PERFORMANCE OF CONVOLUTIONAL CODED OFDM IN MULTIPATH CHANNEL

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Declaration

I declare that this written submission represents my ideas in my own words, and where others' ideas or words have been included, I have adequately cited and referenced the original sources. I also declare that I have adhered to all principles of academic honesty and integrity and have not misrepresented or fabricated or falsified any idea/data/fact/source in my submission. I understand that any violation of the above will be a cause for disciplinary action by the Institute and can also evoke penal action from the sources that have thus not been properly cited, or from whom proper permission has not been taken when needed.

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Approval Sheet

This thesis entitled PERFORAMENCE OF CONVOLUTIONAL CODED OFDM IN MULTIPATH CHANNELS by Ramesh G is approved for the degree of Master of Technology from IIT Hyderabad.

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Last but not least, I would like to thank my parents for their endless love and support.

Dedicated to

Abstract

Wimax and 4G-LTE are the two Promising Technologies proposed to provide high data rates around 100 Mbps. So as to satisfy the multimedia services demands from increasing mobile users. Both these Technologies employs OFDM modulation and Multiple Antenna Technology to achieve high data rates. Data in general will be coded before transmitted through wireless channel. So as to reduce the reception errors i.e. signal is channel coded to minimize the BER. This thesis evaluates the performance of coded OFDM scheme and studies how coded OFDM performs better with respect to BER than non-coded OFDM. In this thesis Convolution coding scheme was employed at the transmitter side and Viterbi decoder at Receiver end. over a slow varying Rayleigh channel. BPSK is used to modulate the incoming data symbols and results are validates via simulator.

Nomenclature

AMPS	Advanced Mobile Phone Services
AWGN	Additive White Gaussian Noise
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
CP	Cyclic prefix
DFT	Discrete Fourier Transform.
FDMA	Frequency Division Multiple Access
GSM	Global System for Mobile Communication
IDFT	Inverse Discrete Fourier Transform
FFT	Fast Fourier Transform
IFFT	Inverse Fast Fourier Transform
ICI	Inter Carrier Interference
ISI	Inter Symbol Interference
LS	Least Square
MCM	Multicarrier Modulation
MMSE	Minimum Mean Square Estimation
OFDM	Orthogonal Frequency Division Multiplexing
PSK	Phase Shift Keying
PSMA	Pilot Symbol Assisted Modulation
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RECT	Rectangular
SCM	Single Carrier Modulation
SNR	Signal to Noise Ratio

Contents

Chapter 1

Introduction

Wireless communication is been promising means of communication over last two decades. Today almost all of the world population are using mobile phones as a means of communication. Where itself results how rapidly is wireless industry scaling over a period of time. Wireless communication has enabled many references required to meet the needs of increasing traffic many technologies and methodologies are evolved over a period of time now we are in era where single electronic device is going to be hub of many application. 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 wired alternatives, and the need for mobility proves especially challenging for reliable communications.

The fading phenomenon occurs in radio transmission channels. It is due to the presence of multipaths that varies during the transmission [1].There are many techniques used to compensate for fading channel impairments [2], [3]. Use of error control coding is one of the important techniques. It is used to enhance the efficiency and accuracy of information transmitted. In a communication system, data is transferred from a transmitter to a receiver across a physical medium of transmission or channel. The channel is generally affected by noise or fading which introduces errors in the data being transferred. Channel coding is a technique used for correcting errors introduced in the channel. It is done by encoding the data to be transmitted and introducing redundancy in it such that the decoder can later reconstruct the data transmitted using the redundant information. If the error control coding is doing its job properly, the bit error rate at the output should be less than the bit error probability at the decoder input [5]. In my thesis convolution code is used as an error control code. The Viterbi algorithm was proposed in 1967 as a method of decoding convolution codes [6]. Viterbi decoding is considered and the bit error rate performance is evaluated for convolution code and it is compared with the bit error rate for uncoded signal under AWGN channel and slow Rayleigh fading channel.

1.1 Organization of Thesis:

Chapter 1: Introduction

Chapter 2: The basic principle of OFDM system is discussed in Chapter-2. OFDM Communication system including its generation and reception, advantages, and Attenuations, and implementation of the system. Orthogonal Frequency Division Multiplexing (OFDM) as a transmission technique is known to possess a lot of strength, compared to any other transmission technique, such as high spectral efficiency, robustness to the channel fading and immune to impulse interference.

Chapter 3: which give the introduction to wireless environmental. Fading channels discretion like Frequency selective fading, Rayleigh fading than Generating fading model by using jakes model and AWGN fading. System model description.

Chapter 4: Convolutional coding and Decoding, How the convolutional encoder will generates code words and it’s functioning along with State diagram-tree diagram-trellis diagram-Viterbi decoding

Chapter 5: Results and conclusions

References

Chapter 2

Orthogonal Frequency Division Multiplexing

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 elaborates the basics and implementation of OFDM in real time systems. The chapter also discusses the propagation characteristics of a mobile communication channel.

2.2 Why OFDM

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.

Two periodic signals are orthogonal when the integral of their product over one period is equal to zero. For the case of continuous time:

$$\int_0^T \cos(2\Pi n f o t)\cos(2\Pi m f o t)\,dt = 0,$$

For the case of discrete time:

$$\sum_{k=0}^{N-1} \cos\left(\frac{2\pi k n}{N}\right)\cos\left(\frac{2\pi k m}{N}\right)dt = 0,$$

Where $m \neq n$ in both cases.

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 2.1.

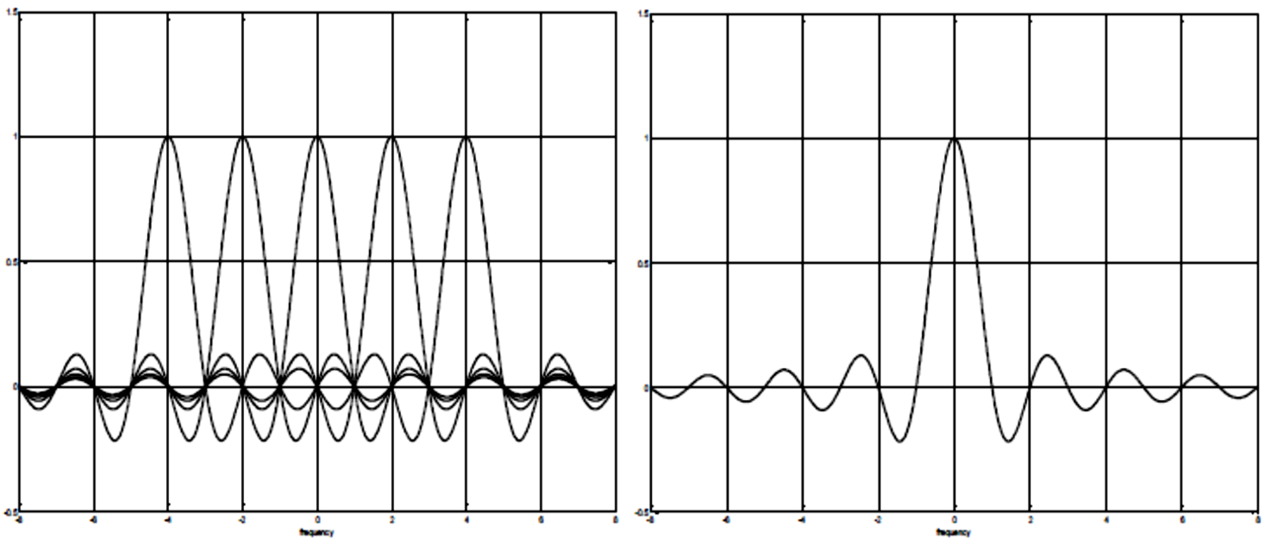

Fig 2.1 Spectra of OFDM signal a and sub-channel

Each sub – carrier in an OFDM system is a sinusoid with a frequency that is an integer multiple of a fundamental frequency. Each sub – carrier is like a Fourier series component of the composite signal, an OFDM symbol.

The sub – carriers' waveform can be expressed as

$$\mathrm{s(t)} = \cos(2\pi fct + \theta k)$$

$$= a_n \cos(2\pi n f_o t) + b_n \sin(2\pi n f_o t)$$

$$= \sqrt{a_n^2 + b^2 n}\cos(2\pi n f_o t + \varphi),$$

Where $\varphi = \tan^{-1}(\frac{bn}{an})$

The sum of sub carrier is the baseband OFDM signal

$$s(t) = \sum_{n=0}^{N-1} \{a_{ncos(2\pi n f_o t) - b_n \sin(2\pi n f_o t)}\}$$

Each sinc of the frequency spectrum in the Fig 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.

2.3 Channel Impairments

The transmitted signal faces various obstacles and surfaces of reflection, as a result of which the received signals from the same source reach at different times. This gives rise to the formation of echoes which affect the other incoming signals. Inter – symbol interference (ISI) is a form of distortion of a signal in which one symbol interferes with subsequent symbols. This is an unwanted phenomenon as the previous symbols have similar effect as noise, thus making the communication less reliable. ISI is usually caused by multipath propagation or the inherent non – linear frequency response of a channel causing successive symbols to “blur” together. The presence of ISI in the system introduces error in the decision device at the receiver output. Therefore, in the design of the transmitting and receiving filters, the objective is to minimize the effects of ISI and thereby deliver the digital data to its destination with the smallest error rate possible.

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 than high frequency signals. Thus high frequency signals, especially, Ultra High Frequencies (UHF), and
microwave signals require line of sight for adequate signal strength. To overcome the problem of shadowing, transmitters are usually elevated as high as possible to minimize the number of obstructions.

2.4 Principle 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 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. 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”. 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 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 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. 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.

