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INVESTIGATE THE PERFORMANCE OF 16-PHASE SHIFT KEYING (PSK) MODULATION TECHNIQUE IN AWGN ENVIRONMENT WITH AND WITHOUT CHANNEL CODING

ALI OTHMAN ALBAJI
Department of Electronics & Telecommunications, The Higher Institute of science & Technology- Suk Algumaa, Tripoli, Libya. Email: ahmedali@graduate.utm.my/sabos_in@yahoo.com

ROZEHA BT. A. RASHID
Department of Telecommunication Software and Systems (TeSS) Research Group, Faculty of Engineering, Universiti Teknologi Malaysia, Johor Bahru, Malaysia. E-mail: rozeha@fke.utm.my

ALZHARI ALSHAREEF
Department of Electronic and Telecommunication, National Authority for Technical Education Technical Faculties Administration Engineering Technology Faculty, Libya. E-mail: tabaqah7@gmail.com

AKRAM H. JEBRIL
Department of Communication Engineering, Technical College of Civil Aviation & Meteorology, Esbea, Tripoli, Libya. E-mail: jebrillakram@gmail.com

ABDUSALAMA DAHO
Department of Electrical and Electronic, Sebha University Faculty of Engineering, Sebha, Libya.
E-mail: abdu.daho@sebhau.edu.ly

YASEEN HADI ALI
Department of Computer Techniques Engineering, Telecommunication Software and System, Alsalam University College, Baghdad, Iraq. E-mail: yaseenhadi5@gmail.com.

Abstract

This paper aims to investigate the performance of 16-Phase Shift Keying (PSK) modulation technique in AWGN environment with and without channel coding. The parameters are chosen to observe the error performance of the system. A first-order error correction results in a maximum absolute phase error from 0dB to 15dB at an IF of up to 400 MHz, without any additional or external adjustments. Using MATLAB Simulink has given us a brief description of how the system is modeled in the MATLAB environment and what parameters are chosen to observe the error performance of the system. Findings showed that the convolutional-coded modulation is more efficient than the uncoded modulation. An increase in the code rate of the encoder increases the BER.

Keywords: PSK, AWGN, Performance, MATLAB, BER

1.INTRODUCTION

Communication means transmission, reception and analysis of information by electronics means. There are lot of equipment around us like telephone, radio, mobile, television etc. which involves the communication. Communication provides senses for ships on the high seas, aircraft in flight, and rocket and satellite in space. Communication through a wireless telephone keeps a car driver in touch [1]. There are lots of applications of communication in the world. Communication is classified into two

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parts viz. Analog Communication and Digital Communication. The increasing number of technologies and the need for faster data transmission are some of the factors that have led to the development of new communication systems. In order to meet the increasing demand for data transmission, the engineers of these systems are working on developing various technologies that can handle high data rates [2]. In order to ensure that the information sent by a receiver is always accurate, a communication system needs to transfer the data from the transmitter to the receiver in a way that's both reliable and accurate. Unfortunately, this can be very challenging due to the presence of distortion. One of the most common methods that can be used to achieve a minimum error-free transmission is by implementing error control coding. This type of coding can be used in order to ensure that the information sent by a receiver is always accurate [3].

Claude Shannon introduced the concept of data transmission codes in 1948 [4]. He defined the theoretical channel capacity, which refers to the maximum amount of data that can be transmitted across a given channel. According to Shannon and his colleagues, the transmission rate required by a communication is less than the capacity of a channel C, which is measured in bits per second. A coding scheme cannot achieve reliable transmission if the R is greater than C. The main objective of channel coding is to ensure that the information sent across a channel is encoded in a way that can be detected and rectified by channel noise [5]. There are two types of error correction codes that can be used to combat these errors. The automatic repeat request is a type of error detection system that tries to determine the presence of these errors by analyzing the data. If the errors are found at the receiver side, the receiver sends an acknowledgement to the transmitter. The receiver then resends the data until the errors are resolved.

Forward error correction, also known as FEC, is a second type of error correction that can be performed to correct or detect errors [6]. In most applications, this type of error correction is not required. For instance, in a broadcasting system, it is usually not feasible to retransmit data. With the use of simplex communication, FEC codes can be the only solution.

The main purpose of a channel code is to add redundancy to the data stream in a digital communication system. This process can be done by implementing extra bits in the channel to minimize the error in the data stream. Ideally, the channel codes should have high bit rate and low complexity to maximize data throughput and minimize the energy consumption of the transmission.

The advantages of computational codes are typically found in combination with other techniques, such as channel coding. Convolutional codes are typically described by two parameters: the code rate and the constraint length. The code rate is a ratio between the number of input bits and the number of output bits. The constraint length is represented by K, which is the number of shift registers that are used in the coding part of a convolutional algorithm [7]. The output of a given code depends on the current k input and the past input bits. The decoding of such codes is carried out using the widely

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used Viterbi algorithm, which is based on the same principle. Due to the popularity of computational codes, there have been many different approaches to extend and modify this basic coding scheme. Convolutional codes are different from block codes in that they have memory and outputs that are dependent on the previous input blocks.

A convolutional encoder is a type of algorithm used in FEC to improve the performance of the Bit Error Ratio. It can be described as a finite state machine. The use of block codes and convolutional codes helps minimize the random errors and prevent burst errors. Concatenated codes are typically composed of the outer and inner codes. In a communication system, low-latency is very important when it comes to implementing a speech transmission procedure using a combination of block and convolutional codes [8].

2.LITERATURE REVIEW

Daniël van der Veer et al proposed the receiver architecture of the BLR standard that improves the frame detection by 4.4dB.It also provides better resilience to frequency offset. The BLR standard adds two rates 1/2 and two rate 1/2 repetition codes to the GMSK modulation. The two accelerators that are used for these codes are the repetition and the Viterbi decoder. The BER performance of the Digi-Key's Viterbi decoder was compared with that of the matched and low-pass filters [9]. The former has a better bit error rate for values below 16 dB, while the latter performs better for higher values. The repetition decoder was also implemented as an accelerator, similar to the design of the Digi-Key's Viterbi decoder. Although the BER performance of the repetition decoder was worse than expected, it can be used as an MLSE equalizer. The advantage of using the matched filter is that it can reduce the ISI introduced by the filter [9]. The BER performance when using the matched filter and the equalizer is better than when using the same combination of filters. However, when using the matched filter and the combined output of the two components, the BER is only slightly better than when using the same combination of filters. The combination of the low-pass and matched filters is worse than the one used for the computation of the convolutional decoding. Based on the measurements of the various receiver configurations, it was concluded that the receiver with the matched and equalizer BER curve is better than the theoretical DQPSK curve.

