Efficient approach to design a wireless receiver for successfully decoding of transmitter data under AWGN environment

By

ADEL ISSA FARAJ BEN ISSA

Electrical & Computer Engineering Department

Submitted to the Graduate School of Science and Engineering

in partial fulfillment of the requirements for the degree of Master

ALTINBAŞ UNIVERSITY

[2018]

This is to certify that we have read this thesis and that in our opinion it is fully adequate, in scope and quality, as a thesis for the degree of ....

Academic Title Name SURNAME	Academic Title Name SURNAME
Co-Supervisor	Supervisor

Examining Committee Members (first name belongs to the chairperson of the jury and the second name belongs to supervisor)

Academic Title Name SURNAME	Faculty, University	__________________
Academic Title Name SURNAME	Faculty, University	__________________
Academic Title Name SURNAME	Faculty, University	__________________
Academic Title Name SURNAME	Faculty, University	__________________
Academic Title Name SURNAME	Faculty, University	__________________

I certify that this thesis satisfies all the requirements as a thesis for the degree of ....

Academic Title Name SURNAME

Head of Department

Approval Date of Graduate School of
Science and Engineering: ____/____/____

I hereby declare that all information in this document has been obtained and presented in accordance with academic rules and ethical conduct. I also declare that, as required by these rules and conduct, I have fully cited and referenced all material and results that are not original to this work.

Adel Issa Ben Issa

[Signature]

DEDICATION

Dedication for to my mother , wife and my children Nour,Hala,Jana,and Bana who has always been with me through the hard times ,without whom this thesis would have been completed

ACKNOWLEDGEMENTS

I would first like to thank my advisor asst.Prof Dogu Cagdas The door to his office was always open whenever I ran into a trouble spot or had a question about my research or writing. He consistently allowed this thesis to be my own work, but steered me in the right the direction whenever he thought I needed it.

Finally, I must express my very profound gratitude to my mother and to my wife for providing me with unfailing support and continuous encouragement throughout my years of study and through the process of researching and writing this thesis. This accomplishment would not have been possible without them. Thank you.

Author

[ADEL BEN ISSA]

ABSTRACT

Wireless communications imply free-space signal transmissions and receiving that signal by a remote user or through a station. In optimum transmission, which does not involve noise, end-user/s can receive signals with the same power without information losses; however, in actual scenario, noise is inevitable in wireless communications. Digital modulation is performed to formulate an electrical version that suits free space transmission. The digital data stream can be efficiently transmitted from one station to another by breaking it into smaller blocks called as packets. The packet-switching circuits result in better noise immunity. Furthermore, variations in signal power and noise power provide better results on the receiver's end.

Trellis coding modulation is applied for effective signal transmission. In this project, we have proposed a complete design of encoder and decoder methods to transmit data at Trellis code rate 3/2. A trellis encoder performs like a Finite State Machine (FSM) using convolution coding and later, the decoder works for Maximum Likelihood (ML) detection with the help of a state table and trellis diagram while both the algorithms are simulated using MATLAB.

Efficient approach to design a wireless receiver for successfully decoding of transmitter data under AWGN environment

Author: ADEL BEN ISSA

M.S. Electrical & Computer Engineering Department, Istanbul Altınbaş University,

Supervisor: asst.prof: Dogu Cagdas

Date: [2018]

Table of Contents

LIST OF TABLES

Pages

LIST OF FIGURES

Pages

Pages

LIST OF ABBREVIATIONS

AWGN	Additive White Gaussian Noise
ISI	Inter-Symbol Interference
BER	Bit Error Rates
SNR	Signal-to-Noise Ratio
BPSK	Binary Phase Shift Keying
MIMO	Multiple Input Multiple Output
SISO	Single Input Single Output
QPSK	Quadrature Phase Shift Keying
MC-CDMA	Multi-Carrier Code Division Multiple Access
DS-CDMA	Direct Sequence Code Division Multiple Access
OFDM	Orthogonal Frequency Division Modulation
TWDP	Two Wave Diffused Power
DPSK	Differential Phase Shift Keying
QAM	Quadrature Amplitude Modulation
BCH	Bose-Chaudhuri-Hochquenghem
BICM	Bit-interleaved Coded Modulation
CSI	Channel State Information
ABICM	Adaptive Bit inter-leaved coded modulation
ATCM	Adaptive Trellis Coded Modulation
QoS	High Quality of Service
ZPS	Zero-Padding Schemes

SSD	Signal Space Diversity
DAC	Digital to Analog Conversion
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Accesses
FDM	Frequency Division Multiplexing
FDMA	Frequency Division Multiple Access
CDMA	Code Division Multiple Access
AMPS	Advanced Mobile Phone System
CDM	Code Division Multiplexing
GSM	Global System for Mobile
SDM	Space Division Multiple
SDMA	Space Division Multiple Access
TCM	Trellis Coded Modulation
FEC	Forward Error Correction
FSM	Finite State Machine

1 Introduction

Outline

A deep fade is taking place in the context of wireless communications because in this case, power of noise is more as compared to power of transmission. A diversity approach has been proposed to overcome the said fading effects, and the diversity approach is carried out by simply letting receiver detect multiple versions of the same transmitted information by sending it from a different transmitter antenna. Noise power or data rates adapt to achieve the Shannon capacity; so, modulation methods as well as data rates will be adaptable as per fading/noise levels.

The technique of channel states reveals the states of channel to the transmitter so that power per bit will be relative in terms of level of noise. The channel input constrains are the essential considerations for coding method that is used to code transmission of data in that particular channel having AWGN noise.

The approaches, which were used earlier for communication system designs help detecting a signal through receivers, and for applying strategies to correct error at demodulator for obtaining transmitted data through a noisy received signal. A few years ago, coding schemes were introduced with modulation, and the combination is well known as coding modulation while the same involves in signal generation and digitally modulating the same through a digital/binary stream. Then, the channel coding is applied on that stream on the receiver. This decoding is followed by demodulation, which is used for extracting data.

Problem statement

Since transmission of data over a short distance is under consideration for this study, the phenomena including multipath delay, phase or amplitude fluctuations are termed as fading. The fading impact is either light, which is termed as "small-scale" fading and it might severely affect the data, and this is called as "large-scale" fading. It is possible to neglect small-scale fading while large-scale fading exists in the form of a large variation in a multi-path over the distance of transmission and random modulation of frequency that takes place because of doppler effects within the multipath rays.

Fading channel may have echoes or time desperation, which results from the multi-path propagation delay. Fading is limited with some factors such as the situation of surrounding objects in communication pairs (transmitter and receiver), the receiver speed and the signal transmission bandwidth. Inter-symbol interference (ISI) can distort signal because many symbols interfere a single symbol.

This phenomenon is unwanted because previous symbols create the same impact like noise, so, it reduces the reliability of communication. ISI has impact on the pulse spreading beyond the allotted interval of that pulse, and eventually, the starched pulse will be dominating the neighbouring pulses over the given time interval; therefore, it develops inter-sample interface.

The inherent response of non-linear frequency or the multipath propagation leads towards inter-sample interface in many cases and hence, the receiver will be tuff when the signal generates large error.

Objectives

Increasing spectral efficiency of optical coherent channel in communication systems using coded modulation is a main objective.

The coded modulation performance will be presented with different error rate through Rayleigh and AWGN channels. In this project, use of Trellis coding has been proposed because the data bits exist in many parallel streams each representing a code word, which is then coded further by using trellis coding algorithm.

System performance may be examined by permitting end-to-end transmission of data through additional Gaussian fading and noise channel. The signal-to-noise ratio has an effect on transferred data that can be analyzed to assure noise immunity of the system. A study of analytical approach related to orthogonolity (serial to parallel) of data transmission may also start when simulation provides reliable transmission.

Use of code words helps in the code modulator's performance evaluation with different N (code length). It is possible through assuring that chosen values of Ns has insignificant effect on the obtained results. The 10e5 bits of input will be examined for optimum performance.

Thesis organization

This thesis comprises of seven chapters. The project outline and description is given in the first chapter, research problem and objectives are also stated in the same chapter. Chapter two demonstrates previous studies, which were conducted for evaluating the communication systems' performances with respect to fading and noisy channel. Chapter two consists of literature review.

The communication systems' terminologies and the methods pertaining to the communication hierarchy will be disclosed in chapter three. Chapter four “Methodology” is about coding modulation techniques as well as decoding methods, their performances and uses. Chapter five “Simulation and results” contains the entire system simulation and the monitoring tools to measure the performance within the designed system and criteria.

Chapter six “Conclusion” contains the conclusion of this work regarding how the results are obtained and includes system impacts as well as recommendations. Finally, the project contributions and publications with the used references will be listed in chapter seven under “References and Publications”. The literature survey on telecommunication systems in Chapter two has multiple factors, in which, degrading the performance, noise such AWGN, fading and shadowing are significant topics because they may threaten any communication system.

2 Literature Survey

Telecommunication system is subject to multiple factors, in which, degrading the performance, noise such AWGN, fading and shadowing are the major issues, which may threaten any communication system. Optimal diversity is proposed to implement a channel that resists fading. A receiver may detect super imposition of the signals coming from diverse antennas. This chapter illustrates the fading and noise issues in any communication system and different approaches to combat with them. Furthermore, the coded modulation is proposed because it enhances bit error rates (BER) so that the communication system perform better as a whole, which is proven during the recent researches in this domain.

In [1], author has tested the performance of different communication systems under fading channel effects. He added a different modulation technique for fading channels and hence, he analyzed their performance such as BER per iteration. After measuring the BER for the particular communication system, we found that BER (bit error rate) increases with high SNR, whereas the resulting BER from lower signal-to-noise ratio (SNR) remains small. This result looks logical since small SNR made the AWGN dominate the BER.

In other word, BER is possible to enhance through improving the SNR. Afterwards, the performance resulted by the whole channels are compared for evaluating a more accurate model, which is more reliable under noise. A binary phase shift keying (BPSK) modulation helps analyzing the performance by establishing the plot of SNR versus BER under various channel conditions.

In [2], a wireless system's multi-path fading channel has been proposed and examined with different inputs. A complex and real data was provided as an input System Matlab-simulated and analyzed for fading effects and noise effects. The simulation has yielded improved performance with greater SNR at the input, similarly an SNR-BER graph has been made, which clearly shows BER improvement with higher SNR.

In [3], the author mentioned that multiple input-output transmissions have a noticeable impact on wireless transmission and it was mentioned that MIMO application is a key technique of advanced future transmission, which uses several transmitter antennas while the same number of antennas are placed on receiver-side.

The diversity approach permitting the transmission of data at a greater rate with the help of post-modern wireless communication, within which [3], a system is simulated with different configuration of antennas in RX and TX and hence, data transmission occurs under AWGN, RICIAN channels of fading. A zero-forcing receiver is required to handle transmission on remote side of the communication system.

In this system, a QPSK modulation technique was used and BER was analyzed within the mentioned system conditions. The outcomes exhibit that the BER enhancement is dominating at present of AWGN with RICIAN channel, which is more noticeable than the standalone AWGN. The research ends with a diverse approach, which is important for increasing system's performance and it is achievable by installing some transmitted antennas on the transmitter end as well as on the receiver end.

According to the author, [4] the spectrum availability for the daily application has become a challenging task for the radio frequency researchers. The same is accomplished through fast involving applications and production technologies. As some proposed solutions for the spectrum constrains have efficient utilization; a frequency spectrum can be divided and used in the application with care. Many studies have been conducted on this topic. The cognitive radio-based IEEE 802.22 protocol is created for providing an effective usage of available resources of radio spectrum, and this technology is set in wide rural area network and works for searching frequency bands of the television broadcasting network, which ensures zero interference with original data on the network.

The author has presented this technology WRAN by integrating of many modulations like 64-QAM, QPSK and 16 QAM modulations; the data is treated with coding algorithm. Later, it was modulated in a transmission-ready form. The channel is set as AWGN with RICIAN fading effects and hence, it is received by particular receiver.

The system was analyzed with Matlab the outcomes were plotted in the form of SNR curves versus Bit Error Rate. The results were recorded and refined by more and more SNR. In other word, BER improves with increase in the SNR, which is more than the noise power. .

In [5], the research [3] mentions the importance of diversity for the communication systems and here, the author focused on the combination technology, which produces the received signal's super-position. Author also states advantages of wireless communication diversity as it is a method to improve the system's performance.

In the current study, post-detection methods of multipath rays are discussed; signals are detected by multiple antennas and then combined over the fading Rayleigh channel. In the current study, an orthogonal frequency division multiplexing technique is involved. The system was simulated to analyze the post detection combination technique in the above system characteristics while the BER has been improved using PDC method with large channel number.

Results show that the spatial diversity is possible to enhance with high channel order of configuration that decreases the BER automatically at the rated SNR (signal-to-noise ratio), which summarizes this research as: The OFDM system performance increases by the deployment of larger number of antenna configuration, for example: L=2 has lesser performance as compared to L>2.

In [6], MC-CDMA (Multi-Carrier Code Division Multiple Access) is from the recent approaches for allowing multiple hosts with more data rates to communicate with other developed wireless technologies. This study has the implementation and designs of MC-CDMA on various fading channels and Rayleigh, Nakagami and Rician Fading channels have been demonstrated.

The performance is studied and referred to as DS-CDMA results (Direct Sequence Code Division Multiple Access) from the BER curves and time-efficiency plots of many fading channels. Lesser BER and bigger time efficiency results in MC-CDMA and DS-CDMA, which conveys that the MC-CDMA eliminates the selective frequency fading issue in CDMA; however it perfectly participates in creating multi-path fading circumstances.

According to the further approaches, minimizing PAPR uses OFDM on different equalization alternatives, employs various spreading codes, defines FEC at receiver and transmitter, which is achievable to enhance of MC-CDMA performance in these circumstances.

In [7], author has mentioned that wireless communication is having a great significance on the frequency and the fading phenomenon, which demands a good scrutiny to make it gain efficient data at the receiver-end system.

Various parameters affect this process, which are termed as "shadowing" and multi-path propagations. This case yields the instability or swings at the signal of receiver; however, it limits the latest data transmission methods. It happens in the signal phase or amplitude or even in a phase of receiving the signal at antennas.

The wireless communication performance analyses are about performing a cumbersome and complicated task, which is always necessary. Various fading models are utilized perfectly for interpreting the impact of channel fluctuations. Two Wave Diffused Power (TWDP) is a common fading model, which is utilized as frame for other fading channels, for example, Rician and Rayleigh etc.

This type of channel is beneficial for understanding the real world routines. Two wave diffused power fading contains two components of multipath with existence of propagating signals. Within this approach, the essential part will be the evaluation of DPSK and BPSK candidates' performance in two wave diffused power fading environment.

Available fading parameters, PDF density function constructed by Bessel function, and the order of approximation are used to verify the proper SNR at the receiver's output. This project later uses various parameters coming after comparison between the two modulation schemes. This approach conveys different parameters such as the SC performance at the receiver with DPSK and BPSK modulations on fading channels like TWDP, which is understood and produced using MATLAB. The parameters consider either the ABER or PDF outage.

The numerical equations, which are available, are being utilized. Such interesting parameters are selected for illustration purpose. The choice of combiners as diversity methods enhances the SNR at the output in the presence of interferences such as co-channel and noise. Furthermore, SNR versus probability of outage, and ABER plot are determined.

The service quality QoS is ensured by wireless communication and it is largely enhanced by using optimum modulation scheme selection. And hence, enhanced wireless coverage and minimized power utilization are ensured. Techniques that contain a simplification of modulation/de-modulation processes and design result in substantial economic benefits.

In [8], cellular communications or mobile networks have become important for today's telecommunication schemes, said by Arthur .However, the systems are usually affected by multipath propagation. One such study is presented for calculating the symbol error rates and BER probability of M-ary M-QAM (Quadrature Amplitude Modulation) over slow fading, which is identically Gaussian and independent Rayleigh fading channels.

Since fading is considered as a significant wireless communication constraint, the diversity modulation has been utilized for efficiently transferring the message signals. The simulation reveals SER/BER and EnB. For LTE cellular mobile network, 64-QAM is a significant factor, which ensures higher data rate for air interface. Combining the MIMO and OFDM helps in successfully combating with the extra wireless effects of computational complexity and producing 100Mbps or more speed (maximum data rate). M-QAM SER and BER in AWGN and Rayleigh fading channels have been studied.