2.4.1 Cyclic Prefix

The Cyclic Prefix or Guard Interval is a periodic extension of the last part of an OFDM symbol that is added to the front of the symbol in the transmitter, and is removed at the receiver before demodulation.

The cyclic prefix has to two important benefits –

- The cyclic prefix acts as a guard interval. It eliminates the inter – symbol interference from the previous symbol.
- It acts as a repetition of the end of the symbol thus allowing the linear convolution of a frequency – selective multipath channel to be modeled as circular convolution which in turn maybe transformed to the frequency domain using a discrete Fourier transform. This approach allows for simple frequency – domain processing such as channel estimation and equalization.

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Fig 2.2 Cyclic prefix

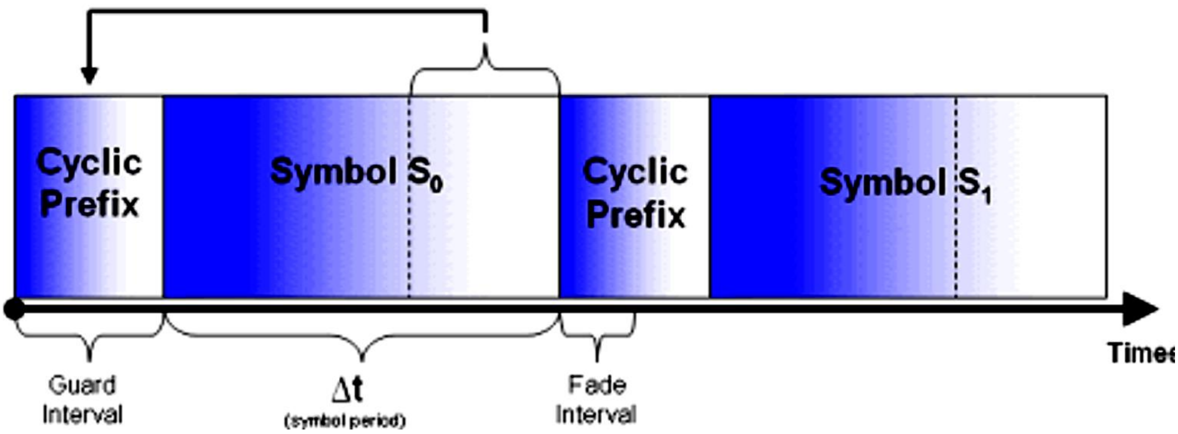


Fig 2.3 Cyclic prefix of the two OFDM symbol

2.5 Implementation of OFDM

An OFDM system was modeled using Matlab 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. The OFDM system modeled using Matlab is shown in Figure 2.4.

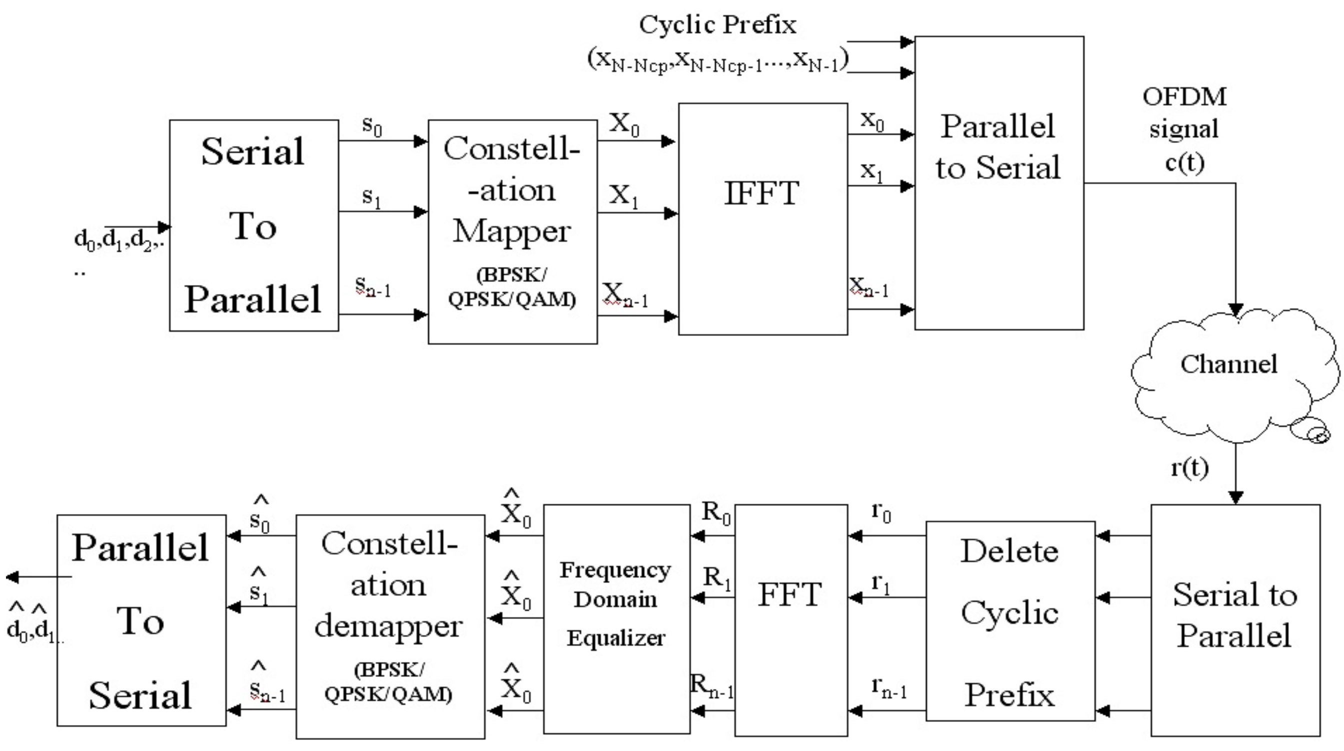


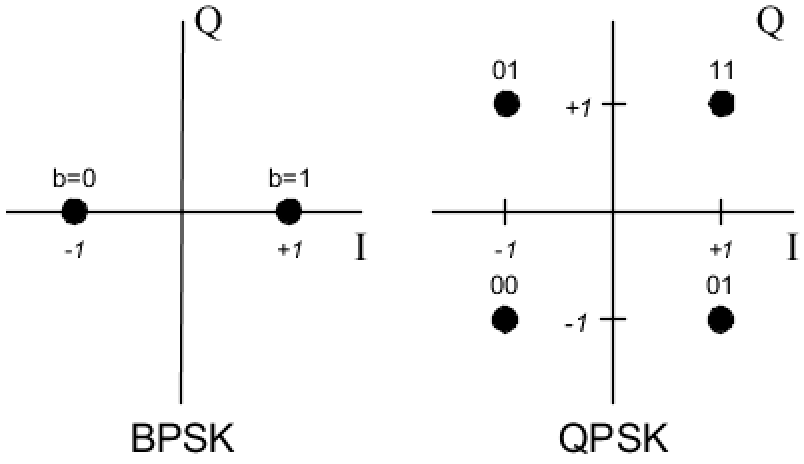
Fig 2.4 Modulation and Demodulation in OFDM system

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. The data is now mapped on to the subcarriers accordingly. 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.

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 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.

2.6 QAM Mapper

Once the signal has been coded, it enters the constellation mapper block. All wireless communication systems use a modulation scheme to map coded bits to a form that can be effectively transmitted over the communication channel. Thus, the bits are mapped to a subcarrier amplitude and phase, which is represented by a complex in-phase and quadrature-phase (IQ) vector. The IQ plot for a modulation scheme shows the transmitted vector for all data word combinations. Types of digital modulation include BPSK, QPSK, etc. The constellation maps for BPSK, and QPSK, modulations are shown in Figure 2.5.



(a) (b)

Fig: 2.5: Constellation Maps: (a) BPSK, and (b) QPSK

The constellation mapped data is subsequently modulated onto all allocated data carriers in order of increasing frequency offset index.

2.7 Discrete Fourier Transform

The Fast Fourier Transform (FFT) is an effective algorithm for the implementation of the DFT. Forward FFT takes a random signal, multiplies it successively by complex exponentials over the range of frequencies, sums each product and plots the results as a coefficient of that frequency. The coefficients are called a spectrum and represent ―how much‖ of that frequency is present in the input signal.

FFT can be written in sinusoids as:

$$x(k) = \sum_{n=0}^{N-1} X(n)\sin(\frac{2\pi kn}{N}) + j\sum_{n=0}^{N-1} X(n)\cos(\frac{2\pi kn}{N})$$

Here $X(n)$, are coefficients of the sines and cosines of frequency $2\pi k/N$, where k is the index of the frequencies over the N frequencies and n is the time index. $x(k)$ is the value of the spectrum for the k^{th} frequency and $X(n)$ is the value of the signal at time n. The IFFT takes a frequency spectrum and converts it to a time domain signal by again successively multiplying it by a range of sinusoids. The equation for an IFFT is:

$$X(n) = \sum_{n=0}^{N-1} x(k)\sin(\frac{2\pi kn}{N}) - j\sum_{n=0}^{N-1} x(k)\cos(\frac{2\pi kn}{N})$$

The IFFT is used to produce a time domain signal, as the symbols obtained after modulation can be considered the amplitudes of a certain range of sinusoids. This means that each of the discrete samples before applying the IFFT algorithm corresponds to an individual subcarrier. Besides ensuring the orthogonality of the OFDM subcarriers, the IFFT represents also a rapid way for modulating these subcarriers in parallel, and thus, the use of multiple modulators and demodulators, which spend a lot of time and resources to perform this operation, is avoided.