Vincent Tran et al ,the performance of the OFDM system in multipath fading channel and AWGN channel after implementing LDPC for channel coding is compared with that of the other two systems [10]. Simulation results show that the system's BER performance significantly improved after implementing the LDPC technique. In addition, the system's performance also increased in comparison with that of the other two systems.

Deepan Govindaraj et al an algorithm that generates a trellis that describes the ISI using a faster communication system than Nyquist is used [11]. It considers the constellation, the pulse shape, and the time compression factor, and outputs a matrix

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containing necessary states and branch metrics. This ensures that the channel output is coded according to the necessary state.

Samridhiet al, Through the AWGN channel, a simulation of the performance of various turbo and convolution codes was conducted. Various data rates, block lengths, modulations, and iterations were used for analysis. There are multiple annotations [12]. For instance, in the case of soft and hard decision Viterbi decoding, the former performs better with increasing data rate. It has been observed that with increasing number of iterations, turbo codes outperform their counterparts in terms of performance. However, there are also multiple constraints that prevent them from performing well. For instance, in the case of QSSK, it is preferred over higher schemes due to the noise constraints. Due to the complexity of the codes, it is often difficult for the decoder to distinguish between noise and symbols. With increasing number of iterations and block length escalation, the performance of turbo codes can be improved. Some of the factors that can improve the performance of turbo codes are early termination and block length escalation.

V. S. Sreekanth et al , a process known as puncturing is used to create a m/n rate code from a basic low-rate encoder. It involves deleting some bits from the output. A puncturing matrix is used to remove the bits from the output. The reduction of complexity in the coding process has led to the emergence of a new class of iterated short code called turbo codes, which closely follows the theoretical limits of Shannon's theorem [13]. Compared to the long-convolutional codes, the new generation of turbo codes has less complexity and performance. A concatenation with an outer algebra called Reed-Solomon is used to address the issue of error floors in the designs of turbo codes. The performance of the decoder is improved by implementing a modified version of the Viterbi algorithm.

Nejwa El Maammar et al , presents a detailed analysis of the traceback memory management in the various types of VUTERBI decoders. It shows the performance comparison between the different hard decisions that are used in the algorithm when they are associated with memory or not. This is very useful for channel choosing. Based on the simulations, we can determine the minimum value of the traceback depth, which is comparable to the performance of infinite survivor path memory [14]. It has been observed that performing traceback depth of around 15 provides the obtained BER with a close relative to the BER with finite survivor memory.

Buthanani Dlodlo et al, The TC-DSTBC system uses a transmission matrix to transmit data. It is similar to a conventional DSTBC. The goal of the TC-DSTBC system is to use a dynamic mapping procedure instead of a fixed one to map additional data onto the expanded space-time Block Code (STBC). This method ensures that the system can maintain its bandwidth efficiency while reducing the error performance of the traditional DSTBC [15]. To achieve this, a new mapping technique known as square M-ary Quadrature Amplitude Modulation (M-QAM) is proposed. The performance of the TC-DSTBC system against the conventional DSTBC has been compared. It shows that it

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can achieve a bandwidth efficiency of around 12.5 to 8.3% for 16-QAM and 64-QAM.According to the Monte Carlo simulation results, the four receive antennas of the TC-DSTBC system maintain their BER performance at high signal-to-noise ratios. This is because they are distributed under the same distribution and independent set of receive antennas. In addition, the system can also benefit from the advantages of the square M-QAM modulation order and the additive white noise conditions.

Semardeep Dhaliwal et al , the effects of constraint length and code rate on the performance of a convolutional encoder have been studied. In the simulation results, it has been shown that the BER of a coded system using the Viterbi decoding algorithm is better than that of an uncoded system. It also shows that the increase in the code rate leads to a decrease in the BER of the system [16]. The performance of a coded system using the constraint length 8 algorithm is significantly better than that of an uncoded system.

Adel Issa Faraj Ben Issa et al , when light fading effects appear in an image, the resulting negative and positive format is transmitted through AWGN channel. Due to the noise generated by the channel, the system stops receiving the transmitted signal. However, the experimental results show that higher energy can eliminate the noise impact and increase bit noise [17]. A coding modulation technique is used to create a unique code for each input word. It can be implemented by implementing a signal bits stream algorithm that is designed to enhance the performance of the signal. If a portion of the signal gets affected by the noise, the other information of the signal will remain safe.

3. OVERVIEW OF COMMUNICATION SYSTEM

Transmitter, channel and receiver these are the main aspects of any communication systems. Following figure 1. represents a general block diagram of a communication system.

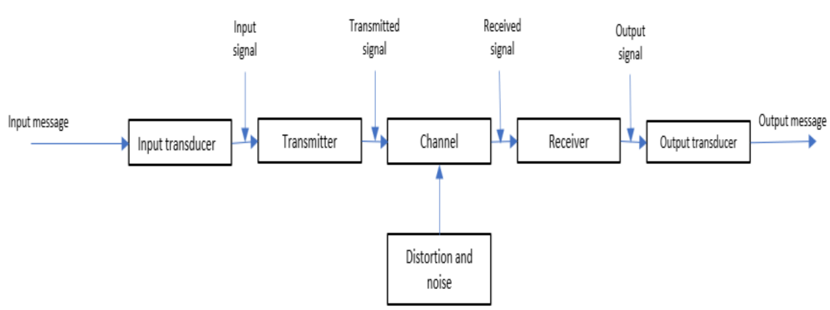

Figure 1. General block diagram of a communication system [18]

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Information or a message is generated by a source for example the message can be in the form of voice, music, text, e-mail etc. If data is in non-electric form, then it must be converted into an electric form by an input transducer which is generally called as baseband signal. This baseband signal is converted into a suitable form for transmission using transmitter. Channel or medium is another important aspect of communication system through which the information is passed. In wired medium i.e., metallic wired connection between transmitter and receiver, the information is passed through the medium in the form of electric voltages or currents. In another wired medium such as optical fiber cable, data is passed through the medium in the form of lights. And in wireless communication, the data is transmitted in the form of electromagnetic energy. External disturbance gets added during the transmission of signal through the channel. The receiver reprocesses the signal received by the channel by reversing the signal modifications done at the transmitter side and removing the external noise added by channel. Output transducer is used to convert the electric form of data into its original non-electric form.

