the simulations the signals generated were not scaled to any particular sample rate, thus can be considered to be frequency normalized. Three carrier modulation methods were tested to compare their performances. This was to show a trade off between system capacity and system robustness. DBPSK gives 1 b/Hz spectral efficiency and is the most durable method, however system capacity can be increased using DQPSK (2 b/Hz) and D16PSK (4 b/Hz) but at the cost of a higher BER. The modulation method used is shown as BPSK, QPSK, and 16PSK on all of the simulation plots, because the differential encoding was considered to be an integral part of any OFDM transmission.

Table: 3.1 OFDM system parameters used for simulations

Parameter	Value
Carrier Modulation used	BPSK,QPSK,16PSK
FFT size	2048
Number of carrier used	800
Guard Time	512 samples (25%)
Guard Period Type	Half zero signal, half a cyclic extension of the symbol

3.3 Simualtion Results

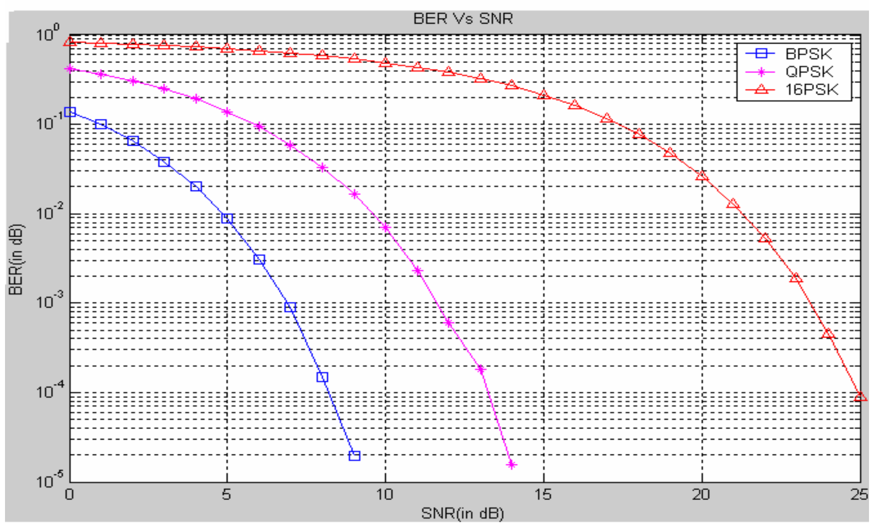


Figure 3.2 BER versus SNR for OFDM using BSPK, QPSK and 16PSK

It was found that the SNR performance of OFDM is similar to a standard single carrier digital transmission. This is to be expected, as the transmitted signal is similar to a standard Frequency Division Multiplexing (FDM) system. Figure 3.2 shows the results from the simulations. The results show that using QPSK the transmission can tolerate a SNR of >10-12 dB. The bit error rate BER gets rapidly worse as the SNR drops below 6 dB. However, using BPSK allows the BER to be improved in a noisy channel, at the expense of transmission data capacity. Using BPSK the OFDM transmission can tolerate a SNR of >6-8 dB. In a low noise link, using 16PSK can increase the capacity. If the SNR is >25 dB 16PSK can be used, doubling the data capacity compared with QPSK.

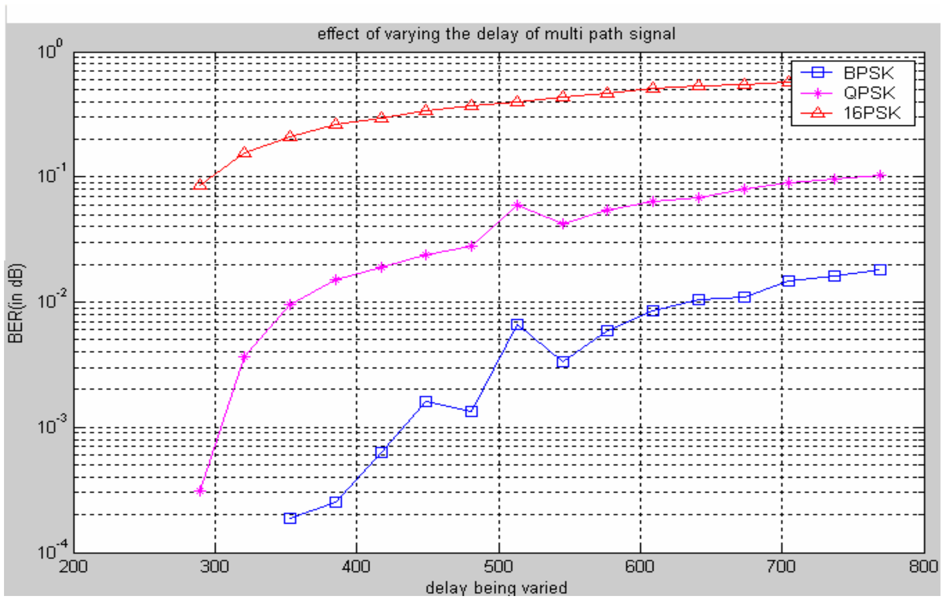


Figure 3.3 Delay Spread tolerance of ODFM

For this simulation the OFDM signal was tested with a multipath signal containing a single reflected echo. The reflected signal was made 3 dB weaker then the direct signal as weaker reflections then this did not cause measurable errors, especially for BPSK. Figure 3.3 shows the simulation results.

It can be seen from Figure 3.3 that the BER is very low for a delay spread of less then approximately 256 samples. In a practical system (i.e. one with a 1.25 MHz bandwidth) this delay spread would correspond to ~80 msec. This delay spread would be for a reflection with 24 km extra path length. It is very unlikely that any reflection, which has traveled an extra 24 km, would only be attenuated by 3 dB as used in the simulation, thus these results show extreme multipath conditions. The guard period used for the simulations consisted of 256 samples of zero amplitude, and 256 samples of a cyclic extension of the symbol. The results show that the tolerable delay spread matches the time of the cyclic extension of the guard period. It was verified that the tolerance is due to the cyclic extension not the zeroed period with other simulations. These test however are not shown to conserve space.

For a delay spread that is longer than the effective guard period, the BER rises rapidly due to the inter-symbol interference. The maximum BER that will occur is when the delay spread is very long (greater then the symbol time) as this will result in strong inter-symbol interference. In a practical system the length of the guard period can be chosen depending on the required multipath delay spread immunity required.

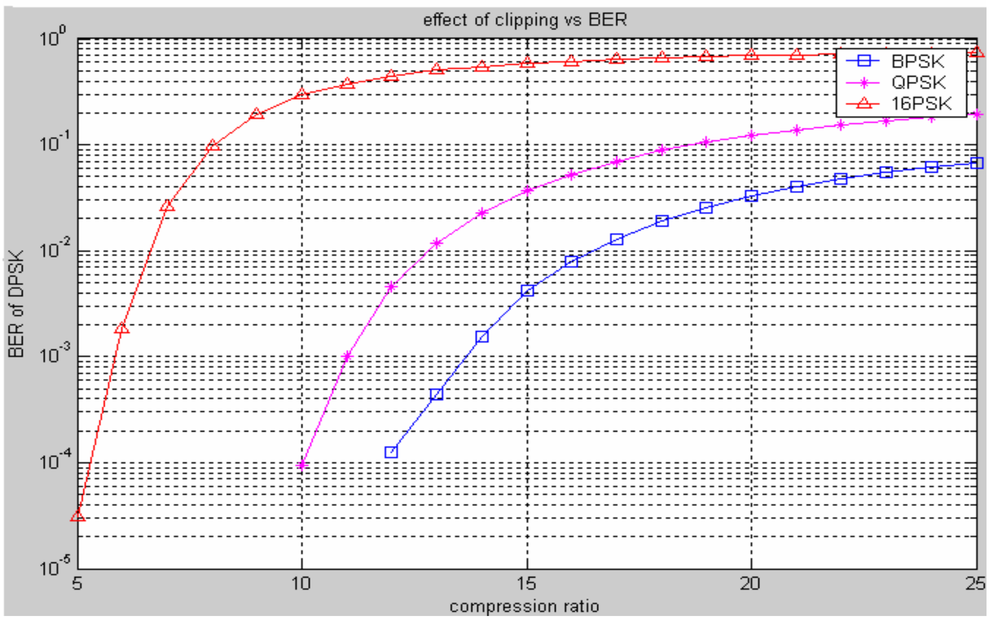


Figure 3.4 Effect of peak power clipping for OFDM

It was found that the transmitted OFDM signal could be heavily clipped with little effect on the received BER. In fact, the signal could the clipped by up to 9 dB without a significant increase in the BER. This means that the signal is highly resistant to clipping distortions caused by the power amplifier used in transmitting the signal. It also means that the signal can be purposely clipped by up to 6 dB so that the peak to RMS ratio can be reduced allowing an increased transmitted power.

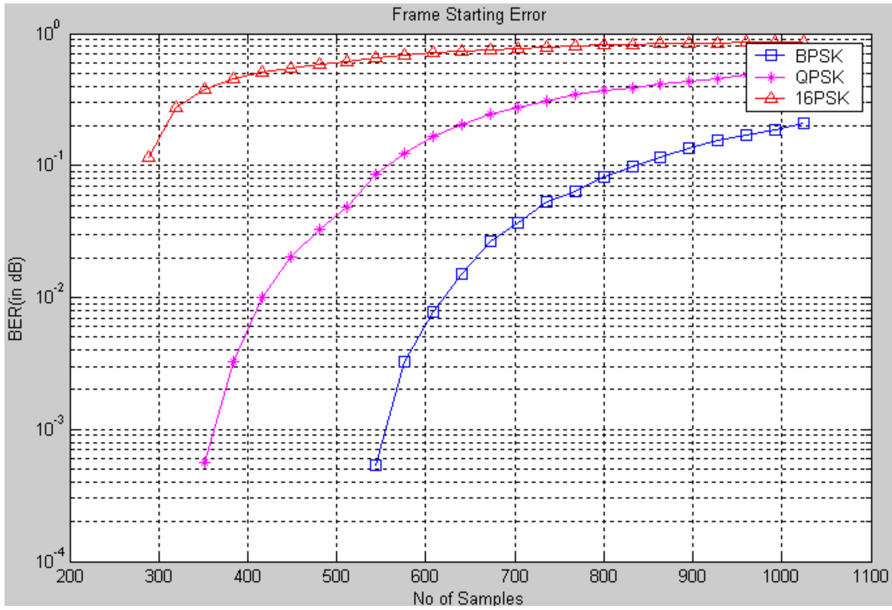


Figure 3.5 Effect of frame synchronization error on the received OFDM signal.

Figure 3.5 shows the effect of start time error on the received BER. This shows that the starting time can have an error of up to 256 samples before there is any effect of the BER. This length matches the cyclic extension period of the guard interval, and is due to the guard period maintaining the orthogonality of the signal.