2.8 ADVANTAGES & DISADVANTAGES OF AN OFDM SYSTEM

Advantages

- Due to increase in symbol duration, there is a reduction in delay spread. Addition of guard band almost removes the ISI and ICI in the system.
- Conversion of the channel into many narrowly spaced orthogonal sub – carriers render it immune to frequency selective fading.
- As it is evident from the spectral pattern of an OFDM system, orthogonally placing the sub – carriers lead to high spectral efficiency.
- Can be efficiently implemented using IFFT.

Disadvantages

- These systems are highly sensitive to Doppler shifts which affect the carrier frequency offsets, resulting in ICI.
- Presence of a large number of sub – carriers with varying amplitude results in a high Peak – to – Average Power Ratio (PAPR) of the system, which in turn hampers the efficiency of the RF amplifier.

Chapter 3

Wireless channel and System Model

3.1 Introduction:

The rapid fluctuation of the amplitude of a signal over a relatively small distance is referred to as fading. Interference between two or more versions of the transmitted signal can result in different propagation delays at the receiver and this is known as multipath. Some of the causes of multipath as pointed out in [28] are: atmospheric ducting, ionospheric reflection and refraction, and reflection from water bodies and terrestrial objects such as mountains and buildings. Due to the relative motion between the mobile and the base station, each multipath wave experiences an apparent shift in frequency. The shift in received signal frequency due to motion is called the Doppler shift, and is directly proportional to the velocity and direction of motion of the mobile with respect to the direction of arrival of the received multipath wave [36].

The factors influencing small scale fading are:

1. Multipath propagation

2. Speed of the mobile

3. Speed of surrounding objects

4. The transmission bandwidth of the signal

The classification of fading is based on the relationship between the signal parameters and the channel parameters. The channel is typically characterized by its impulse response which contains all the necessary information required to analyse or simulate any type of radio transmission through the channel [36].

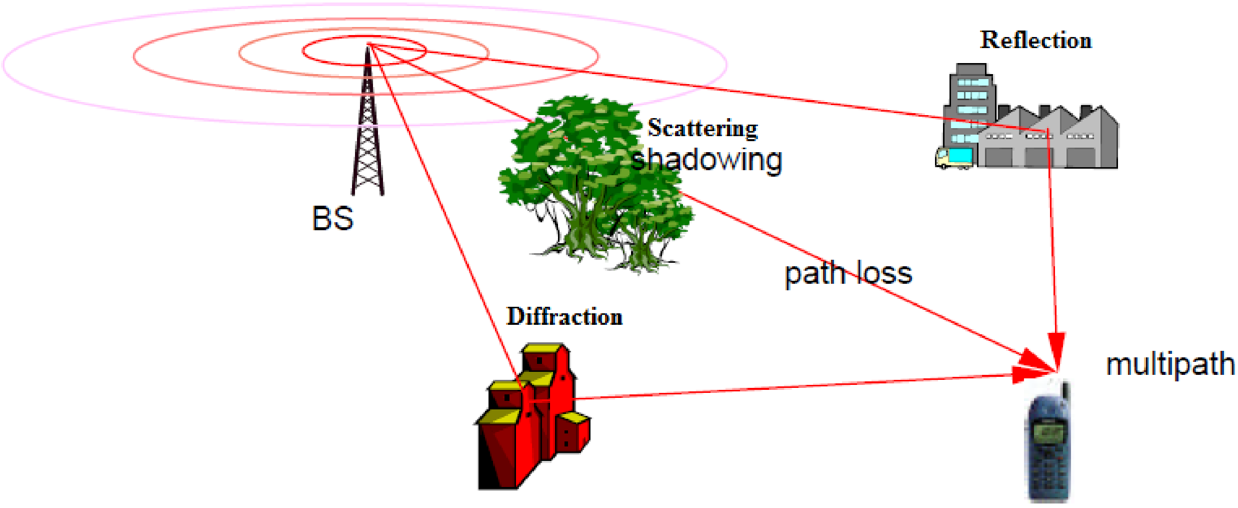


Figure 3.1 Multipath Scattering and Shadowing

3.2 Fading Channel

The measure of how quickly the channel response de-correlates is called the coherence time. When the coherence time is large compared to the symbol duration of the signal, then the channel is referred to as slow fading. Fast fading is the opposite of slow fading and occurs when the coherence time is small or comparable to the symbol duration. Another classification of the fading process depends on the relationship between the delay spread of the channel which is a measure of its time depressiveness and the symbol duration. When the delay spread is much smaller than the symbol duration the fading is classified as flat and when it is not it is termed as frequency selective fading [35].

Doppler shift is caused by the relative motion between the receiver and the transmitter. Doppler spread B_D is a measure of the spectral broadening caused by the time rate of change of the mobile radio channel and is defined as the range of frequencies over which the received Doppler spectrum is essentially non-zero. When a pure sinusoidal tone of frequency f_c is transmitted, the received signal spectrum, called the Doppler spectrum, will have components in the range $(f_c - f_d)$ to $(f_c + f_d)$ where f_d is the

Doppler shift. The amount of spectral broadening depends on f_d which is a function of the relative velocity of the mobile and the angle θ between the direction of motion of the mobile and the direction of arrival of the scattered waves [8].

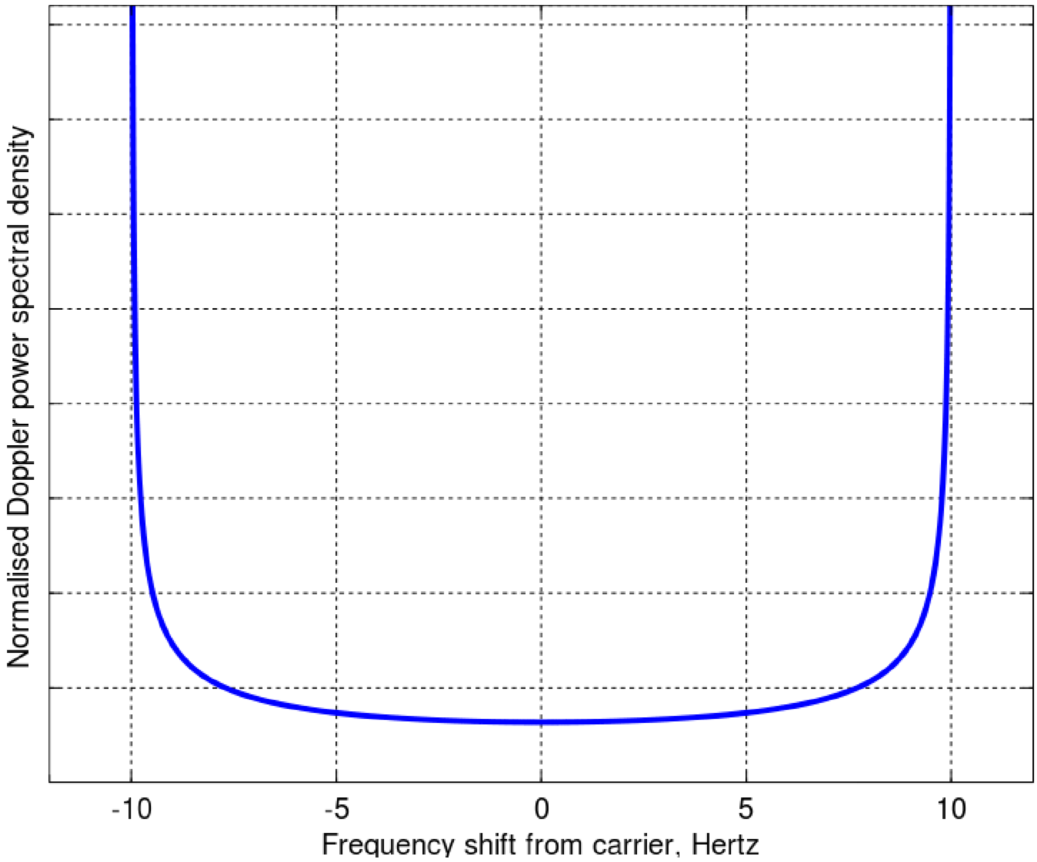


Figure 3.2: Doppler power spectral density of Rayleigh fading with a maximum Doppler shift of 10Hz.

3.3 Frequency Selective Fading

Fading is considered to be flat when the symbol duration of the signal is much larger than the delay spread of the channel. This is desirable for communication, unfortunately, for high data rate applications the signal bandwidth increases and the symbol period is on the order of a few microseconds. The frequency selective fading channel can be modelled as an L tap filter depicted in Figure 3.3. is the number of L resolvable paths provided by the channel and is a measure of the diversity available in the channel.

