CHARACTERISTICS OF DIFFERENT NODES OF PROPAGATION CHANNEL AND ITS EFFECT ON DIFFERENT MODULATION TECHNIQUES

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A project report submitted in partial fulfilment of the requirements for the Degree of
Bachelor of Technology
(Electronics Engineering)

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Under the guidance of

Prof. RANJAN MISHRA

Approved

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Dean

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CERTIFICATE

This is to certify that the work contained in this report titled ***"CHARACTERISTICS OF DIFFERENT NODES OF PROPAGATION CHANNEL AND ITS EFFECT ON DIFFERENT MODULATION TECHNIQUES''*** has been carried out by ***ASHOK KUMAR and GAURAV SAINI*** under my/our supervision and has not been submitted elsewhere for a degree.

Date Date

Acknowledgement

On the submission of our Thesis report of **"CHARACTERISTICS OF DIFFERENT NODES OF PROPAGATION CHANNEL AND ITS EFFECT ON DIFFERENT MODULATION TECHNIQUES''**, we would like to express our gratitude & sincere thanks to our mentor **Prof. RANJAN MISHRA**, for his constant motivation and support during the course of our project work. We would like to sincerely thank him for his support and guidance in providing us a clear understanding about the intricate details of the topic and strengthening the foundation throughout the work.

Finally, a word of thanks to all the faculty members of Department of Electronics and Communication Engineering, for their help and support in various ways for overall completion of our thesis.

Abstract

Communication Engineering has become the most vital field of Engineering in our daily life. Digital communication basically involves transfers of information from one place to another or from one point of time to another using different technique but during transmission error may be introduced by the channel which makes data degradable for user. Due to channel induced SNR, the received signal may not be replicated the same transmitted signal.

Modulation is the process of the transfer of data over a medium. Generally a digital communication system is used to transport digital information between two or more nodes. In radio communication this is usually done by adjusting a physical characteristic of a sine carrier, the frequency, phase or amplitude. This is performed in real system with a modulator at the transmitter end to detect the resultant modulation on receiver end. Hence, modulation can be objectively defined as the process of transforming information so that it can be successfully sent through a medium.

Most communication systems fall into one of three categories: bandwidth efficient, cost efficient, or power efficient. Bandwidth efficient describes the ability of a modulation scheme to accommodate information within a limited bandwidth. Power efficient describes the capability of the system to reliably send data at the lowest practical level. In most of the systems, there is a high priority on bandwidth efficiency. The parameter to be optimized depends of the particular system.

MATALB simulation is done on different types of modulation scheme to understand the performance and analysis of digital modulation technique. This thesis deals with the current digital modulation techniques used in industry. This report also justifies the qualitative and quantitative selection of one modulation technique over the other. All the experiment, and related data collected were obtained using MATLAB and SIMULINK. Understanding of different techniques is done using “EMONA KIT” available in digital Communication Lab.

CONTENT

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CHAPTER 1

INTRODUCTION

Modulation is the process of facilitating the transfer of information over a medium. Generally, a digital communication system is used to transfer digital information between two or more ends. In Digital communications this is usually achieved by adjusting a physical characteristic of a sine carrier, the frequency, phase, amplitude or a combination of them. This is performed in real systems with a modulator at the transmitting end to impose the physical change to the carrier and a demodulator at the receiving end to detect the resultant modulation on receiver end. Hence, modulation can be defined as the process of transforming data, so that it can be successfully sent through a medium.

Most communications systems fall into one of three categories: bandwidth efficiency, power efficiency, or cost efficiency. Bandwidth efficiency explains the capability of a modulation scheme to accommodate data within a limited bandwidth. Power efficiency explains the capability of the system to reliably send information at the lowest real time power level. In most of the systems, there is a higher priority on the efficiency of bandwidth. The bandwidth parameter, which has to be optimized rely on the availability of the particular system.

For designers of digital terrestrial radio waves, their higher priority is better bandwidth efficiency with low BER. They have a lot of power available and are not concerned with efficiency of power, they are not generally concerned with the cost of receiver or complexity because they do not have to build complex of them. But, engineers of hand-held cellular phones put a higher priority on power efficiency because these phones need to run by a battery. Cost is also considered because mobile phones must be low-price to attract more users. Likely, these systems decrease some bandwidth efficiency to get power and cost efficiency.

Every time one of these parameters (bandwidth, power or cost) is increased, another one reduces, or becomes more difficult or does not perform well in a harsh environment. For example, in the case of microwaves, cost is the main system priority. Less-cost radios will always be loved by users. In earlier days, it was easy to make a radio low-cost by reducing power and bandwidth efficiency. This is not possible now. The wave spectrum is very essential and operators who could not use the spectrum efficiently could lose their existing licenses or lose out in the competition for new operators. These are the traditions that must be considered in digital communications design.

1.1 Brief Description.

The system is based upon some major devices

1. **The Emona Telecoms-Trainer 101 (ETT101) Kit.**

   It consisted of Basic Analog Communications AM, FM, DSB, SSB, PAM, TDM, PWM, Super heterodyne, Speech in communication, PLL, QAM, SNR Principles. Digital Communication Spectrum, Line Coding, Delta Modulation, Noise Generation, SNR Concept.

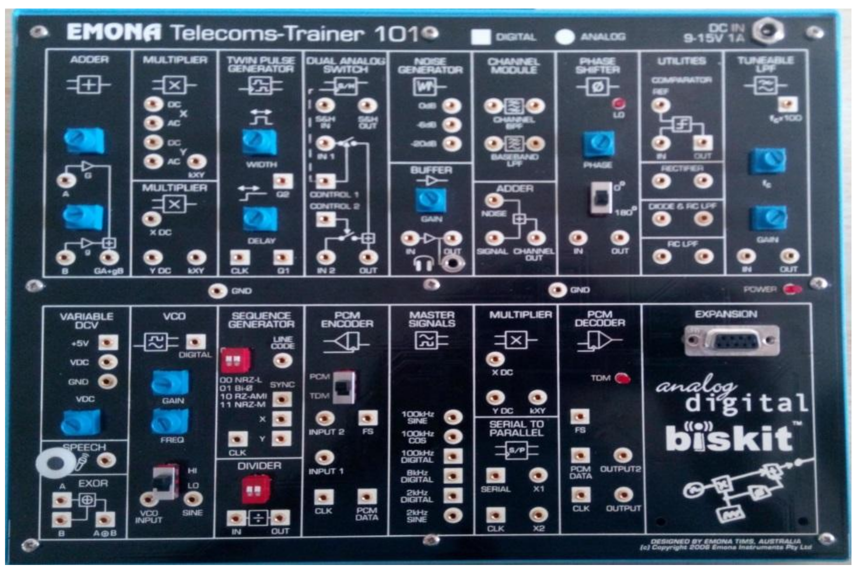

Fig. 1.1 EMONA KIT

Oscilloscope:

An oscilloscope's primary function is to provide a graph of a signal's voltage and time. This is used for finding such things as clock frequency, duty cycle of pulse-width-modulated signal, propagation delay, or signal rise time and fall time. It can also tell you to the presence of glitches in your logic switches.

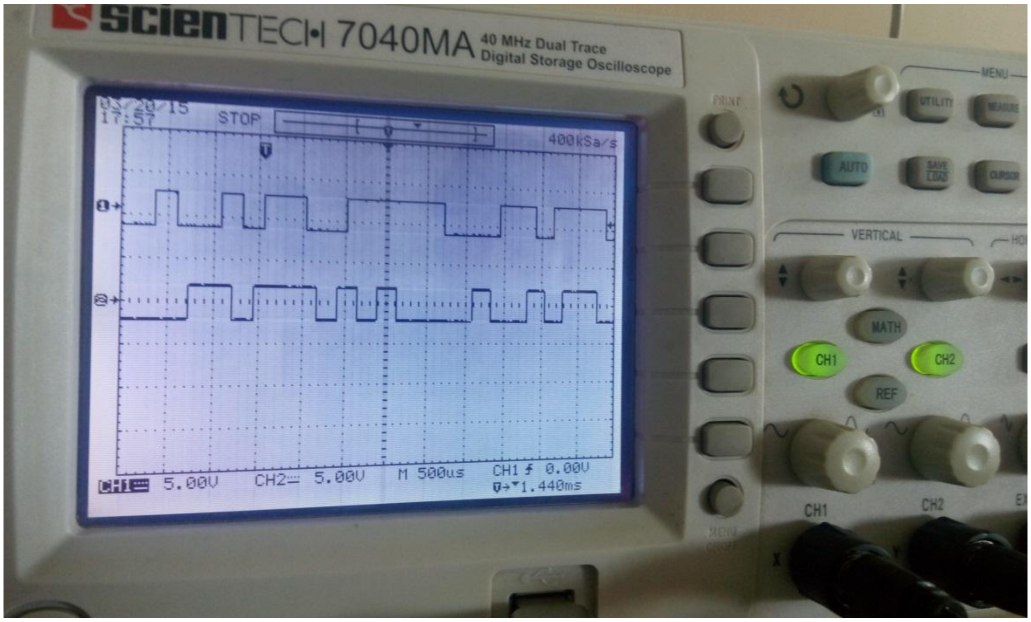

Fig. 1.2 Oscilloscope

1.2 System block diagram:

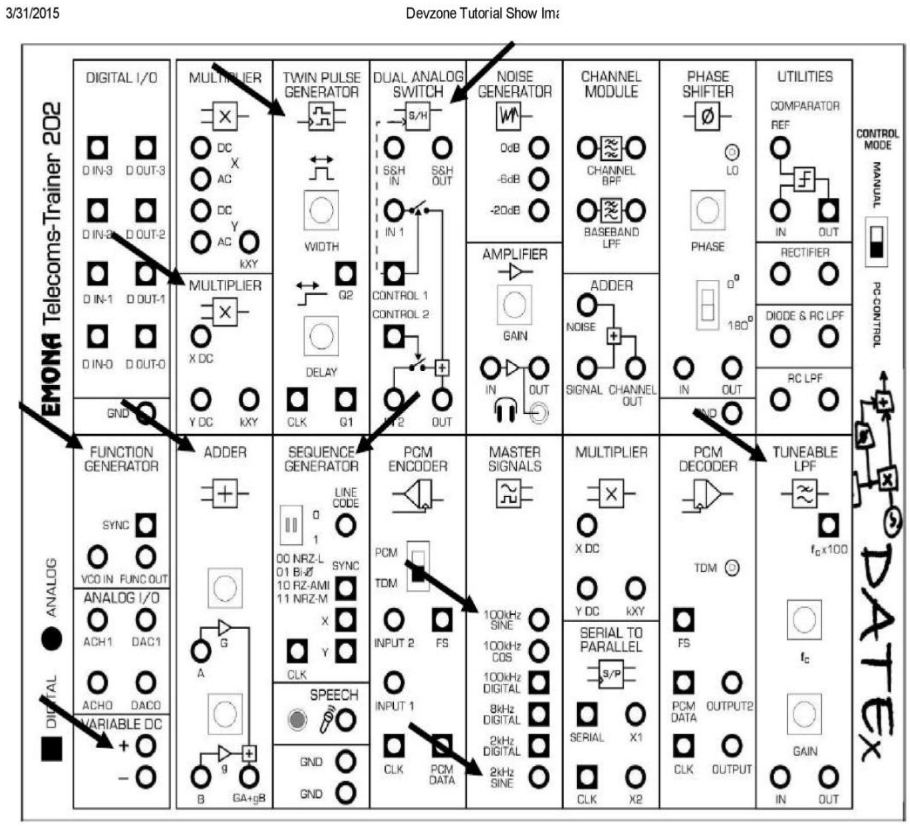

Fig. 1.3 Block Diagram

CHAPTER-2

LITERATURE SURVEY

Usage of the benefits of electrical communications in general and digital communications in particular, is an integral part of our daily life now. A large number of applications due to R&D in digital communications have already started influencing our daily life activities directly or indirectly. Popularity of the Web and T.V. are only two of the most important examples to prove this point. Actually, it may not be an overstatement now that ‘Data highways’ are considered as essential part of national infrastructure in the march of a modern world. It is, however, important to elaborate that isolated developments only in the field of electrical communications have not caused this process. Unbelievable progresses and technical achievements in several related fields in electronics engineering and computer engineering have actually made applications of several principles and theories of communication engineering feasible for implementation and usage

2.1 Research works:

International Journal of Engineering and Innovative Technology (IJEIT) Volume 2, Issue 2, August 2012

The migration to 4G networks will bring a new level of expectation to wireless communications. As after digital wireless revolution made mobile phones available for everyone, the higher speeds and packet delivery of 4G networks will make high quality multimedia available everywhere. The key to achieving this higher level of service delivery is a new air interface. Orthogonal Frequency Division Multiplexing (OFDM) is an alternative wireless modulation technology to CDMA. OFDM is a digital modulation and multiplexing technique. In this paper, we have discussed various digital modulation techniques such as BPSK (2bits), QPSK (4 bits), QAM, 16 QAM and 64 QAM. We have designed simulation environment in MATLAB with various configurations of OFDM technique. The main objective of our work is to measure Bit Error Rate with different modulation schemes and come to the best configuration to achieve better utilization of bandwidth. We have studied existing configurations with analog and digital modulation techniques and compared the results. The driving force behind the need to satisfy this requirement is the explosion in mobile telephone, Internet and multimedia services coupled with a limited radio spectrum.

International Journal of Engineering and Innovative Technology (IJEIT) Volume 2, Issue 7, July 2012, " Performance Investigation for Different Modulation Techniques"

The technology needed to tackle the challenges to make services like high speed data, video and multimedia traffic as well as voice signals available is popularly known as the Third Generation (3G) Cellular Systems. Downlink transmission (base station to mobile terminal) using high data rate Mary Quadrature Amplitude Modulation (QAM), Quadrature Phase Shift Keying (QPSK) and Binary Phase Shift Keying (BPSK) modulation schemes are considered in a Wideband-Code Division Multiple Access (W-CDMA) system. The performances of these modulation techniques are evaluated when the system is subjected to a number of users as well as noise and interference in the channel. Additive White Noise Gaussian (AWGN) and multipath Rayleigh fading are considered in the channel. Computer simulation tool, MATLAB, is used throughout the research to evaluate Bit-Error-Rate (BER) for W-CDMA system models**.**

International Journal of Computer Networks & Communications (IJCNC) Vol.6, No.2, March 2014

To achieve better calculative performance in optical fiber communication and for simplicity of implementation different digital modulation, detection and multiplexing techniques are used. These techniques maximize the spectral efficiency. This paper reviews a tabular comparative analysis with 3D graphical representation for different optical digital modulation formats and multiplexing techniques within and beyond 400 Gb/s. In this particular article we survey about different parameters related to digital fiber optic communication

CHAPTER-3

MODULATION TECHNIQUES:

3. Digital Modulation Techniques

In digital communications, the modulation process corresponds to switching or keying the amplitude, frequency, or phase of a sine carrier wave according to incoming digital data.

Three basic

1. Digital modulation techniques

2. Amplitude-shift keying (ASK) -special case of AM

3. Frequency-shift keying (FSK) -special case of

3.1.1. ASK [Amplitude Shift Keying]:

In a binary ASK system symbol '1' and '0' are transmitted as

$$S(t) = \sqrt{\frac{2E_b}{T_b}} Cos 2\pi f t \quad \text{for symbol 1}$$

$$S_2(t) = 0 \text{ for symbol 0}$$

Amplitude-shift keying (ASK) is a form of amplitude modulation that represents digital data as variations in the amplitude of a carrier wave. In an ASK system, the binary symbol 1 is represented by transmitting a fixed-amplitude carrier wave and fixed frequency for a bit duration of T seconds. If the signal value is 1 then the carrier signal will be transmitted; otherwise, a signal value of 0 will be transmitted.

Any digital modulation scheme uses a finite number of distinct signals to represent digital data. ASK uses a finite number of amplitudes, each assigned a unique pattern of binary digits. Usually, each amplitude encodes an equal number of bits. Each pattern of bits forms the symbol that is represented by the particular amplitude. The demodulator, which is designed specifically for the symbol-set used by the modulator, determines the amplitude of the received signal and maps it back to the symbol it represents, thus recovering the original data. Frequency and phase of the carrier are kept constant.

