Bit Error Rate as a Performance Indicator for Fading Channel Communication

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Abstract

MIMO is one of the technical breakthroughs in modern digital communication and also known as volume to volume wireless link. In wireless communication the issue of fading and interference is a major issue. An equalizer will help in solving the issue. Inter symbol Interference (ISI) is created by multipath signal prorogation within time dispersive channels and Equalization compensates it. In this paper we have investigated the BER performance characteristics of different equalizers namely, ZF,SD,ML MMSE ,DFE, and MLSE equalizers. It is also shown that, at low BERs, both the MLSE algorithm and the DFE algorithm suffer from limitations. The DFE error performance is detected bits fed back than with correct bits fed back. Finally, it is observed that during the MLSE portion of the simulation, it dynamically updates the estimated channel response. MATLAB is used for simulation purpose. Using adaptive equalization, compensation for the time dispersion introduced by the channel can be achieved.

Keywords: BER(Bit Error Rate),MIMO(Multiple Input Multiple Output),ZF(Zero Forcing),ISI(Inter Symbol Interference),ML(Maximum Likely hood),MMSE(Minimum Mean Square Error)DFE(Decision Feedback Equalizer)

1. INTRODUCTION

The rising demand of multimedia services and the development of Internet related contents lead to increasing curiosity to high speed communications. In the ceaseless search for increased capacity in a wireless communication channel it has been shown that by using MIMO (Multiple Input Multiple Output) system architecture it is possible to increase that capacity considerably. The MIMO is very likely beneficial since it enables support of more antennas and larger bandwidths and it simplifies equalization in MIMO systems. Usually fading is considered as a problem in wireless communication but MIMO channels uses the fading to increase the capacity of entire system. Fading of the signal can be mitigated by different diversity techniques. The data rate and spectrum efficiency of wireless mobile communications have been significantly improved over the last decade or so. Recently, the advanced systems such as 3GPP LTE and terrestrial digital TV broadcasting have been sophisticatedly developed using OFDM and CDMA technology. In general, most mobile communication systems transmit bits of information in the radio space to the receiver. The radio channels in mobile radio systems are usually multipath fadingchannels, which cause inter-symbol interference (ISI) in the Received signal. To remove ISI from the signal, there is a needof strong equalizer which requires knowledge on the channel impulse response (CIR).[1]Hence, there is a need for the development of novel practical, low complexity equalization techniques and for understanding their potentials and limitationswhen used in wireless communication systems characterized by very high data rates, high mobility and the presence of multiple antennas.[10]In radio channels, a variety of adaptive equalizers can be used to cancel interference while providing diversity. Since the mobile fading channel is random and time varying, equalizers must track the time varying characteristics of the mobile channel, and thus are called adaptive equalizers. The general operating modes of an adaptive equalizer include training and tracking. First, a known, fixed-length training sequence is sent by the transmitter so that the receiver's equalizer may adapt to a proper setting for minimum bit error rate (BER) detection.

OVERVIEW OF MIMO- SYSTEM The *MIMO system transmits different signals from each transmit element so that the receiving antenna array receives a superposition of all the transmitted signals. All signals are transmitted from all elements once and the receiver solves a linear equation system to demodulate the message. Multiplexing (MIMO) system isan effective solution to improve communication quality, performance, capacity, and transmission rate. MIMO is under intensive investigation by researchers*
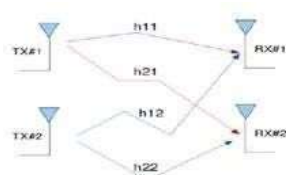

Fig-1 Transmit 2 Receive (2×2) MIMO

We consider the system where the transmitter has n_t antennas and the receiver has n_r antennas. Let $h_{m,n}$, be a complex number corresponding to the channel gain between transmit antenna n and the receive antenna m. If at a certain time instant complex signals $\{x_1, x_2, x_3, \ldots x_{n_t}\}$

are transmitted via the n_t antennas, the received signals at antenna m can be expressed as

$$y_m = \sum_{n=1}^{n_c} h_{m,n} + e_m \qquad \{1\}$$

Where e_m is a noise signal. This relation is easily expressed in a matrix structure. Let x and y be n_t and n_r vectors containing the transmitter and receiver data, respectively. Define the following $n_r \times n_t$ channel gain matrix

$$H = \begin{bmatrix} h1,1 & \cdots & h1,nt \\ \vdots & \ddots & \vdots \\ hnr,1 & \cdots & hnr,nt \end{bmatrix} \qquad (2)$$

Then we have, $y = Hx + e \qquad \{3\}$

Where, $e = [e_1, e_2, e_3, \ldots]^T$ which Is a vector of noise sample

3. FLAT RAYLEIGH FADING MODEL

The propagating radio signals are affected by the physical channel in various ways. To specify a situation of frequency- flat Rayleigh fading, a model is introduced called flat Rayleigh fading model. This is a reference model, Fast fading component has Rayleigh density function if there is no direct path from signalling parts. Rayleigh distribution is as follows,

$$f_{Rayleigh}(r) = \begin{cases} \frac{r}{\sigma^2} \exp\left(-\frac{r^2}{2\sigma^2}\right) & 0 \leq r \leq \infty \end{cases} \qquad (4)$$

