[*Gocher* et al., 5(3): March, 2018*]
ISSN: 2349-5197
Impact Factor: 3.765

INTERNATIONAL JOURNAL OF RESEARCH SCIENCE & MANAGEMENT

ENHANCEMENT CHANNEL ESTIMATION TECHNIQUES FOR OFDM SYSTEMS WITH REALISTIC INDOOR FADING CHANNELS

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DOI: 10.5281/zenodo.1193966

Keywords: LTE; MIMO; OFDM; cyclic prefix; zero padding; LS; LMMSE; Lr LMMSE.

Abstract
This dissertation deals with the channel estimation techniques for orthogonal frequency division multiplexing (OFDM) systems such as in IEEE 802.11. Although there has been a great amount of research in this area, characterization of typical wireless indoor environments and design of channel estimation schemes that are both robust and practical for such channel conditions have not been thoroughly investigated. It is well known that the minimum mean-square-error (MMSE) estimator provides the best mean-square-error (MSE) performance given a priori knowledge of channel statistics and operating signal- to-noise ratio (SNR). However, the channel statistics are usually unknown and the MMSE estimator has too much computational complexity to be realized in practical systems. In this work, we propose two simple channel estimation techniques: one that is based on mod-ifying the channel correlation matrix from the MMSE estimator and the other one with averaging window based on the LS estimates. We also study the characteristics of several realistic indoor channel models that are of potential use for wireless local area networks (LANs). The first method, namely MMSE-exponential-Rhh, does not depend heavily on the channel statistics and yet offer performance improvement compared to that of the LS estimator. The simulation results also show that the second method, namely averaging window (AW) estimator, provides the best performance at moderate SNR range.

Introduction
Wireless communications is an emerging field. It has grown at a tremendous rate in the last ten to twenty years. It is having lot of ideas for increasing capacity and BER performance**.** All emerged based on wireless technology to provide higher throughput, immense mobility, longer range, robust backbone to thereat. The number of telecommunications innovations grew rapidly during the last half of the 20th century. Currently there is widespread and growing use of cellular phones, cordless phones, digital satellite systems, and personal mobile radio networks. Wireless communications occurs at many different frequencies, from underwater communication at extremely low frequencies .There are various technology comes in the picture to improve the performance of the wireless communication like single carrier communication and multicarrier communication [1].

OFDM is a multicarrier modulation technique used for high data rate in wireless applications that is suitable for eliminating ISI .The main merit of OFDM is the fact that the radio channel is divided into many narrow-band, low-rate, frequency-nonselective sub-channels or subcarriers, so that multiple symbols can be transmitted in parallel, while maintaining a high spectral efficiency. Each subcarrier may deliver information for a different user, resulting in a simple multiple access scheme known as Orthogonal Frequency Division Multiple Access (OFDMA) . This enables different media such as video, graphics, speech, text, or other data to be transmitted within the same radio link, depending on the specific types of services and their Quality-of-Service (QoS) requirements. Orthogonal Frequency Division Multiplexing (OFDM) is a special case of multicarrier transmission, in where a single data stream is transmitted over a number of lower rate subcarriers. In single carrier system if signal get fade or interfered then entire link gets failed where as in multicarrier system only a small percentage of the subcarriers will be affected. The main reason to use OFDM is to increase the robustness against the selective fading or narrowband interference.

Orthogonal Frequency Division Multiplexing (OFDM)

Orthogonal Frequency Division Multiplexing (OFDM) is very similar to the well known and used technique of Frequency Division Multiplexing (FDM). OFDM uses the principles of FDM to allow multiple messages to be sent over a single radio channel. However in a much more controlled manner, is allowing an improved spectral efficiency [2].

A simple example of FDM is the use of different frequencies for each FM (Frequency Modulation) radio stations. The all stations transmit at the same time but do not interfere with each other because they transmit using different carrier frequencies. Additionally they are bandwidth limited and are spaced sufficiently far apart in frequency so that their transmitted signals do not overlap in the frequency domain. This filtered signal can then be demodulated to recover the original transmitted information. At the receiver, each signal is individually received by using a frequency tunable band pass filter to selectively remove all the signals except for the station of interest[3].

OFDM is different from FDM in several ways. In the conventional broadcasting each radio station transmits on a different frequency, it effectively using FDM to maintain a separation between the stations. There is however no coordination or synchronization between each of these stations. With an OFDM transmission such as DAB, information signals from multiple stations are combined into a single multiplexed stream of data. All the subcarriers within the OFDM signal are time and frequency synchronized to each other, allowing the interference between subcarriers to be carefully controlled. This data is then transmitted using an OFDM ensemble that is made up from a dense packing of many subcarriers. These multiple subcarriers overlap in the frequency domain, but do not cause Inter-Carrier Interference (ICI) due to the orthogonal nature of the modulation. Typically with FDM the transmission signals need to have a large frequency guard-band between channels to prevent interference. However with OFDM the orthogonal packing of the subcarriers greatly reduces this guard band, in improving the spectral efficiency. This lowers the overall spectral efficiency [4].