In [9], the radio spectrum is limited to wireless systems, which keep on growing every day. The spectrum utilization is perfectly possible and effectively done through CR (cognitive radio). CR is using the cognitive property, which is sensing the channel/spectrum to verify that it is available for new users or not. With this research, Arthur has studied BPSK modulation performance using the fading channels presented by Nakagami, Rayleigh, and Rician, and analyzed it using BER.

Matlab was utilized for creating this system for transmitting data under the above mentioned fading channels and conducting comparative analytical study of results. The outcomes of this study are revealed that transmission of data BPSK under AWGN shows best performance amongst the others.

In [10], there is an effort to assure reliable signal reception using wireless, which is significant for reception as well as transmission of information.

For mitigating the degrading impact of a transmission signal, error correction helps and assures reliable recovery of erroneous data bits, which is possible during the deep fading of a channel. The current study discusses how Bose-Chaudhuri-Hochquenghem (BCH) and Reed Solomon (RS) codes actually work with Rayleigh fading.

Researchers carried out investigations to understand the impact of error correction codes for QAM modulation order. This simulation was accomplished through random data encoding, which was generated using BCH and RS codes. Later, it was modulated through changing modulation orders before transmission in a Rayleigh distributed channel. Through a receiver, a signal was decoded and demodulated.

The received BERs were found for correction for retrieving the genuine signal that was transmitted before. System evaluation was carried out to find BER and the outcomes have proven that BCH code works better in case of greater modulation orders. The RS and BCH simulation results of correlated Rayleigh fading are shown.

MATLAB was used to conduct simulation, and the M-QAM modulation was applied in orders such as 16, 32 and 64. The RS and BCH code performances have been analyzed based on BER with the help of random binary data. Outcomes show that for the un-coded signals, when modulation order increases, the performance gets better.

When RS and BCH apply, BER performance declined with respect to order of modulation but it was noticed that the erroneous bits were found in case of BCH 64-QAM, which were fewer in comparison with rest of the signals having different SNR levels. This study has further stressed the necessity for correction and error detection.

Coded modulation

In [11], Bit-interleaved coded modulation (BICM) has been given as a flexible coding mechanism/modulation system that helps designers to independently select modulation constellation. This happens when channel encoder output and the modulator input remain separate on bit-level inter-leaver. For improving spectral efficiency, it is possible to combine BICM and the higher-order modulations like phase shift keying or QAM.

BICM is suitable for fading channels that adds the minor issue of channel capacity whenever comparison is drawn with fading channels and white Gaussian noise. The current study proposes an adaptive BICM technique, which is modelled with a modulation scheme of OFDM and verifies with various channel models.

In [12], it was observed that at present, the industries require some special techniques due to excessive wireless communication, said by the author. These techniques should solve the communication-related problems and challenges, and improves the quality of service. When the transmitter releases a signal towards a receiver, it shows random fluctuations generally known as noise. This is true both for frequency and time domains.

We should predict and eliminate noise. This requirement to predict and measure noise is called as channel state information commonly denoted by CSI. It predicts the extent to noise. But, here the measurement will be observed after some time units. A transmitter chooses right modulation method that depends on CSI. This technique is termed as adoption.

So here, both receiver and transmitter have to send confirmation about receiving a signal. This method is called as CSI feedback. When different techniques are applied together, they are called as adaptive bit inter-leaved coded modulation (ABICM).

Several additional techniques including BICM, UAM, and ATCM exist for the same purpose. BICM has foundations in the concept of Bhattacharya bound, which is applicable to minimum constellation distances and non-adaptive BICM. It determines the transmission power as well as the constellation size. ATCM (Adaptive Trellis Coded Modulation) shows improved performances but it is now considered as out-dated. Generally, ABICM is proposed because it is expurgation bounded and supported through fading prediction.

This ABICM process increases BER precision and provides improved performance and spectral efficiency.

In [13], OFDM gives the foundation to all 4G wireless technologies. These technologies assure multi-carrier modulations. A wireless channel's fading nature can be either destructive or constructive because it has a tendency to degrade the system's performance.

The spectral performances and efficiencies are possible to enhance through modulation schemes or adapting data rates. ABICM (Adaptive Bit-interleaved coded modulation) has been introduced in the current study having OFDM with zero-padding schemes (ZPS). The study deals with ABICM-OFDM-ZPS utilizing expurgated bound that gives correct estimations of BER for static fading including indoor

wireless.
Simulation results show ABICM-OFDM-ZPS performances and in this case, we conclude that it shows improved performance along with reasonable complexity. In [14], a study aims to demonstrate the BICM with soft decision iterative decoding (BICM-ID). In comparison with TCM (trellis coded modulation), BICM-ID provides more diversity with some free Euclidean distance.

With iterative decoding, soft bit decisions help improving conditional inter-signal Euclidean distances, which results in greater coding gains in comparison with turbo TCM both in Rayleigh and Gaussian fading channels having lesser complexity of the system. The researchers resolved some critical design problems, which improved decoding as well as provided analytical performance limits having ideal feedback assumption. The BICM-ID properties show extensive simulations and greater SNRs while the BICM-ID performances converge to error-free feedbacks.

In [15], the presented study investigates BICM-ID in case of bandwidth-efficient transmission and showed that the BER is decreased with the help of iterations between a simple channel decoder and a multi-level demapper. For achieving substantial turbo-gain, the binary indices' assignment strategy is critical.

In the current study, we resolved the issue of selecting a suitable index for arbitrary and higher-order signal constellations. A new process has been proposed, which works with the help of binary switching algorithm for optimized mappings. It has out-performed the previous ones.

In [16], a study to investigate the application of BICM and BICM-ID in orthogonal-frequency-division-multiplexing (OFDM) systems having signal space diversity (SSD) was presented. In this case, the outcome was correlated fading over sub-carriers. Initially, tight bound on asymptotic error-performance for all sub-carriers was derived. It was utilized for establishing improved achievable asymptotic performances through SSD. The pre-coding with sub-groups having a minimum number of sub-carriers for each group while channel taps were enough for obtaining this best asymptotic error performance, and at the same time, it substantially decreases the receiver complexity.

The optimum joint sub-carrier group-and-rotation matrix design can be determined, if we solve Vandermonde linear system. The given examples illustrate agreement between many simulation and analytical results. In [19], new BICM variant has been proposed, which is called as parallel BICM. The L identical binary codes exist in parallel and use mapper, a binary dither signal, and a newly-proposed finite inter-leaver. In comparison with earlier approaches, this scheme is independent of ideal, infinite-length inter-leaver assumptions. Over a memory less channel, the latest scheme has proven to be equivalent to a binary memory less channel for any block length.

Therefore, new scheme facilitates the design coded modulation schemes with the help of a basic binary code. In the nutshell, the coded modulation scheme shows specific performance, which needs analytical evaluated using binary codes in a binary channel. Later, the new scheme was theoretically analyzed considering error exponent, capacity and channel dispersion.

The scheme has identical capacity as that of BICM. The scheme's error exponent was numerically compared with the proposed mismatched-decoding exponent BCIM analysis.

3 Communication systems

3.1 Introduction

Communication defines today's life events irrespective of their kinds; generally, communication does exist, and it takes place through transmission process, in which, a signal is generated and sent across channel to the receiver.

Communication system basically contains a transmitter, channel and a receiver. The given block diagram (Figure 3.1) depicts the general communication system.

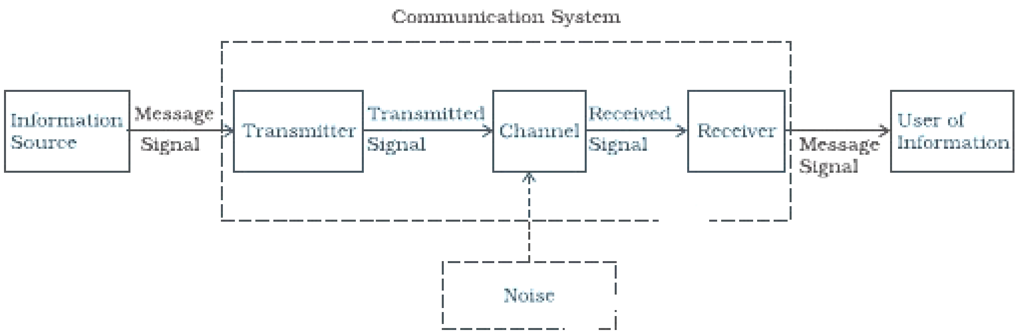

Figure 3.1: Principal structure of a communication system

Usually, receiver and transmitter might be near or far from each other.

The channel is representing the transmission medium, which acts as a physical medium like free space of copper wire, optical cables etc. The channel is influenced through the noise resources and it depends on a channel's nature such as wireless channel or wire channel having different noise parameters can serve as examples.

A transmission signal is generated when the information source produces the information stream; later, the transmitter may transform the signal nature into a suitable form. The term modulation is used in a transmitter for converting a signal in suitable format. In the same way, a receiver performs a reverse process of the transmitter where a signal is de-modulated and filtered.

The receiver tries to recover the signal as precisely as possible from the transmitter. The information system cannot produce the signal that is transmittable by physical channel, while the signal only radiates when it is transformed into electrical form; wire or wireless channel, which are utilized for conveying the signals to receivers.

Two terminals can communicate privately without involving any other terminal called as point-to-point communication. Alternatively, a transmitter can contact multiple receivers called as broadcasting where a signal is accessed by all the receivers.

3.2 System infrastructure

It is an entity that is responsible to transform information digits into electrical form, which is usable by physical channel, and this definition is valid for digital communication systems. Basically, the transducer can play the role of signal conversion from original form into different forms, which are suitable for transmission by means of physical mediums.

- Signal: Information bits generated by information source can form a magnetic field component called as signal. These signals can be analog including pressure, temperature, sound signal, and light signal. These signals can be sensed by a human and monitored with the help of a physical system, and besides, mathematical form can be derived for any physical quantity.

- Another signal type is digital signal, which is in fact a binary signal, and it is most familiar where the signal seems as rectangular wave containing zeros and ones. For digitizing any analog signal, we can call the processed such as ADC (analog to digital conversion) for example: the temperature that changes the form that is understandable by computers by firstly detecting the temperature using special sensors. This sensor is included with such tools to scale the analog temperature fluctuation into the equivalent level of digital format.

- Similarly, term DAC is stands for digital to analog conversion where the computer output is changed into electrical form, which is understandable by humans such as the voice that leaves the computer and enters the speakers.

- Noise: It is an essential element for every communication system, which interrupts smooth transmission process.

- Noise signal: It is a random process that destroys the transmitted information. Different resources of noise can be noticed in a communication system. Thermal noise takes place due to electronic radiation; otherwise, external noise is presented such as AWGN.

- Transmitter: Performing the steps for transmitting information from its source to its pre-planned destination is the function of a transmitter.

- Receiver: It extracts the needed information from a receiver signal, which comes from a transmitter through the channel. Noise may be eliminated at this point in a communication system.

- Modulation: It integrates a transmitter and converts the data into electrical form. Two modulation types exist and named as analog and digital modulation analog. In analog modulation, carrier signal with higher frequency is used.

- In digital modulation, the source bits are directed into modulators such as BPSK, QPSK while in binary phase shift keying, the zeros and ones are scaled with analog signal. In this way, two bits are transmitted by one signal.

- Demodulation: This process is available on the receiver-end and it is utilized for retrieving genuine data, which is sent by the transmitter. Demodulation with analog signals is possible through processing and changing a signal through local oscillator, and hence, it is actually reverse modulation.

- Repeater: It consists of a combination of receivers and transmitters, which repeatedly transmit a signal between multiple transmitters and receivers. The repeater is utilized to extend the level of communication by amplifying the original signal from first transmitter and repeat it to many receivers. The carrier wave may change with different transmissions. Figure 3.2 depicts the satellite communication repeaters.

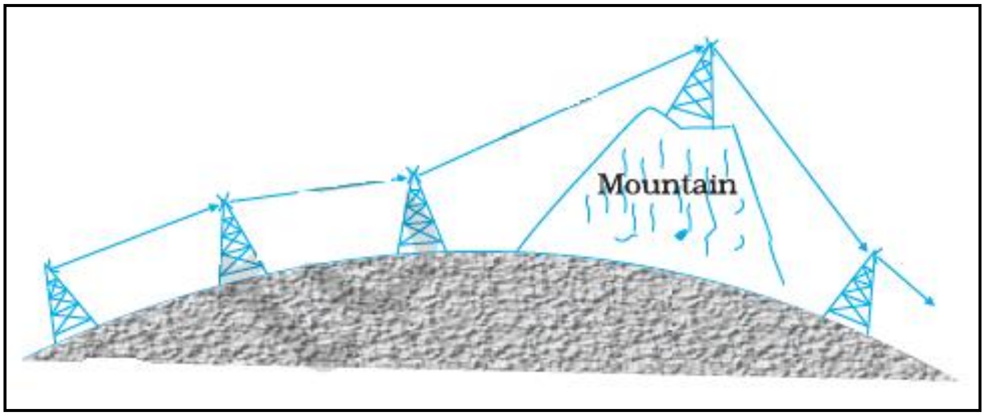

Figure 3.2: RF transmission fashion.

3.3 Medium of communication

A channel is very important for any kind of communication system. Many channels exist such as Rayleigh fading channel, Additive white Gaussians noise channel, and Rician fading channel. Rayleigh fading means multiplicative distortion h(t) of a signal s(t), since $y(t) = h(t)\cdot s(t)+ n(t)$

Here, y(t) represents waveform and noise is n(t). A wireless is generally unstable; therefore, fading takes place because of multipath propagation.

Multipath propagation shows quick rapid phase amplitude fluctuations. Multiple paths are created when the reflectors surround a receiver and transmitter because the transmitted signals can traverse. Consequently, a receiver has a superposition in transmitted signal copies and every one of them traverses towards a different path.

Each signal copy experiences different delays, attenuations, and phase shifts when it travels from towards the receiver. It might lead to a destructive or constructive interference, which either amplifies it or attenuates the power of a signal on the receiver side. Fading can be large-scale or small-scale.

In case of multi-path time delays, the small-scale fading process can be either frequency selective fading or flat fading. If a signal bandwidth is lower as compared to the channel bandwidth while the delay spread is lower as compared to a relative symbol period, in that case, flat fading will occur but when the signal bandwidth is more than the channel bandwidth, the delay spread is more as compared to the relative symbol period as frequency-selective fading takes place.

Based on Doppler-spread low-scale fading can be fast or slow. Fading takes place slowly when the channel's coherence time is more as compared to the channel's delay constraint. The changes in phase and amplitude are generally constant during the time when they are used.

Slow fading takes place through occurrences like shadowing when a major obstruction like a building or a hill exists within the path of the main signal between the receiver and transmitter. Fast fading takes place for small channel coherence time. The phase change and amplitude vary during the different tenures of its use.

In case of fast-fading channels, a transmitter might get benefit out of the changing channel conditions for different time diversities that improve the communication robustness. Nakagami fading model includes multi-path scattering having substantial delay-time spread having varying reflected wave clusters.

In a single cluster, the individual phases of reflected waves exist in a random way; however, the delay time is generally equal in case of all the waves. Consequently, every cumulated cluster signal envelop is distributed with Rayleigh distribution. The mean of the time delay significantly differs for different clusters. When the delay time significantly increases as compared to the bit time in a digital link, different clusters create substantial inter-symbol interference; therefore in that case, the multi-path self-interference becomes closer to the co-channel interference.

Rayleigh fading model states that fading takes place through multi-path reception. Rayleigh fading model also assumes a random signal magnitude, which moves through transmission medium almost randomly. In this case, Rayleigh fading is helpful because many things can interrupt a radio signal before its arrival to the receiver.

Rayleigh fading applies whenever no dominant line-of-sight propagation exists between a receiver and transmitter. Rician model takes into account a dominant wave, which is possibly a phase sum of either two or more-than-two dominant signals. These combined signals show deterministic treatment in a predictable method. In this case, the dominant wave is possibly subjected to shadow attenuation.