In any practical system, the timing error made be either early or late, thus any receiver would aim for the middle of the expected starting time to allow for an error of ±128 samples. In addition, if the signal is subject to any multipath delay spread, this will reduce the effective stable time of the guard period, thus reducing the starting time error tolerance.

Simulation with 4-QAM

The OFDM system was simulated with the following parameters

Tu=224e-6; %useful OFDM symbol period

T=Tu/2048; %baseband elementary period

G=0; %choice of 1/4, 1/8, 1/15 and 1/32

delta=G*Tu; %guard band duration

Ts=Tu+delta; %total OFDM symbol period

Kmax=1705; %number of subcarriers

Kmin=0;

FS=4096; %IFFT/FFT length

q=10; %carrier period to elementary period ratio

fc=q*1/T; %carrier frequency

Rs=4*fc; %simulation period

t=0:1/Rs:Tu;

tt=0:T/2:Tu;

Repeat = 68; % one OFDM frame (68 OFDM symbols) is sent, symbol by symbol

SNR_dB = 0:2:16

The results obtained are shown in figures 3.6, 3.7 and 3.8

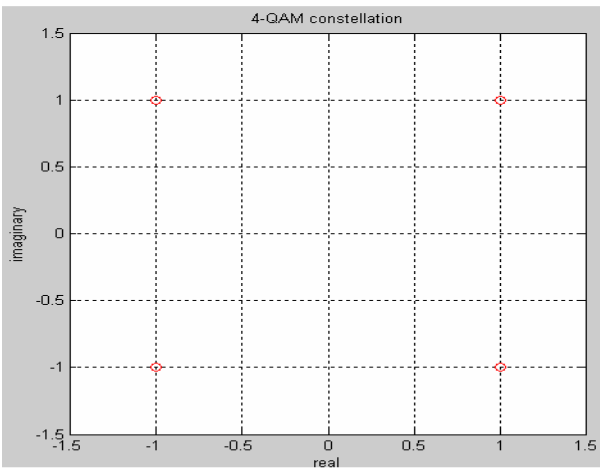


Figure 3.6 4 QAM constellation

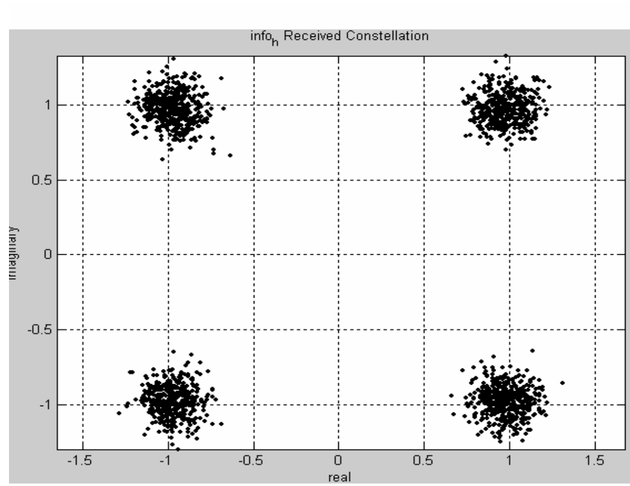


figure 3.7 received 4 QAM constellation

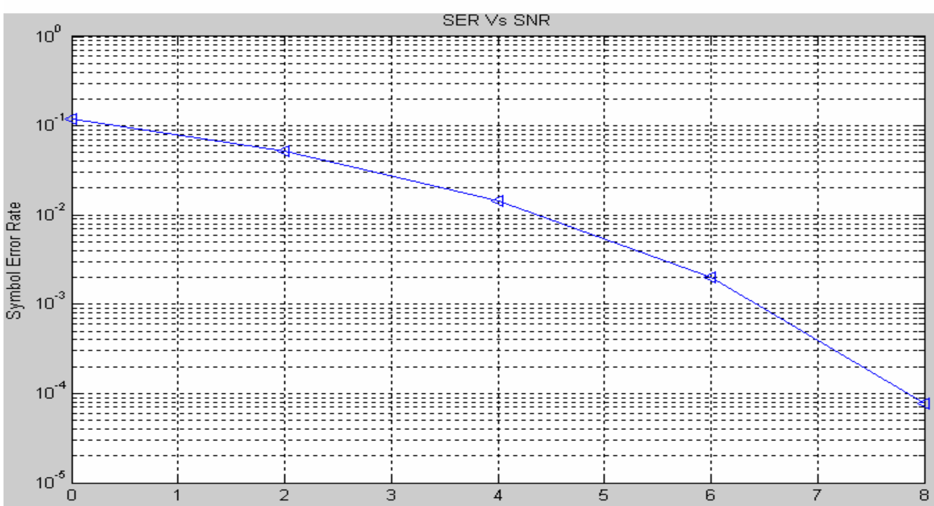


Figure 3.7 symbol error rate versus SNR (in dB)

Simulation with 64 QAM

Here the OFDM system was simulated with the following parameters.

N = 64; % number of carriers - 48 data, 4 pilot, 12 unused

L = 8; % oversampling factor

Symbols = 100; % number of OFDM symbols to simulate

Rate=1/2/4/8-------------BPSK/QPSK/16QAM/64QAM

The figures 3.8, 3.9, 3.10 represent the OFDM constellation for 64 carriers, OFDM time domain signal and approximate OFDM spectrum

