Implementation of Digital Modulation Technique and Calculate the Bit Error Rate Performance using Matlab

Ali Kamal Taqi [1]

Abstract— With the increasing demand in communication, it has become necessary to give better and efficient service to users by using better technique. This paper demonstrates different modulation technique including Amplitude Shift Keying (ASK), Frequency Shift Keying (FSK), Phase Shift Keying (PSK) and analyze the Bit Error Rate (BER) for different modulation schemes such as Binary Phase Shift Keying (BPSK), Quadrature Phase Shift Keying (QPSK), Octal Phase Shift Keying (8PSK), and Trellis Coded Modulation (TCM). By Choosing a reliable modulation scheme and better coding technique the enhancement of the performance can be obtained in transmitter and receiver of the system. Simulated result is shown to analyze and compare the performance of these systems by using additive white Gaussian noise channel (AWGN). Finally, the different modulation schemes are compared on the basis of BER and best modulation scheme is determined. From analysis of modulation techniques, the system could use more appropriate modulation technique to suit the channel quality, thus we can deliver the optimum and efficient system parameters. Both Matlab Code and Simulink have been used for simulation.

Index Terms— Digital Modulation, ASK, FSK, PSK, Bit Error Rate (BER), TCM.

I. INTRODUCTION

In the simplest type, a transmission-reception system may be a three-block system, consisting of a) a transmitter, b) a transmission medium and c) a receiver. If we predict of a mix of the transmission device and reception device within the sort of a 'transceiver' and if (as is sometimes the case) the transmission medium permits signal each ways that, we tend to are in an exceedingly position to consider a both-way (bi-directional) communication system. For simple description, we are going to discuss a few unidirectional transmission-reception systems with the implicit assumption that, once understood, the concepts are used for developing / analyzing two-way communication systems. So, our representative communication system, in an exceedingly straightforward type, once more consists of 3 completely different entities, viz. a transmitter, a channel and a receiver. A data communication system has many distinctive options when put next with associate analog communication system. each analog (such as voice signal) and digital signals (such as knowledge generated by computers) is communicated over a digital gear mechanism. once the signal is analog in nature, constant discrete-time discrete-amplitude illustration is feasible when the initial process of sampling and quantization. So, each a digital signal and a quantal analog signal are of comparable kind, i.e. discrete-time-discrete-amplitude signals. A key feature of a data communication system is that a way of 'information', with acceptable unit of live, is related to such signals. This image, attributable to Claude E. Shannon, ends up in many fascinating schematic description of a data communication system. parenthetically, take into account Fig.1.1 that shows the signal supply at the transmission finish as constant 'Information Source' and also the receiving user as associate 'Information sink'. the general purpose of the data communication system is 'to collect data from the supply and perform necessary signal process specified the data is delivered to the top user (information sink) with acceptable quality'. One could note of the compromising phrase 'acceptable quality' and surprise why a digital gear mechanism mustn't deliver precisely the same data to the sink as accepted from the supply. A broad and general answer to such question at now is: well, it depends on the designer's understanding of the 'channel' (Fig. 1.1) and the way the designer will translate his data to style the signal process algorithms / techniques within the 'Encoder' and 'decoder' blocks in Fig. 1.1 we tend to hope to select up many basic nonetheless smart approaches to amass the higher than skills. However, pioneering add the 1940-s and 1950-s have established a bottom-line to the look for 'a unflawed (equivalently, 'error-less') data communication system' delivery out many profound theorems (which currently get into the name of knowledge Theory) to ascertain that, whereas error-less transmission of knowledge will ne'er be secured, the other 'acceptable quality', haphazardly getting ready to error-less transmission could also be doable. This 'possibility' of virtually error-less data transmission has driven important analysis over the last 5 decades in multiple connected areas similar to, a) digital modulation schemes, b) error management techniques, c) optimum receiver style, d) modeling and characterization of channel so forth. As a result, varieties of electronic communication systems are designed and place to use over the years and therefore the overall performance has improved considerably [1].

[1] *Computer Engineering Department, University of Technology, Iraq*
Email: alikamal1@yahoo.com

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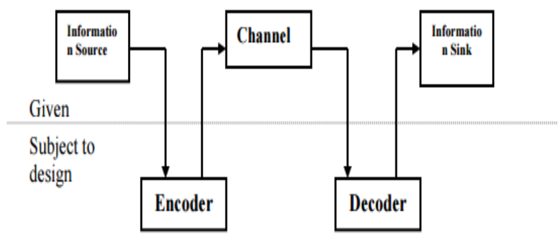

Figure 1.1: Basic block diagram of a digital communication system

II. Digital Communication System

It is attainable to expand our basic 'three-entity' description of an electronic communication system in multiple ways in which. as an example, Fig. 2.1 shows a somewhat elaborate diagram expressly showing the vital processes of 'modulation demodulation', 'source coding-decoding' and 'channel cryptography – decoding'. A reader might have multiple queries with reference to this sort of abstraction. as an example, once 'information' has got to be sent over an outsized distance, it's a standard information that the signal ought to be amplified in terms of power and so began the physical transmission medium. Diagrams of the kind in Figs. 1.1 and 2.1 have no explicit reference to such issues. However, the issue here is more of suitable representation of a system for clarity rather than a module-by-module replication of an operational digital communication system.