Like AM, ASK is also linear and sensitive to atmospheric noise, distortions, propagation conditions on different routes in PSTN, etc. Both ASK modulation and demodulation processes are relatively inexpensive. The ASK technique is also commonly used to transmit digital data over optical fiber. For LED transmitters, binary 1 is represented by a short pulse of light and binary 0 by the absence of light. Laser transmitters normally have a fixed "bias" current that causes the device to emit a low light level. This low level represents binary 0, while a higher-amplitude light wave represents binary 1.

The simplest and most common form of ASK operates as a switch, using the presence of a carrier wave to indicate a binary one and its absence to indicate a binary zero. This type of modulation is called on-off keying (OOK), and is used at radio frequencies to transmit Morse code (referred to as continuous wave operation). When AM is used for multiplexing digital data, it is known as amplitude shift keying (ASK). Other names include: on-off keying, continuous wave and interrupted continuous wave. An essential part of electronic communications and telecommunications is the ability to share the channel. This is true regardless of whether the channel is copper wire, optical fiber or free-space. If it's not shared then there can only ever be one person transmitting on it at a time. Think about the implications of this for a moment. Without the ability to share, there could only be one radio or TV station in each area. Only one mobile phone owner could use their phone in each cell at any one time. And there would only be the same number of phone calls between any two cities as the number of copper wires or optical fibers that connected them.

So sharing the channel is essential and there are several methods of doing so. One is called time division multiplexing (TDM) and involves giving the users exclusive access to the channel for short periods of time. On the face of it, this type of sharing might seem impractical. Imagine giving all mobile phone users in a cell just a minute or so to make their call then having to wait until their turn comes around again. However, TDM works well when the access time is extremely short (less than a second) and the rate of the sharing is fast. This allows multiple users to appear to have access all at the same time.

Notice that the ASK signal's upper and lower limits (the envelopes) are the same shape as the data stream (though the lower envelope is inverted). This is simultaneously an advantage and a disadvantage of ASK. Recovery of the data stream can be implemented using a simple envelope detector. However, noise on the channel can change the envelopes' shape enough for the receiver to interpret the logic levels incorrectly causing errors i.e. analog AM communications have the same problem and the errors are heard as a hiss, crackles and pops. Shows what an ASK signal looks like time-coincident with the digital signal that has been used to generate it.

Here you'll examine the operation of an alternative method that involves using the digital signal to switch the carrier's connection to the channel on and off.

The simplest and most common form of amplitude Shift Keying operates as a switch using the presence of a carrier wave to indicate a binary one and its carrier wave to indicate a binary zero. This type of modulation is called On-Off keying.

In ASK logic levels are represented by different amplitudes of signals. Usually one amplitude is zero for logic digital zero while logic 1 represented by the actual amplitude of some sine wave.

The transmission of digital signals is increasing at a rapid rate. Low-frequency analogue signals are often converted to digital format (PAM) before transmission. The source signals are generally referred to as baseband signals. Of course, we can send analogue and digital signals directly over a medium. From electro-magnetic theory, for efficient radiation of electrical energy from an antenna it must be at least in the order of magnitude of a wavelength in size; $c = f\lambda$, where c is the velocity of light, f is the signal frequency and λ is the wavelength. For a 1 kHz audio signal, the wavelength is 300 km. An antenna of this size is not practical for efficient transmission. The low-frequency signal is often frequency-translated to a higher frequency range for efficient transmission. The process is called modulation. The use of a higher frequency range reduces antenna size. In the modulation process, the baseband signals constitute the modulating signal and the high-frequency carrier signal is a sinusoidal waveform. There are three basic ways of modulating a sine wave carrier. For binary digital modulation, they are called binary amplitude-shift keying (BASK), binary frequency-shift keying (BFSK) and binary phase shift keying (BPSK). Modulation also leads to the possibility of frequency multiplexing. In a frequency-multiplexed system, individual signals are transmitted over adjacent, non-overlapping frequency bands. They are therefore transmitted in parallel and simultaneously in time. If we operate at higher carrier frequencies, more bandwidth is available for frequency-multiplexing more signals

3.1.2. FSK [Frequency Shift Keying]:

In a binary FSK system symbol '1' and '0' are transmitted as

$$S(t) = \sqrt{\frac{2E_b}{T_b}}\, Cos 2\pi f_1 t \quad \text{for symbol 1}$$

$$S_2(t) = \sqrt{\frac{2E_b}{T_b}}\, Cos 2\pi f_2 t \quad \text{for symbol 0}$$

The FSK signal switches between two frequencies. The frequency of the signal that corresponds with logic-Os in the digital data called the space frequency is usually lower than the modulator's nominal carrier frequency. The frequency of the signal that corresponds with logic-ls in the digital data called the mark frequency is usually higher than the modulator's nominal carrier frequency. The modulator doesn't output a signal at the carrier frequency, hence the reference here to it as being the "nominal" carrier frequency.

FSK generation can be handled by conventional FM modulator circuits and the voltage controlled oscillator (VCO) is commonly used. Similarly, FSK demodulation can be and led by conventional FM demodulators such as the zero crossing detector and the phase-locked loop. Alternatively, if the FSK signal is passed through a sufficiently selective filter, the two sine waves that make it up can be individually picked out. Considered on their own, each signal is an ASK signal and so the data can be recovered by passing either one of them through an envelope detector. As its name suggests, a frequency shift keyed transmitter has its frequency shifted by the message. Although there could be more than two frequencies involved in an FSK signal, in this experiment the message will be a binary bit stream, and so only two frequencies will be involved. The word 'keyed' suggests that the message is of the 'on-off' (mark-space) variety, such as one (historically) generated by a mores key, or more likely in the present context, a binary sequence between two fixed-frequency oscillators to produce the mark and space frequencies. While this method is sometimes used, the constraint that transitions from mark to space and vice versa must be phase continuous ("glitch" free) requires that the shift and keying rate be interrelated. A synchronous FSK signal which has a shift in Hertz equal to an exact integral multiple (n = 1, 2,...) of the keying rate in bauds, is the most common form of coherent FSK. Coherent FSK is capable of superior error performance but non-coherent FSK is simpler to generate and is used for the majority of FSK transmissions. Non-coherent FSK has no special phase relationship between consecutive elements, and, in general, the phase varies randomly. Many different coding schemes are used to transmit data with FSK. They can be classified into two major groups: synchronous and asynchronous. Synchronous transmissions have mark-to-space and space-to-mark trans element. The length of the latch bit may be very long between characters, especially in the case of manually generated characters where the operator types more slowly than the system can transmit characters. The non-integer minimum latch element length of 142 elements and the random nature of manual character generation emphasizes the asynchronous nature of this scheme. A common synchronous system uses Moore ARQ coding. The Moore code is a 7-bit-percharacter code with no start or stop elements. Bit synchronization is maintained by using a reference clock which tracks the keying speed of the received signal. Character synchronization is maintained by sending periodic “idle” or “dummy” characters between valid data characters. FREQUENCY DIVISION MULTIPLEX (FDM) Several FSK signals can be transmitted simultaneously in a given frequency band by assigning different center frequencies to each of the FSK signals. This method of simultaneous transmission is called FDM. In the radio spectrum several audio FSK signals are often combined for transmission by a single-sideband transmitter. This form of FDM is often called Voice Frequency Telegraph (VFT). To minimize bandwidth, the individual FSK channels usually have “narrow” shifts of between 50 and 200 Hertz. A typical FDM system. HF radio systems usually transmit 16 tone channels, but 24 or more tone channels per 3-kHz sideband are possible. Telephone standard is typically 12 or 24 tone channels per 3-kHz sideband (voice band). HF radio is subject to random multi-path signal fading. The relatively narrow-band nature of this fading phenomenon causes only one or two tone channels to be severely affected at any moment. As a defense against fading, it is common practice to run duplicate data in tone channels in the same FDM group. If the duplicate channels are separated by 12 kHz in synchronism with a reference clock. Asynchronous signals do not require a reference clock but instead rely on special bit patterns to control timing during decoding. Compares synchronous and asynchronous keying. A very common asynchronous coding system is the 5-bit Baud code with leading start (release) and trailing stop (latch) elements. Originally designed for use with mechanical teleprinters, the system is “latched” until a “release” element is received, causing the printer to interpret the next 5 element intervals as code bits. The binary values of the 5 bits correspond to a particular character. In the two character patterns correspond to the characters “C” and “W” respectively. The 5 “information” bits are immediately followed by a stop or “latch” bit lasting a minimum of 1.42 element lengths. The latch bit stops the printing decoder until the decoder is again started by the next “release” element. The length of the latch bit may be very long between characters, especially in the case of manually generated characters where the operator types more slowly than the system can transmit characters. The non-integer minimum latch element length of 142 elements and the random nature of manual character generation emphasizes the asynchronous nature of this scheme. A common synchronous system uses Moore ARQ coding. The Moore code is a 7-bit-percharacter code with no start or stop elements. Bit synchronization is maintained by using a reference clock which tracks the keying speed of the received signal. Character synchronization is maintained by sending periodic “idle” or “dummy” characters between valid data characters. 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As a defense against fading, it is common practice to run duplicate data in tone channels in the same FDM group. If the duplicate channels are separated by 1 kHz.

3.1.3 PSK [Phase Shift Keying]:

In a binary PSK system the pair of signals $S_1(t)$ and $S_2(t)$ are used to represent binary symbol ‘1’ and ‘0’ respectively.

$$S1(t) = \sqrt{2Eb/Tb}\cos 2\Pi fct$$

$$S2(t) = \sqrt{2E/Tb}\cos 2\Pi fct$$

Phase shift keying, PSK, is widely used these days within a whole raft of radio communications systems. It is particularly well suited to the growing area of data communications. PSK, phase shift keying enables data to be carried on a radio communications signal in a more efficient manner than Frequency Shift Keying, FSK, and some other forms of modulation.

With more forms of communications transferring from analogue formats to digital formats, data communications is growing in importance, and along with it the various forms of modulation that can be used to carry data.

There are several flavours of phase shift keying, PSK that are available for use. Each form has its own advantages and disadvantages, and a choice of the optimum format has to be made for each radio communications system that is designed. To make the right choice it is necessary to have a knowledge and understanding of the way in which PSK works. Phase Shift Keying, PSK, basics Like any form of shift keying, there are defined states or points that are used for signalling the data bits. The basic form of binary phase shift keying is known as Binary Phase Shift Keying (BPSK) or it is occasionally called Phase Reversal Keying (PRK). A digital signal alternating between +1 and -1 (or 1 and 0) will create phase reversals, i.e. 180 degree phase shifts as the data shifts state. The problem with phase shift keying is that the receiver cannot know the exact phase of the transmitted signal to determine whether it is in a mark or space condition. This would not be possible even if the transmitter and receiver clocks were accurately linked because the path length would determine the exact phase of the received signal. To overcome this problem PSK systems use a differential method for encoding the data onto the carrier. This is accomplished, for example, by making a change in phase equal to a one, and no phase change equal to a zero. Further improvements can be made upon this basic system and a number of other types of phase shift keying have been developed. One simple improvement can be made by making a change in phase by 90 degrees in one direction for a one, and 90 degrees the other way for a zero. This retains the 180 degree phase reversal between one and zero states, but gives a distinct change for a zero. In a basic system not using this process it may be possible to loose synchronisation if a long series of zeros are sent. This is because the phase will not change state for this occurrence.

There are many variations on the basic idea of phase shift keying. Each one has its own advantages and disadvantages enabling system designers to choose the one most applicable for any given circumstances. Other common forms include QPSK (Quadrature phase shift keying) where four phase states are used, each at 90 degrees to the other, 8-PSK where there are eight states and so forth. Phase shift keying, PSK, is widely used these days within a whole raft of radio communications systems. It is particularly well suited to the growing area of data communications. PSK, phase shift keying enables data to be carried on a radio communications signal in a more efficient manner than Frequency Shift Keying, FSK, and some other forms of modulation. With more forms of communications transferring from analogue formats to digital formats, data communications is growing in importance, and along with it the various forms of modulation that can be used to carry data. There are several flavours of phase shift keying, PSK that are available for use. Each form has its own advantages and disadvantages, and a choice of the optimum format has to be made for each radio communications system that is designed. To make the right choice it is necessary to have a knowledge and understanding of the way in which PSK works. What is PSK, Phase Shift Keying.

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Phase Shift Keying, PSK, basics

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(BPSK) or it is occasionally called Phase Reversal Keying (PRK). A digital signal alternating between +1 and -1 (or 1 and 0) will create phase reversals, i.e. 180 degree phase shifts as the data shifts state.

3.1.4 BPSK

In phase shift keying (PSK), the phase of a carrier is changed according to the modulating waveform which is a digital signal. In BPSK, the transmitted signal is a sinusoid of fixed amplitude. It has one fixed phase when the data is at one level and when the data is at the other level, phase is different by 180 degree.

In digital modulation techniques a set of basic functions are chosen for a particular modulation scheme. Generally the basic functions are orthogonal to each other. Basis functions can be derived using ‘Gram Schmidt orthogonalization procedure.Once the basis function are chosen, any vector in the signal space can be represented as a linear combination of the basis functions.
In Binary Phase Shift Keying (BPSK) only one sinusoid is taken as basis function modulation. Modulation is achieved by varying the phase of the basis function depending on the message bits. The following equation outlines BPSK modulation technique. The constellation diagram of BPSK will show the constellation points lying entirely on the x axis. It has no projection on the y axis. This means that the BPSK modulated signal will have an in-phase component (I) but no quadrature component (Q).