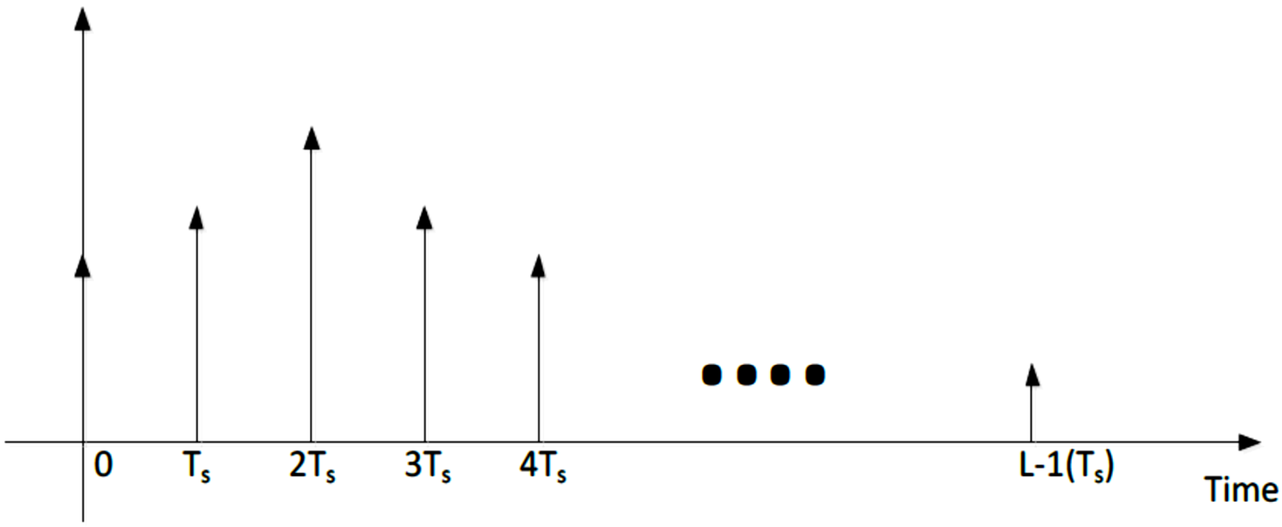


Figure 3.3: L Taps Channel Model

$$L = \left| \frac{T_d}{T_s} \right| + 1 \qquad (3.1)$$

Where T_d is the delay spread of the channel and T_s is the symbol duration. The impulse response of the channel can be then expressed as:

$$h(\tau,t) = \sum_{k=1}^{L} h_k(t)\delta(\tau - kT_s) \qquad (3.2)$$

The usual model assumed for frequency selective fading is Wide Sense Stationary with Uncorrelated Scattering (WSSUS). This implies that the tap gains are uncorrelated [7].

3.4 Rayleigh Fading Channel

The equivalent complex baseband received signal $r(t)$ in a multipath channel can be

Expressed as:

$$r(t) = \sum_{k=1}^{N} a_k(t)e^{j\theta_k(t)}s(t - \tau_k) + n(t) \qquad (3.3)$$

Where, a_k, θ_k and τ_k are the multiplicative gain, phase shift and the delay of the k^{th} path, N denotes the number of paths $s(t)$ is the transmitted signal and $n(t)$ is the Additive White Gaussian Noise term.

When the path delays are small compared to the symbol duration $(t-\tau_k) \approx s(t)$ and the received signal can be expressed as:

$$r(t) = \sum_{k=1}^{N} a_k(t) e^{j\theta_k(t)} s(t-\tau_k) + n(t) \qquad (3.4)$$

$$r(t) = g(t)s(t) + n(t) \qquad (3.5)$$

Where

$$g(t) = x(t) + jy(t) \qquad (3.6)$$

$$x(t) = \sum_{k=1}^{N} a_k(t) \cos\theta_k(t) \qquad (3.7)$$

$$y(t) = \sum_{k=1}^{N} a_k(t) \sin\theta_k(t) \qquad (3.8)$$

From the above equation we can see that the original transmitted signal is modulated by a random time varying scale factor $g(t).x(t)$ is the in-phase component and $y(t)$ is the quadrature component of the gain. When the number of paths is large we can use the Central Limit Theorem to show that $x(t)$ and $y(t)$ are independent Gaussian random processes. This type of fading is known as Rayleigh fading as the envelope of the scale factor $(g(t))$ follows a Rayleigh distribution

$$f_R(r) = \frac{r}{\sigma^2} e^{-\frac{r^2}{2\sigma^2}}, r \geq 0 \qquad (3.9)$$

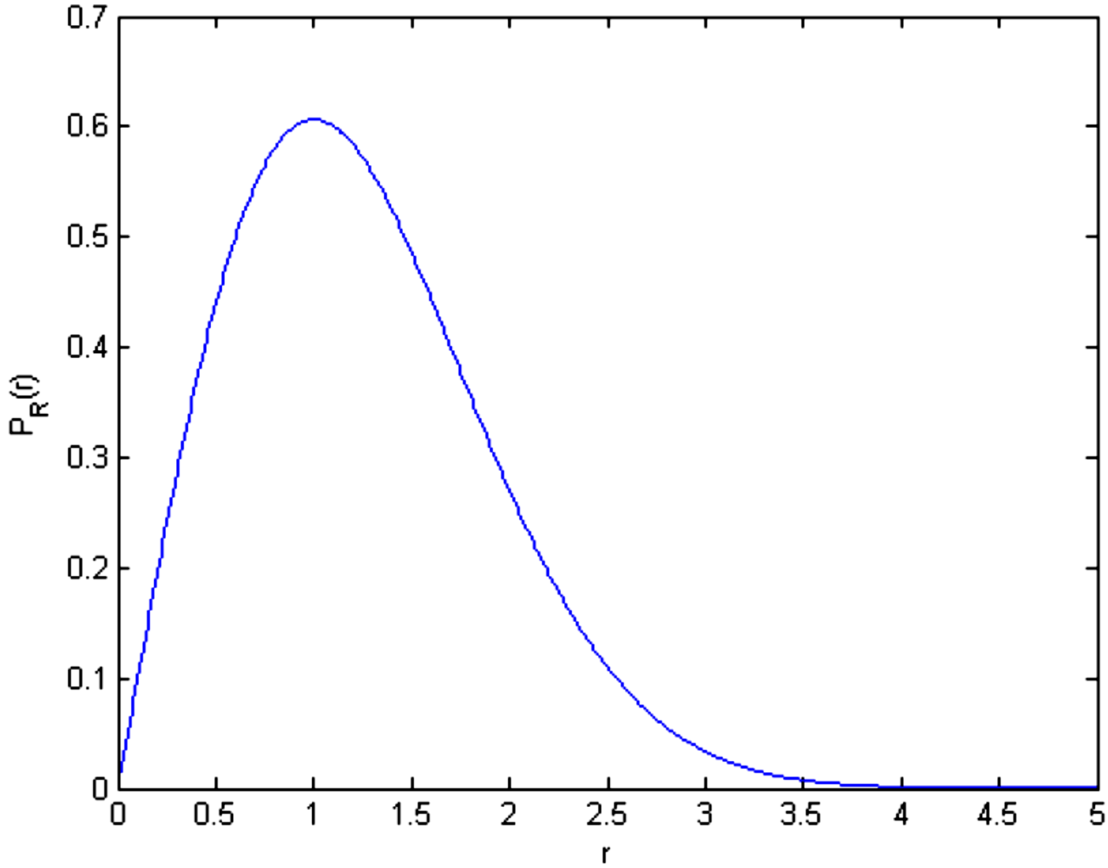


Figure 3.4: PDF of Rayleigh Fading Envelope

The phases θ_k are uniformly distributed in the interval $[0, 2\pi]$, and independent for each path. This type of fading is the most commonly dealt with type of fading in the literature and is a good model for urban areas where there is no dominant or line-of-sight path available between the transmitter and the receiver.

3.5 Additive White Gaussian Noise Channel

This is a channel model in which the only impairment to communication is a linear addition of wideband or white noise with a constant spectral density and a Gaussian distribution of amplitude. The model does not account for fading, frequency selectivity, interference, nonlinearity or dispersion. However, it produces simple and tractable mathematical models which are useful for gaining insight into the underlying behavior of a system before these other phenomena are considered [4].

Wideband Gaussian noise comes from many natural sources, such as the thermal vibrations of atoms in conductors, shot noise, black body radiation from the earth and other warm objects, and from celestial sources such as the Sun.

3.6 Generating Fading (Jakes' Model)

From the definition of Rayleigh fading given above, it is possible for one to generate this model by generating two independent Gaussian random variables namely: $x(t)$ and $y(t)$. However, sometimes only the amplitude fluctuations are of interest. Note that this is for link level simulations of wireless communication only. The aim of generating Rayleigh fading is to produce a signal that has the same Doppler spectrum

Jakes' model is based on summing sinusoids as defined by the following equations:

$$g(t) = x(t) + jy(t)$$

$$g(t) = \sqrt{2}\left\{\left[2\sum_{n=1}^{M}\cos\beta_n \cos 2\pi f_n t + \sqrt{2}\cos a \cos 2\pi f_m t\right] + j\left[2\sum_{n=1}^{M}\cos\beta_n \cos 2\pi f_n t + \sqrt{2}\cos a \cos 2\pi f_m t\right]\right\}$$

(3.10)

Where

$$a_n = \hat{\phi}_n = -\hat{\phi}_n \qquad (3.11)$$

$$\beta_n = \hat{\phi}_n = -\hat{\phi}_n \qquad (3.12)$$

$\hat{\phi}$ is the random phase given by

$$\hat{\phi} = -2\pi(f_c + f_m)\tau_n$$

Where

$f_m = {}^{v}\!/\!_{\lambda_c}$ is the maximum Doppler frequency

f_c is the carrier frequency

There are M low frequency oscillators with frequency $f_n = f_m \cos(2\pi n / N)$, $n = 1, 2, 3....., M$ where $M = \frac{1}{2}(\frac{N}{2} - 1)$ and where N is the number of sinusoids. The amplitudes of the oscillators are all unity except for the oscillator at frequency f_m which has amplitude ${}^{1}\!/\!_{\sqrt{2}}$. It is desirable that the phase of $g(t) = x(t) + jy(t)$ be uniformly distributed. This can be accomplished using time averaging described in [9].

3.7 System Model

An OFDM system was modeled using Matlab to allow various parameters of the system to be varied and tested. The aim of doing the simulations was to measure the BER performance of coded OFDM and uncoded OFDM under different channel conditions. And to compare the performance of the OFDM system in different multipath channels. Below shown in Figure 3.5. A brief description of the model is provided below.

