

Performance of BER for BPSK and DPSK (Coherent and Non-Coherent) Modulation in Turbo-Coded OFDM with Channel Equalization

Saruti Gupta, Geetanjali Wasson

Abstract—With the increasing demand of wireless communication, there will be growing need of design of high-speed wireless communication system. But gross frequency selective fading of wireless channel is a problem difficult to solve in design of system. Orthogonal frequency division multiplexing (OFDM) is quite effective to eliminate frequency selective fading of wireless channel, and is a simple and effective technique, so OFDM has been one of the most important techniques for high-speed wireless communication system. Then, the influence of cyclic prefix and adaptive channel equalization on the system performance is analyzed in various fading channel. However in fading environments the bit error rate (BER) increases. The performance can be improved by using some kind of channel coding. This form of OFDM is called coded-OFDM (COFDM). The turbo coding also allows achieving the Shannon's bound Performance.

In this paper we will design the OFDM system with channel Equalization under the powerful concatenated turbo codes to it. This will help to maintain the system performance under a desired bit error rate, as there were errors occurring in burst form which eventually degrades the efficiency of the system. This paper deals with the optimization to analyze the comparative study of BER for the BPSK and DPSK (coherent and non-coherent) modulation scheme to achieve higher data speed and less probability of error for robust and reliable communication under our proposed system model.

Index Terms— OFDM, cyclic prefix, AWGN, Rayleigh, equalization, turbo encoder, turbo decoder, BPSK, DPSK, BER,

I. INTRODUCTION

One of promising techniques for high-speed data transmission is OFDM (orthogonal frequency division multiplexing), a kind of multicarrier modulation scheme. It removes the detrimental effect of multipath fading by partitioning the wideband fading channel into flat narrow band channels[3]. Digital modulation techniques contribute to the evolution of our mobile wireless communications by increasing the capacity, speed as well as the quality of the wireless network. In this paper, we focus on the BPSK and DPSK (coherent and non-coherent) modulation digital modulation technique over AWGN, Rayleigh and rician fading channels. Further high bit error rates of the wireless communication system require employing Robust and powerful forward error correction (FEC) methods on the data transferred to reducing the error in transmission. In this paper we will see how performance of an OFDM system can be improved by adding turbo codes to it. The turbo coding also allows achieving the Shannon's bound Performance.

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This will help to maintain the system performance under a desired bit error rate, as there were errors occurring in burst form in OFDM which eventually degrades the efficiency of the system. Moreover, we can easily overcome the major disadvantages of OFDM i.e. ISI (inter symbol interference) and ICI (Inter carrier Interference) by implementing cyclic prefix and channel equalization through LMS equalizer. Channel equalization is proposed via LMS adaptive algorithm to mitigate the effects of ISI and decrease the probability of error that occurs at the receiver side[17]. Our aim is to optimize and analyse the comparative study of BER for the BPSK and DPSK (coherent and non-coherent) modulation scheme through our proposed system under OFDM technique with channel equalization under turbo coded encoding.

This paper deals with the optimization to analyze the comparative study of BER for the BPSK and DPSK (coherent and non-coherent) modulation scheme to achieve higher data speed and less probability of error for robust and reliable communication.

Section II explains the basic principles of OFDM with channel Equalization in various fading environments employed with the BPSK and DPSK (coherent and non-coherent) modulation scheme. Section III contains brief discussion about turbo encoder and decoder. Section IV explains implementation of coded OFDM with Turbo and RS codes with simulation results and analysis. The paper is concluded in Section V.

II. OFDM

A. BASIC PRINCIPLE

The flexibility of OFDM physical layer technologies provides opportunities to use advanced techniques in several LAN, MAN, as IEEE802-11a, HIPERLAN/2 and IEEE802.16e [1]. Orthogonal frequency division multiplexing (OFDM) is a Special case of multicarrier transmission, where a single data stream is transmitted over a number of lower rate subcarriers.

OFDM is most preferred for high speed communication in multipath environment due to its immunity to ISI. OFDM avoids ISI problem by sending many low speed transmissions simultaneously. OFDM is presently used in a number of commercial wired and wireless applications. However the performance of OFDM in fading environments degrades due to various impairments e.g. frequency selective fading, time selective fading and other propagation effects.

The basic principle of OFDM is to split a high rate data stream into a number of lower rate streams that are transmitted simultaneously over a number of orthogonal sub-carriers [2].

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Orthogonality is achieved by the fact that carriers are placed exactly at the nulls in the modulation spectra of each other. Here, the increase of symbol duration for the lower rate parallel subcarriers reduces the relative amount of dispersion in time caused by multi-path delay spread.

TABLE I OFDM Specification

Parameter	Value
FFT size	64
Number of sub-carrier	52
FFT sampling Frequency	20 MHZ
Cyclic prefix duration	0.8 us
Number of OFDM symbols	100
Modulation	BPSK and DPSK (coherent and non-coherent)

The specification of OFDM used in our design implementation is given in table I.