It is an important assumption for satellite channel modelling. Other than dominant components, mobile antennas have the capacity to receive magnanimous numbers of scattered as well as reflected waves.

3.4 MIMO vs. SISO channel

For wireless communications, increased data rate is possible through utilizing MIMO systems.

The MIMO system consists of several antennas on receiver as well as transmitter side. Its performance is enhanced using diversity in comparison with the single antenna systems. Some future wireless communication trends demand for MIMO development. They are mentioned below:

- Need of "high data rate" and "high link quality"
- Limited RF spectrum
- Need to improve the limited capacity and transmission
- Time & frequency limits

The use of several antennas has three benefits: formation of beam, spatial diversity and SDM (space division multiplexing). MIMO systems use spatial diversity through many spatially-apart antennas for creating a denser scattering environment. This system's capacity is possible to enhance through further spatial channels with the help of space-time coding.

The MIMO systems are implemented in many processes for obtaining the diversity gains in order to deal with signal-fading or gaining additional capacity. The MIMOs reduce the BER/data rates with the help of many receiving and transmitting antennas. It is possible to improve the BER and better data rate is attainable with the help of STBC (space-time block coding), which is a main MIMO system scheme.

Here OSTBC3 has been utilized for space-time coding. Major STBC functions include spatial multiplexing and diversity. The improved performance needs diversity-multiplexing trade-offs. In broadband wireless communications, appropriate STBC implementation helps improving diversity as well as performance of a space time (ST) coding system with the help of many parameters including type of trellis codes and channel fading.

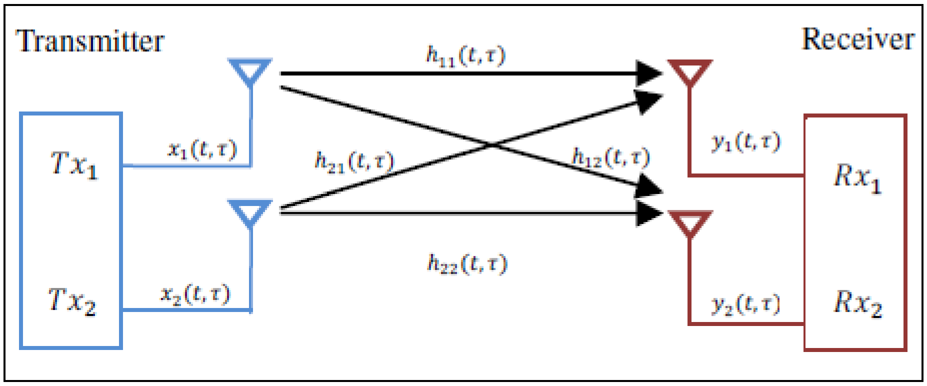

Figure 3.3: Basic representation of M X N MIMO System.

The fundamental M X N MIMO systems representation is given in Figure 3.3. For this project, we have considered the impact of antenna configuration on MIMO system performance with Racian and AWGN channels. Data impairment was observed during transmission that took place because of the channel noise and/or multi-path fading.

3.5 Multiplexing

In multiplexing techniques, several communication systems are linked with each other because they share common channels. Figure 3.4 illustrates the core multiplexing principle to share common channels with the help of K (users' number). The frequency division, time division, code division and spatial division are the popular methods to execute multi-channel signalling with a single channel.

We will discuss them briefly with their histories

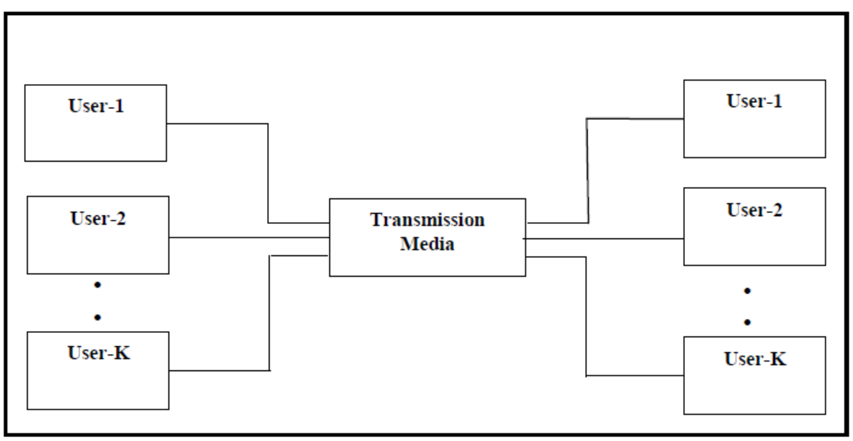

Figure 3.4: Principle of multiplexing.

3.5.1 Time Division Multiplexing and Time Division Multiple Accesses

When telegraph was declared as a commercially successful technology back in 1840s, telegraphic codes (lines, dots & dashes) become necessary to carry out simultaneous transmissions of several signals through just one channel that builds capacity.

F. C. Bakewell, M. B. Farmer, and A. V. Newton proposed several processes, which could assure implementation of TDM with the help of synchronously-driven rotational commutors [3]. B. Meyer, P. Lacour, J. M. E. Baudot, and P. B. Delany deserve credit for technical as well as theoretical improvements. Willard M. Miner was the first to introduce a TDM system that worked with the help of speedy rotating commutators, which is shown in Figure 3.5.

At that time, sampling theorem was not yet introduced, so, an experiment showed that the commutator speed is closer to upper frequency components of speech, and so, the best result was obtained. When the use of telephone became commonplace, multiplex signal transmission opportunities emerged. Issue of time-division multiplex for telephonic communication is linked with the sampling problem.

Until now, it was unclear whether samples can be instantaneous or have non-zero durations. Willard Miner conducted some previous researches on telephone multiplexing and he determined the sampling frequency through trial and error. Raabe's thesis was published in 1939 and what he did was beyond experimentation.

He also has demonstrated a better description of sampling with finite-duration pulses and low-pass and band-pass signals. Raabe showed that many channels carried telephonic signals, which can be reconstructed and multiplexed while having a little error, given that it meets a particular condition. The condition was called Raabe's condition that requires sampling frequency to be two times as much as a multiplexed signal's highest frequency.

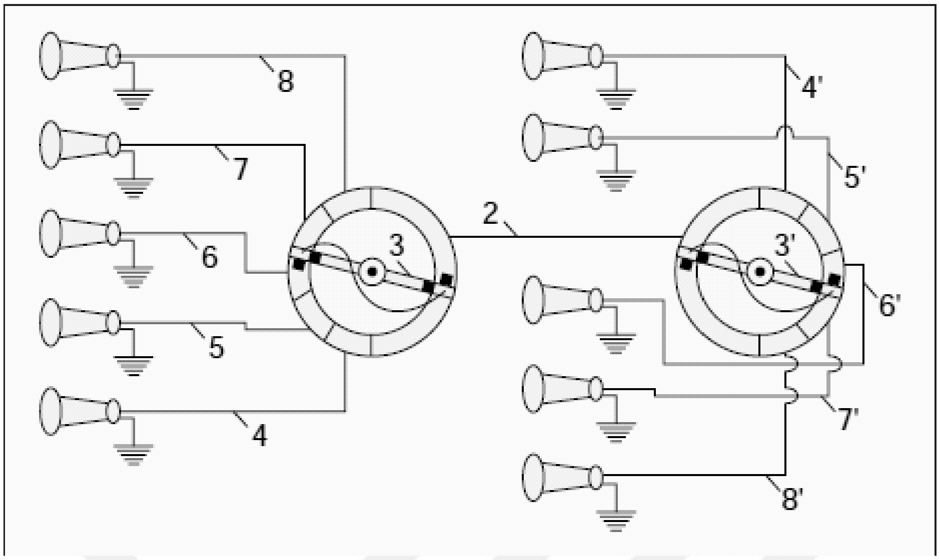

Figure 3.5: Block diagram of commutator based TDM system.

With the advent of satellite communications, time division multiple accesses (TDMA) was a method, which attracted interest for wireless communication. Several access points are part of a specific multiplexing, in which, at least two signals coming from different sources, are sent by a radio to a single transponder. Figure 3.6 illustrates TDMA principle to accommodate N-channels.

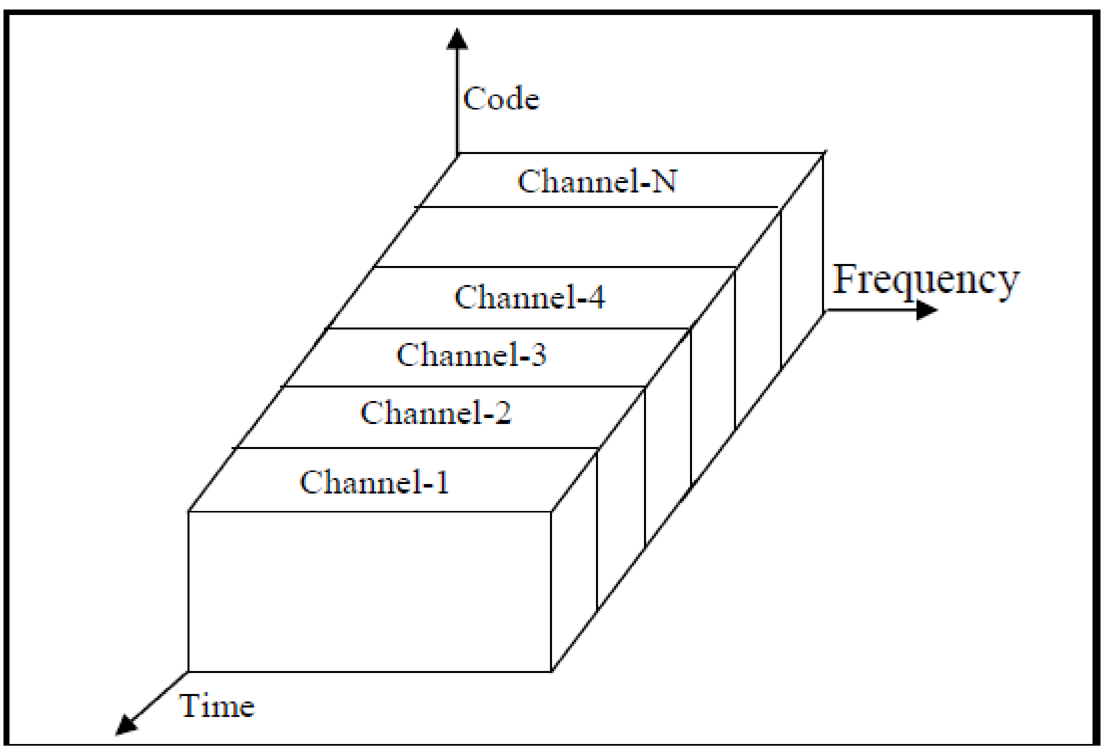

Figure 3.6: Principle of TDMA.

3.5.2 Frequency Division Multiplexing and Frequency Division Multiple Access

Many works pertaining to the frequency division multiplexing (FDM) have appeared in journals so far, which describe the FDM evolutionary history. The researchers and writers claimed that Alexander Graham Bell evolved FDM during his experiment on a telegraph multiplexing system Figure 3.7.

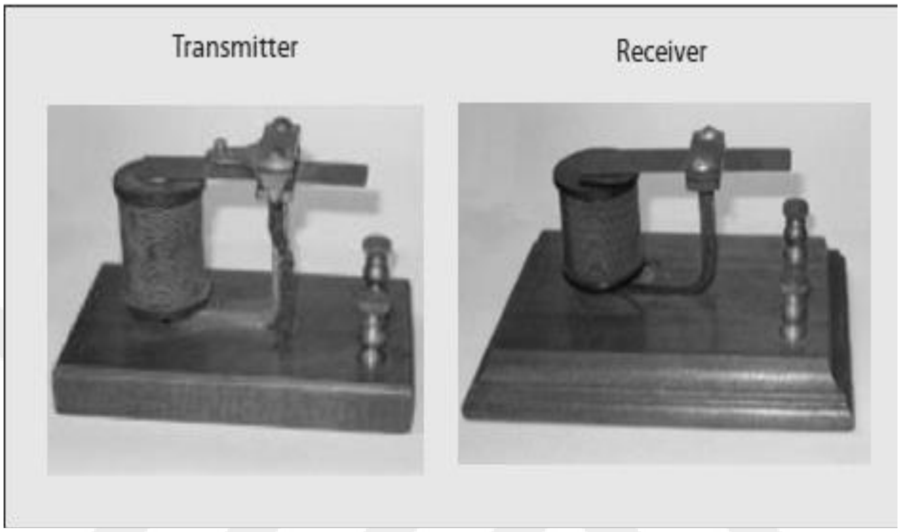

Figure 3.7: Bell's harmonic telegraph system.

Just like TDMA, FDMA also got its recognition after its commercial launch of satellite and cellular service in 1970 and 1980 respectively. A geostationary satellite uses earth coverage antenna, which can see approximately one-third of the earth's surface. The satellite can see any point within its viewing area and works as a means of multiple accesses. Figure 3.8 shows FDMA with the help of a block diagram.

FDMA was the first multiple access process in the field of satellite communications. When satellite communications began in the 1960s, a greater part of communication traffic was telephony, which was communicated through satellite. All signals were analog, and analog multiplexing was used on earth stations for combining large numbers of telephone channels into a single baseband signal that could modulate through a single radio frequency.

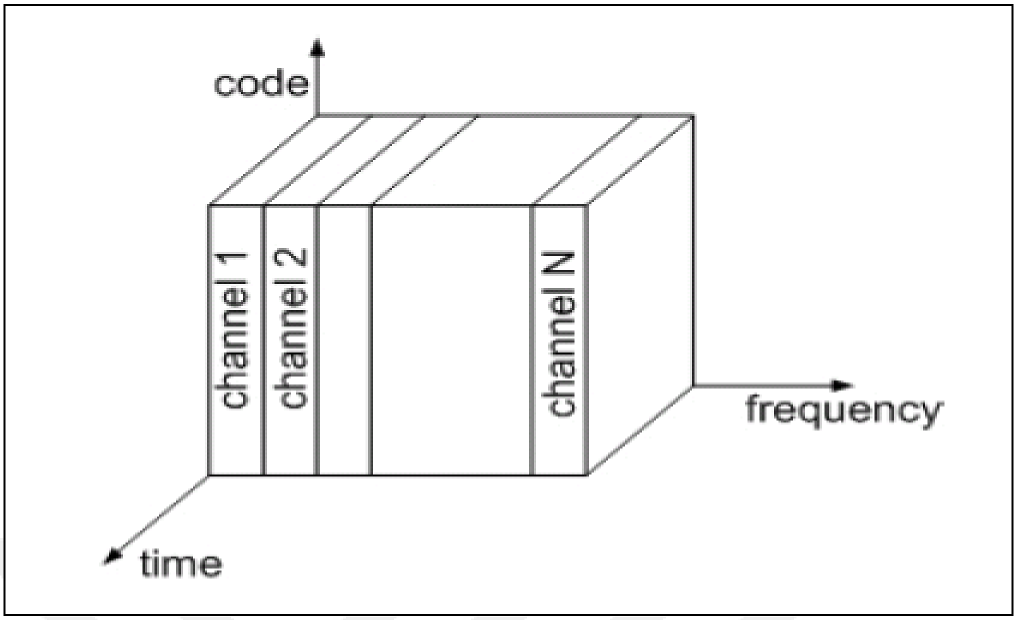

Figure 3.8: Principle of FDMA.

3.5.3 Code Division Multiplexing and Code Division Multiple Access

When frequency reuse concept was introduced in cellular technology, it greatly expanded the efficiency and capacity. The first cellular service started operating in 1980 using analog radio transmission in the form of advanced mobile phone system (AMPS).

Within a few years, cellular systems began to face the challenge of capacity saturation as more and more users started demanding mobile services. To solve this problem and to accommodate more users, a new set of digital wireless technology called TDMA and GSM (Global system for mobile) was developed. But just as TDMA was being standardized, an even better solution was found in CDMA. Figure 3.9 illustrates the principle behind CDMA technology.