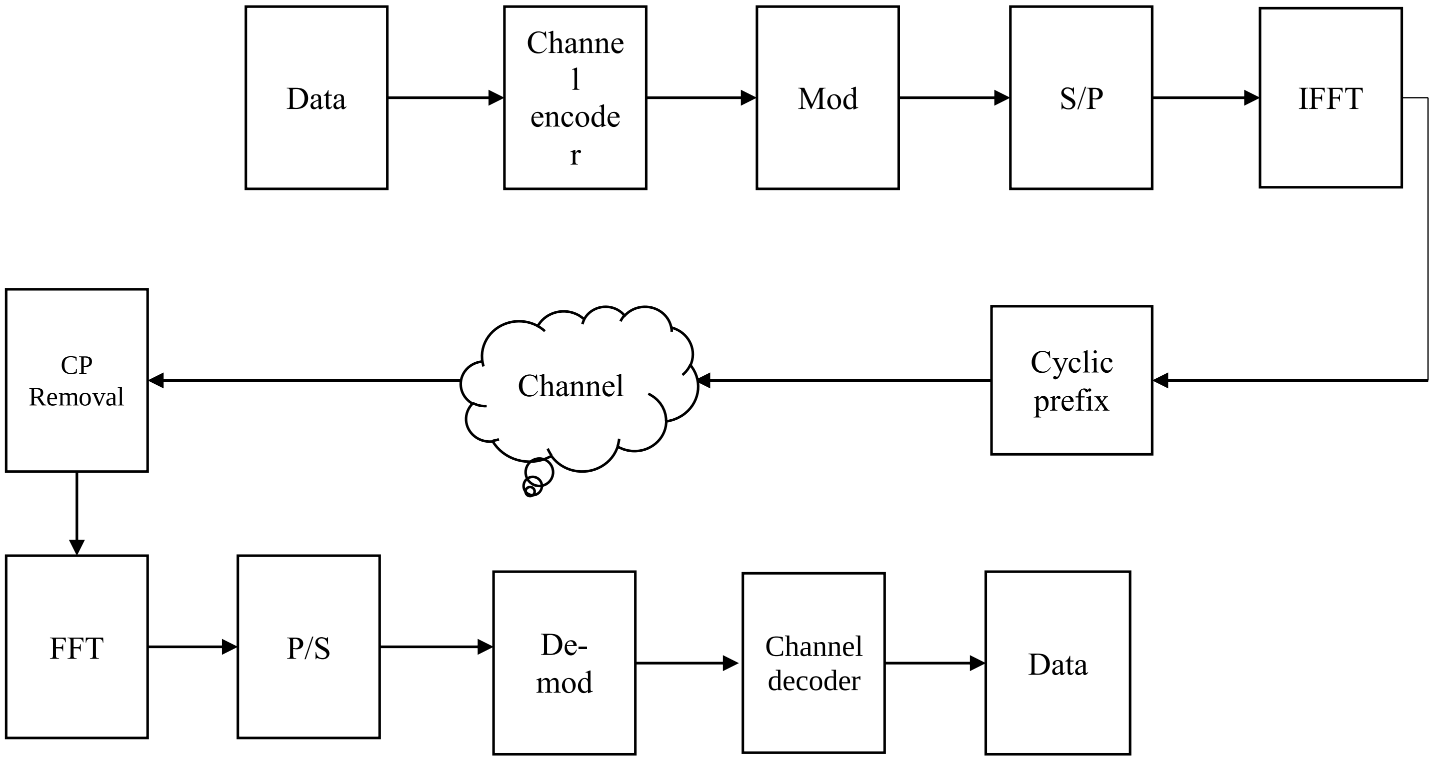


Fig 3.5: System model used for simulation

Channel encoder:

Channel encoder add some redundant bits to incoming data stream according to some predefined algorithm which facilitates reception with minimal errors.

Modulator

Binary Phase Shift Keying was employed to modulate the bits from channel encoder where data on each symbol is mapped to corresponding phase angle.

Serial to parallel conversion

Serial data stream is formatted into the word size required for transmission, and shifted into a parallel format. The data is then transmitted in parallel by assigning each data word to one carrier in the transmission.

Inverse Fourier Transform

An inverse Fourier transform generates a time domain waveform which is then up-converted to RF and aired through wireless channel.

Cyclic prefix

In order to account for Inter Symbol Interference which may arise because of spreading effect of multipath channel few last samples of ofdm symbol are duplicated and prepended to OFDM symbol.

Channel

Slow varying Rayleigh fading channel model which capture the effects of wireless channel between transmitter and receiver is applied to transmitted signal

CP Removal:

The cyclic prefix added at transmitter should be removed before further processing of received signal

An inverse operation (DFT) to IDFT at transmitter is performed at receiver to extract data samples on each subcarrier

Demodulation:

The extracted data samples are serialized and applied to MAP detector to recover the transmitted information.

Chapter 4

Convolution coding and decoding

4.1 Introduction

Convolutional codes are widely used as channel codes in practical communication systems for error correction. The encoded bits depend on the current k input bits and a few past input bits. The main decoding strategy for convolutional codes is based on the widely used Viterbi algorithm. As a result of the wide acceptance of convolutional codes, there have been several approaches to modify and extend this basic coding scheme. Trellis coded modulation (TCM) and turbo codes are two such examples. In TCM, redundancy is added by combining coding and modulation into a single operation. This is achieved without any reduction in data rate or expansion in bandwidth as required by only error correcting coding schemes.

4.2 Convolutional Encoding

Convolutional codes are a family of error correcting codes which add redundant information based on the block of data they are processing. Convolutionally encoding data is basically accomplished using shift registers and associated combinatorial logic that perform modulo-two addition. A convolutional code is specified by C (n, k, K), in which ‘k’ length each information symbol to be encoded is transformed into an ‘n’ bit symbol, where ‘k/n’ is the code rate and the transformation is a function of the last information symbols, where ‘K’ is the constraint length of the code [21].

4.3 Structure of the Convolutional Code

Convolutional codes are commonly described using two parameters: the code rate and the constraint length. The code rate, k/n, is expressed as a ratio of the number of bits into the convolutional encoder (k) to the number of channel symbols output by the convolutional encoder (n) in a given encoder cycle. The constraint length parameter, K, denotes the "length" of the convolutional encoder, i.e. how many k-bit stages are available to feed the combinatorial logic that produces the output symbols. Closely related to K is the parameter k, which indicates how many encoder cycles an input bit is retained and used for encoding after it first appears at the input to the convolutional encoder. The k parameter can be thought of as the memory length of the encoder.

A simple convolutional encoder is shown in Fig. 4.1. The information bits are fed in small groups of k-bits at a time to a shift register. The output encoded bits are obtained by modulo-2 addition (EXCLUSIVE-OR operation) of the input information bits and the contents of the shift registers which are a few previous information bits. If the encoder generates a group of ‘n’ encoded bits per group of ‘k’ information bits, the code rate R is commonly defined as R = k/n. in below fig 4.1.it is taken as k = 1 and n = 2. The number, K of elements in the shift register which decides for how many codewords one information bit will affect the encoder output, is known as the constraint length of the code. For the present example,K = 3.

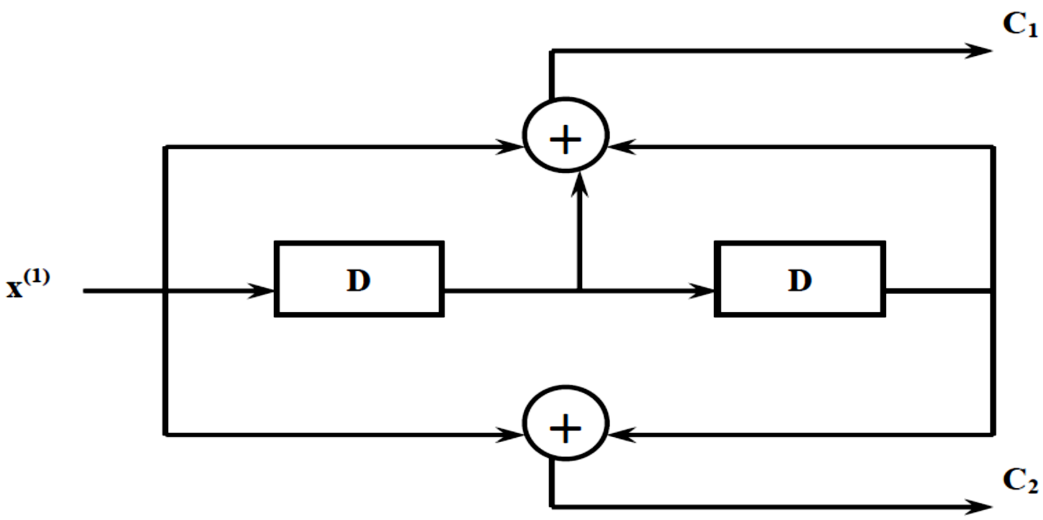


Fig 4.1: A convolutional encoder with k=1, n=2 and r=1/2

The shift register of the encoder is initialized to all-zero-state before encoding operation starts. It is easy to verify that encoded sequence is 00 11 10 00 01 ….for an input message sequence of 01011….

The operation of a convolutional encoder can be explained in several but equivalent ways such as, by

a) State diagram representation,

b) Tree diagram representation

c) Trellis diagram representation.

4.4 State Diagram Representation:

In Figure 4.1, the number of combinations of bits in the registers (D) are called the states of the code and are defined by 2^K. The C (1,3,2) in our example has $2^2 = 4$ states which are 00,01,10,11. Number of states are independent of rate of the code. The transition of an encoder from one state to another, as caused by input bits, is depicted in the state diagram Fig. 4.2. A new input bit causes a transition from one state to another. The path information between the states, denoted as k/c1c2, represents input information bit 'k' and the corresponding output bits (c1c2). Again, it is not difficult to verify from the state diagram that an input information sequence k = (1011) generates an encoded sequence c = (11, 10, 00, 01).

