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Bit Error Rate reduction in MIMO systems using Equalization techniques

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Abstract---Multiple transmit and receive antennas can be used to form multiple-input multiple-output (MIMO) channels to increase the capacity and data rate. Fading effects are the major effects to be considered at the receiver. Fading effects must be mitigated at the receiver before demodulation by using equalization techniques. Equalization compensates for inter symbol interference (ISI) created by multipath within time dispersive channels which is the major factor responsible for bit error rate. In this paper, different equalization techniques are presented for reduction in Bit Error Rate (BER).

***Index Terms*--BER (Bit Error Rate), DFE equalizer, Equalization algorithms, ISI (intersymbol interference), LMS equalizer, MIMO, ZF (Zero Forcing Equalizer).**

I. INTRODUCTION

Wireless communication systems require different techniques of signal processing that improve the link performance in aggressive mobile environments. The mobile radio channel is dynamic due to multipath propagation and Doppler spread, these effects have a strong negative impact on the bit error rate of any modulation technique. Equalization, diversity, and channel coding are basically three techniques which can be used independently to improve received signal quality and link performance over small-scale times and distances. Inter symbol interference (ISI) must be compensated by using different equalization techniques created by multi path within time dispersive channels. If the modulation bandwidth exceeds the coherence bandwidth of the channel, intersymbol interference occurs and pulses spread in time into adjacent symbols and cause pulse broadening. An equalizer within a receiver compensates for the average range of expected channel amplitude and delay characteristics. Equalizers must be adaptive since the channel is generally unknown and vary with time. Equalizers are used to mitigate the effects of fading. These are usually employed to reduce the depth and duration of the fades experienced by a receiver in a local area which are due to motion. Multiple-input multiple-output (MIMO) communication systems offer an enormous increase in spectral efficiency. In high-data rate MIMO applications with frequency-selective fading, the receiver faces a challenging task to detect multiple transmit signals in the presence of noise, channel-induced inter-symbol interference (ISI) [1].

II. MIMO SYSTEM MODEL

MIMO is a narrowband technology that uses multiple transmits and receives antennas. If H is the channel matrix then we have

$$Y = Hx + n \qquad (1)$$

The number of independent channels that a signal travels from the sender to the receiver is called as the diversity gain. The proper operation of MIMO systems requires careful design, with the encoded signals received from each transmitting antenna and the multiple communication channels achieving specified orthogonality conditions.[2]. 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 H_{ij} . Due to the wireless nature of the system, each receive antenna receive transmission from all transmitter. By slowly time-varying, we assume the channel remain constant over a block of data consists of N symbols. We further assume that all channel responses to be of finite length v, v < N. The input/output relationship can be then described by the following equation:

$$y_j(k) = \sum_{i=1}^{M_T} h_{ij}(k) * x_i(k) + n_j(k), \; j=1 \qquad (2)$$

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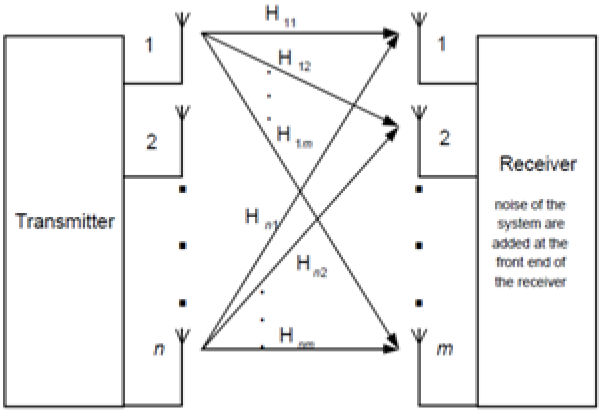

Fig.1 A multi-input multi-output system

Where y_j(k) is the output of the jth receiving antenna,x_i(k) is the input at the ith transmitting antenna.

A. 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 unperturbed 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.

III. EQUALIZATION

The inter-symbol interference caused by multipath MIMO channels distorts the MIMO transmitted signal which causes bit errors at receiver. To minimize this ISI, equalization is needed. Equalizer minimizes the error between actual output and desired output by continuous updating its filter coefficients [3]. Equalization can be done in both time and frequency domain. Equalization in frequency domain is simpler to use as compared to time domain. Equalizers basically use any adaptive algorithm for reducing the effects of deep fades as well as intersymbol interference. In this paper various equalizer's performance is compared in terms of bit error rates (BER) giving the number of operations and their advantages as well as disadvantages.