4.TYPICAL DIGITAL COMMUNICATION

If a continuous or analog signal is provided as an input to the process of modulation then it is called as Analog Communication whereas if a digital signal is provided as an input to modulation is called a Digital Communication. Figure 1.2 represents a typical digital communication block diagram.

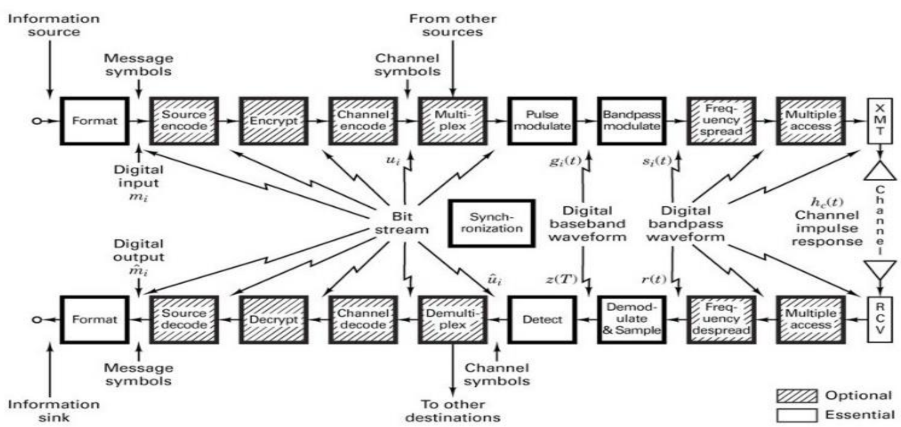

Figure 2: Typical block diagram of a digital communication system [19]

Upper row of operations is a transmitter and lower row of operations denotes a receiver. Various optional operations like source coding-decoding, encryption-decryption, channel encoding-decoding, frequency spreading-dispreading are performed.

a) Transmitter

As far as this research is concerned, there are two major operations in transmitter. One in source encoding and other one is channel encoding. Source encoder is used to remove the redundancy in the information if any or it is used to compress the data

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whereas the channel encoder is used to detect and correct the data. Formatting block converts the data into suitable form for further process. One can transmit number of messages simultaneously with limited resources using local and global technique, multiplexing and multiple accesses respectively. Then the data is transmitted using wired or wireless medium.

b) Receiver

Reverse operations are performed in the receiver to extract the original information. Here again, the channel decoder and source decoder are important for finding out and correcting the errors and adding redundant data in the information respectively.

5.OBJECTIVE

In the research, a digital communication system is analyzed with and without using channel coding technique. Responses of the system for both scenarios are observed and compared.

1. Global Objective:

1. To investigate the performance of 16-Phase Shift Keying (PSK) modulation technique in AWGN environment with and without channel codding.

2. Local Objectives

1. To build a communication system transmitter with and without channel coding technique
2. To modulate the signal using 16- Phase Shift Keying (16-PSK) modulation scheme.
3. To design a receiver circuitry to extract original information in both cases (with and without channel coding).
4. To calculate Bit Error Rate (BER) for different values of Signal to Noise Ratio (SNRs).

6.BACKGROUND

In this part, types of source and channel coding along with different modulation and demodulation schemes are discussed which are required for successful and efficient transmission of information.

a) Source Coding

SNR and bandwidth are two important factors that determine performance of communication system. Digital systems adopt aggressive measures to lower the source data rate as compared to analog systems [19]. Many messages consist of unwanted data which is commonly called as redundancy. If these redundancies are transmitted, then it would be wastage of resources like bandwidth etc. So removal of such unnecessary information is required in many cases at initial stages. This is done by source coding technique. Source coding reduces the redundancy based on the

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predictability of a message source. Thus this block is the first block after proper formatting of data.

b) Types of Source Coding

Objective of source coding is to use codes which are as short as possible for the representation of data. The more predictable message contains more unwanted information, thus a shorter code is required and less predictable information require larger codes. There are several types of source coding listed below.

c) Huffman coding

This is a probabilistic approach of source coding. Based on the probability of symbols, a code-word is assigned to it. Higher the probability, shorter the length of code-words and lower the probability, larger the length of code-word.

d) Shannon Fanon coding

The approach of this type of coding is same as that of Huffman coding technique. This one is also a probability based source coding technique. Only the method to obtain the code-word is different than the Huffman coding.

e) Lampel Ziv (LZ) and Lampel Ziv Welch (LZW)

These types are dictionary based approaches where code-words are formed with the help of the pattern in which the symbols are appeared.

f) Channel Coding

To combat errors that arise from noise and interferences, some extra, unwanted data needs to be added systematically in the transmitter such as the receivers can rely on the unnecessary data i.e. redundancy to correct errors caused by channel distortion and noise. This is done by channel coding or error correction coding. Error correction coding intentionally inserts redundancies in the compressed data such that if error occurs upon detection, then redundancy can help to correct the erroneous data. Each error correction coding technique involves the use of channel encoder in the transmitter and a decoding algorithm in receiver.

7. TYPES OF CHANNEL CODING

Different types of error control codes are listed below.

