

Channel Estimation in OFDM Mobile Wireless Channel Using ZF, ZF-SIC, ZF-SIC-OO, MMSE, MMSE-SIC, MMSE-SIC-OO, LS & ALMMSE Method

Arif Azad

Mtech 4th Sem (Communication System)
Electronics and Communication
SRIST, Jabalpur, India
Email id: arifazad.in@gmail.com

Prof. Urooz Jabeen
Assistant Professor

Electronic and Communication Dept.
SRIST, Jabalpur, India

Prof. Md. Suhel
Assistant Professor

Electronic and Communication Dept.
SRIST, Jabalpur, India

Prof. Amit Pathak
Assistant Professor

Electronic and Communication Dept.
SRIST, Jabalpur, India

Abstract: Multiple Input Multiple Output (MIMO) Orthogonal Frequency Division Multiplexing (OFDM) systems have emerged as wide application technology in wireless communication systems for increasing data rate and the system performance. The effect of fading & interference can be reduced to increase the capacity of the link. MIMO systems uses multiple (input) Transmit and multiple (output) Receive antennas which exploit the multipath propagation in the rich scattering environment. The matrix channel plays very pivotal role in the throughput of a MIMO link since the modulation, data rate, power allocation & the antenna weights are the various dependent parameters on the channel gain. When data rate is been transmitted at high bit rate, the channel impulse response can be extended over many symbol periods which leads to Inter-Symbol Interference (ISI). ISI always causing an issue for signal recovery in wireless communication. In order to reduce complexity of MIMO system, various detection algorithm such as Zero Forcing (ZF), Minimum Mean Square Error (MMSE), Maximum Likelihood (ML) and a Novel algorithm namely ALMMSE is proposed that reduce Bit Error Rate (BER) by using spatial multiplexing. BPSK modulation is taken here as basic for simulation. All the simulations has been done by using MATLAB that shows BER vs. Signal to Noise Ratio (SNR) curve of equalizer exceeds that of ZF, ZF-SIC, ZF-SIC-OO, MMSE, MMSE-SIC, MMSE-SIC-OO, ML, LS and ALMMSE equalizer. In this paper antenna 2x2 configuration is used.

KEYWORDS: MIMO, OFDM, of ZF, ZF-SIC, ZF-SIC-OO, MMSE, MMSE-SIC, MMSE-SIC-OO, ML, LS, ALMMSE, Rayleigh fading channel.

I. INTRODUCTION

Channel estimation is an important technique especially in mobile wireless network systems where the wireless channel changes over time, usually caused by transmitter and/or receiver being in motion at vehicular speed. Mobile wireless communication is adversely affected by the multipath interference resulting from reflections from surroundings, such as hills, buildings and other obstacles. In order to provide reliability and high data rates at the receiver, the system needs an accurate estimate of the time-varying channel. Furthermore, mobile wireless systems are one of the main technologies which used to provide services such as data communication, voice, and video with quality of service (QoS) for both mobile users and nomadic. The knowledge of the impulse response of mobile wireless propagation channels in the estimator is an aid in acquiring important information for testing, designing or planning wireless communication systems.

Channel estimation is based on the training sequence of bits and which is unique for a certain transmitter and which is repeated in every transmitted burst [3]. The channel estimator gives the knowledge on the channel impulse response (CIR) to the detector and it estimates separately the CIR for each burst by exploiting transmitted bits and corresponding received bits. Signal detectors must have knowledge concerning the channel impulse response (CIR) of the radio link with known transmitted sequences, which can be done by a separate channel estimator. The channel estimator is shown in fig 1.

The channel estimation method based on OFDM technique. Orthogonal frequency division multiplexing can accommodate high data rate in the mobile wireless systems in order to handle multimedia services. It is important to understand the OFDM technology because the channel estimation is an integral part of OFDM system.

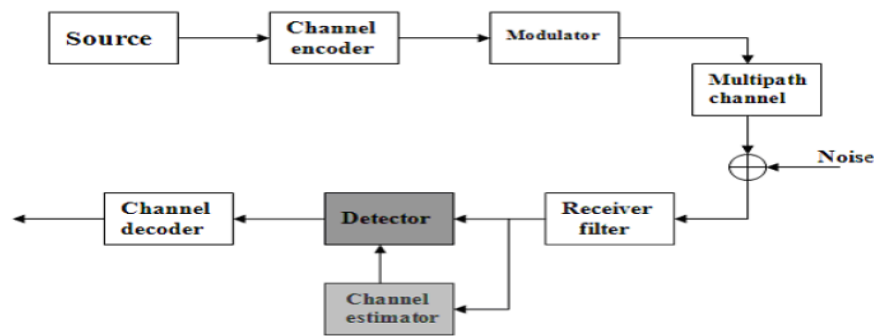


Figure 1. The block diagram of the channel estimator

OFDM technology can be used effectively to avoid the effect of frequency-selective fading and narrowband interference from parallel closely spaced frequencies in mobile networks. One of the desirable features of OFDM is its robustness to the multipath induced intersymbol interference. On the other hand due to the frequency selective fading of the dispersive wireless channel, some sub channels may face deep fades and degrade the overall system performance. In order to compensate the frequency selectivity, techniques such as error correcting code and diversity have to be used [4]-[6].