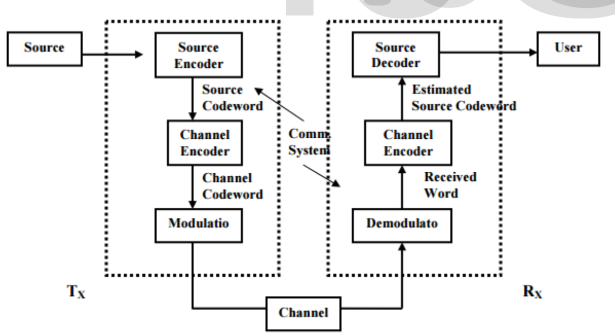

Figure 2.1: A possible breakup of the pervious diagram (following Shannon's ideas)

To elaborate this potentially useful style of representation, let us note that we have hardly discussed about the third entity of our model, viz. the 'channel'. One can define several types of channel. For example, the 'channel' in Fig. 2.1 should more appropriately be called as a 'modulation channel' with an understanding that the actual transmission medium (called 'physical channel'), any electromagnetic (or otherwise) transmission- reception operations, amplifiers at the transmission and reception ends and any other necessary signal processing units are combined together to form this 'modulation channel'. We will see later that a modulation channel usually accepts modulated signals as analog waveforms at its inputs and delivers another version of the modulated signal in the form of analog waveforms. Such channels are also referred as 'waveform channels'. The 'channel' in Fig. 1.1, on the other hand, appears to accept some 'encoded' information from the source and deliver some 'decoded' information to the sink. Both the figures are potentially useful for describing the same digital communication system. On comparison of the two figures, the reader is encouraged to infer that the 'channel' in Fig. 1.1 includes the 'modulation channel' and the modulation-demodulation operations of Fig. 2.1 The 'channel' of Fig. 1.1 is widely denoted as a 'discrete channel', implying that it accepts discrete-time-discrete-amplitude signals and also delivers discrete-time discrete-amplitude signals [2].

III. Modulation Technique

A. *Amplitude-Shift Keying (ASK) Modulation*

The transmission of digital signals is increasing at a speedy rate. Low-frequency analogue signals area unit typically born-again to digital format (PAM) before transmission. The supply signals area unit typically noted as baseband signals. Of course, we will send analogue and digital signals directly over a medium. From electro-magnetic theory, for economical radiation of current from associate degree antenna it should be a minimum of within the order of magnitude of a wavelength in size;

$c = f\lambda$, wherever c is that the rate of sunshine, f is that the signal frequency and λ is that the wavelength. For a 1kHz audio signal, the wavelength is three hundred kilometer. associate degree antenna of this size isn't sensible for economical transmission. The low-frequency signal is commonly frequency-translated to the next frequency vary for economical transmission. the method is named modulation. the utilization of the next frequency vary reduces antenna size.

In the modulation method, the baseband signals represent the modulating signal and therefore the high-frequency carrier signal may be a curved wave. There are a unit 3 basic ways that of modulating a undulation carrier. For binary digital modulation, they're referred to as binary amplitude-shift keying (BASK), binary frequency-shift keying (BFSK) and binary phase-shift keying (BPSK). Modulation conjointly results in the likelihood of frequency multiplexing. in a very frequency-multiplexed system, individual signals area unit transmitted over adjacent, non-overlapping frequency bands. they're thus transmitted in parallel and at the same time in time. If we tend to operate at higher carrier frequencies, a lot of information measure is obtainable for frequency-multiplexing a lot of signals.

A binary amplitude-shift keying (BASK) signal is outlined by

$$s(t) = A\, m(t) \cos 2\pi f_c t, \qquad 0 \le t \le T$$

Where A is constant, m(t)=1 or 0 , f_c is the carrier frequency and T is the bit duration. It has a power P= $A^2/2$, so that A= $\sqrt{2P}$ this equation can be written as

$$S(t)= \sqrt{2P} \cos 2\pi f_c t \qquad 0 \le t \le T$$
$$= \sqrt{PT}\sqrt{2/T} \cos 2\pi f_c t \qquad 0 \le t \le T$$
$$= \sqrt{E}\, \sqrt{2/T} \cos 2\pi f_c t \qquad 0 \le t \le T$$

The output of the Matlab code for BASK modulation shown in figure (3.1) has the following parameters: Carrier frequency=8 with Message frequency=4

