ANALYSIS OF MIMO-OFDM SYSTEM FOR ESTIMATION OF BER IN AWGN CHANNEL

Submitted in partial fulfillment of the requirements
of the degree of

Bachelor of Engineering

By

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Project Report Approval for B.E

This project report entitled ***ANALYSIS OF MIMO-OFDM SYSTEM FOR ESTIMATION OF BER IN AWGN CHANNEL*** by***DAKHANI JASMINEJEELANI, KAWWAL SANA ISMAIL, PATHAN AIMAN ABDUL BASIT*** is approved for thedegree of ***Bachelor of Engineering***.

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Declaration

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

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Acknowledgement

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1. Abstract

Signal deterioration in a communication channel is mainly due to Multipath Fading. MIMO help for the best quality of signal and OFDM for attaining lower bandwidth. Multiple Input Multiple Output (MIMO) systems use more than one antenna at both ends of the communication link.

Over the past decade, the use of MIMO system has rapidly gained popularity due to its enhanced performance capabilities of improved Reliability, Spatial Diversity Gain and Spatial Multiplexing Gain. Orthogonal Frequency Division Multiplexing (OFDM) is one of the best digital modulation schemes, where signal is divided into number of narrow band signals to obtain spectrum efficiency and minimizing the Inter Symbol Interference (ISI). Thus, combining MIMO and OFDM technologies will improve spectral efficiency, Link reliability, spectral gain and data rate.

The main aim of MIMO-OFDM systems is to combine OFDM technology with the techniques of MIMO systems. The growing demand of multimedia services and the growth of Internet related contents lead to increasing interest to high speed communications. OFDM is a type of multichannel modulation that divides a given channel into many parallel sub channels or subcarriers, so that multiple symbols are sent in parallel.

The proposed MIMO-OFDM system has distinguished advantages over the conventional SISO , SIMO and MISO systems and this is being implemented in 4G cellular and in various other emerging communication technologies.

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2. Introduction

Wireless communication involves the transfer of data without the use of wires. The earliest uses of wireless technology were very limited and data transferred was minimal, hence the available spectrum was sufficient. Communication technology is growing day by day, and system devices are reducing in size as well as being capable of increased processing power. With the growth in wireless communication, customers are demanding more improved and attractive applications, due to which there is a need for improvement in capacity of the available wireless spectrum. Many major technologies have been developed to meet user requirements. 4G/WiMAX technology will provide high data rate and capacity.

Designing the future wireless system with multiple input and output is emerging in a high rate. Multiple Transmit and receive antenna are now widely used to form Multiple Input Multiple Output (MIMO) system used in wireless communications offers various benefits such as higher capacity (bits/s/Hz) through spatial multiplexing scheme and better transmission quality (Bit Error Rate, outage) through transmit diversity scheme(Space Time Block Coding).

The wireless communication devices must have very high spectrum efficiency and the capacity of overcoming the channel fading in the environment of multi-path channel. In wireless telecommunications, multipath is the propagation phenomenon that results in radio signals' reaching the receiving antenna by two or more paths. Causes of multipath include atmospheric ducting, ionosphere reflection and refraction, and reflection from water bodies and terrestrial objects such as mountains and buildings.

The effects of multipath include constructive and destructive interference, and phase shifting of the signal. Since the shape of the signal conveys the information being transmitted, the receiver will make mistakes when demodulating the signal's information. If the delays caused by multipath are great enough, bit errors in the packet will occur. The receiver won't be able to distinguish the symbols and interpret the corresponding bits correctly. This leads to an error in the symbol decoding. It is very difficult to match these requirements using the traditional modulation technique including MIMO & Orthogonal Frequency Division Multiplexing (OFDM); however the hybrid MIMO-OFDM system can meet these requirements.

Hybridization of MIMO-OFDM system is a combination of MIMO and OFDM technologies. MIMO is an antenna technology which uses multiple antennas at both the receiver and transmitter side. OFDM is the one of the best digital modulation techniques which splits the signal into several narrow band channels to obtain spectral efficiency. Some of the features of 4G technologies are supporting multimedia, video streaming, internet and other broadband services.

MIMO systems take advantage of the multiple signals to improve the quality and reliability of the transmitted information signal as the information in wireless channels is mainly affected by multipath fading. Multipath results in the multiple copies of the transmitted information at the receiver with some delays. OFDM uses the spectrum very efficiently by overlapping the sub carriers. It increases the data rate, reduces the ISI (Inter Symbol Interference) and utilizes the spectrum very effectively which is required for transmission of video and other multimedia messages.

a. Multiple Antenna techniques

SISO

Single-Input Single-Output is the classical method in wireless communication and the most common antenna configuration, using one antenna at transmitter and one at the receiver. It is used in radio, TV broadcast and in technology as WiFi, Bluetooth.

SIMO

Single-Input Multiple-Output is the system using one antenna at transmitter and multiple antennas at the receiver. It provides receiver diversity which receive the strongest signal from several transmit antennas. Generally, it is used in Uplink environment.

In Multiple-Input Single-Output two or more number of antennas are used in the transmitter and one antenna at the receiver. It provides transmit diversity because of multiple antenna at a transmitter side. MISO technology has applications in WLAN, MAN and digital television (DTV). Commonly, it is used in downlink scenarios.

MIMO

Multiple-Input Multiple-Output uses multiple antennas at both sides which provides transmit diversity and receiver diversity. It's applicable in every kind of networks like PAN, LAN, WLAN, WAN, MAN. MIMO system can be applied in different ways to receive either a diversity gain, capacity gain or to overcome signal fading.

b.Why MIMO?

The typical aspirations of a system designer are high data rate, low bit error rate, low power consumption, low cost and easy implement ability. The MIMO system ensures us very high data rates even more than 1Gbps while minimizing the bit error rate. By Shanon's theorem the rate of transmission is always less than or equal to the capacity. Practically it is less than the capacity. The capacity depends on the bandwidth of the channel and SNR of the channel. Both the bandwidth and signal to noise ratio are characteristics of the channel.

The SNR can be improved either by reducing noise power or by increasing signal power. Reduction in noise power is not possible while increase in signal power requires more power for transmission which should be avoided for a good design. The improvement of bandwidth is not possible. However there are techniques like OFDM (orthogonal frequency division multiplexing) which assure us efficient use of the channel i.e., spectral efficiency. But however the use of multiple antennas at the transmitter and at the receiver that is use of MIMO meets the ongoing requirements in 4G. The bit error rate in MIMO is very less as compared to conventional SISO systems.

c. OFDM System

It is a digital multi-carrier modulation technique.It converts frequency selective channel into parallel flat sub-channels. It Reduces multipath fading and cope with ISI. Cyclic prefixing is used in this techniqueThe steps OFDM follows:

Modulation → Conversion(S/P) → Insertion

d.MIMO-OFDM

Orthogonal Frequency Division Multiplexing (OFDM) is one of the most promising physical layer technologies for high data rate wireless communications due to its robustness to frequency selective fading, high spectral efficiency, and low computational complexity. OFDM can be used in conjunction with a Multiple-Input Multiple-Output (MIMO) transceiver to increase the diversity gain and/or the system capacity by exploiting spatial domain. Because the OFDM system effectively provides numerous parallel narrowband channels, MIMO-OFDM is considered a key technology in emerging high-data rate systems such as 4G, IEEE 802.16, and IEEE 802.11n.

MIMO communication uses multiple antennas at both the transmitter and receiver to exploit the spatial domain for spatial multiplexing and/or spatial diversity. Spatial multiplexing has been generally used to increase the capacity of a MIMO link by transmitting independent data streams in the same time slot and frequency band simultaneously from each transmit antenna, and differentiating multiple data streams at the receiver using channel information about each propagation path, future standards need to specify both bandwidth requirements and type of signaling that achieves the data rate required for minimal predefined qualities of services for future applications.

Frequency bands used by mobile devices are strictly specified by responsible regulatory bodies, which set limits on the bandwidth available for communication. Therefore, a very natural and important question is what the maximum data rate is (equivalently, information rate) at which reliable communication over a mobile channel of a given bandwidth is attainable. This quantity is known as the channel capacity .for the well-known expression for the maximum data rate that can be achieved, for reliable communication. That is the average bit error rate (BER) can be made arbitrarily close to zero by use of channel coding, for transmissions up to the maximum achievable rate. For mobile channels, that are time-varying and dispersive in time and frequency however the channel capacity derivation is still an open research area. In this context we point out the lack of equivalent vector channel models for realistic continuous-time SISO and MIMO

CONVENTIONAL MIMO-OFDM SYSTEM

The conventional MIMO-OFDM system is shown in figure 1. The proposed system consists of 2 transmit and 2 receive antennas. In transmitter side initially maps the sequence of bits in each spatial stream to constellation points. Then it is encoded in to digital form. A pilot sequence is inserted and used for the channel estimation. Then, a cyclic prefix is inserted in front of the OFDM symbol at the last step of OFDM modulation block. The time length of the cyclic prefix should be greater than the maximum delay spread of the channel. The purpose of adding cyclic prefix is to guard the OFDM symbol against Inter Symbol Interference (ISI),hence, this cyclic prefix is called the guard interval of the OFDM symbols. The MIMO coding can use several encoders such as STBC, STTC coding. In this paper, the conventional MIMO-OFDM system is implemented using STTC with two transmit and two receive antennas

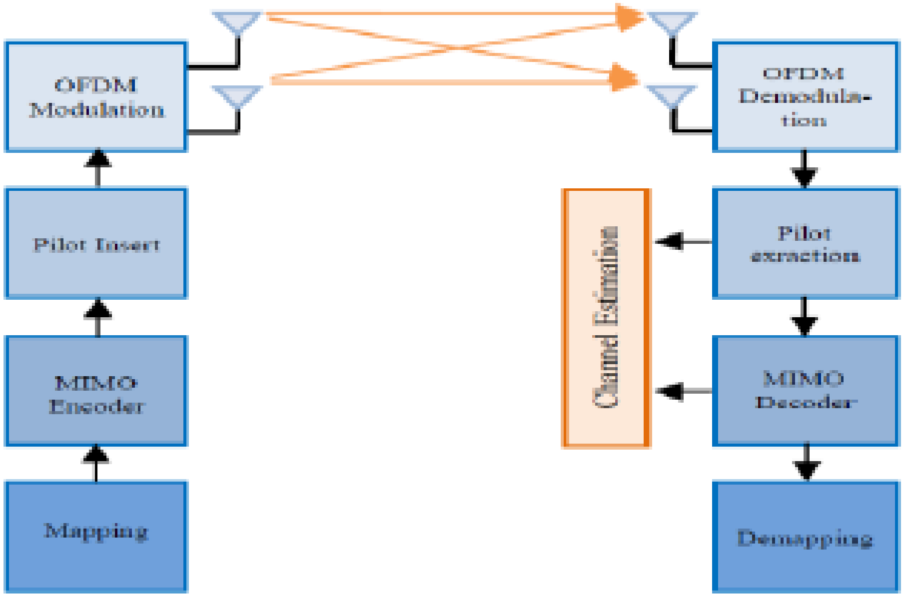
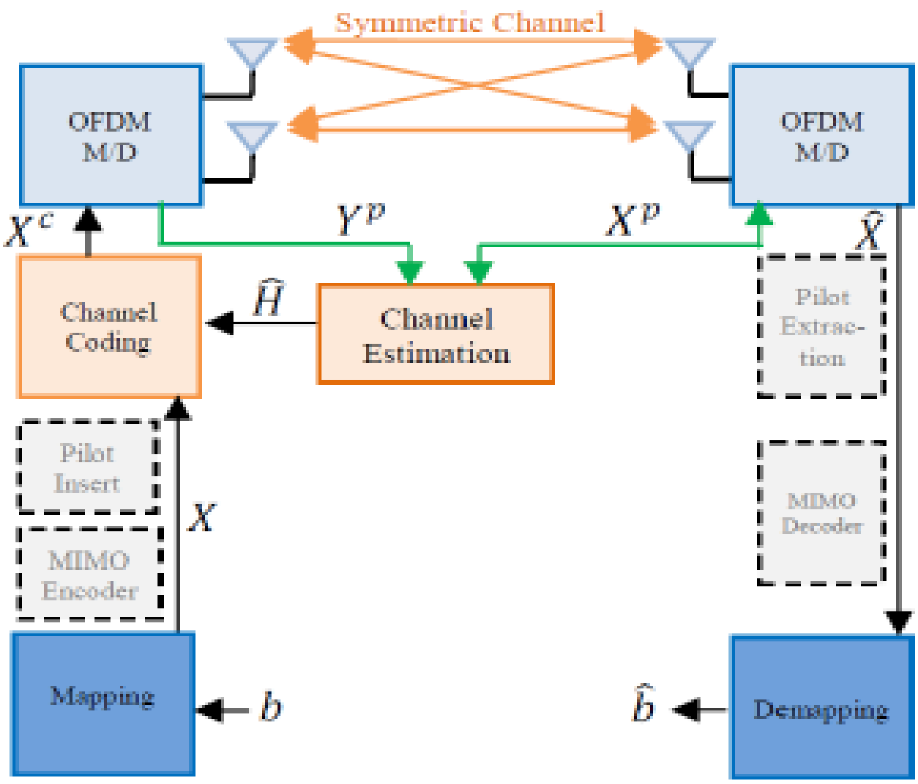


Figure 1.Conventional MIMO-OFDM system model

<u>NEW TRANSMISSION MODEL</u>

The new transmission model is suitable for symmetric channels, such as the transmission between two transmit antennas and two receive antennas. The proposed MIMO-OFDM model is shown in the following figure 2.