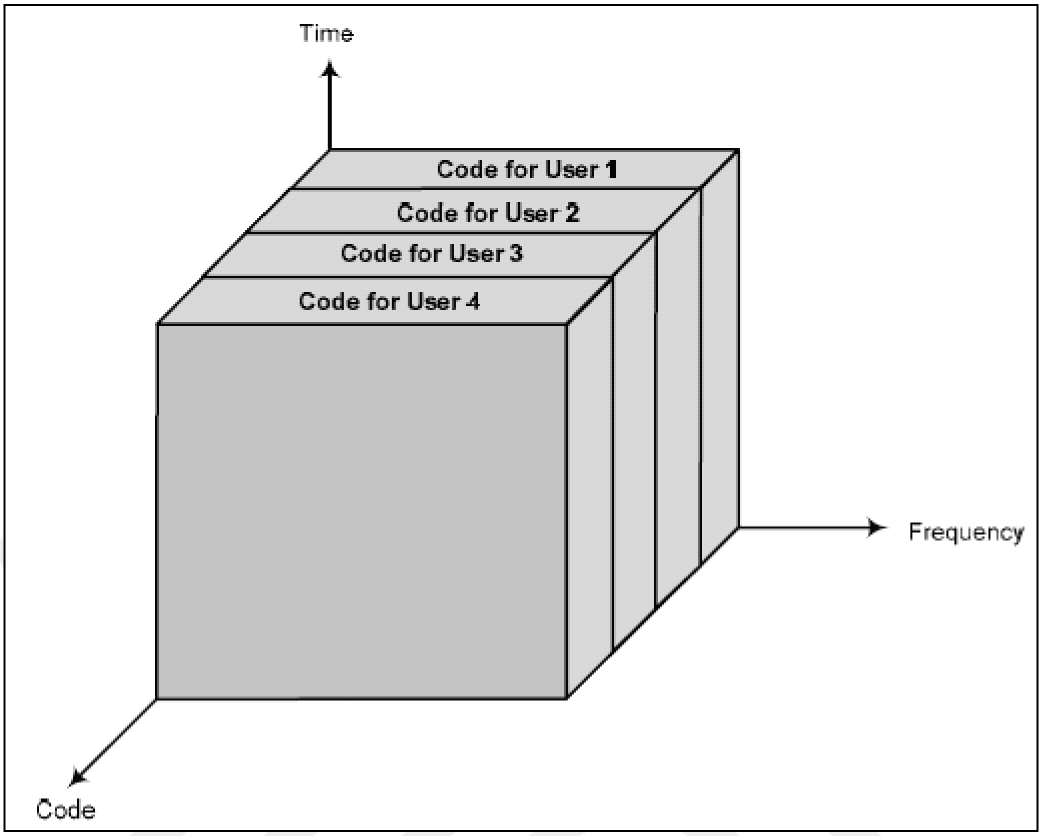

Figure 3.9: Concept of CDMA.

3.5.4 Space Division Multiplexing and Space Division Multiple Access

Space division multiple access (SDMA) uses spatial separation of the users for optimizing performance and bandwidth under noise. In this case, each user radiates power, which is controlled by SDM using a spot beam antenna.

These areas may be served by similar frequency or different frequencies. Figure 3.10 is SDM block diagram. For limited co-channel interference, cells should be sufficiently separated. When multiple antennas are used on transmitter as well as receiver ends, SDM recovers mixed up data streams.

The multiple-input multiple-output (MIMO) technology is a method of SDM, which is used for wireless communication design. Spatial dimensions are created through a different receiver and transmitter, which combat with multi-path fading.

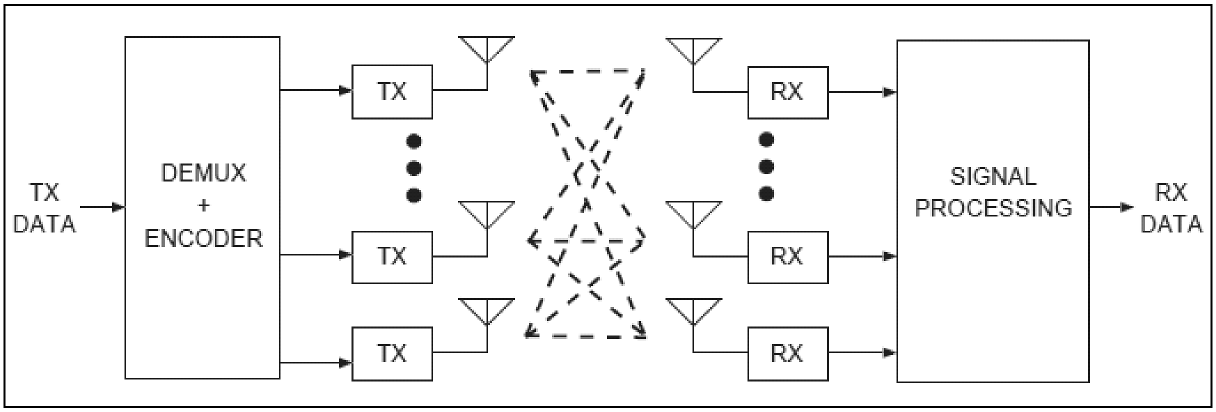

Figure 3.10: Block diagram or SDMA System.

4 Methodology

4.1 Modulation Scheme

During information transmission, modulation is a process, in which, signals are available for physical channel. It shifts the function of signal transformation using a genuine form into another form for use in a channel. Modulation falls into two categories: analog and digital.

Analog signal is a modulated signal having frequency (Fx), which performs the multiplication with higher frequency to plays a role of carrier. Another type of modulation is digital, which is utilized with signals of digital nature. Frequency modulation, amplitude modulation and phase modulation are the main analog modulation schemes. Binary shift and phase-shift keying, and QAM are the fundamental forms of digital modulation. It is possible to write a transmission signal by de-combining it in real/imaginary parts, assuming that S(t) is a transmitted signal, the signal will be represented in the following expression:

$$S(t) = Si(t) + j.Sq(t)$$

The transmitted signal M(t) is obtained by just taking the real part of complex carrier part (ωct)) of the signal is:

$$M(t) = Real\,[\,S(t).e(-jwct)]$$

$$M(t) = Si(t).\cos(wct) + Sq(t).\sin\,(wct)$$

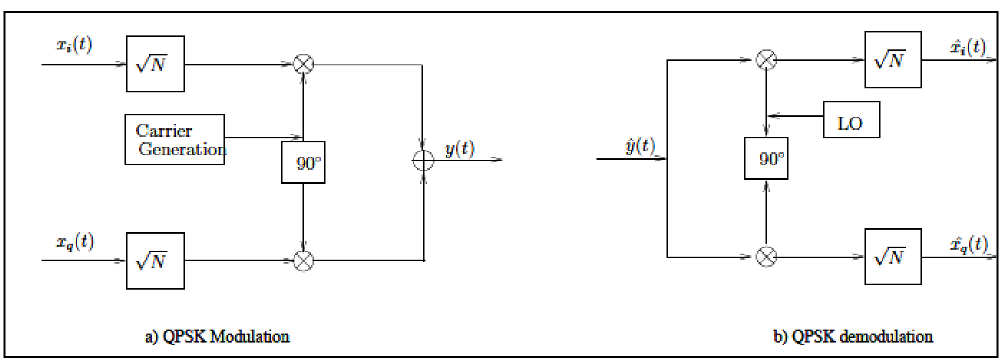

Figure 4.1: Modulation and Demodulation of QPSK

Signal generation and detection of Quadrature phase-shift keying signal. It is described in figure 4.1, which is a constellation diagram showing the pattern of this kind of signals and determines the direct QPSK signal and then mapping them into the phase angle (in-phase or quadrature). Constellation diagram is illustrated in Figure

4.2

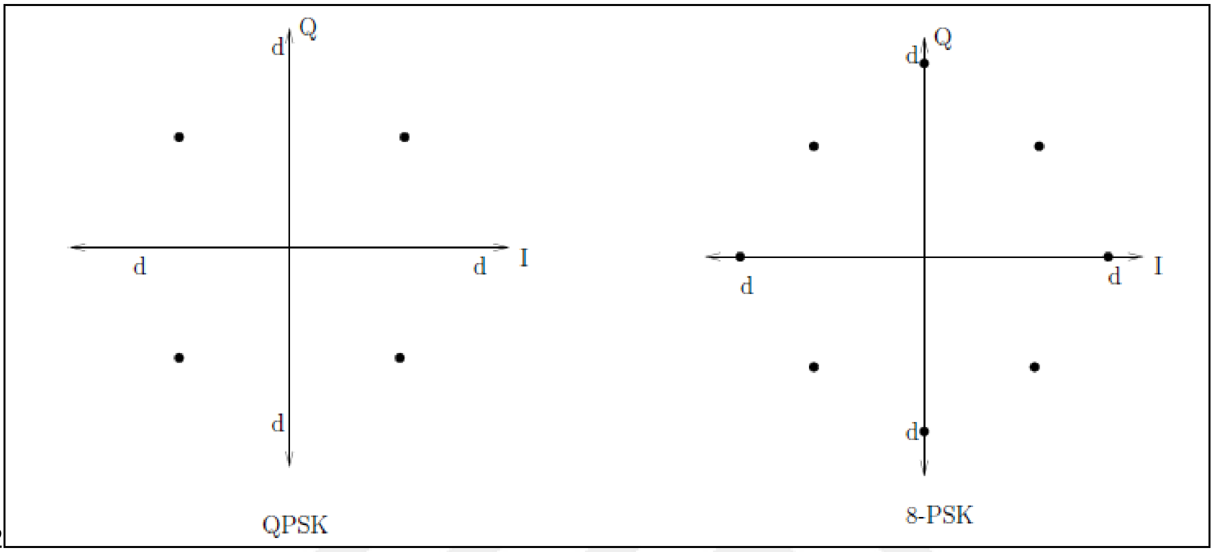

Figure 4.2: Constellation Diagram for 4 and 8-PSK

From M-ary modulation, BPSK modulation is a specific case of those categories, and modulation takes place by amplitude and phase changes are illustrated through a constellation diagram. For BPSK, four possible bits are possible to transmit whereas two bits were transmitted with BSK.

Signal of carrier will be having a single value, called as θi = 2(*i*-1)**π/M;* the term *I is equal to* 1, 2, 3, 4.., *M integers*. Signalling duration for one possible M signals can be given as:

$$s_i(t) = \sqrt{\frac{2E}{T}} cos\left(2\pi f_c t + \frac{2\pi}{M}(i-1)\right) i = 1, 2, 3, 4 \cdots , M$$

Where E is the energy per symbol and fc is the carrier frequency.

4.2 Capacity

In any channel, the term capacity is linked with maximum information limit, which can pass through that channel at the output called as throughput.

The Shannon's capacity equation is expressed as:

$$\eta_{Bmax} = \frac{C}{B_p} = log_2(1 + SNR) bits/channeluse$$

This formula is referring to the input of Gaussian distributed bit stream heading towards a channel, which could be supposed as continuous-output continuous-input memory-less category of the channels while the term CCMC refers to the channel capacity.

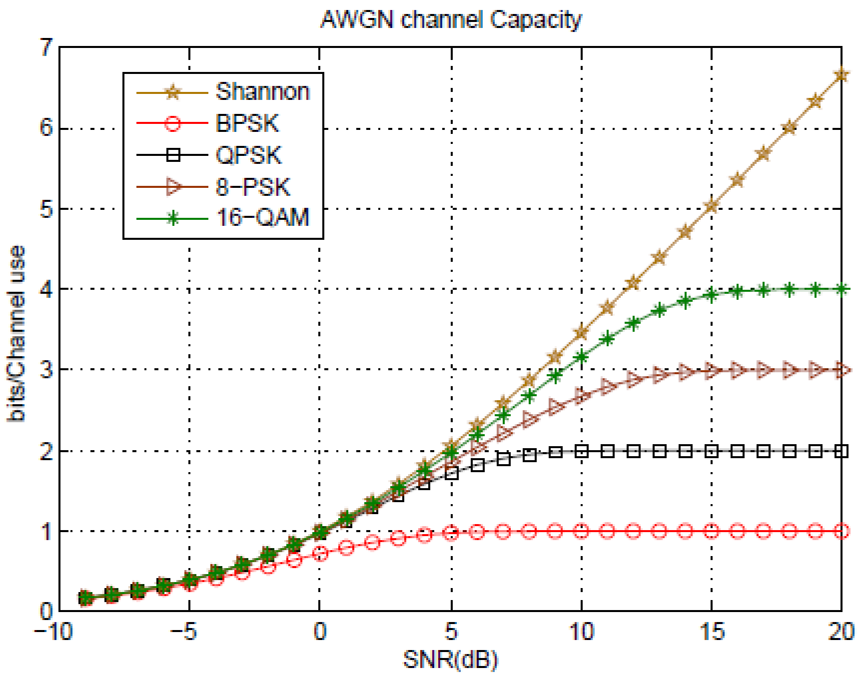

Figure 4.3: DCMC Channel Capacity for Different Constellations Under AWGN Channel

The DCMC mainly limits a channel through log2M at a stage where SNR of that channel is rising while M represents points on a constellation diagram. The channel capacities under varying different modulations in AWGN are illustrated in Figure 4.3.

4.3 Mapping Schemes

Bitmapping procedure shows the method to map bits to corresponding symbols before the transmission. Two major classifications of bit-mapping methods exist including Gray Mapping and Natural Mapping (set partitioning). Gray Mapping takes place when the constellation points' hamming distance is 1.

Gray Mapping successfully takes place but it is unable to minimize the BER, so, we mainly use natural mapping schemes. The illustration of mapping schemes is given in Figure 4.4. The analysis of natural Mapping scheme has been given in a later section.

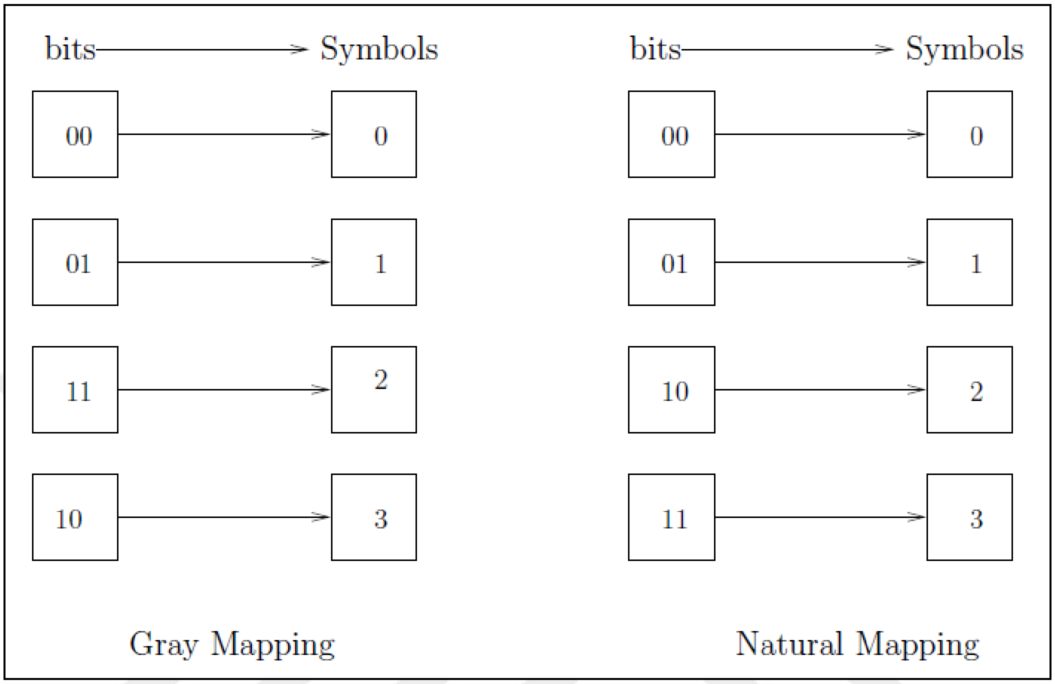

Figure 4.4: Bit mapping relation

4.4 7-PSK

7 PSK is similar to 8 PSK modulation systems other than the fact that 7 PSK systems have 7 constellation points in addition to one on the origin. The 7 PSK modulation signal is as follows:

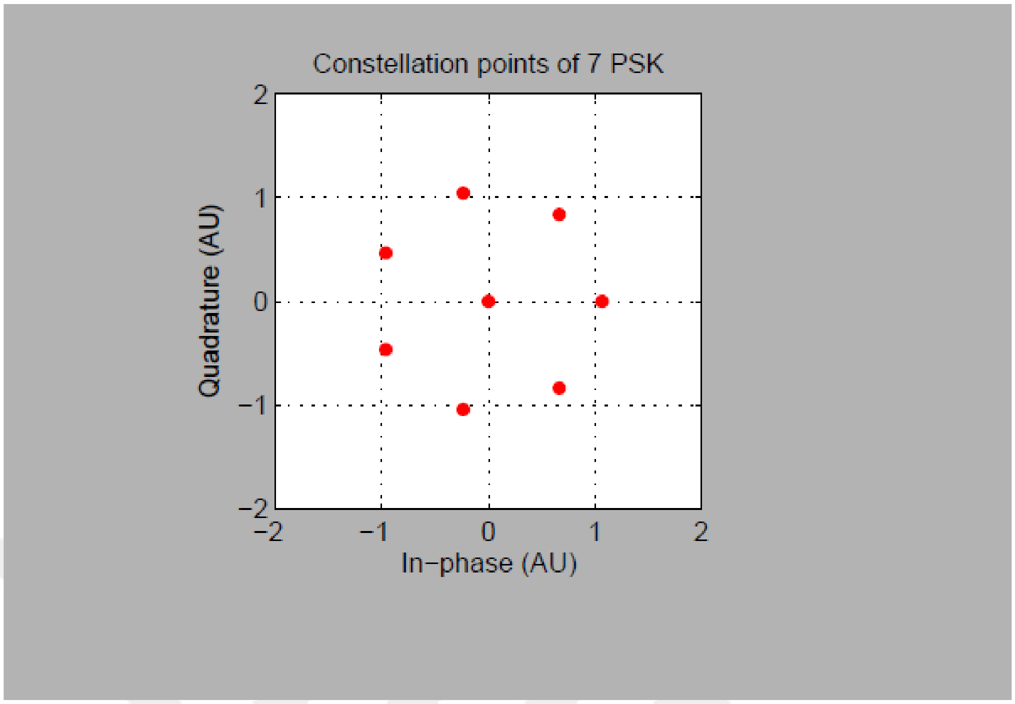

Figure 4.5: 7-PSK constellation diagram.

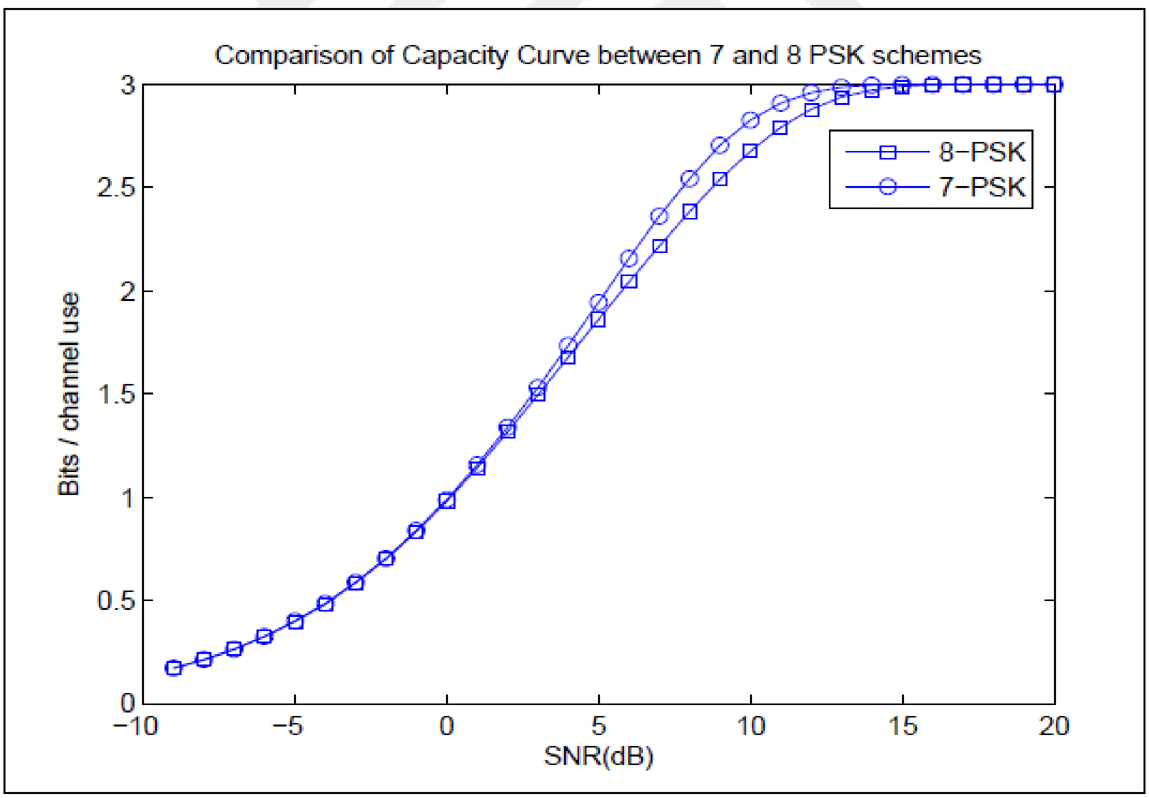

Figure 4.6: 7-PSK and 8-PSK capacity comparison.

$$s_i(t) = \begin{cases} \sqrt{\frac{2E}{T}} cos\left(2\pi f_c t + \frac{2\pi}{7}(i-1)\right) & (1 \leqslant i \leqslant 7) \\ 0 & (m = 8) \end{cases}$$

Where E represents energy of each respective waveforms when (1 <i < 7) for a transmission signal. The constellation points for the 7 PSK are illustrated in Figure 3.5. Here, the conventional difference between 8 PSK is a centre point in a constellation diagram. The 7-PSK can achieve a capacity that is much higher than that of the 8-PSK with capacitive efficient modulation scheme.

The 8-PSK-7-PSK capacity comparison is given in Figure 4.6.

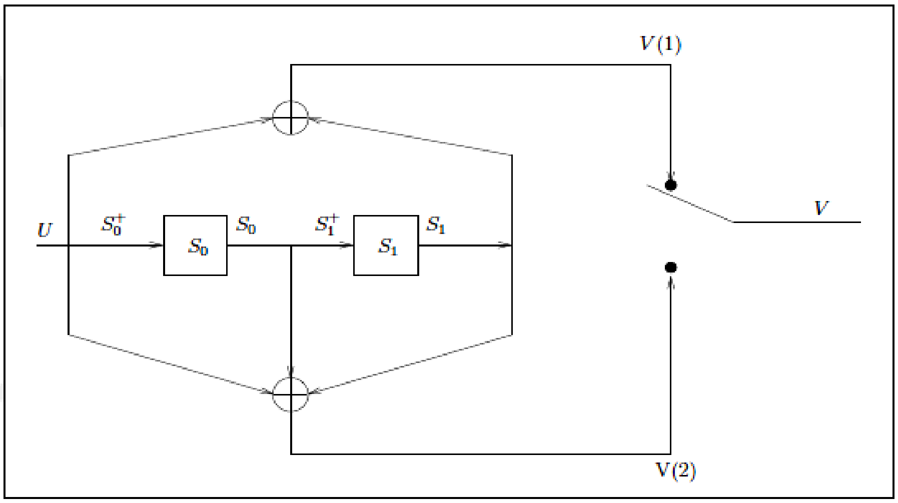

Figure 4.7: Convolutional Encoder of rate ½.

4.5 Convolutional Code

In general, k/n convolution encoder has M-element shift register, k per input information bit and n output coded bits, which are given by the linear register content and bits of input data.

Generally encoder having rate 1/n is used, having binary convolution codes .The "n" generating polynomial has been described by the specific connections to the register stage. Upon clocking a shift register, the signal moves to next state and so on. The generator polynomials constitute a binary pattern, which indicates absence/presence of a specific relationship with a specific shift register.

For example by referring the figure 4.7, a generator polynomial will be:

$$g_1 = \left[[1 \quad 0 \quad 1]\right], g_2 = \left[[1 \quad 1 \quad 1]\right]$$

4.6 State and Trellis Diagram

It is a significant method for characterizing a state-machine operation, in which, a convolution encoder (CC) encoder refers to a state diagram in Figure 4.8.

The diagram fully describes the encoder operation including how the state has been transferred from first state another, and which path it can choose when it is moving from a path to another. Now, considering the above mentioned convolution encoder that has a couple of bits in two shift registers at any time. There are four states and the state transition takes place through input bit U.

The constraint imposed through a respective encoder restricts them to only two legitimate state transitions, which depends on the input type. Another simple way of encoder representing is portraying its trellis diagram, which is given in Figure 4.8. On left, four states are portrayed. State transition is controlled through input bits and state transitions, which take place because of logical "0" which is shown in Figure 3.8.

If we denote convolution encoder input as U, next state be S+0, S+ and the output be V, the State Transition equation will be as follows:

$$S_0^+ = U \qquad S_1^+ = S_0$$

$$V_1 = U \oplus S_1 \qquad V_2 = U \oplus S_0 \oplus S_1$$

Since here the memory elements (m) number is 2, so 2m = 22 = 4 states (S0, S1). These states will change accordingly to the input information word U to get the output (Vi) and the next states (S0, S1) respectively as demonstrated in table 3.1.

Information U	Initial State S_0S_1	Final State $S_0^+S_1^+$	Output $V_1V_2]$
0	00	00	00
1	00	10	11
0	01	00	11
1	01	10	00
0	10	01	10
1	10	11	01
0	11	01	01
1	11	11	10

Table 4.1: State Transition Table.

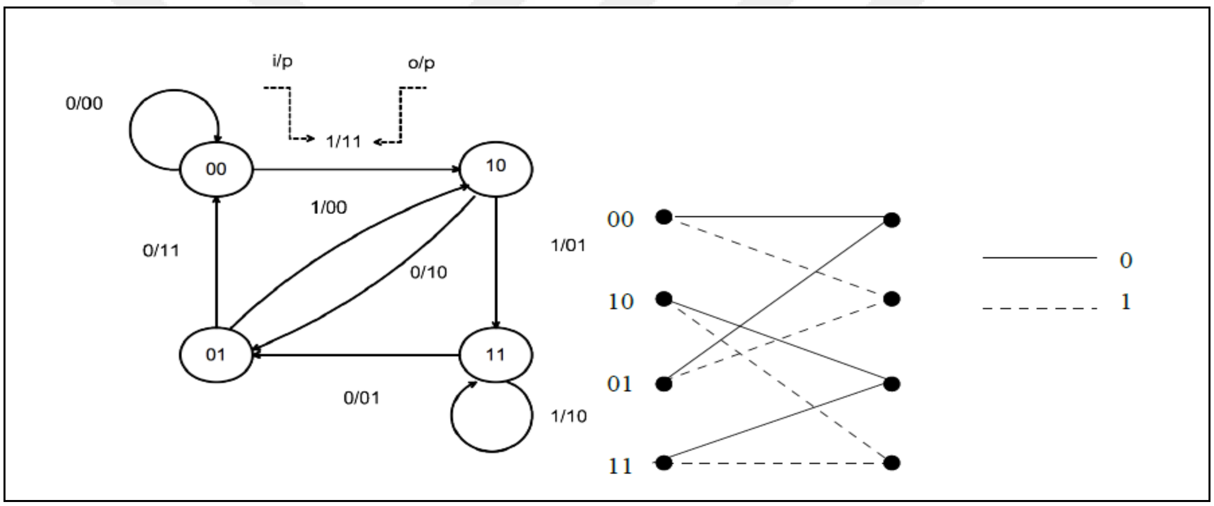

Figure 4.8: State Diagram and One Stage of Trellis.

4.7 Viterbi Algorithm (VA)

Andrew Viterbi [9] developed it and it is established that the algorithm calculates highest likelihood code sequence from the received data. This algorithm is used in several applications such as GSM phones, space probes etc. It takes place either by hard or soft decoding which is explained in later subsections.

4.7.1 Viterbi Hard Decoding

Here si(k) represents stage i trellis. Each state si(k) is associated in a state metric or branch metric M(si(k)) and a trellis yk path.

A Viterbi algorithm states “At time i, the likely paths yk i finally coincides at i-l. In the beginning of Viterbi, the algorithm operates from the zero state, and later it makes a comparison with the output for encoded sequence of the trellis based on Hamming distance (HD), which is nothing but represents different bit position between two binary sequences.

Now, for output symbol 10, the associated hamming distance will be 1 against both 00 and 11 of the encoded sequence. During this stage, the decoder is unable to express any preference regarding 0s and 1s. This Hamming distance is actually a Viterbi decoding context and known as branch metric.

Now when we proceed toward a next symbol, hamming distance will be computed over again for a received signal and four legitimate paths. This distance yields a new branch metric, which is linked with 2nd stage. When the encoded symbol has two genuine bits as inputs, the obtained branch-metric will be added with previous branch metric to obtain the path metric.

A lower Hamming distance indicates a high similarity between encoded and received sequences, which is a characteristic of the most likely encoded sequence, since possibility of a high numbers of error exponentially decreases.

4.7.2 Viterbi Soft Decoding

Hard viterbi decoding depends on received coded symbol location, and in this case, a coded bit can be found when a received symbol is more than 0 and coded bit will be 1. When a received symbol is approximately 0, received coded bit will be 0.

For soft-decision decoding, the estimated coded bit as well as Hamming distance are found along with the distance between transmitted and received symbols. It is possible through 8 confidence levels, for example, +4 scale shows maximum confidence in a demodulator’s decision to have binary 1 and -4 for the lowest possible confidence. In fact, if the demodulator output is -4, the low confidence implies a high probability of a binary zero.
Bearing this eight level confidence scale, the received bits are decoded.

4.8 Puncturing Convolution codes

This procedure systematically deletes or stops a few output bits in a high-rate-encoder. Since, a low-rate trellis structure remains the same; the average rate of information bits for each symbol sequence does not change.

Consequently, a putout sequence belongs to the higher rate punctured convolution codes (PC) codes. A goal of puncturing is that the same decoder is utilized for a verity of high rate codes. Decoding a PC code uses Viterbi algorithm, which is described earlier by the insertion of the "deleted" symbols in the position that were not send. It is known to be de-puncturing. The deleted symbols are marked by a special flag.

The Viterbi Algorithm, which has been discussed in the previous chapters, increases the likelihood that the received data does not assure minimum symbol error; therefore, a further decoding algorithm BCJR is utilized to find the probability of an incoming bit, so it is a better choice. The details of algorithm are observable in the coming sections.

4.9 Trellis Coded Modulation

The concept behind this kind of modulation [2] is that in place of sending symbol “m” when the modulation is accomplished, extra parity bit is added for making the constellation points twice as before and at the same time, the effort is to sustain the same effective throughput. For instance, when two 4-level information bits under Phase Shift Keying (PSK) exist, they introduce parity through scaling genuine constellation points to eight, i.e. by making it to 8 PSK.

In consequence, redundant bits are taken using expanded constellation diagram in place of improving a signal rate (bandwidth). Ungerboeck [2] has sufficiently explained how to use TCM schemes for redundant non-binary modulations along with finite state Forward Error Correction (FEC) encoder, which selects the coded signal sequence.

The extra bits formed by corresponding convolution encoders restrict the possible state transformation among the consecutive phase to a certain legitimate constellation. A receiver decodes noisy input signals using trellis-based soft-decision detection system.

The term “Trellis” describes this scheme because the overall operation can be described by a corresponding state transition diagram similar to binary convolution encoder. In case of TCM, trellis branches having labels according to modulated phases.

4.9.1 TCM Principle

The TCM can be illustrated with the help of 8-state trellis code 8 PSK because it is simple and creates better understanding.