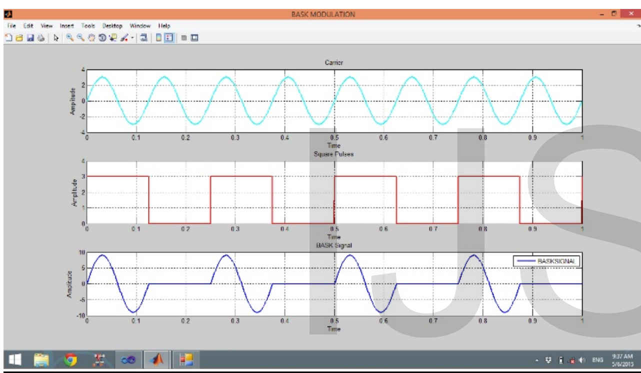

Figure 3.1: BASK Modulation

B. *Frequency-Shift Keying (FSK) Modulation*

Frequency-shift keying (FSK) may be a modulation theme during which digital data is transmitted through distinct frequency changes of a radio emission. the only FSK is binary FSK (BFSK). BFSK uses a combine of distinct frequencies to transmit binary (0s and 1s) data
A binary frequency-shift keying (BFSK) signal can be defined by

$$S(t) = \begin{cases} A \cos 2\pi f_0 t \\ A \cos 2\pi f_1 t \end{cases}$$

where A is a constant, f_0 and f_1 are the transmitted frequencies, and T is the bit duration. duration. The output of the Matlab code for BFSK modulation shown in figure (3.2) has the following parameters: 1st Carrier frequency=16, 2nd Carrier frequency=8 with Message frequency=4

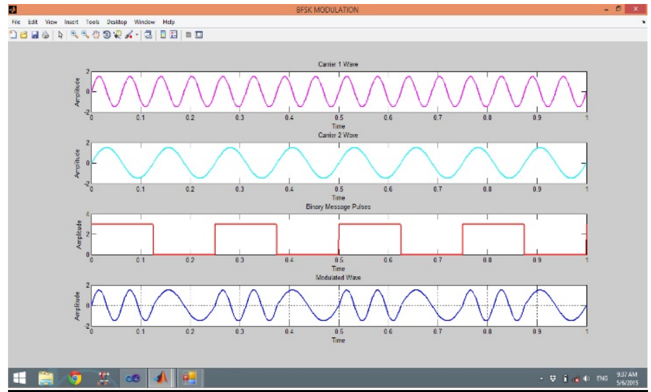

Figure 3.2: BFSK Modulation

C. *Phase-Shift Keying (PSK) Modulation*

Phase-shift keying (PSK) may be a digital modulation theme that conveys information by dynamical, or modulating, the section of a reference signal (the carrier wave). Any digital modulation theme uses a finite range of distinct signals to represent digital information
A *binary phase-shift keying* (BPSK) signal can be defined by

$$s(t) = A\, m(t) \cos 2\pi f_c t \qquad 0 \le t \le T$$

where A is a constant, $m(t)$ = +1 or -1, f_c is the carrier frequency, and T is the bit duration. The signal has a power $P = A^2/2$, so that $A = \sqrt{2P}$.

The output of the Matlab code for BPSK modulation shown in figure (3.3) has the following parameters: Carrier frequency=8 with Message frequency=4

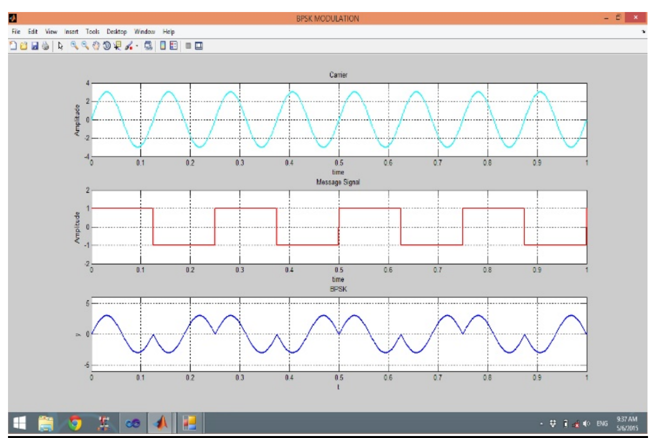

Figure 3.3: BPSK Modulation

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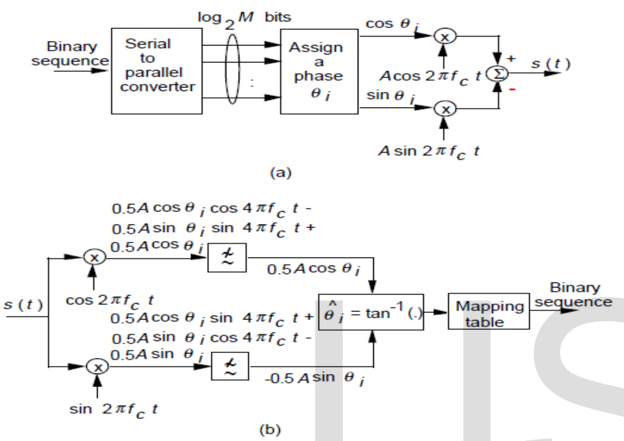
D. M-ary Phase-Shift Keying (M - PSK)

An *M-ary phase-shift keying* (*M*-PSK) signal can be defined by

$$s(t) = \begin{cases} A\cos\left(2\pi f_c t + \theta_j + \theta'\right), & 0 \le t \le T \\ 0 & elsewhere \end{cases}$$

where $\theta_j = \frac{2\pi}{M} i$

for i = 0, 1, ..., M - 1. Here, A is a constant, f_c is the carrier frequency, θ' is the initial phase angle, and T is the symbol duration.



(a) *M*-PSK modulator, and (b) coherent demodulator.