In this new MIMO-OFDM model, the channel parameters are estimated from a pilot data transmitted by the receiver side. These estimated parameters are used by a special channel coding block to adapt the transmitter signal to the diverse channel impairments and variations. To reduce the system complexity we have removed the pilot insert, the pilot extraction, the MIMO encoder and the MIMO decoder from the conventional MIMO-OFDM scheme. The channel coding is based on the channel parameters, this channel in our case is between two transmit antennas and two receive antennas, and it can be modeled as shown in the figure 2. First, the receiver send a pilot signal to the transmitter, which can expressed as follows:

$$\begin{cases} Y_1^p = H_{11}.X_1^p + H_{21}.X_2^p + N_1^p \\ Y_2^p = H_{12}.X_1^p + H_{22}.X_2^p + N_2^p \end{cases} \quad (1)$$

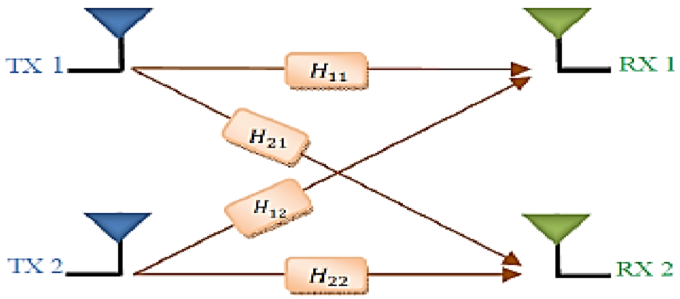


Figure 3. 2 x 2 channel model

Where:

X_1^p and X_2^p are the orthogonal transmitted pilot signals from the transmit antenna TX1 and TX2 respectively.

Y_1^p and Y_2^p are the received pilot signals on the receive antenna RX1 and RX2 respectively.

Y_1^2 and Y_2^2 are the received information at time slot 2 on receive antenna RX1 and RX2 respectively.

H_{ij} is the channel from j^{th} transmit antenna *TXj* to i^{th} receive antenna *RXi* with i and j ε {1,2}.

N_1^p and N_2^p are the noise components on receive antenna RX1 and RX2 respectively.

N_1^2 and N_2^2 are the noise at time slot 2 on the receive antenna RX1 and RX2 respectively.

Let us also define the pilot received signal Y^p the matrix channel H the pilot transmitted signal X^p and the noise vector N^p as follows respectively.

$$Y^p = \begin{bmatrix} Y_1^p \\ Y_2^p \end{bmatrix} \qquad \text{H} = \begin{bmatrix} H_{11} & H_{12} \\ H_{21} & H_{22} \end{bmatrix}$$

$$X^p = \begin{bmatrix} X_1^p \\ X_2^p \end{bmatrix} \text{ and } N^p = \begin{bmatrix} N_1^p \\ N_2^p \end{bmatrix}$$

By using the above notations, equation (1) can be rewritten as

$$Y^p = H'.X^p + N^p \qquad (2)$$

Using the transmitted pilot signal X^p and the received pilot signal Y^p, the channel parameters are estimated as following

$$\hat{H}_{11} = \left(Y_1^p.X_1^p\right) / \left(X_1^p\right)^2$$

$$= \left(H_{11}.X_1^p.X_1^p + H_{21}.X_2^p.X_1^p\right) + N_1^p.X_1^p / \left(X_1^p\right)^2$$

$$= H_{11} + \frac{N_1^p}{X_1^p} \qquad (3)$$

The term $X_2^p.X_1^p = 0$ because the pilots X_2^p and X_1^p are chosen to be orthogonal signals.

In addition, if the pilot signal power is $\left\|X_1^p\right\|^2 >> 1$ then

$$\hat{H}_{11} \approx H_{11} \qquad (4)$$

Similarly, all channel parameters H_{ij} can be easily deducted

$$\hat{H}_{21} = \left(Y_1^p.X_2^p\right) / \left(X_2^p\right)^2$$

$$= \left(H_{11}.X_1^p.X_2^p + H_{21}.X_2^p.X_2^p\right) + N_2^p.X_2^p / \left(X_2^p\right)^2$$

$$= H_{21} + \frac{N_2^p}{X_2^p}$$

$$\hat{H}_{21} \approx H_{21}, \qquad \left(\text{if } \left\|X_2^p\right\|^2 >> 1\right) \qquad (5)$$

$$\hat{H}_{12} = \left(Y_2^p.X_1^p\right) / \left(X_1^p\right)^2$$

$$= \left(H_{12}.X_1^p.X_2^p + H_{22}.X_2^p.X_1^p\right) + N_2^p.X_1^p / \left(X_2^p\right)^2$$

$$= H_{12} + \frac{N_1^p}{X_1^p}$$

$$\hat{H}_{12} \approx H_{12} \qquad \left(\text{if } \left\|X_1^p\right\|^2 >> 1\right) \qquad (6)$$

$$\hat{H}_{22} = \left(Y_2^p . X_2^p\right) / \left(X_2^p\right)^2$$
$$= \left(H_{12}.X_1^p.X_2^p + H_{22}.X_2^p.X_2^p\right) + N_2^p.X_2^p / \left(X_2^p\right)^2$$
$$= \mathrm{H}_{22} + \frac{\mathrm{N}_2^p}{X_2^p}$$
$$\hat{\mathrm{H}}_{22} \approx \mathrm{H}_{22} \qquad \left(if \ \left\|X_2^p\right\|^2 >> 1\right) \qquad (7)$$

By combining results obtained from equations (3), (5), (6) and (7), a more compact expression can be easily written

$$\begin{bmatrix} \hat{H}_{11} & \hat{H}_{21} \\ \hat{H}_{12} & \hat{H}_{22} \end{bmatrix} = \left(\begin{bmatrix} Y_1^p \\ Y_2^p \end{bmatrix} . [X_1^p \ X_2^p]\right) \begin{bmatrix} 1/\left(X_1^p\right)^2 & 0 \\ 0 & 1/\left(X_2^p\right)^2 \end{bmatrix} \qquad (8)$$

$$\begin{bmatrix} \hat{H}_{11} & \hat{H}_{21} \\ \hat{H}_{12} & \hat{H}_{22} \end{bmatrix} = \hat{H} = \begin{bmatrix} H_{11} + \dfrac{N_1^p}{X_1^p} & H_{21} + \dfrac{N_2^p}{X_2^p} \\ H_{12} + \dfrac{N_1^p}{X_1^p} & H_{22} + \dfrac{N_2^p}{X_2^p} \end{bmatrix} \qquad (9)$$

If $\left(\left\|X_1^p\right\|^2 and \left\|X_2^p\right\|^2 >> 1\right)$ then

$$\hat{\mathrm{H}} \approx \mathrm{H} \qquad (10)$$

Moreover, equation (9) can be further simplified and rewritten as follows.

$$\hat{\mathrm{H}} = \left((\mathrm{Y}^{\mathrm{P}}.(X^{\mathrm{P}})).A\right) \qquad (11)$$

Where,

$$A = \begin{bmatrix} 1/\left(X_1^p\right)^2 & 0 \\ 0 & 1/\left(X_2^p\right)^2 \end{bmatrix} \qquad (12)$$

Finally, the channel can be easily estimated using the following expression

$$\hat{\mathrm{H}} = A'. \left(Y^p.(X^p)'\right)' \qquad (13)$$
$$\Leftrightarrow \hat{\mathrm{H}} = A'. \left(X^p.(Y^p)'\right) \qquad (14)$$
$$\Leftrightarrow \hat{\mathrm{H}} = A. \left(X^p.(Y^p)'\right) \qquad (15)$$

$$X^c = \hat{H}^{-1}.\ \mathrm{X} \approx H^{-1}.X \qquad (16)$$

The received signal of the second time slot is given by the following equation (17)

$$\begin{cases} Y_1^2 = H_{11}.X_1^c + H_{12}.X_2^c + N_1^2 \\ Y_2^2 = H_{22}.X_1^c + H_{22}.X_2^c + N_2^2 \end{cases} \qquad (17)$$

$$\Leftrightarrow \begin{bmatrix} Y_1^2 \\ Y_2^2 \end{bmatrix} = \begin{bmatrix} H_{11} & H_{12} \\ H_{21} & H_{22} \end{bmatrix} . \begin{bmatrix} X_1^c \\ X_2^c \end{bmatrix} + \begin{bmatrix} N_1^2 \\ N_2^2 \end{bmatrix} \qquad (18)$$

$$\Leftrightarrow \mathrm{Y} = H.X^c + \mathrm{N} \qquad (19)$$

The advantage of this channel coding is that there is no need to perform the channel estimation and MIMO encoding at the receiver, because going through the channel the received signal becomes;

$$\mathrm{Y} = H.X^c + \mathrm{N} = \mathrm{H}.\hat{\mathrm{H}}^{-1}.X + N$$

$$\Leftrightarrow \mathrm{Y} = \hat{X} \approx X + \mathrm{N} \qquad (20)$$

So we can directly demodulate the received signal to find the estimation of the original transmitted symbol .

3. Literature survey

a. Transmission scheme for MIMO-OFDM system

This contribution introduces a new transmission scheme for multiple-input multiple-output (MIMO) orthogonal frequency division multiplexing (OFDM) systems. The new scheme is efficient and suitable especially for symmetric channels such as the link between two base stations or between two antennas on radio beam transmission. The principle is based on the estimation of channel parameters of a pilot data send by the receiver to the transmitter. Then, the transmitter codes the transmitted signal using the estimated channel parameters to adapt the signal to the channel variations. Conducted Monte-Carlo simulation results show that the proposed scheme has better performance, in terms of bandwidth efficiency and complexity, compared to the conventional MIMO-OFDM scheme methods in the case of a symmetric channel.

b. MIMO-OFDM wireless systems: basics, perspectives, and challenges

Multiple-input multiple-output (MIMO) wireless technology in combination with orthogonal frequency division multiplexing (MIMOOFDM) is an attractive air-interface solution for next-generation wireless local area networks (WLANs), wireless metropolitan area networks (WMANs), and fourth-generation mobile cellular wireless systems. This article provides an overview of the basics of MIMO-OFDM technology and focuses on space-frequency signaling, receiver design, multiuser systems, and hardware implementation aspects. We conclude with a discussion of relevant open areas for further research.

c. Evaluation of BER for AWGN, rayleigh and rician fading channels under various modulation schemes

Several transmission modes are defined in IEEE 802.11 a/b/g WLAN standards. A very few transmission modes are considering for IEEE 802.11 a/b/g in physical layer parameters and wireless channel characteristics. In this paper, we evaluated the performance of available transmission modes in IEEE 802.11b. However, the performance analysis can be done Straight forward using the evaluation of IEEE 802.11b. The performance of transmission modes are evaluated by calculating the probability of Bit Error Rate (BER) versus the Signal Noise Ratio (SNR) under the frequently used three wireless channel models (AWGN, Rayleigh and Rician). We consider the data modulation and data rate to analyze the performance that is BER vs. SNR. We also consider multipath received signals. The simulation results had shown the performance of transmission modes under different channel models and the number of antennas. Based on simulation results, we observed that some transmission modes are not efficient in IEEE 802.11b. The evaluation of performance confirms the increase in the coverage area of the physical layer in the 802.11b WLAN devices.

d. BER comparison ofrayleigh fading channel with Almouti space time block coding and the method of MRC

This paper proposes a technique which uses chaotic communication system combined with Rayleigh fading channel, for secure communications and to improve the system performance by mitigating interference. For secure communications, chaotic sequences are used. Many chaotic communication systems have been proposed, but most of them show a poor performance under realistic channel conditions (i.e.

noise and multipath fading).This paper proposes a wireless communication structure based on two coupled chaotic systems. In order to enhance the errorrate performance of MIMO-OFDM system, Rayleigh fading channel is used. Evaluation and comparison of the performances of MIMO- OFDM system in the AWGN (Additive White Gaussian Noise) channel, Rayleigh fading channel and the MRC diversity methods are provided.Results are verified and analyzed for two cases, one when we used 1*2 and 2*1 diversity scheme for Rayleigh fading channel with Almouti scheme is used in the proposed system and second when diversity method of MRC is used in the proposed system. Computer simulations are done to verify the performance of the proposed approach. A simulation tool with a Graphical User Interface (GUI) which implements these algorithms is also developed to provide ease in the execution.