Let $\{s_{n,k}\}_{k=0}^{N-1}$ with $E|s_{n,k}|^2 = \sigma_s^2$ be the complex symbols to be transmitted at the n th OFDM block, then the OFDM modulated signal can be represented by

$$s_n(t) = \sum_{k=0}^{N-1} s_{n,k} e^{j2\pi k \Delta f t}, \quad 0 \leq t \leq T_s \quad (1)$$

where T_s , Δf , and N are the symbol duration, the subchannel space, and the number of subchannels of OFDM signals, respectively.

For the receiver to demodulate the OFDM signal, the symbol duration should be long enough such that $T_s \Delta f = 1$, which is also called the orthogonal condition since it makes $e^{-j2\pi k \Delta f t}$ orthogonal to each other for different k . With the orthogonal condition, the transmitted symbols $s_{n,k}$ can be detected at the receiver by

$$s_{n,k} = 1/T_s \int s_n(t) e^{-j2\pi k \Delta f t} dt \quad (2)$$

if there is no channel distortion.

The sampled version of the baseband OFDM signal $s(t)$ in (1) can be expressed as

$$S_n(mT_s/N) = \sum_{k=0}^{N-1} s_{n,k} e^{j2\pi k \Delta f m T_s/N} = \sum_{k=0}^{N-1} s_{n,k} e^{j2\pi k m/N} \quad (3)$$

which is actually the inverse discrete Fourier transform (IDFT) of the transmitted symbols $\{s_{n,k}\}_{k=0}^{N-1}$ and can efficiently be calculated by fast Fourier transform (FFT). It can easily be seen that demodulation at the receiver can be performed using DFT instead of the integral in (2).

If we add the guard time with no transmission then it creates problems for IFFT and FFT, which results in inter-carrier interference (ICI)[3]. A delayed version of one subcarrier can interfere with another subcarrier in the next symbol period. This can be avoided by extending the symbol into the guard period that precedes it. And this is known as a cyclic prefix. With the CP, the transmitted signal is extended to $T = T_g + T_s$ and can be expressed as

$$s_n'(t) = \sum_{k=0}^{N-1} s_{n,k} e^{j2\pi k \Delta f t}, \quad -T_g \leq t \leq T_s.$$

It is obvious that $s_n'(t) = s_n(t + T_s)$ for $-T_g \leq t \leq 0$, which is why it is called the CP.

The impulse response of a wireless channel can be expressed by

$$h(t) = \sum_i \gamma_i \delta(t - \tau_i) \quad (4)$$

where τ_i and γ_i are the delay and the complex amplitude of the i th path, respectively. Then, the received signal can be expressed as

$$x_n(t) = \sum_i \gamma_i s_n'(t - \tau_i) + n(t)$$

where $n(t)$ represents the additive white Gaussian noise (AWGN) at the receiver. As demonstrated in Fig. 1, $x_n(t)$ consists of only the signal component from the n th OFDM block when $\tau_l \leq t \leq \tau_u$, where $\tau_l = -T_g + \tau_m$, $\tau_u = T_s + \tau_m$, $\tau_m = \min_i \{\tau_i\}$, and $\tau_M = \max_i \{\tau_i\}$; otherwise, the received signal consists of signals from different OFDM blocks.

If $\tau_l \leq 0$ and $\tau_u \geq T_s$, then

$$\begin{aligned} x_{n,k} &= 1/T_s \int x_n(t) e^{-j2\pi f_k t} dt \\ &= 1/T_s \int (\sum_i \gamma_i s_n'(t - \tau_i) + n(t)) e^{-j2\pi f_k t} dt \\ &= H_k s_{n,k} + n_k \end{aligned} \quad (5)$$

for $0 \leq k \leq N - 1$ and all n , where H_k denotes the frequency response of the wireless channel at the k th subchannel and is defined as

$$H_k = \sum_i \gamma_i e^{-j2\pi k \Delta f \tau_i}$$

and n_k is the impact of AWGN and is defined as

$$n_k = 1/T_s \int n(t) e^{-j2\pi f_k t} dt.$$

It can be proved that n_k are independent identically distributed complex circular Gaussian with zero mean and variance σ_{2n} . With H_k , transmitted symbols can be estimated.

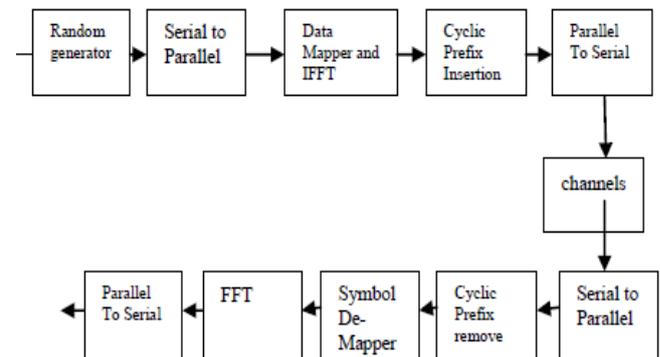


Fig. 1 OFDM Block Diagram

For single carrier systems, the received signal is the convolution of the transmitted sequences or symbols and the impulse response of wireless channels in addition to AWGN, whereas the impact of the channel is only a multiplicative distortion at each subchannel for OFDM systems,

which makes signal detection in OFDM systems very simple and is also one of the reasons why OFDM is very popular nowadays.