Doppler shift is caused by the relative motion between the transmitter and receiver. It is considered as Doppler shift in the environment of single-path while Doppler spread in the environment of multipath. For OFDM system, the orthogonality between subcarriers is destroyed by Doppler spread. The greater the relative motion is the stronger the orthogonality is destroyed. Doppler Spread cannot ignore in the scenario [7]. So it is increasingly important to solve the Doppler frequency-spread in high-mobility environment.

Multiple Input Multiple Output (MIMO) systems have been recently emerged as a key technology in wireless communication systems for increasing both data rates and performance. MIMO is an antenna technology that used both in transmitter and receiver equipment for wireless radio communication. MIMO uses multiple antennas to send multiple parallel signals for transmission. Wireless communication technology has shown that when multiple antennas at both transmitter and receiver are employed it provides the possibility of higher data rates compared to single antenna systems [1][2]. MIMO exploits the space dimension to improve wireless system capacity, range and reliability. In the never ending search for increased capacity in a wireless communication channel it has been shown that by using MIMO OFDM systems [3] it is possible to increase that capacity substantially. Especially in broadband application where Inter-symbol Interference is a critical factor. Equalizers are employed to reduce such interference. MIMO system transmits different signals from each transmit element so that the receiving antenna array receives a superposition of all the transmitted signals. MIMO has eminent features which offer significant increment in data throughput and link range without additional bandwidth and increase transmit power. Modern wireless communication standards such as IEEE 802.11n (Wi-Fi), 4G, 3GPP Long Term Evolution (LTE) and WiMAX has become an important part due to these properties [4-5]. Orthogonal frequency division multiplexing (OFDM) reduces receiver complexity in wireless broadband systems.

The use of MIMO technology in combination with OFDM therefore seems an attractive solution for future wireless communication. In OFDM, A Novel detection algorithm for performance analysis of MIMO-OFDM systems using equalizer over a Rayleigh fading channel one user is dedicated to a particular time slot to access all the sub carriers and all carriers are transmitted in parallel with same amplitude i.e. in OFDM, the user device can transmit using the entire carrier at a time. Orthogonal Frequency Division Multiplexing is a multicarrier modulation technique in which FFT space is divided into orthogonal sub-carriers. In this technique, the whole transmission bandwidth is split into many closely spaced narrow sub-channels until overlapped and are transmitted in parallel. Each sub-channel is narrow enough so that it experiences a flat fading although the overall radio propagation environment is frequency selective.

In MIMO wireless communication multipath fading is usual phenomenon that causes ISI from the transmitted signal. To remove ISI [6] from transmitted signal, BER reduction is compulsory. In this paper different equalization approach called Zero Forcing (ZF), Minimum Mean Square Error (MMSE), Maximum Likelihood (ML) and Adaptive Linear Minimum Mean Square Error LMMSE equalizers has been discussed. The channel is a Rayleigh fading channel and the modulation as BPSK has been taken. In this Paper, we used 2x2 perfect codes i.e number of transmit antenna is 2 and number of receiver antenna 2.

II. Orthogonal Frequency Division Multiplexing (OFDM)

Orthogonal frequency division multiplexing can accommodate high data rate in the mobile wireless systems in order to handle multimedia services. It is important to understand the OFDM technology because the channel estimation is an integral part of OFDM system. OFDM technology can be used effectively to avoid the effect of frequency-selective fading and narrowband interference from parallel closely spaced frequencies in mobile networks. If there is no orthogonality in the channel, inter-channel interference (ICI) can be experienced. With these vital advantages, OFDM technology has been widely used by many wireless

standards such as WLAN, WMAN, and DVB [8]. In OFDM scheme, complex filters are not required and time-spreading can be used without any complications in OFDM scheme.

Orthogonal Frequency Division Multiplexing (OFDM) is one of the promising applications, which reduces the multipath fading and makes complex equalizers unnecessary [9]. The concept of using parallel-data transmission and frequency division multiplexing (FDM) was first published in the mid of 1960s. The basic idea was to use parallel data and FDM with overlapping subchannel to avoid the use of high-speed equalization to combat impulsive noise and multipath distortion and fully utilize bandwidth.