1. Linear Block Codes
2. Cyclic Codes
3. Convolutional Codes

Detailed description of convolutional code is provided here**:**

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a) Convolutional Codes

In block code, block of ‘k’ bit length is converted into a block of ‘n’ bit length code-word (where n > k) by adding n-k parity bits. And ‘n’ bit length code-word is generated by an encoder in any particular time unit which depends on ‘k’ bit length block of message generated within that time span. But in convolutional coding, ‘n’ bit length code-word is generated by an encoder in a particular time unit, depends not only on the ‘k’ bit length block of message generated at the same time unit but also on the data generated at previous time instant. This code can correct random errors, burst errors or both the errors.

b) Convolutional encoder

An encoder with constraint length ‘N’ consist of ‘N’ shift registers along with some ‘q’ number of modulo-2 adders. Following figure shows a convolutional encoder with N = 3 and q = 2. Data is applied as an input to one of the shift register and coded output is obtained at the commutator output.

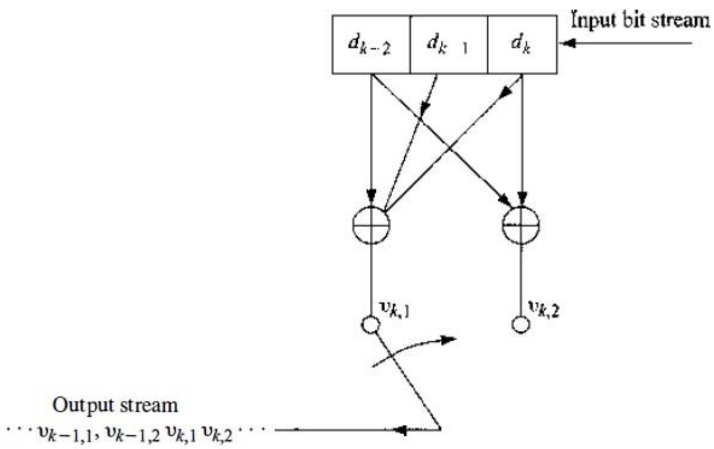

Figure 3: Convolutional Encoder using Registers [18]

State is the contents present in the remaining N-1 shift registers. For different instances and inputs, different states can be observed. The state transition can be visualized using code tree, trellis diagram. The trellis diagram starts from all 0s in the shift registers and makes transitions corresponding to each input data bit. The encoder of a binary convolutional code with rate 1/n, measured in bits per symbol, may be viewed as finite state machine that consist ‘N’ shift registers with ‘q’ number of modulo-2 adders. L-bit message sequence produces a coded output of length q(L+N) bits. The code rate is given by,

$$r = \frac{L}{q(L+N)}\ bits/symbol$$

Typically, L >> N, thus code rate is generally

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$$r \approx \frac{1}{q}\ bits/symbol$$

A constraint length is the number of shifts over which a single input bit can influence the encoder output. In above encoder, memory of encoder is equal to N input bits and K = N + 1 shift are required for a message bit to enter the shift register and finally come out.

c) Modulation Techniques

There are various methods available for the transmission of digital data over a band pass channel. Different digital modulation schemes are listed below.

a) Amplitude Shift Keying (ASK)

b) M-ary Phase Shift Keying (M-PSK)

c) M-ary Quadrature Amplitude Modulation (M-QAM)

In this research, we are going to use 16-PSK modulation and demodulation technique so description about M-PSK is specified below.

d) M-ary Phase Shift Keying

In this type of modulation, phase of carrier changes according to digital modulating signal. 'M' denotes the number of constellation points or number of symbols. In M-ary PSK, the phase of carrier can consider one of the 'M' possible values, namely,

$$\theta_i = \frac{2(i-1)\pi}{M}\ \ where\ i = 1, 2, \ldots\ldots, M$$

During each signaling interval of duration T, one of the M possible signals [18],

$$s_i(t) = \sqrt{\frac{2E}{T}}\cos\left(2\pi f_c t + \frac{2\pi}{M}(i-1)\right)\ \ where\ i = 1, 2, \ldots, M$$

Is sent where 'E' is the energy per symbol. According to Gram-Schmidt process, this signal can be expressed in terms of two basic functions. Thus the signal constellation of M-ary PSK is two dimensional. M signals are equally spaced and arranged in a circular fashion with radius √E as shown in figure 2.2 (for M=8).

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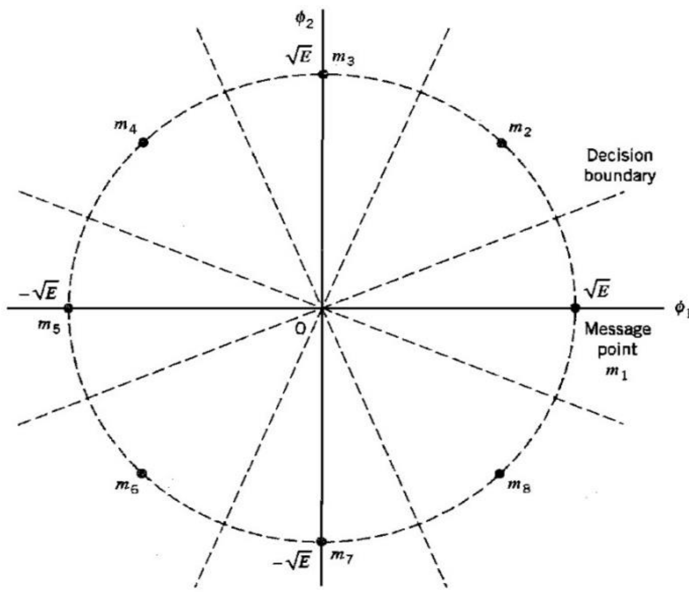

Figure 4: Signal space diagram for 8-PSK modulation [18]

Euclidean distance of two points in M-ary PSK is given as

$d_{Eu}=2\sqrt{E}\,\sin(\pi/M)$

And average probability of symbol error is expressed in terms of error function as

$P_s \cong \mathrm{erfc}(\sqrt{E/N_0}\,\sin(\pi/M))$

Bandwidth Efficiency of M-ary Phase Shift Keying

The power spectra of M-ary PSK for three different values of M is shown below.