Firstly serial string of binary data is modulated and transformed into N-way parallel bit stream, corresponding to N different sub-carriers [5]. According to the channel conditions that is considering multipath fading for the use of modulation BPSK and DPSK (coherent and non-coherent) mapping this paper selects the appropriate sub-carrier into the lead frequency after the IFFT, and through the string conversion, a practical and effective OFDM data symbol will be gotten, adding a cyclic prefix to get a complete OFDM symbol. The signal will be sent into the channel. The part of the process of receiving and sending processes are on the contrary, firstly it removes the cyclic prefix [6], assuming the signal synchronization has been completed, and then after a string and converted through the FFT operation to be transmitted on each sub-carrier signal after the inverse mapping of the constellation points to get the various sub-bits of information on the carrier, and finally through and convert the final string of binary bit stream to receive information [7].

B. MODULATION

The study focuses on analysis BER (Bit error rate) of BPSK and DPSK (coherent and non-coherent) modulation scheme.

BPSK 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. 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.

Probability of error for BPSK

$$P_e = Q \sqrt{\frac{2 E_B}{N_0}}$$

The BDPSK modulation is the first binary numeral baseband signal differential coding, transforms the basic code into the relative code, then carries on the absolute phase modulation, The main methods are coherent demodulation and non-coherent demodulation. A phase change of ¼ across adjacent symbol intervals represents the bit value “1” while no phase change across adjacent symbol intervals represents “0”.

Probability of error for DPSK

$$P_e = \frac{1}{2} e^{(-EB/NO)}$$

C. CHANNEL EQUALIZATION

Inserting an equalizer realized as an adaptive system, the influence of variable delay and multipath could be mitigated in order to remove or reduce considerably the guard interval and to gain some spectral efficiency.

Concerning the adaptive algorithm, we employ a recently developed in [8] method based on adaptive filtering with averaging for parameter update. Comparison with one of the most frequently used algorithms – Least Mean Squares (LMS) [9] shows that the present design has three very attractive features: high adaptation rate, relatively low computational complexity and robustness in fixed-point implementations. The equalizer works in two modes. First, in training mode for channel estimation. Estimation error is defined as the difference between the estimate and the original signal available through the pilot signals. Second, in

decision direction fashion, when the estimation error is determined as the difference between the estimate and the detected data symbols at the decision device output.

D. CHANNELS

The channels focused under our design implementation are considering AWGN, Rayleigh and Rician channel.

1) *AWGN Channel:* Additive white Gaussian Noise (AWGN) is a channel model in which the only impairment to communication is a linear addition of wideband or white noise with a constant spectral density (expressed as watts per hertz of bandwidth) and a Gaussian distribution of amplitude. The model does not account for fading, frequency, selectivity, interference, nonlinearity or dispersion.

2) *Rayleigh Channel:* In a multipath environment, it is reasonably intuitive to visualize that an impulse transmitted from the transmitter will reach the receiver as a train of pulses. When there are large numbers of paths, applying Central Limit Theorem, each path can be modelled as circularly complex Gaussian random variable with time as the variable [7]. This model is called Rayleigh fading channel model. The channel model is reasonable for an environment where there are large numbers of reflectors [10]. The channel is modelled as n-tap channels with each the real and imaginary part of each tap being an independent Gaussian random variable. The impulse response is,

$$h(t) = \frac{1}{\sqrt{n}} h(t-t_1) + h(t-t_2) + \dots + h(t-t_n)$$

Where, $h(t-t_1)$ is the channel coefficient of the first tap, $h(t-t_2)$ is the channel coefficient of the second tap and so on. The real and imaginary part of each tap is an independent Gaussian random variable with mean 0 and variance 1/2. The term $1/\sqrt{n}$ is for normalizing the average channel power over multiple channel realizations to 1.

3) *Rician Channel:* Rician fading is similar to Rayleigh fading except for the fact that there exists a strong line of sight component along with reflected waves [5]. In the presence of such a path, if the kd is the strength of the LOS component.

The envelope in this case has a Rician density function given by:-

$$f(r) = \frac{r}{\sigma^2} \frac{r^2 + k_d}{2 \sigma^2} I_0 \frac{r k_d}{\sigma^2}, r > 0$$

Where $I_0(\cdot)$ is the 0th order modified Bessel function of the first kind.

III. TURBO CODES

Turbo codes are well known to be one of the error correction techniques which achieve closer results to the Shannon limit. Turbo codes use the parallel concatenated encoding Scheme. Turbo codes, as presented in, are formed by two parallel recursive systematic convolutional encoders separated by an inner interleaver at the transmitter and an iterative.

The parameters used in our design implementation is given in table II.

TABLE II Turbo Codes Specification

Parameter	Value
Turbo codes Rate	1/2
Turbo decoder	Log -MAP
Code generator	K=5; G=(21) ₈ ; F=(37) ₈
Number of iterations	2

A Posteriori Probability (APP) decoder at the receiver. In a concatenated coding scheme, probability of error decreases exponentially while decoding complexity increases algebraically.