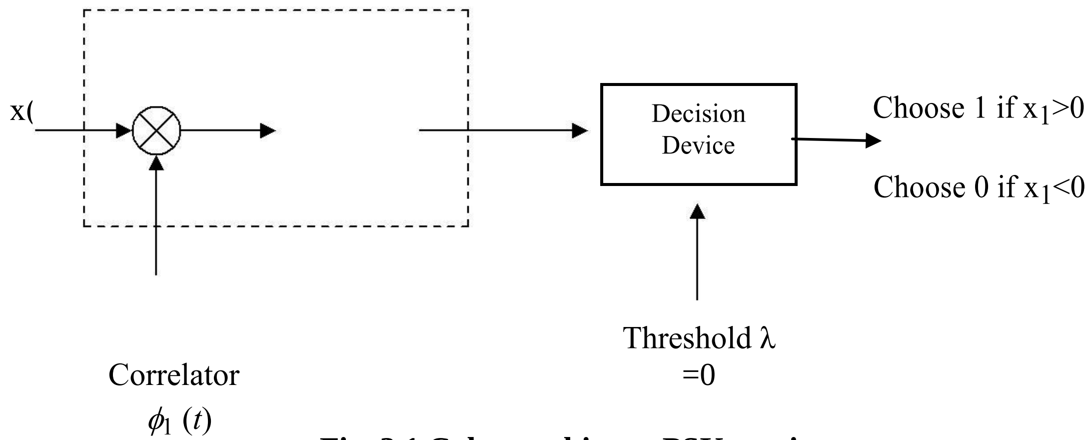


Fig. 3.1 Coherent binary PSK receiver

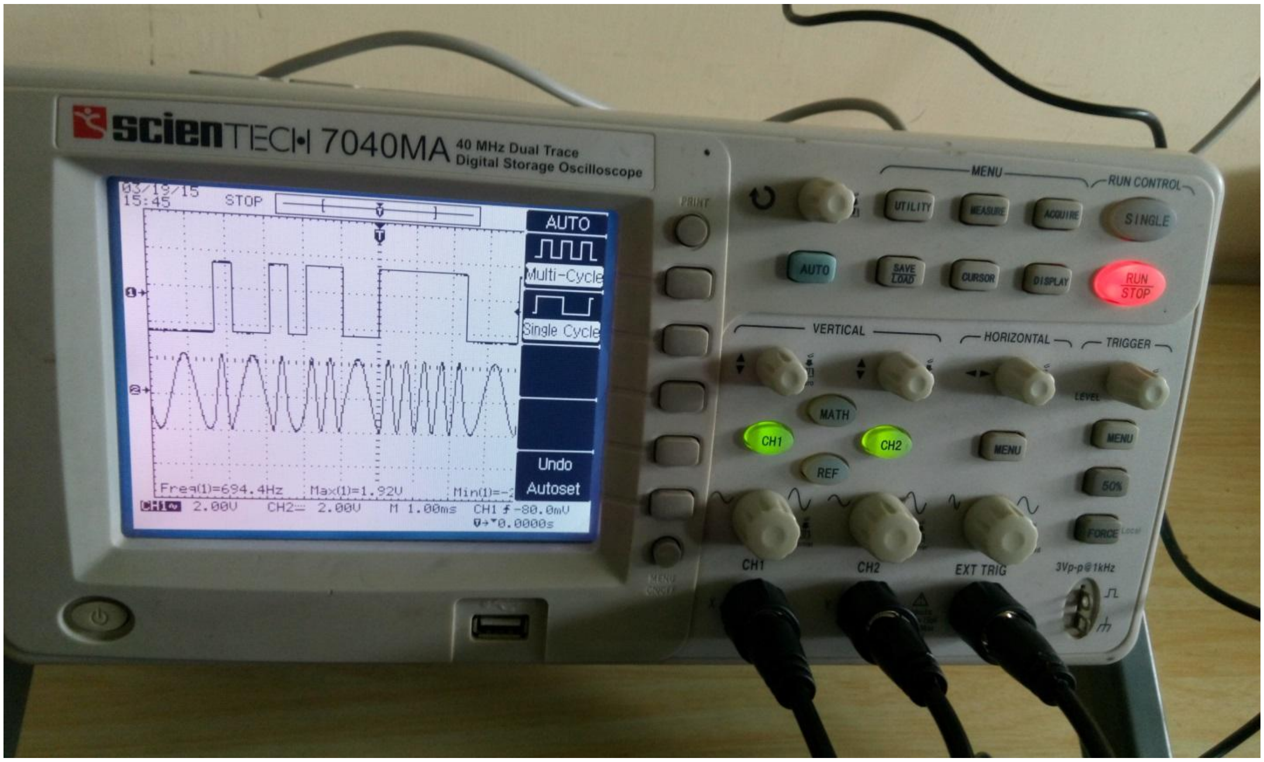


Fig 3.2 FSK waveform

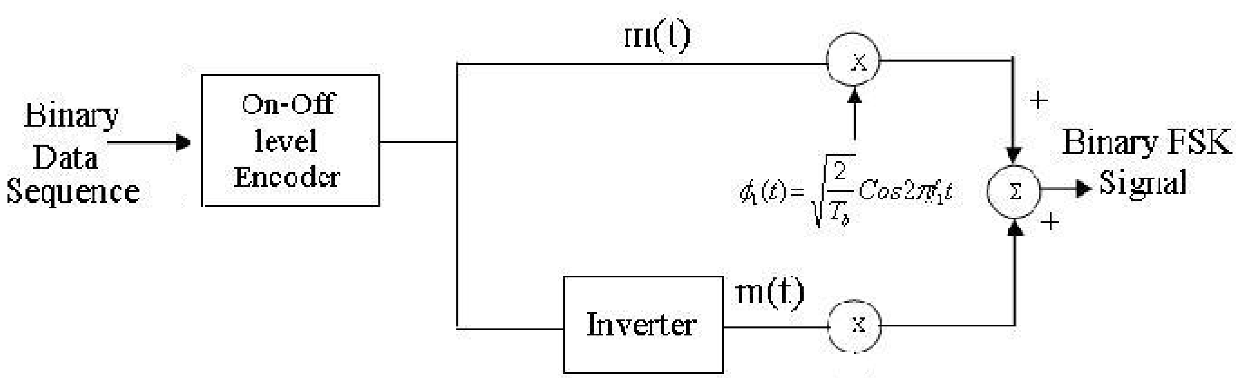


Fig. 3.3 FSK transmitter

Frequency-shift keying (FSK) allows digital information to be transmitted by changes or shifts in the frequency of a carrier signal, most commonly an analog carrier sine wave. There are two binary states in a signal, zero (0) and one (1), each of which is represented by an analog wave form. This binary data is converted by a modem into an FSK signal, which can be transmitted via telephone lines, fiber optics or wireless media.FSK is commonly used for caller ID and remote metering applications.FSK is also known as frequency modulation (FM).

Frequency shift keying (FSK) is the most common form of digital modulation in the high-frequency radio spectrum, and has important applications in telephone circuits. This article provides a general tutorial on FSK in its many forms. Both modulation and demodulation schemes will be discuses Binary FSK (usually referred to simply as FSK) is a modulation scheme typically used to send digital information between digital equipment such as teleprinters and computers. The data are transmitted by shifting the frequency of a continuous carrier in a binary manner to one or the other of two discrete frequencies. One frequency is designated as the “mark” frequency and the other as the “space” frequency. The mark and space correspond to binary one and zero, respectively. By convention, mark corresponds to the higher radio frequency. Minimum duration of a mark or space condition is called the element length. Typical values for element length are between 5 and 22 milliseconds, but element lengths of less than 1 microsecond and greater than 1 second have been used. Bandwidth constraints in telephone channels and signal propagation considerations in HF channels generally require the element length to be greater than 0.5millisecond. An alternate way of specifying element length is in terms of the keying speed. The keying speed in “bauds” is equal to the inverse of the element length in seconds. For example, an element length of 20 milliseconds (.02 seconds) is equivalent to a 50-baud keying speed.

The Sequence Generator module is used to model a digital signal and its SYNC output is used to trigger the scope to provide a stable display. The function generator's VCO facility is used to generate the FSK signal. Then it can be recovered using any of the FM demodulation schemes. However, as the FSK signal switches back and forth between just two frequencies we can use a method of demodulating it that cannot be used to demodulate speech-encoded FM signals.

Frequency-shift keying is a type of digital binary communication technique. It is identical to FM modulation using a digital binary signal as the message m(t). Thus, a binary 1 represents one frequency, and a binary 0 represents another frequency. The FSK signal deviates from the carrier frequency depending on the binary message m(t). For example, assume m(t) can take on the values 1 or -1. When m(t) = 1, the FSK signal would deviate on the lower side of the carrier frequency. When m(t) = -1, the FSK signal would deviate to the higher side of the carrier frequency. The frequency deviation of the FSK signal is given by the FM equation:

Generation and Detection:

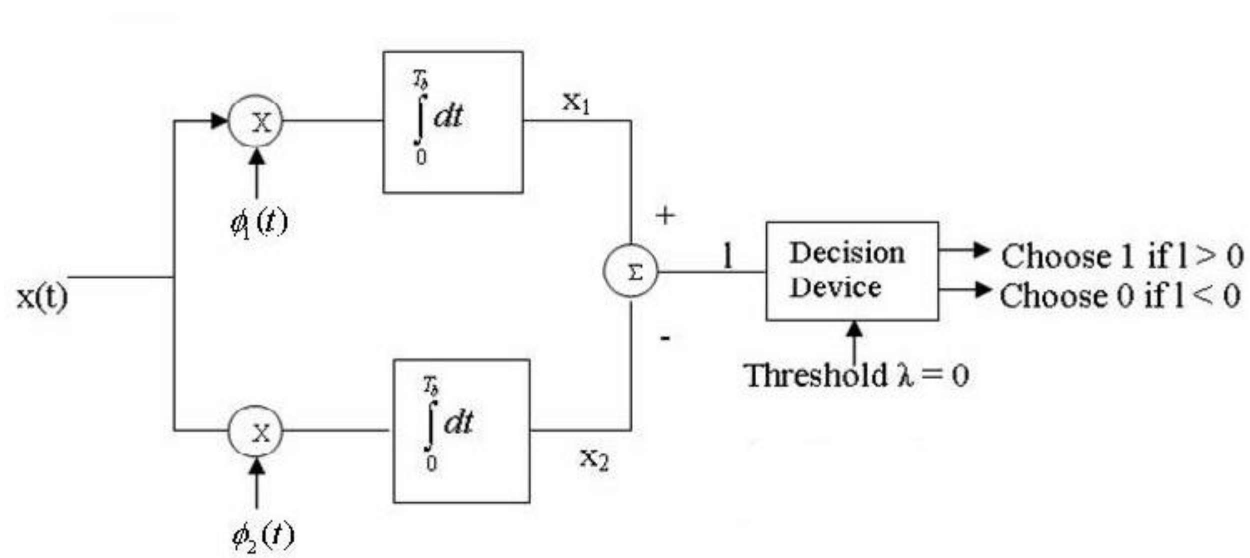


Fig. 3.4 FSK Receiver

A binary FSK Transmitter is as shown in fig. (a). The incoming binary data sequence is applied to on-off level encoder. The output of encoder is $\sqrt{E_b}$ volts for symbol 1 and 0 volts for symbol '0'. When we have symbol 1 the upper channel is

Switched on with oscillator frequency f_1, for symbol '0', because of inverter the lower channel is switched on with oscillator frequency f_2. These two frequencies are combined using an adder circuit and then transmitted. The transmitted signal is nothing but required BFSK signal. The detector consists of two correlators. The incoming noisy BFSK signal x(t) is common to both correlator. The Coherent reference signal $\square_1$ (*t*) *and* $\square_2$ (*t*) are supplied to upper and lower correlators respectively. The correlator outputs are then subtracted one from the other and resulting a random vector 'l' (l=x_1 - x_2). The output 'l' is compared with threshold of zero volts.

If l > 0, the receiver decides in favour of symbol 1.

If l < 0, the receiver decides in favour of symbol 0.

Probability of Error Calculation:-

In binary FSK system the basic functions are given by

$$\phi_1(t) = \sqrt{\frac{2}{T_b}} Cos 2\pi f_1 t \qquad 0 \le t \le T_B$$

$$\phi_2(t) = \sqrt{\frac{2}{T_b}} Cos 2\pi f_2 t \qquad 0 \le t \le T_B$$

The transmitted signals $S_1(t)$ and $S_2(t)$ are given by

$$S_1(t) = \sqrt{E_b}\,\phi_1(t) \qquad \text{for symbol 1}$$

$$S_2(t) = \sqrt{E_b}\,\phi_2(t) \qquad \text{for symbol 0}$$

Therefore Binary FSK system has 2 dimensional signal space with two messages $S_1(t)$ and $S_2(t)$, [N=2 , m=2]

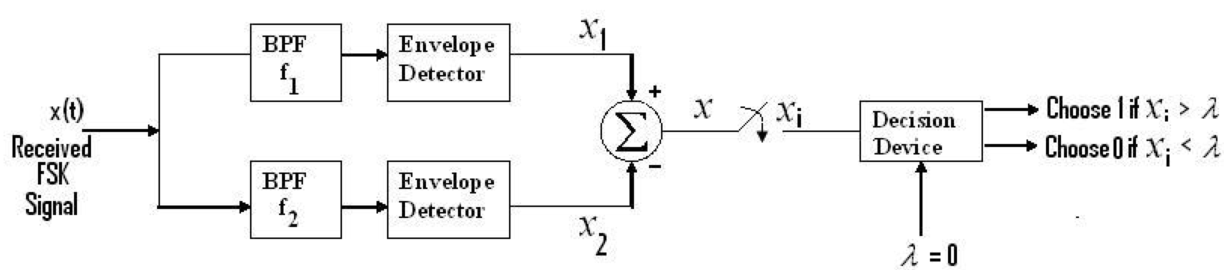


Fig. 3.5 Non- Coherenent FSK Demodulation

Communicate '0's and '1's,The output of filter is envelope detected and then baseband detected using an integratration and dump technique. The detector is simply calculating which of two possible sinusoids is stronger at the receiver. If we take the difference of the outputs of the two envelope detectors the result is bipolar baseband.

The outputs of envelope detector are sampled at $t = kT_b$ and their values are compared with the threshold and a decision will be made in favour of symbol 1 or 0.

3.1.5 Differential Phase Shift Keying:- [DPSK]

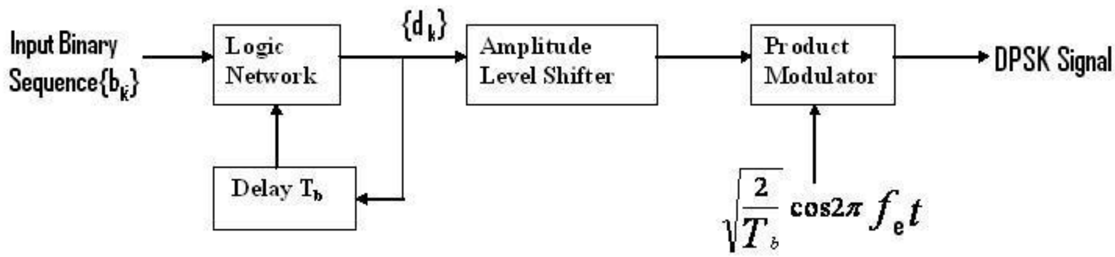


Fig. 3.6 DPSK Transmitter

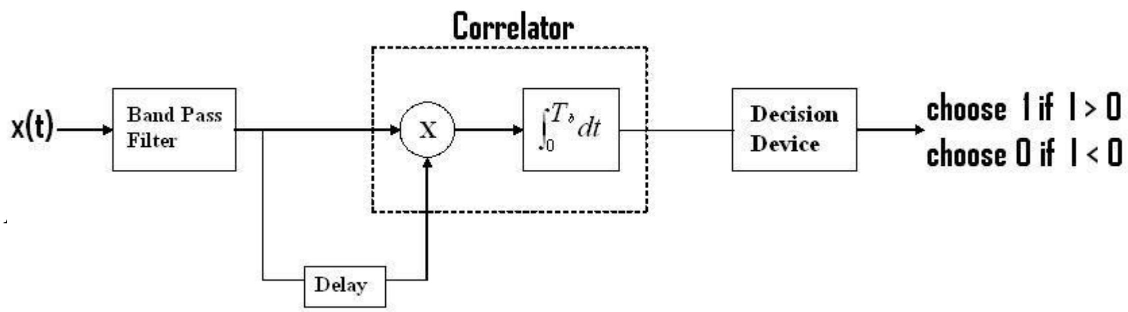


Fig. 3.7 DPSK Receiver

It eliminates the need for coherent reference signal at the receiver by combining two basic operations at the transmitter

(1) Input binary wave is differentially encoded and

(2) Phase shift keying

Hence the name differential phase shifts keying [DPSK]. To send symbol ‘0’ we phase advance the current signal waveform by 180^0 and to send symbol 1 we leave the phase of the current signal waveform unchanged. The differential encoding process at the transmitter input starts with an arbitrary first but, securing as reference and thereafter the differentially encoded sequence$\{d_k\}$ is generated by using the logical equation.

$$d_k = d_{k-1}\ b_k \oplus d_{k-1}\ b_k$$

Where b_k is the input binary digit at time kT_b and d_{k-1} is the previous value of the differentially encoded digit. Table elaborates the logical functions involved in the generation of DPSK signal.

Input Binary Sequence $\{b_K\}$	1	0	0	1	0	0	1	1
Differentially Encoded sequence $\{d_K\}$ 1	1	0	1	1	0	1	1	1
Transmitted Phase 0	0	Π	0	0	Π	0	0	0
Received Sequence (Demodulated Sequence)	1	0	0	1	0	0	1	1

Table 3.1

The received signal is first passed through a BPF centered at carrier frequency f_c to limit noise power. The filter output and its delay version are applied to correlator the resulting output of correlator is proportional to the cosine of the difference between the carrier phase angles in the two correlator inputs. Finally the correlator output is compared with threshold of '0' volts.