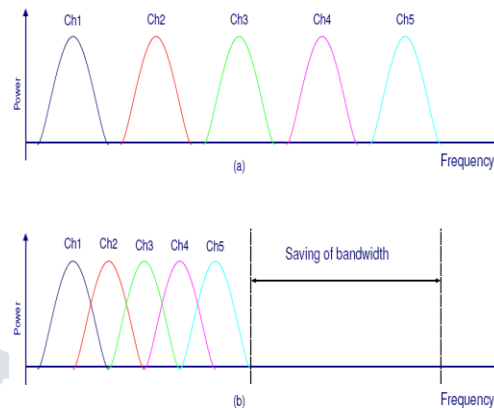


Figure 2 : Concept of OFDM Signal: (a) conventional multicarrier technique (FDM) , and (b) orthogonal frequency division multiplexing technique.

In Figure 2, we can observe the difference between non-overlapping multicarrier modulation technique and overlapping modulation technique. From figure 2(b) it is very clear that by using overlapping modulation technique we can save much more bandwidth than the non-overlapping one [9]. Weinstein and Ebert [10] applied the discrete Fourier transform (DFT) to parallel data transmission system as part of the modulation and demodulation process. In multicarrier transmission, bandwidth divided in many non-overlapping subcarriers but not essential that all subcarriers are orthogonal to each other as shown in figure 3 [9].

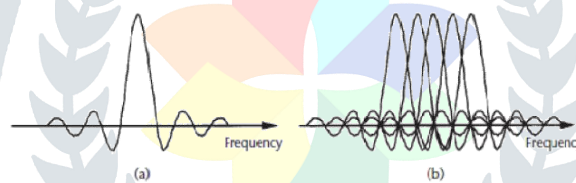


Figure 3: Spectra of (a) an OFDM sub-channel, and (b) an OFDM Signal

Orthogonal frequency-division multiplexing (OFDM) is a key technique in the mobile applications of 3G/4G system due to its capability of high rate transmission and robustness to inter-symbol-interference (ISI). It could combat the frequency selective fading effectively because of narrow bandwidth of each subcarrier. Furthermore, its implementation is low-complexity because the signal processing architectures need only IFFT, FFT and simple frequency-domain equalization. However, these advantages no longer exist when the channel response varies rapidly in time-domain because of the mobility of user equipment.

III. Channel and Signal Model

The method of channel estimation in which the Pilot Sequence are added after the signal is modulated and before the signal is transmitted. At the receiver pilot sequences are removed and the signal is extracted from the received signal.

Step 1 For a transmitted signal the $X(n)$ is generated which shows the number of sub carrier is generated.

$$X(n) = \frac{1}{N} \sum_{i=0}^{N-1} d_i e^{j\frac{2\pi}{N}in}$$

Where: N= Number of sub-carriers.

Step 2 The signal is transmitted and the Channel impulse response is:

$$h(n) = \frac{1}{\sqrt{N_P}} \sum_{p=0}^{N_P-1} \alpha_p e^{j\frac{2\pi}{N}f_{D_P}n} \delta(n-\tau_p)$$

Where, α_p , f_{D_P} and τ_p are the complex amplitude, normalized Doppler shift and time delay for the p th multipath arrival respectively.

Step 3 The signal is received at the receiver is recovered by performing the inverse of the transmitted signal & performs the LS & LMMSE method.

The signal received by the convolution of the input signal and channel impulse response is:

$$y(n) = \frac{1}{N\sqrt{N_P}} \sum_{p=0}^{N_P-1} \alpha_p e^{j\frac{2\pi}{N}fD_p^n} \sum_{i=0}^{N-1} d_i e^{j\frac{2\pi}{N}i(n-\tau_p)} + w(n) \quad 3.3$$

Where, $w(n)$ is additive white Gaussian noise (AWGN).

Method Used For Training Based Channel Estimation

Usually the Channel Impulse Response (CIR) is estimated based on the known training sequence or pilot sequences, which is transmitted over the channel in every transmission burst. The receiver can utilise the known training bits and the corresponding received samples for estimating CIR typically for each burst separately. There are a few different approaches of channel estimation. In this work the many channel estimation method used for the estimation of the error or distortion occurs in the transmitted signal. A modified and improved Linear Minimum Mean Square Error (LMMSE) and Least Square (LS), Zero Forcing Estimation with ML (Maximal Likelihood) & MRC (Maximal Ratio Combining) methods are used.

Usually the pilot/ training based channel estimation is based on known sequence of bits, which is unique for a certain transmitter and which is repeated in every transmission burst. Thus, the channel estimator is able to estimate CIR for each burst separately by exploiting the known transmitted bits and the corresponding received samples. Detectors comprising of linear detectors (LD) and non-linear detectors (NLD). Linear detectors (LD) consists of zero forcing (ZF) & Minimum mean square error (MMSE) and Non-linear detectors (NLD) consists of Zero forcing-successive Interference cancellation (ZF-SIC) & Minimum mean square error-successive Interference cancellation (MMSE-SIC) [7].