4. Problem statement

The available spectrum is fixed but the demand for high data rate and high reliability is growing day by day to suite the more improved and attractive applications. In this project, we implement and optimize the 2x2 MIMO-OFDM transmitter and receiver for different embedded platform. The idea behind developing the MIMO-OFDM transmitter is to get the advantage of both MIMO and the OFDM technology in achieving higher data rate, spectral efficiency and reliability. To reduce the bit error rate of MIMO-OFDM from 10^-2 to 10^-5 using 2x2 diversity scheme.

5. Methodology

Wireless communication using multiple-input multiple-output (MIMO) system enables increased spectral efficiency for a given total transmit power. The capacity is increased by introducing additional spatial channels that are exploited by using space-time coding. Multiple antennas are used at both the source (transmitter) and the destination (receiver) side. The antennas at each end of the communications system are combined to minimize errors and optimize data speed.

6. System requirements

The following are the system requirement specifications arrived for different subsystems of the system:

• MIMO system typically consists of m transmitting antennas and n receiving antennas.

• In this system channel is assumed to be a static channel and that the channel is known perfectly at the receiver for all the systems being developed.

• OFDM system generally consists of different sub blocks namely FFT block, Cyclic prefix and parallel to serial conversion.

• Both spatial diversity and spatial multiplexing techniques should be supported .Channel is assumed to be a static channel and is known perfectly at the receiver for all the systems being developed.

7. Software

MATLAB

MATLAB is a high-performance language for technical computing. It integrates computation,visualization, and programming in an easy-to-use environment where problems and solutions areexpressed in familiar mathematical notation. Typical uses include:

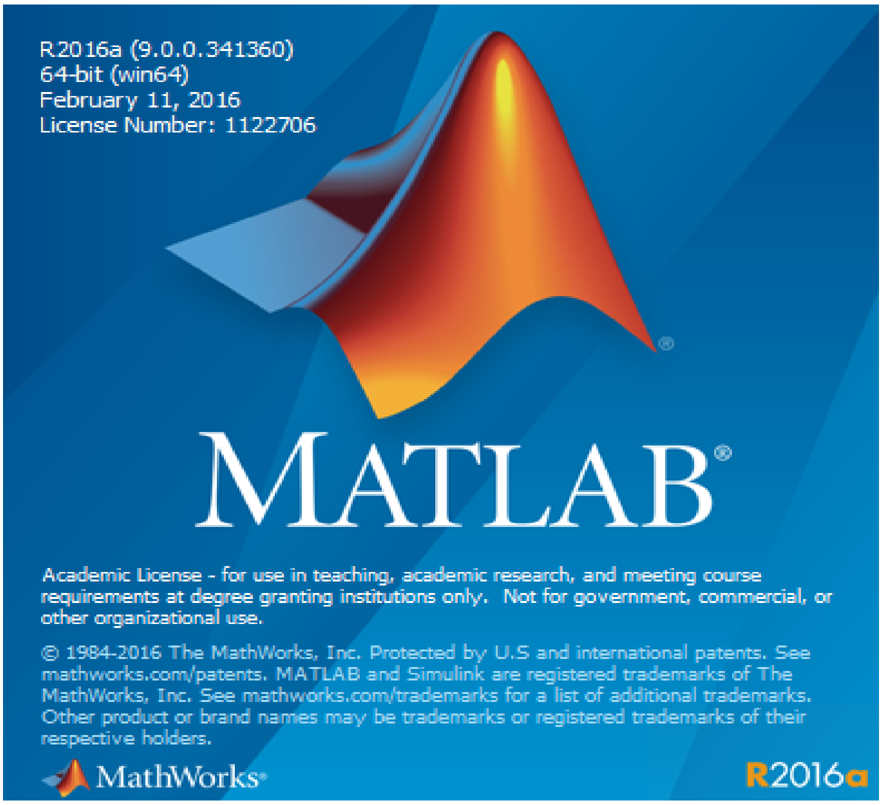
- Math and computation
- Algorithm development
- Modeling, simulation, and prototyping
- Data analysis, exploration, and visualization
- Scientific and engineering graphics
- Application development, including Graphical User Interface building
- MATLAB is an interactive system whose basic data element is an array that does not requiredimensioning. This allows you to solve many technical computing problems, especially thosewith matrix and vector formulations, in a fraction of the time it would take to write a program ina scalar non-interactive language such as C or Fortran.

The name MATLAB stands for matrix laboratory. MATLAB was originally written to provideeasy access to matrix software developed by the LINPACK and EISPACK projects, whichtogether represent the state-of-the-art in software for matrix computation.

MATLAB has evolved over a period of years with input from many users. In universityenvironments, it is the standard instructional tool for introductory and advanced courses inmathematics, engineering, and science. In industry, MATLAB is the tool of choice for highproductivity research, development, and analysis.

MATLAB features a family of application-specific solutions called toolboxes. Very important tomost users of MATLAB, toolboxes allow you to learn and apply specialized technology.Toolboxes arecomprehensive collections of MATLAB functions (M-files) that extend theMATLAB environment to solve particular classes of problems. Areas in which toolboxes areavailable include signal processing, control systems, neural networks, fuzzy logic, wavelets,simulation, and many others.

The MATLAB System



The MATLAB system consists of five main parts:
The MATLAB language.

This is a high-level matrix/array language with control flow statements, functions, datastructures, input/output, and object-oriented programming features. It allows both "programmingin the small" to rapidly create quick and dirty throw-away programs, and "programming in thelarge" to create complete large and complex application programs.
The MATLAB working environment.

This is the set of tools and facilities that you work with as the MATLAB user or programmer. Itincludes facilities for managing the variables in your workspace and importing and exportingdata. It also includes tools for developing, managing, debugging, and profiling M-files,MATLAB's applications.

Handle Graphics.

This is the MATLAB graphics system. It includes high-level commands for two-dimensional andthree-dimensional data visualization, image processing, animation, and presentation graphics. Italso includes low-level commands that allow you to fully customize the appearance of graphicsas well as to build complete Graphical User Interfaces on your MATLAB applications.

The MATLAB mathematical function library:

This is a vast collection of computational algorithms ranging from elementary functions likesum, sine, cosine, and complex arithmetic, to more sophisticated functions like matrix inverse,matrix eigenvalues, Bessel functions, and fast Fourier transforms.

The MATLAB Application Program Interface (API):

This is a library that allows you to write C and Fortran programs that interact with MATLAB. It
include facilities for calling routines from MATLAB (dynamic linking), calling MATLAB as acomputational engine, and for reading and writing MAT-files.

APPLICATIONS:

- Communications Systems
- Computational Biology
- Computational Finance
- Control Systems
- Digital Signal Processing
- Embedded Systems
- FPGA Design and Co-design
- Image and Video Processing

8. Experimental specifications

PARAMETERS	SPECIFICATIONS
CHANNEL MODEL	AWGN CHANNEL
MODULATION	BPSK
DIVERSITY SCHEME	2x2 ANTENNAS
SEPARATION DISTANCE	½
ANTENNAS TRANSMITTING POWER	EQUALLY
FFT SIZE	512
NUMBER OF DATA SUBCARRIERS	324
NUMBER OF PILOT SUBCARRIERS	60
NUMBER OF GUARD BAND SUB CARRIERS	92
CYCLIC PREFIX OR GUARD TIME	1/8

9. BER v/s SNR

In digital transmission, the number of bit errors is the number of received bits of a data stream over a communication channel that has been altered due to noise, interference, distortion or bit synchronization errors.

The bit error rate or bit error ratio (BER) is the number of bit error divided by the total number of transferred bits during a studied time interval. BER is a unit less performance measure, often expressed as a percentage. The bit error probability pe is the expectation value of the BER. The BER can be considered as an approximate estimate of the bit error probability. This estimate is accurate for a long time interval and a high number of biterrors. Measuring the bit error rate helps people choose the appropriate forward error correction codes. Since most such codes only bit-flips, but not bit – insertions or bit detection, the hamming distance metric is the appropriate way to measure the number of bit errors. The BER may beimproved by choosing strong signal strength by choosing a slow and robust modulation scheme or line coding scheme, and by applying channel coding schemes such as redundant forward error correction codes.

Binary symmetric channel which is used in analysis of decoding error probability in case of non bursty bit errors on Additive white Gaussian noise (AWGN) channel without fading. A worst case scenario is a completely random channel, where noise totally dominates over the useful signal. In a noisy channel, the BER is often expressed as a function of the normalized carrier-to-noise ratio measured denoted Eb/N0 that is energy permit to noise power spectral density ratio, or Es/N0 that is energy per modulation symbol to noise spectral density.

As the name implies, a bit error rate is defined as the rate at which errors occur in a transmission system. This can be directly translated into the number of errors that occur in a string of a stated number of bits. The definition of bit error rate can be translated into a simple formula:

"BER = number of errors / total number of bits sent"

If the medium between the transmitter and receiver is good and the signal to noise ratio is high, then the bit error rate will be very small - possibly insignificant and having no noticeable effect on the overall system However if noise can be detected, then there is chance that the bit error rate will need to be considered. The main reasons for the degradation of a data channel and the corresponding bit error rate, BER is noise and changes to the propagation path (where radio signal paths are used). Both effects have a random element to them, the noise following a Gaussian probability function while the propagation model follows a Rayleigh model. This means that analysis of the channel characteristics are normally undertaken using statistical
analysis techniques.

Signal to noise ratios and Eb/No figures are parameters that are more associated with radio links and radio communications systems. In terms of this, the bit error rate, BER, can also be defined in terms of the probability of error or POE. The determine this, three other variables are used. They are the error function (erf), the energy in one bit, Eb, and the noise power spectral density (which is the noise power in a 1 Hz bandwidth).

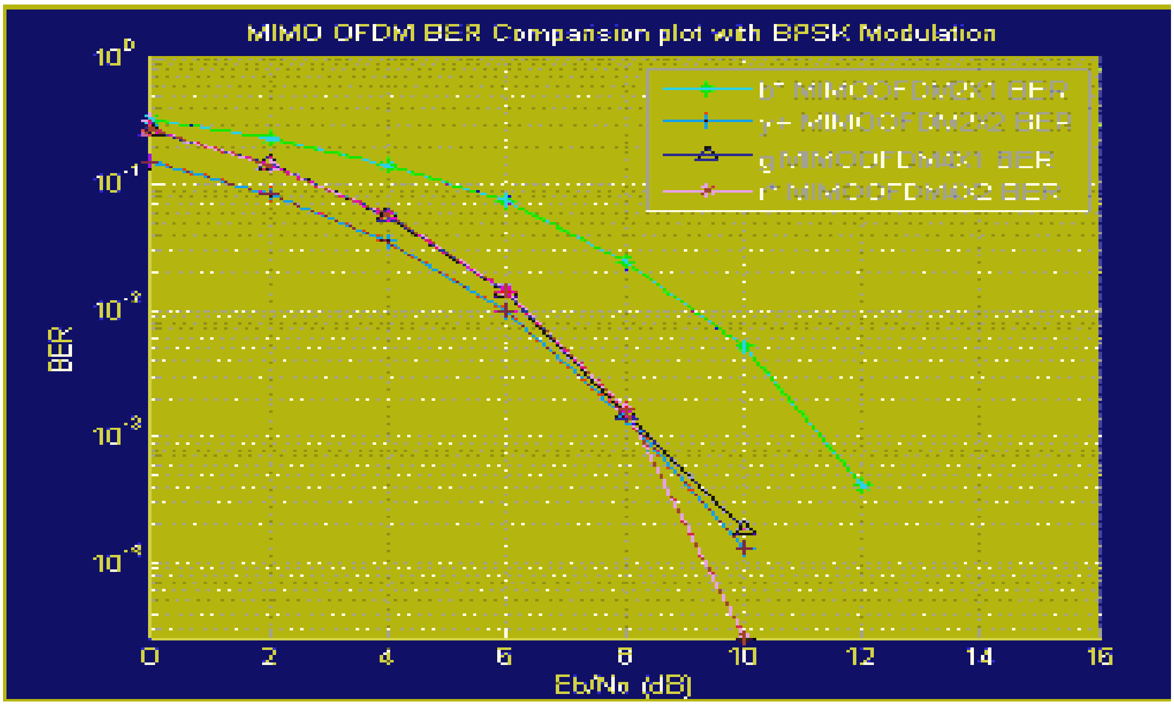
It should be noted that each different type of modulation has its own value for the error function. This is because each type of modulation performs differently in the presence of noise. In particular, higher order modulation schemes (e.g. 64QAM, etc) that are able to carry higher data rates are not as robust in the presence of noise. Lower order modulation formats (e.g. BPSK, QPSK, etc.) offer lower data rates but are more robust. The energy per bit, Eb, can be determined by dividing the carrier power by the bit rate and is a measure of energy with the dimensions of Joules. No is a power per Hertz and therefore this has the dimensions of power (joules per second) divided by seconds). Looking at the dimensions of the ratio Eb/No all the dimensions cancel out to give a dimensionless ratio. It is important to note that POE is proportional to Eb/Noand is a form of signal to noise ratio.