If correlator output is +ve -- A decision is in favour of symbol '1'

If correlator output is -ve --- A decision is made in favour of symbol '0'

3.1.6 COHERENT QUADRIPHASE – SHIFT KEYING

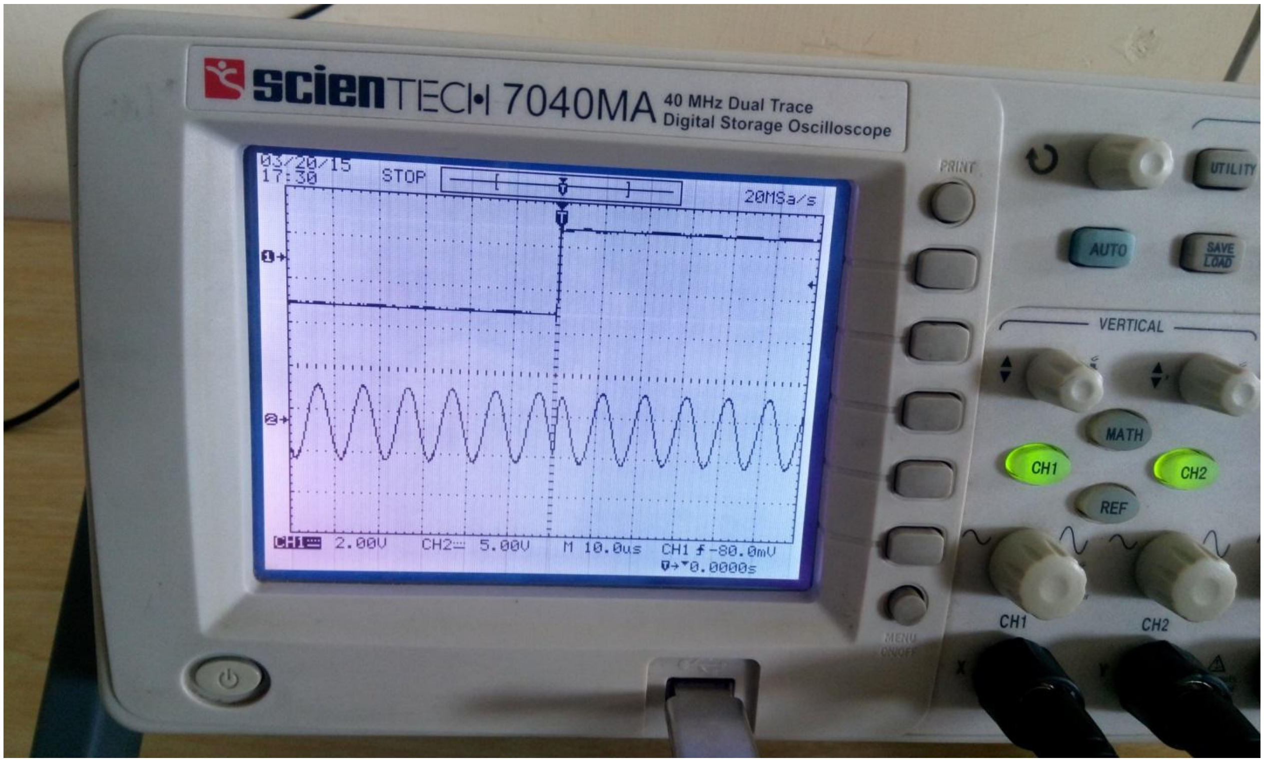


Fig. 3.8 QPSK Waveform

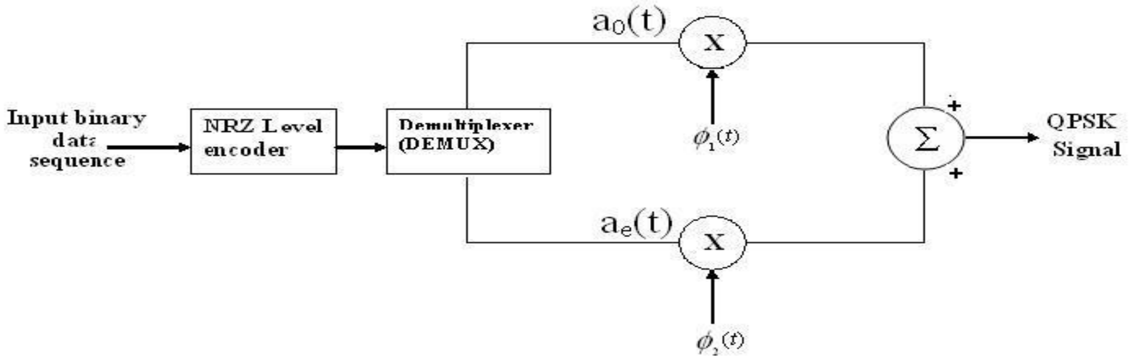


Fig. 3.9 QPSK Transmitter

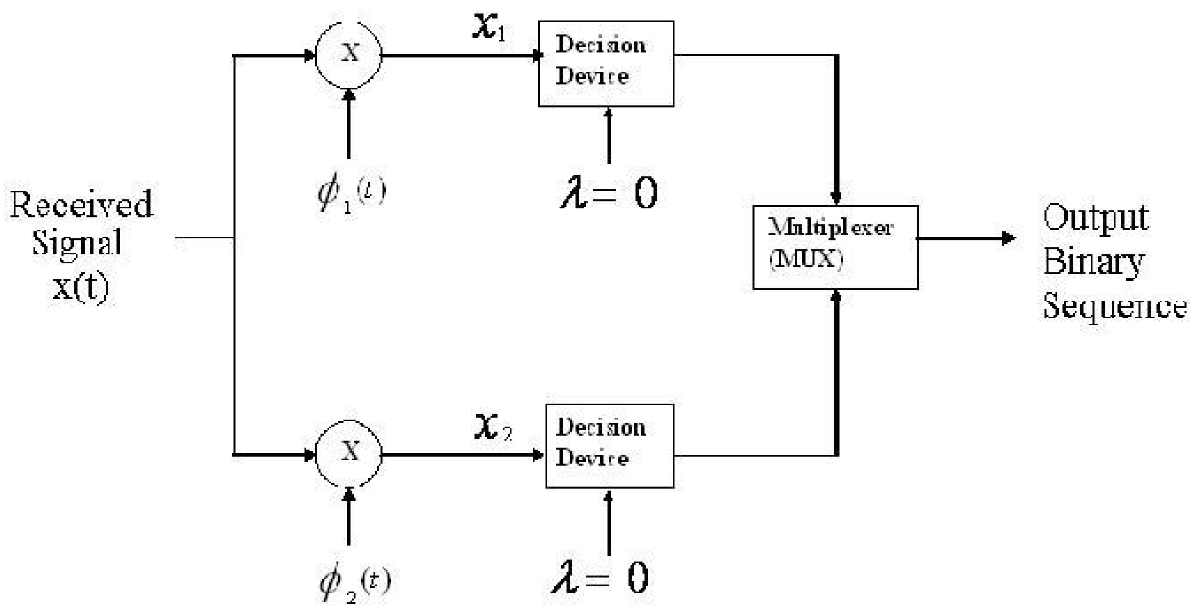


Fig. 3.10 QPSK Receiver

QPSK (Quadrature Phase Shift Keying) is type of phase shift keying. Unlike BPSK which is a DSBCS modulation scheme with digital information for the message, QPSK is also a DSBCS modulation scheme but it sends two bits of digital information a time (without the use of another carrier frequency).

The amount of radio frequency spectrum required to transmit QPSK reliably is half that required for BPSK signals, which in turn makes room for more users on the channel. Each adjacent symbol only differs by one bit, sometimes known as quaternary or quadriphase PSK or 4-PSK, or 4-QAM. QPSK uses four points on the constellation diagram, equispaced around a circle. With four phases, QPSK can encode two bits per symbol shown in the diagram to minimize the BER- twice the rate of BPSK. Analysis shows that this may be used either to double the data rate compared to a BPSK system while maintaining the bandwidth of the signal or to maintain the data rate of BPSK but half the bandwidth needed. The implementation of QPSK is more general than that of BPSK and also indicates the implementation of higher order PSK. Writing the symbols in the constellation diagram in terms of sine and cosine waves used to transmit them. This yields the four phases π/4, 3π/4, 5π/4 and 7π/4 as needed. At the input of the modulator, the digital data's even bits (i.e., bits 0,2,4 and so on) are stripped from the data stream by a "bit-splitter" and are multiplied with a carrier to generate a BPSK signal (called PSK_I). At the same time, the data's odd bits (i.e., bits 1,3,5 and so on) are stripped from the data stream and are multiplied with the same carrier to generate a second BPSK signal (called PSK_Q). However, the PSK_Q signal's carrier is phase shifted by 90^o before being modulated.

The two BPSK signals are then simply added together for transmission and, as they have the same carrier frequency, they occupy the same portion of the radio frequency spectrum. While this suggests that the two sets of signals would be irretrievably mixed, the required 90^o of phase separation between the carriers allows the sidebands to be separated by the receiver using phase discrimination.

Each adjacent symbol only differs by one bit, sometimes known as quaternary or quadriphase PSK or 4-PSK, or 4-QAM. QPSK uses four points on the constellation diagram, equispaced around a circle. With four phases, QPSK can encode two bits per symbol shown in the diagram to minimize the BER- twice the rate of BPSK. Analysis shows that this may be used either to double the data rate compared to a BPSK system while maintaining the bandwidth of the signal or to maintain the data rate of BPSK but half the bandwidth needed. The implementation of QPSK is more general than that of BPSK and also indicates the implementation of higher order PSK. Writing the symbols in the constellation diagram in terms of sine and cosine waves used to transmit them. This yields the four phases $\pi/4$, $3\pi/4$, $5\pi/4$ and $7\pi/4$ as needed. This results in a two dimensional signal space with unit basis functions. The first basis function is used as the in-phase component of the signal and the second as the quadrature component of the signal. Hence the signal constellation consists of the signal- space 4 points.

It is to be noticed that the arrangement uses two product detectors to simultaneously demodulate the two BPSK signals. This simultaneously recovers the pairs of bits in the original data. The two signals are cleaned-up using a comparator or some other signal conditioners then the bits are put back in order using a 2-bit parallel to serial converter.

In order to understand how each detector picks out only one of the BPSK signals and not both of them, recall that the product detection of DSBSC signals is "phase sensitive". That is, recovery of the message is optimal if the transmitted and local carriers are in phase with each another. But the recovered message is attenuated if the two carriers are not exactly in phase. Importantly, if the phase error is 90^o, the amplitude of the recovered message is zero.

It is to be noticed that the product detectors in the figure share the carrier but one of them is phase shifted 90^o.That being the case, once the phase of the local carrier for one of the product detectors matches the phase of the transmission carrier for one of the BPSK signals, there is automatically a 90^o phase error between the detector's local carrier and the transmission carrier of the other BPSK signal. So the detector recovers the data on the BPSK signal that it's matched to and rejects the other BPSK signal. QPSK (Quadrature Phase Shift Keying) is type of phase shift keying. Unlike BPSK which is a DSBCS modulation scheme with digital information for the message, QPSK is also a DSBCS modulation scheme but it sends two bits of digital information a time (without the use of another carrier frequency).The amount of radio frequency spectrum required to transmit QPSK reliably is half that required for BPSK signals, which in turn makes room for more users on the channel.

Data modulators, especially those intended to produce constant envelope output signals, are "high-leverage" components in that even very small deviations from ideal in their behavior can lead to large degradations in overall system performance. Therefore, successful simulation of wireless communication systems depends upon the use of modulator models that capture all of the significant deviations from ideal behavior. In the "usual" development of data modulation techniques as presented in most communications texts, the various techniques are presented in order of complexity, starting with the simplest. Thus BPSK would be presented first, then QPSK followed by m-PSK, and so on.

Because of its relationship to complex-envelope representations of signals, quadrature modulation plays a central role in simulation of wireless communication systems and models for quadrature modulators, and demodulators serve as building blocks for most other types of data modulators and demodulators. Therefore, this chapter begins with a discussion of quadrature phase shift keying (QPSK) and uses this discussion as a vehicle for development of generic models for quadrature modulation and demodulation. The discussion then moves to binary phase shift keying (BPSK) and shows how this simpler format is modeled using the generic quadrature modulation models. A similar approach is then taken for developing models for multiple phase shift keying (m-PSK), minimum shift keying (MSK), and frequency shift .

Modulation is the process of sending data signal over carrier signal to minimize the noise or fading effect. They are mainly divided into two categories i.e., analog and digital. In analog modulation carrier signal is modulated with the help of analog data signal and in digital it modulates with digital signal. Digital modulation is called shift keying because in this, the carrier signal is shifted in amplitude, frequency or phase by digital input signal. Different PSKs can be obtained by M-ary PSK, where M is the no. of states or no. of phase shifts which is depend upon the no. of signals are combined for modulation.

In QPSK two signals are combined for modulation. BER of QPSK is better than higher order PSK signals such as 8-PSK, 16-QAM, 32-QAM etc. which are easily affected by noise. At higher order PSK, larger bandwidth is require for higher data transfer rate and consume more power, whereas QPSK is more bandwidth as well as power efficient. There are many applications where QPSK modulator is used, out of which few are of battery operated devices such as Bluetooth, TDMA cellular communication, Medical Implant Communication Services (MICS) etc. therefore it is necessary to minimize the power consumption of these devices so that the battery will last for longer time. It can be reduce by reducing size of circuit or reducing the speed of operation.

We can see, day by day the digital devices are getting smaller, many research is going on to minimize the size of the design which reduce the power consumption of the device, hence cost of the device will get reduce. If we observe the QPSK output waveform, we found that there is a sinusoidal wave shifting with the change in symbol and the phase angle is same for different symbols as shown in table 1, means we can say that, for specific symbol there will be specific phase shift output signal. So, instead of generating the phase shift by multiplying data signal with carrier one, we will just store the signal in ROM and call it for specific symbol from specific phase.

It means the output waveform will be the same sinusoidal signal with starting from specific phase angle. In this case we don’t require more than one ROM to store our signal since it is only the sinusoidal signal. We just have to start the output signal from different phase angle according to input symbol (00, 01, 11, 10). The phase shift angle may be 00 , 900 , 1800 and 2700 or it may be 450 , 1350 , 2250 and 3150 . In proposed block diagram as shown, we are shifting signal from

450 . The constellation diagram shows the phase angle and amplitude of signal for different symbols. In above diagram the phase angles are 450 , 1350 , 2250 and 3150 for “00”, “01”, “11” and “10” respectively. The symbols are taken as gray codes i.e. one bit change per symbol or 900 phase shift per symbol.

A. Carrier Source It provides a sinusoidal carrier signal of specific frequency which is modulated by the data signal. A ROM is used to store the amplitude values of the signal which can be read by using VHDL coding for FPGA to produce sine signal. On board frequency is in MHz which is high to observe the output signal on DSO, a shift register can be developing to provide various frequency signals.

B. Phase Shifter It shifts the sine signal into four different angles as shown in f. It is nothing but the ROM which stored the sinusoidal signal and we are sending the signal at the output with different starting point of signal or different phase angle which is specific for different symbols. Actually it is a DMUX which takes one input as carrier signal i.e. sinusoidal signal and giving output as different phase shifted sinusoidal signal. These output are selected by two select lines which are two input signals I and Q.

C. Shift Register Practically we take two input data signals to modulate them with same carrier signal, but on board we need to generate two signals from a random signal. Here we are taking one data signal and separating it into two signal i.e. Serial in Parallel out (SIPO).

D. Multiplexer It selects only one signal at a time out of four shifted signals and gives as output QPSK signal. The signal is selected by two select lines I and Q, means output will be the phase shifted signal according to the input symbol. E. DAC All the above blocks can be dump into FPGA kit which gives output digital QPSK signal so, we need to convert it into analog form using DAC. The waveform can be observed on DSO. The ROM contains values of sinusoidal signal, it requires a clock to read the values i.e. one value per clock pulse, similarly shift register require a clock signal to take input data signal then T = 1 fc * x = Td Where, T = Time period of sine signal x = No. of values stored in the ROM fc = clock frequency applied to carrier signal Td = Time period of data signal

To divide the frequency a divider block with factor x is required to design, it will take the input clock frequency fd which is to be divided by x to get the frequency equal to freq. of sine signal. In this paper, Active HDL tool is use to generate the blocks with VHDL coding. The FPGA kit will be use to implement these blocks on hardware to observe the output waveform. But the FPGA is a digital circuit which provide digital output signal hence we need a DAC to covert this signal to analog one and it can be observe on DSO. Before this the Simulink software is used to verify the block diagram and check its output waveforms.

A QPSK modem was created using FPGA. In this paper, modulator and demodulator is embedded on same FPGA kit. In the design, direct digital synthesis principle was presented. In this paper, simulation results of output wave were only introduced (Song and Yao, 2010). In the other work, QPSK modulator was presented. Resource utilization in device was only presented. However, all the information obtained were about power consumption.

Data rate of a designed FPGA based QPSK modulator is about 2 Mbps. In this design, for carrier signal, using square wave signal QPSK modulator was created .This paper mainly concentrates on the hardware realization of modulators for BPSK and QPSK techniques. These modulator schemes were created by the fastest data transfer method. In the design, each sample of output signal is obtained approximately, each clock time using mux-based algorithm. One of the most important issues is power consumption and resource utilization on FPGA device. Although, in numerous studies, FPGA-based QPSK modulator or BPSK modulator was presented, resource utilization or power consumption issues were not discussed.

In order to reveal these unclear issues, such as resource utilization and power consumption, device utilization was also presented in this work. Comparison of resource utilization and power consumption was realized and the simulation results were presented. Additionally, the design of this study was compared to those of previous works. Before this the Simulink software is used to verify the block diagram and check its output waveforms. QPSK (Quadrature Phase Shift Keying) is type of phase shift keying. Unlike BPSK which is a DSBCS modulation scheme with digital information for the message, QPSK is also a DSBCS modulation scheme but it sends two bits of digital information a time (without the use of another carrier frequency).