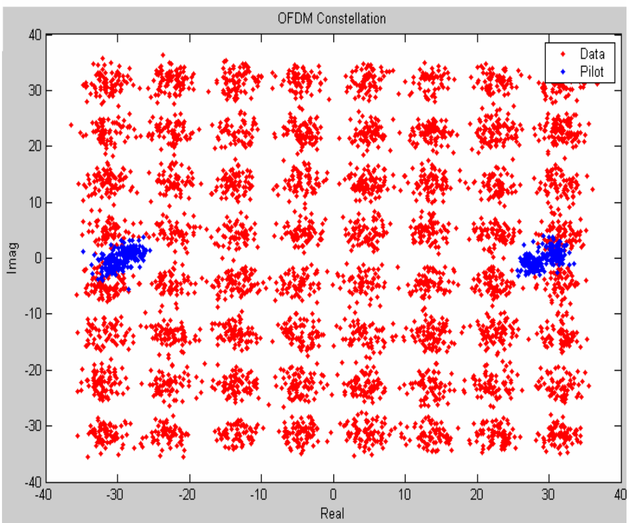


Figure 3.8 OFDM constellation

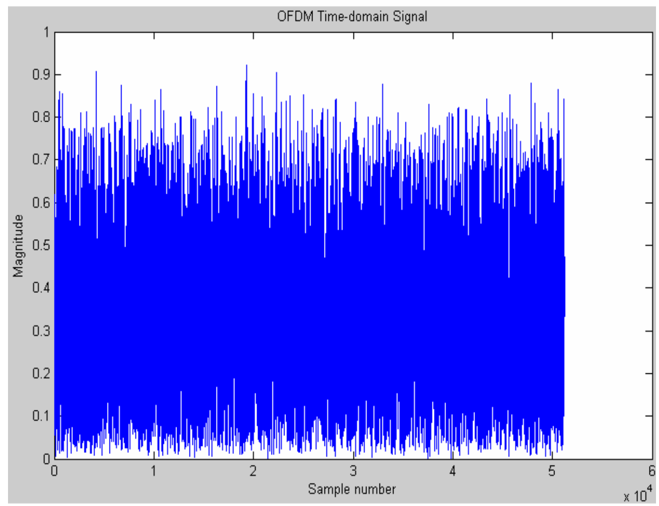


Figure 3.9 OFDM time domain signal

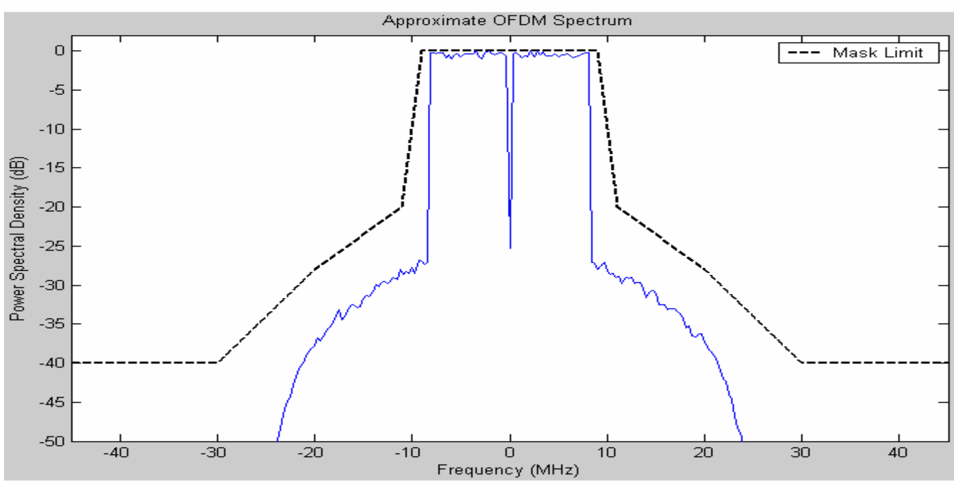


Figure 3.10 approximate OFDM spectrum

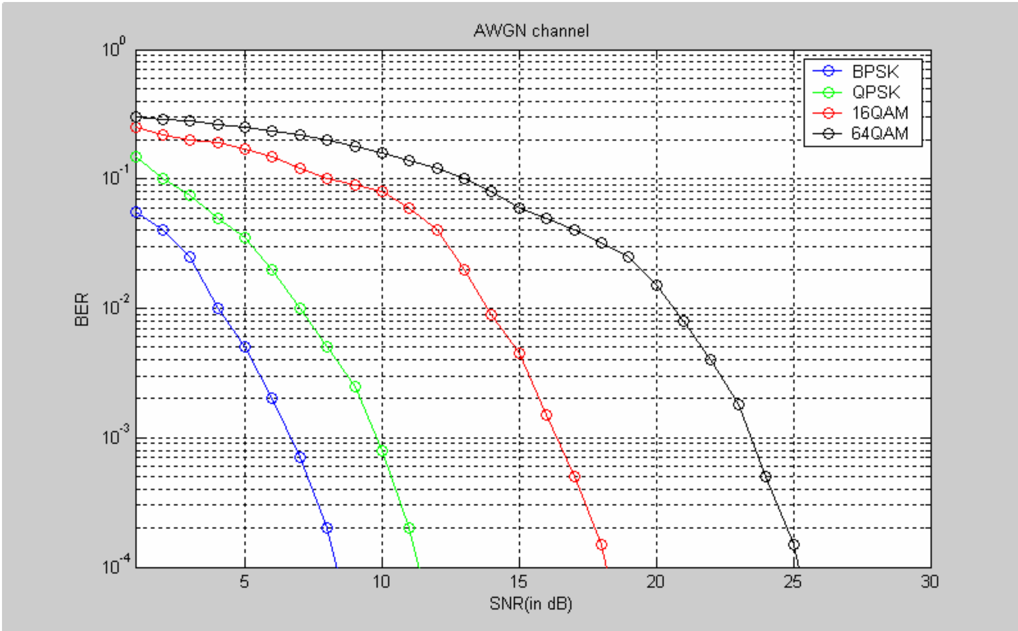


Figure 3.11 BPSK BER performance of OFDM in an AWGN channel

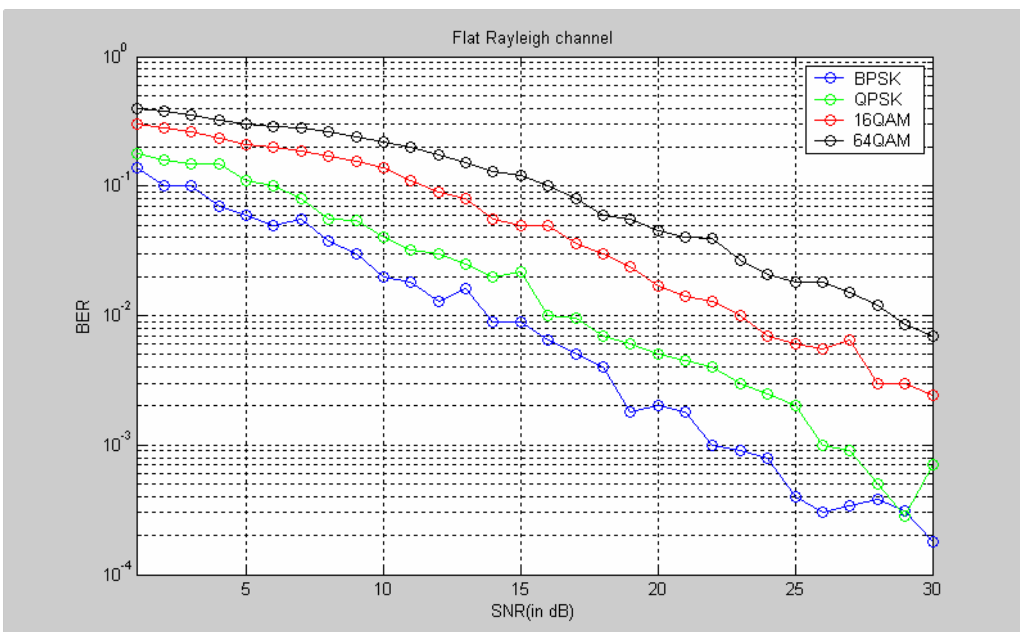


Figure 3.12 OFDM performance in a fast flat Rayleigh faded channel

3.4 Conclusion

In this chapter the OFDM Modem was implemented in matlab and the performance was evaluated for differet simulation parameters. Here three carrier modulation methods were tested to compare their performances. This was to show a trade off between system capacity and system robustness. The following points can be drawn from the study presented in this chapter.

- BPSK gives 1 b/Hz spectral efficiency and is the most durable method but system capacity is less.
- QPSK can be used to increase system capacity up to 2b/Hz but BER is more compared to BPSK
- 16 PSK can increase the system capacity further up to 4b/Hz but here also BER is compared to BPSK and QPSK.

Chapter 4

CHANNEL CODING IN OFDM

4.1 Introduction:

OFDM has recently received increased attention due to its capability of supporting high-data-rate communication in frequency selective fading environments which cause inter-symbol interference (ISI). Instead of using a complicated equalizer as in the conventional single carrier systems, the ISI in OFDM can be eliminated by adding a guard interval which significantly simplifies the receiver structure. However, in order to take advantage of the diversity provided by the multi-path fading, appropriate frequency interleaving and coding is necessary. Therefore, coding becomes an inseparable part in most OFDM applications and a considerable amount of research has focused on optimum encoder, decoder, and interleaver design for information transmission via OFDM over fading environments.

This chapter discusses the performance of the OFDM with different coding techniques and a combination of coding techniques. Following the introduction this chapter is organized as under Section 4.2 describes the different coding techniques Section 4.3 describes about the burst error correction techniques. Section 4.4 provides concluding remarks.

4.2 Different CODING Techniques:

FEC is widely used in digital telecommunications systems to make it easier to encode and decode signals. Coding (FEC techniques) [27] introduces redundancy in data and adding delay. Simplest error-detecting technique is parity check bit coding. i.e. adding an extra binary digit at the end of each word. But it will be effective only if the probability of error is very low.

Coding allows us to reduce the information bit error rate while maintaining a fixed transmission rate. In principle it allows us to reach Shanon's limit with ingenious coding and enough unlimited complexity. The more commonly employed codes are:

1. Block Codes
    1. Single Parity-Check Bit Code
    2. Repeated Codes
    3. Hadamard Code
    4. Hamming Code
    5. Extended Codes
    6. Cyclic Codes
    7. The Golay Code
    8. BCH Codes

Block codes operate on a block of bits. Using a preset algorithm, we take a group of bits and add a coded part to make a larger block. This block is checked at the receiver. The receiver then makes a decision about the validity of the received sequence.

Block Codes

.

Block codes [27]are referred to as (n, k) codes. A block of k information bits are coded to become a block of n bits.

n=k + r where r is the number of parity bits. k is the number of information bits

Suppose a code word has the form

$a_1a_2a_3\ldots\ldots\ldots\ldots a_kc_1c_2\ldots\ldots\ldots c_r$

Generation of a block code starts with a selection of the number r of the parity bits to be added and thereafter with the specification of an H matrix.

$$H = \left| \begin{array}{cccc|cccc} h_{11} & h_{12} & \ldots & h_{1k} & 1 & 0 & \ldots & 0 \\ h_{21} & h_{22} & \ldots & h_{2k} & 0 & 1 & \ldots & 0 \\ \ldots & & & & \ldots & & & \\ h_{r1} & h_{r2} & \ldots & h_{rk} & 0 & 0 & \ldots & 1 \end{array} \right|$$

$$\underbrace{\qquad r*k \qquad}_{\text{h sub matrix}} \qquad \underbrace{\qquad r*r \qquad}_{\text{Identity matrix, I}}$$

4.2.1 Single Parity-Check Bit Code:

It is an example of an block code [27]. Here parity check is selected to satisfy the equation

a1+a2+…….ak+c1=0

$h_{11} = h_{12} = \ldots\ldots. = h_{1k} = 1$

All other equations of the set are identically zero. Here r=1 and n=k+1.

This is a single error correcting code and will not correct errors.

4.2.2 Repeated Codes:

Here a binary 0 is encoded as a sequence of (2t+1) zeros, and a binary 1 as a similar number of 1's. Thus, k=1, r=2t, and n=2t+1.

For t=1, r=2 and n=3

4.2.3 Hadamard Code:

Code word in a Hadamard code [27] are the rows of a Hadamard matrix. The Hadamard matrix is a square (n*n) matrix in which $n=2^k$, k is number of bits in the uncoded word. One code word consists of all zeros and all the other codewords have n/2, 0's and n/2, 1's. Further each codeword differs from another codeword in n/2 places and for this reason the codewords are orthogonal to one another.

The Hadamard matrix which provides two codewords is

$$M2=\begin{bmatrix} 0 & 0 \\ 0 & 1 \end{bmatrix}$$

And the codewords are 00 and 01.

The matrix M4 which provides 4 code words is

$$M4= \begin{bmatrix} M2 & M2 \\ M2 & M2^* \end{bmatrix} = \begin{bmatrix} 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 1 \\ 0 & 0 & 1 & 1 \\ 0 & 1 & 1 & 0 \end{bmatrix}$$

The hamming distance of an orthogonal code is $d_{min}=\frac{n}{2}=2^k\text{-}1$

Thus the number of errors that can be corrected with a Hadamard code is $t=\frac{n}{4}-1$

4.2.4 Hamming Code:

Hamming codes [28] are most widely used linear block codes. A Hamming code is generally specified as (2n-1, 2n-n-1). The size of the block is equal to 2n-1. The number of information bits in the block is equal to 2n-n-1 and the number of overhead bits is equal to n. All Hamming codes are able to detect three errors and correct one. Common sizes of Hamming codes are (7, 4), (15, 11) and (31, 26). All of these have the same Hamming distance.Hamming code is a code in which dmin=3 so that t=1, i.e., a single error can be corrected. $n=2^r-1$; $k=2^r-1-r$;

Lets take a look the (7, 4) Hamming code. In this code, the length of the information sequence is 4 bits. So there are only 16 possible sequences. The table below lists all 16 of these sequences.

Table 4.1- 16 possible sequences from a 4 bit word

Sequence Number	i0	i1	i3	i4
1	0	0	0	1
2	0	0	1	0
3	0	0	1	1
4	0	1	0	0
5	0	1	0	1
6	0	1	1	0
7	0	1	1	1
8	1	0	0	0
9	1	0	0	1
10	1	0	1	0
11	1	0	1	1
12	1	1	0	0
13	1	1	0	0
14	1	1	0	1
15	1	1	1	1
16	0	0	0	0

.

H is called the parity matrix of the code. It has n rows and n-k columns. For a code of (7, 4), it has 7 rows and 3 columns. We can reorder the rows of this matrix any which way we want. The only thing to keep in mind is that we want the basis rows to be together either on top or at the bottom. Each reordering will give us a new Hamming code. Following are just two ways we can order the rows of H, each of these will result in a different code.

$$H = \begin{bmatrix} 1 & 1 & 1 \\ 0 & 1 & 1 \\ 1 & 0 & 1 \\ 1 & 1 & 0 \\ 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 1 \end{bmatrix} \quad \text{a different ordering of } H = \begin{bmatrix} 1 & 1 & 1 \\ 0 & 1 & 1 \\ 1 & 0 & 1 \\ 1 & 1 & 0 \\ 0 & 0 & 1 \\ 0 & 1 & 0 \\ 1 & 0 & 0 \end{bmatrix}$$

Now we take the upper part of this matrix (under the line) and create a new matrix called

$$G = \begin{bmatrix} 1 & 0 & 0 & 0 & 1 & 1 & 1 \\ 0 & 1 & 0 & 0 & 0 & 1 & 1 \\ 0 & 0 & 1 & 0 & 1 & 0 & 1 \\ 0 & 0 & 0 & 1 & 1 & 1 & 0 \end{bmatrix}$$

Take upper part of the H matrix (the one on the left) and add a unit matrix to the right of it, making a new matrix of size (4 x 7). This is our generator matrix, the one that tells us how the 16 info sequences will be mapped to the 16 valid codewords.

Rearrangement of the rows of H will lead to a different G matrix and hence will change the mapping. The essential point to this is that two block codes of the same size may create an entirely different mapping depending on how the rows are ordered, but each mapping has the same Hamming weight and has the same error correction/detection capability.Now all we have to do is multiply the information vector d with matrix G to get a codeword c. c = d. G

For information sequence [0, 1, 1, 0], we get the following transmitted codeword of length 7.

$$C = [0\ 1\ 1\ 0] \begin{bmatrix} 1 & 0 & 0 & 0 & 1 & 1 & 1 \\ 0 & 1 & 0 & 0 & 0 & 1 & 1 \\ 0 & 0 & 1 & 0 & 1 & 0 & 1 \\ 0 & 0 & 0 & 1 & 1 & 1 & 0 \end{bmatrix} = [0\ 1\ 1\ 0\ 1\ 1\ 0]$$

4.2.5 Extended Codes:

All codes can be extended [27], that is, starting with a parity check matrix H, a new, extended, matrix He(n+1, k) code.