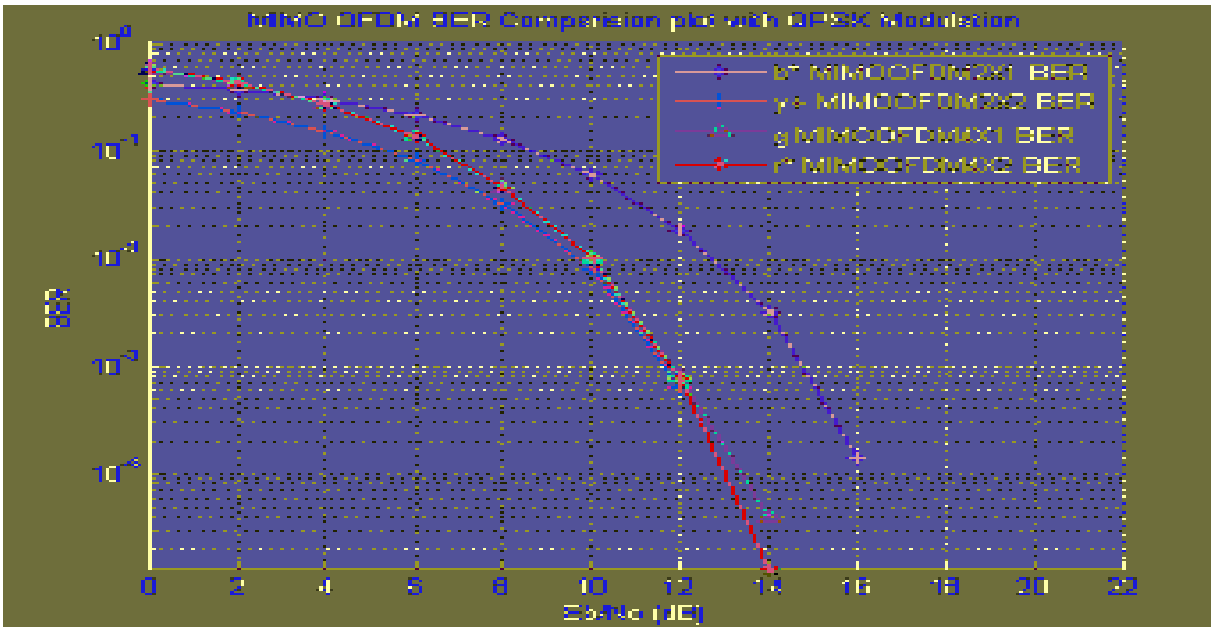
Lower order modulation schemes can be used, but this is at the expense of data throughput. It is necessary to balance all the available factors to achieve a satisfactory bit error rate. Normally it is not possible to achieve all the requirements and some trade-offs are required. However, even with a bit error rate below what is ideally required, further trade-offs can be made in terms of the levels of error correction that are introduced into the data being transmitted. Although more redundant data has to be sent with higher levels of error correction, this can help mask the effects of any bit errors that occur, thereby improving the overall bit error rate.

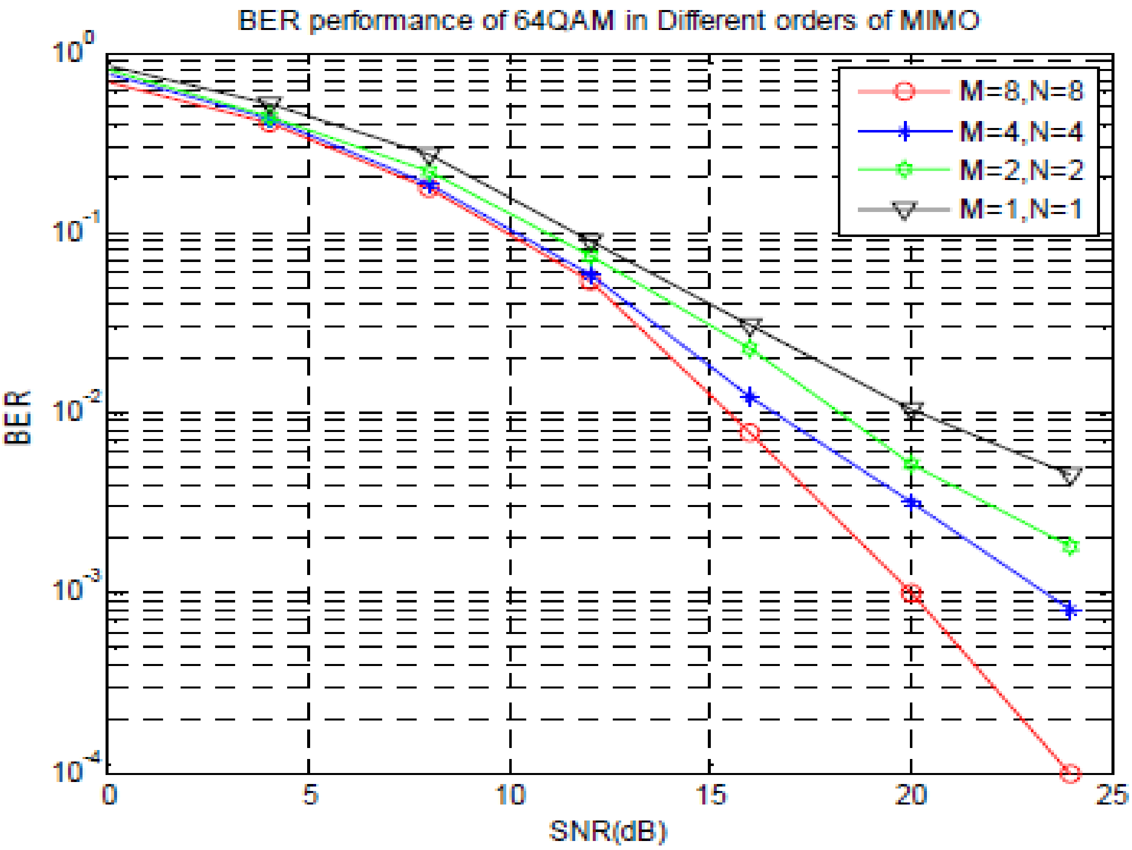
Bit error rate BER is a parameter which gives an excellent indication of the performance of a data link such as radio or fibre optic system. As one of the main parameters of interest in any data link is the number of errors that occur, the bit error rate is a key parameter. Knowledge of the BER also enables other features of the link such as the power and bandwidth, etc to be tailored to enable the required performance to be obtained.