CHAPTER-4

Characteristics of signal propagation in a wireless channel

Under ideal conditions, in wireless communication also the exact replica of the transmitted signal will be obtained at the receiver. But naturally this is not the case as a lot of parameters like attenuation factor, fading, Doppler Effect etc. have to be taken care of while considering a wireless radio transmission channel. The actual signal received at the receiver end of a mobile communication system has various reflections of the actual signal, reflected from various buildings and other structures present in the path of the signal, effects of Doppler effect when the transmitter or receiver or both are moving, fading, attenuation, addition of noise and so on. In order to understand how a wireless communication system works we need to understand how the channel changes the characteristics of the signal.

4.1 Attenuation

It is the loss of signal strength as the signal is transmitted from one point to another. It depends on various parameters like the frequency at which the signal is transmitted, the power of the transmitted signal, blocks in the path of the signal, distance between the transmitter and the receiver, multipath fading effects. Whenever an object comes in the way of the line of sight of transmission, shadowing happens. The object can be a structure like a building or a tower or it might be an environmental attenuation factor. These are the factors of attenuation caused by the environment due to Geography, climate etc like mountains, oceans etc which leads to alterations in the transmitted signal.

4.2 Multipath effects

There can be many reflecting objects in the path of the signal. The all reflect the signal which results in the receiver receiving some or all of the reflected signals. If the signals are in phase they reinforce each other but if the signals are out of phase they might cancel each other out resulting in signal loss. This is the case when the independent noise signal induced in each of the versions of the reflected signals are ignores. Considering the noise factor along with multipath effects one can very easily see that the signal obtained at the receiver may not have much of a resemblance with the transmitted signal unless some measures are taken to reduce such signal impairing effects of the channel. When the transmitter, receiver or both are moving the problem of Doppler shift also comes into picture. There are several problems caused by multipath effect like multipath fading, Doppler shift, delay spread, frequency selective fading etc.

4.3 DIFFERENT RADIO CHANNELS

The complex nature of wireless channel used in real time radio communication has lead the scientists working in the field of communication to develop channel models that help us to understand the real time system better. These channel models include only one or more factors of the wireless channel so that we can use it to model schemes or modulation techniques that can overcome the problem to an extent. It helps us to study the effect of one signal impairment factor at a time and can be used to show how it affects the signal. Also the channel model can be used as a medium to devise ways to overcome the effect of these channel impairment factor under consideration.

In this project we have studied the effect of various communication channels on PSK modulated signals. The channels covered under the purview of this project include Additive White Gaussian Noise (AWGN) channel, multipath fading channels like Rayleigh and Rician fading channels. These channels and their characteristics are explained below.

4.3.1 AWGN CHANNEL

Additive White Gaussian Noise (AWGN) channel is an analog channel that adds white noise to the signal transmitted through it. White noise has noise signals of amplitude following Gaussian distribution curve and has constant spectral density. It is called additive since it adds noise to the system. The word white is used to point to the analogy to the white colour in the spectrum of visible light since like white light which has uniform emissions under all frequencies; white noise has uniform power under all frequencies.

This model is of utmost importance while considering satellite communications since the space adds noise to the transmitted signal. It is insufficient while considering mobile radio communications since it does not take into consideration the different types of fading that occour due to hindrances in the signal path. But it is used to study the same to simulate the effect of background noise in mobile communication systems. AWGN channel provides maximum corruption of the signal. So if the system works optimally under the noise provided by AWGN channel it can work in almost all practical situations involving noise.

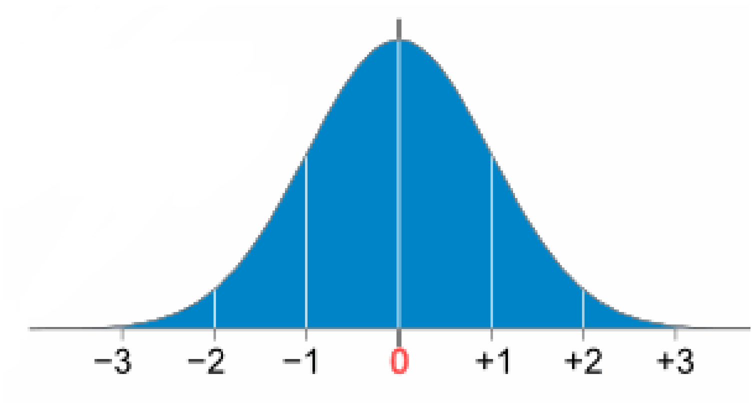
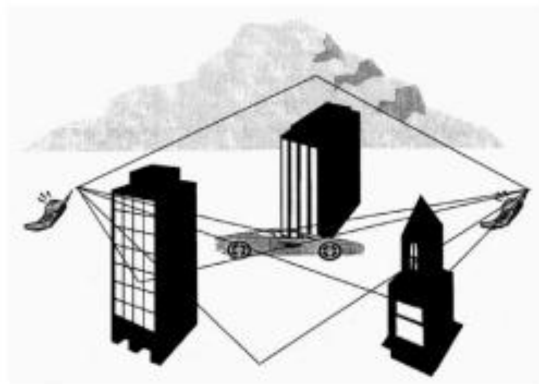


Fig. 4.1 Gaussian Curve

A standard normal distribution curve is as shown above and the AWGN channel add noise that follows this curve.

4.3.2 FADING CHANNELS

Fading channels take into consideration the effects of multipath fading. There are several reflecting objects in the path of the signal in a real time system. They may include billboards, trees, buildings or geographical parameters like oceans (oceans are highly reflecting bodies which lead to a high level of multipath fading related problems) or hillocks as can be seen from the figure below.



Multipath fading channel

Fig. 4.2 Multipath Fading

In mobile radio communications in urban areas the condition is worse as the caller as well as the called subscriber might be moving say for example in cars. The automobiles which are moving on the road also serve as reflecting media, thus responsible for a complex mixture of multipath faded signals being received by the mobile phones.

There are two main types of multipath fading as given below.

1. Flat fading - This type of fading equally changes the frequencies present in a given channel. That is, this type of fading affects the amplitude of the signal or its rise or fall over a time.
2. Frequency Selective fading

In this type of fading the channel affects the each different frequency passing through it differently. Across the channel, the amplitude and phase of the signal varies. There are chances of receiving relatively deep nulls which can lead to problems in receiving. Balancing the signal’s amplitude may be like a solution of the problem but it is not enough to just maintain the amplitude of the signal but some sort of equalization be necessary for overcoming this effect. Some techniques like OFDM overcome this problem by spreading the data over the channel to make sure that only a fraction of the data is lost due to these types of nulls. It can be reconstituted by forward error correction techniques and thus it helps overcome the above mentioned problem of frequency selective fading.

An example of frequency selective fading can be observed in evening receptions of medium wave broadcast stations via ground and sky wave. The signals received through two different means of propagation have different phases which cause problems in detecting the received signal. Since path length affects multipath fading, different frequencies in the AM broadcast signal are change in a different way resulting in distortion. Even high frequency band affected by the problem of selective multipath fading.

They reflect the signal with the end result being that the receiver receives a signal which is either the constructively or destructively interfered version of the reflected signals obtained from various points of reflection. Several daily life processes like multipath scattering effects, Doppler shifts due to the motion of Tx/ Rx, time dispersion etc are taken care of in the two fading channels are explained below.

4.3.3 Rayleigh Fading channel

This channel model gives a an almost real picture of how the signal is affected by multipath fading effects especially in the case of non line of sight communication.

Rayleigh fading refers to the form of fading that is often experienced in an environment where a large number of reflecting elements are present. The Rayleigh fading model uses a statistical approach to examine the propagation, and may be used in a number of harsh environments.

The Rayleigh fading model is normally viewed as a suitable approach to take when analyzing and prediction radio wave propagation performance for areas such as cellular communications in a well built up urban environment where there are many reflections from buildings, etc. High frequency ionospheric radio wave propagation where reflections (or more exactly refractions) occur at many points within the ionosphere is also another area where Rayleigh fading model applied properly. It is also better to apply the Rayleigh fading model for tropospheric radio propagation so, there are various reflection points and the signal may follow a variety of different paths.

The Rayleigh propagation model is most applicable to instances where there are many different signal paths, none of which is effective. In this process all the signal paths will vary and can have an impact on the overall signal at the receiver.

Rayleigh fading basics

The Rayleigh fading model is particularly useful in scenarios where the signal may be considered to be scattered between the transmitter and receiver end. In this form of scenario there is not a single signal path that effective and a statistical approach is required to the analysis of the overall nature of the radio communications channel.

Rayleigh fading is a model that can be used to describe the form of fading that occurs when multipath propagation exists. In any high altitude environment a wireless signal will travel via a number of different paths from the transmitter to the receiver. The most general path is the direct path, or LOS path.

However there will be many objects around the direct path. These objects may use to reflect, refract, etc. As an outcome of this, there are various other paths by which the data may reach the destination.

When the signals reach the receiver, the resultant signal is a collection of all the signals that have reached the receiver end via the different paths that are available there. These signals will all combine together, phase of the signal being essential. Depends on the way in which these signals combine together, the signal will change in strength. If they all are in same phase with each other they will add together. Though this is not generally the case, as some will be in same phase and others are out of phase, this depends on the various path lengths, and hence some will try to add to the resultant signal, whereas others will try to subtract from it.

As there is often movement of the transmitter or the receiver this can cause the path lengths to change and according to this, the signal level will change. Also if any object is used for reflection or refraction of any part of the signal moves, then this will also causes change. This happens because some of the path lengths will vary and in turn this will mean their relative phases will vary, giving rise to a variation in the addition of all the received signals.

The Rayleigh fading model can be used to examine radio signal propagation on the basis of statics. It works best under conditions when there is no dominant signal (e.g. direct line of sight signal), and in many instances cellular telephones being used in a dense urban environment fall into this category. Other examples where no dominant path generally exists are for ionospheric propagation where the signal reaches the receiver via a huge number of individual paths. Propagation using tropospheric ducting also exhibits the same patterns. Accordingly all these examples are ideal for the use of the Rayleigh fading or propagation model.

Rayleigh channel allows us

- to compute the probability density function of the received amplitude.
- to compute the Doppler spread and the rate of channel fluctuations.
- to build a channel simulator
- to predict the effect of multiple interfering signals in a mobile channel
- to find outage probabilities in a cellular network

If the set of reflected waves are dominated by one strong component, Rician fading is a more appropriate model.

4.3.4 Rician fading

The model behind Rician fading is similar to that for Rayleigh fading, except that in Rician fading a strong dominant component is present. This dominant component can for instance be the line-of-sight wave. Refined Rician models also consider thatκ

- that the dominant wave can be a phasor sum of two or more dominant signals, e.g. the line-of-sight, plus a ground reflection. This combined signal is then mostly treated as a deterministic (fully predictable) process, and that
- the dominant wave can also be subject to shadow attenuation. This is a popular assumption in the modelling of satellite channels.

Besides the dominant component, the mobile antenna receives a large number of reflected and scattered waves.

Rician Distribution

Consider two Gaussian random variables X and Y. Here X models the specular component (LOS) and Y models the random/scatter component. By definition, X has non-zero mean (m), Y has zero mean and both have equal variance σ^2. Then the transformation, $Z = \sqrt{X^2 + Y^2}$ is Rician Distributed.

The ratio of power of specular component to the power of random component is called Rician κ factor and it is defined as $\kappa = \frac{m}{2\sigma^2}$

It can be immediately ascertained that Rayleigh Fading is related to central Chi Square distribution (due to zero mean) and the Rician Fading is related to non-central Chi Square distribution (due to non zero mean).

Signal Model

To model the system let us use a sinusoidal carrier

S (t)= cos $\boldsymbol{\omega_c t}$

This signal received over a Rician multipath channel can be expressed as

$$\mathbf{v(t) = C \cos \omega_c t + S^N_{n=1} \; r_n \cos (\omega_c t + f_n)}$$

where

C is the amplitude of the line-of-sight component

r_n is the amplitude of the *n*-th reflected wave

f_n is the phase of the *n*-th reflected wave

n = 1 .. *N* identify the reflected, scattered waves.

Rayleigh fading is recovered for $C = 0$

Rician factor

The Rician *K*-factor is defined as the ratio of signal power in dominant component over the (local-mean) scattered power. In the expression for the received signal, the power in the line-of-sight equals $C^2/2$. In indoor channels with an unobstructed line-of-sight between transmit and receive antenna the *K*-factor is between, say, 4 and 12 dB. Rayleigh fading is recovered for $K = 0$ (-infinity dB).

Applications

Examples of Rician fading are found in

- Microcellular channels
- Vehicle to Vehicle communication
- Indoor propagation
- Satellite channels

Therefore form the observations made above it is clear that:

Rayleigh fading channel is used when there are one or more faded paths between transmitter and receiver

Rician fading channel is used when there is direct line of sight path from transmitter to receiver.