He consists of the matrix with an added row consisting of all 1's and an added right-hand column consisting of all 0's except for the bottommost elements which remains a 1.

Here, $d_{emin}=d_{min}+1$;

4.2.6 Cyclic Codes:

They use generator polynomials to generate the code word [27].

4.2.7 Golay Code:

It is also an example of an cyclic code.

Golay code [27]is an (23,12) cyclic code whose generating function is

$$g(x)=x^{11}+x^{9}+x^{7}+x^{6}+x^{5}+x+1$$

Its extended form is (24,12) code. With $d_{min}=7$ and $d_{min}=8$ for extended.

4.2.8 BCH Codes:

BCH code [27] employs k information bits, r parity check bits and therefore number of bits in a codeword is n=k+r. Furthermore, the number of errors which can be corrected in an n-bit code word is $t=r/m$;

Where m is an integer related to n as $n=2^{m}-1$.

4.3 Burst Error Correction Techniques

The parity bits added in the block codes will correct a limited number of bit errors in each codeword. When the bit errors are closely clustered we say that the errors occur in bursts.

4.3.1 Block Interleaving

A primary technique which is effective in overcoming error bursts is interleaving. The principle of interleaving [27] is illustrated in Figure 4.1:

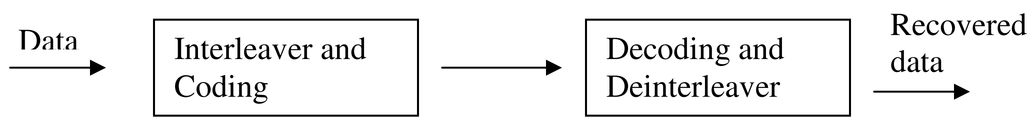
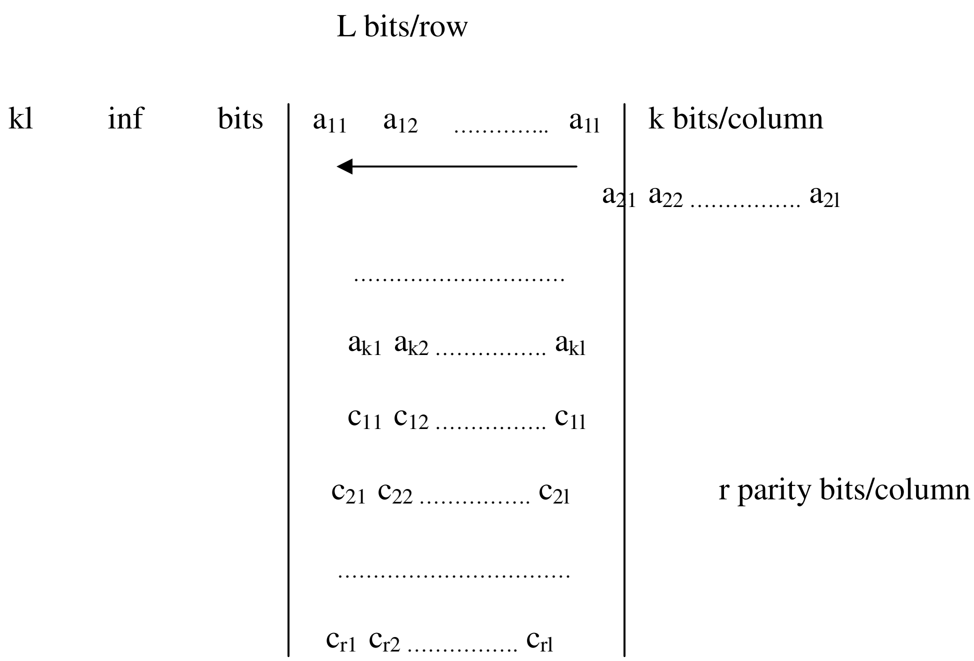


Figure 4.1 Interleaving block diagram

Before the data stream is applied to the channel, the data goes through a process of interleaving and error correction coding. At the receiving end, the data is decoded i.e., the data bits are evaluated in manner to take advantage of error correcting and detecting features which result from the coding and the process of interleaving is undone.



As in above Figure a group of kl bits is loaded into a shift register which is organized into k rows with l bits per row. The data stream is entered into the storage element at a_{11}. At each shift each bit moves one position to the right while the bit in the rightmost storage element moves to the leftmost stage of the next row. Thus for example as indicated by the arrow the content of a_{11} moves to a_{21}. When kl bits have been entered the register is full the first bit being in a_{kl} and the last bit in . At this point the data stream is diverted to a second similar shift register and a process of coding is apple to the data held stored in the first register. Here the information bits in a column (ex a_{11} a_{12} ………… a_{l1}) are viewed as the bits of an uncoded part to which parity bits are to be added. Thus the codeword a_{11} a_{12} ………… a_{l1} c_{11} c_{12} …………… c_{l1} is formed, thereby generating a codeword with k information bits and r parity bits.

When the coding is completed, the entire content of the k*l information register as well as the r*l parity bits are transmitted over the channel. Generally the bit by bit serial transmission is carried out row by row in the order

c_{rl}……. $c_{r1.}$ ……. c_{1l}……. c_{11} a_{kl} … .a_{k1} ………a_{12} a_{11}

The received data is again stored in the same order as in the transmitter and error correction decoding is performed. The parity bits are discarded and the data bits are shifted out of the register.

4.3.2 Reed-Solomon (RS) CODE:

The RS block code [27] is organized on the basis of groups of bits. Such groups of bits are referred to as symbols. We store sequences of m individual bits which appear serially in a bit stream and thereafter operate with the m-bit sequence rather than individual bits. These are called m-bit symbols. Even if an error occurs in a single bit of an symbol, the entire symbol is in error. RS code has k information symbols, r parity symbols and a total codeword length of n (=k+r) symbols. The number of symbols in the codeword is arranged to be $n=2^m -1$. The RS code is able to correct errors in t symbols where t=r/2.

4.3.3 Convolutional Coding:

Convolutional codes [29] are referred to as continuous codes as they operate on a certain number of bits continuously. Interleaving has mitigating properties for fading channels and works well in conjunction with these two types of coding.

Coding and decoding with Convolutional Codes

Convolutional codes are commonly specified by three parameters; (n,k,m).

n = number of output bits

k = number of input bits

m = number of memory registers

The quantity k/n called the code rate, is a measure of the efficiency of the code. Commonly k and n parameters range from 1 to 8, m from 2 to 10 and the code rate from 1/8 to 7/8 except for deep space applications where code rates as low as 1/100 or even longer have been employed.

Often the manufacturers of convolutional code chips specify the code by parameters ($n,k,$L), The quantity L is called the constraint length of the code and is defined by

Constraint Length, L = $k\ (m\text{-}1)$

The constraint length L represents the number of bits in the encoder memory that affect the generation of the n output bits. The constraint length L is also referred to by the capital letter K, which can be confusing with the lower case k, which represents the number of input bits. In some books K is defined as equal to product the of k and m. Often in commercial spec, the codes are specified by (r, K), where r = the code rate k/n and K is the constraint length. The constraint length K however is equal to L – 1, as defined in this paper. I will be referring to convolutional codes as (n,k,m) and not as (r,K).

Code parameters and the structure of the convolutional code

The convolutional code structure is easy to draw from its parameters. First draw m boxes representing the m memory registers. Then draw n modulo-2 adders to represent the n output bits. Now connect the memory registers to the adders using the generator polynomial as shown in the Fig. 4.2

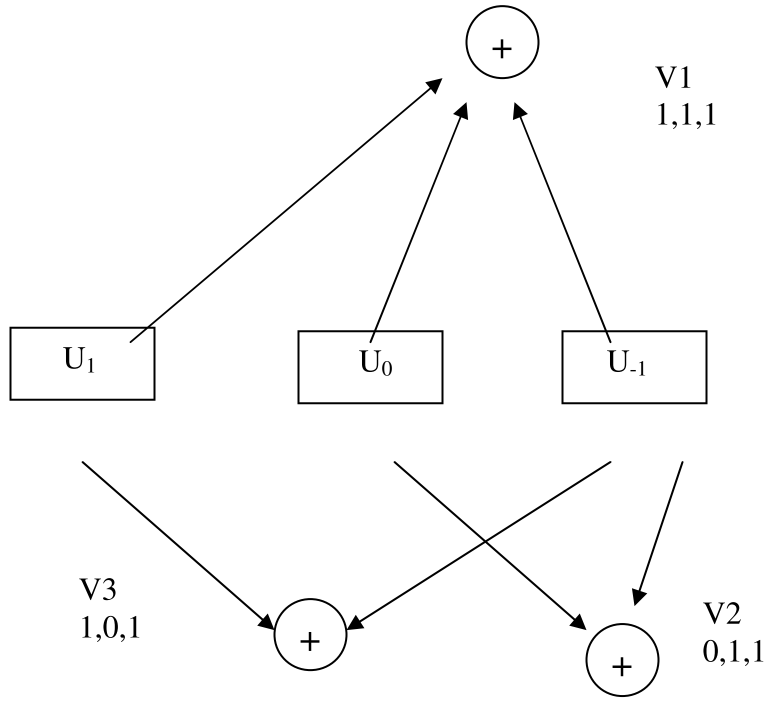


Fig: 4.2 convoultional coder

This is a rate 1/3 code. Each input bit is coded into 3 output bits. The constraint length of the code is 2. The 3 output bits are produced by the 3 modulo-2 adders by adding up certain bits in the memory registers. The selection of which bits are to be added to produce the output bit is called the generator polynomial (g) for that output bit. For example, the first output bit has a generator polynomial of (1,1,1). The output bit 2 has a generator polynomial of (0,1,1) and the third output bit has a polynomial of (1,0,1). The output bits just the sum of these bits.

$v_1 = \text{mod2}\,(u_1 + u_0 + u_{-1})$; $v_2 = \text{mod2}\,(u0 + u_{-1})$

$v_3 = \text{mod2}\,(u_1 + u_{-1})$

Punctured Codes:

For the special case of $k = 1$, the codes of rates ½, 1/3, ¼, 1/5, 1/7 are sometimes *called mother codes [29]*. We can combine these single bit input codes to produce punctured codes which give us code rates other than $1/n$.

By using two rate ½ codes together, and then just not transmitting one of the output bits we can convert this rate ½ implementation into a 2/3 rate code. 2 bits come and 3 go out. This concept is called puncturing. On the receive side, dummy bits that do not affect the decoding metric are inserted in the appropriate places before decoding

States of a code

For a (2,1,4) code with a constraint length of 3.