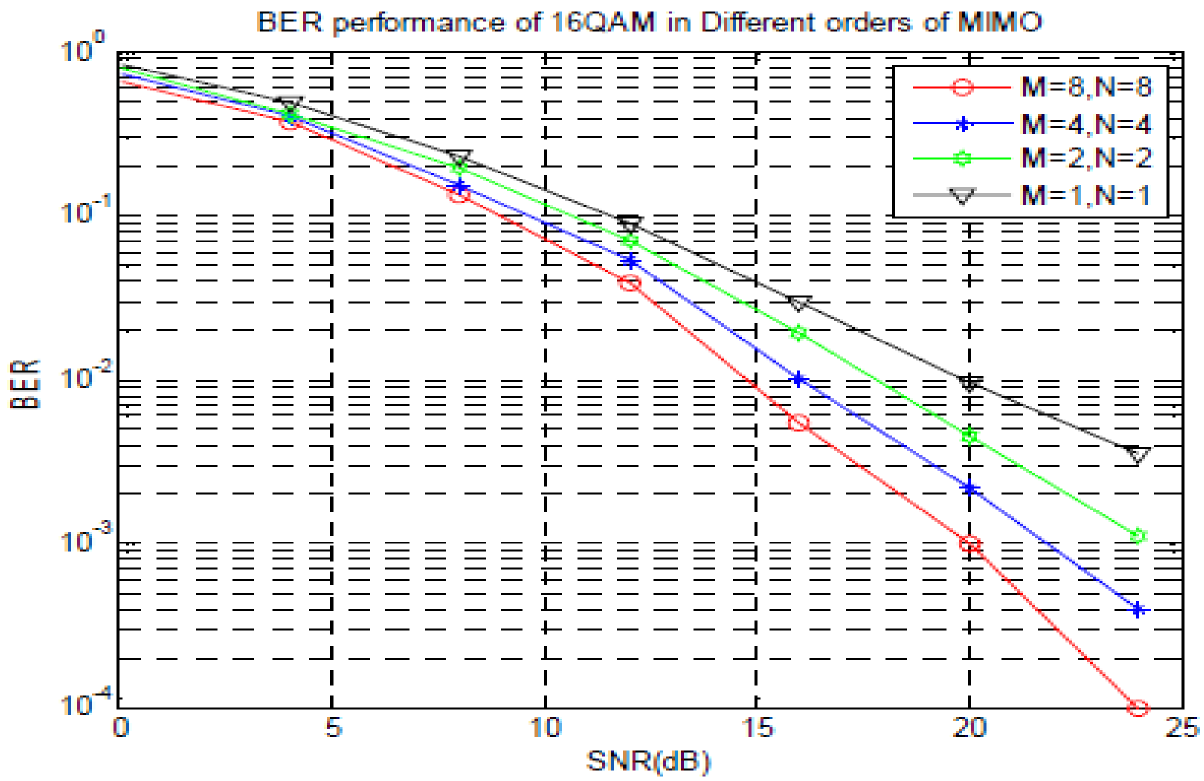
10. Comparative analysis

MODULATION SCHEME	SNR	BER
BPSK	10dB	10^-5
QPSK	10dB	10^-2.9
QAM	10dB	10^-3
64-QAM	10dB	10^-1.8

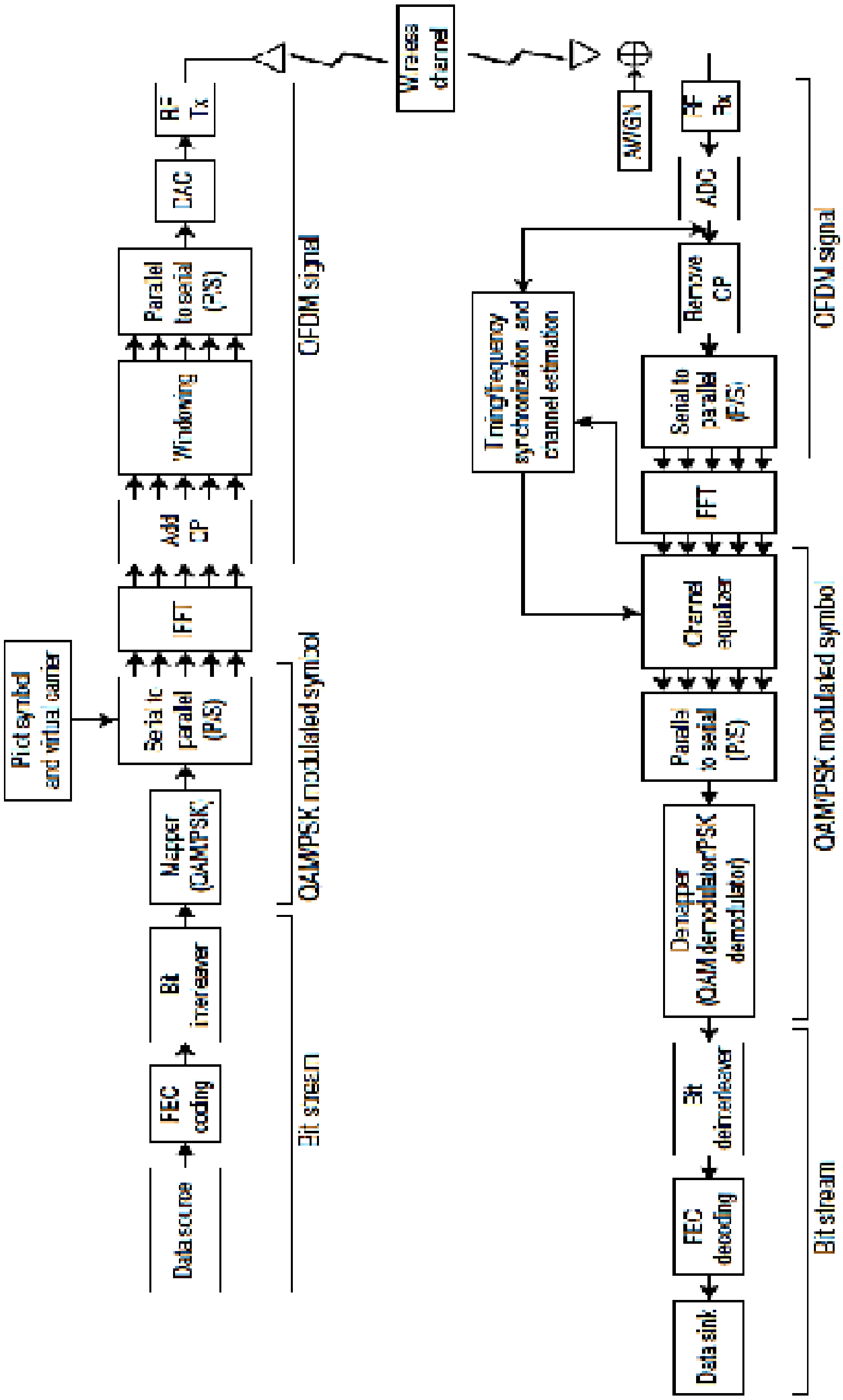



MIMO OFDM BER Comparision plot with QPSK Modulation
b* MIMOOFDM2X1 BER
y+ MIMOOFDM2X2 BER
g MIMOOFDM4X1 BER
r* MIMOOFDM4X2 BER
BER
Eb/No (dB)
0
2
4
6
8
10
12
14
16
18
20
22


BER performance of 64QAM in Different orders of MIMO
M=8,N=8
M=4,N=4
M=2,N=2
M=1,N=1
BER
SNR(dB)
10^{0}
10^{-1}
10^{-2}
10^{-3}
10^{-4}
0
5
10
15
20
25


BER performance of 16QAM in Different orders of MIMO
10^0
10^{-1}
10^{-2}
10^{-3}
10^{-4}
BER
0
5
10
15
20
25
SNR(dB)
M=8,N=8
M=4,N=4
M=2,N=2
M=1,N=1

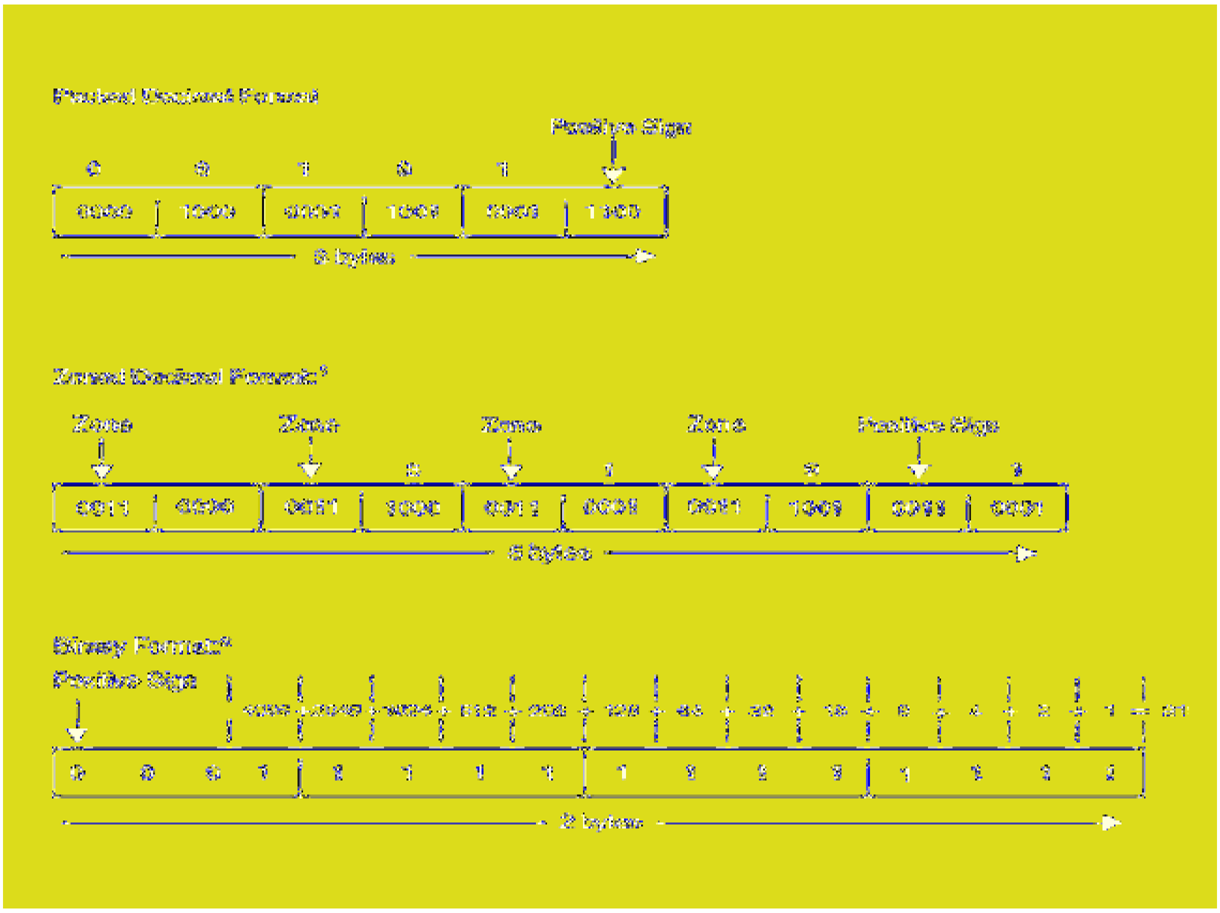
11. Design



Data source
FEC coding
Bit interleaver
Mapper (QAM/PSK)
Pilot symbol and virtual carrier
Serial to parallel (S/P)
IFFT
Add CP
Windowing
Parallel to serial (P/S)
DAC
RF Tx
Bit stream
QAM/PSK modulated symbol
OFDM signal
Wireless channel
AWGN
RF Rx
ADC
Remove CP
Timing/frequency synchronization and channel estimation
Serial to parallel (S/P)
FFT
Channel equalizer
Parallel to serial (P/S)
Demapper (QAM demodulator/PSK demodulator)
Bit deinterleaver
FEC decoding
Data sink
OFDM signal
QAM/PSK modulated symbol
Bit stream

Binary Format

Binary format means that the sign (positive or negative) is in the leftmost bit of the field and the integer value is in the remaining bits of the field. Positive numbers have a zero in the sign bit; negative numbers have a one in the sign bit and are in twos complement form. In binary format, each field must be either 2 or 4 bytes long. A binary field can be from one to nine digits in length and can be defined with decimal positions. If the length of the field is from one to four digits, the compiler assumes a binary field length of 2 bytes. If the length of the field is from five to nine digits, the compiler assumes a binary field length of 4 bytes.



Input and Output Sizes

If the convolutional code uses an alphabet of 2*n* possible symbols, this block's input vector lengthis L**n* for some positive integer *L*. Similarly, if the decoded data uses an alphabet of 2*k* possibleoutput symbols, this block's output vector length is L**k*.This block accepts a column vector input signal with any positive integer value for *L*. Forvariable-sized inputs, the L can vary during simulation. The operation of the block is governedby the operation mode parameter."

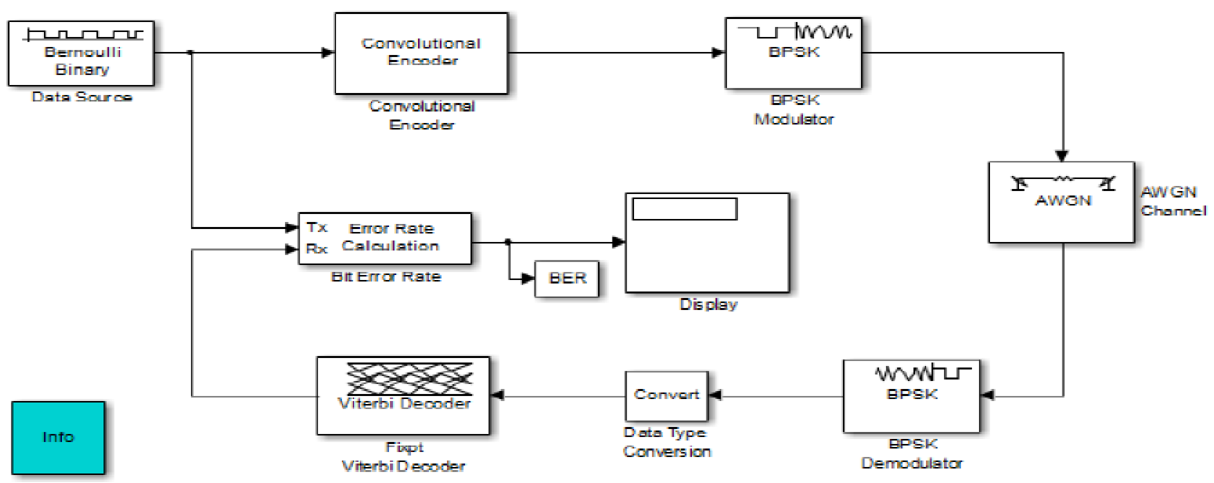
Input Values and Decision Types

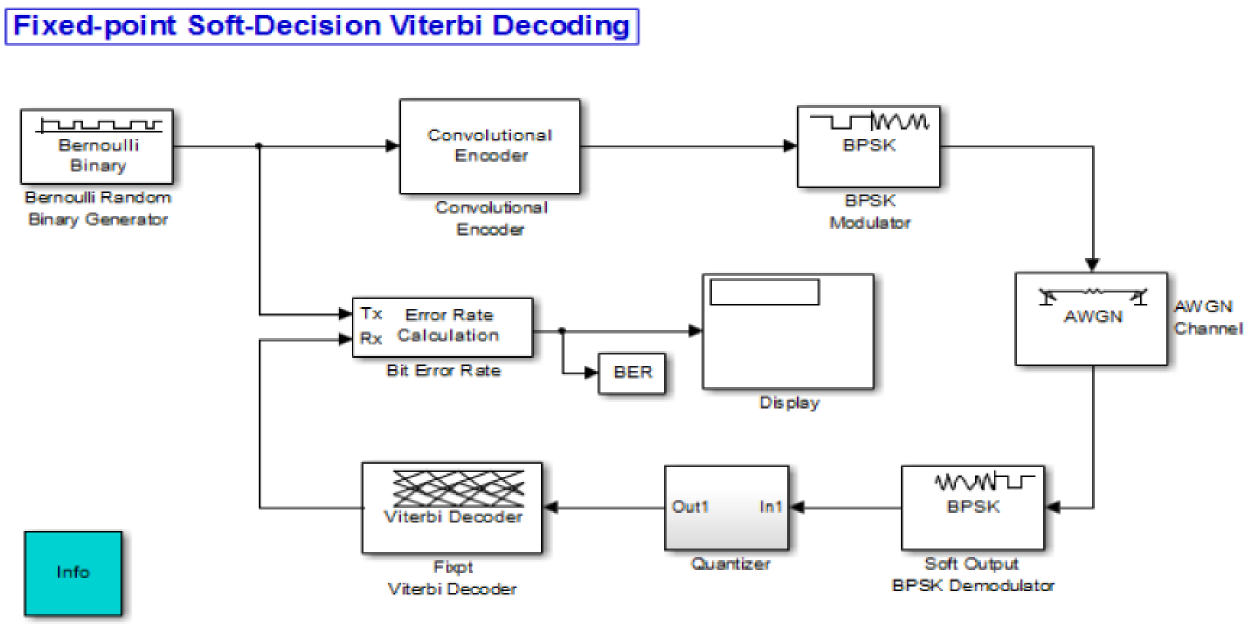
The entries of the input vector are either bipolar, binary, or integer data, depending onthe Decision type parameter.

Fixed-Point Viterbi Decoding Examples

The following two example models showcase the fixed-point Viterbi decoder block used for bothhard- and soft-decision convolutional decoding.If you are reading this reference page in the MATLAB® Help Browser, click Fixed-point Hard-Decision Viterbi Decoding and Fixed-point Soft-Decision Viterbi Decoding to open the models.