A. TURBO ENCODER

The turbo encoder shown in Fig. 2 is formed by concatenation of two constituent codes in parallel and then by separation two codes by an interleaver. The recursive systematic convolutional code is usually used as a constituent code. The information bits are first encoded by a recursive systematic convolutional code, and then, after passing through an interleaver, are encoded by a second systematic convolutional encoder[11]. The code sequences are formed by the information bits, followed by the parity check bits generated by both encoders. In Fig. 2., the encoder takes data sequence d_k an input sequence and puts out three components: d_k , information bits, $x_{1,k}$, parity bit of the first encoder, and $x_{2,k}$, parity bit of the second encoder.

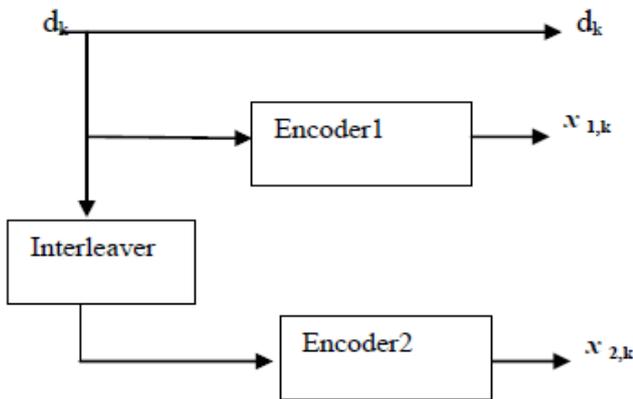


Fig 2.Turbo encoder

B. Turbo Decoder

The input to this turbo decoder is a sequence of received code values, $R_k = \{y_k^s, y_k^p\}$ from the demodulator. The turbo decoder consists of two component decoder – DEC1 to decode sequence from ENC1, and DEC2 to decode sequences from ENC2. Each of these decoders is a Maximum A Posteriori (MAP) decoder. DEC1 takes as its input the received sequence systematic values y_k^s and the received sequence parity values y_k^p belonging to the first encoder ENC1. The output of DEC1 is a sequence of soft estimates EXT1 of the transmitted data bits d_k . EXT1 is called extrinsic data, it does not contain any information which was given to DEC1 by DEC2. This information is interleaved, and then passed to the second decoder DEC2

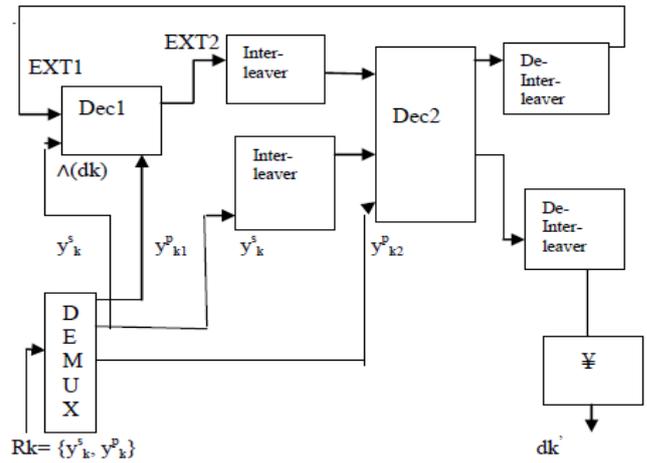


Fig 3.Turbo decoder

DEC2 takes as its input the (interleaved) systematic received values y_k^s and the sequence of received parity values y_k^p from the second encoder ENC2, along with the interleaved form of the extrinsic information EXT1, provided by the first decoder[12].

Here, the iterative decoding process adds greatly to the BER performance of turbo codes. However, after several iterations, the two decoders' estimates of d_k will tend to converge. At this point, DEC2 outputs a value $\Lambda(dk)$; a log likelihood representation of the estimate of d_k .

$\Lambda(dk)$ is de-interleaved so that its sequence coincides with that of the systematic and first parity streams[13]. Then a simple threshold operation is performed on the result, to produce hard decision estimates, dk , for the transmitted bits.

Forward Pass-Calculation of α State Probability

At each node, the current state probability, α is calculated by multiplying the state probability at the previous node $\alpha_{k-1}(m)$ by the branch transition probability, $\gamma_{k-1}(m', m)$, given the received code pair $R_k = \{y_k^s, y_k^p\}$, $\alpha(m)$ is calculated as:

$$\alpha(m) = \frac{\sum_m \sum_{i=1}^1 \gamma_i((y_k^s, y_k^p), m', m) \alpha_{k-1}(m')}{\sum_m \sum_{m'} \sum_{i=1}^1 \gamma_i((y_k^s, y_k^p), m', m) \alpha_{k-1}(m')} \quad (1)$$

Where m is the current state, m' is the previous state and i is the data bit ('0' or '1') corresponding to each branch exiting aNode.