The number of combinations of bits in the shaded registers are called the states of the code and are defined by

Number of states = 2^L

Where L = the constraint length of the code and is equal to $k\,(m - 1)$.
his technique allows us to produce codes of many different rates using just one simple hardware. Although we can also directly construct a code of rate 2/3, the advantage of a punctured code is that the rates can be changed dynamically (through software) depending on the channel condition such as rain, etc. A fixed implementation, although easier, does not allow this flexibility.

TURBO CODES

Turbo codes [30] were first presented at the International Conference on Communications in 1993. Until then, it was widely believed that to achieve near Shannon's bound performance, one would need to implement a decoder with infinite complexity or close. Parallel concatenated codes, as they are also known, can be implemented by using either block codes (PCBC) or convolutional codes (PCCC). PCCC resulted from the combination of three ideas that were known to all in the coding community:

- The transforming of commonly used non-systematic convolutional codes into systematic convolutional codes.
- The utilization of soft input soft output decoding. Instead of using hard decisions, the decoder uses the probabilities of the received data to generate soft output which also contain information about the degree of certainty of the output bits.
- Encoders and decoders working on permuted versions of the same information. This is achieved by using an interleaver.

An iterative decoding algorithm centered around the last two concept would refine its output with each pass, thus resembling the turbo engine used in airplanes. Hence, the name Turbo was used to refer to the process.

Turbo Encoding :

Recursive Systematic Convolutional Codes (RSC)

Convolutional encoding results from passing the information to be encrypted through a linear shift register as shown in Figure 3.1 below. The encoder shown here is nonsystematic because no version of the uncoded input is part of the output. Convolutional encoder can be represented by their generator polynomials. For the encoder below, g(1) = [111] and g(2) = [101].

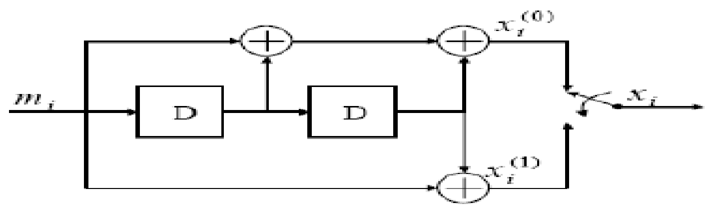
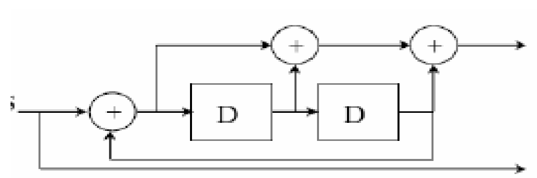


Fig: 4.3 constraint length k=2 convolutional encoder

Convolutional encoding is a continuous process where the output depends on the K previous inputs of the encoder. The linear shift register introduces a deterministic component to the randomly generated input. This component can be tracked through a trellis, which we will introduce shortly. For Turbo codes, the recursive systematic convolutional codes were chosen as they exhibit better performance at low signal to noise ratios (SNR).

RSC-1: input bits



Output bits

Fig: 4.4 Recursive systematic convolutional encoder

For decoding purposes, convolutional encoders can be seen as finite state machines (FSM) with the content of the shift register indicating the state of the machine, and the outputs being a function of the current state as well as the input to the encoder. The

code's behavior can also be described by using a trellis diagram. In a trellis diagram, all possible transitions between states are shown along side with the input and output associated with it. Transitions not drawn on the trellis do not represent valid codewords and are therefore classified as errors.

Encoding of Parallel Concatenated Convolutional Codes :

Turbo codes were presented by Berrou, Glavieux and Thitimajshima [11] in 1993. They are the result of the parallel concatenation of two or more RSC. For the scope of this thesis, we will only consider the case where two RSC are used. The information is encoded by the first recursive systematic encoder, interleaved and then encoded by the second RSC at the same time. The size of the interleaver determines the length of the codeword.

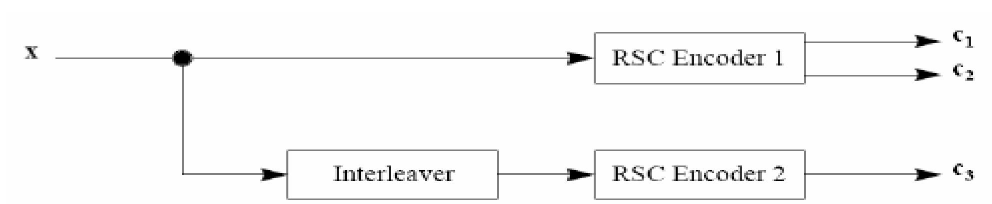


Figure: 4.5 Turbo Encoding Scheme

A block MxN interleaver can be used. In that case, the M bits would be fed into the interleaver column-wise and N bits would be read out row-wise. The interleaver would then alleviate burst errors by spreading them so that one error occurs every M bits and thus reduce the correlation between its input and output. The presence of the interleaver adds to a difficult trellis termination problem. The trellis of a conventional convolutional encoder can be terminated by appending a few zeros at the end of the input sequence. For the recursive variety of encoders, the termination bits depend on the on the state of the encoder as we are trying to force it back to the zero state. Therefore, the tails bits cannot be known until the encoder has completely encoded the data. Moreover, the additional bits used for trellis termination of RSC #1 will be interleaved and therefore useless in terminating RSC #2. They become data for the latter. One can see how difficult it becomes to successfully compute a sequence of tail bits that will terminate both trellis [13]. One solution is to only terminate the trellis of RSC #1 and leave the other open [12].

This alternative is perhaps the easiest to implement and we will chose it in our simulation. One can modify a turbo code (mother code) in order to achieve different code rates with the resulting code (punctured code). Puncturing patterns decide which parity bits are to be retained after puncturing. Commonly used patterns include selecting the x^{th} bit every 2*k parity bits, k > 0. For most rates, when commonly used patterns are applied to both parity sequences, turbo codes exhibit very good performance [30].

4.4 Simulation of an COFDM system:

OFDM simulation model used [31]

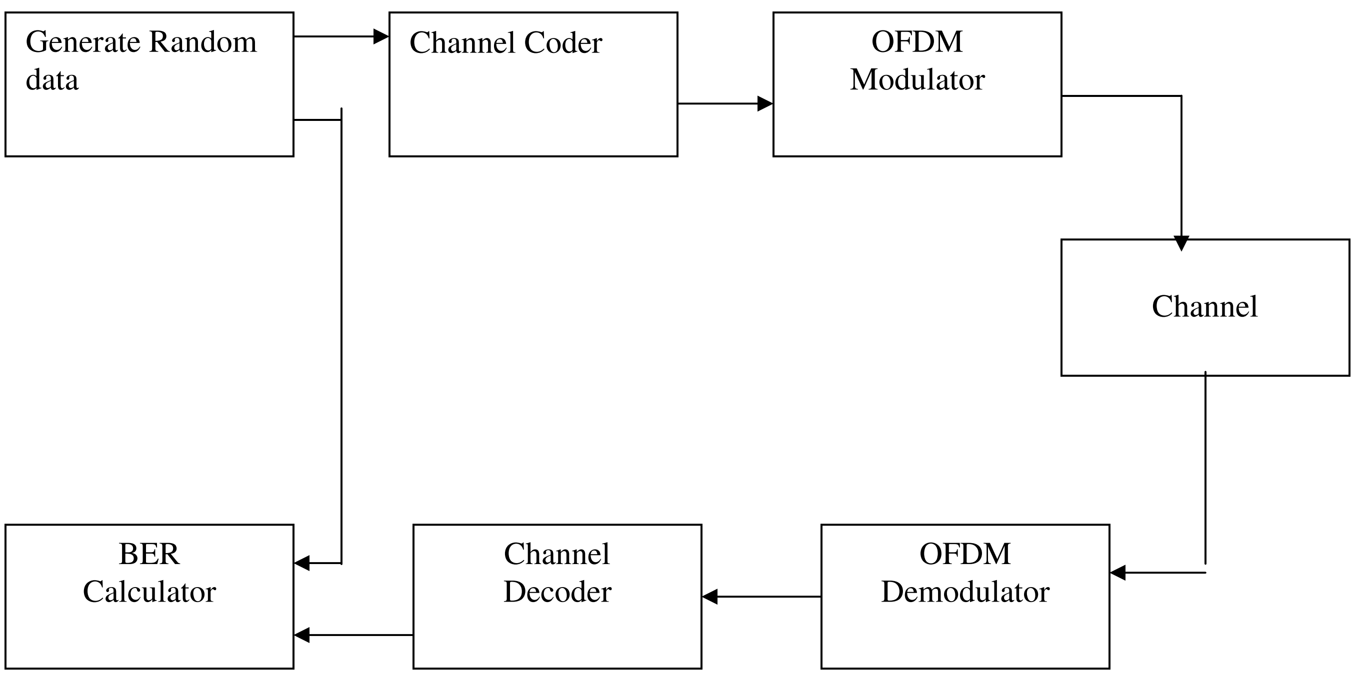


Figure: 4.6 OFDM Simulation Model used

Here in the COFDM simulation model (refer to figure 4.6) first random data is generated and is passed through a channel coder. From there it is passed to a OFDM modulator and then passed to a channel. Next the reverse process is performed at the receiver side.

After channel coding is undone BER is calculated to estimate the performance of COFDM system.

Here, in channel coder different coding techniques are used to estimate the performance of COFDM system. The simulation environment is shown in the following table (Table 4.1).

Table 4.1 OFDM simulation parameters used

Number of Subcarriers	52
OFDM symbol duration	64 sample=3.2usec
Guard Interval	16 sample = 800 nsec
Modulation Type	BPSK
IFFT size	64
CC code	½, (133,171)
Reed Solomon rate	(224,255)
Delay Profile	Exponential 4 tap

4.5 Results

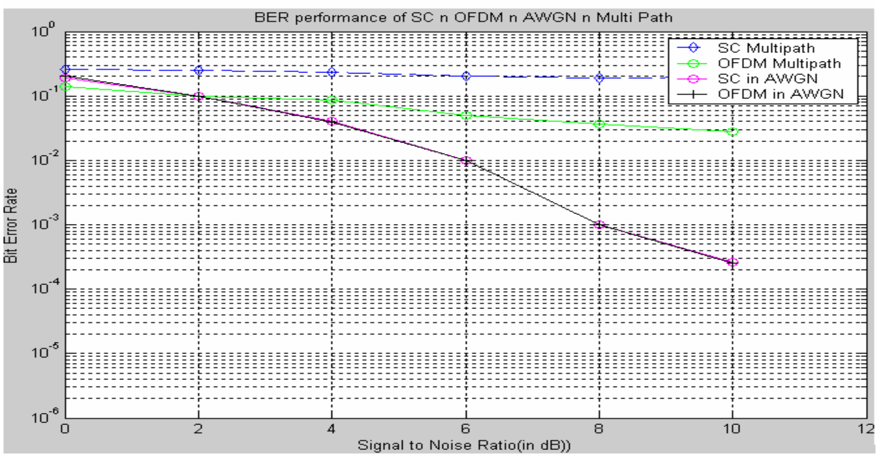


Figure 4.7 BER versus SNR for OFDM and SC in AWGN and Multipath

From the above figure (figure 4.7) we can say that under AWGN channels both SC and OFDM fair equally. But under Multipath conditions OFDM fares better than SC due to its inherent advantages in Multipath compared to SC.