Fixed-point Hard-Decision Viterbi Decoding





Comparisons Between Hard and Soft-Decision Decoding

The two models are set up to run from within BERTool to generate a simulation curve thatcompares the BER performance for hard-decision versus soft-decision decoding.

To generate simulation results for doc_fixpt_vitharddec.mdl, do the following:

a. Type bertool at the MATLAB command prompt.
b. Go to the **Monte Carlo** pane.
c. Set the **Eb/No range** to 2:5.
d. Set the **Simulation model** to doc_fixpt_vitharddec.mdl. Make sure that the model is on path.
e. Set the **BER variable name** to BER.
f. Set the **Number of errors** to 100, and the **Number of bits** to 1e6.
g. Press **Run** and a plot generates.

Serial to parallel converter

This is a simple method to convert a serial data that have been entered to the circuit into parallel one. The reverse parallel to serial can be done in a similar manner. In fact, the first one is useful in computer systems architecture. For example, in ALUs, data can be entered serially and the input, for ALU operations, can be applied in parallel.

BPSK Modulation and Demodulation

BPSK (also sometimes called PRK, phase reversal keying, or 2PSK) is the simplest form of phase shift keying (PSK). It uses two phases which are separated by 180° and so can also be termed 2-PSK. It does not particularly matter exactly where the constellation points are positioned, and in this figure they are shown on the real axis, at 0° and 180°. This modulation is the most robust of all the PSKs since it takes the highest level of noise or distortion to make the demodulator reach an incorrect decision. It is, however, only able to modulate at 1 bit/symbol and so is unsuitable for high data-rate applications.

In the presence of an arbitrary phase-shift introduced by the communications channel, the demodulator is unable to tell which constellation point is which. As a result, the data is often differentially encoded prior to modulation.

BPSK is functionally equivalent to 2-QAM modulation.

Implementation

The general form for BPSK follows the equation:

$$s_n(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi ft + \pi(1-n)), n = 0, 1.$$

This yields two phases, 0 and π. In the specific form, binary data is often conveyed with the following signals:

$$s_0(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi ft + \pi) = -\sqrt{\frac{2E_b}{T_b}} \cos(2\pi ft) \text{ for binary "0"}$$

$$s_1(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi ft) \text{ for binary "1"}$$

where *f* is the frequency of the base band.

Hence, the signal space can be represented by the single basis function

$$\phi(t) = \sqrt{\frac{2}{T_b}} \cos(2\pi ft)$$

where 1 is represented by $\sqrt{E_b}\phi(t)$ and 0 is represented by $-\sqrt{E_b}\phi(t)$. This assignment is, of course, arbitrary.

This use of this basis function is shown at the end of the next section in a signal timing diagram. The topmost signal is a BPSK-modulated cosine wave that the BPSK modulator would produce. The bit-stream that causes this output is shown above the signal (the other parts of this figure are relevant only to QPSK). After modulation, the base band signal will be moved to the high frequency band by multiplying $\cos(2\pi f_c t)$.

.

Bit error rate

The bit error rate (BER) of BPSK in AWGN can be calculated as:

$$P_b = Q\left(\sqrt{\frac{2E_b}{N_0}}\right) \text{ or } P_e = \frac{1}{2}\operatorname{erfc}\left(\sqrt{\frac{E_b}{N_0}}\right)$$

Since there is only one bit per symbol, this is also the symbol error rate.

FFT and IFFT

The OFDM transmitter and receiver contain Inverse Fast Fourier Transform (IFFT) and Fast Fourier Transform (FFT), respectively. The IFFT/FFT algorithms are chosen due to their execution speed, flexibility and precision . For real time systems the execution speed is the main concern. The IFFT block provides orthogonality between adjacent subcarriers. The orthogonality makes the signal frame relatively secure to the fading caused by natural multipath environment. As a result OFDM system has become very popular in modern telecommunication systems.

The main objective of this paper is to design IFFT/FFT blocks for OFDM, because these are main blocks for modulation and demodulation in OFDM transmitter and receiver. The OFDM signal is generated by implementing the Inverse Fast Fourier Transform (IFFT) at the transmitter which is used to convert frequency domain to time domain and Fast FourierTransform (FFT) which is used to convert time domain to frequency domain at the receiver side is implemented. The basic equation of the FFT is

$$X(k)=\sum_{n=0}^{N-1} x(n)e^{-j2\pi kn/N},\ k=0,\ldots,N-1$$

On the other hand, the Inverse FFT equation is

$$x(n)=\frac{1}{N}\sum_{k=0}^{N-1} X(k)e^{j2\pi kn/N},\ n=0,\ldots,N-1$$

Where, N is the transform size or the number of sample points in the data frame. X(k) is the frequency output of the FFT at kth point where k=0, 1, …, N-1 and x(n) is the time sample at nth point with n=0, 1,…, N-1.

Pilot signal

"Pilot tone" redirects here. For pilot tones in motion picture sound recording systems, see Pilottone.In telecommunications, a **pilot signal** is a signal, usually a single frequency, transmitted over a communications system for supervisory, control, equalization, continuity, synchronization, orreference purposes. Spectrum of an FM broadcast signal. The pilot tone is the orange verticalline on the right of the spectrogram.

In FM stereo broadcasting, a **pilot tone** of 19 kHz indicatesthat there is stereophonic information at 38 kHz (19×2, the second harmonic of the pilot). Thereceiver doubles the frequency of the pilot tone and uses it as a phase reference to demodulatethe stereo information. If no 19 kHz pilot tone is present, then any signals in the 23-53 kHz rangeare ignored by a stereo receiver. A guard band of ±4 kHz (15-23 kHz) protects the pilot tonefrom interference from the baseband audio signal (50 Hz-15 kHz) and from the lower sidebandof the double sideband stereo information (23-53 kHz). The third harmonic of the pilot (19×3, or57 kHz) is used for Radio Data System.

In AM stereo, the bandwidth is too narrow to accommodate subcarriers, so the modulation itself is changed, and the pilot tone is infrasonic(below the normal hearing range, instead of above it) at a frequency of 25 Hz. In color television,the color burst placed between each pair of video fields is the pilot signal to indicate that thereare color subcarriers present. In the NTSC television system, a pilot tone of 15.7342657 kHz isused to indicate the presence of MTS stereo. In some analog video formats (Frequencymodulation is the standard method for recording the luminance part of the signal, and is used torecord a composite video signal in Direct color systems), e.g. Video 2000 and some Hi-bandformats a pilot tone is added to the signal to detect and correct time base errors.Sometimes it isnecessary to employ several independent pilot frequencies. Most radio relay systems use radio orcontinuity pilots of their own but transmit also the pilot frequencies belonging to the carrierfrequency multiplex system.

Parallel to serial

When data is to be sent over a large distance then the parallel data are send serially over single wire(plus ground). To send parallel data serially from microprocessor or microcontroller or DSP chips, parallel to serial converter is required. Not only is parallel to serial converter required at the microprocessor chip level but also in other parts of communication system. For telephone system for example, parallel to serial and serial to parallel converters are required for implementing modulation/demodulation.

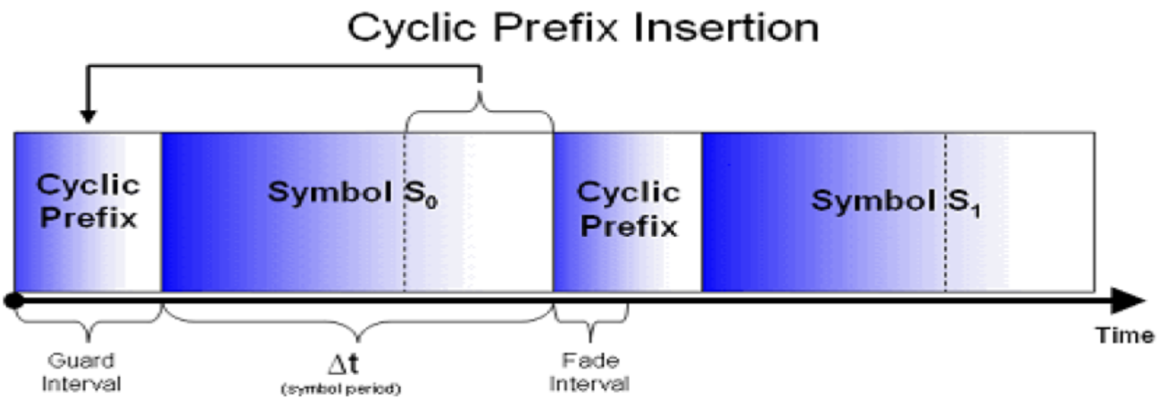
Add cyclic prefix

The term ***cyclic prefix*** refers to the prefixing of a symbol with a repetition of the end. Althoughthe receiver is typically configured to discard the cyclic prefix samples, the cyclic prefix servestwo purposes.

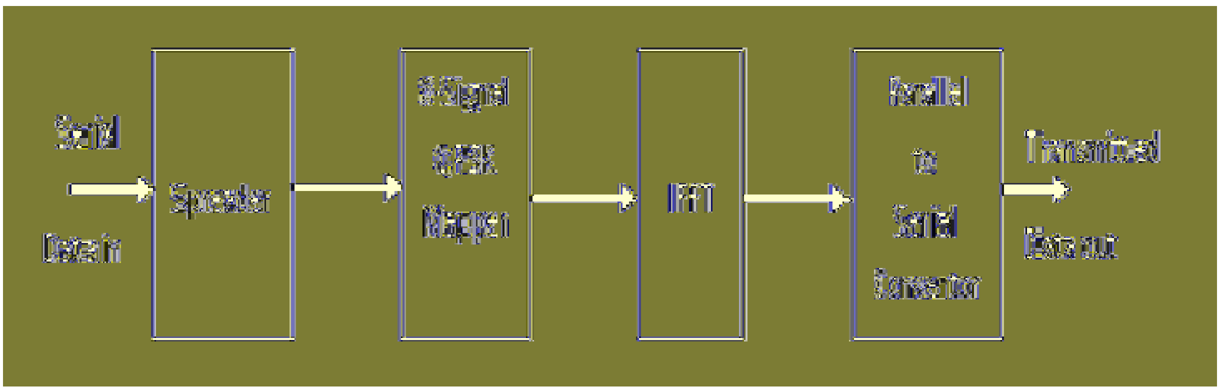
- As a guard interval, it eliminates the inter symbol interference from the previous symbol.
- As a repetition of the end of the symbol, it allows the linear convolution of a frequencyselective multipath channel to be

modeled as circular convolution, which in turn may betransformedto the frequency domain using a discrete Fourier transform. This approachallows for simple frequency-domain processing, such as channel estimation andequalization.

- In order for the cyclic prefix to be effective (i.e. to serve its aforementioned objectives), thelength of the cyclic prefix must be at least equal to the length of the multipath channel. Althoughthe concept of cyclic prefix has been traditionally associated with OFDM systems, the cyclicprefix is now also used in single carrier systems to improve the robustness to multipathpropagation



OFDM Transmitter

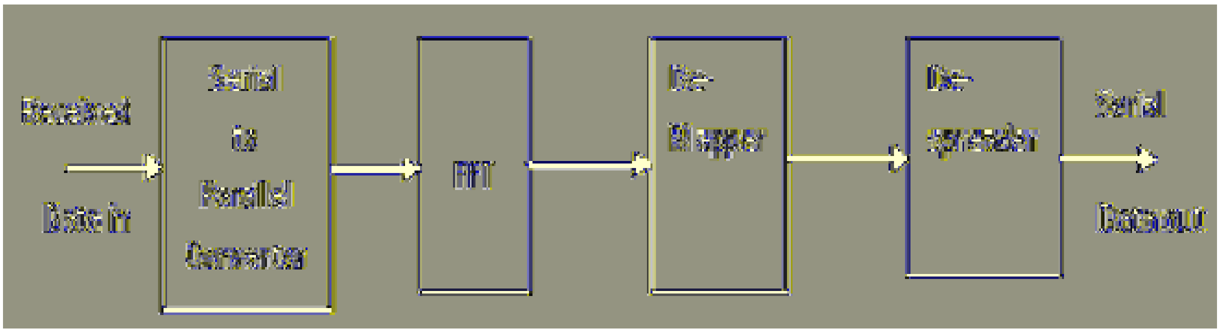


IFFT

The IFFT transform a spectrum (amplitude and phase of each component) into a time domainsignal. An IFFT converts a number of complex data points, of length that is power of 2, into the same number of points intime domain. Each datapoint in frequency spectrum used for an FFT or IFFT operation is called a bin.