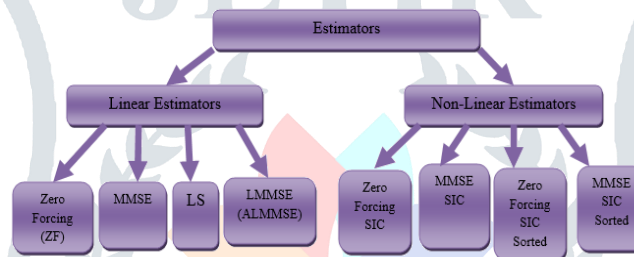


Figure: 4 Classification of Various Estimation Techniques

a) Zero Forcing

Zero Forcing (ZF) is the form of linear equalization algorithm used in communication systems which applies the inverse of the frequency response to the received signal, to restore signal after the channel. It has many useful applications. In ZF knowing the channel allows recovery of the two or more streams which will be received on top of each other on each antenna. The name Zero Forcing corresponds to bringing down the intersymbol interference (ISI) to zero in the noise free case. This is very much useful when ISI is significant compared to noise. For a channel with frequency response $F(f)$ the zero forcing equalizer $C(f)$ is constructed by $C(f) = 1/F(f)$.

The combination of channel and equalizer gives a flat frequency response and linear phase $F(f)C(f) = 1$.

Actually zero-forcing equalization does not work in most the applications, for the following reasons:

1. Even though the channel impulse response has finite length, the impulse response of the equalizer needs to be infinitely long.
2. At some frequencies the received signal may be weak. To compensate, the magnitude of the zero-forcing filter ("gain") grows very large. As a consequence, any noise added after the channel gets boosted by a large factor and destroys the overall signal-to-noise ratio. Furthermore, the channel may have zeroes in its frequency response that cannot be inverted at all. (Gain $\rightarrow 0$ still equals 0).

This second term is many a times more limiting condition. These problems are addressed in the LMMSE equalizer by making a small modification to the denominator of $C(f)$:

$$C(f) = 1/(F(f) + k),$$

Where: k is related to the channel response and the signal SNR.

If the channel response (or channel transfer function) for a particular channel is $H(s)$ then the input signal is multiplied by the reciprocal of it. This is intended to remove the effect of channel from the received signal, in particular the intersymbol interference (ISI).

The ZF removes all ISI, and is perfect when the channel is noiseless. However, when the channel is noisy, the ZF will amplify the noise greatly at frequencies f where channel response $H(j2\pi f)$ has a small magnitude (near zeroes of the channel) to attempt to

invert the channel completely. A more balanced linear equalizer in this case is the minimum mean-square error equalizer, which does not usually eliminate ISI completely but instead minimizes the total power of the noise and ISI components in the output.

b) **Successive Interference Cancellation with optimal ordering:** In Successive Interference Cancellation, receiver arbitrarily takes one of the estimated symbols, and subtract its effect from the received symbol y_1 3.7 Also we can subtract the effect of \hat{x}_1 first or \hat{x}_2 first. To make that decision, we have to find out the transmit symbols (after multiplication with the channel) which came at higher power at the receiver. The received power at the both the antennas corresponding to the transmitted symbol x_1 is, $P_{x_1} = |h_{1,1}|^2 + |h_{2,1}|^2$.

The received power at the both the antennas corresponding to the transmitted symbol x_2 is,

$$P_{x_2} = |h_{1,2}|^2 + |h_{2,2}|^2.$$

If $P_{x_1} > P_{x_2}$ then the receiver decides to remove the effect of \hat{x}_1 from the received vector y_1 3.8 and then re-estimate \hat{x}_2 .

In matrix notation, $r = hx_2 + n$

c) Minimum Mean Square Error (MMSE)

The MMSE estimator employs the second-order statistics of the channel conditions to minimize the mean-square error. Denote by R_{gg} , R_{HH} , R_{yy} and the auto-covariance matrix of g , H and y respectively, and by R_{gy} the cross covariance matrix between g & y . Also denote by σ_N^2 the noise variance $E |N|^2$. Assume the channel vector g and the noise are N uncorrelated, it is derived that:

$$R_{HH} = E \bar{H} H^H = E (Fg)^H = F R_{gg} F^H$$

$$\hat{H}_{MMSE} = \hat{g}_{MMSE} = F [(F^H X^H)^{-1} R_{gg}^{-1} \sigma_N^2 + X F]^{-1} \bar{Y}$$

$$\hat{H}_{MMSE} = [R_{HH} + \sigma_N^2 X X^H]^{-1} \hat{H}_{LS}$$

The MMSE estimator yields much better performance than LS estimators, especially under the low SNR scenarios. A major drawback of the MMSE estimator is its high computational complexity, especially if matrix inversions are needed each time the data in changes. 3.12 3.11

d) Least Square (LS)

Least square estimation is used to minimize the square distance between the received signal and the transmitted signal. The least square estimates (LS) of the channel at the pilot subcarriers given by the following equation: $H_P^{LS} = (X_P^H X_P)^{-1} X_P^H y_P$ 3.13

