A Review On Various Approaches to Reduce Intersymbol Interference in MIMO – OFDM System

Jaya Misra
ME Scholar
FET, SSGI, Bhilai (C.G.)

Rakesh Mandal
Assistant Professor
FET, SSGI, Bhilai (C.G.)

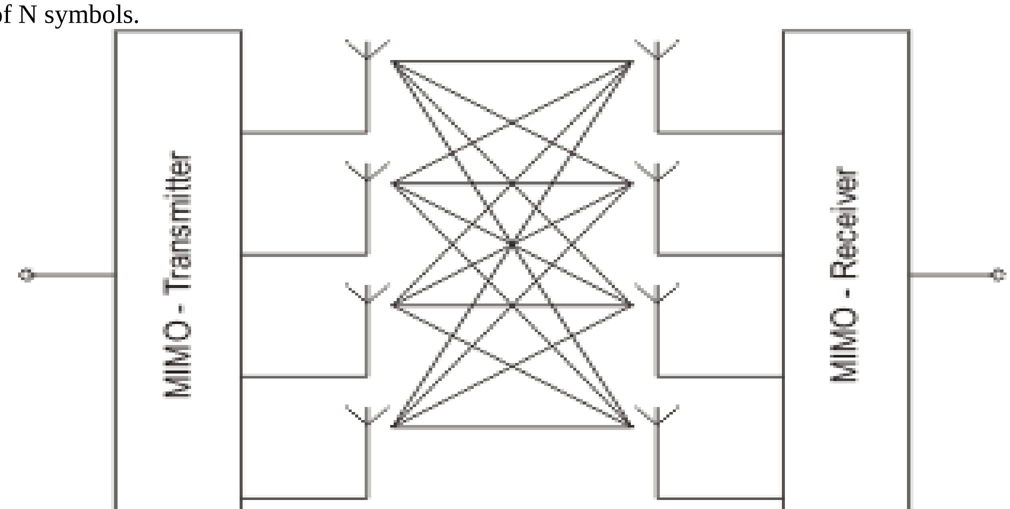
Abstract

MIMO-OFDM (Multiple Input Multiple Output Orthogonal Frequency Division Multiplexing) system has been recognized as one of the most popular and competitive technique in a wireless environment now a days. The performance is calculated in terms of Bit Error Rate (BER) versus the Signal to Noise Ratio (SNR). In this paper we discuss various algorithm to reduce ISI .we discuss the BER performance of the MIMO-OFDM system with four different equalizers (ZF, LMS, RLS and MMSE). The multicarrier modulation is used, which have advantages like inter symbol interference (ISI) reduction, high data rate, greater reliability, and good performance in multipath fading.
Keywords: OFDM, ISI, Rayleigh Fading Channel, BER, Signal To Noise Ratio (SNR).

I. INTRODUCTION

A. MIMO SYSTEM MODEL:

MIMO is a narrowband technology that uses multiple transmits and receives antennas. If H is the channel matrix then we have $Y = Hx + n$. The number of independent channels that a signal travels from the sender to the receiver is called as the diversity gain. In order to operate MIMO systems properly, we require careful design, with the encoded signals collected from each transmitting antenna and the multiple communication channels achieving specified orthogonality conditions.[4]. The better combination of number of transmitting and receiving antenna for MIMO systems in BPSK modulation technique that satisfy the good SNR is to be investigated primarily. The following multi-antenna MIMO communication system consist of n transmit antenna and m receive antenna, and in some case with a slowly time-varying channel . Due to the wireless nature of the system, each receive antenna receive transmission from all transmitter. By slowly time-varying, we consider the channel remain constant over a block of data consists of N symbols.



Wireless Channel

Bit error rate The measure of performance of any communication system is usually bit error rate (BER).Bit Error Rate is given as follows BER= Errors/Total Number of Bits.With a strong signal and an undistributed signal path, this number so small as to be insignificant. It becomes significant when we want to maintain an adequate signal-to-noise ratio in the presence of inadequate transmission through electronic circuitry and the medium for propagation

B. OFDM:

A general problem found in high speed communication is inter-symbol interference .ISI occurs when a transmission interferes with itself and the receiver cannot decode the transmission correctly.[13] This paper will focus on Orthogonal Frequency Division Multiplexing (OFDM). OFDM is especially suitable for high speed communication due to its resistance to ISI.As communication systems boost their information transfer speed the time for each transmission. The primary objective of our study is reducing the ISI problem in the wireless communication. One solution can be Orthogonal Frequency Division Multiplexing (OFDM).The idea of OFDM [12] is to distribute the high rate data stream into many low rate data streams that are transmitted in a parallel way over many sub Channels. Thus, in a sub channel, the symbol duration is low as compared to the maximum delay of the channel and hence, ISI can be handled. Orthogonal Frequency Division Multiplexing (OFDM) is a multicarrier transmission technique that has been recently recognized as an excellent method for high speed bi-directional wireless data communication.