CHAPTER 5

RESULTS

5.1. AWGN Channel simulation

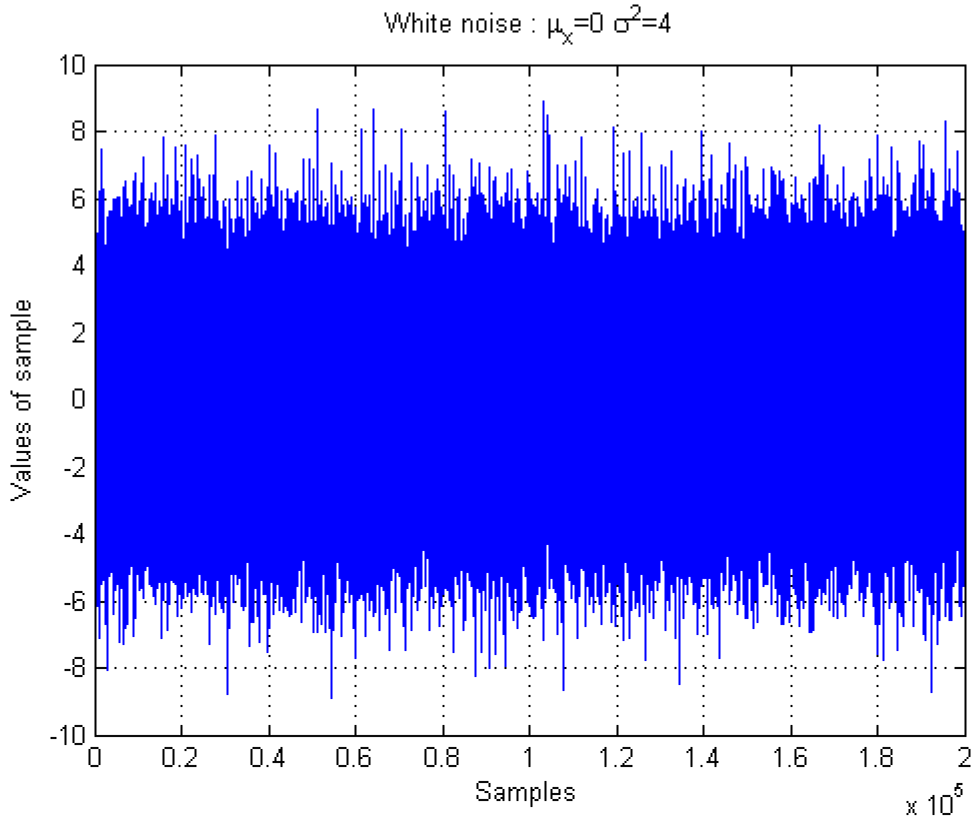


Fig. 5.1 Simulation of BPSK

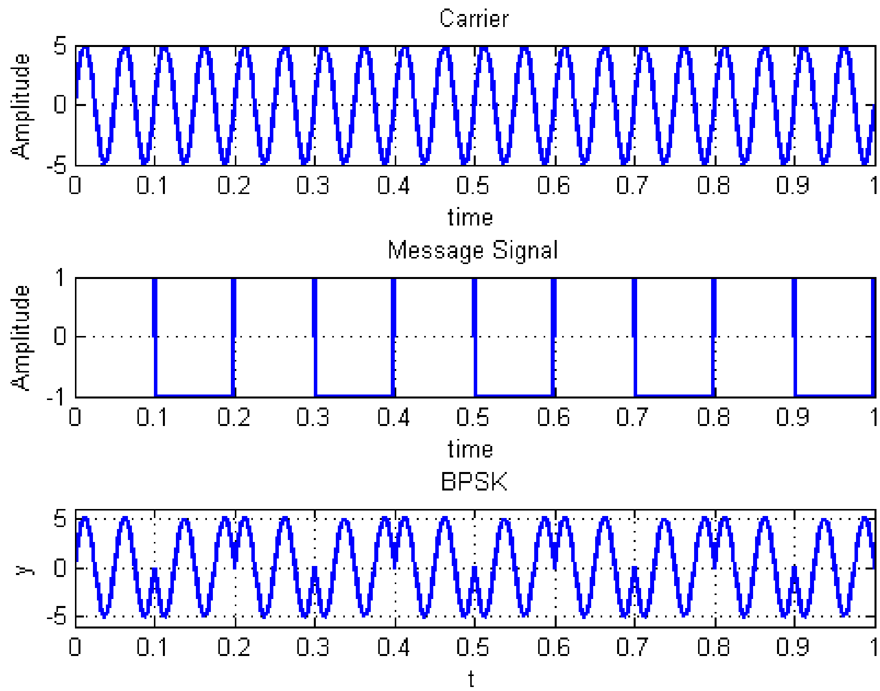


Fig. 5.2 Simulation of BER of BPSK on AWGN channel

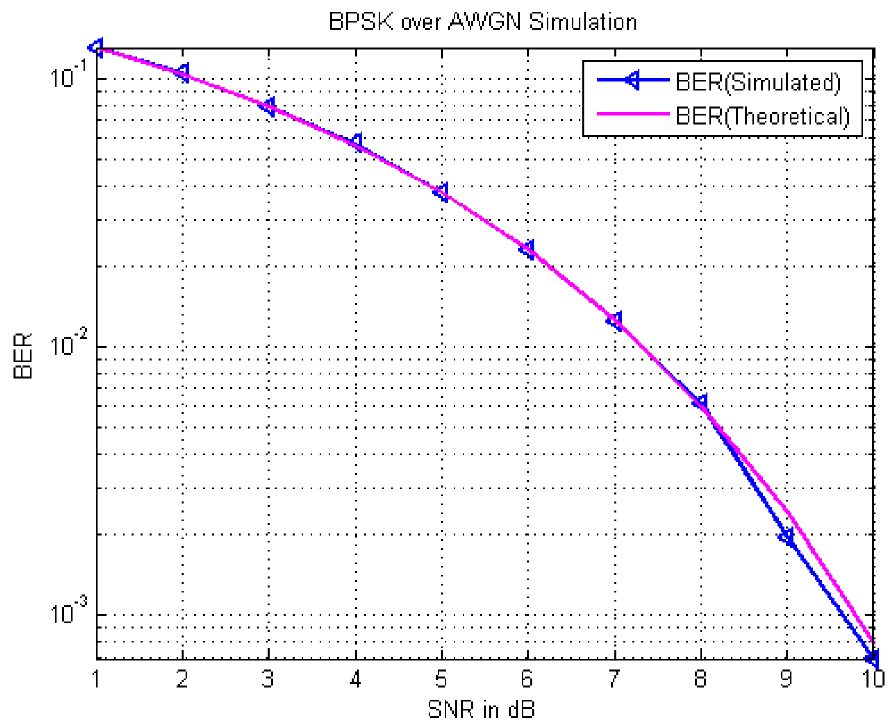


Fig. 5.3 Simulation of BER of QPSK on AWGN channel

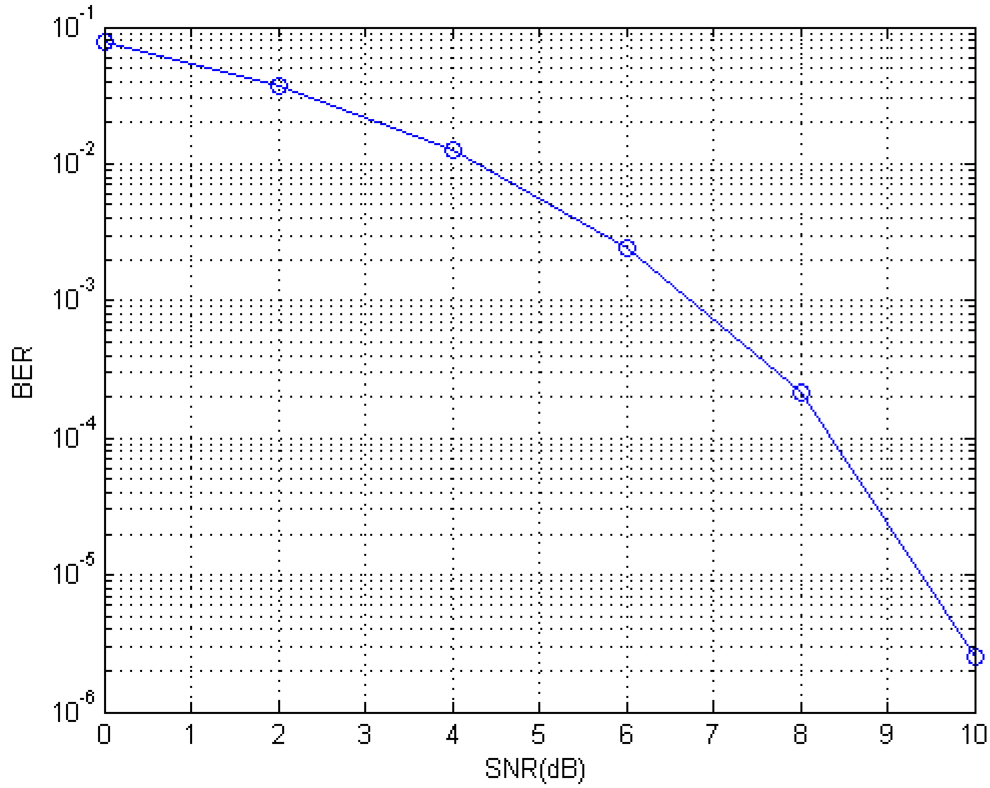


Fig. 5.4 Simulation of 8psk over AWGN channel

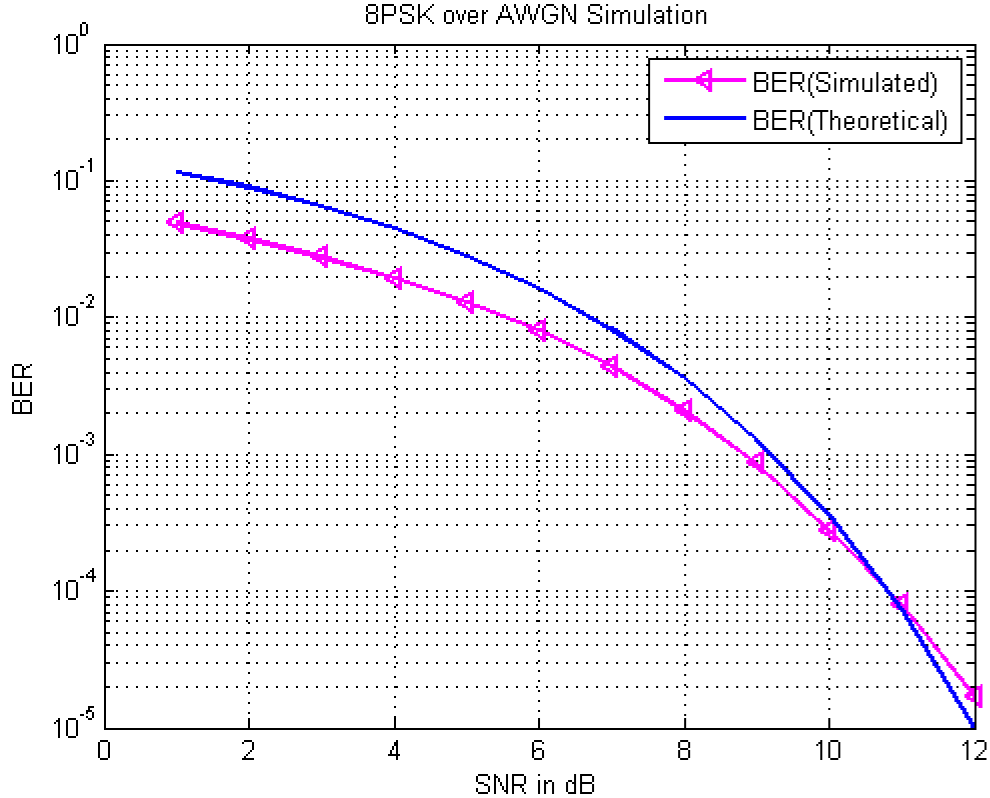


Fig. 5.5 Simulation of 16PSK on AWGN channel

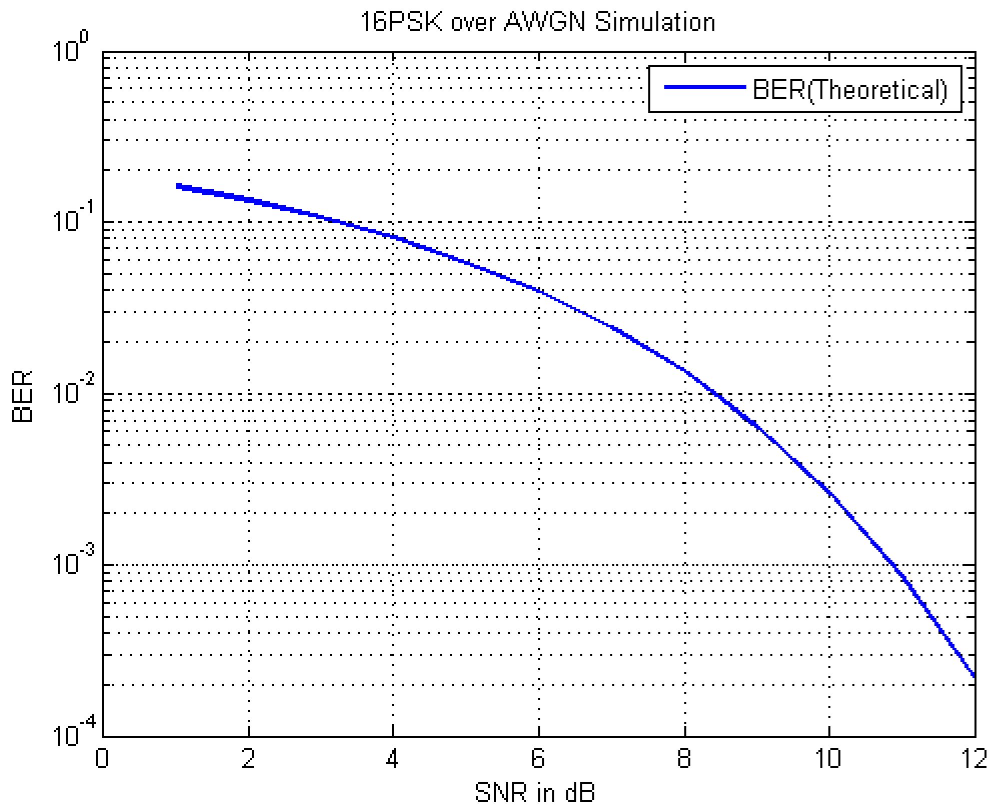


Fig. 5.6 Comparison of performance of BPSK on AWGN and Rayleigh channel

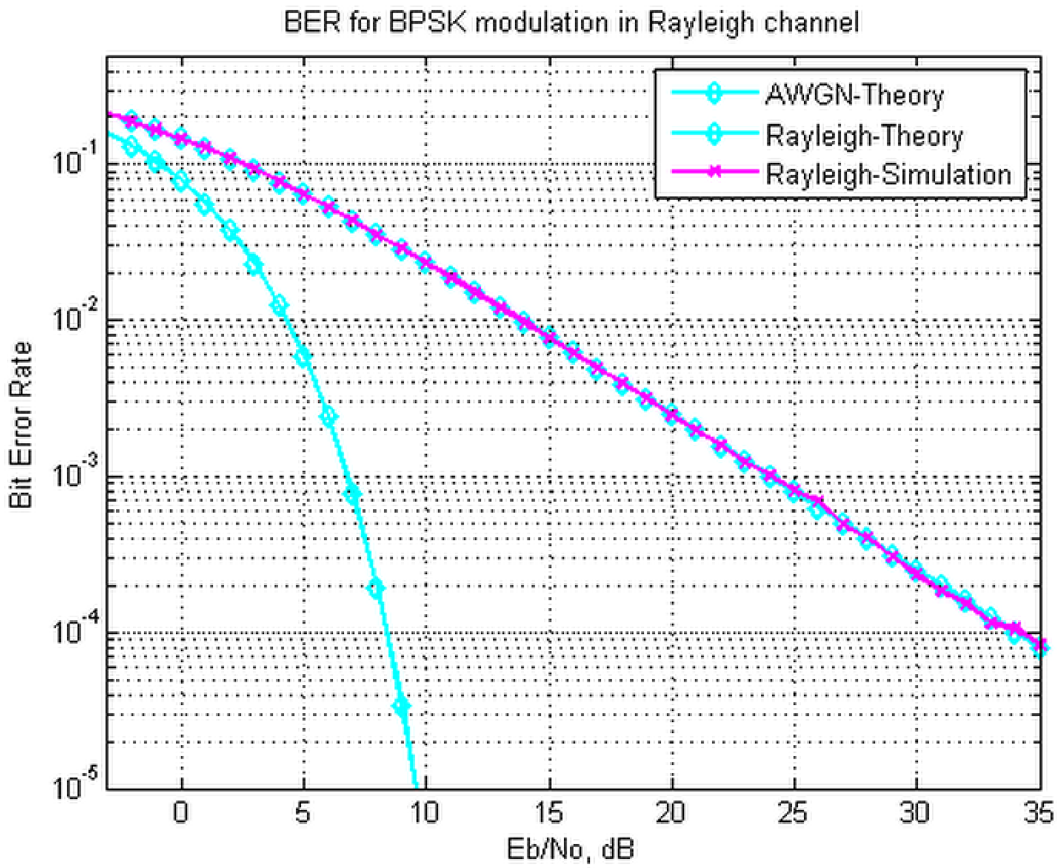


Fig. 5.7 QPSK over Rayleigh channel

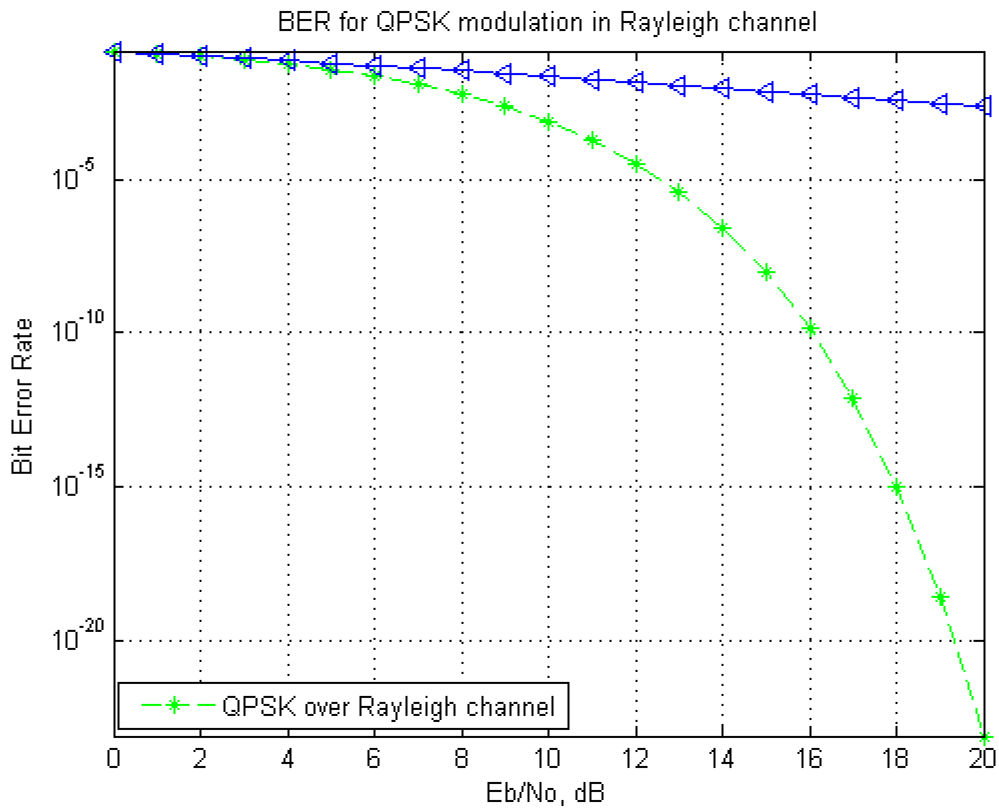


Fig. 5.8 QPSK Over Rayleigh Channel

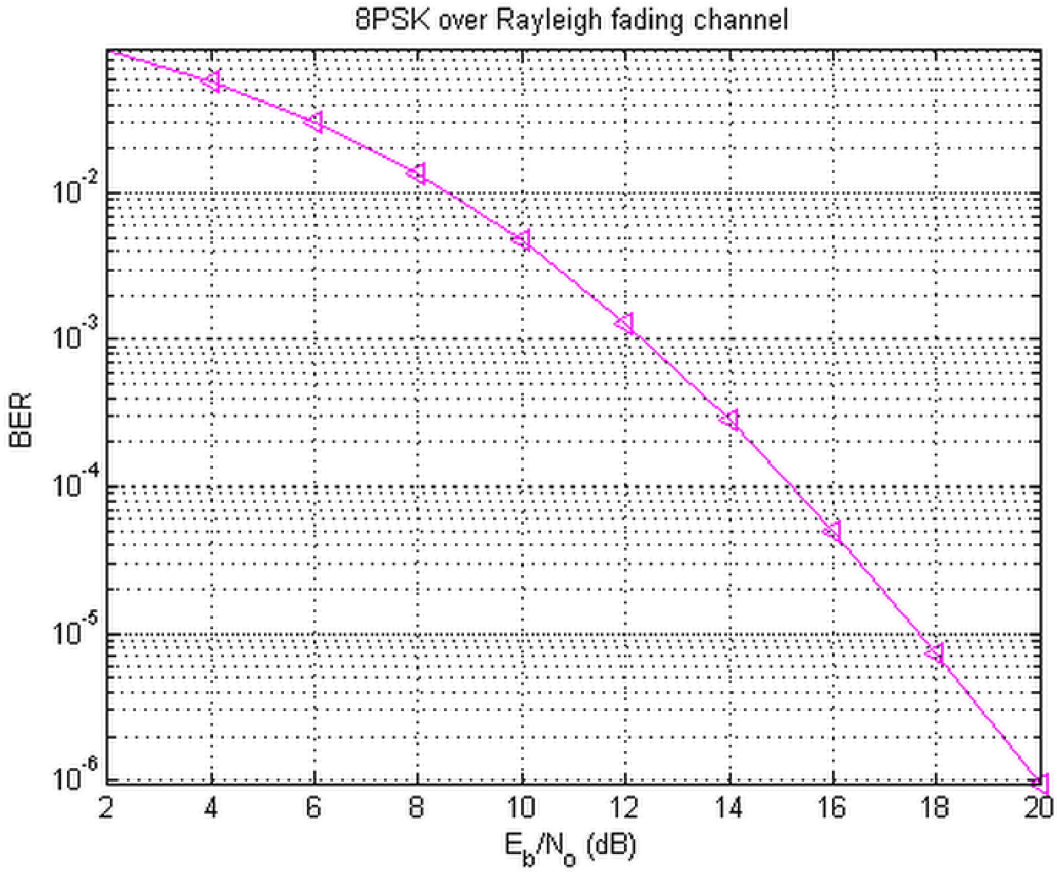


Fig. 5.9 BPSK Over Rayleigh Channel

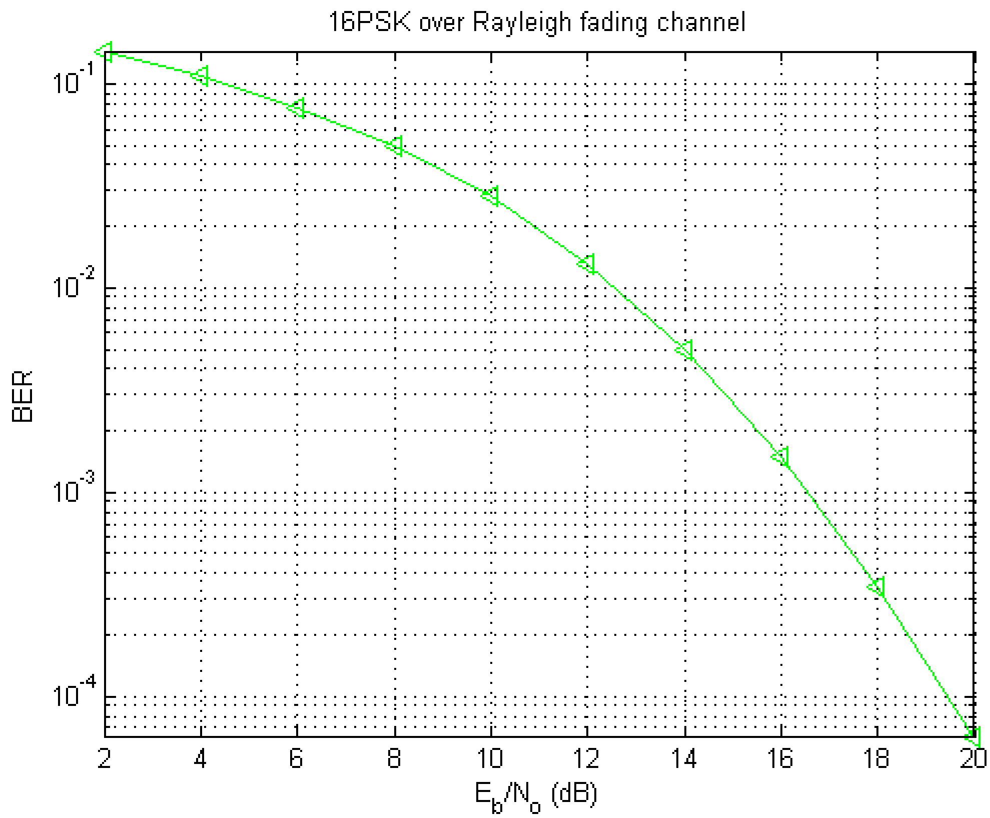


Fig. 5.10 16PSK Over Rayleigh Channel

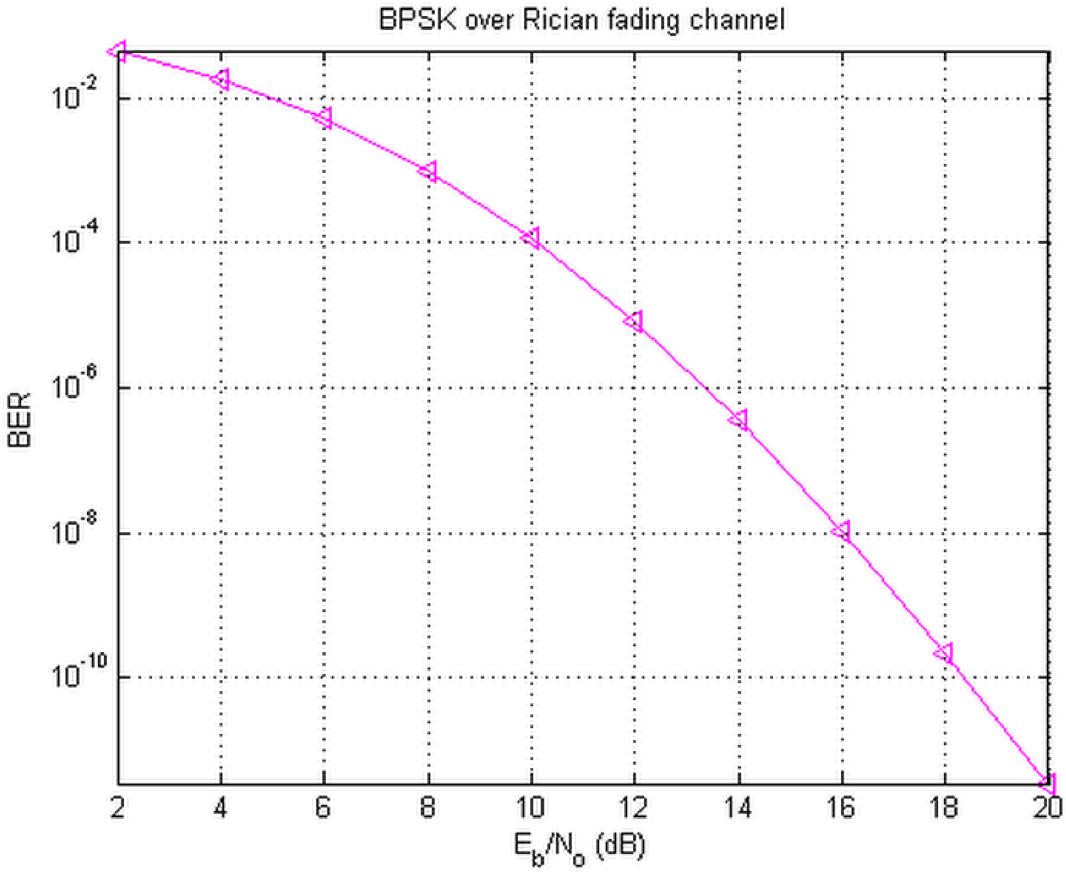


Fig. 5.11 BPSK Over Rician Channel