OFDM transmitter system and its parameter

OFDM transmitter consists of following number of sub blocks and can be explained as,

1. Serial to Parallel Converter

The data is considered to be in frequency domain unlike other systems which handle the data in the time domain. This frequency domain high-rate data stream is serial-to-parallel converted into a data block Sk= [Sk[0].. Sk[M-1]] for modulation onto M parallel subcarriers. This increases the symbol duration (Ts) on each subcarrier by a factor of approximately M, in such that it becomes significantly longer than the channel delay spread (τ-max) .

2. SymbolMapping

Symbols are then generated for each parallel stream using phase or amplitude modulation techniques such as QPSK, 16 QAM or 64QAM etc. The M parallel data streams are independently modulated resulting in the complex vector Xk=[Xk[0]..Xk[M-T]]T

Time Domain Conversion of the Data Stream:

The symbols generated (Xk) from each stream are then converted into time domain signal using Inverse Fourier Transform (IFFT) resulting in a set of N complex time domain samples Xk= [Xk[0]..Xk[N-1]]T. (However, in a practical OFDM system, is the number of processed subcarriers is greater than the number of modulated sub-carriers (i.e. N = M), with the un modulated sub-carriers being padded with zeros.)

3. Bit rate and symbol rate

To understand and compare different PSK modulation format efficiencies, the difference between bit rate and symbol rate is important. The signal bandwidth for the communication channel need depends on the symbol rate, not on the bit rate. Bit rate is the frequency of a system bit stream. For the example a radio with an 8 bit sampler, sampling at 10 kHz for voice. The basic bit stream rate in the radio would be eight bits multiplied by 10K samples per second or 80 Kbits per second [6].

Symbol Rate = Bit rate /No. of bits transmitted with each symbol
If one bit is transmitted per symbol with BPSK, and the symbol rate would be the same as the bit rate of 80 Kbits per second. If two bits are transmitted per symbol in QPSK, for the symbol rate would be half of the bit rate or 40 Kbits per second. The symbol rate is sometimes called baud rate. Note these terms are often confused. That baud rate is not the same as bit rate. This is why modulation formats that are more complex and use a higher number of states can send the same information over a narrower piece of the RF spectrum. The bite error rate (BER) of BPSK in AWGN can be calculated as.

$$P_b = Q(\sqrt{\frac{2E_b}{N_O}}) \quad \text{.....................................(3.12)}$$

Where No/2 = noise power spectral density (W/Hz)
E_b= PsTb is the energy contained in a bit duration.
Where Ps = power of sinusoid of amplitude A = ½ A^2
So that $A = \sqrt{2p^s}$
Thus the transmitted signal is either

$$V_{BPSK}(t) = \sqrt{2p^s}\cos(\omega_o t) \quad \text{...........................(3.13)}$$

$$= \sqrt{2p^s}\cos(\omega_o t + t \quad \text{...........................(3.14)}$$

The probability of bit-error for QPSK is the same as for BPSK:

$$P_b - Q\left(\sqrt{\frac{2E_b}{N_b}}\right) \quad \text{...(3.15)}$$

However, with two bits per symbol, the symbol error rate is increased:

$$P_s = 1 - (1 - P_s)2$$

$$P_s = 2Q\left(\sqrt{\frac{2E_b}{N_b}}\right) - Q2\left(\sqrt{\frac{2E_b}{N_b}}\right) \quad \text{..(3.16)}$$

If the signal-to-noise ratio is high (as is necessary for practical QPSK systems)[5].

Additive White Gaussian Noise Channel

AWGN channel is an additive noise channel. Instead of the logical values (0 and 1) which have been used to describe data, we consider real values should be considered to explain the AWGN channel. Assigning {-1,+1} to the binary bit values {1,0}, the channel input is a stream of values from {-1,+1}. The channel output is modeled as adding white Gaussian noise (Gaussian distributed variable with zero mean and non-zero variance) to the input values. The effect of AWGN channel on bits is given in Figure 3.12. The effect of the channel can also be formulated below, where $\tilde{N}$ is a Gaussian random variable, $\left|\tilde{X}\right|$is the channel input, Ŷis the output, and σ^2 is the variance of the Gaussian random variable. The AWGN channel is described by the ratio of the signal power to the noise power, $SNR = \frac{x^2}{\sigma^2}$, with $x_n$as symbol energy. However to compare different coding schemes with different rates, $Eb/N0 = \frac{x_n^2}{2.R\sigma^2}$ is used, where N_0 is the noise spectral density, $E_b$is energy per bit, and *R* is the code rate [28]. In this thesis, the *SNR* value is used for E_b/N_0 value, and the change in the code rate is taken into account while calculating the SNR value. $\hat{Y} = \left|\tilde{X}\right| + \tilde{N}(0, \sigma^2)$