When 2-bit transmission takes place, 8 PSK is used. A 2/3 rate convolution coder has higher hamming distance for a respective constraint length [19] and a Gray coding as the mapping scheme. Yet, there are problems when this approach is used. As Ungerboeck [2] suggested that the encoder design should have high Euclidean Distance (ED) but in case of Gray-code mapping, that is unable to convert the large HD into larger ED, and so, the permeation of binary output significantly influences the ED.

This is the good time to explain the concepts of Euclidean Distance(ED) and Hamming distance (HD). For signal constellation, ED represents distance between points on a constellation diagram. In Figure 2.9 do, d1, and d2 represents the minimum ED. Like real numbers, binary numbers have the concept of distance.

Let's compare two binary numbers; say 1011 and 0100, their hamming distance is 4, which is obtained after comparing and adding them.

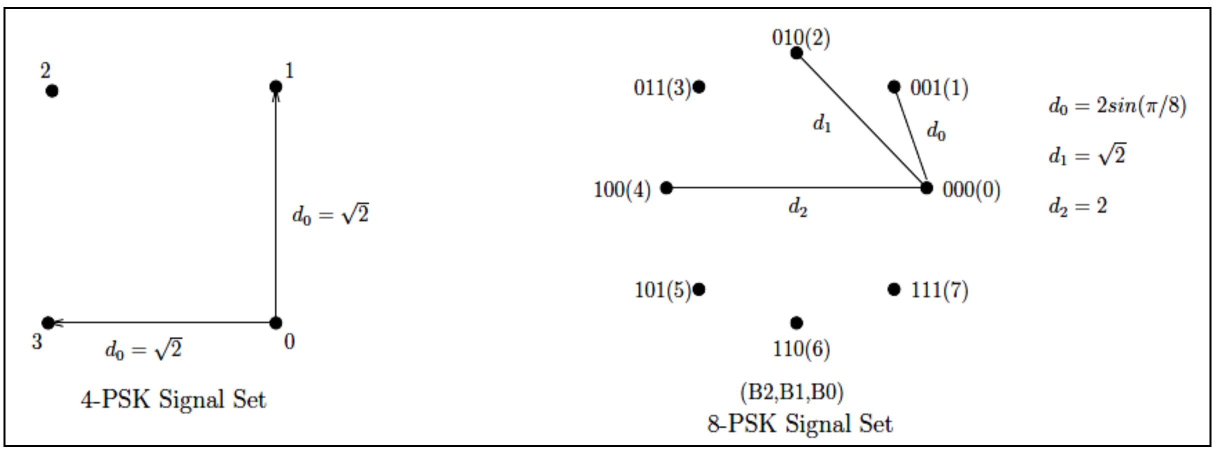

Figure 4.9: Constellation Diagram for 4 and 8-PSK.

5 Simulation and results

5.1 Outline

In order to investigate communication system performance over AWGN and Reyleigth fading process, the said system works with simulation by Matlab software.

Firstly, random bits and terms for source are generated, which are transmitted through fading channels and AWGN to adjust its characteristics to the noise parameters after being combined with a transmitted signal. This structure acts as channel of transmission. A receiver is also designed to perform a filtering to recall the original properties as much as it can.

During this stage of simulation, system testing is performed through AWGN fading channel without involving any coding scheme in the channel; it is only limited to modulation/de-modulation approaches on a transmitter/ receiver respectively. On a further stage of simulation, a coding scheme is designed to perform channel coding and results are shown after decoding with many properties.

5.2 Source of signals

A signal is formed using several bits, firstly a (10e5) bits generation randomly (random here stands for bit wised amplitude) takes place. The generated signals are later processed by QPSK modulator to perform the mapping of those signals (bits) to (-1 and +1) figure 5.1. The following diagram is showing the performed process in the first simulation stage (Figure 5.2).

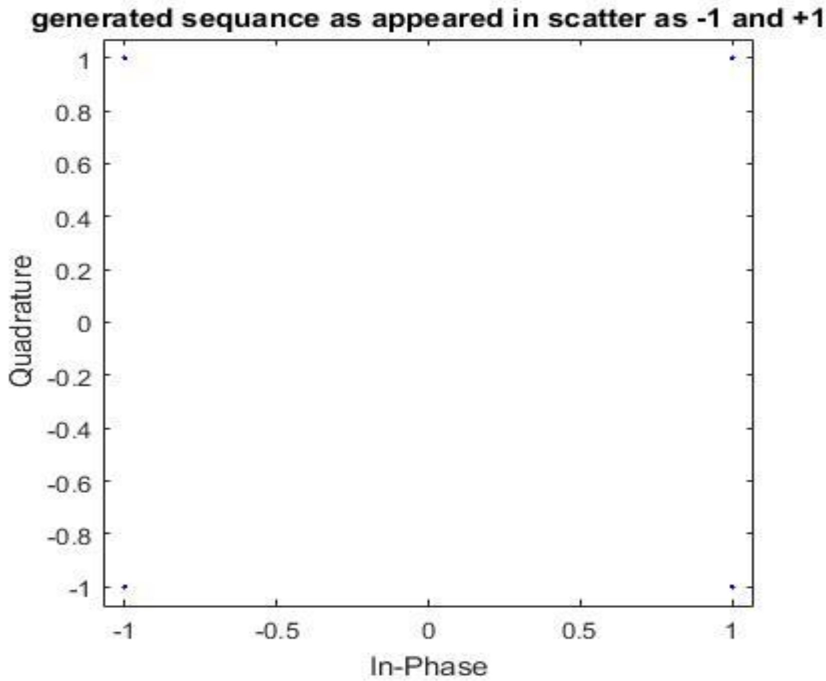

Figure 5.1: QPSK signal at the modulator output.

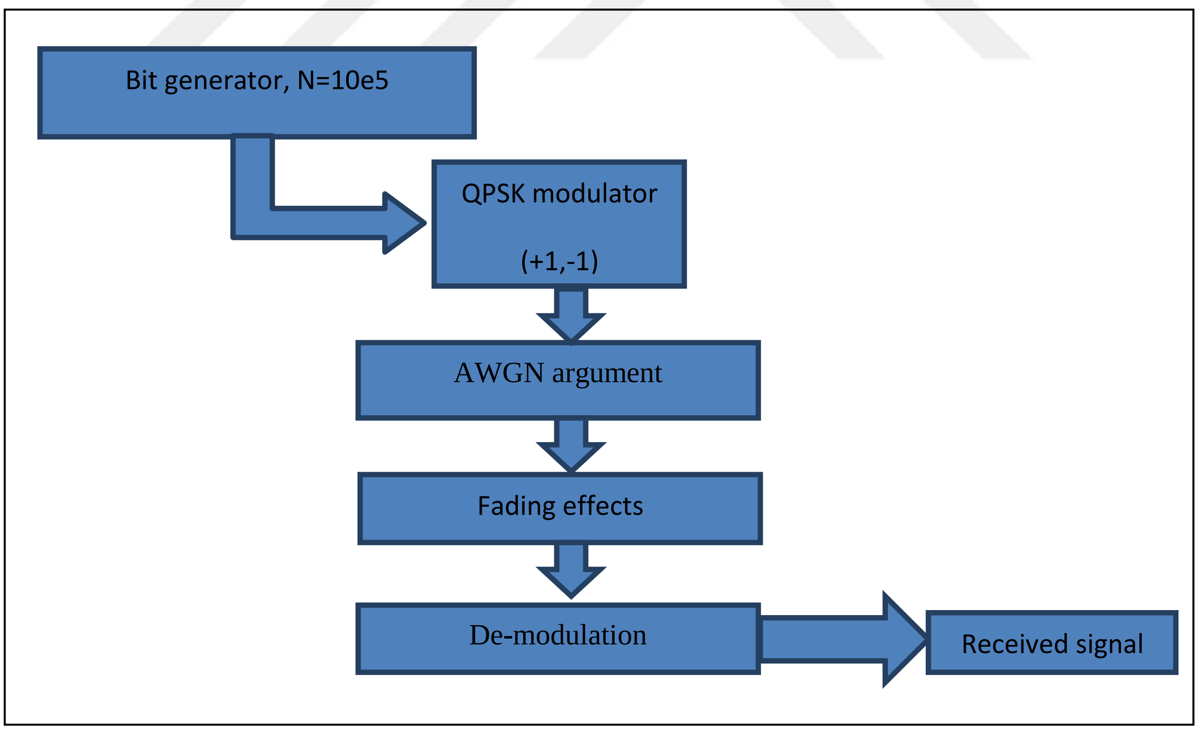

Figure 5.2: system structure.

5.2.1 Impact of noise

Per bit energy to noise power is a significant communication and data transmission parameter.

This depicts SNR, also termed as per bit SNR.

When we compare BER as a standard performance measure, and when we ignore bandwidth, according to the description, Eb represents energy of the signal that is linked with data bits. It equals signal power when it is divided by the users' bit rate. When the unit of signal power is watt and bit rate in bps, the resulting Eb will be in joules (watt-seconds).

N0 represents noise spectral density while noise power for every hertz of bandwidth is calculated in joules (watts/hertz). They are same as Eb, and Eb/N0 has no dimension; therefore, its unit is decibel. Eb/N0 shows a system's power efficiency irrespective of error, modulation type, and bandwidth. It leaves no confusion about what real definition of bandwidth is and what is its applicability.

If the bandwidth of a signal is clear, Eb/N0 equals SNR. In this case, the bits mean "used" bits having nothing to do with error-correction information and modulation type. Eb/N0 should be cautiously applied on limited-interference channels because the additional white noise (having the noise density: N0) exists while the interference may not be noise-like. Some spread-spectrum technologies such as CDMA experience noise-like interference, which is expressed by I0 while thermal noise N0 generates the total ratio Eb/(N0 + I0). Eb/N0 is linked with CNR (carrier- noise ratio), so SNR of a signal before detection and after receiving filter will be:

$$\frac{C}{N} = \frac{Eb}{No} .* \frac{f}{B}$$

Here:

"F" is the channel data rate (net bit rate), and

"B" is the channel bandwidth

The equivalent expression in logarithmic form (dB):

$$CNR(dB) = 10\log\left(\frac{Eb}{No}\right) + 10\log\left(\frac{f}{B}\right)$$

Eb/N0 is considered a usual measure to denote per symbol energy to noise-power spectral density (Es/N0):

$$\frac{Eb}{No} = Es/\rho No$$

ρ

Where Es represents energy/ symbol and its unit is joule while and ρ has nominal efficiency in hertz or bits.

Es/N0 generally applied to analyze digital modulation processes. These two quotients are interlinked as per following:

$$\frac{Es}{No} = \frac{Eb}{No} * log2\,(M)$$

Here, M represents the counting of alternative modulation symbols, such as M = 8 for 8PSK and 4 for QPSK. It is important to note that the currently discussed unit is energy per bit not energy/information bit.

Es/N0 is further express-able as:

$$\frac{Es}{No} = \frac{E}{N} * B/fs$$

Here:

- “C/N” shows CNR (carrier-to-noise ratio) or SNR.
- “B” represents channel bandwidth (Hz).
- “Fs” means symbols/ second.

As the generated bits are modulated with PBSK modulator, resulted signal is transmitted by AWGN and fading channel, so the noise component will be added into signal so the resulted output will be obtained, which is depicted in Figure 5.3.

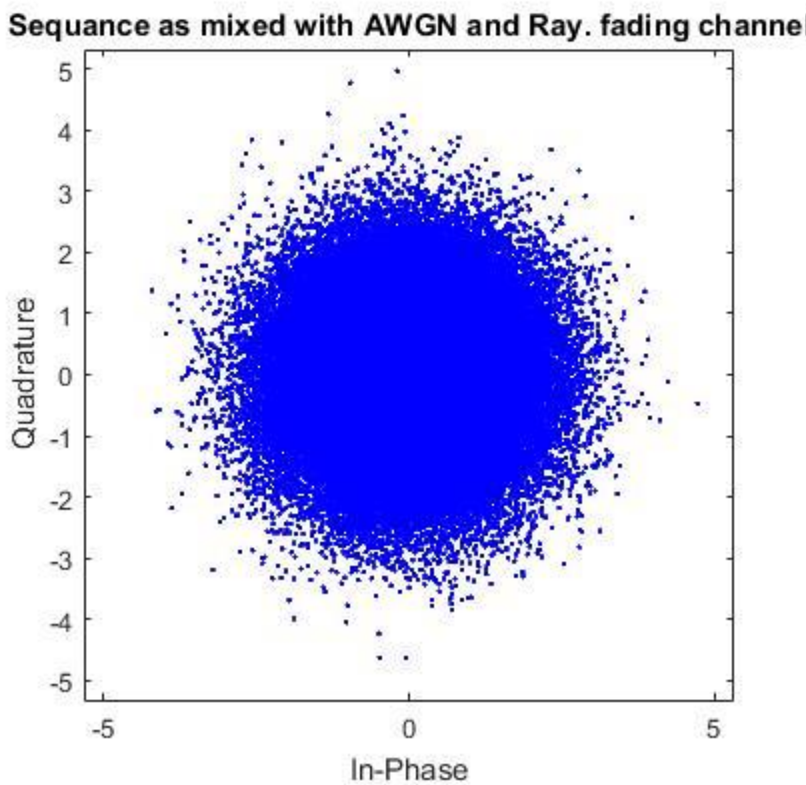

Figure 5.3: received signal having noise effects.

The figure 5.1 is depicting the generated bits that are arranged as -1 and +1, so, Matlab can create a “scatter plot” where all bits with +1 segregate from those having value -1. From plot in figure 5.1, all bits are gathered in two groups (-1,+1) accumulatively. In figure 5.4, noise is added here into a modulated signal, so, the analogy of the figure 5.1 is demolished and bits are scattered around as Figure 5.4 shows.

For recovering original bits, firstly bit-to-noise energy ratio is taken as variable range (1 through 20) so that we will be having twenty different transmitted signals for testing. In other word, signal is transmitted under different energy of bit and we have tried twenty possibilities to examine signal performance over a noisy channel. Later, the signal was received and demodulator recovered originally transmitted bits.

For each EtN value, a different output was calculated and the same is plotted in Figure 5.5.

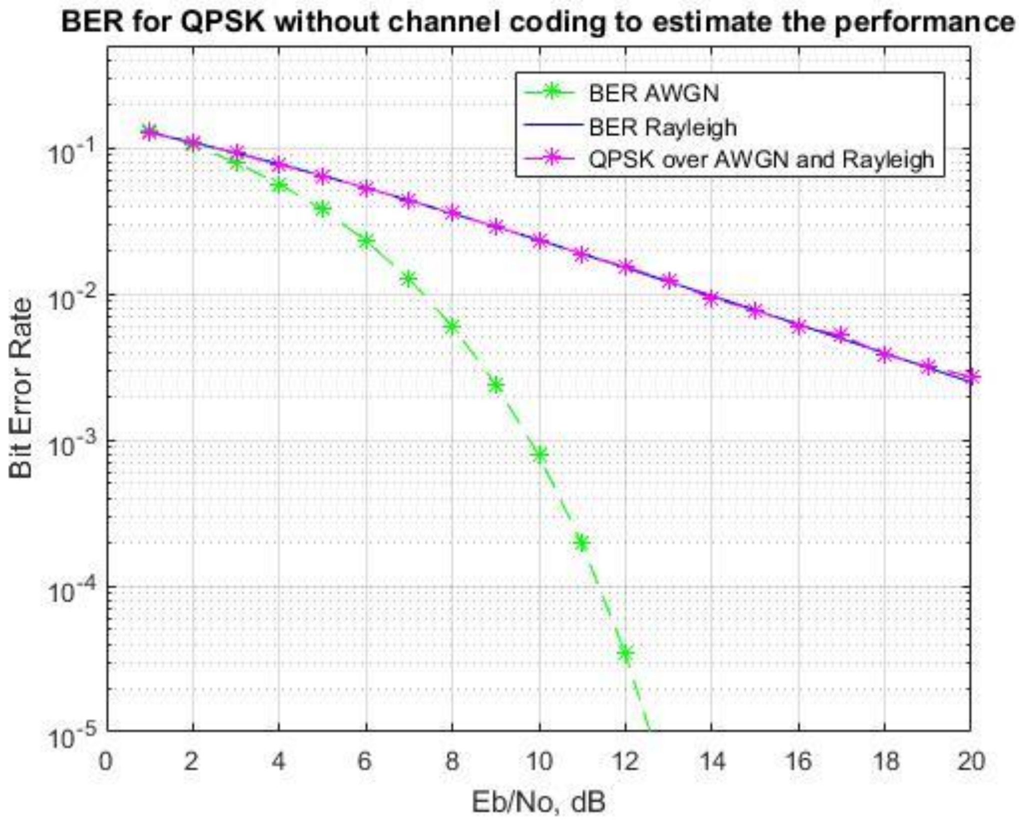

Figure 5.4: QPSK receives signal when noise elimination takes place at 0 to 20 EtN

5.3 Channel coding

In this part of process, a signal was undergoing with complete coding techniques as it was transmitted through a modulator towards the channel.