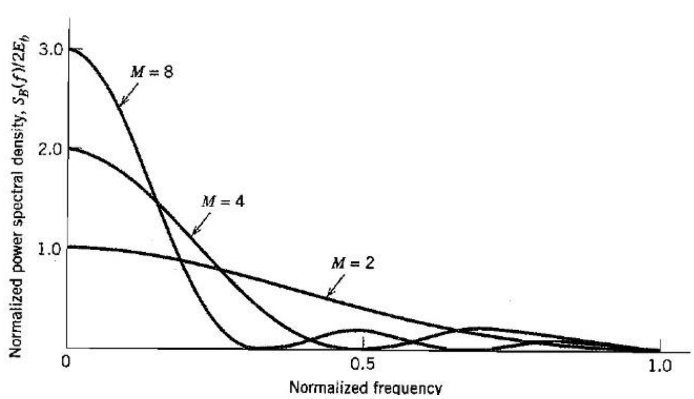

Figure 5: Power Spectra of M-ary PSK modulation [18]

If 'T' is the symbol duration, R_b is the bit rate, the channel bandwidth B required for efficient transmission is expressed as [19]

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$$B = \frac{2R_b}{\log_2 M}$$

E. Based on this, bandwidth efficiency is given by

$$\rho = \frac{R_b}{B} = \frac{\log_2 M}{2}$$

e) Decoding at Receiver

Source decoding and channel decoding techniques are designed at receiver side which are used to separate out message form transmitted signal. One of the important channel decoding method related to Convolutional code is described below.

- **Maximum Likelihood Decoding (Viterbi Algorithm):**

It is well known hard decoding technique which is more flexible. The decoder structure is relatively simple for short constraint length, making decoding feasible at relatively high rate at of up to 10Gbit/s [2]. In AWGN channel, the maximum likelihood receiver requires selecting a code-word closest to the received code-word. Each path in a trellis diagram represents a valid code-word. The objective of this decoding technique is to find the best path through trellis diagram which is closest to the received data sequence. A minimum hamming distance path represents the most likely sequence. As far as computational complexity and storage is concerned, Viterbi algorithm is the best approach. To achieve low error probability, longer constraint length is required so Viterbi algorithm is not useful.

f) Demodulation Process

In M-ary PSK demodulation technique, original data is separated out from disturbed data with the help of threshold in constellation diagram. Output of demodulation is the channel encoded data and it can be erroneous.

8.METHODOLOGY

Digital communication system is analyzed using MATLAB Simulink. This gives us a brief description about how the system is modelled in MATLAB environment and what parameters are chosen to observe error performance of system.

a) Data source

A digital data is generated by “Bernoulli Binary Generator” block in MATLAB Simulink. The block along with the properties of the block is shown in figure 3.

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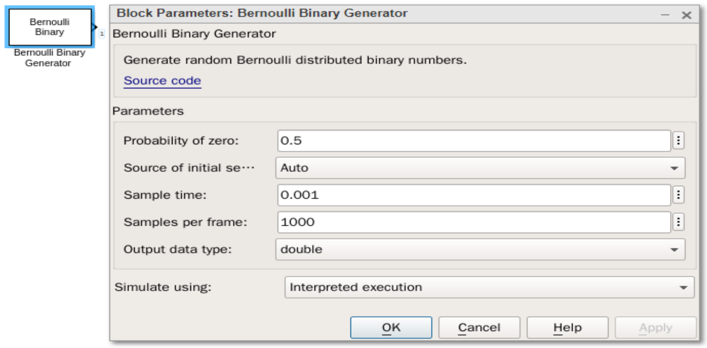

Figure 6: Bernoulli Binary Generator

Sample time in block parameters is simply a bit duration. Here it is set to 0.001 simulation time. Total number of samples per frame are considered as 1000. Rest of the parameters are set to default values. The output of this block is a random binary data and length of the data depends upon the simulation time. This data is transmitted and referred at receiver side to find out bit error rate (BER).

b) Chanel Encoding

For coded transmitted data, input data is provided to the channel encoder first and then needs to be modulated. A convolutional encoder is used as a channel encoder. The block along with the parameters is shown in figure 3.

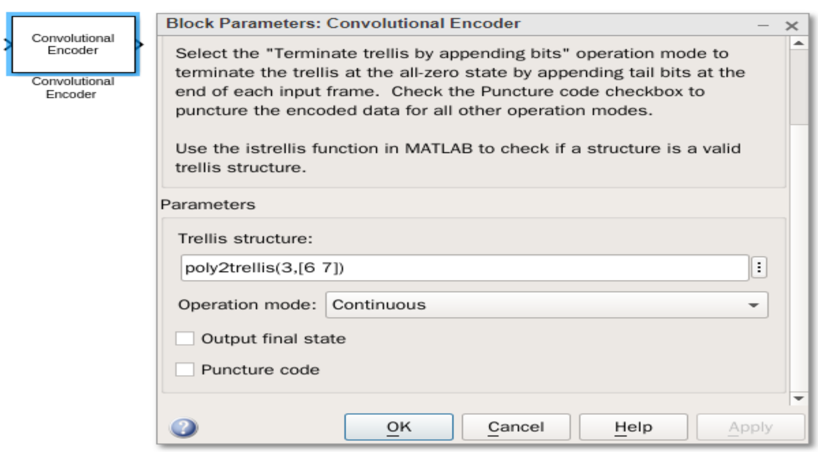

Figure 7: Convolutional Encoder

Number of symbols in M-ary PSK modulator are 16. Thus, the value of M-ary number is chosen as 16. Input type is selected as bit because digital data which is simply a series of 0s and 1s. Constellation ordering is set to Gray i.e. default scheme.

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c) Transmission Medium

The modulated data is transmitted through AWGN channel. Thus the medium is wired. Signal to Noise Ratio (SNR) is selected here. For different values of SNRs BER is observed. The block with parameters is shown in figure 3.