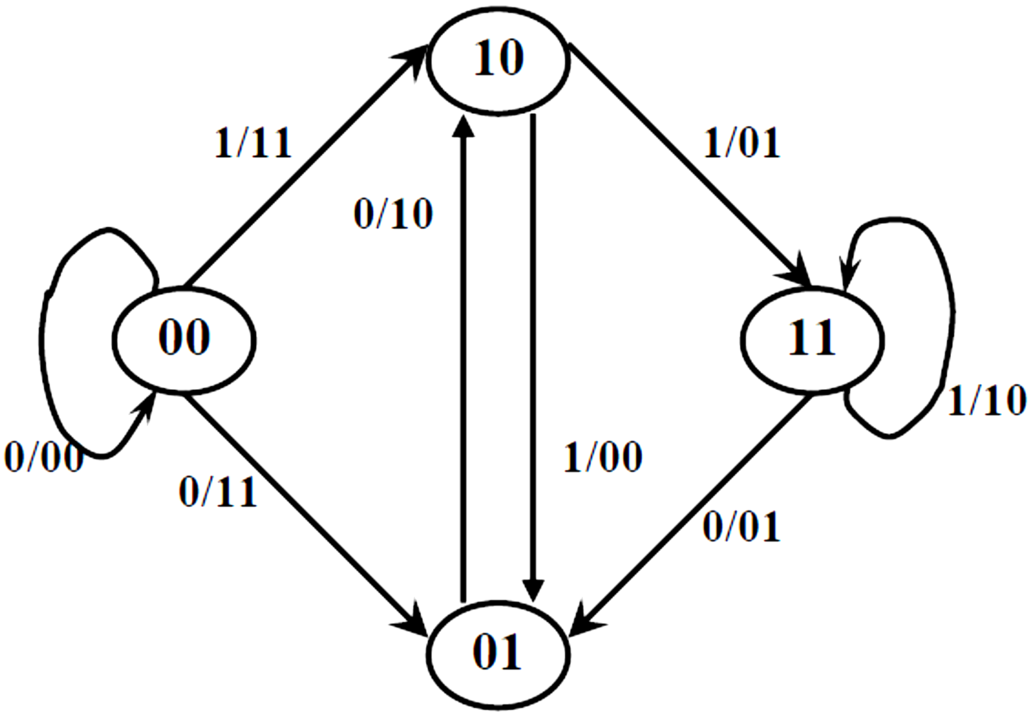


Fig.4.2 State diagram representation for the encoder

4.5 Tree Diagram Representation

The tree diagram representation shows all possible information and encoded sequences for the convolutional encoder. Fig. 4.3 shows the tree diagram for the encoder in Fig. 4.3. The encoded bits are labeled on the branches of the tree. Given an input sequence, the encoded sequence can be directly read from the tree. As an example, an input sequence (1011) results in the encoded sequence (11, 10, 00, 01).

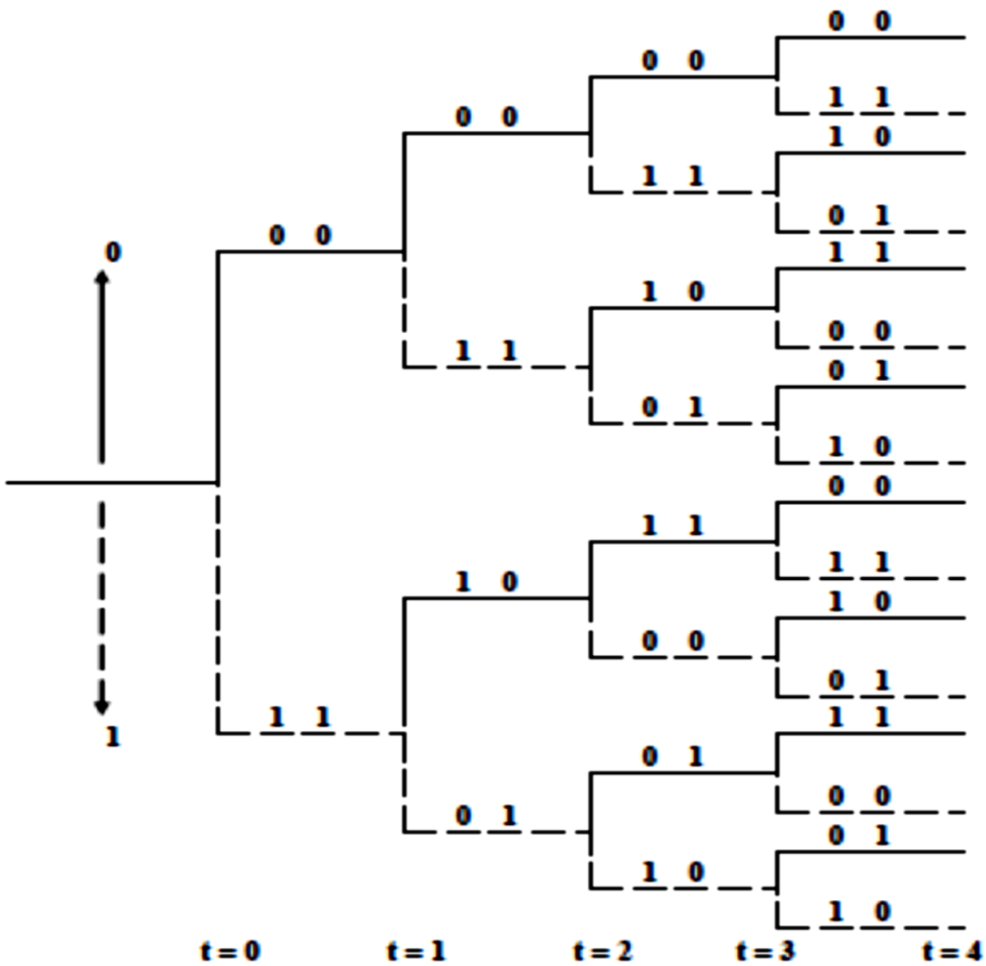


Fig. 4.3: A tree diagram for the encoder

4.6 Trellis Diagram Representation

The trellis diagram of a convolutional code is obtained from its state diagram. All state transitions at each time step are explicitly shown in the diagram to retain the time dimension, as is present in the corresponding tree diagram. Usually, supporting descriptions on state transitions, corresponding input and output bits etc. are labeled in the trellis diagram. It is interesting to note that the trellis diagram, which describes the operation of the encoder, is very convenient for describing the behavior of the corresponding decoder, especially when the famous 'Viterbi Algorithm (VA)' is followed.

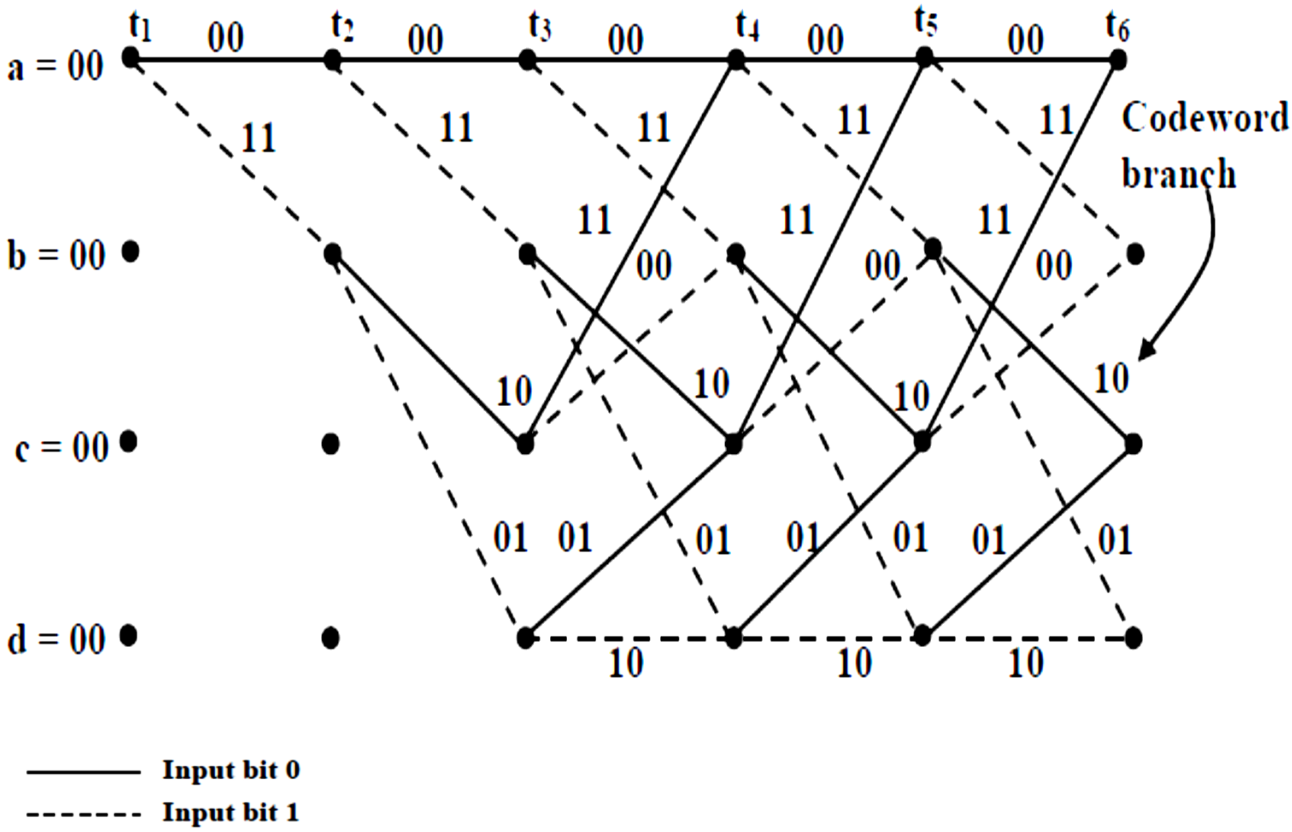


Fig 4.4 Trellis Diagram

4.7 Viterbi decoding

This decoder uses Viterbi algorithm for decoding a bit stream that has been encoded using a convolutional code. It was developed by Andrew J. Viterbi and was published in an IEEE transaction in 1967 [6]. The use of the Viterbi algorithm for decoding covolutionally coded data has become very popular since then. According to [1], the Viterbi algorithm consists of three major parts:

1) Branch metric calculation.
2) Path metric calculation.
3) Trace back operation.