Backward Pass-Calculation of β State Probabilities. Then the trellis is traversed in the reverse direction[14]. Again the probability of each branch being taken is calculated. The current state probability $\beta_k(m)$ is found by multiplying the probability of arriving in the previous state $\beta_{k+1}(m)$ by the probability of taking the current state transition $\beta_{k+1}(m', m)$, given the current received values $R_{k+1} = \{y_{k+1}^s, y_{k+1}^p\}$. This is expressed as follows

$$\beta_k(m) = \frac{\sum_m \sum_{i=1}^1 \gamma_i((y_{k+1}^s, y_{k+1}^p), m', m) \alpha_{k+1}(m')}{\sum_m \sum_{m'} \sum_{i=1}^1 \gamma_i((y_{k+1}^s, y_{k+1}^p), m', m) \alpha_{k+1}(m')} \quad (2)$$

Where the symbol has the same meaning as before, but β is the backward state probability. The transition probability for each branch between nodes is given by the equation

$$\gamma_i((y_k^s, y_k^p), m', m) = p((y_k^s, y_k^p) | d_k = I, m', m)$$

$$q(d_k = I, m', m) \pi(m' | m)$$

c) Calculation of Log Likelihood Probabilities $\Lambda(d_k)$
The soft estimate is represented as a log likelihood ratio (LLR), $\Lambda(d_k)$, is calculated as follows:

$$\Lambda(d_k) = \frac{\ln \sum_m \sum_{m'} \gamma_i((y_k^s, y_k^p), m', m) \alpha_{k-1}(m') \beta_k(m)}{\sum_m \sum_{m'} \gamma_o((y_k^s, y_k^p), m', m) \alpha_{k-1}(m') \beta_k(m)} \quad (3)$$

$\Lambda(d_k)$ represents the probability that the current data bit is a '0' (if $\Lambda(d_k)$ is negative) or a '1' (if $\Lambda(d_k)$ is positive). After a number of iterations, de-interleaved value of $\Lambda(d_k)$ from DEC2 is converted to a hard decision estimate, $\Lambda(d_k)$, of the transmitted data bit. This forms the output of the final turbo decoder stage.

IV. DESIGN IMPLEMENTATION

Here, we propose the system model with OFDM technique and channel equalization under turbo coded encoding.

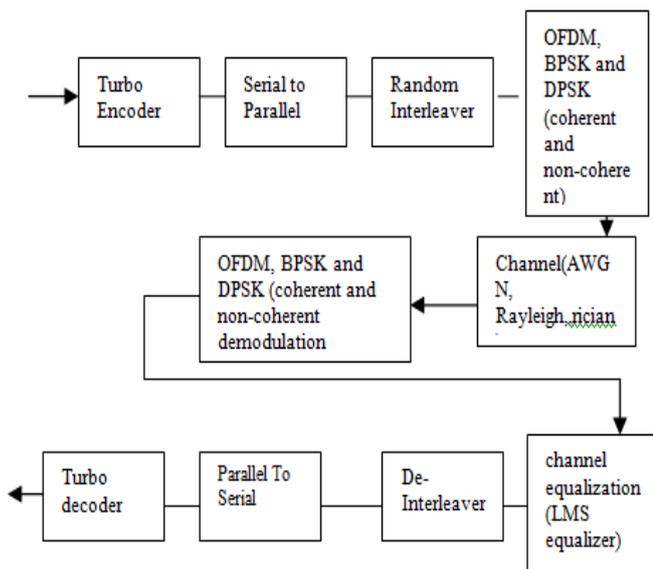


Fig 4. Design implementation

Our aim to optimize and analyze the comparative study of BER for the BPSK and DPSK (coherent and non-coherent) modulation scheme is achieved through our proposed algorithm which can be simulated by the MATLAB tool.

The input data sequence is first encoded by turbo encoder, the outputs of the encoder are fed to the OFDM/ BPSK and DPSK (coherent and non-coherent) modulation system. The transmitted data is fed to the multipath fading channels where first the demodulation process with symbol recovery from OFDM is carried out. The bits then decoded and de-interleaved and received at the output. The BER of the given system is analyzed.