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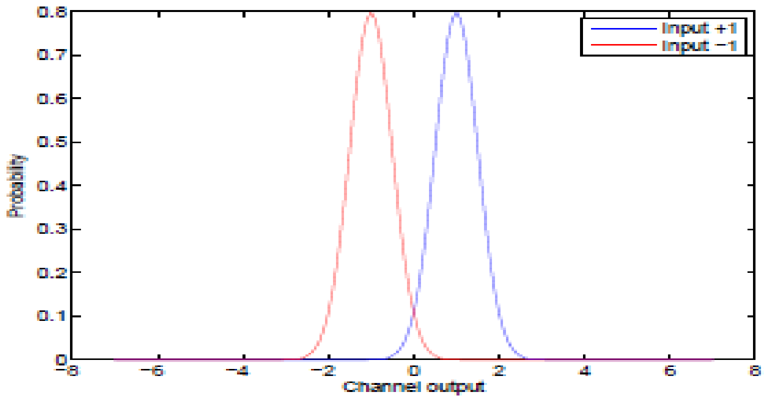

Figure: 4 the probability distribution of AWGN channel output

Flat fading

In a multipath environment, if the difference in the time delay of the number of paths is less than the reciprocal of the transmission bandwidth, the paths cannot be individually resolved. These paths also have random phases. They add up at the receiver according to their relative strengths and phases. The envelope of the received signal is therefore a random variable. This random nature of the received signal envelope is referred to as fading and may be described by different statistical models. These models are described below.

- **Rayleigh Distribution:**

As discussed earlier, the mobile antenna, instead of receiving the signal over one line-of-sight path, receives a number of reflected and scattered waves, as shown in Fig.1. Because of the varying path lengths, the phases are random, and consequently, the instantaneous received power becomes a random variable[6]. In the case of an un-modulated carrier, the transmitted signal at frequency ω_c reaches the receiver via a number of paths, the i^{th} path having an amplitude a_i, and a phase ϕ_i. The received signal $s(t)$ can be expressed as

$$s(t) = \mathrm{Re}\left\{\sum_{i=1}^{N} a_i e^{j(\omega_c t+\phi_i)}\right\} = \sum_{i=1}^{N} a_i \cos(\omega_c t+\phi_i) \quad \ldots\ldots(3.21)$$

where N is the number of paths. The phase ϕ_i depends on the varying path lengths, changing by 2π when the path length changes by a wavelength. Therefore, the phases are uniformly distributed over $[0,2\pi]$.

Effect of motion

Let the i^{th} reflected wave with amplitude a_i and phase ϕ_i arrive at the receiver from an angle ψ_i relative to the direction of motion of the antenna. The Doppler shift of this wave is given by

$$\omega_{di} = \frac{\omega_c v}{c}\cos\psi_i \quad \ldots\ldots(3.22)$$

Where v is the velocity of the mobile, c is the speed of light ($3\mathrm{x}10^8$ m/s), and the ψ_i's are uniformly distributed over $[0,2\pi]$[7].

The received signal s(t) can now be written as

$$s(\mathrm{t}) = \sum_{i=1}^{N} a_i \cos(\omega_c t + \omega_{di} t + \phi_i) \quad \ldots\ldots(3.23)$$

$$s(t) = I(t)\cos\omega_c t - Q(t)\sin\omega_c t \quad \ldots\ldots(3.24)$$

Where the in phase and quadrature components are respectively given as

$$I(t) = \sum_{i=1}^{N} a_i \cos(\omega_{d\,i} t + \phi_i) \quad \text{……………………….. (3.25)}$$

$$Q(t) = \sum_{i=1}^{N} a_i \sin(\omega_{d\,i} t + \phi_i) \quad \text{…………..…. ……………….. (3.26)}$$

Probability density functions of the received signal envelope.

If N is sufficiently large, by virtue of the central limit theorem, the in phase and quadrature components I(t) and Q(t) will be independent Gaussian processes which can be completely characterized by their mean and autocorrelation function. In this case, the means of I(t) and Q(t) are zero. Furthermore, I(t) and Q(t) will have equal variances, σ^2, given by the mean-square value or the mean power. The envelope, r(t), of I(t) and Q(t) is obtained by demodulating the signal s(t) as shown in Fig.2. The received signal envelope is given by

$$r(t) = \sqrt{I^2(t) + Q^2(t)} \quad \text{…………………………… (3.27)}$$

And the phase θ is given by

$$\theta = \arctan\left(\frac{Q(t)}{I(t)}\right) \quad \text{…..........................(3.28)}$$

The probability density function (pdf) of the received signal amplitude (envelope), f(r), can be shown to be Rayleigh [5] given by

$$f(r) = \frac{r}{\sigma^2} \exp\left\{-\frac{r^2}{2\sigma^2}\right\}, \qquad r \geq 0 \text{……………..(3.29)}$$