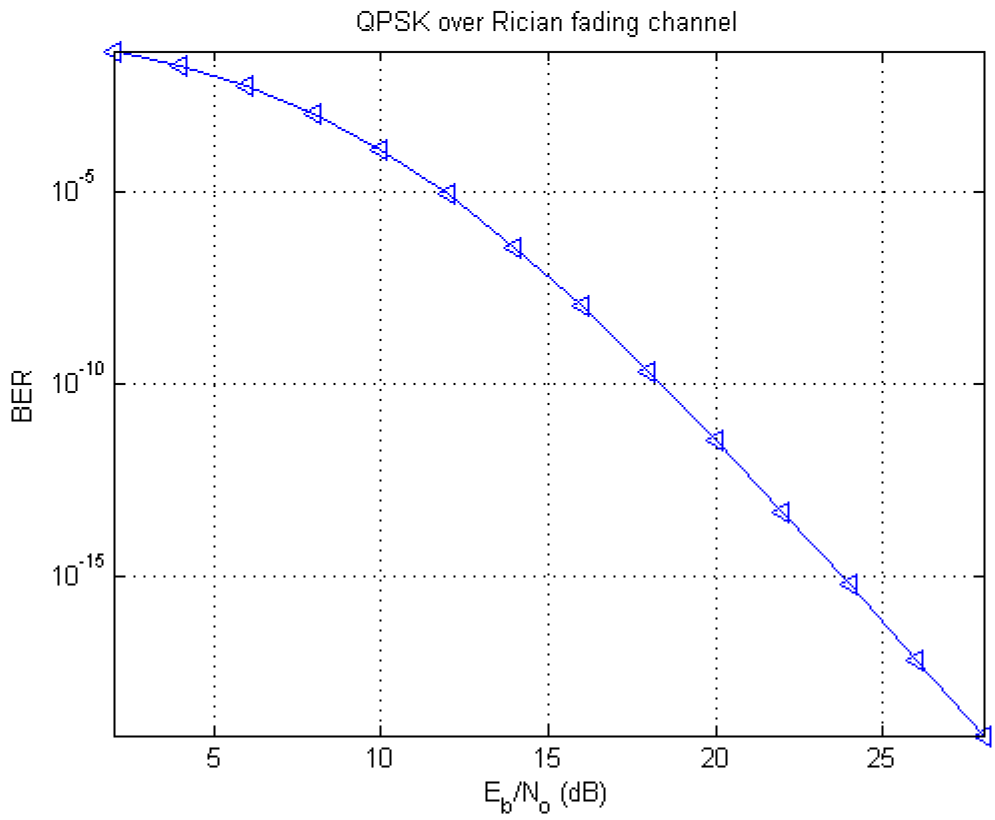


Fig. 5.12 QPSK Over Rician Channel

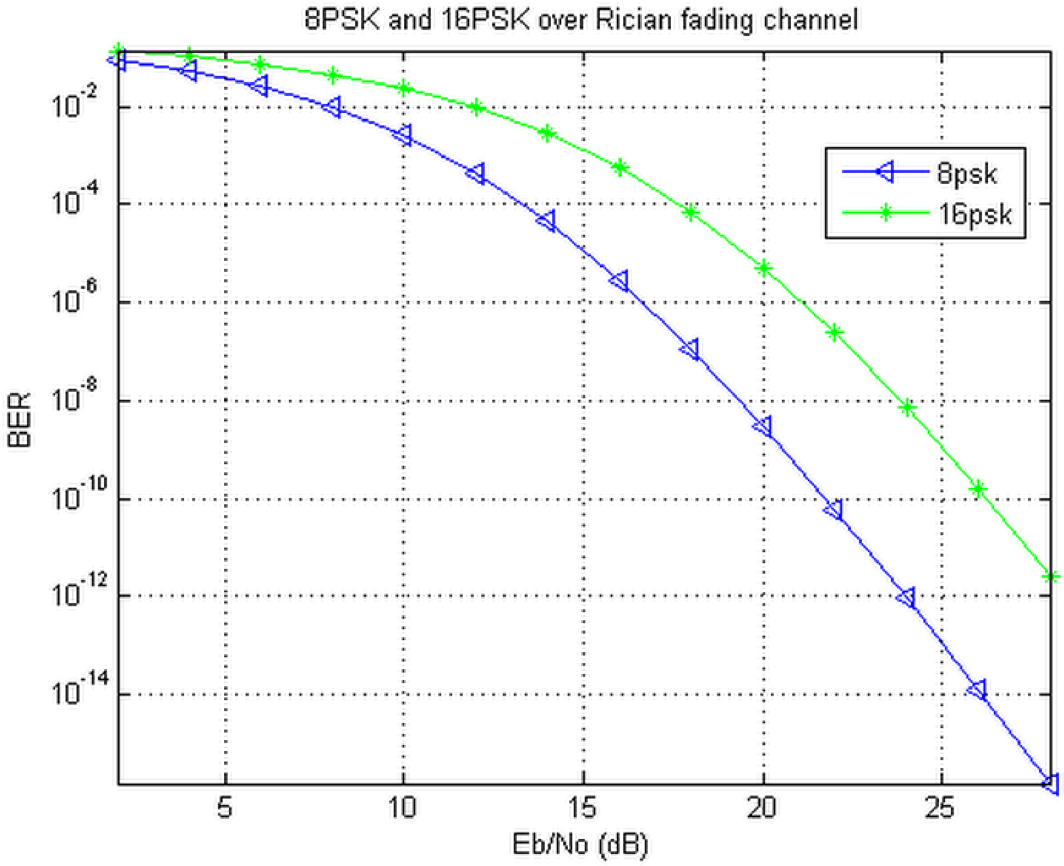


Fig. 5.13 BPSK And 16PSK Over Rician Channel

CHAPTER 6

CONCLUSION

From the results obtained, it can be concluded that as the level of encoding rises in PSK, higher BER is induced in the channel. That is as we move from BPSK to higher order PSKs the BER value is increased as shown in the graph below. This is because in higher order PSKs one symbol itself carries several bits encoded in it. So if one symbol is received in error, several bits can be in error depending on the level of encoding. It is also seen that BPSK and QPSK gives identical performance in terms of BER which makes QPSK the better choice for practical systems as for the same BER performance, it performs encoding also which can be used to reduce the bandwidth required or to double the data rate for a given bandwidth. This observation justifies the use of QPSK in OFDM systems. The graph showing this comparison is included below.