It is also used for wireless digital audio and video broadcasting .It is based on Frequency Division Multiplexing (FDM), which is a technology that uses multiple frequencies to simultaneously transmit multiple signals in parallel Each signal has its own frequency range (sub carrier), which is then modulated by input. Each subcarrier is separated by a guard band to ensure that they do not overlap. These sub-carrier are then demodulated at the receiver by using filters to separate the bands

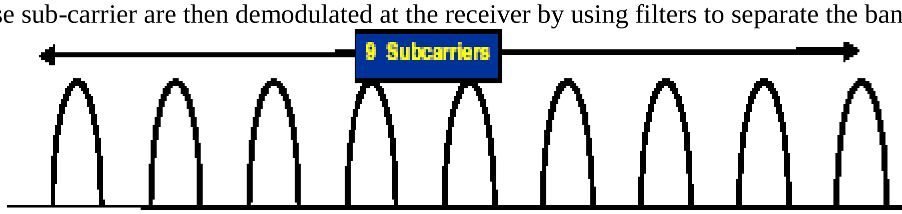


Fig. 1: FDM

OFDM is similar to FDM but much more spectrally efficient by spacing the sub channel much more spectrally efficient by spacing much closer together. This is done by computing frequencies that are orthogonal, which means that they are perpendicular in mathematical terms, allowing the spectrum of each sub-channel to overlap another without Interfering with it. In the effect of this is seen as the required bandwidth is greatly reduced by removing guard bands and allowing signals to overlap .In order to demodulate the signal ,a discrete Fourier transform (DFT) is needed .Fast Fourier transform (FFT) chips are commercially available making this a relatively easy operation .

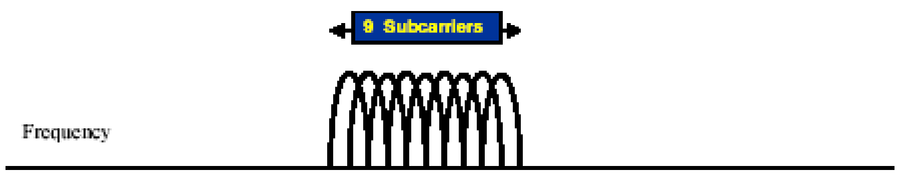


Fig. 2 : OFDM

C. Intersymbol Interference:

ISI is a form of distortion of a signal in which one symbol interferes with subsequent symbols. This is an undesirable phenomenon as the previous symbols have similar effect as noise, thus making the communication less reliable. ISI is usually caused by multipath propagation or the inherent non-linear frequency response of a channel causing successive symbols to "mixed" together. The presence of ISI in the system gives errors in the decision device at the receiver output. Thus, in the design of the transmitting and receiving filters, the purpose to reduce the effects of ISI, and thereby deliver the required data to its destination with the smallest error rate possible

D. Bit error rate:

The measure of performance of any communication system is usually bit error rate (BER).Bit Error Rate is given as follows BER= Errors/Total Number of Bits .With a strong signal and an undisturbed signal path, this number should be as small as to be insignificant. It becomes significant when we want to maintain an adequate signal-to-noise ratio in the presence of inadequate transmission through electronic circuitry and the medium for propagation.

E. Signal to Noise Ratio, SNR

The noise performance and hence the signal to noise ratio is a key parameter for any radio receiver. The SNR is often defined as a measure of the sensitivity performance of a receiver. This is of main importance in all applications from simple broadcast receivers to those used in cellular or wireless communications as well as in fixed or mobile radio communications and satellite radio communication.

There are a number of ways in which the noise performance, and hence the sensitivity of a radio receiver can be measured. The most general method is to compare the signal and noise levels for a known signal level, i.e. the signal to noise (S/N) ratio or SNR. Obviously the greater the difference between the signal and the unwanted noise, the better the system is.

F. Channel Equalization:
A typical communication system design involves first passing the signal to be transmitted through a whitening filter to reduce redundancy or correlation and then transmitting the resultant whitened signal. At the receiver, the received signal is passed through the inverse whitening filter and the original signal is thus restored. However, the channel will affect the transmitted signal because of a) Channel noise b) Channel dispersion leading to inter symbol interference. For example, by reflection of the transmitted signal from different objects such as buildings in the transmission path, leads to echoes of the transmitted signal appearing in the receiver. Therefore, it is necessary to pass the received signal through equalizing filter to reduce the dispersion effect. Equalization compensates for Inter symbol Interference (ISI) created by multi path within time dispersive Channel message signal whitening signal receiver.