The cumulative distribution function (cdf) for the envelope is given by

$$F(r) = \int_0^r f(r)dr = 1 - \exp\left\{-\frac{r^2}{2\sigma^2}\right\} \quad \text{………………………… (3.30)}$$

The mean and the variance of the Rayleigh distribution are $\sigma\sqrt{\pi/2}$ and $(2-\pi/2)\sigma^2$, respectively. The phase θ is uniformly distributed over $[0,2\pi]$. The instantaneous power is thus exponential. The Rayleigh distribution is a commonly accepted model for small-scale amplitude fluctuations in the absence of a direct *line-of-sight* (LOS) path, due to its simple theoretical and empirical justifications [8].

- **Rician Distribution:**

The Rician distribution is observed when, in addition to the multipath components, there exists a direct path between the transmitter and the receiver. Such a direct path or line-of-sight component is shown in Fig.1. In the presence of such a path, the transmitted signal can be written as

$$s(t) = \sum_{i=1}^{N-1} a_i \cos(\omega_c t + \omega_{d\,i} t + \phi_i) + k_d \cos(\omega_c t + \omega_d t) \quad \text{………(3.31)}$$

Where the constant k_d is the strength of the direct component, ω_d is the Doppler shift along the LOS path, and ω_{di} are the Doppler shifts along the indirect paths[9].

Probability density function of the received signal envelope

The derivation for the probability density function here is similar to that for the Rayleigh case. If N is sufficiently large, then by virtue of the central limit theorem, the in phase and quadrature components *I(t)* and *Q(t)* are independent Gaussian processes which can be characterized by their mean and autocorrelation function. In the Rician case, the mean values of *I(t)* and *Q(t)* will not be zero because of the presence of the direct component. The envelope *r(t)*, of *I(t)* and *Q(t)* is obtained by demodulating the signal *s(t)*. The envelope in this case has a Rician density function given by [5]

$$f(r) = \frac{r}{\sigma^2}\exp\left\{-\frac{r^2 + k_d^{\ 2}}{2\sigma^2}\right\} I_o\left(\frac{rk_d}{\sigma^2}\right), \qquad r \geq 03 \text{ ....(3.32)}$$

Where $I_0()$ is the zeroth-order modified Bessel function of the first kind. The cumulative distribution of the Rician random variable is given as

$$F(r) = 1 - Q\left(\frac{k_d}{\sigma}, \frac{r}{\sigma}\right), \qquad r \geq 0 \text{ .... (3.33)}$$

Where *Q*(,) is the Marcum's Q function [4,6]. The Rician distribution is often described in terms of the Rician factor *K*, defined as the ratio between the deterministic signal power (from the direct path) and the diffuse signal power (from the indirect paths)[10]. K is usually expressed in decibels as

$$K(dB) = 10\log_{10}\left(\frac{k_d^{\ 2}}{2\sigma^2}\right) \text{ ....(3.34)}$$

In equation *(12)*, if k_d goes to zero (or if $k_d^{\ 2}/2\sigma^2 \ll r^2/2\sigma^2$), the direct path is eliminated and the envelope distribution becomes Rayleigh, with $K(dB) = -\infty$. On the other hand, if the LOS path is much stronger than all the indirect paths combined, r and θ are both approximately normal, with $K(dB) >> 1$. The Rician pdf for different values of K is shown in Fig.3, where K = 0 corresponds to the Rayleigh density function. When the envelope is Rician, the instantaneous power follows a non-central chi-square distribution with two degrees of freedom[11].

The Rician fading channel is constructed using the Rician block in Matlab R2012a using the Rician channel function with different values of Rician factor, which is also known as (k) factor. The Doppler frequency sift is 5.6GHZ. The signal is passed through this channel using the filter function with different SNR values the same is done for AWGN channel. The channel is constructed using AWGN channel function with the same sampling period and here we have taken the snr value 20. The comparison of the different results is done. It can be easily visualized that the variation of the BER is reduced with different values of Doppler sift .The packet loss is reduced and the BER is improved [12].

Table 1 Parameters'

K value	Allow of snr
K=1	0.010
K=10	0.015
K=100	0.045

Table 4 Parameters values.