4.7.1 Branch Metric Calculation:

- The pair of received bits (for (n=2)), are compared with the corresponding branches in the trellis and the distance metrics are calculated. For hard decision decoding, Hamming distances are calculated. Suppose if the received pair of bits are '11′ and the hamming distance to {'00′,'01′,'10′,'11′} outputs of the trellis are 2,1,1,0 respectively.
- Branch metric is sum of path metric of the previous sate and hamming distance required for the transition

4.7.2 Hamming Distance calculation

- For decoding consider two Received coded bits at a time y_i and compute the hamming distance between all possible combinations of two bits. The number of differing bits can be computed by XOR-ing y_i with 00,01,10,11 and then counting the number 1's
- $hd_i,00$ is the number of 1's in $00 \otimes y_i$
- $hd_i,01$ is the number of 1's in $01 \otimes y_i$
- $hd_i,10$ is the number of 1's in $10 \otimes y_i$
- $hd_i,11$ is the number of 1's in $11 \otimes y_i$

4.7.3 Path Metric Calculation:

- Path metrics are calculated using a procedure called ACS (Add-Compare-Select). This procedure is repeated for every encoder state.
- From the two available branch matrices the one with minimum metric value is chosen. This operation is called as add compare and select (ACS) unit

- Add – for a given state, we know two states on the previous step which can move to this state, and the output bit pairs that correspond to these transitions. To calculate new path metrics, we add the previous path metrics with the corresponding branch metrics.

State 00 can be reached from two branches

- Sate 00 with output 00. The branch metric for this transition is

$$bm_{i,00,00} = pm_{i-1,00} + hd_{i,}00$$

- State 01 with output 11. The branch metrics for this transition is

$$bm_{i,00,01} = pm_{i-1,01} + hd_{i,}11$$

The path metric for state 00 is chosen based which is minimum out of two

$$pm_{i,00} = \min(bm_{i,00,00}\ , bm_{i,00,01})$$

The survivor path for 00 is stored in survivor path metric

$$sv_{i,00} \quad = 00, bm_{i,00,00} < bm_{i,00,01}$$

$$= 01, bm_{i,00,00} > bm_{i,00,01}$$

State 01 can be reached from two branches:

State 10 with output 10. The branch metric for this transition is

$$bm_{i,01,10} = pm_{i-1,10} + hd_{i,}10$$

State 11 with output 01. The branch metric for this transition is

$$bm_{i,01,11} = pm_{i-1,11} + hd_{i,}01$$

The path metric for state 01 is chosen based which is minimum out of the two

$$pm_{i,01} = \min(bm_{i,01,10}, bm_{i,01,11})$$

The survivor path for state 01 is stored in survivor path metric

$$\begin{aligned} sv_{i,01} &= 10, bm_{i,01,10} < bm_{i,01,11} \\ &= 11, bm_{i,01,10} > bm_{i,01,11} \end{aligned}$$

Each state can be reached from two possible states (show by red and blue lines)

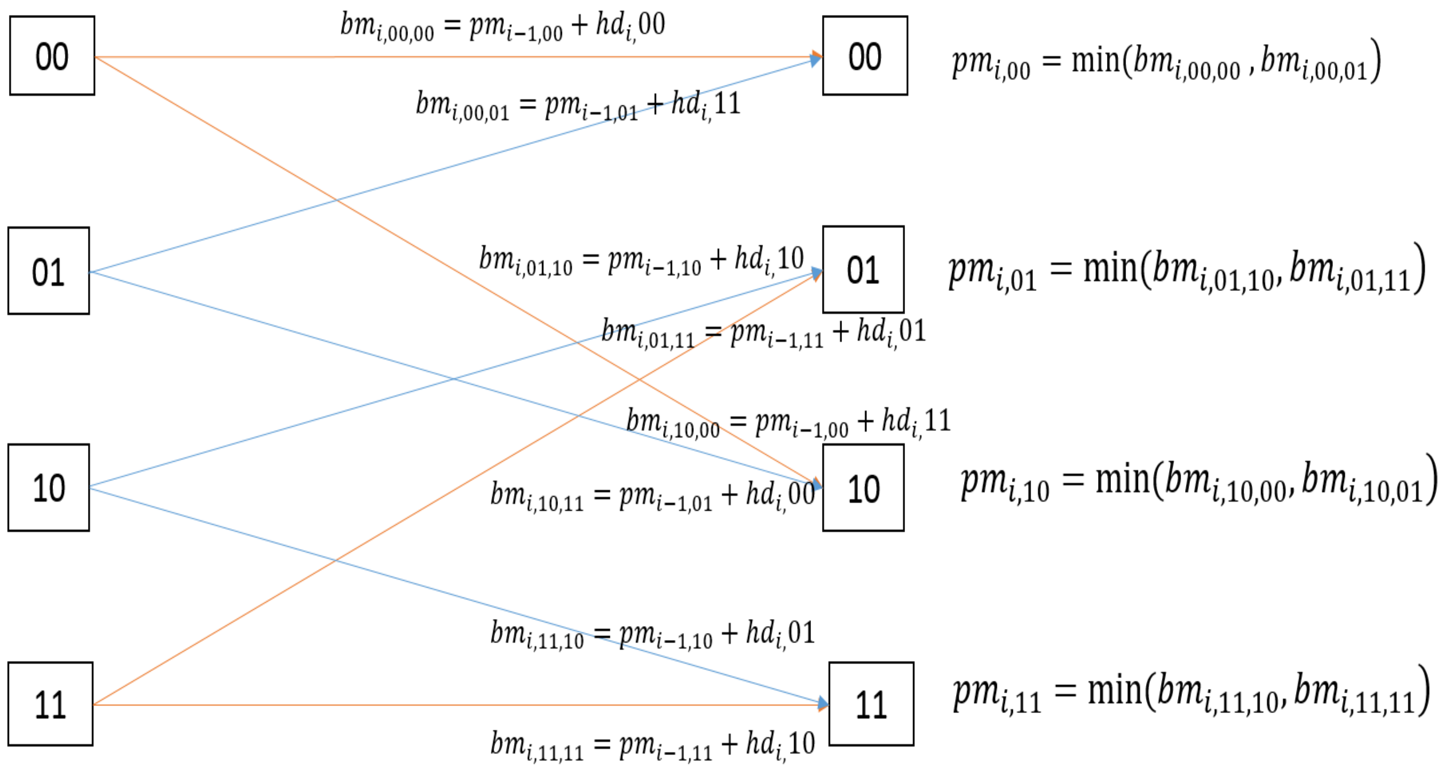


Fig: 4.5.Branch metric and path metric computation Representation

State 10 can be reached from two branches:

State 00 with output 11. The branch metric for this transition is

$$bm_{i,10,00} = pm_{i-1,00} + hd_{i,}11$$

State 01 with output 00. The branch metric for this transition is

$$bm_{i,10,11} = pm_{i-1,01} + hd_{i,}00$$

The path metric for state 10 is chosen based which is minimum out of the two

$$pm_{i,10} = \min(bm_{i,10,00}, bm_{i,10,01})$$

The survivor path for state 01 is stored in survivor path metric

$$sv_{i,10} = 00, bm_{i,10,00} < bm_{i,10,01}$$

$$= 01, bm_{i,10,00} > bm_{i,10,01}$$

State 11 can be reached from two branches:

State 10 with output 01. The branch metric for this transition is

$$bm_{i,11,10} = pm_{i-1,10} + hd_{i,}01$$

State 11 with output 10. The branch metric for this transition is

$$bm_{i,11,11} = pm_{i-1,11} + hd_{i,}10$$

The path metric for state 01 is chosen based which is minimum out of the two

$$pm_{i,11} = \min(bm_{i,11,10}, bm_{i,11,11})$$

The survivor path for state 01 is stored in survivor path metric

$$sv_{i,11} = 10, bm_{i,11,10} < bm_{i,11,11}$$

$$= 11, bm_{i,11,10} > bm_{i,11,11}$$

4.8 Trace back operation:

- Once the survivor path is computed $\left(\frac{N}{2} + K - 1\right)$ times, the decoding algorithm can start trying to estimate the input sequence
- So start from the last survivor path at index $\left(\frac{N}{2} + K - 1\right)$ for sate 00. From the survivor path find the previous sate corresponding to the current state.

- From the knowledge of current state and previous state the input sate can be determined .continue tracking back through the survivor path and estimate the input sequence till index=1

Current state	00	01	10	11
00	0	0	X	X
01	X	X	0	0
10	1	1	X	X
11	X	X	1	1

Table: 4.1. Trace back table

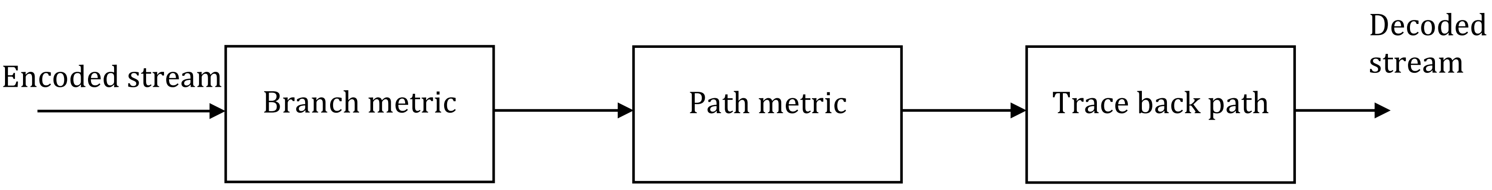


Fig 4.6 Viterbi decoder data Flow

Chapter 5

Results and conclusions

The simulations are carried out using MATLAB software. The performance is simulated and evaluated for BPSK systems. Based on data generated by computer simulation of BPSK modulation techniques for BER calculation the following results are obtained.