A) Least Mean Square (LMS) Algorithm

A more robust equalizer is the LMS equalizer where the mean square error (MSE) between the desired equalizer output and the actual equalizer output.

$$\hat{e}_k = d_k - \hat{d}_k \quad (3)$$

$$e_k = X_k - Y_k^T W_k = X_k - Y_k^T Y_k \quad (4)$$

To compute the mean square error at time instant k,

$$\zeta = E[e_k^* e_k] \quad (5)$$

The LMS algorithm seeks to minimize the mean square error given in above equation. For a specific channel condition, the prediction error e_k is dependent on the tap gain vector W_N , so the MSE of an equalizer is a function of W_N . Let the cost function J(W_N) denote the mean squared error as a function of tap gain vector W_N . In order to minimize the MSE, it is required to set the derivative of Equation to zero.

$$dJ(W_N) = -2p_N + 2R_{NN}W_N = 0 \quad (6)$$

Simplifying Equation

$$R_{NN}\hat{W}_N = p_N \quad (7)$$

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Equation is a classic result, and is called the normal equation, since the error is minimized and is made orthogonal (normal) to the projection related to the desired signal.When Equation is satisfied, the MMSE of the equalizer is

$$J_{opt} = J(\hat{W}_N) = E[x_k x_k^*] - p_N^T \hat{W}_N \quad (8)$$

To obtain the optimal tap gain vector, the normal equation must be solved iteratively as the equalizer converges to an acceptably small value of J_{opt}. There are several ways to do this, and many variants of the LMS algorithm have been built upon the solution of equalization. One obvious technique is to calculate

$$W = R_{NN}^{-1} p_N \quad (9)$$

The filter weights are updated by the update equations given below. Let the variable n denote the sequence of iterations, LMS is computed iteratively b

$$\hat{d}_k(n) = w_N^T(n)\, y_N(n) \quad (10)$$

$$e_k = x_k(n) - \hat{d}_k(n) \quad (11)$$

$$w_N(n+1) = w_N(n) - \alpha e_k^*(n)\, y_N(n) \quad (12)$$

Where the subscript N denotes the number of delay stages in the equalizer, and α is the step size which controls the convergence rate and stability of the algorithm. The Least mean square algorithm in the equalizer maximizes the signal to distortion ratio at its output within the constraints of equalizerilter length of the . If an input signal has a time dispersion characteristic that is greater than the propagation hindrance through the equalizer, then the equalizer will be unable to reduce distortion. The convergence rate of the LMS algorithm is slow due to the fact that there is only one parameter, the step size α , that controls the adaptation rate.

B) Zero-Forcing (ZF) Equalization

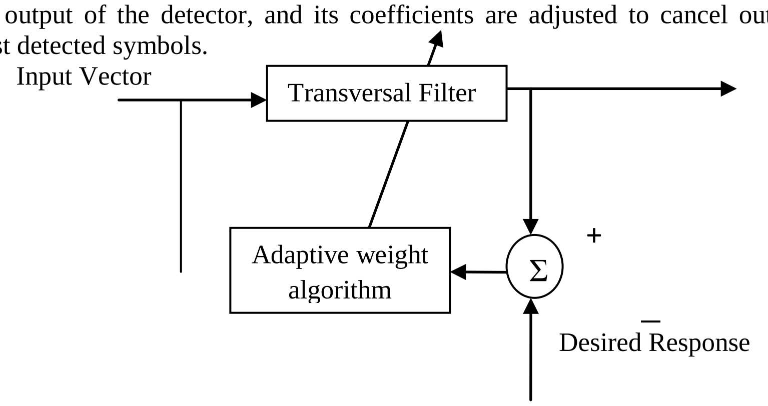
In ZF equalizer [3] the coefficients are chosen to force the samples of the combined channel and equalizer impulse response to zero. The combined response of the channel with the equalizer is given by

$$H_{CH}(f)\, H_{EQ}(f) = 1 \quad (13)$$

Where H_{CH}(f) is folded frequency response of the MIMO-channel and H_{EQ}(f) is frequency response of equalizer. In this case, the equalizer filter compensates for the channel-induced ISI as well as the ISI brought about by the transmitter and receiver filters. Zero-Forcing filter designed using the equation above does not eliminate all ISI because the filter is of finite length.

C) Decision Feedback Equalizer (DFE)

In DFE [3] once an input symbol has been detected, the intersymbol interference that it induces on future symbols is predicted and subtracted before detection of subsequent symbols. DFE is realized in direct transversal form which consists of feed forward filter (FFF) and a feedback filter (FBF). The FBF is driven by decision on the output of the detector, and its coefficients are adjusted to cancel out the ISI on the current symbol from past detected symbols.



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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 influence of input samples, a forgetting factor for the influence of each sample is used. First process is the 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.

$$r(n) = wH(n)\, C(n) \qquad (14)$$

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

$$e(n) = d(n) - r(n) \qquad (15)$$

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.

$$d(n) = wr(n) \qquad (16)$$

$$w(\, nr + 1\,) = w(\, nr\,) + \mu C(n)\, e^{*}(n) \qquad (17)$$

Where nr is number of iterations and μ is step size, which controls 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.RLS algorithm outperforms other algorithms to predict the channel coefficients. It is observed that the difference between the actual and predicted values obtained using RLS algorithm is very very less and hence we can infer that the tracking is efficient if we use RLS algorithm. The comparison of various algorithms, complexity and advantages details are given

TABLE .1: COMPARISONS OF VARIOUS ALGORITHMS

Algorithm	Number of Multiply Operations	Advantages	Disadvantages
Least Mean squares (LMS)	2N+1	Low computational complexity, simple program	Slow convergence, poor tracking
Recursive Least squares (RLS)	$1.5N^2$+6.5N	Fast convergence ,excellent tracking ability	High computational complexity
Zero Forcing Algorithm (ZF)		Performs well for static channels with high SNR	Neglects the effect of noise altogether

IV. CONCLUSION

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

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