II. ALGORITHM

A. Zero Forcing Equalizer:
This is a linear equalization algorithm used in communication systems, which inverts the frequency response of the channel at the receiver to restore the signal before the channel [2]. ZF algorithm considers as the signal of each transmitting antenna output as the desired signal, and consider the remaining part as a disruption, so the mutual interference between the different transmitting antennas can be completely removed. ZF equalizers ignore the additive noise and may considerably amplify noise for channels with spectral nulls. Mathematical equation of sub-channel in the MIMO-OFDM system is as follows:

$$R\,(k) = H\,(k)\,X\,(k) + n\,(k) \qquad (1)$$

Where, , $X\ k$ and $n\ k$ respectively expresses output signal, the input signal and noise vector of the k sub-channels in MIMO-OFDM system. The relation between input (k) and output signal (k) as in eq. 1 exploits that this is a linear equalizer. A ZF algorithm for MIMO OFDM is the most common and basic algorithm, and the basic concept of ZF algorithm is kept of MIMO-channel interference by multiplying received signal and the inverse matrix of channel matrix. Zero- Forcing solution of MIMO-OFDM system is as follows:

$$X_{ZF} = H^{-1}R = x + H^{-1}\,n \qquad (2)$$

in which H^{-1} is the channel matrix for the generalized inverse matrix.

B. LMS algorithm:
Usually, the adaptive algorithm consists of a transfer filter for processing the input single and an algorithm unit for update the transfer filter's coefficients. $x(n)$ is the input signal; $w(n) = [\ w_0\ \ w_1,\ w_2 ......\ w_l]$ is the vector of the transfer filter's coefficients; $d(n)$ is the desired output of the transfer filter; $y(n)$ is the output of the transfer filter; $e(n)$ is the error value, and it can be written as:

$$e(n) = d(n) - y(n) \qquad (3)$$

For LMS algorithm, the method to update the coefficients of the transfer filter is given as follows:

$$w(n) = w(n+1) + \mu^{*} x(n)^{*} e(n) \qquad (4)$$

μ is the step of LMS algorithm.

The main drawback of the "pure" LMS algorithm is that it is sensitive to the scaling of its input $x(n)$. This makes it very hard (if not impossible) to choose a learning rate μ that guarantees stability of the algorithm.

C. RLS Algorithm
RLS (recursive least squares) algorithm is used for determining the coefficients of an adaptive filter. RLS algorithm uses information from all past input samples to approximate the autocorrelation matrix of the input vector. To decrease the impact of input samples, a forgetting factor for the effect of each sample is used.The First process is filtering in which RLS computes the output of a linear filter in response to an input signal and generates an estimation error. Second is the adjustment of parameters of the filter in accordance with the estimation error.

$$r(n) = wH(n)\,C(n) \qquad (5)$$

The above equation describes the filtering portion of the algorithm. Transversal filter is used to compute error estimates given by the following equation.

$$e(n) = d(n) – r(n) \qquad (6)$$

Where d(n) is the desired response and is given by "(16)". Equation (17) describes the adaptive operation in which the tap-weight vector is updated by incrementing its old value by an amount equal to the complex conjugate of the estimation error.

$$d(n) = wr(n) \qquad (7)$$

$$w(\,nr + 1\,) = w(\,nr\,) + \mu C(n)\, e^{*}(n) \qquad (8)$$

Where nr is number of iterations and μ is step size, which manages the convergence rate and stability of algorithm.RLS is an adaptive algorithm based on the idea of least squares. RLS algorithm is used to remove the influence of old measurements. It is observed that the difference between the actual and predicted values obtained using RLS algorithm is very less and hence we can infer that the tracking is efficient if we use RLS algorithm.

D. MMSE Algorithm

1) *Minimum mean square error (MMSE) equalizer:*

Minimum mean square error (MMSE) equalizer: A MMSE estimator is a method in which it minimizes the mean square error (MSE), which is a universal measure of estimator quality [12]. The most important characteristic of MMSE equalizer is that it does not usually eliminate ISI totally but instead of minimizes the total power of the noise and ISI components in the output.

Let us assume that x be an unknown random variable and R be a known random variable, then

$$R = HX + n \ . \qquad (9)$$

An estimator $x\ (R)$ is any function of the measurement y, and its mean square error is given by

$$MSE = \{(X - X)2\} \qquad (10)$$

where the expectation is taken over both X and R.The comparison of various algorithms, advantages and disadvantages details are given

Table - 1
Comparisons of Various Algorithms

ALGORITHMS	*ADVANTAGES*	*DISADVANTAGES*
Zero Forcing (ZF)	*Performs well for static channels with high SNR*	*Neglects the effect of noise altogether*
Least Mean squares (LMS)	*Low computational complexity, simple program*	*Slow convergence, poor tracking*
Recursive Least squares (RLS)	*Fast convergence ,excellent tracking ability*	*High computational complexity*
Minimum mean square error (MMSE)	*High adaptation rate, low computational complexity*	*Residual ISI remains*

III. CONCLUSION

This paper has coverd the basic concepts of equalization used in wireless communication to reduce bit error rate which occurs due to intersymbol interference or fading effects. Fading effects in the channels are equalized by using four equalization algorithms namely Least Mean Squares (LMS), Zero Forcing Algorithm (ZF) and Recursive Least Squares (RLS) and minimum mean square error (MMSE). Each algorithm discussed has its own advantages and disadvantages for appropriate reduction in bit error rate and hence ISI.

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