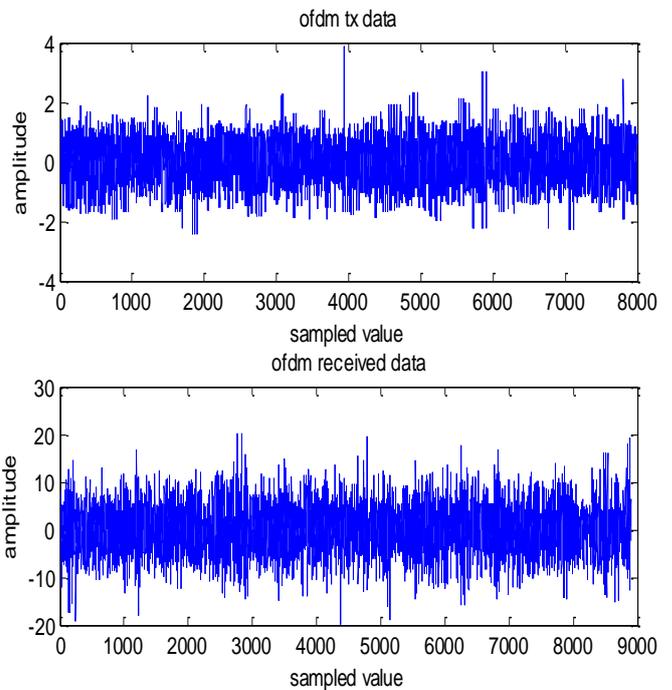


Fig 5. waveforms of transmitted and received OFDM data

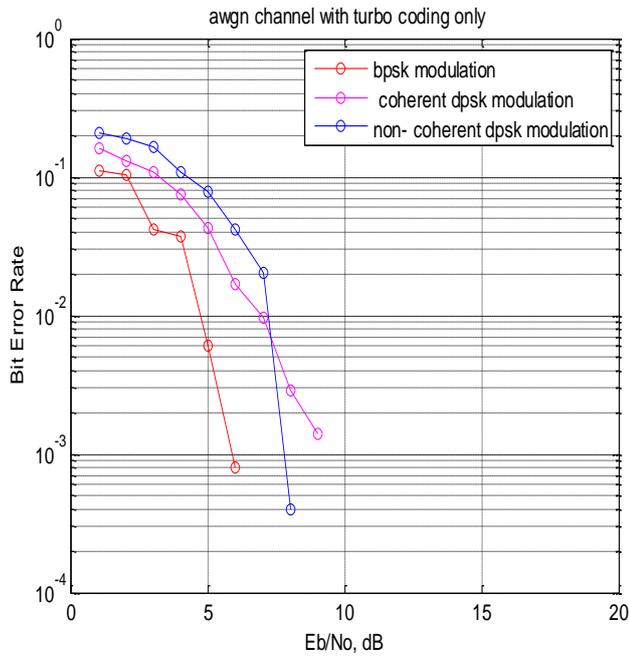
In the fig. waveforms of transmitted OFDM data and received sampled values are plotted.

Simulations of BER analysis for our proposed system given for BPSK and DPSK (coherent and non-coherent) modulations are performed.

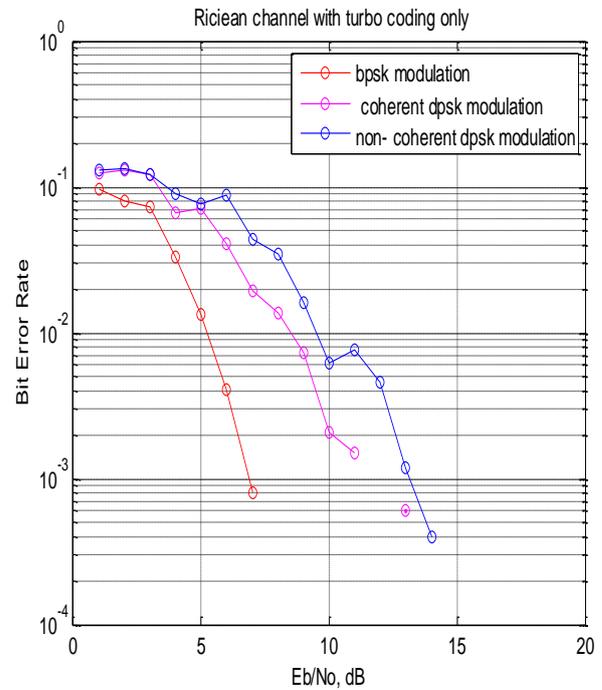
Since OFDM suffers from the problem of ISI. So, in order to combat ISI and ICI we have proposed a cyclic prefix and guard time insertion between OFDM symbols, in which the length of the guard time is made longer than the length of delay spread. The influence of variable delay and multipath could be mitigated through channel coding technique. Further the performance of serial concatenation is overpowered by the parallel concatenation of codes. According to the study, it is known that burst errors can deteriorate the performance of any communication system. The burst errors can happen either by deep frequency fades or by impulsive noise. We have also investigated that in OFDM bit errors occur in burst form rather than independent, and burst errors extensively degrade the performance of the system.

Turbo codes use the parallel concatenated encoding scheme. Hence, turbo codes are used with the system. Turbo codes parameters used are: Code rate : 1/2; SISO Decoder : Log-MAP; Code Generator : K=5; G=(21)₈; F=(37)₈; Interleaver : pseudo random interleaver.

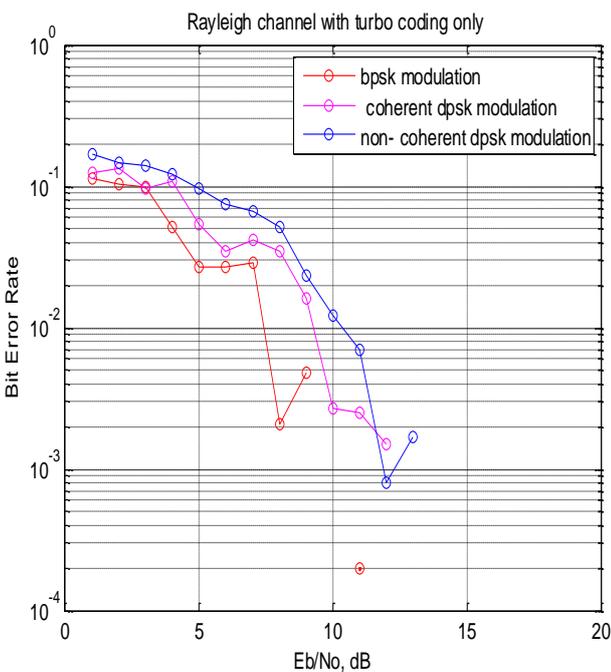
Performance of BER for BPSK and DPSK (Coherent and Non-Coherent) Modulation in Turbo-Coded OFDM with Channel Equalization