The transmission procedure takes place as follows:

MIMO: The bits that are generated from modulator are taking shape 0s and 1s because the coding scheme of this project must be supplied with Binary format. However, signal is generated to be serially transmitted through a channel to improve the transmission, which might be broken into parallel arranged groups shown below.

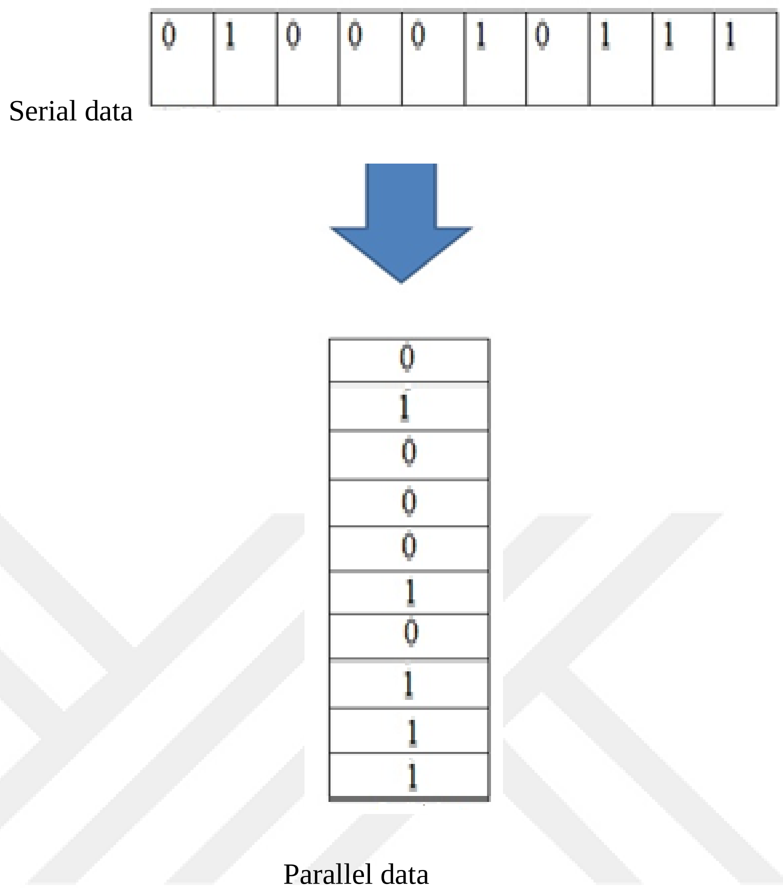

Figure 5.5: Conversion of bits from serial to parallel form before transmission.

- When the signals convert into parallel form, the encoder must be ready to accept the same and generate the code words to start signal transmission.
- In the current project, we used trellis coding for performing the channel coding prior to transmission.
- Encoder involves a generator matrix to generate a code word.

5.4 Trellis encoder

Since there is encoder-enforced limitation, limited transitions with specific phasor sequence are possible. Now, the above mentioned rules will be much clearer with this example. For example, let the correct path be all zero-path, and the shortest distance out of the two paths diverges and then remerges, which can be calculated by finding least square distance/free Euclidian distance.

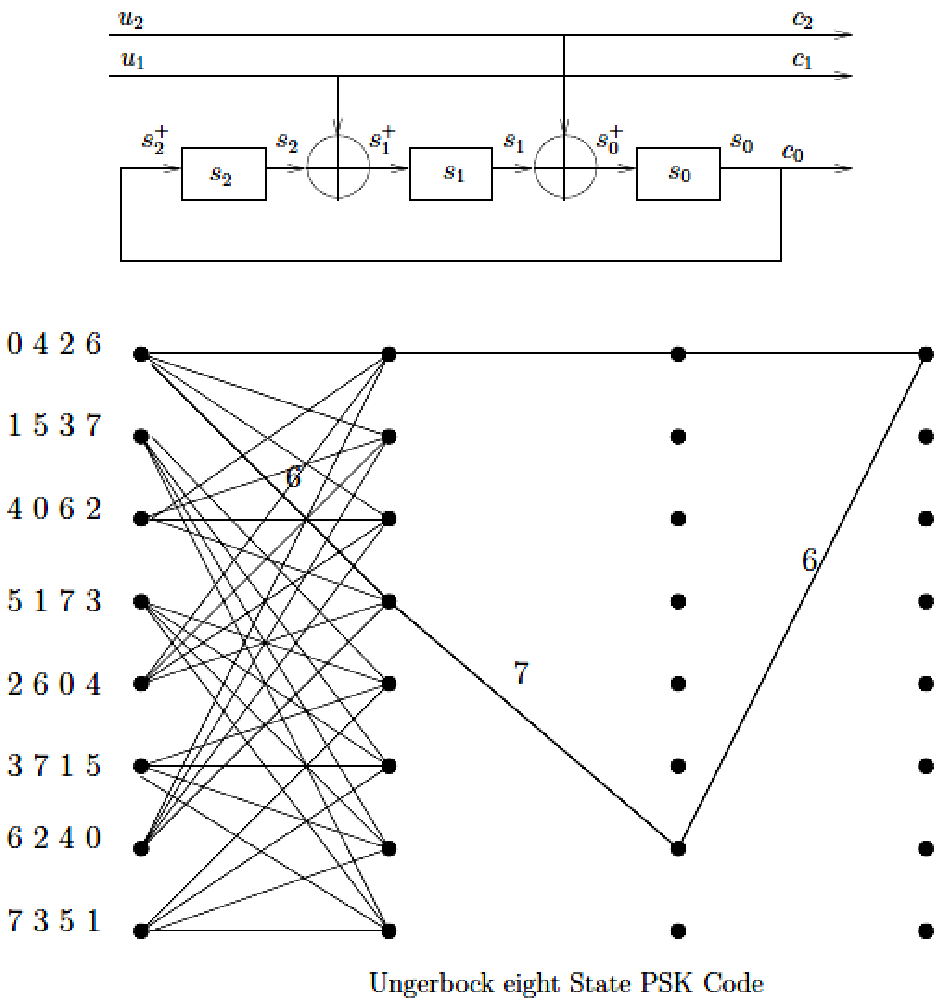

Figure 5.6: Trellis and encoder for the eight-state 8PSK.

5.4.1 Encoder design

This stage is about encoder design that is capable to generate a code word for each input set; however, code may be the encoder output. For our project, we have implemented encoding, which is illustrated in figure 5.8.

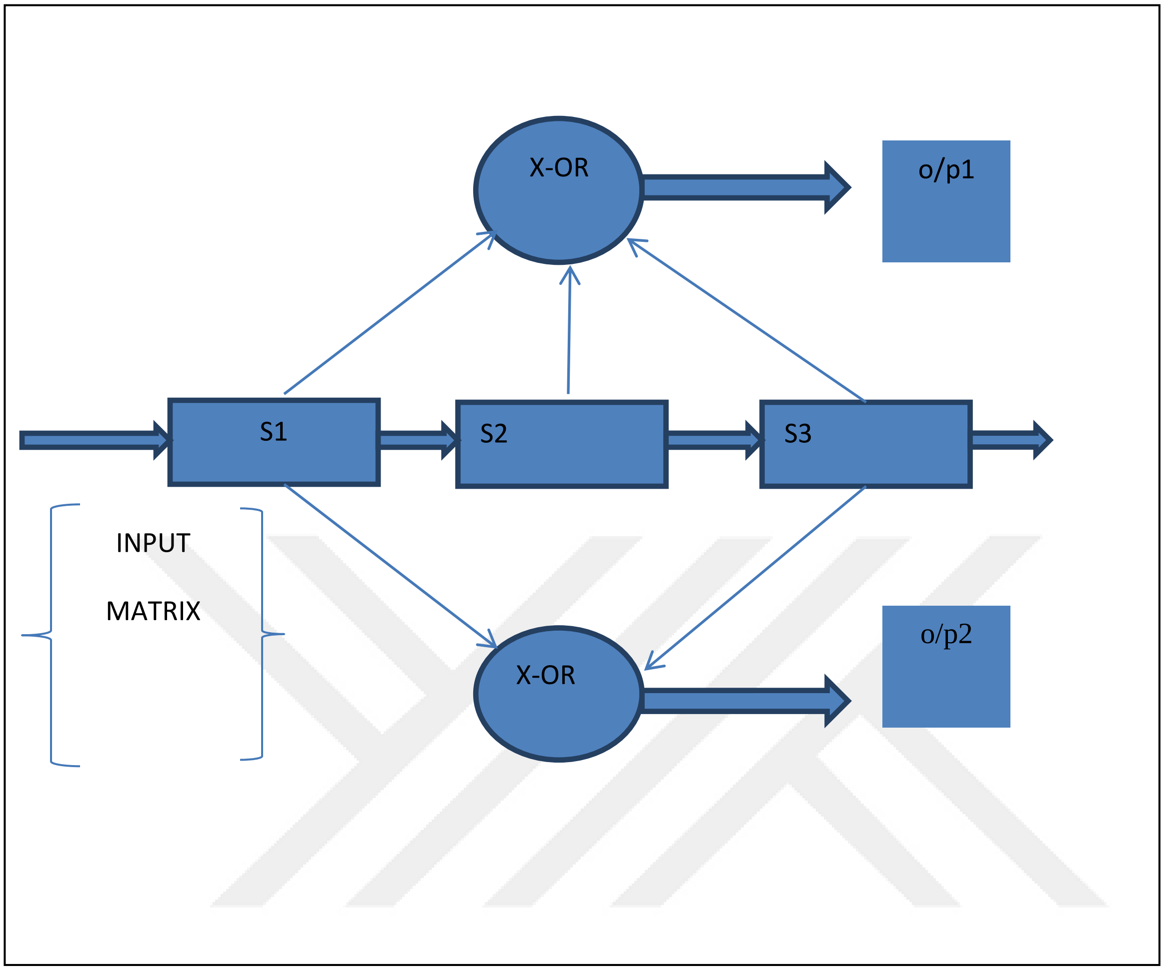

Figure 5.7: convolution encoder.

It is designed by interfacing shift-register as well as XOR gate for calculating the output.

There are two different outputs, which were achieved as illustrated in the figure given above:

O/P1= S1 X_OR S2 X_OR S3

O/P2= S1 X_OR S3

From that, all the code words will be generated here, however as we got two XOR in this design, so each bit will be coded as results in two bits; i.e. when a code word of 10 bits is entered into the encoder, a twenty bit code will be generated. The code structure is shown in trellis diagram as depicted in figure 5.9.

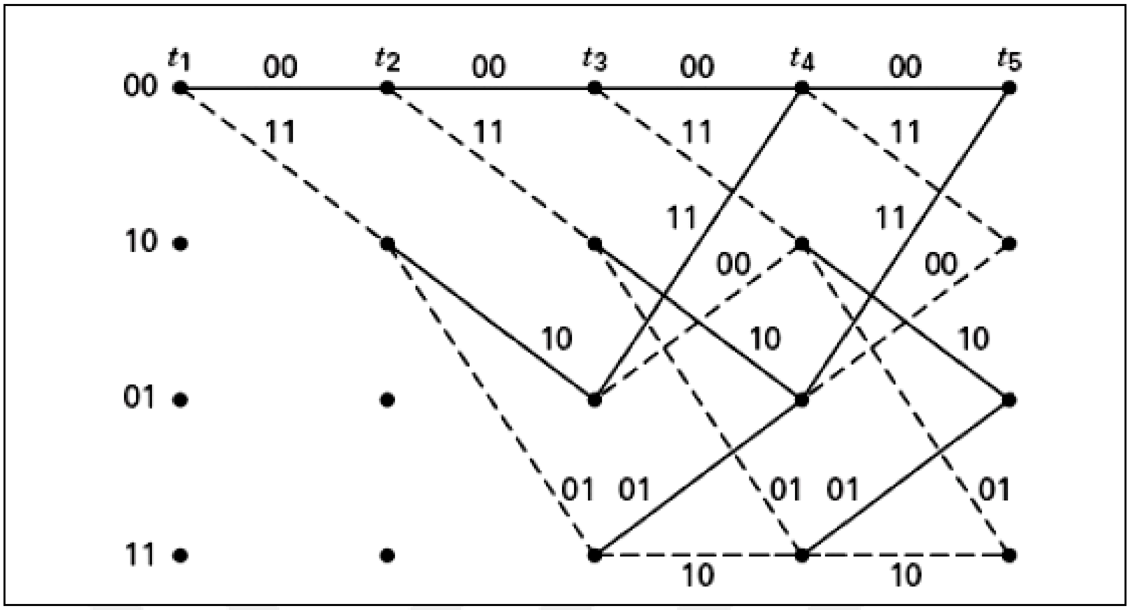

Figure 5.8: Code trellis representation (diagram).

However, encoded bit stream takes a look, which is depicted in figure 5.10:

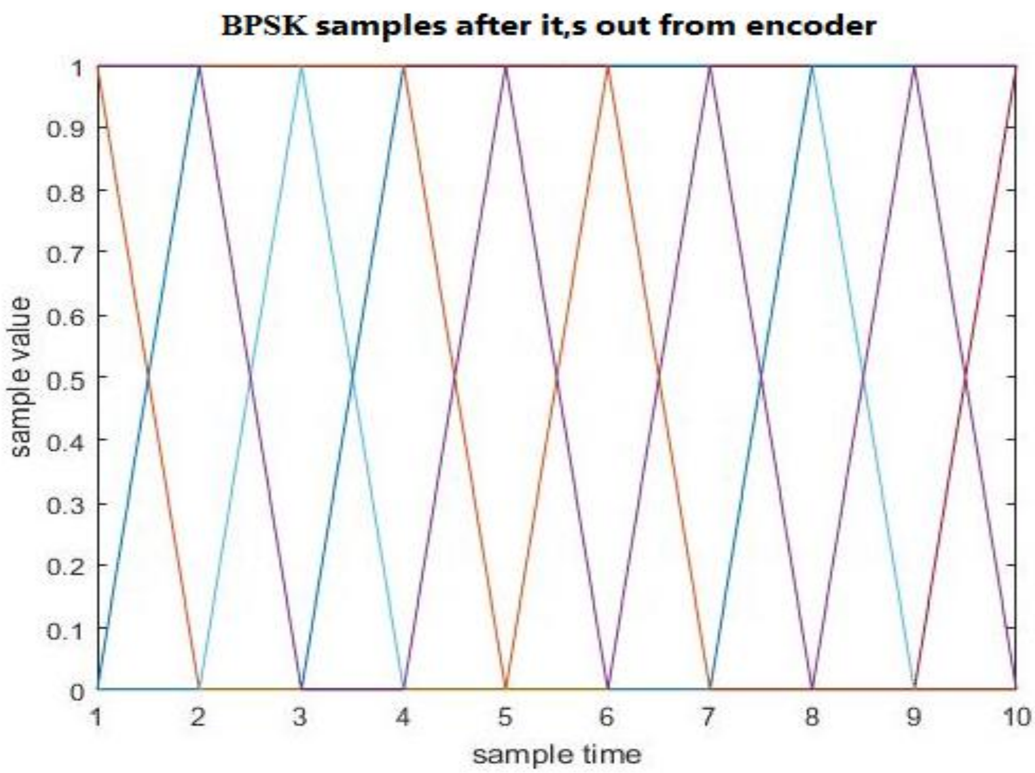

Figure 5.9: encoder output stream.