Total number of subchannels	256
Total number of Pilots	32
Data subchannels	224
Guard interval length	64

Modulation	2
Pilot position interval	8
Channel length	16
Iteration in each evaluation	500

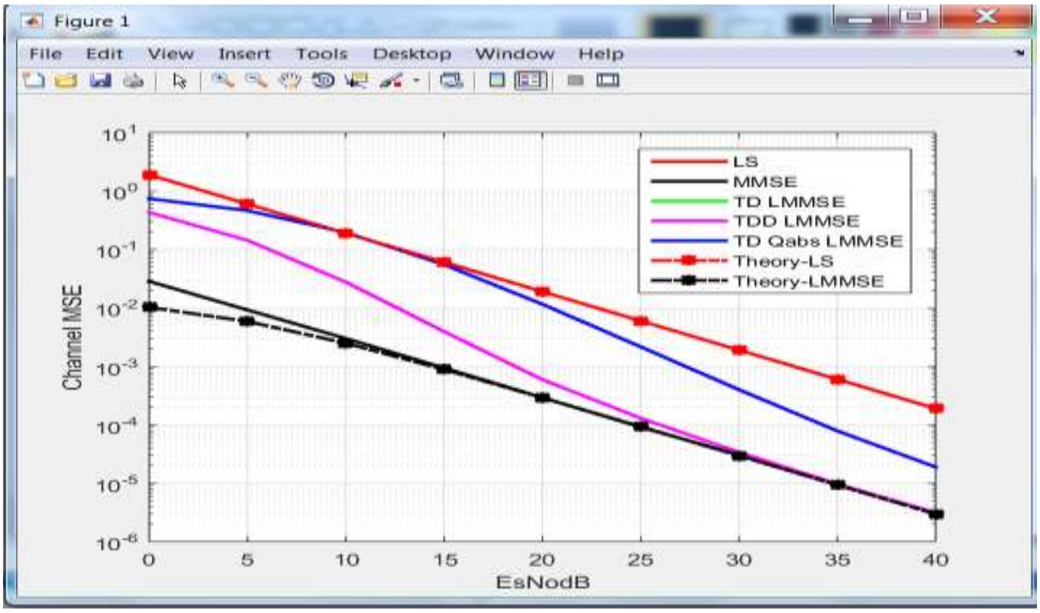

Figure: 5.1Channel MSE and E/N Plot.

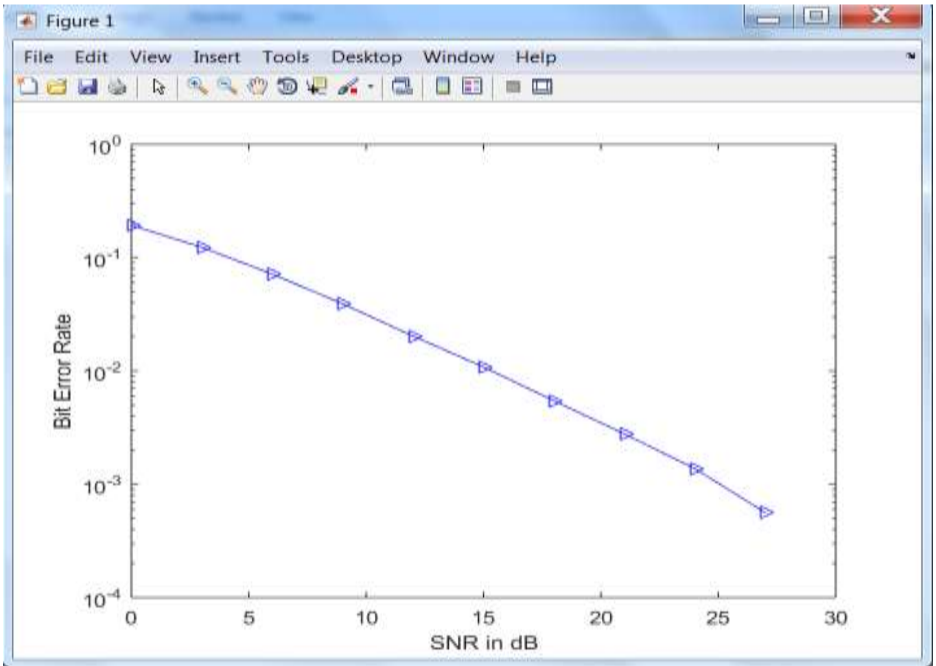

Figure: 5.2 BER and SNR channel error reduction.