Channel estimation is based on standard LS techniques. We can write the transmitted and the received signals in vector form as:

$$r_n = [r_{n,0}, r_{n,1}, \dots, r_{n,K-1}]$$

$$s_n = [s_{n,0}, s_{n,1}, \dots, s_{n,K-1}]$$
3.14

Where r_n and s_n are the vectors containing samples and respectively for $k = 0, 1, \dots, K-1$, and K is the total number of sub-carriers in an OFDM symbol. By simply $r_{n,k} / s_{n,k}$, dividing and we get the frequency response of the channel plus some noise. In this way, we can express the estimated channel frequency response by:

$$\hat{H}_{n,k} = \frac{r_{n,k}}{s_{n,k}}, \text{ for } k = 0, 1, \dots, K-1$$

Since the transmitted signal is BPSK with unit magnitude is

$$\frac{1}{s_{n,k}} = s_{n,k}^*$$
3.16

We can rewrite the above Equation as:

$$\hat{H}_{n,k} = r_{n,k} s_{n,k}^*, \text{ for } k = 0, 1, \dots, K-1$$
3.17

While implementing this estimation technique, the frequency responses of the channels corresponding to each OFDM blocks and sub-carriers are assumed to be independent of each other. Consequently, none of the correlation properties of the channel is used and the estimation is based on a Gaussian channel model. The block diagram of the LS channel estimator is shown in Figure 5.

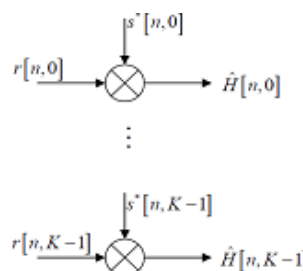


Figure 5: LS Channel Estimator

The advantage of LS algorithm is its simplicity, because no consideration of noise and ICI. So, without using any knowledge of the statistics of the channels, the LS estimators are calculated with very low complexity, but obviously it suffers from a high MSE. LS method, in general, is utilized to get initial channel estimates at the pilot subcarriers, which are then further improved via different methods.

e) **Adaptive Linear Mean Minimum Square Error (ALMMSE)**

The LMMSE channel estimator is designed to minimize the estimation MSE. The LMMSE estimate of the channel responses given as:

$$\hat{h}_{lmmse} = R_{hh} R_{h_{ls} h_{ls}}^{-1} \hat{h}_{ls} = R_{hh} (R_{hh} + \sigma_n^2 (XX^H)^{-1})^{-1} \hat{h}_{ls}$$

Where:

$$\hat{h}_{ls} = X^{-1} y = \begin{bmatrix} y_0 & y_1 & \dots & y_{N-1} \\ x_0 & x_1 & \dots & x_{N-1} \end{bmatrix}^T$$

is the least-squares (LS) estimate of h , σ_n^2 is the variance of the additive channel noise, and the covariance matrices are:

$$R_{hh} = E\{hh^H\}$$

$$R_{h_{ls} h_{ls}} = E\{\hat{h}_{ls} \hat{h}_{ls}^H\} \quad R_{h_{ls} h_{ls}} = E\{\hat{h}_{ls} \hat{h}_{ls}^H\}$$

The covariance matrices are:

$$R_{HHp} = E\{hh^H\}$$

The LMMSE estimator (12) is of considerable complexity, since a matrix inversion is needed every time the training data in X changes. We reduce the complexity of this estimator by averaging over the transmitted data [10], i.e. we replace the term $(XX^H)^{-1}$ in (Eq.) with its

expectation $E(XX^H)^{-1}$. Assuming the same signal constellation on all tones and equal probability on all constellation points, we get $E(XX^H)^{-1} = (1/x_k^2)2I$, here I is the identity matrix. Defining the average signal-to-noise ratio as:

$$SNR = E|x_k^2|/\sigma_n^2$$

We obtain a simplified estimator as:

$$H_P^{LMMSE} = R_{hh} \left(R_{hh} + \frac{\beta}{SNR} I_P \right)^{-1} H_P^{LS}$$

$$\beta = E|x_k^2|E|1/x_k|^2$$

Where, β = Scaling Factor and given as:

I_P = Identity Matrix

R_{HHp} represent the cross-correlation matrix between all sub-carriers and the subcarriers with reference signals. $R_{H_p H_p}$ represent the auto-correlation matrix of the sub-carriers with reference signals. The high complexity of LMMSE estimator is due to the inverse matrix. Every time data changes, inversion is needed. The complexity of this estimator can be reduced by averaging the transmitted data. Then, the estimation can be done more accurately.