(a)



(c)

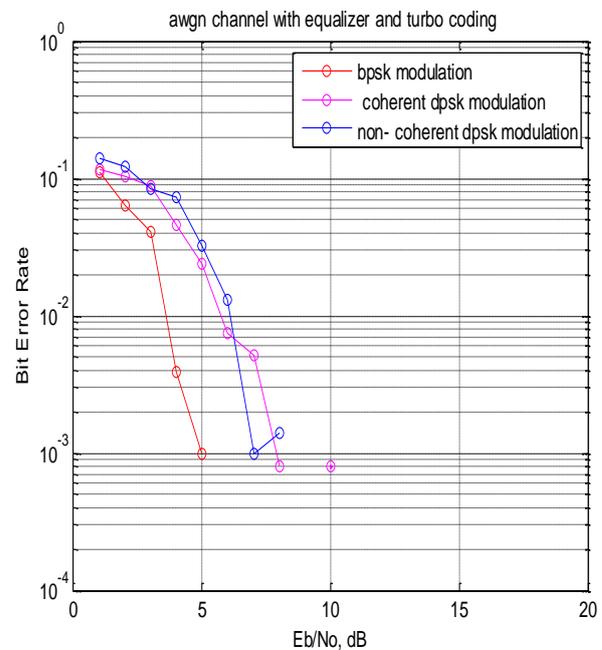


(b)

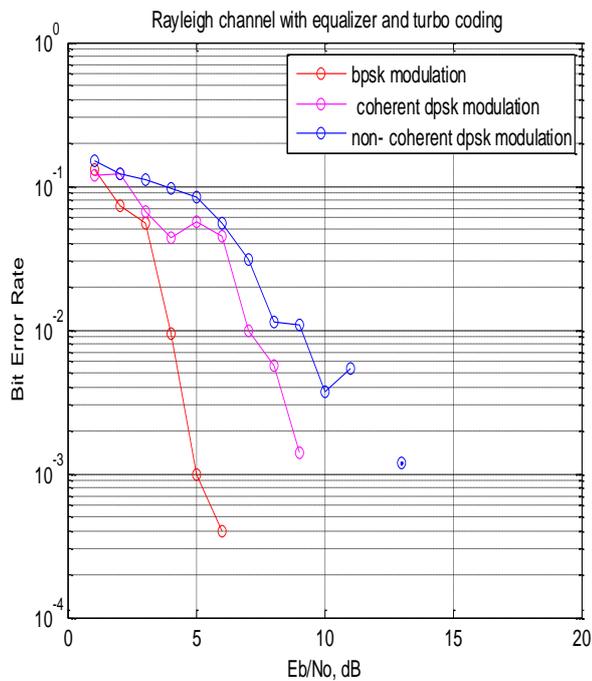
Simulations of the proposed turbo coded system show the BER analysis of the BPSK and DPSK (coherent and non-coherent) modulations. The BER performance of BPSK is better than that of DPSK. Further the analysis show that the BER performance of coherent DPSK is better than that of the Non-coherent DPSK.

Fig 6. BER Comparison of COFDM using turbo codes (a) AWGN (b) Rayleigh (c) rician Fading Environments

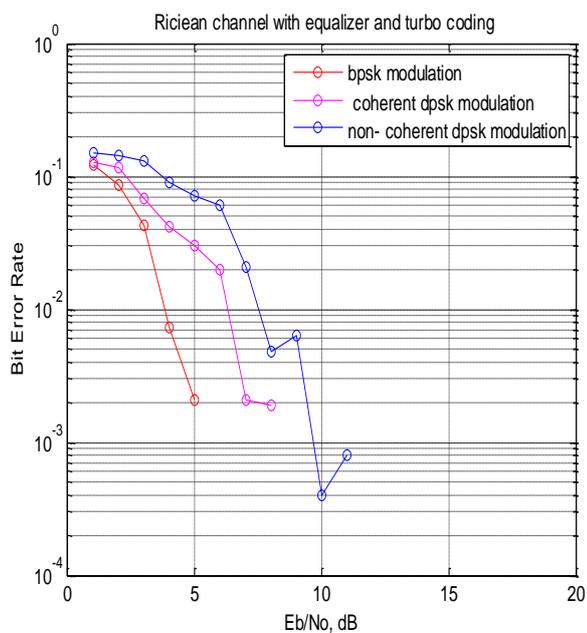
Further improvement in the BER analysis can be obtained through channel equalization via LMS equalizers. Inserting an equalizer realized as an adaptive system, the influence of variable delay and multipath could be mitigated in order to remove or reduce considerably the guard interval and to gain some spectral efficiency. Equalization is to mitigate the effects of ISI to decrease the probability of error that occurs without suppression of ISI.