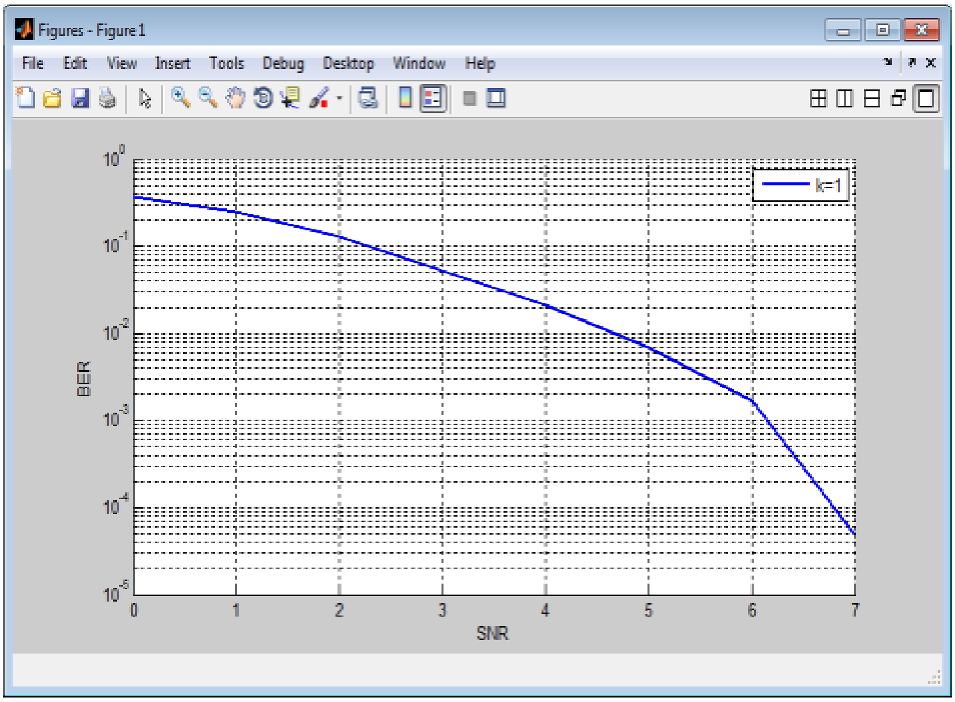

Figure: 5.3 BER Vs SNR using AWGN channel with the value of k factor=20

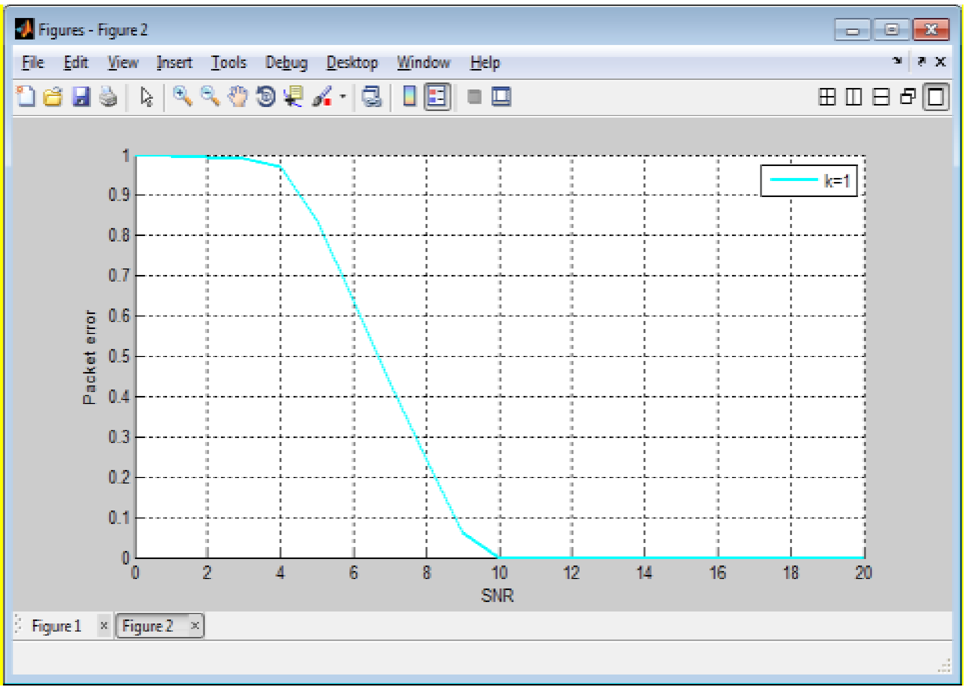

Figure: 5.4 Packet Error Vs SNR using AWGN channel with the value of k factor=20

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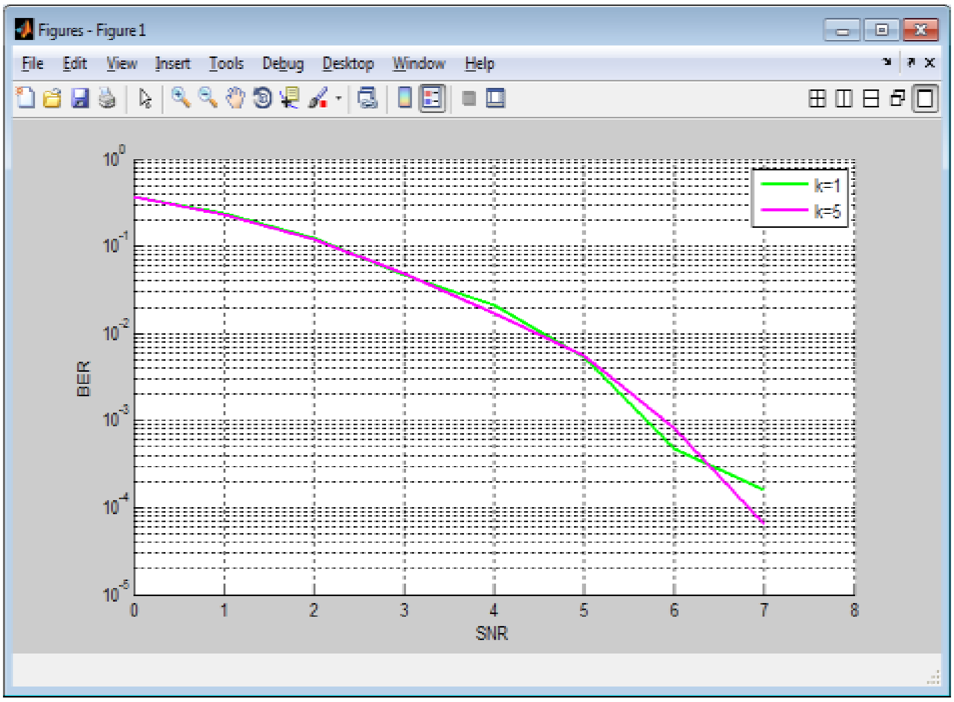