The output of the Matlab code for QPSK modulation shown in figure (3.4) has the following parameters: Carrier frequency=8

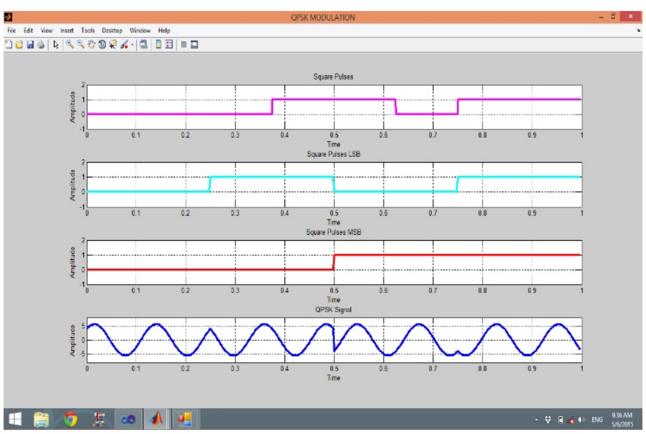


Figure 3.4: QPSK Modulation

The output of the Matlab code for 8PSK modulation shown in figure (3.5) has the following parameters: Carrier frequency=8 with Message input= [0 2 5 1 6 3 4 7]

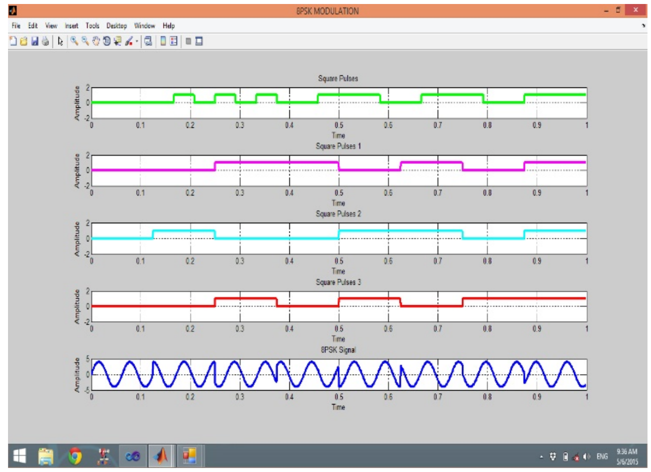


Figure 3.5: 8PSK Modulation

IV. BIT ERROR RATE

The bit error rate (BER), or perhaps more appropriately the bit error ratio, is the number of bits received in error divided by the total number of bits transferred. We can estimate the BER by calculating the probability that a bit will be incorrectly received due to noise.

A. BPSK BER

The output of the Matlab code for BPSK BER shown in figure (4.1) has the following parameters: SNR from 1 to 20, Number of Bit=1000000 and Energy Bit=1

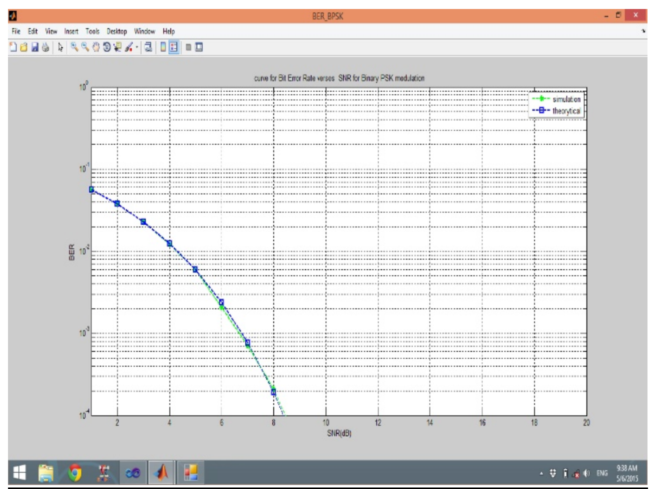


Figure 4.1: BPSK Bit Error Rate

The Matlab Simulink block diagram of the for BPSK shown in figure (4.2) with transmitted and received bits in figure (4.3) and the BER performance in figure (4.4)