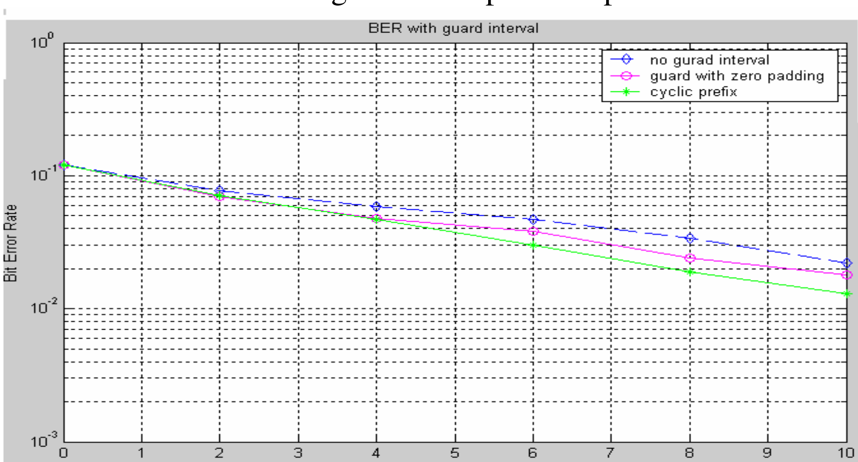


Figure 4.8 BER with changing guard interval

In the figure 4.8 BER was plotted for three cases:

1) no guard interval
2) guard interval with zero padding
3) with cyclic prefix

From this figure (Figure 4.8) we can conclude that with cyclic prefix the BER was very low compared to the other two schemes.

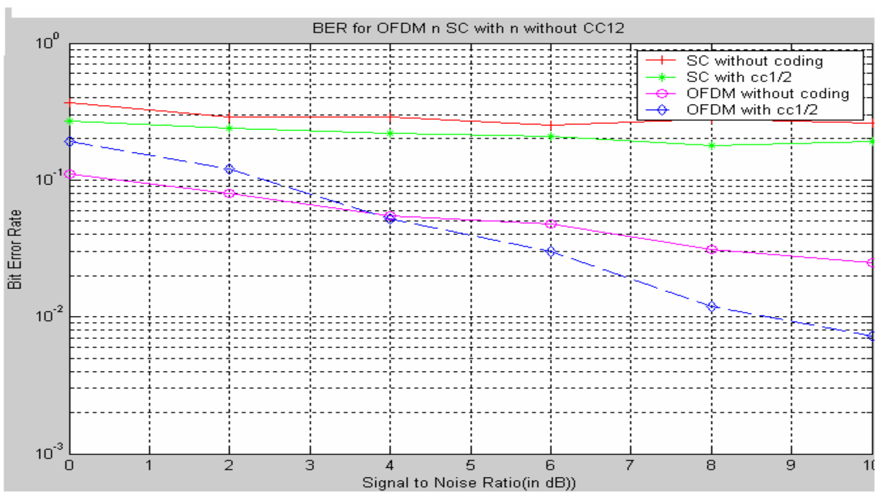


Figure 4.9 OFDM and SC performance with and without coding

From the above figure (figure 4.9), it can be concluded that OFDM and SC doesn't fair well without coding. With coding OFDM outperforms SC in BER performance.

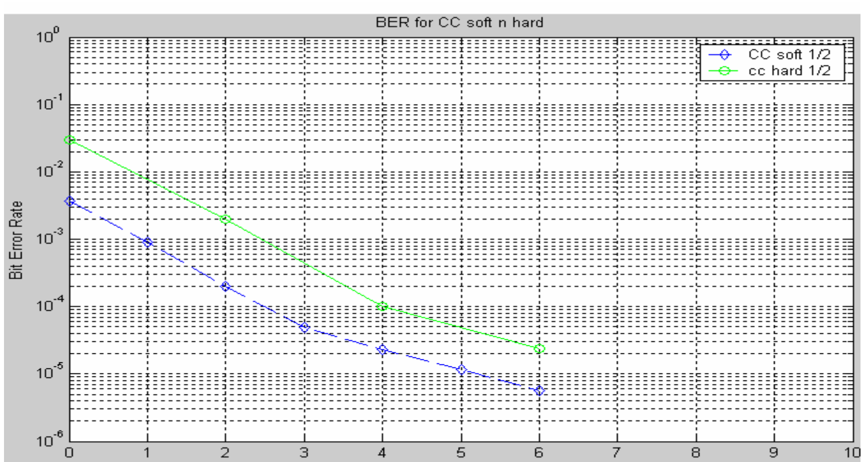


Figure 4.10 BER Versus SNR for CC hard and soft

From the figure 4.10 it can be concluded that CC soft decoding performs well compared to CC hard decoding.

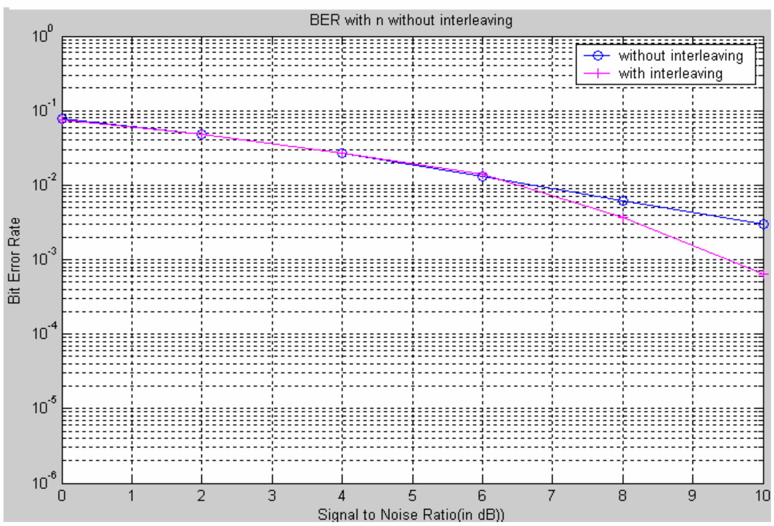


Figure 4.11 BER with and without interleaving

From the above figure (figure 4.11) it can be concluded that the BER performance with interleaving performs better compared to without interleaving.

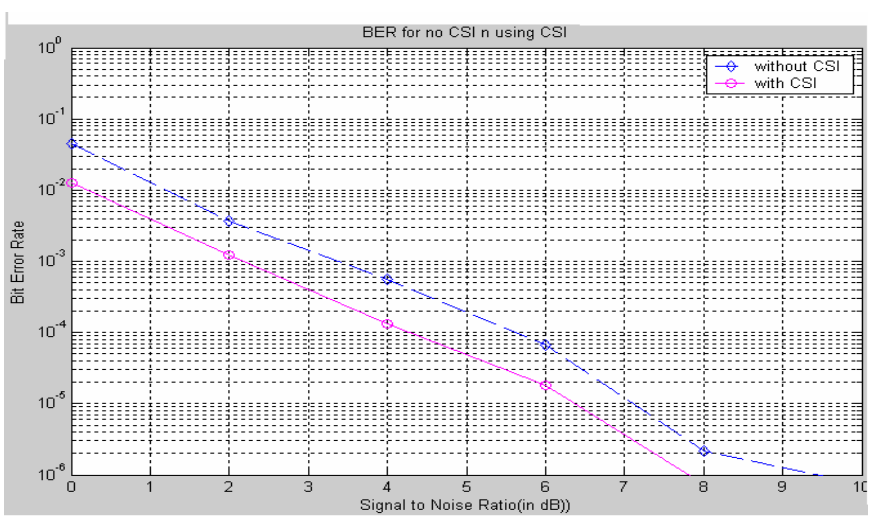


Figure 4.12 BER versus SNR for with and without CSI

From the above figure (figure 4.12) it can be concluded that the BER performance with CSI performs better compared to without CSI.

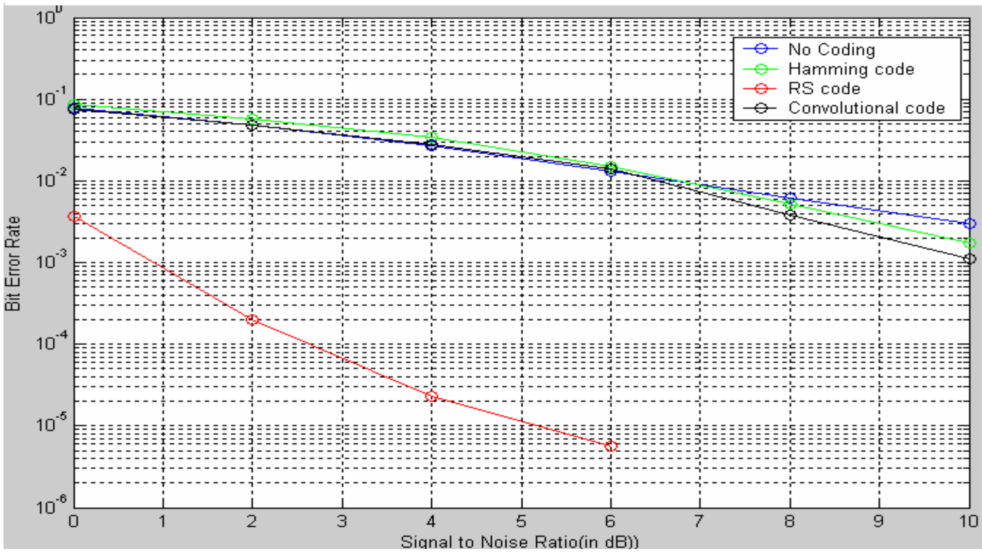


Figure 4.13 BER versus SNR for No coding, Hamming coding, RS code, Convolutional code

From the above figure (figure 4.13) it can be concluded that Convolutional coding performs better than RS and Hamming coding.