The Inverse FastFourier Transform(IFFT) performs N-Point IFFT operation for the received constellation points from the QPSKMapper.. The output is ofN time domain samples. After N-point computation these values are passed through parallel toserial convertor

OFDM Receiver



FFT

FFT Converters time domain to frequency domain. The parallel symbols which are receivedfrom serial to parallelconverter perform N-Point FFT operation and sends to de mapper.

SYMBOL DETECTION

Symbol detection in multi-input multi-output (MIMO)communication systems using differentparticles warm optimization (PSO) algorithms is presented. This approach is particularlyattractive as particles warm intelligence is well suited for real-time applications, where lowcomplexity and fast convergenceis of absolute importance. While an optimal maximumlikelihood (ML) detection using an exhaustive search method is prohibitively complex, PSOassistedMIMO detection algorithms give near-optimal bit error rate (BER) performance with asignificant reduction in ML complexity.

12. Working

The input serial binary data will be processed by a data scrambler and then channel coding is applied to the input data to improve the BER (bit error rate)performance of the system. The encoded data stream is further interleaved to reduce the burstsymbol error rate. Dependent on the channel condition like fading, base modulationmodes such as BPSK (binary phase shift keying), QPSK (quadrature phase shift keying) andQAM are adaptively used to boost the data rate. The modulationmode can be changed even during the transmission of dataframes. The resulting complex numbers are grouped into column vectors which have the samenumber of elements as the FFT size, *N*. For simplicityof presentation and ease of understanding, we choose to use matrix and vector to describe themathematical model.

$$\vec{S}(m) = \begin{bmatrix} S(mN) \\ \vdots \\ S(mN+N-1) \end{bmatrix}_{N\times 1},$$

Let *S*(*m*) represent the *m*-th OFDM symbol in the frequency domain i.e.where*m* is the index of OFDM symbols. We assume that the complex-valued elements *fS*(*mN*)*;S*(*mN*+ 1)*,……,S*(*mN*+ *N* ¡ 1)*g* of ~*S*(*m*) are zero mean and uncorrelated random variableswhose sample space is the signal constellation of the base modulation (BPSK, QPSK and QAM).

To achieve the same average power for all mappings, a normalization factor *K*MOD [37] ismultiplied to each elements of *S*(*m*) such that the average power of the mappings is normalizedto unity. To obtain the time domain samples, as shown by the IDFT block in Figure 1.5, an IFFT(inverse fast Fourier transform) operation is represented by a matrix multiplication.Let F*N* be the *N*-point DFT matrix whose (*p; q*)-th elements is

$$e^{-j\frac{2\pi}{N}(p-1)(q-1)}$$

The resulting time domain samples ~*s*(*m*) can be described by

$$\begin{aligned} \vec{s}(m) &= \begin{bmatrix} s(mN) \\ \vdots \\ s(mN+N-1) \end{bmatrix}_{N\times 1} \\ &= (\tfrac{1}{N})\mathbf{F}_N^H \vec{S}(m). \end{aligned}$$

Compared to the costly and complicated modulation and multiplexing of conventional FDMsystems, OFDM systems easily implement them by using FFT in baseband processing. Tocombat the multipath delay spread in wireless channels, the time-domain samples *s*(*m*) iscyclically extended by copying the last *Ng* samples and pasting them to the front, as shown infigure.

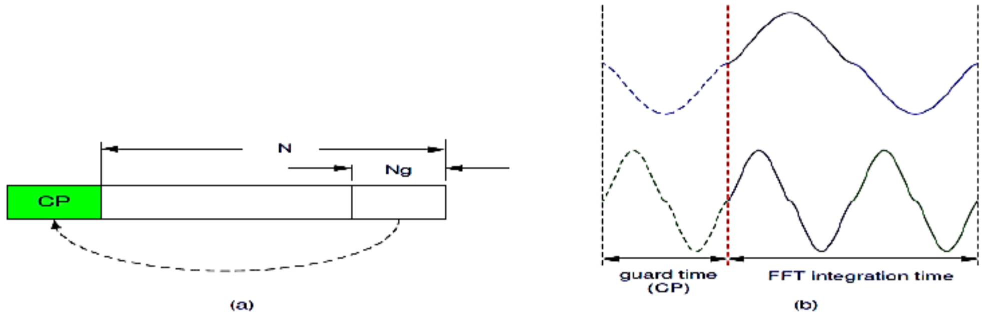


Figure (a) Concept of CP.

(b) OFDM symbol with cyclic extension.

Let $u(m)$ denote the cyclically extended OFDM symbol as

$$\vec{u}(m) = \begin{bmatrix} u(mN_{tot}) \\ \vdots \\ u(mN_{tot} + N_{tot} - 1) \end{bmatrix} = \begin{bmatrix} CP \\ \vec{s}(m) \end{bmatrix}_{N_{tot} \times 1}$$

Where;

$$N_{tot} = N + N_g$$

is the length of $u(m)$. In the form of matrix, the CP insertion can bereadily expressed as a matrix product of $s(m)$ and an N×N matrix ACP . By straightcomputation, it holds that;

where

$$\vec{u}(m) = \mathbf{A}_{CP}\vec{s}(m),$$

$$\mathbf{A}_{CP} = \begin{bmatrix} \mathbf{0} & \mathbf{I}_{N_g} \\ \mathbf{I}_{N-N_g} & \mathbf{0} \\ \mathbf{0} & \mathbf{I}_{N_g} \end{bmatrix}_{(N+N_g)\times N}$$

One of the challenges from the harsh wireless channels is the multipath delay spread.If the delayspread is relatively large compared to the symbol duration, then a delayed copy of a previoussymbol will overlap the current one which implies severe ISI. To eliminate the ISI almostcompletely, a CP is introduced for each OFDM symbol and the length of CP, Ng must be chosenlonger than the experienced delay spread, L, i.e.,$Ng > L$.

In addition, CP is capable of maintaining the orthogonalityamong subcarriers whichimplies zero ICI. It is because the OFDM symbol is cyclically extended and this ensures that thedelayed replicas of the OFDM symbol always have an integer number of cycles within the FFTinterval, as long as the delay is smaller than the CP. Nomatter where the FFT window starts,provided that it is within the CP, there will be always one ortwo complete cycles within FFT integration time for the symbol on top and at belowrespectively.

In IEEE 802.11a standard [37], Ng is at least 16. The obtained OFDM symbol(including the CP) $u(m)$, must be converted to the analogue domain by anDAC (digital-to-analog converter) and then up-converted for RF transmission since it is currentlynot practical to generate the OFDM symbol directly at RF rates. To remain in the discrete-timedomain, the OFDM symbol could be up-sampled and added to a discrete carrier frequency.

Thiscarrier could be an IF (intermediate frequency) whose sample rate is handled by currenttechnology. It could then be converted to analog and increased to the transmit frequencyusing analog frequency conversion methods. Alternatively, the OFDM modulation could beimmediately converted to analog and directly increased to the desired RF transmitt frequency.

Either way has its advantages and disadvantages. Cost, power consumption and complexity mustbe taken into consideration for the selected technique.The RF signal is transmitted over the air. For the wireless channel, it is assumed in this thesis asa quasi-static frequency-selective Rayleigh fading channel. It indicates that the channelremains constant during the transmission of one OFDMsymbol. Suppose that the multipath channel can be modeled by a discrete-time basebandequivalent (L¡1)th-order FIR (Finite impulse response) filter with filter taps.

$$\{h_0, h_1, \ldots, h_l, \ldots, h_{L-1}\}$$

It is further assumed that the channel impulse response, i.e.,the equivalent FIR filter taps, are independent zero mean complex Gaussian random variableswith variance of per dimension. The ensemble of

$$\{\mathcal{P}_0, \ldots, \mathcal{P}_l, \ldots, \mathcal{P}_{L-1}\}$$

is thePDP (power delay profile) of the channel and usually the total power of the PDP is normalizedto be 1 as the unit average channel attenuation. Denote the CIR (channel impulse response)vector *hm*as

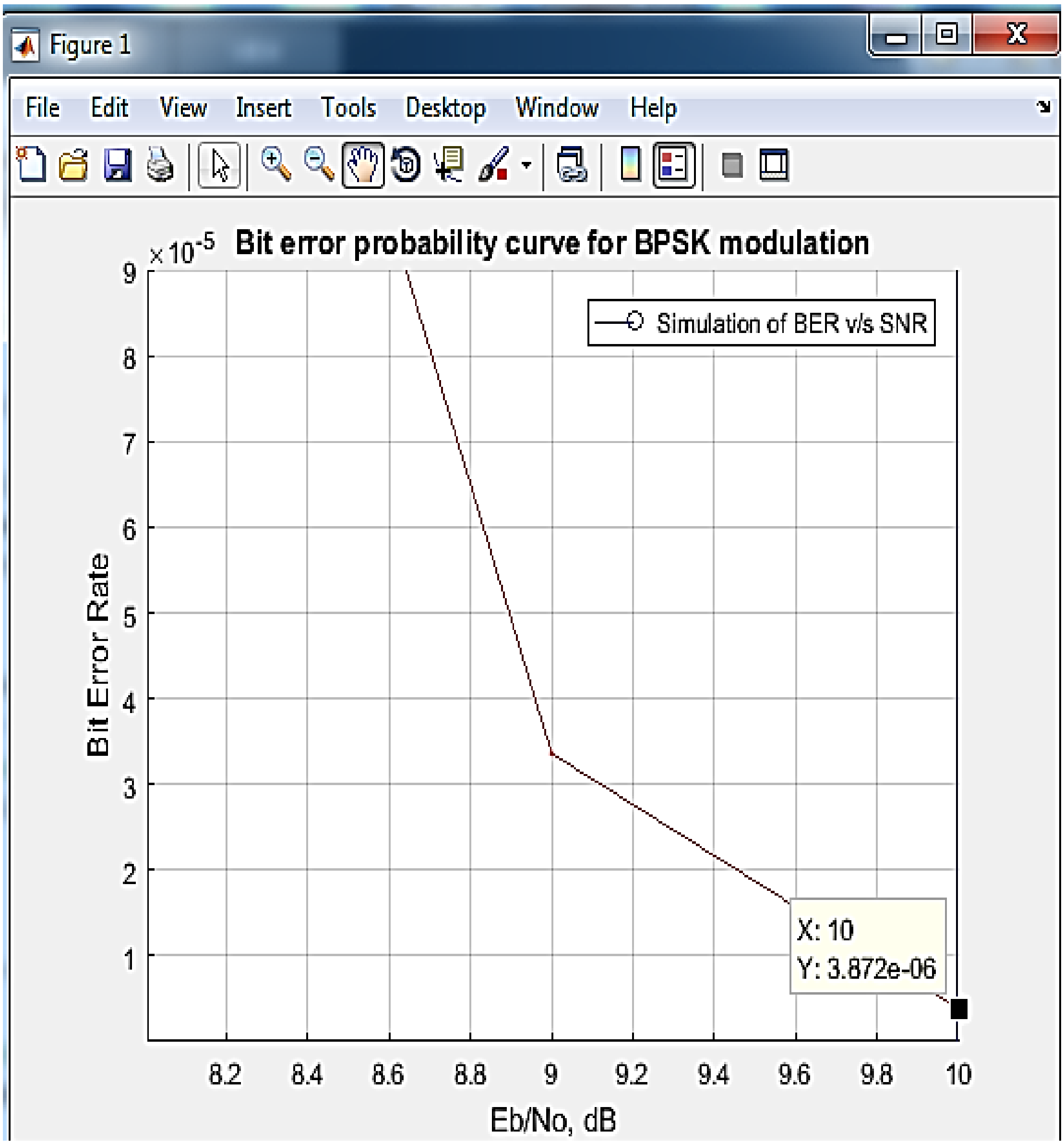
$$\vec{h}_m = \begin{bmatrix} h_{0,m} \\ \vdots \\ h_{L-1,m} \end{bmatrix}_{L\times 1}$$

Where, the subscript *m* is kept to imply that the channel may vary from one OFDM symbol to thenext one. Then the complex baseband equivalent received signal can be represented by adiscrete-time convolution as

$$r(mN_{tot} + n) = \sum_{l=0}^{L-1} h_{l,m} u(mN_{tot} + n - l) + v(mN_{tot} + n)$$

13. Results

➢ **For Eb/No=10dB, Bit Error Rate=3.87×10^{-06}**



14. Future scope

OFDM and MIMO will serve as the physical layer of two key technologies for future mobile communication systems: UMTS LTE and WiMax. OFDM and MIMO will help in the evolving standards for LTE and WiMax. LTE is the 4G evolution of cellular systems, while WiMax is a technology that is expected to deliver last mile wireless broadband access. Background to LTE: HSPA Release 7

- LTE Design Goals
- Frame Structure
- Downlink MIMO Modes
- Physical Resource Block
- LTE uplink: Single Carrier (SC) – FDMA
- WiMax: Overview of MAC layer, MAC protocol data units
- Frame Structure
- Ranging
- Quality of Service Classification
- Scalable OFDMA
- Adaptive Modulation and Coding
- OFDMA Channelization: PUSC, FUSC, AMC, Matlab example
- Multiple Antenna Technology in WiMax
- Power consumption pattern
- Design complexity of receiver
- Future 4G or a dominant interface between 4G and 5G
- Co-operative Partial Transmit Sequence(CO-PTS) system for cognitive radios to improve simulation accuracy and efficiency can be used
- Multi user MIMO systems
- MIMO-OFDM channel estimations using blind methods

15. Conclusion

MIMO-OFDM is a powerful modulation technique used for high data rate, and is able to eliminate ISI. It is computationally efficient due to the use of FFT techniques to implement modulation and demodulation functions. The performance of MIMO-OFDM is tested for two digital modulation technique BPSK using MATLAB/SIMULINK toolbox. Computer simulations are done to verify the performance of the proposed approach. A simulation tool with a Graphical User Interface (GUI) which implements these algorithms is also developed to provide ease in the execution.