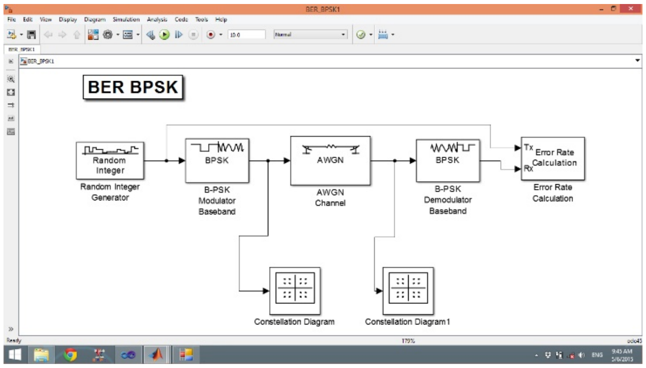


Figure 4.2: BPSK Block Diagram

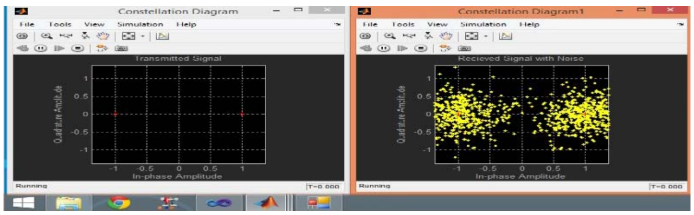


Figure 4.3: BPSK Transmitted and Received Bits

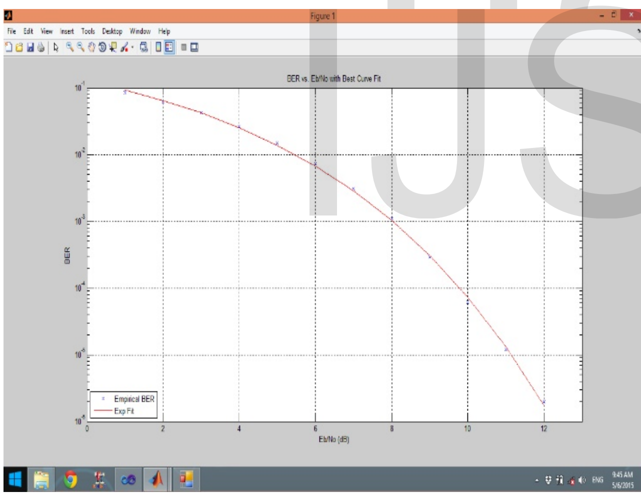


Figure 4.4: BPSK BER Performance

B. QPSK BER

The output of the Matlab code for QPSK BER shown in figure (4.5) has the following parameters: SNR from 1 to 20 , Number of Bit=1000000 and Energy Bit=1

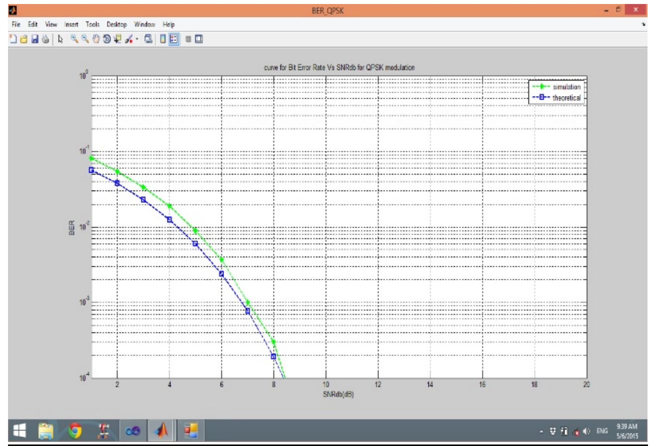


Figure 4.5: QPSK Bit Error Rate

The Matlab Simulink block diagram of the for QPSK shown in figure (4.6) with transmitted and received bits in figure (4.7) and the BER performance in figure (4.8)

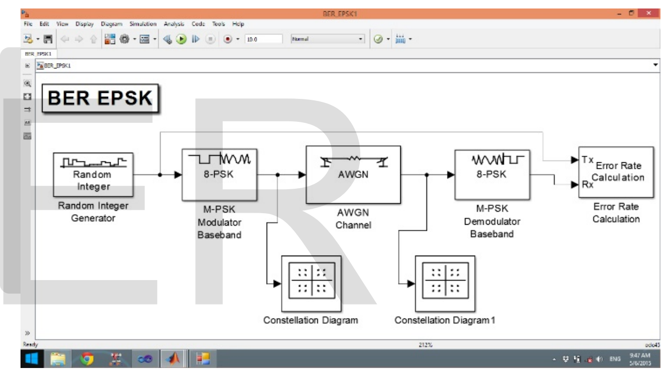


Figure 4.6: QPSK Block Diagram

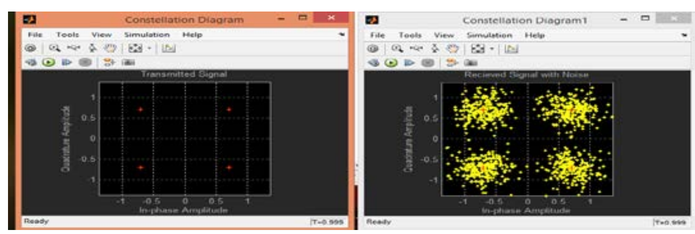


Figure 4.7: QPSK Transmitted and Received Bits

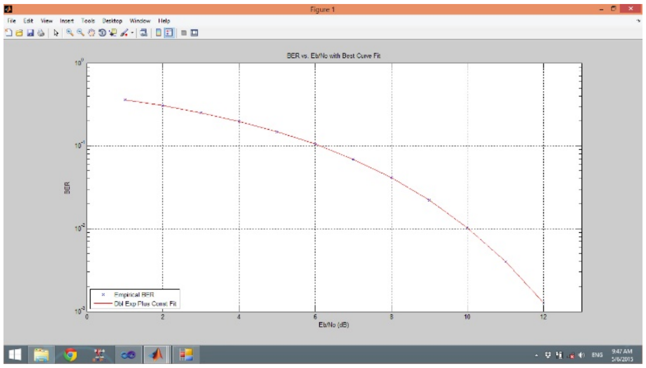


Figure 4.8: QPSK BER Performance

C. 8PSK BER

The output of the Matlab code for 8PSK BER shown in figure (4.9) has the following parameters: SNR from 1 to 20 , Number of Bit=1000000 and Energy Bit=1