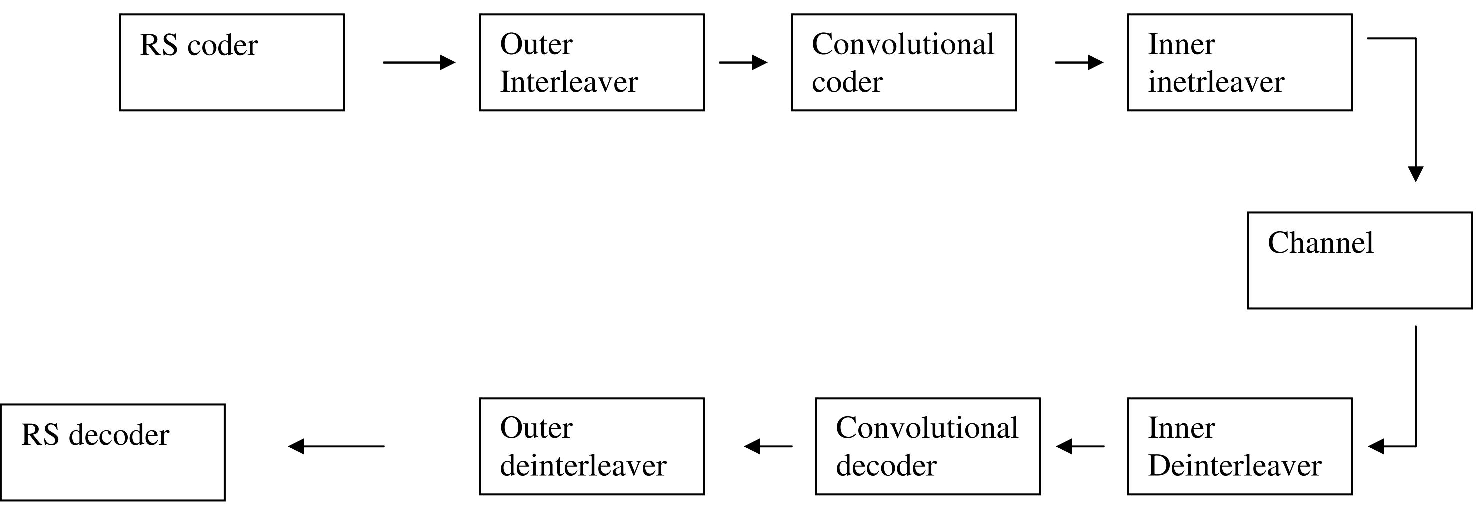


Figure 4.14 RS-CC block diagram

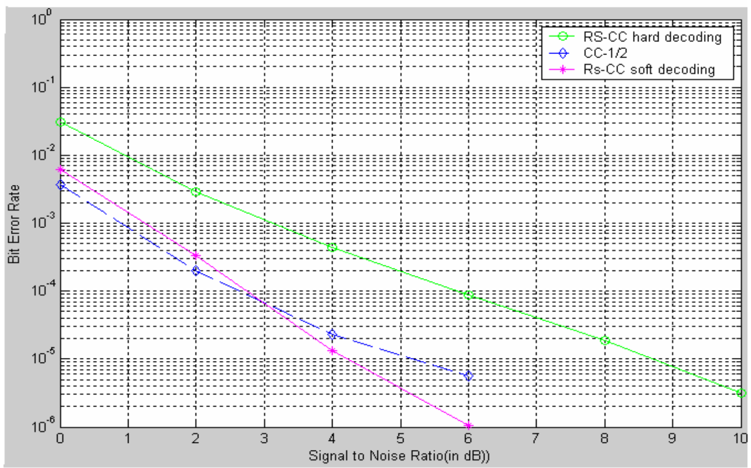


Figure 4.14 BER Vs SNR for CC and RS-CC for hard and soft decoding

From the above figure (figure 4.14) it can be concluded that Reed Solomon Convolutional soft decoding performs better than Convolutional coding and RS-CC soft decoding.

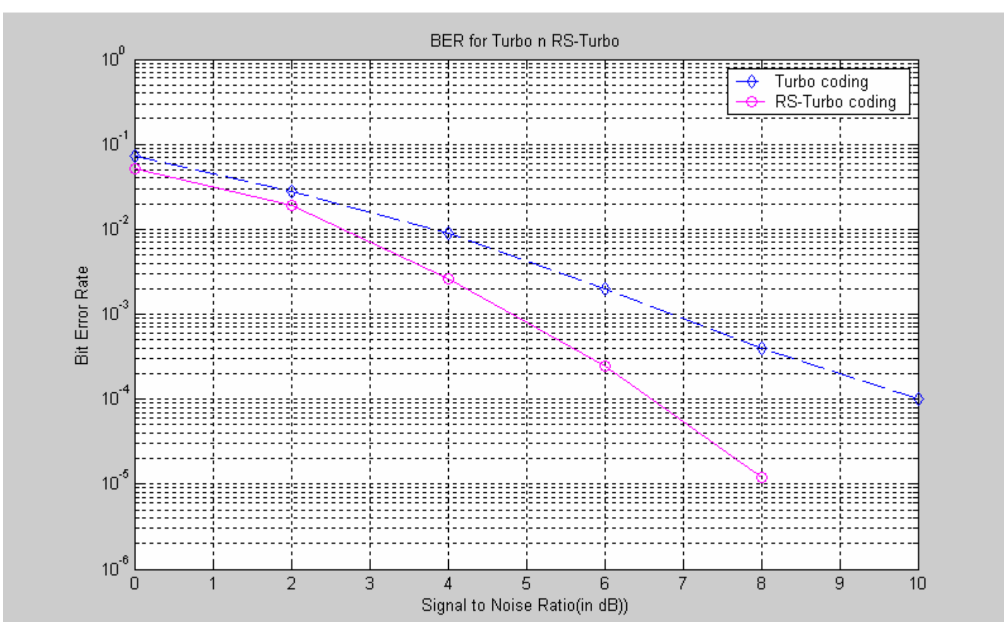


Figure 4.15 BER Vs SNR for Turbo coding and RS-Turbo coding

From the above figure (figure 4.15) it can be concluded that RS-Turbo coding performs better than Turbo coding.

4.5 Conclusion:

Here OFDM and SC are compared and analysed in Multipath and AWGN environments. Next for channel coding different coding techniques are analysed and their results are compared. Of all the coding techniques discussed convolutional coding fares better. Turbo coding also functions better. Next a combination of coding techniques like RS-CC and RS-Turbo are discussed. Of these RS-Turbo fares better than all the coding techniques discussed.

Chapter 5

SPACE FREQUENCY CODED OFDM

5.1 INTRODUCTION:

High data rate reliable transmission over wireless channels is seemingly possible due to the advent of Space-time codes. Space-time codes rely on transmit diversity and are particularly suitable when the signal undergoes frequency flat fading due to the channel. This chapter discusses the performance of the OFDM with space frequency coding technique. Following the introduction this chapter is organized as under
Section 5.2 describes the space frequency coding technique. Section 5.3 describes about overloaded OFDM. Section 5.4 provides concluding remarks.

5.2 SPACE FREQUNENCY CODING TECHNIQUE

In the original space-time code scheme [32], Alamouti showed that it is possible to obtain the same diversity as with multiple receivers. Since then, transmit diversity has been pursued with great interest among the research community. However, the fundamental assumption based on the which the scheme works is that that channel is frequency flat, i.e., the coherence bandwidth of the channel is much smaller than bandwidth of the signal which may not be true in wideband communication [2-5]. This assumption may not be generally true in wideband communication systems. For example, high-data rates are made possible with increased resources in terms of bandwidth in WWAN and outdoor wireless WANs. There is need to developing new technologies for providing wideband wireless communications.

OFDM has matured into a very practicable technique and has been incorporated into the IEEE 802.11a [33]. OFDM splits the channel into sub-channels equal to the no. of carriers under use. Each subchannel is treated independently and the multiplexed modulated symbols are sent over each carrier. This operation is performed via IFFT at the transmitter side and with FFT at the receiver side. This is another interesting aspect of OFDM.

Thus marrying OFDM with Space-time codes appears vary natural in frequency selective fading scenarios. OFDM splits the channel into near frequency -flat subchannels and Space-time codes exploit the transmit diversity under these frequency-flat sub-channels. Together they form a promising technological alternative for high data rate broadband communications. This space-frequency coded OFDM system [6] is shown in Fig. 5.1.

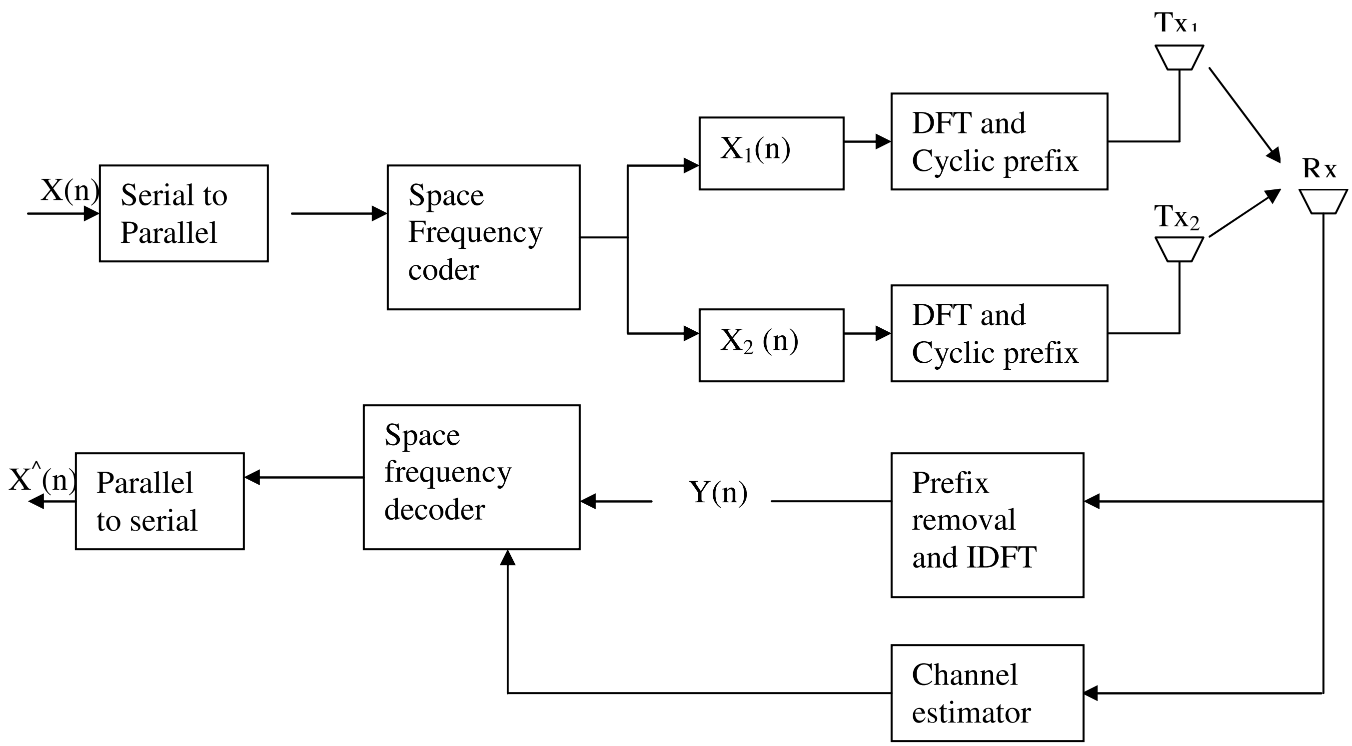


Figure 5.1 A space frequency coded OFDM system

Here, we simulate 2 x 1 Alamouti space-time code with OFDM (which is called as space-fequency code, as the space-time codes are transmitted over another carrier rather than another time-slot). A frequency selective Raleigh channel is assumed. Information bits are M-ary PSK modulated, which are then converted into space-frequency codes. These are multiplexed to form OFDM frames for final transmission. Symbol error rates are computed at different bit-SNRs for different M in the M-ary PSK. Later on, ways are investigated to improve the system. In particular, the performance of the system in which information about few more symbols is spread over all the carriers by varying both the phase and energy of that particular constellation in a controlled manner is studied. Details about the encoding and decoding mechanisms are given. Symbol error rates are also compared.