The performance of LMMSE estimator is much better than LS estimator, especially under the lower E_b/N_0 and ALMMSE estimator could gain 10-15 dB more of performance than LS [2]. However, because of the required matrix inversions, the computation is very complex when the number of subcarriers of OFDM system increases. Therefore, an important drawback of the ALMMSE estimator can be the high computational complexity.

f) **Maximal-Ratio Combining (MRC)**

MRC is the method of combining diversity in as follows:

1. The signals from each channel are added together,
2. The gain of each channel is made proportional to the RMS signal level and inversely proportional to the mean square noise level in that channel.
3. Different proportionality constants are used for each channel.

It is also known as ratio-squared combining and pre-detection combining. Maximal-ratio combining is the optimum combiner for independent AWGN channels. MRC can restore a signal to its original shape.

In MRC the equalized symbol is,

$$\hat{x}_2 = \frac{h^H r}{h^H h} \quad 3.26$$

Else if $P_{x1} \leq P_{x2}$ the receiver decides to subtract effect of \hat{x}_2 from the received vector y_1 and y_2 , and then re-estimate \hat{x}_1 .

In matrix notation, $r = hx_1 + n$

Optimal way of combining the information from multiple copies of the received symbols in receive diversity case is to

$$\hat{x}_1 = \frac{\mathbf{h}^H \mathbf{r}}{\mathbf{h}^H \mathbf{h}} \quad 3.28$$

apply Maximal Ratio Combining (MRC). The equalized symbol is,

II. SYSTEM DESIGN OF MIMO-OFDM SPATIAL MULTIPLEXING

The channel estimation system design for a 2x2 MIMO-OFDM Spatial Multiplexing in the frequency domain with 128 subcarriers, so that the model does not need the IFFT/ FFT. Channel estimation system design, in detail, is shown in Figure 6.

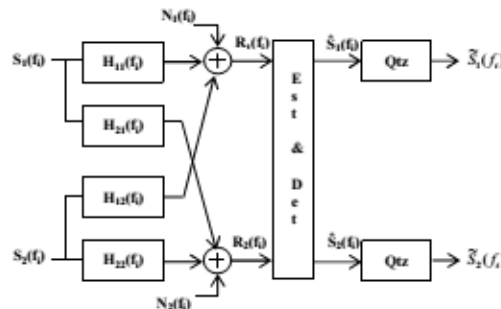


Figure 6: Design of a 2x2 MIMO-OFDM models in Frequency Domain

Flow Chart of Transmitted Signal

The flow chart for the transmitted signal is shown in figure 7. in which the generation of the transmitted signal is shown.

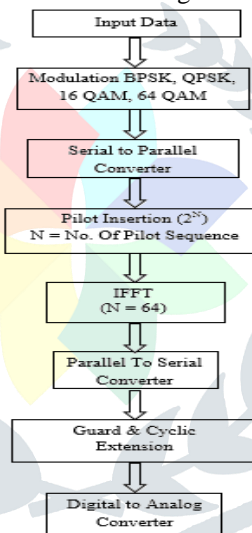


Figure 7: Flow chart for the transmitted signal

Flow Chart of Received Signal: Flow chart for the received signal is shown in the figure 8, when the transmitted signal is received it containing some amount of noise with the received signal. The AWGN (Additive White Gaussian Noise) is inserted in the signal before the signal is received. The received signal gets from the convolution of the transmitted signal and the Channel Impulse Response (CIR). The channel estimator also gives the knowledge of the channel impulse response (CIR) to the detector; it estimates separately the CIR the each burst exploiting transmitted bits corresponding receiver bits. Then, receive signal is converted from analog to digital form and then remove the guard inserted and cyclic extension from the signal. And next the signal are converted from the serial to parallel and Fast Fourier Transform (FFT) of order $N = 64$ is performed and then the LS and LMMSE method is used for the channel estimation and again the parallel signal is converted from parallel to serial form. Apply the de-modulation and finally get the data at the output.

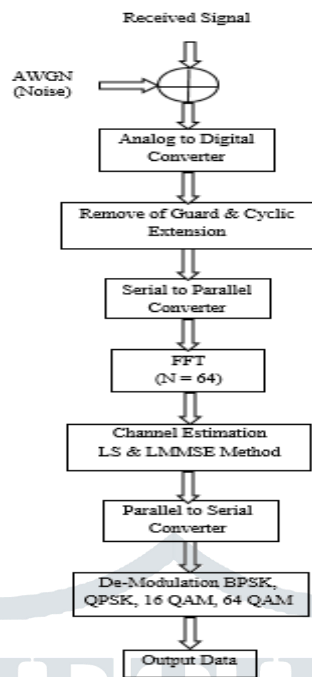


Figure 8: Flow chart for the received signal

IV. SIMULATION RESULTS

The MATLAB/SIMULINK Communication tool box is used for implement the proposed model. The AWGN (Additive White Gaussian Noise) is inserted in the signal before the signal is received. The received signal gets from the convolution of the transmitted signal and the Channel Impulse Response (CIR). The channel estimator also gives the knowledge of the channel impulse response (CIR) to the detector; it estimates separately the CIR the each burst exploiting transmitted bits corresponding receiver bits. Then, receive signal is converted from analog to digital form and then remove the guard inserted and cyclic extension from the signal. And next the signal are converted from the serial to parallel and Fast Fourier Transform (FFT) of order $N = 64$ is performed and then the ZF, ZF-SIC, ZF-SIC-OO, LS & ALMMSE methods are used for the channel estimation and again the parallel signal is converted from parallel to serial form. Apply the de-modulation and finally get the data at the output. BER for a 2x2 OFDM system using above methods have been calculated.