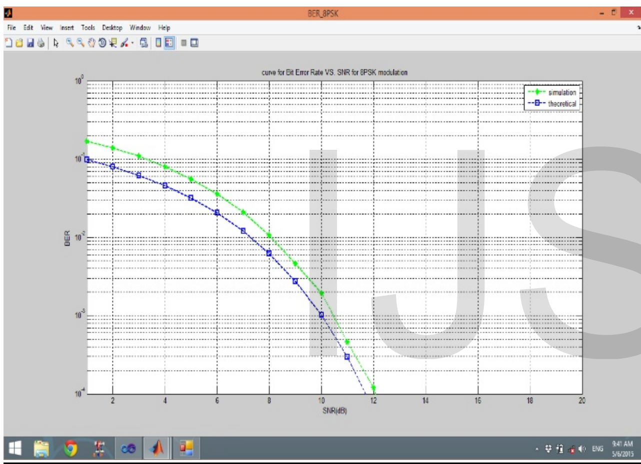


Figure 4.9: 8PSK Bit Error Rate

The Matlab Simulink block diagram of the for QPSK shown in figure (4.10) with transmitted and received bits in figure (4.11) and the BER performance in figure (4.12)

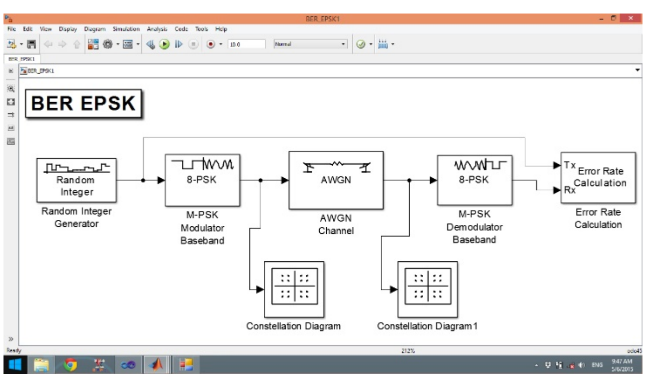


Figure 4.10: 8PSK Block Diagram

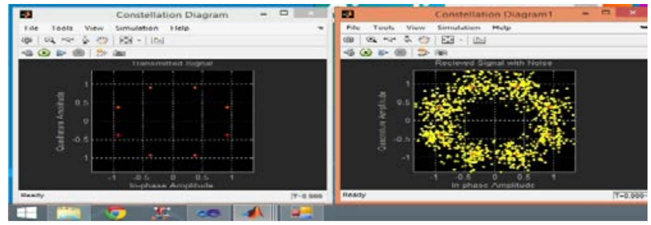


Figure 4.11: 8PSK Transmitted and Received Bits

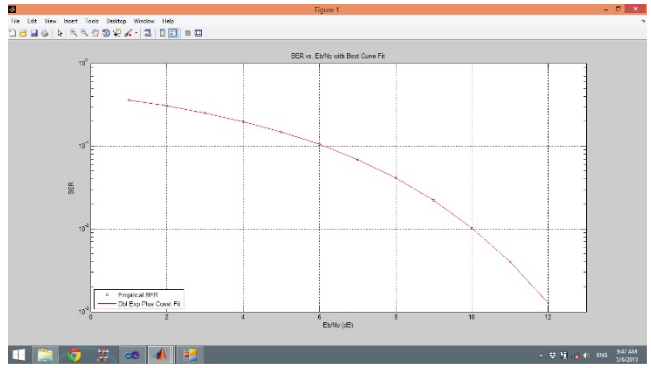


Figure 4.12: 8PSK BER Performance

using Matlab Simulink for modeling the modulation process and Bit Error Rate performance by using the following block parameter showing in table 3.1

Table 3.1: Input Parameter Table of the Matlab Simulink Block

Block Name	Parameter Name	BPSK | QPSK | 8PSK
Random Integer Generator	M-ary Number Initial Seed Sample Time Sample per Frame Output data type	2 | 4 | 8 37 1 1e6 Double
BPSK modulator Baseband	M-ary Number Phase offset (rad) Constellation ordering Input Type Output Type	2 | 4| 8 Pi/2 | Pi/4 | Pi/8 Binary Integer Double
AWGN Channel	Initial Seed Mode Eb/No (dB) Number of bit per Symbol Input Signal Power (Watt) Symbol Period	37 Signal to Noise Ratio (Eb/No) EbNodB 1 1
MPSK Demodulator Baseband	M-ary Number Phase offset (rad) Constellation ordering Output Type	2 | 4 | 8 Pi/2 | Pi/4 | Pi/8 Binary Integer
Error Rate Calculation	Receive Delay Computation Delay Computation Mode Output Data Variable Name Stop Simulation Target Number of Errors Maximum Number of Symbols	0 0 Entire Frame Workspace BER_DATA Checked Inf 1e6

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V. CONVOLUTIONAL ENCODER TCM [3]

A Convolutional encoder accepts a sequence of n bits and it produces at its output l binary coded digits at any time. In general, a convolutional encoder with a memory of L bits may be considered as a finite-state machine (or finite memory system rather than memory less system, as in the case of the block encoder) with 2^l possible states. The state of the encoder at any time instant is determined by the contents of its store (delay unit) at that time instant.