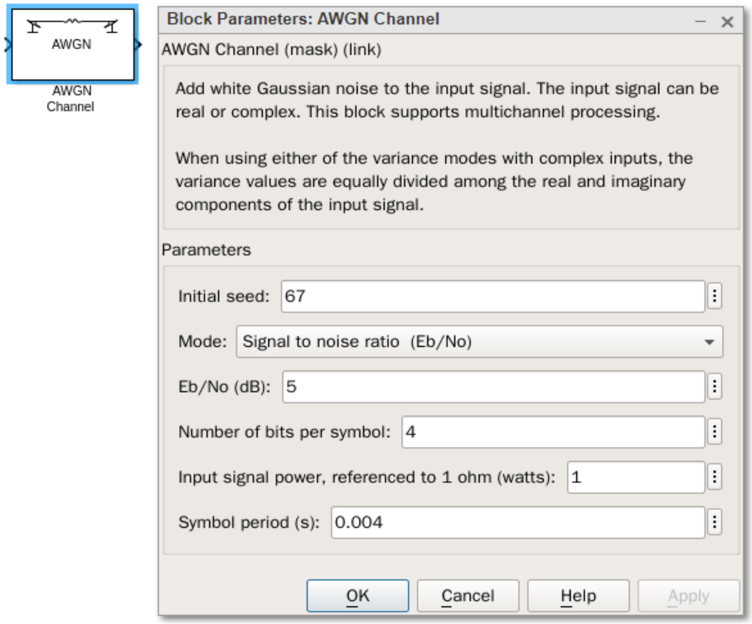

Figure 8: AWGN Channel

SNR is set to different values starting from 0 dB to 15 dB. For 16-PSK modulation schemes, 4 number of bits are used to represent a symbol and accordingly a symbol period will be 0.004 seconds. Just before the demodulation process, the signal is observed using constellation diagram.

d) Demodulation Technique

M-PSK Baseband Demodulator is used to demodulate the coded and uncoded data. Properties of demodulator block must be same as the modulator block.

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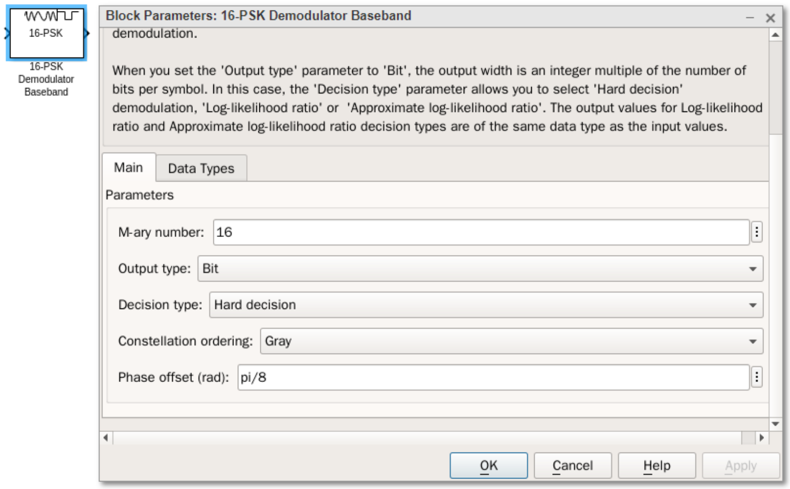

Figure 9: M-PSK Baseband Demodulator

Hard decision method is used in demodulation process as the data is binary. Rest of the parameters are same as that of modulator.

e) Channel Decoding Technique

A Viterbi decoder is used for decoding of convolutional codes. Figure 3.6 shows the block along with the parameters.

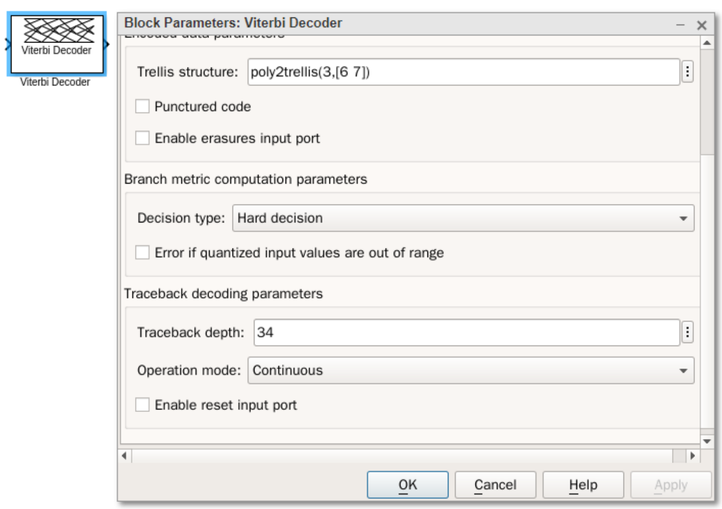

Figure 10: Viterbi Decoder

Same 4-state trellis structure is used here and “Hard Decision” type is selected as specified in problem statement.

f) Error Analysis

The transmitted and extracted data is compared and then number of erroneous bits are calculated. Error Rate Calculation block is used to find out errors. To check errors in

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coded data, we need to insert a delay equal to the trace-back depth in Viterbi decoder block. For uncoded data, receiver delay must be 0. The block along with the parameters is shown in figure 3.7.

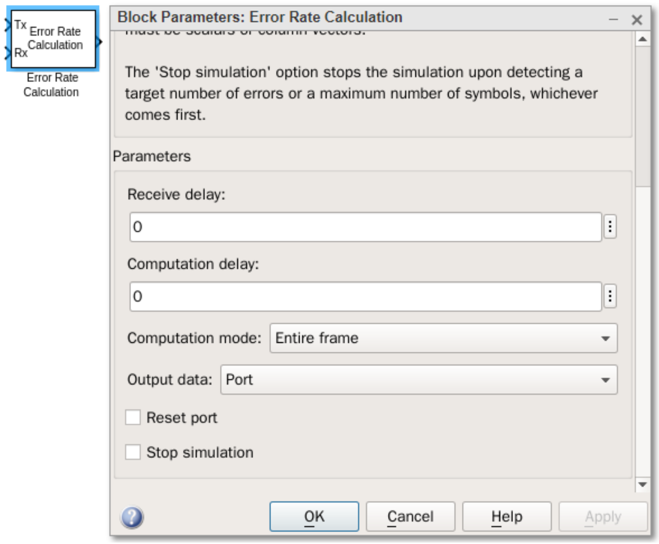

Figure 11: Error Rate Calculation

g) Implementation

The communication system is built in Simulink environment using above blocks. Two approaches viz. performance of coded and uncoded are designed. The entire model is shown in figure 3.8.