a) Using ML (Maximum Likelihood) Algorithm

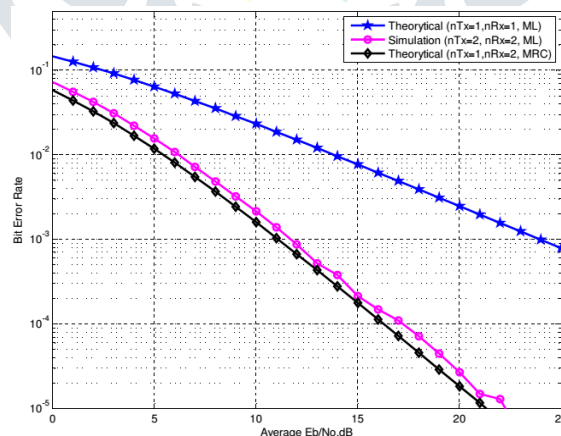


Figure :9 Graph between BER and SNR using ML Equalizer.

b) Using ZF (Zero Forcing) Algorithm

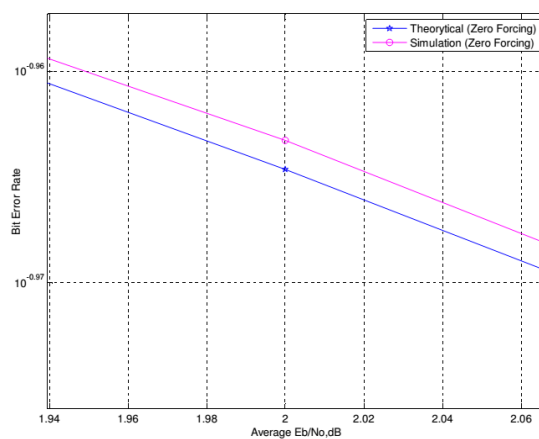


Figure :10 BER and SNR using ZF Estimator

c) Using ZF-SIC (Zero Forcing) Successive Interference Cancellation

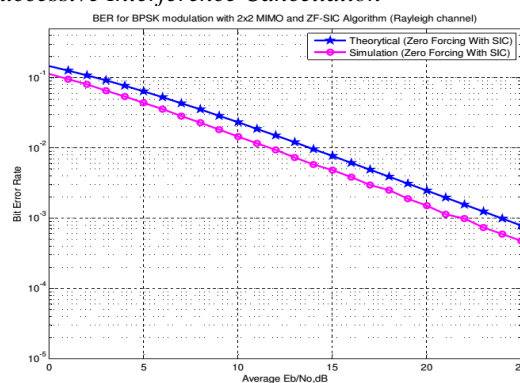


Figure :11 BER for ZF-SIC Algorithm (Rayleigh channel)

d) Using ZF-SIC (Zero Forcing) Successive Interference Cancellation with Optimal Ordering

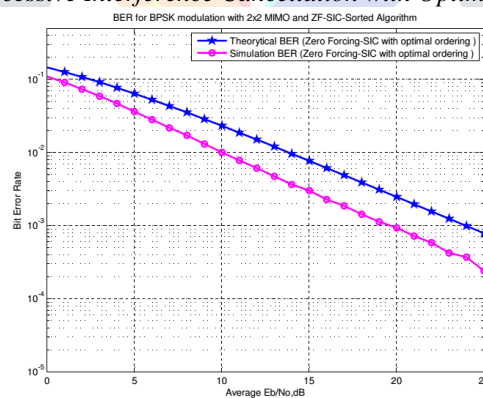


Figure :12 BER for ZF-SIC-Optimum Ordering Algorithm

e) Using Minimum Mean Square Error (MMSE) Estimator

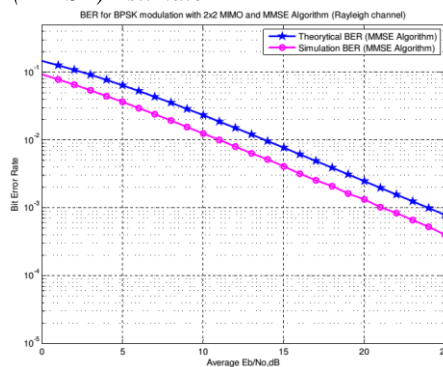


Figure :13 BER for MMSE Algorithm

f) Using Minimum Mean Square Error with Successive Interference Cancellation (MMSE-SIC) Estimator