System Description:

No. of carriers: N_c

Total Bandwidth: W Hz

Input symbols (M-ary coded) in a frame: $X_o\ X_1,\ldots,X_{Nc}$

Modulation: M-ary PSK

No. of Tx: 2

No. of Rx: 1

Space time code: Alamouti 2 x 2 code

$$h_i(n) = \sum_{l=1}^{L} a_i(n)\delta(t\text{-}l/W)\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(4)$$

Channel Model [34,35, 36]:

No. of taps: L = 6

Where a_t are zero mean complex Gaussian random variables with variance 1/L

$$H_i\,(k) = \sum_{j=1}^{Nc} h_i\,(j)\,\exp(-\,j2\pi kn\,/\,N_c)\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots.(5)$$

Space-frequency code at Tx -1 (total N_c symbols) : $X_1 = [X_0 - X_1^*, \ldots.. X_{Nc-2} - X_{Nc-1}^*]^T$

Space-frequency code at Tx -2 (total N_c symbols):$X_2 = [X_1, X_0^*, \ldots, X_{Nc-1}, X^*_{Nc-2}]^T$

Received symbols at the receiver:

$Y_e = A_{1e}X_{1e} + A_{1e}X_{2e} + N_e$

$Y_o = A_{lo}X_{lo} + A_{2o}X_{2o} + N_o$ where

$X_{ie} = X_i(2k)$

$X_{io} = X_i(2k+1)$, $k = 0,1,\ldots Nc/2$

and N ,Y ,A defined likewise.

Λ is a diagonal matrix with *Hi(k)* as its diagonal elements

The estimated (decoded) symbols (assuming that two adjacent sub-channels have approximately same frequency response) after stripping the cyclic prefix and performing FFT operation on the received symbols, are [6],

$$\hat{\mathbf{X}}_e = (|\mathbf{\Lambda}_{1e}|^2 + |\mathbf{\Lambda}_{2e}|^2)\mathbf{X}_e + \mathbf{\Lambda}^*{}_{1e}\mathbf{N}_e + \mathbf{\Lambda}_{1e}\mathbf{N}^*{}_o$$
$$\hat{\mathbf{X}}_e = (|\mathbf{\Lambda}_o|^2 + |\mathbf{\Lambda}_{2o}|^2)\mathbf{X}_o + \mathbf{\Lambda}^*{}_{2e}\mathbf{N}_e - \mathbf{\Lambda}_{1o}\mathbf{N}^*{}_o$$

This equation enables us to study the performance of the scheme completely in the constellation domain.

Performance analysis

We studied the performance of the above scheme with different M-ary groupings and different number of carriers. We varied the SNR from 0 to 20dB. M is varied at from 2 to 4. As we can see from the graph in Fig. 2, the SER is decaying with almost unity slope for all the schemes except the case with Nc=16. The reason might be that, there bandwidth is too small to assume that the channels are frequency flat and also the assumption that adjacent channels have approximately same frequency response may also be violated. The effect of employing carriers from 256 to 512 did not seem very significant as the capacity almost linearly adds up and the sub-channels would tend to be frequency flat. However, this constant SER is at the expense of bandwidth. The effect of varying M from 2 to 4 did shift the SER curve by about 2dB which is expected in M-ary modulation. To achieve the same SER, more power is required so as to push the constellation further away from the origin.

5.3 Over-loaded OFDM

We tried to spread a message symbol onto all the carriers. An explanation is sought to suggest a way of spreading the message symbols. In the OFDM case, message symbols are modulated on to separate carriers which are orthogonal. Suppose if we use time-varying signals to modulate these message symbols, we can overload the time-frequency plane i.e., besides all the carriers, we can modulate some additional symbols using these time-varying signals. In the context of M-ary FSK, this affects the constellation in two ways:

The phase as well as radius of the constellation is varied. In the regular M-ary OFDM, the constellation is cylinder. The radius is equal to the symbol energy and each section of the cylinder corresponds to a carrier. When chirp signals are used to modulate additional symbols, then, the phase of the message symbol in a particular is corrupted by the phase of the chirp signal and phase of the message symbols this chirp signal is carrying. The radius is affected by the addition of instantaneous energy of the chirp signal. The modified constellation can be represented as:

These modified symbols 'are now used to form the space-frequency codes. The decoding is performed by inverting the weight matrix to obtain an approximate Nc+Ne constellation vectors which is subjected to M-ary PSK detection. In the simulation studies, we have used BPSK modulation with Nc=256 and Ne=l. The choice of the weight matrix affects the performance matrix severely. It is of much interest to study how to do design the weight matrix for a given modulation and given number of carriers. Our initial results were not promising and a careful study to construct such pseudo orthogonal weight matrix is of great interest.

$$\begin{bmatrix} S'_o \\ : \\ S'_{Nc} \end{bmatrix} = \begin{bmatrix} 1 \ 0 \ldots 0 \ W_1^0 \ldots W_{Ne} \\ : \\ 0 \ \ldots \ 1 \ W_1^{Nc} \ldots W_{ne}^{Nc} \end{bmatrix} \begin{bmatrix} S_o \\ : \\ S_{Nc} \\ S_{Nc+1} \\ : \\ S_{Nc+Ne} \end{bmatrix}$$

S_i : mapped symbol $i \in [0, Nc-1]$

W_i^k : weight of ith symbol over ith carrier

$$\sum_{i=0}^{Nc-1} W_i^k = 1$$

5.4 Simulation Results

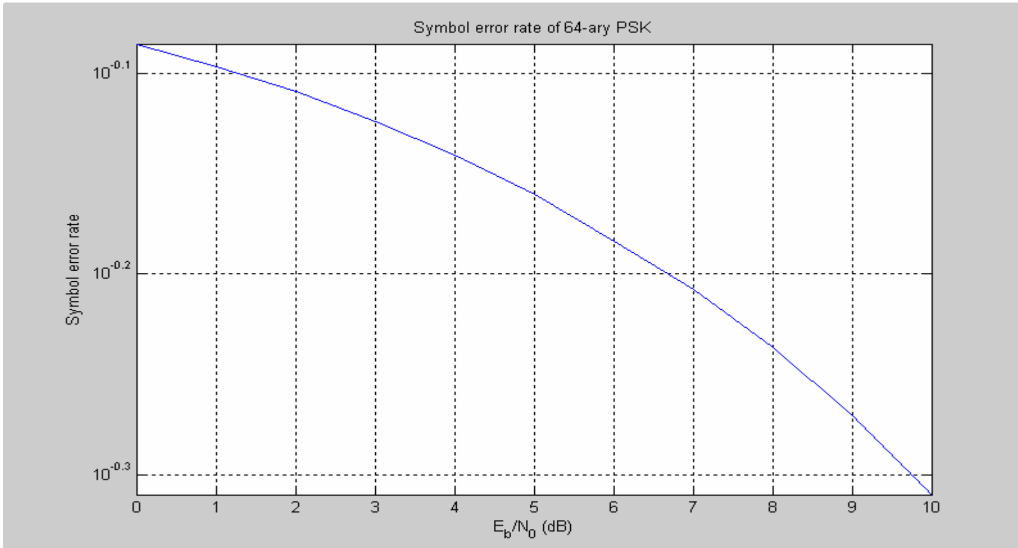


Figure 5.2 SER Vs EbNo for 64 ary PSK

5.5 Conclusions:

For large number of carriers for a given band-width, the SER rate is almost the same. This is because, the sub-channels are frequency flat, and each sub-channel can be used to its full capacity. The assumption that adjacent sub-channels are identical is also valid. Increasing M, the SER shifts by 2dB. This is expected because, to achieve the same SER, more power is required to push the constellation further away from the origin to offer more resolution or detection capability. The intial weight matrix we have chosen was not performing as expected. We expect that there should be no degradation in the SER because, the Ne symbols are spread over all the carriers and all the carriers suffers independent fading. So we expect frequency diversity to offer some advantage here. This lets us think to design new weight pseudo orthogonal matrices.

Chapter 6

CONCLUSION

6.1 Introduction

The work undertaken in this thesis primarily discusses coded OFDM systems and SFC OFDM system. The implementation of OFDM model is presented. The capability of OFDM in Rayleigh faded channels have been analyzed. This thesis analyzes OFDM system and the effect of channel coding in reducing BER. Along with this soft decoding and decoding with CSI is also studied. Besides, performance of convolutional codes Turbo codes in OFDM systems is compared and compared. Besides, the performance of convolution and turbo codes in OFDM systems is compared. This chapter summarizes the work reported in this thesis, specifying the limitations of the study and provides some pointers to future development.

Following this introduction section 6.2 lists the achievements from the work undertaken. Section 6.3 provides the limitations and section 6.4 presents few pointers towards the future work.

6.2 Achievement of the thesis

The work presented in this thesis can be classified into two parts. The first part presents the measure of the performance of OFDM system in AWGN and frequency selective channels and the other part is dedicated for COFDM. Major points of the thesis, highlighting the contributions at each stage, are presented below.
Chapter 2 of this thesis discusses about the fundamental concepts and principles of OFDM Chapter 3 of this thesis presents the implementation of OFDM. We investigated OFDM as an attractive means of transmitting high rate information over highly dispersive

radio channels. Basic components of OFDM Modems as well as a number of design factors, which influence the achievable performance were studied. OFDM Modem's performance, when communicating over conventional AWGN channels and frequency selective channels using various modulation schemes has been studied.
Chapter 5 of this thesis discusses the performance of COFDM using different coding techniques. In this chapter the different coding techniques are discussed.

6.3 Limitations of the work

This section presents some of the limitations of the work reported in this thesis.

In this thesis the implementation of OFDM model has been validated by simulations in matlab.The work undertaken in this thesis only considered Hamming coding, RS coding, convolutional coding and Turbo coding. Other coding techniques like Golay codes, complimentary codes, LDPC and TCM were not considered. Other issues like adjacent channel interference (ACI), co-channel interference (CCI), Rayleigh fading with different delays, were not considered. PAPR reduction techniques are not considered.

6.4 Scope for the further research

By concluding this thesis, the following are some pointers for further works can be undertaken.

The suggested area in which research can be undertaken follows from the limitation of the work presented in this chapter. Different efficient coding and modulation schemes could be considered in the simulation.

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