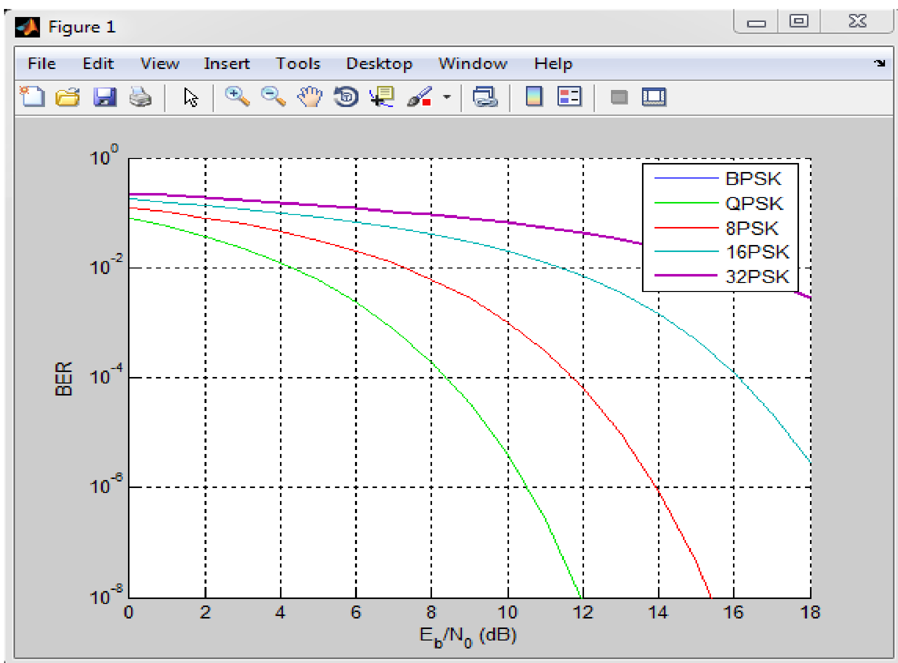


Fig. 6.1 Comparision

Also AWGN channel induces maximum amount of distortion in a signal while Rayleigh channel induces maximum error due to multipath fading. If a modulation scheme is devised which gives good performance with AWGN and Rayleigh channels, it can perform well in almost all real life situations. Rician and Rayleigh fading channels induce more error in the signal compared to AWGN channel. They differ from each other in terms of line of sight. Since Rician channel involves line of sight communication, it gives a much better performance than Rayleigh channel as seen from the graphs. The performance of rician channel depends on its diversity order. As diversity order increases, the error rate also increases.

The BER analysis supporting these observations is given below.

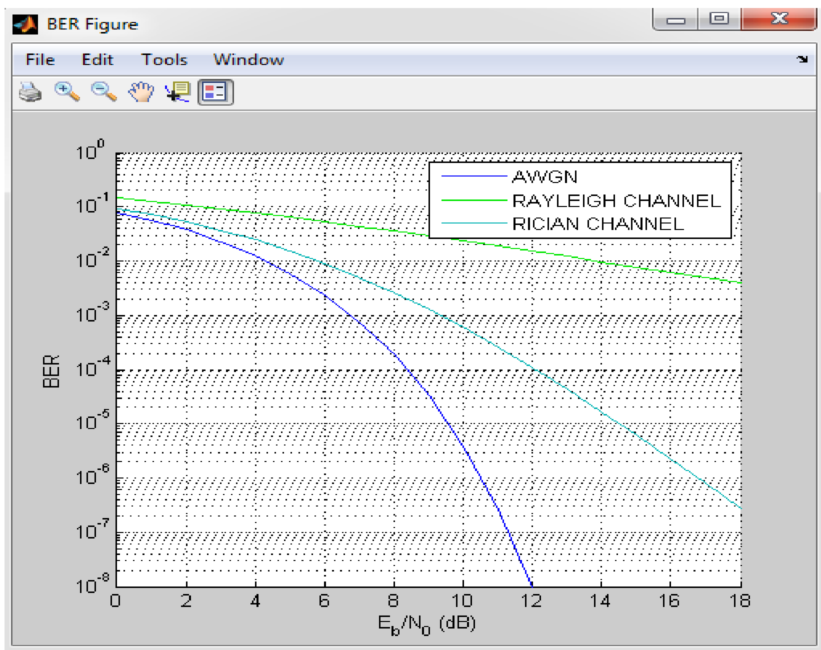


Fig. 6.2 BER Analysis

Application

The above mentioned modulation schemes are applied in a wide variety of digital communication techniques. The analysis performed is crucial while deciding the modulation.

CHAPTER 7

FUTURE SCOPE

As mentioned above, the results obtained in this project are detrimental in deciding the type of Modulation scheme to be used when a transmission scheme is designed for a particular type of channel.

As a further study, the effect of interference on these PSKs can also be studied. Various coding techniques like Reed Solomon coding are applied to the signals to improve their performance. An analysis of such coding schemes can be performed for a better understanding of the way they improve the performance of the system.

A study on various other modulation schemes like SCFDMA can be conducted. The CDMA technique can also be modelled using MATLAB for a better understanding of the underlying technology of 3G communication.

REFERENCES

[1] Iraji, S. and Lilleberg, J. "EM-based multiuser detection and parallel interference cancellation for space-time block coded WCDMA systems employing 16-QAM over multipath fading channels, IEEE conference on Personal, Indoor and Mobile Radio Communications, vol.2, pp.688-692, 2002.

[2] Min-yan Song, Yang Xiao and Habermann, J. "High data rate wireless system, IEEE conference on Signal Processing, vol.2, pp.1344-1350, 2002.

[3] Rosmansyah Y, Sweeney P and Tafazolli, R, "Air-interface techniques for achieving high data rates for UMTS, IEEE conference on 3G Mobile Communication Technologies, pp.368-372, 2001.

[4] Mohr, W. and Becher, R, "Mobile communications beyond third generation", IEEE conference on Vehicular Technology Conference, vol.2, pp.654-661, 2000.

[5] Vikram Kaul, Wenfeng Zhang, Roy D. Yates, "multi-cell WCDMA signal processing simulation", IEEE, 2000.

[6] M. A. Masud, M. Samsuzzaman and M. A.Rahman, "Bit Error Rate Performance Analysis on Modulation Techniques of Wideband Code Division Multiple Access", Journal of Telecommunications, 2010

[7] Mun Geon Kyeong, Jae Joon Park, "3G-Class Space-Time Array Transceiver for High Data Rate Service WCDMA Smart Antenna System", IEEE, 2000.

[8] Rappaport, T. S., Wireless Communications: Principles & Practice, 2nd edition, Prentice Hall.

[9] Bernard Sklar, "Digital Communications: Fundamentals and Applications", Prentice-Hall, 2nd Edition.

[10] Erik Dahlman, Bjorn Gudmundson, Matts Nilsson, and Johan Skold, "UMTS/IMT-2000 Based on Wideband CDMA", IEEE Communications Magazine.

APENDIX

MATLAB CODE:

1. Code for simulation of AWGN channel

```
clear all;
clc;
L=200000;                          %  Length for the random signal sample
sigma=2;
mu=0;
X=sigma*randn(L,1)+mu;             %White Gaussian noise generation
plot(X);
title(['White noise : \mu_x=',num2str(mu),' \sigma^2=',num2str(sigma^2)])
xlabel('Samples')                  % X- axis labelling
ylabel('Values of sample')         % Y- axis labelling
grid on;
```

2. Code for simulation of BPSK

```
clear all;
clc;
set(0,'defaultlinelinewidth',2); % setting line width
A=5;
t=0:.001:1;
f1=20;                 % Frequency of carrier signal
f2=5;                  % Frequency of message signal
x=A.*sin(2*pi*f1*t);         %equation for carrier signal
subplot(3,1,1);
plot(t,x);
xlabel('time');
ylabel('Amplitude');
title('Carrier');
```

```
grid on;
u=square(2*pi*f2*t);          % equation for message signal
subplot(3,1,2);
plot(t,u);
xlabel('time');
ylabel('Amplitude');
title('Message Signal');
grid on;
v=x.*u;              %carrier wave multiplied with message signal
subplot(3,1,3);
plot(t,v);
axis([0 1 -6 6]);
xlabel('t');
ylabel('y');
title('BPSK');
grid on;
```

3. BPSK on AWGN channel

```
close all;
clear all;
SNRdB=1:10;                                   %Signal to Noise Ratio
SNR=10.^(SNRdB/10);                           %Signal to Noise Ratio on Linear Scale
Bit_Length=10^5;                              %Number of Bits Transmitted
BER_Simulated=zeros(1,length(SNRdB));      % Bit Error Rate (Simulated)
%Using Soft detection scheme
%if o/p >=0, +1
%else -1
%Error detection
%BPSK Transmission over AWGN channel
parfor k=1:length(SNR);
    x=(2*floor(2*rand(1,Bit_Length)))-1;
    y=(sqrt(SNR(k))*x)+randn(1,Bit_Length);
```

```
  BER_Simulated(k)=length(find((y.*x)<0));                  %Total number of bits in error
end
BER_Simulated=BER_Simulated/Bit_Length;
semilogy(SNRdB,BER_Simulated,'b-<', 'linewidth' ,2.0);
hold on
semilogy(SNRdB,qfunc(sqrt(SNR)),'m-','linewidth',2.0);      % Bit Error Rate (Theoretical)
title('BPSK over AWGN Simulation');
xlabel('SNR in dB');
ylabel('BER');
legend('BER(Simulated)','BER(Theoretical)')
grid
```

4. Program for simulation of BER of QPSK on AWGN channel

```
clear all;
close all;
 l=1e6;
SNRdB=0:2:10;
SNR=10.^(SNRdB/10);
 for n=1:length(SNRdB)
    si=2*(round(rand(1,l))-0.5);              %In-phase symbol generation
    sq=2*(round(rand(1,l))-0.5);              %Quadrature symbol generation
    s=si+j*sq;                                %Adding the two parallel symbol streams
    w=(1/sqrt(2*SNR(n)))*(randn(1,l)+j*randn(1,l)); %Random noise generation
    r=s+w;                                    %Received signal
    si_=sign(real(r));                        %In-phase demodulation
    sq_=sign(imag(r));                        %Quadrature demodulation
    ber1=(l-sum(si==si_))/l;                  %In-phase BER calculation
    ber2=(l-sum(sq==sq_))/l;                  %Quadrature BER calculation
    ber(n)=mean([ber1 ber2]);                 %Overall BER
end
```

```
semilogy(SNRdB, ber,'o-')                    %Plot the BER
xlabel('SNR(dB)')                            %Label for x-axis
ylabel('BER')                                %Label for y-axis
grid on                                      %Turning the grid on
```

5. Program for simulation of 8PSK on AWGN channel

```
close all;
clear all;
clc;
SNRdB=1:12;                                  %Signal to Noise Ratio in dB
SNR=10.^(SNRdB/10);                          %Signal to Noise Ratio in Linear Scale
Bit_Length=10^6;                             %No. of Bits Transmitted
BER_Simulated=zeros(1,length(SNRdB));        %Simulated Bit Error Rate

%Detection Scheme:(Soft Detection)
%+1 if o/p >=0
%-1 if o/p<0
%Error if input and output are of different signs

%% BPSK Transmission over AWGN channel
parfor k=1:length(SNR);
   x=(3*floor(3*rand(1,Bit_Length)))-1;
   y=((3/2)*sqrt(3*SNR(k))*x*sin(pi/8))+randn(1,Bit_Length);

   BER_Simulated(k)=length(find((y.*x)<0));          %Total number of bits in error
end
BER_Simulated=BER_Simulated/Bit_Length;
semilogy(SNRdB,BER_Simulated,'m-<', 'linewidth' ,2.0);
hold on

semilogy(SNRdB,qfunc(sqrt((3*SNR)*sin(pi/8))),'b-');      %Theoretical Bit Error Rate
```

```
title('8PSK over AWGN Simulation');xlabel('SNR in dB');ylabel('BER');
legend('BER(Simulated)','BER(Theoretical)')
grid
```

6. Simulation of 16PSK on AWGN channel

```
close all;
clear all;
clc;
SNRdB=1:12;                          %Signal to Noise Ratio in dB
SNR=10.^(SNRdB/10);                        %Signal to Noise Ratio in Linear Scale

semilogy(SNRdB,qfunc(sqrt((4*SNR)*sin(pi/16))),'b-','linewidth',2.0);     %Theoretical Bit Error
Rate
title('16PSK over AWGN Simulation');xlabel('SNR in dB');ylabel('BER');
legend('BER(Theoretical)')
grid
```

6. BPSK on rayliegh and AWGN channel

```
clear
N = 10^6 % number of bits or symbols
% Transmitter
ip = rand(1,N)>0.5; % generating 0,1 with equal probability
s = 2*ip-1; % BPSK modulation 0 -> -1; 1 -> 0
Eb_N0_dB = [-3:35]; % multiple Eb/N0 values
for ii = 1:length(Eb_N0_dB)
  n = 1/sqrt(2)*[randn(1,N) + j*randn(1,N)]; % white gaussian noise, 0dB variance
  h = 1/sqrt(2)*[randn(1,N) + j*randn(1,N)]; % Rayleigh channel
  % Channel and noise Noise addition
  y = h.*s + 10^(-Eb_N0_dB(ii)/20)*n;
  % equalization
  yHat = y./h;
```

```
   % receiver - hard decision decoding
   ipHat = real(yHat)>0;
   % counting the errors
   nErr(ii) = size(find([ip- ipHat]),2);
end
simBer = nErr/N; % simulated ber
theoryBerAWGN = 0.5*erfc(sqrt(10.^(Eb_N0_dB/10))); % theoretical ber
EbN0Lin = 10.^(Eb_N0_dB/10);
theoryBer = 0.5.*(1-sqrt(EbN0Lin./(EbN0Lin+1)));
% plot
close all
figure
semilogy(Eb_N0_dB,theoryBerAWGN,'cd-','LineWidth',2);
hold on
semilogy(Eb_N0_dB,theoryBer,'cd-','LineWidth',2);
semilogy(Eb_N0_dB,simBer,'mx-','LineWidth',2);
axis([-3 35 10^-5 0.5])
grid on
legend('AWGN-Theory','Rayleigh-Theory', 'Rayleigh-Simulation');
xlabel('Eb/No, dB');
ylabel('Bit Error Rate');
title('BER for BPSK modulation in Rayleigh channel');
```

7. QPSK over Rayleigh channel

```
clear all;
close all;
N = 10^5;   % Total no of of bits per simulation per SNR_dB
Eb_No_dB = [0:20]; % multiple Eb/N0 values
EbNo_lin = 10.^(Eb_No_dB/10);
Ber_AWGN = 0.5*erfc(sqrt(EbNo_lin/2)); % theoretical BER of QPSK over AWGN channel
```

```
Ber_Rayl = 0.5.*(1-sqrt(EbNo_lin./(EbNo_lin+1))); % theoretical BER of QPSK over Rayleigh channel % plotting the simulated results
semilogy(Eb_No_dB, Ber_AWGN,'g--*');
legend('QPSK over AWGN channel');
hold on semilogy(Eb_No_dB, Ber_Rayl,'b-<');
legend('QPSK over Rayleigh channel');
axis([0 20 10^-5 0.5])
 xlabel('Eb/No, dB');
ylabel('Bit Error Rate');
title('BER for QPSK modulation in Rayleigh channel');
axis tight;
grid on;
```

8. PSK on Rayleigh channel

```
close all;
clear all;
EbNo=2:2:20;
ber = berfading(EbNo,'psk',8,1)
semilogy(EbNo,ber,'m-<');

title('8PSK over Rayleigh fading channel');
xlabel('E_b/N_o (dB)');
ylabel('BER');
axis tight;
grid on;
```

9. SIMULATION OF 16PSK ON RAYLEIGH CHANNEL

```
close all;
clear all;
EbNo=2:2:20;
ber = berfading(EbNo,'psk',16,1)
semilogy(EbNo,ber,'g-<');
```

```
title('16PSK over Rayleigh fading channel');
xlabel('E_b/N_o (dB)');
ylabel('BER');
axis tight;
grid on;
```

10. simulation on Rician fading channel

```
close all;
clear all;
EbNo=2:2:20;
ber = berfading(EbNo,'psk',2,5)
semilogy(EbNo,ber,'m-<');

title('BPSK over Rician fading channel');
xlabel('E_b/N_o (dB)');
ylabel('BER');
axis tight;
grid on;
```

11. QPSK ON RICIAN CHANNEL

```
close all;
clear all;
EbNo=2:2:28;
ber = berfading(EbNo,'psk',4,10)
semilogy(EbNo,ber,'b-<');

title('QPSK over Rician fading channel');
xlabel('E_b/N_o (dB)');
ylabel('BER');
axis tight;
grid on;
```

12. 8PSK &16PSK on Rician Channel simulation

```
close all;
```

```
clear all;
EbNo=2:2:28;
ber = berfading(EbNo,'psk',8,10)
ber2 = berfading(EbNo,'psk',16,10)
semilogy(EbNo,ber,'b-<');

hold on;

semilogy(EbNo,ber2,'g-*');

title('8PSK and 16PSK over Rician fading channel');
xlabel('Eb/No (dB)');
ylabel('BER');
axis tight;
grid on;
```