Let the n input digits form the n-component vector

$$a_i = [a_{i,1} \quad a_{i,2} \; .\,.\,.\,.\,.\,.\,.\,.\,.\,. \; a_{i,n}] \qquad ......(5.1)$$

And let the l output digits form the l-component vector

$$\beta_i = [\,\beta_{i,0} \;\; \beta_{i,1} \; .\,.\,.\,.\,.\,.\,.\,.\,. \beta_{i,l-1}] \qquad ......(5.2)$$

Also let the state of the encoder be defined as the L-component vector

$$\mu_i = [\,\mu_{i,0} \;\; \mu_{i,1} \, .\,.\,.\,.\,.\,.\,.\,.\,. \mu_{i,L-1}] \qquad ......(5.3)$$

Where the binary digits $\alpha_{i,h}$,$\beta_{i,h}$ and $\mu_{i,h}$ may take any one of their two possible values 0 and 1.

The operation of the convolutional encoder may now be described as follows: for each input sequence α_i ,the encoder generates the sequence β_i at its output ,while changing its state from μ_i to its next state μ_{i+1} .since for every n input bits , l bits are produced by the encoder ,so the rate of the convolutional encoder is R=n/ l.

As an example, Figure (5.1) shows a 4-state convolutional the minimum squared Euclidean distance sometime called the free Euclidean distance is defined as

$$d_1^2 = Min_{i=j}.\, |S_i\text{-}S_j|^2 \qquad \text{for all I,j} \qquad ......(5.4)$$

Where S_i and S_j assume all valid pairs of coded sequences that the convolutional encoder/modulator combination can produce and excludes all the cases whrer the two sequences are identical . and $|S_i\text{-}S_j|$ is the unitary distance between the two sequences S_i and S_j

The asymptotic coding gain of the coded system over the corresponding uncoded system is given by

$$G_c \text{ (dB)}=10Log_{10}(d_f^2 \,/ d_{un}^2) \qquad ......(5.5)$$

Where d_f is given by Eq. 5.4, and d_{un} is the minimum Euclidean distance of the uncoded system .Here Eq.5.5 assume that the average transmitted signal energy of the coded and uncoded system is the same.

For an example to find the d_f and G_c for a coded 4-PSK signal with signal constellation (M=4) .let us consider the encoder in Figure (5.2) and its state-transition diagram in Figure(5.3).

The d_f can be calculated by assuming the correct state as the all-zero state.

$$d_f^2 = d^2(0,3) + d^2(0,0) + d^2(0,2) \qquad ......(5.6)$$

$$=2+0+4=6$$

Where d^2(i,j) is the square Euclidean distance between the signal points i and j , and I (or j) is the decimal representation of the output coded digits.

The asymptotic coding gain of the coded 4-PSK (in this example) over the uncoded 2-PSK with d_{un}^2 =4 is

$$G_c = 10\,Log_{10} \;\; (6/4)= 1.7609 \text{ dB} \qquad ......(5.7)$$

Thus an advantage of about 1.76 dB in tolerance to AWGN can be obtain with coding

Viterbi Decoder

The Viterbi decoding technique is one approach to the maximum likelihood detection of convolutional codes in general, the Viterbi decoder operates by tracing through a trellis identical to that at the encoder in an attempt to emulate the encoder's behavior. At any given instant the decoder does not know the state of the encoder and does not try to decode this immediately. The decoder examines all possible branches to each state in the trellis stage. For each state .it computes a likelihood score, known as the branch cost (branch metric), from the received data and the data corresponding to each branch.

Each state has associated with it accost, which is the sum of the surviving branches cost up to that state (the surviving branches are the branches which produce the smallest new state cost). To determine the surviving branches, the branches cost is added to the state cost in the previous stage of the trellis.

After evaluating a number of stages of the trellis the surviving paths in the latest stage will originate (with a high probability) from a single state in the first stage of the trellis. After this. The most likely state of the encoder in the first stage will be *known, and hence it can be deduced the original input data, even* though a decision has not yet been made for the latest