(a)



(b)



(c)

Fig 7. BER Comparison of COFDM with channel equalization using turbo codes (a) AWGN(b) Rayleigh(c) rician Fading Environments

Further Simulations show more better performance in BER analysis of the BPSK and DPSK(coherent and non-coherent) modulations .The BER performance of BPSK is better than that of DPSK.Further the analysis show that the BER performance of coherent DPSK is better than that of the Non –coherent DPSK[16].The BER analysis results show more decrease in the probability of error of the modulation scheme.

V. CONCLUSION

Hence in our paper through our proposed work BER performance of the modulation scheme has been optimized to

achieve higher data speed and less probability of error for robust and reliable communication .A comparative study of BER for the BPSK and DPSK (coherent and non-coherent) modulation scheme is analysed. The BER performance of BPSK is better than that of DPSK.Further the analysis show that the BER performance of coherent DPSK is better than that of the Non –coherent DPSK.Further results with parallel concatenated encoding scheme i.e turbo codes to combat the burst errors occurring in the OFDM system has been simulated.To further decrease the probability of the OFDM system we use adaptive equalization via lms equalizer to improve the BER performance of the given modulation schemes.Future work can be extended to higher order modulation schemes and with MIMO systems.

REFERENCES

1. Sklar B. Rayleigh fading channels in Mobile digital communication systems Part II: Mitigation, IEEE communication Magazine, 1997, vol.35(9):148-155.
2. Arun Agarwal, S. K. Patra, Senior Member IEEE —Performance prediction of OFDM based Digital Audio Broadcasting system using Channel protection mechanismsl in IEEE journal © 2011.
3. M.X. Chang and Y.T. Su, “Performance Analysis of Equalized OFDM Systems in Rayleigh Fading”, in IEEE Transactions Wireless Communication, vol.I, No. 4, Oct. 2002,pp. 721–732.
4. Jin Goog Kim,Tae Joon and Jong Tae Lim., —Channel estimation for OFDM over Fast Rayleigh Fading Channels,l Proceedings of world Academy of science and technology, vol. 21, pp. 455-458, Jan. 2007.
5. Zhengdao Wang,“OFDM or single carrier block transmission,” IEEE Trans. On comm., vol. 52, no. 3, pp.380-394, mar-2004.
6. Z.Wang and G. B. Giannakis, —Wireless multicarrier communications, IEEE Signal Processing Mag., pp. 29–48, May 2000.
7. L. J. Cimini, Jr., “Analysis and simulation of a digital mobile channel using orthogonal frequency division multiplexing,” *IEEE Trans. Commun.*, vol. COM-33, no. 7, pp. 665–675, Jul. 1985.
8. Armour S., A. Nix, D. Bull, Pre-FFT equalizer design for OFDM, Electronics Letters, vol. 35, Apr. 1999, pp. 539-540.
9. Iliev G., N. Kasabov, Channel equalization using adaptive filtering with averaging, Proc. 5th Joint Conference on Information Sciences, vol. 2, Atlantic City, USA, Mar. 2000, pp. 870-873.
10. Y.-S. Choi, P. J. Voltz, and F. A. Cassara, “On channel estimation and detection for multicarrier signals in fast and selective Rayleigh fading channels,” *IEEE Trans. Commun.*, vol. 49, no. 8, pp. 1375–1387, Aug. 2001.
11. C. Berrou, A. Glavieux, and P. Thitimajshima, —Near Shannon Limit Error-Correcting Coding: Turbo Codesl, Proceedings of the IEEE International Conference on Communications, ICC '93, Geneva., pp. 1064-1070, May 1993.
12. Hanjong Kim,l Performance improvement of Block Turbo Coded OFDM System Using channel state informationl the 23rd international conference on circuits/systems, computers and communications (ITC-CSCC 2008).
13. W. J. Blackert, E. K. Hall, and S. G. Wilson, “Turbo Code Termination and Interleaver Conditions”, *IEE Electronics Letters*, vol. 31, no. 24, pp.2082–2084, Nov 1995.
14. C. Berrou and A. Glavieux, " Near optimum error correcting coding and decoding: Turbo-codes," *IEEE Trans. Commun.*, vol 44, no. 10,Oct, 1996: 126 1- 127 1
15. M. K. Gupta, Vishwas Sharma —To improve BER of turbo coded OFDM channel over noisy channel in Journal of Theoretical and Applied Information Technology © 2005 - 2009 JATIT.
16. Yu Tang, Xiao-lan Lv “Research on the modulation and demodulation of BPSK and BDPSK simulator based on Matlab”Pg no -1239 - 1241 ,IEEE transactions ,2011.
17. Joshi, Alok and Saini, Davinder S, "COFDM performance in various Multipath fading environment ", in proceedings of IEEE International Conference (ICCAE 2010) ,vol.3, pp 127-131,Singapore, Feb 26-28 ,2010.