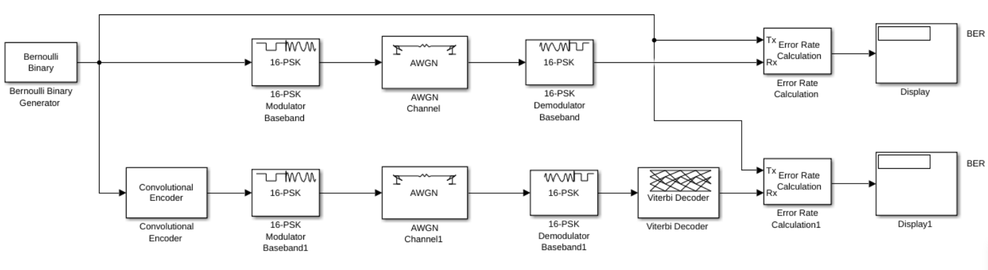

Figure 12: Simulink Model of Communication System

9. RESULTS AND DISCUSSION

For different values of SNRs starting from 0 dB to 15 dB with the increment of 1 dB, BER is observed using “Display” block in MATLAB. The simulation time is kept at 20. Random binary bits are considered as the original information which needs to be transmitted. A bit duration is kept as 0.001. This message acts as an input to the two different approaches; one for with channel coding and other for without channel coding. In later approach, the input data is directly provided to 16-PSK modulator. In this

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modulator, number of constellation points or symbols are 16. At the output of this modulator, if we observe a constellation diagram, 16 points will be arranged in a circular fashion as shown in figure 4.1.

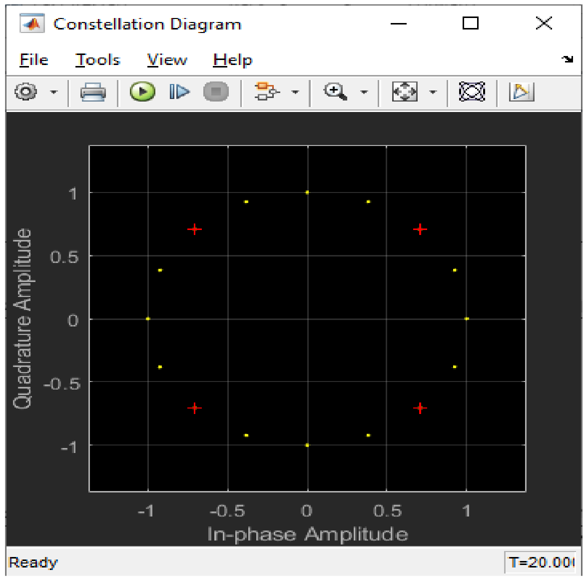

Figure 13: Constellation diagram of modulated signal

On the other hand, the same input data is applied to channel coding technique, “convolution encoder”. A ½-rate convolutional code with at least 4-state trellis is used in this encoder. This encoder adds parity bits. Number of input bits is 1 and the number of output bits is 2. Hence the encoder doubles the length of the input sequence. And the output of channel i.e., convolution encoder is provided to the 16-PSK modulator same as above. Constellation ordering is selected as “Gray”. It denotes one bit change in subsequent symbol.

The modulated signal in both the cases is then transmitted through AWGN channel. Again, constellation diagram is observed at both AWGN channels. The points will be dispersed based on SNR value. More the SNR value, more points are concentrated on reference constellation points.

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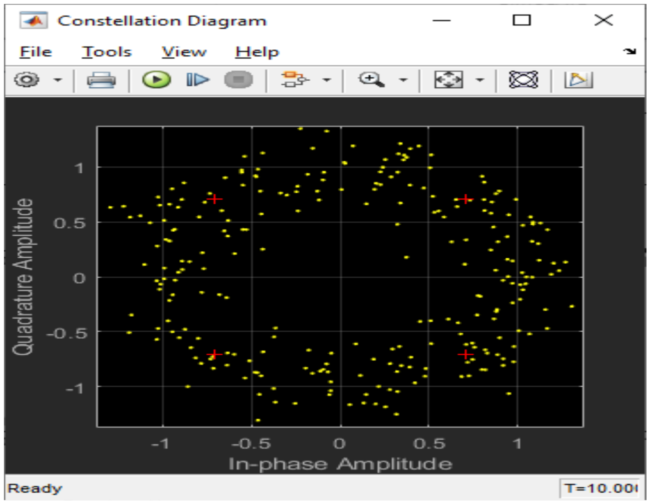