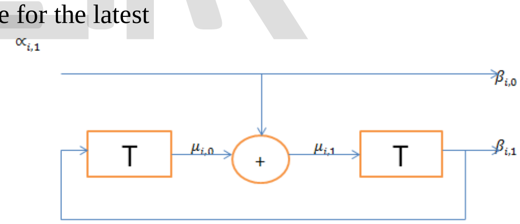

Figure 5.1: Four-State, rate ½ convolutional encoder

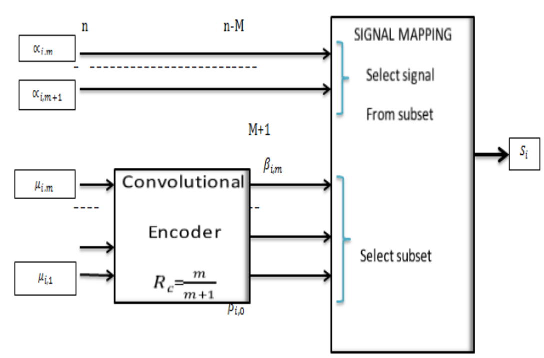

Figure 5.2: General Structure of TCM

Table 5.1: Truth Table of the convolutional encoder

State at time iT μ_i	Input at time iT α_i	Output at time iT β_i	State at Time(i +1)T μ_{i+1}
0	0	0	0
	1	3	1
1	0	0	2
	1	3	3
2	0	1	1
	1	2	0
3	0	1	3
	1	2	2

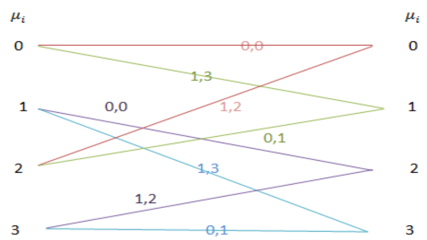

Figure 5.3: State-transition diagram of the encoder i

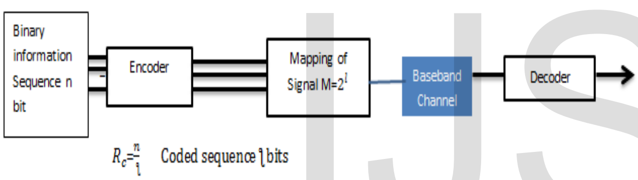

Figure 5.4: Model of combined encoding and modulation

VI.

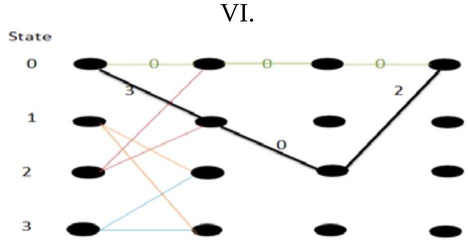

Figure 5.5: calculation the minimum free distance of the encoder in Figure (5.1)

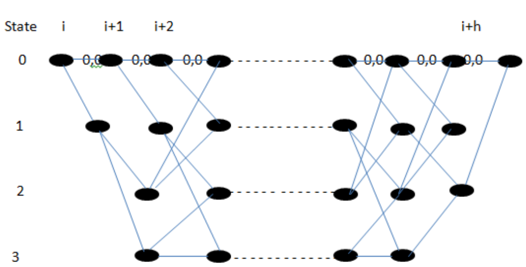

Figure 5.6: Trellis diagram of the encoder assuming the initial and terminal states are zero

The Matlab Simulink block diagram of the for TCM shown in figure (5.7) with the BER performance in figure (5.8)

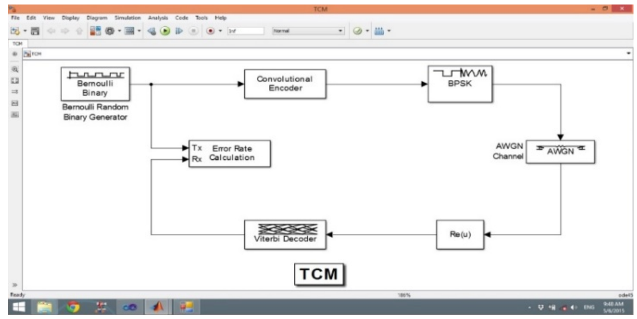

Figure 5.7: TCM Block Diagram

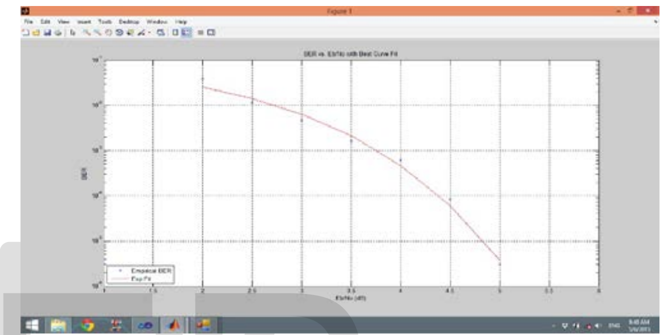

Figure 5.8: TCM BER Performance

VII. Conclusion

In this paper presents, error performance of modulation techniques with AWGN channel are analyzed and BER is calculated. Based on numerical calculation, the BER of BPSK, QPSK, 8PSK and TCM was graphically plotted and compared. The real measurements have achieved the results which have accepted. According to this work, a performance of different modulation techniques and channel coding is analyzed on the basis of BER over AWGN channel. From the analysis of different modulation techniques, we can say BPSK gives better performance with compared to QPSK and 8PSK over AWGN channel. Also we have limitation to increase Eb/No ratio. Hence, for a fixed value of Eb/No, we have to use some kind of coding to improve quality of the transmitted signal by using Trellis Coded Modulation TCM which gives better BER performance compared to uncoded modulation methods.

VIII. References

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