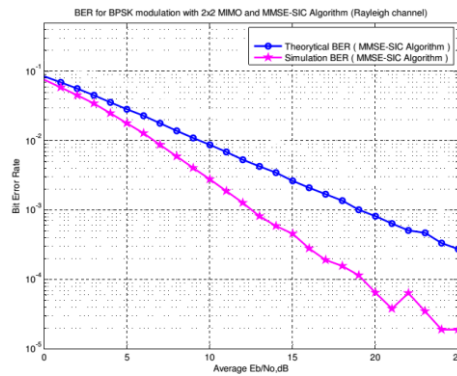


Figure :14 BER for MMSE-SIC Algorithm

g) Using LS (Least-Square) & Adaptive Linear Mean Minimum Square Error (ALMMSE) Estimator

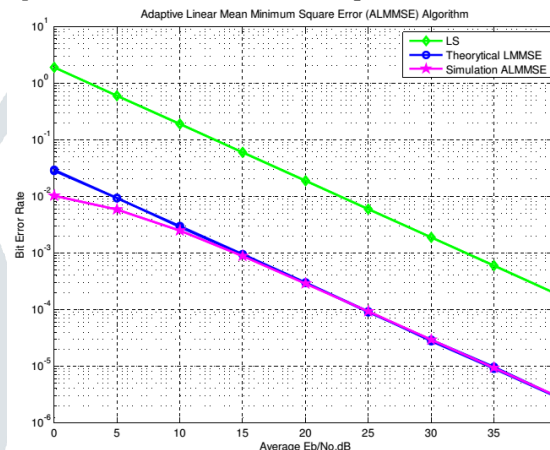


Figure :15 BER for Adaptive Linear Mean Minimum Square Error (ALMMSE) Algorithm

V. RESULTS & COMPARISON

Table 1 shows the comparison of the Comparison of various Reference papers methods with proposed ALMMSE method.

SNR (dB)	BER (dB)								
	ZF [2]	ZF-SIC [2]	ZF-SIC-OO [2]	MMSE [1]	MMSE-SIC [3]	MMSE-SIC-SORT [4]	LC-LMMSE Method [1]	LS Method [4]	Proposed ALMMSE Method
0	0.1461	0.1134	0.1107	0.0923	0.0852	0.0758	0.1259	0.5012	0.0100
2	0.1263	0.0956	0.0915	0.0782	0.0694	0.0592	0.1000	0.3162	0.0050
4	0.0914	0.0657	0.0592	0.0538	0.0456	0.0343	0.0631	0.1995	0.0032
6	0.0642	0.0440	0.0369	0.0362	0.0286	0.0176	0.0316	0.1585	0.0020
8	0.0434	0.0283	0.0224	0.0240	0.0180	0.0087	0.0158	0.1250	0.0013
10	0.0285	0.0183	0.0132	0.0156	0.0108	0.0041	0.0079	0.1023	0.0010
12	0.0189	0.0118	0.0092	0.0100	0.0069	0.0019	0.0070	0.1000	0.0006
14	0.0121	0.0074	0.0058	0.0065	0.0044	0.0009	0.00019	0.0398	0.0002
16	0.0076	0.0047	0.0037	0.0040	0.0027	0.0004	0.00003	0.0063	0.0001
18	0.0049	0.0030	0.0023	0.0027	0.0017	0.0002	0.00002	0.0041	0.00007

Table 1 Comparison of various Reference papers methods with proposed aLMMSE method

VI. CONCLUSION

The channel estimation method, the OFDM transmission technique has emerged as a promising for high data rate and reliable transmission for wireless communication systems. For that the different modulation technique BPSK is used in 2x2 MIMO OFDM, and for the estimation at the receiver the pilot insertion techniques such as ZF (Zero Forcing), Zero Forcing with Successive Interference Cancellation (ZF-SIC), Zero Forcing with Successive Interference Cancellation with Optimum Ordering / Sorted (ZF-SIC-OO), LS (Least Square) and ALMMSE (Adaptive Linear Minimum Mean Square Error) methods are used. BER results for ALMMSE shows better performance vs all other methods. In all the techniques used here, at receiver the estimation is done by using the Pilot Sequences. Although ZF removes all ISI, and is perfect when the channel is noiseless. However, when the channel is noisy,

the ZF will amplify the noise greatly at frequencies f where channel response has a smaller magnitude (near zeroes of the channel) to attempt to invert the channel completely. Due to channel interference, the accuracy of channel estimation is an essential factor for a good receiver. Least Square method reduces BER but the improved ALMMSE is showing best optimized performance of the system for BER as well as SNR as compare theory. At the 10 dB of BER & SNR get the 98% of accuracy which is the main objective.

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