Figure: 5.5 BER Vs SNR using AWGN channel with the value of k factor=50

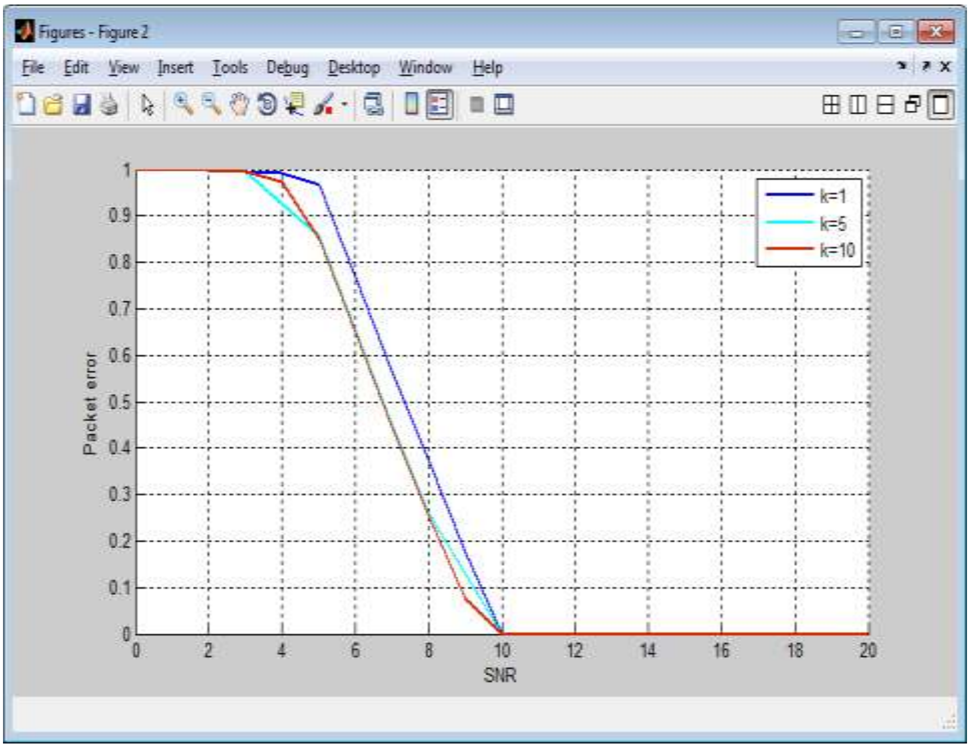

Figure: 5.6 Packet Error Vs SNR using AWGN channel with the value of k factor=50

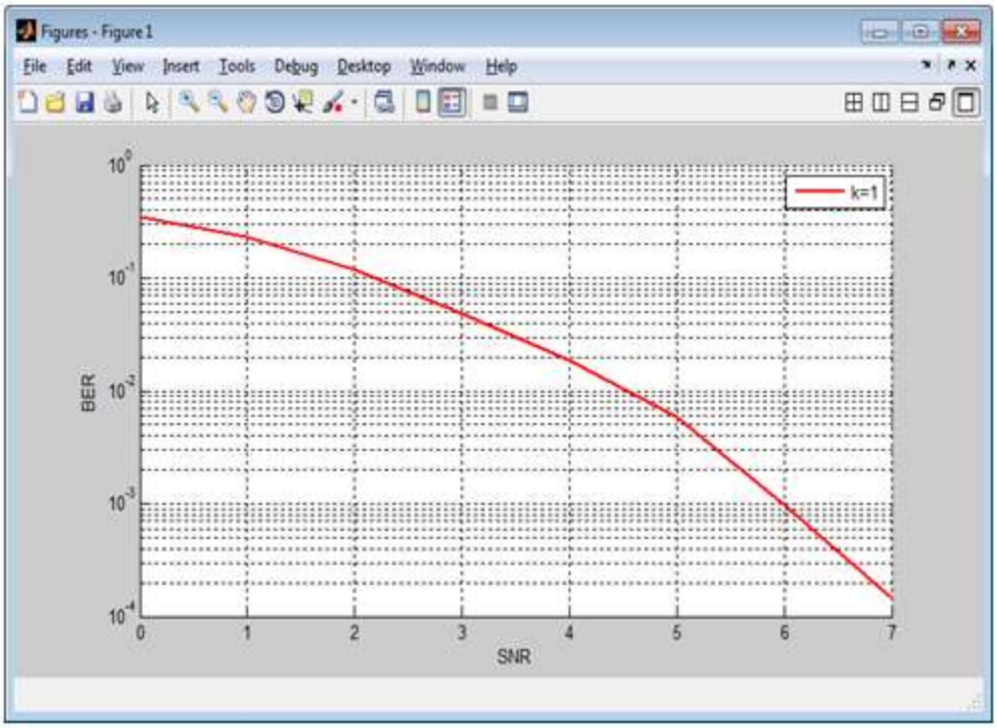

Figure: 5. 7 BER Vs SNR using AWGN channel with the value of k factor=100

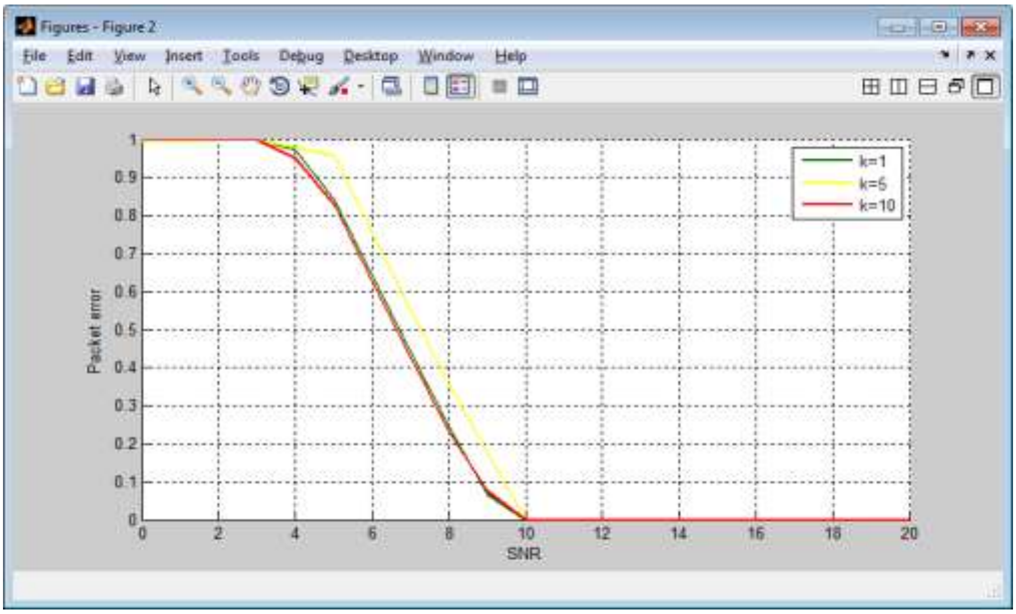

Figure: 5.8 Packet Error Vs SNR using AWGN channel with the value of k factor=100

Conclusion

In the previous work which was discussed in literature review shows that there is a need of improvement in system in terms of Noise level .as we have seen the results of base paper in terms of simulation model (Fig4.19) and the graph of BER Vs E_b/N_o (Fig4.20), our modified work has enhanced the performance of the new designed simulation model. Higher level QAM is implemented by which the noise level is decreased. To compare the performance in terms of BER & Packet error Vs. Doppler shift using different modulation schemes on AWGN Channel and Rician Channel, This system model that is presented in this thesis QAM-OFDM in this system model we adopted OFDM, which is advantageous and so shows the better performance. The simulation results are provided and from which we can evidently conclude that the QAM gives better performance under AWGN and Rician Channel compared to other modulation scheme's and channels. Performance of the system is analyzed under different K factors with different FEC techniques. Simulation results show satisfactory results in terms of better BER values. The throughput and packet error are used to evaluate the performance of the model with different k factor parameter. The performance of MAC Layer is improved with the change in physical layer parameter.

Future Scope
A lot of works can be done for future optimization of Wireless communication especially in OFDM system.

- This work can be extended to increase the performance of the QAM-OFDM system by using the other channel types and other variants of the convolution encoder (2/3, 3/4).
- Furthermore work can be carried by using Power line communication system as the channel. The transfer characteristic performance analysis of the channel is worth being invested.
- Another implementation can be studied by using pilot-based channel. The performances and properties can be tested and verified.
- MIMO is the emerging field these days. MIMO-OFDM has become a research hotpot in the world[13]..

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