Performance analysis of FFT-OFDM and is done using BPSK under AWGN channel. STBC-MIMO system is implemented for 2*2 diversity scheme. STBC-MIMO system as we increase number of receive antennas BER performance of system is improved. System complexity is reduced and the transmission rate is increased. So, the combination of MIMO system and FFT-OFDM system will improve the BER performance of wireless communication system.

MIMO-OFDM system with BPSK scheme is suitable for low capacity, short distance application. While the OFDM with higher modulation scheme is used for large capacity, long distance application at the cost of slight increase in Eb/No. BER is large in BPSK as compared to QAM and it generally depends on applications. In this project, we implemented and optimized the 2x2 MIMO-OFDM transmitter and receiver for different embedded platform. We reduced the bit error rate of MIMO-OFDM from 10^-2 to 10-5 using 2x2 array. We concluded that BPSK modulated MIMO - OFDM system achieves better BER results than QPSK/QAM and other modulated MIMO - OFDM systems for the same bandwidth efficiency.

16. References

- "ShubhangiChaudhary and A. J. Patil, "Performance Analysis of MIMO-STBC with Different Modulation Techniques," *ICTACT Journal on Communication Technology, Volume:03, Issue:01,* pp.510-514, March 2012."

- 'K.Vidhya , K.R.Shankarkumar' "BER Performance of MIMO-OFDM System using STTC" *International Journal of Scientific and Research Publications, Volume 3, Issue 2, February 2013 1 ISSN 2250-3153*

- "MadanLal and HamneetArora"," BER Performance of Different Modulation Schemes for MIMO Systems"*IJCSNS International Journal of Computer Science and Network Security, VOL.11 No.3, March 2011 69*

- "ShrutiTrivedi, Mohd. SarwarRaeen, Shalendra Singh pawar" *"*BER Analysis of MIMO-OFDM System using BPSK Modulation Scheme " *International Journal of Advanced Computer Research (ISSN (print): 2249-7277 ISSN (online): 2277-7970)Volume-2 Number-3 Issue-5 September-2012*208

- MIMO-OFDMWIRELESSCOMMUNICATIONS WITH MATLAB_YongSoo Cho, Chung-Ang, Jaekwon Kim, Won Young Yang, Chung G. Kang

- *VijaykumarKatgi, Asst. Prof. Department of Electronic and Communication Engineering, BKIT, Bhalki, INDIA* "Comparison of MIMO OFDM System with BPSK and QPSK Modulation" *"International Journal on Emerging Technologies* (Special Issue on NCRIET-2015) 6(2): 188-192(2015)

17. Appendix

Program:

```
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%

% Script for computing the BER for BPSK
modulation in a
% AWGN channel with Alamouti Space Time Block
Coding
% Two transmit antenna, two Receive antenna
clear;
c=6;% no of sub carrier channels
bits=54;% bits per channels
n=324;% total no of bits to be transmitted

for i=1:n
data(i)= 2*round(rand)-1;
end


% Converting the series into parallel for the
channels
s = reshape(data,c,bits);

%let
N = 324; % number of bits or symbols

rand('state',100); % initializing the rand()
function
randn('state',200); % initializing the randn()
function

% BPSK modulation for a single channel
```

```
signal1 = s(1,:);
exdata1=[];
%first expand the bit stream
for i=1:length(signal1)
for rep=1:2
  exdata1= [exdata1 signal1(i)];
end
end

signal2 = s(2,:);
%first expand the bit stream
exdata2=[];
for i=1:length(signal2)
for rep=1:2
  exdata2= [exdata2 signal2(i)];
end
end

signal3= s(3,:);
%first expand the bit stream
exdata3=[];
for i=1:length(signal3)
for rep=1:2
  exdata3= [exdata3 signal3(i)];
end
end

signal4= s(4,:);
%first expand the bit stream
exdata4=[];
for i=1:length(signal4)
for rep=1:2
  exdata4= [exdata4 signal4(i)];
end
```

```
end

signal5= s(5,:);
%first expand the bit stream
exdata5=[];
for i=1:length(signal5)
for rep=1:2
  exdata5= [exdata5 signal5(i)];
end
end

signal6= s(6,:);
%first expand the bit stream
exdata6=[];
for i=1:length(signal6)
for rep=1:2
  exdata6= [exdata6 signal6(i)];
end
end

% Transmitter
ip = rand(1,N)>0.5; % generating 0,1 with equal
probability
signal1 = 2*ip-1; % BPSK modulation 0 -> -1; 1 -
> 1
signal2 = 2*ip-1; % BPSK modulation 0 -> -1; 1 -
> 1
signal3 = 2*ip-1; % BPSK modulation 0 -> -1; 1 -
> 1
signal4 = 2*ip-1; % BPSK modulation 0 -> -1; 1 -
> 1
signal5 = 2*ip-1; % BPSK modulation 0 -> -1; 1 -
> 1
signal6 = 2*ip-1; % BPSK modulation 0 -> -1; 1 -
> 1
```

```
n = 1/sqrt(2)*(randn(1,N) + 1i*randn(1,N)); %
white gaussian noise, 0dB variance
Eb_N0_dB = (-3:10); % multiple Eb/N0 values

for ii = 1:length(Eb_N0_dB)
% Noise addition
   y1 = signal1 + 10^(-Eb_N0_dB(ii)/20)*n; %
additive white gaussian noise
 y2 = signal2 + 10^(-Eb_N0_dB(ii)/20)*n; %
additive white gaussian noise
  y3 = signal3 + 10^(-Eb_N0_dB(ii)/20)*n; %
additive white gaussian noise
   y4 = signal4 + 10^(-Eb_N0_dB(ii)/20)*n; %
additive white gaussian noise
    y5 = signal5 + 10^(-Eb_N0_dB(ii)/20)*n; %
additive white gaussian noise
     y6 = signal6 + 10^(-Eb_N0_dB(ii)/20)*n; %
additive white gaussian noise

% receiver - hard decision decoding
   ipHat1 = real(y1)>0;
    ipHat2 = real(y2)>0;
     ipHat3 = real(y3)>0;
      ipHat4 = real(y4)>0;
       ipHat5 = real(y5)>0;
        ipHat6 = real(y6)>0;

% counting the errors
nErr(1) = size(find((ip- ipHat1)),2);
nErr(2) = size(find((ip- ipHat2)),2);
nErr(3) = size(find((ip- ipHat3)),2);
nErr(4) = size(find((ip- ipHat4)),2);
nErr(5) = size(find((ip- ipHat5)),2);
nErr(6) = size(find((ip- ipHat6)),2);
```

```
end
for i=1:6
simBer = nErr(i)/N; % simulated ber
theoryBer = 0.5*erfc(sqrt(10.^(Eb_N0_dB/10))); %
theoretical ber
end

m=10*n;% Generating the carrier signal
ts=.1;
tp=1:ts:11.79;
% Generating the modulated signal 1
carrier1=cos(2*pi*tp);
bpsk_sig1=exdata1.*carrier1;
% Generating the modulated signal 2
carrier2=cos(4*pi*tp);
bpsk_sig2=exdata2.*carrier2;
% Generating the modulated signal 3
carrier3=cos(6*pi*tp);
bpsk_sig3=exdata3.*carrier3;
% Generating the modulated signal 4
carrier4=cos(8*pi*tp);
bpsk_sig4=exdata4.*carrier4;
% Generating the modulated signal 5
carrier5=cos(10*pi*tp);
bpsk_sig5=exdata5.*carrier5;
% Generating the modulated signal
carrier6=cos(12*pi*tp);
bpsk_sig6=exdata6.*carrier6;

% taking the iFFT of each of these signals
if_sig1=ifft(bpsk_sig1);
if_sig2=ifft(bpsk_sig2);
if_sig3=ifft(bpsk_sig3);
if_sig4=ifft(bpsk_sig4);
```

```
if_sig5=ifft(bpsk_sig5);
if_sig6=ifft(bpsk_sig6);

fin(1,:)=if_sig1;
fin(2,:)=if_sig2;
fin(3,:)=if_sig3;
fin(4,:)=if_sig4;
fin(5,:)=if_sig5;
fin(6,:)=if_sig6;

transmit=reshape(fin,1,648);

% generating the noise

p=rand*2*pi;
snr=10;
r=sqrt(-1*(1/snr*log(1 - rand)));
no = (r.* exp(1i*p));
al=rand+1i*rand;% Spreading channel with the
alpha as the variable


for k=2:2:646
for l = 1:2
%al=round(rand)+j*round(rand)
rec(k+l)=transmit(k+l)+al*transmit(k-2+l);
end
end

rxdata=rec + no ;

% Converting from serial to parallel
myrec=reshape(rxdata,6,108);

rxdata1=fft(myrec(1,:));
```

```
rxdata2=fft(myrec(2,:));
rxdata3=fft(myrec(3,:));
rxdata4=fft(myrec(4,:));
rxdata5=fft(myrec(5,:));
rxdata6=fft(myrec(6,:));

% taking the FFT
%begin demodulation
%first multiply receivedbit stream by cosine
wave with carrier frequency


uncarry1=rxdata1.*carrier1;
uncarry2=rxdata2.*carrier2;
uncarry3=rxdata3.*carrier3;
uncarry4=rxdata4.*carrier4;
uncarry5=rxdata5.*carrier5;
uncarry6=rxdata6.*carrier6;


%demodulate by integrating
dec1=[];
dec2=[];
dec3=[];
dec4=[];
dec5=[];
dec6=[];
for inc=1:2:length(uncarry1)
dec=trapz(inc:inc+1,uncarry1(inc:inc+1));
  dec1=[dec1 dec];
end
%2
for inc=1:2:length(uncarry2)
dec=trapz(inc:inc+1,uncarry2(inc:inc+1));
  dec2=[dec2 dec];
```

```
end
%3
for inc=1:2:length(uncarry3)
dec=trapz(inc:inc+1,uncarry3(inc:inc+1));
  dec3=[dec3 dec];
end
%4
for inc=1:2:length(uncarry4)
dec=trapz(inc:inc+1,uncarry4(inc:inc+1));
  dec4=[dec4 dec];
end
%5
for inc=1:2:length(uncarry5)
dec=trapz(inc:inc+1,uncarry5(inc:inc+1));
  dec5=[dec5 dec];
end
%6
for inc=1:2:length(uncarry6)
dec=trapz(inc:inc+1,uncarry6(inc:inc+1));
  dec6=[dec6 dec];
end
final_rec(1,:)=dec1;
final_rec(2,:)=dec2;
final_rec(3,:)=dec3;
final_rec(4,:)=dec4;
final_rec(5,:)=dec5;
final_rec(6,:)=dec6;

fin_rec_parallel=reshape(final_rec,1,324);

%make decision with a threshold of zero
demod=[];
for i=1:length(fin_rec_parallel)
iffin_rec_parallel(i)>0
demod=[demod 1];
```

```
else
demod=[demod -1];
end
end
plot(demod);
gridon
%stem(demod)
%calculate errors
error=0;
for i=1:length(demod)
if data(i)~=demod(i)
error=error+1;
end
end
error;

% plot
close
figure
holdon
ber=error/324;
stem(data)
semilogy(Eb_N0_dB,theoryBer,'r.-');
holdon
semilogy(Eb_N0_dB,simBer,'rx-');
axis([8 10 10^-5 10^-4])
stem(demod,'bx')
gridon
legend('Simulation of BER v/s SNR');
xlabel('Eb/No, dB');
ylabel('Bit Error Rate');
title('Bit error probability curve for BPSK
modulation');
```