The downward slope of BER curve of coded signal is sharper than uncoded signal in the simulated curve. From the cross-sectional point, the coded signal performance is better than uncoded signal. From this simulation it proves that if the data signal is transmitted using convolutional code, the system performance is clearly improved.

Results:

- The simulation result of uncoded signal is evaluated on BER vs. SNR for AWGN channel and the BERs are obtained by varying the values of SNR in the range of 0 to 10 dB

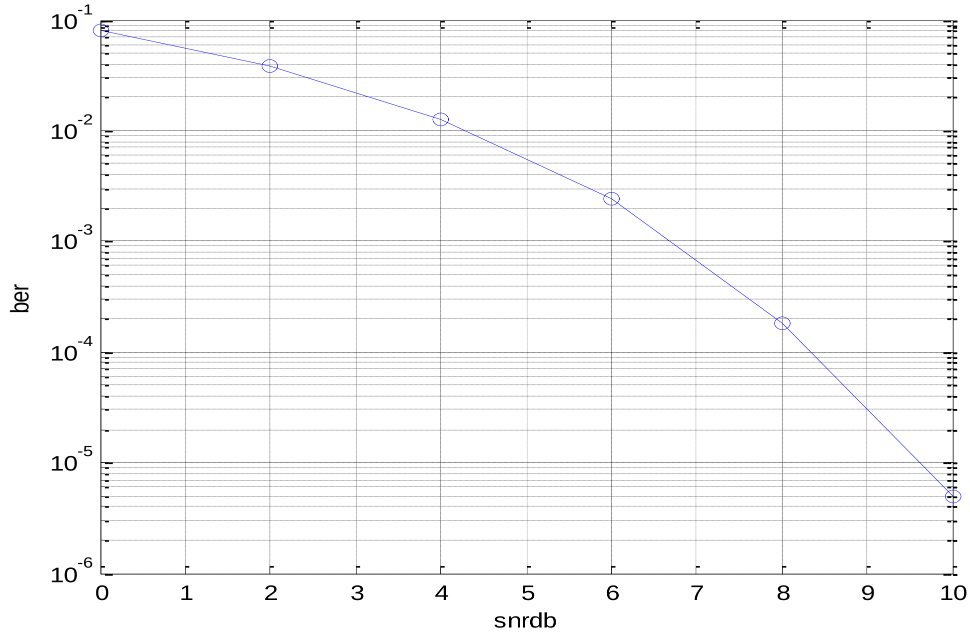


Fig 5.1 BER versus SNR for AWGN uncoded.

- ➢ BER versus SNR over AWGN fading channel for BPSK modulation without convolutional coding technique and With Convolutional codes.

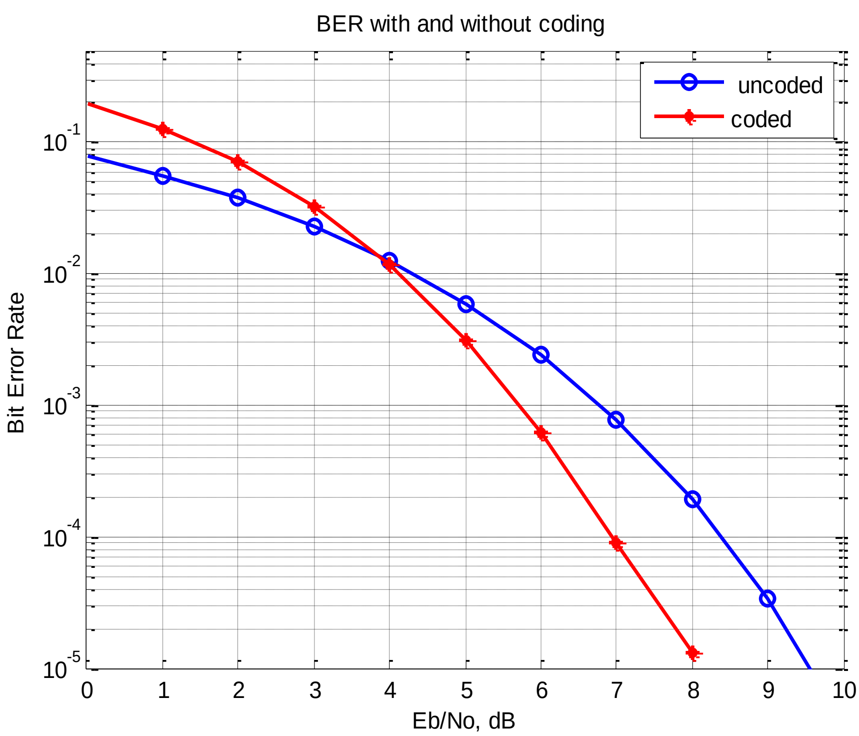


Fig 5.2 BER versus SNR for AWGN uncoded and coded.

➢ The simulation result of uncoded signal is evaluated on BER vs. SNR for Rayleigh fading channel the BERs are obtained by varying the values of SNR in the range of 0 to 10 dB

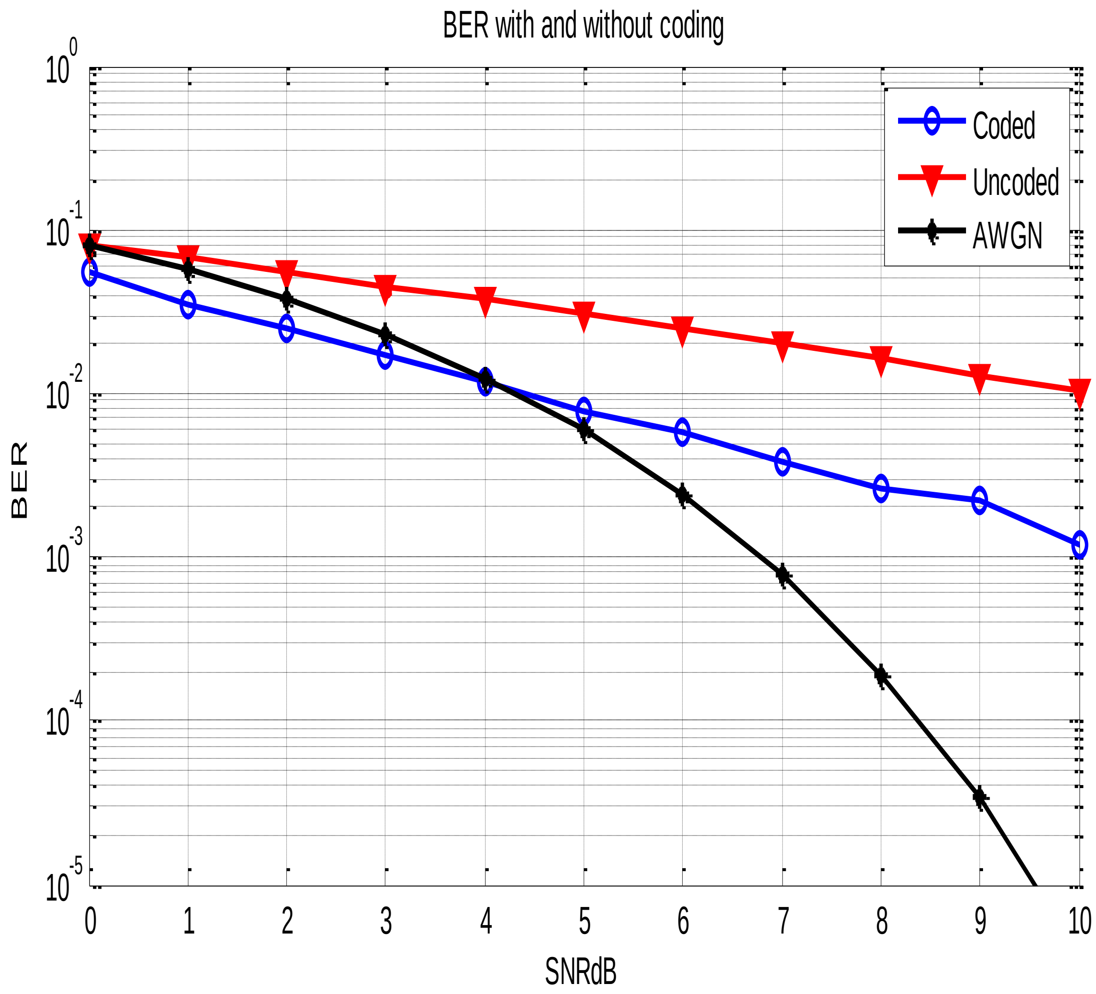


Fig 5.3 BER versus SNR slow Rayleigh fading.

Comparison simulated curve with respect to BER versus SNR over Rayleigh fading channel between the coded and uncoded signal. With uncoded signal of AWGN channel.

Conclusion:

In Digital Communication System Convolution codes are commonly employed for Detection and Correcting the errors in received signal information bits. Perhaps it is widely used coding technique in fading channels in communication systems. In my thesis, the BER performance is obtained for BPSK in AWGN and Rayleigh fading multipath channels. For simulation I have used Convolutional encoder as channel encoder and Viterbi Decoder for decoding the convolutional codes at the receiver. It clearly shows that the performance of the coded signal is better than the uncoded signal in the presence of multipath fading channels.

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