5.5 Viterbi decoder

This decoder type utilizes Viterbi algorithmic function to decode bits, which require encoding with the help of convolution/trellis codes. Decoding can be accomplished using other algorithms such as Fano algorithm.

The Viterbi algorithm consumes more resources; however, it performs better decoding. Normally, it decodes convolutional codes having constraint length k<=10 but practically, the value closer to k=15 is needed. The coding –decoding process can be shown in Figure 5.11.

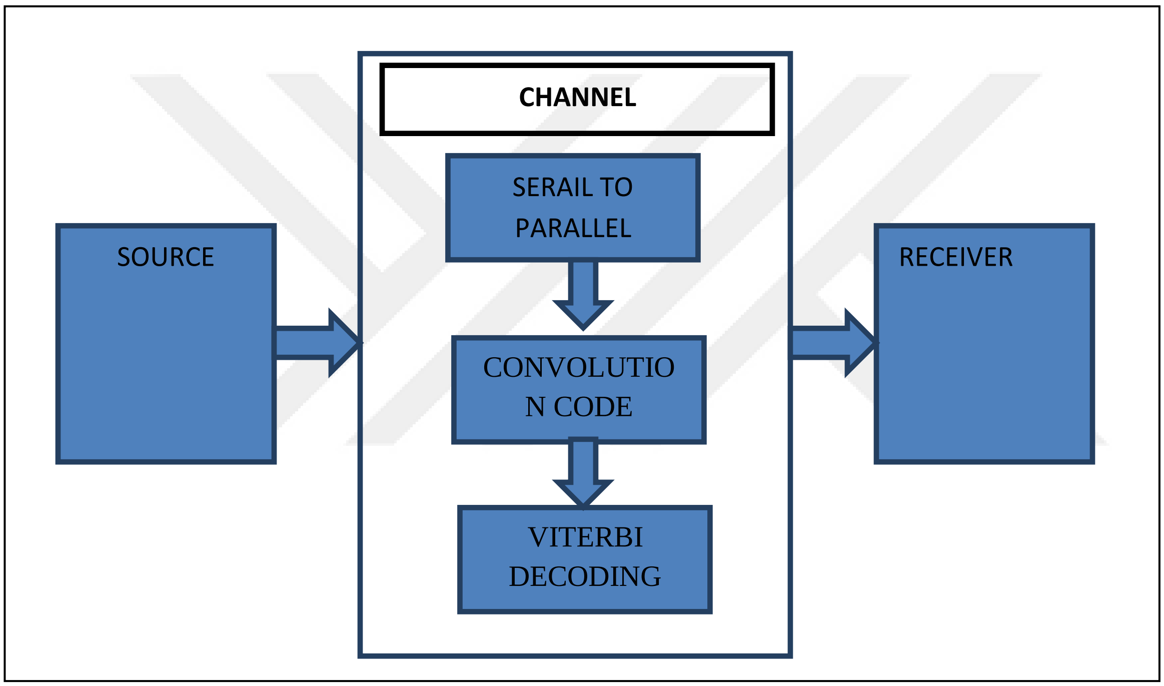

Figure 5.10: Trellis channel coding.

3.6 **Project contribution**

According to the literature survey in chapter II, most of authors have implemented a communication system and tested data transmission under additive white Gaussian noise environments, and things were done at receiver side is variation the signal to noise ratio SNR to improve signal immunity for noise and gain better detection. In our work, we have implemented a wireless communication system where data is sent under AWGN noise and Rayleigh fading, a large number of bits i.e. 10e5 was transmitted under above channel conditions and received with different SNR range, the observation made that large bit power will enhance the receiving signals and improve their noise immunity. Furthermore, a complete design of encoder and decoder was made so that coded modulation is implemented, at meanwhile, concept of packets is also applied into our paradigm so that large stream of bits is breaked into identically length packets, such concept plays essential role to enhance signal immunity to noise and decreases the fading,as well,helps to recover the original signal at receiver when decoder is functioning. In addition, increasing data rates and high quality of service QoS.

However Studies shown that increasing the power of transmitted signal will lead to increasing power consumption , also increasing the number of transmitted antenna and using the concept of multiple input multiple output system will make relatively higher cost as compared with serial input serial output system.

6 Conclusion

The current project performs QPSK modulation of bits in large numbers, and then, the transmission of the resulting positive and negative format took place through AWGN channel when ray-light fading effects come in the image. The system starts receiving the destroyed version of transmitted signal due to noisy channel by varying bit-to-noise energy; the experimental results shown that higher energy will increase the bit noise immunity and eliminate the noise impact on transmission.

We assumed that BtN falls in a range between one decibel to twenty decibels, and we observed that twenty decibels yields optimum performance. Moreover, coding modulation accomplishes where a convolutional encoder is designed to generate a convolutional code for each input code word; the signal bits stream are divided orthogonally for enhancing the signal performance such that if some portion of a signal becomes noise-affected, the other information of that signal will remain safe. The same is implemented by performing serial to parallel transmission.

After doing so, a code is sent to convolutional encoder, and here, the trellis code will be generated for each code word. On the other side, as signal blocks are received on receiver-side, Viterbi decoding recovers the original signal and performs the decoding based on maximum likelihood approach.

7 References

[1]. Ms.Arfiya S.Pathan, 2Mr.Abhay Satmohankar, “A Comparative Study between Multipath Fading Channels”, International Journal of Engineering Development and Research 2015.

[2]. M.S. Chavan1, R.H. Chile2 and S.R. Sawant, “Multipath Fading Channel Modeling and Performance Comparison of Wireless Channel Models”, International Journal of Electronics and Communication Engineering. ISSN 0974-2166 Volume 4, Number 2 (2011).

[3]. Navjot Kaur, Lavish Kansal, “Performance Comparison of MIMO Systems over AWGN and Rician Channels using OSTBC3 with Zero Forcing Receivers”, IJCSI International Journal of Computer Science Issues, Vol. 10, Issue 2, No 2, March 2013.

[4]. Mangesh R. Walke1 , Rekha Mehra, “Performance Analysis of WRAN over Various Channels Using Different Codes and Modulations Techniques”, International Journal of Advance Engineering and Research Development, 2015.

[5]. Tushar Kanti Roy* and Monir Morshed,” Performance Analysis of PDC over Rayleigh Fading Channel for OFDM System”, International Journal of Computing Communication and Networking Research 2016.

[6]. Gagandeep Singh Dhaliwal, Navpreet Kau, “BER based Performance Analysis of MCCDMA over Multipath Channels”, International Journal of Computer Applications (0975 – 8887) Volume 69– No.22, May 2015.

[7]. Prabhpreet Kaur, Kiranpreet Kaur, “Performance Analysis of TWDP Fading System for different Modulations : A Review”, International Journal of Computer Applications (0975 – 8887) 2015.

[8]. Patteti Krishna, Tipparti Anil Kumar and Kalitkar Kishan Rao, “M-QAM BER and SER Analysis of Multipath Fading Channels in Long Term Evolutions (LTE)”, International Journal of Signal Processing, Image Processing and Pattern Recognition Vol.9, No.1 (2016), pp.361-368 http://dx.doi.org/10.14257/ijsip.2016.9.1.34.

[9]. Ahmed Galal Ahmed Mohammed, “Performance Evaluation of BPSK modulation Based Spectrum Sensing over Wireless Fading Channels in Cognitive Radio”, IOSR Journal of Electronics and Communication Engineering (IOSR-JECE) e-ISSN: 2278-2834,p- ISSN: 2278-8735.Volume 9, Issue 6, Ver. IV (Nov - Dec. 2014), PP 24-28.

[10]. Damilare.O Akande* Festus K. Ojo Robert O. Abolade, “Performance of RS and BCH Codes over Correlated Rayleigh Fading Channel using QAM Modulation Technique”, Journal of

Information Engineering and Applications www.iiste.org ISSN 2224-5782 (print) ISSN 2225-0506 (online) Vol.4, No.9, 2014.

[11]. A. MANIKANTA RAGHU1, B. PRABHAKARA RAO, “Performance of Adaptive Bit Interleaved Coded Modulation in Different Fading Environment”, International Journal of Advanced Technology and Innovative Research Volume.07, IssueNo.13, September-2015, Pages: 2620-2625.

[12]. Ramasiva Shankar T.M, 2Aruna Kumari S, “ADAPTIVE BIT INTERLEAVED CODED MODULATION FOR MOBILE RADIO OFDM SYSTEMS BY USING RICIAN FADING CHANNEL”, International Journal of Advanced Technology in Engineering and Science www.ijates.com Volume No.02, Special Issue No. 01, September 2014 ISSN (online): 2348 – 7550.

[13]. C. Padmaja1, Dr.B.L.Malleswari, “Performance analysis of Adaptive Bit-interleaved Coded Modulation in OFDM using Zero Padding Scheme”, IOSR Journal of Mobile Computing & Application (IOSR-JMCA) e-ISSN: 2394-0050, P-ISSN: 2394-0042.Volume 2, Issue 1. (Mar. - Apr. 2015), PP 44-49, www.iosrjournals.org.

[14]. Xiaodong Li, Aik Chindapol, Member, IEEE, and James A. Ritcey, Member, IEEE, “Bit-Interleaved Coded Modulation With Iterative Decoding and 8PSK Signaling”, IEEE TRANSACTIONS ON COMMUNICATIONS, VOL. 50, NO. 8, AUGUST 2002.

[15]. F. Schreckenbach, N. G¨ortz, J. Hagenauer, G. Bauch, “Optimized Symbol Mappings for Bit-Interleaved Coded Modulation with Iterative Decoding”, 2016.

[16]. Bouazza Boubakar, Lego Stephane Y., Garadi Ahmed, Clency Perrine, Rodolphe Vauzelle, “A Novel Constellation Shaping Technique for Bit-Interleaved Coded Modulation”. Springer 2016.

[17]. Lutz H.-J. Lampe1 and Robert F.H. Fischer, “Noncoherent Coded Modulation for Fading Channels”, IEEE TRANSACTIONS ON COMMUNICATIONS, VOL. 44, NO. 18, AUGUST 2015.

[18]. Amir Ingber, Member, IEEE, and Meir Feder, Fellow, IEEE, “Parallel Bit Interleaved Coded Modulation: BICM without Asymptotic Assumptions”, IEEE JOURNAL ON SELECTED AREAS IN COMMUNICATIONS, 2015.

[19]. Menka, “Equalization and Channel Estimation for Cellular MIMO-OFDM”, International Journal of Advanced Research in Computer Science and Software Engineering, Volume 3, Issue 9, September 2013.

[20]. Andreas Ahrensa, Soon Xin Ngb, Volker K¨uhna, Lajos Hanzob, “Modulation-mode assignment for SVD-aided and BICM-assisted spatial division multiplexing”, scinecedirect 2016.

[21]. Jun Tong, Student Member, IEEE, Li Ping, Senior Member, IEEE, and Xiao Ma, “Superposition Coded Modulation With Peak-Power Limitation”, IEEE TRANSACTIONS ON INFORMATION THEORY, VOL. 55, NO. 6, JUNE 2009.

[22]. Albert Guill´en i F`abregas, Alfonso Martinez, “Bit-Interleaved Coded Modulation with Shaping”, International Journal of Advanced Research in Computer Science and Software Engineering, Volume 7, Issue 8, September 2015.

Mathlab Code

```
%%%Performace invistigation Modelation without channel coding%%%
close all;
clear;
clc;
NOB = 10^5;

dir = [];
qua = [];

for k = 1:NOB
             dir = [dir (-1+2*round(rand(1,1)))];
             qua = [qua (-1+2*round(rand(1,1)))];
end
    seq = zeros(2,NOB);
    for k=1:NOB
        seq(1,k)=dir(:,k);
        seq(2,k)=qua(:,k);
    end
    seq;

    fsct=seq';
    scatterplot (fsct);
    title ('generated sequance as appeared in scatter as -1 and +1');

    EtN=[0:20];
    EbN=[];
    for i=1:length (EtN)
        EbN(i)=i;
    end
    EbN;
for k = 1:length(EbN)

    nov= sqrt(1/10^(EbN(k)/10));

   awg=complex(randn(2,NOB),randn(2,NOB));
   rey=complex(randn(2,NOB),randn(2,NOB)); %fading part

   awgn= nov.* awg; %noise AWGN function
   % transmitting of seq into channel
   cseq=awgn+(seq.*rey);

   % reciever end system
   rseq=cseq.*(rey.^(-1));
  % removing the noise parts and trying to recover the original transmitted
  % seq
  rseq;
  le=length (rseq);
  rrseq=real(rseq);
  se=length (rrseq(:,1));
  for kg=1:se
      for j=1:le
      if rrseq(kg,j)< 0
          rrseq(kg,j)=-1;
      else
```

```
          rrseq(kg,j)=1;
      end
      end
  end
  rrseq;
  seq;
  btEr(k)=biterr(seq, rrseq);
end

    ftsct=real(cseq);
    ftsct=ftsct';
    scatterplot (ftsct);
    title ('Sequance as mixed with AWGN and Ray. fading channel');
BER = btEr/(2*NOB);
lnrEbN = 10.^(EbN/10);
BERAWGN = 0.5*erfc(sqrt(lnrEbN/2));
BERREY  = 0.5.*(1-sqrt(lnrEbN./(lnrEbN+1)));
%
%
%%%results
figure (3)
semilogy(EbN,BERAWGN,'g--*');
hold on
semilogy(EbN,BERREY,'b-');
semilogy(EbN,BER,'m--*');
axis([0 20 10^-5 0.5])
%
legend('BER AWGN','BER Rayleigh', 'QPSK over AWGN and Rayleigh');
xlabel('Eb/No, dB');
ylabel('Bit Error Rate');
title('BER for QPSK without channel coding to estimate the performance');
grid on;
%end


%%% This program is to perfomrm of convilutional coding on the signal prior
%%% to transmission of sequance message through AWG noise channel with Ray.
% generate seq for test
% min. message to be coded is of 10 bits ****
% message before coding must be array of (1,n)
% as per convlutional coding PSK modulation will takeplace instead QPSK
clear; clc;
sig=randi([-1 1],1,1e3);
for i=1:length (sig)
    if sig(i)>0
        sig(i)=1;
    else
        sig(i)=0;
    end
end
sig;
%* plotting of PSK sequance
scatterplot (sig);
%     title ('? appeared in scatter as 0 and +1');
        title ('BPSK appeared in scatter as 0 and +1');
%
% check the message sequance Serial to Perallel converssion
lee=length (sig);
```

```
%length of code word
ctr=100;
%
r0=lee/ctr; % number of input code-words
r1=mod(lee,ctr);
if r1==0
    % number of code words
    r0;
    count1=0;
    count2=1;

    mesg=zeros(r0, ctr);
    while (count1 <(ctr*r0) )
           for ii=count1:(count1 + ctr-1)
            %st=mod((ii+1),ctr);
               for jj=count2
                   mesg(jj,mod(ii,ctr)+1)=sig(ii+1);
               end
           end
        count1 =count1 +ctr;
        count2 =(count2)+1;
    end
    mesg;
else
    fprintf('make sure that sequance sample is matching the message length "ctr"');
end
%%%
% convelutional code generator matrix
sle=length (mesg(:,1));
codeword=zeros(r0,(ctr*2));
for i=1:sle
    zrr=mesg(i,:);
    codeword(i,:)=encoder(zrr);
end
  %encoder output print
  codeword ;
  %
  %ploting the codeded sequance
  figure (8)
  plot ((codeword));
    title ('BPSK samples after it,s out from encoder');
    xlabel('sample time');
    ylabel( 'sample value');

  %
  codedeci=zeros(r0,(ctr));
  for i=1:sle
    zrr=mesg(i,:);
    codedeci(i,:)=endeci(zrr);
  end
codedeci;%****


% signal decoding using Veterbi algo.
codff=zeros(r0,(ctr));
  for i=1:sle
    rsyu=codedeci(i,:);
    codff(i,:)=deco(rsyu);
```

```
  end
  %decoder output print
codff;%*****

 sig;
 codeword;
 codff;
```