Figure 14: Constellation diagram at 5 dB without coding

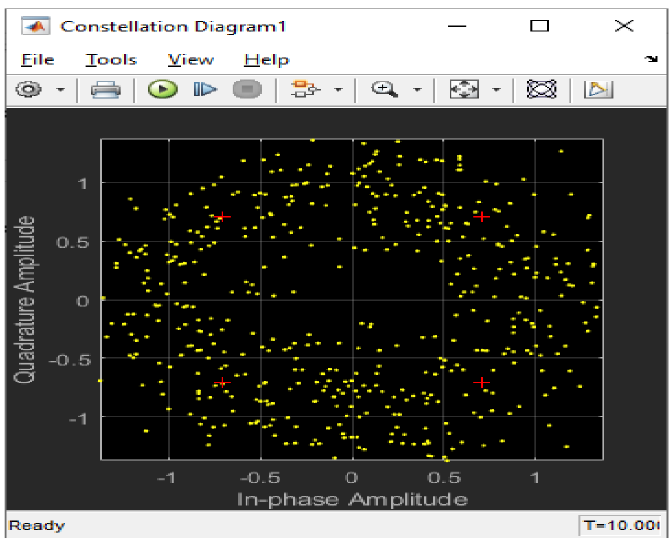

Figure 15: Constellation diagram at 5 dB with coding

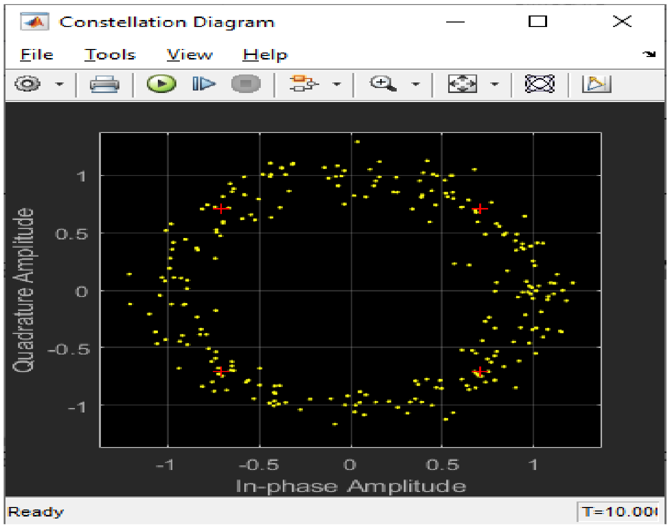

Figure 16: Constellation diagram at 10 dB without coding

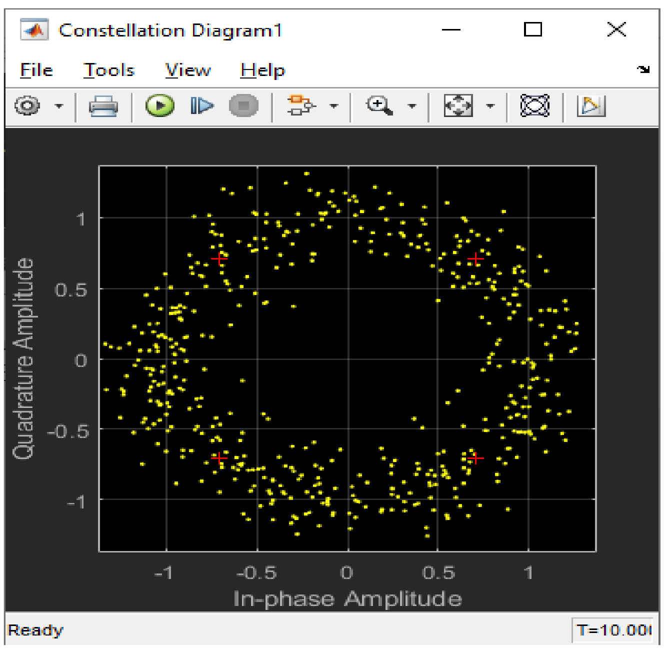

Figure 17: Constellation diagram at 10 dB with coding

10. ERRORS WITHOUT CHANNEL CODING

The value of bit error rate is measured without coding approach for every SNR starting from 0 dB to 15 dB (with increment of 1 dB). The values are summarized in following table 4.1

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Table 1: Ber without Convolutional Coding

Sr. No.	SNR in dB	BER without channel coding
1	0	0.1742
2	1	0.1498
3	2	0.1345
4	3	0.1106
5	4	0.09555
6	5	0.0815
7	6	0.066
8	7	0.052
9	8	0.04164
10	9	0.02927
11	10	0.01964
12	11	0.01318
13	12	0.006545
14	13	0.00309
15	14	0.001909
16	15	0.0006364

Theoretical value of bit error rate of M-PSK modulation is same as the value of BER rate without channel coding.

11. ERRORS WITH CHANNEL CODING

Table 2: BER with convolutional coding

Sr. No.	SNR in dB	BER with channel coding
1	0	0.339
2	1	0.2933
3	2	0.264
4	3	0.2276
5	4	0.1769
6	5	0.1391
7	6	0.094
8	7	0.069
9	8	0.04851
10	9	0.02544
11	10	0.01714
12	11	0.008937
13	12	0.004833
14	13	0.002097
15	14	0.0005471
16	15	0.0001824

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The value of bit error rate is measured with convolutional coding approach for every SNR starting from 0 dB to 15 dB (with increment of 1 dB). The values are summarized in table 4.2. The value of BER drops suddenly in case of channel coding for high signal to noise ratio. For high SNR, channel-coding technique is effective as the extra bits are used to detect and correct the errors if any. Unis MATLAB code SNR v/s BER plot is obtained which is shown in figure 4.6.

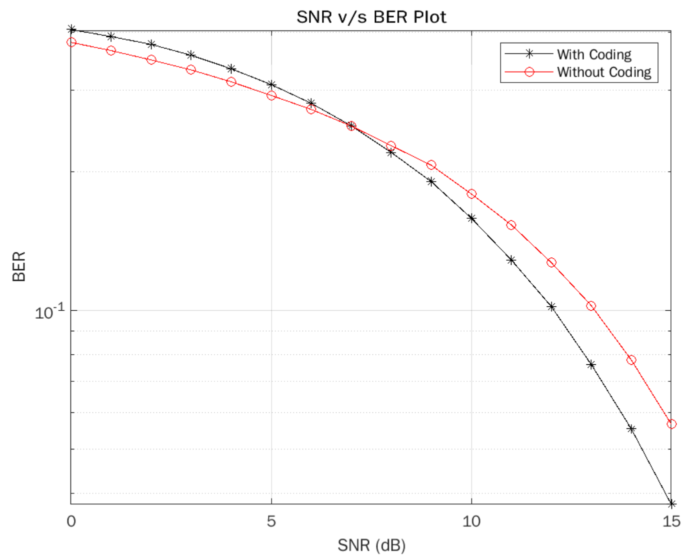

Figure 18: SNR v/s BER Plot

It is clearly observed form this figure 4.6, Bit Error Rate is significantly low in case of channel coding at high SNR.

12. CONCLUSION

Error performance of a communication system with and without channel coding technique specifically convolutional coding technique is analyzed in MATLAB Simulink environment. For lower value of signal to noise ratio, the BER is lesser in case of without convolutional coding approach but it is lower at high value of SNR in case of with coding approach. And this is acceptable as the purpose of parity bits is to detect and correct the errors. Meanwhile the Viterbi decoding technique is effective as far as computational complexity and digital storage is concerned. Unlike block codes, the sender does not send the message bits; it sends only parity bits. So this makes convolutional coded modulation more efficient than uncoded modulation. Increase in the code rate of the encoder, increases the BER.

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