3(A):BER SIMULATION OF BPSK IN A 10-TAP RAYLEIGH FADING CHANNEL :
(a)Generation of random binary sequence.BPSK modulation i.e. bit 0 represented as -1 and bit 1 represented as +1
(b)Assigning to multiple OFDM symbols where data subcarriers from -26 to -1 and +1 to +26 are used, adding cyclic prefix,
 (c)   Convolving each OFDM symbol with a 10-tap Rayleigh fading channel. The frequency response of fading channel on
    each symbol is computed and stored.
(d)Concatenation of multiple symbols to form a long transmit sequence
(e)   Adding White Gaussian Noise. Grouping the received vector into multiple symbols, removing cyclic prefix
(f)Counting the number of bit errors. Repeating for multiple values of .The simulation results are as shown in the plot
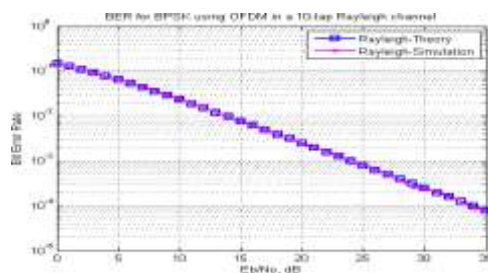
below.

FIG 2:BER for BPSK using OFDM in a
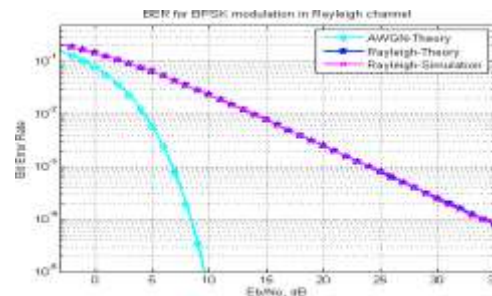10-tap Rayleigh Channel

FIG:3 BER for BPSK in Rayleigh Channel

4. MINIMUM MEAN SQUARE ERROR (MMSE)

In statistics and signal processing, a minimum mean square error (MMSE) estimator describes the approach which minimizes the mean square error (MSE), which is a common measure of estimator quality. Mathematically,
Let X be an unknown random variable, and let Y be a known random variable (the measurement). An estimator is any

function of the measurement Y, and its MSE is given by $MSE = E(X - \bar{X})^2$
Where the expectation is taken over both X and Y. The MMSE estimator is then defined as the estimator achieving minimal MSE.

4(A). MIMO WITH MMSE EQUALIZER

In a 2×2 MIMO channel, probable usage of the available 2 transmit antennas can be as follows:

1. Consider that we have a transmission sequence, for example

2. In normal transmission, we will be sending x_1 in the first time slot, x_2 in the second time slot, and so on.

3.   However, as we now have 2 transmit antennas, we may group the symbols into groups of two. In the first time slot, send x_1 and x_2 from the first and second antenna. In second time slot, send x_3 and x_4 from the first and second antenna; send x_5 and $x6$ in the third time slot and so on.

4.   we are grouping two symbols and sending them in one time slot, we need only n/2 time slots to complete the transmission-data rate is doubled .This forms the simple explanation of a probable MIMO transmission scheme with 2 transmit antennas and 2 receive antennas

4(B).SIMULATION OF MIMO OFDM FOR COMPUTING MINIMUM MEAN SQUARE ERROR (MMSE)

It is clearly seen from the simulation result of MIMO with MMSE without OFDM that, in increase SNR then Probability of error will get decrease. Now we want to find practically the results of MIMO with MMSE using OFDM how this will change the obtainable error From the following simulation results we find that the theoretical performance of Maximal Ratio Combining (MRC) diversity technique is best among all the receiver techniques. And there is a big difference of practical Performance of MMSE. The Synchronization factor in an OFDM system is the most critical one. When the receiver is initially turned on, it is not in synchronization with the transmitter. For this reason, data transmission in an OFDM System might need data to be sent in frames. At the beginning of each frame a null symbol is transmitted, so that the receiver can detect incoming data using simple envelope detection techniques. However, the noise in the signal might interfere with the envelope detection process. In general, it has been found that the receiver synchronizes itself with the transmitter in a time interval less than or equal to the guard interval. The Complexity of performing an FFT is dependent on the size of the FFT. However, it can be seen that because the symbol period increases with a larger FFT, the extra processing required is minimal.For e.g. the 2048-point FFT requires only 1.1 times the time required for processing a 1024-point FFT Now let us think aboutthe other results of different MIMO strategies with MMSE receiver. What a great change we got in 2x3 MIMO with MMSE (without OFDM) as compared to the MRC. We achieved a difference of 1db between the MRC and MMSE. And the MMSE equalizer is now able to show better performance in comparison with MRC. Better results can be obtained if we increase size of the MIMO

Simulation Model

   (a) Generate Random binary sequence 1's& -1's

   (b) Group them into pair of symbols and send two symbols into one time slot

   (c)   Multiply the symbol with the channel and then add white Gussian

   noise (d)Equalized the received symbols

   (e) Perform hard decision decoding and count bit errors

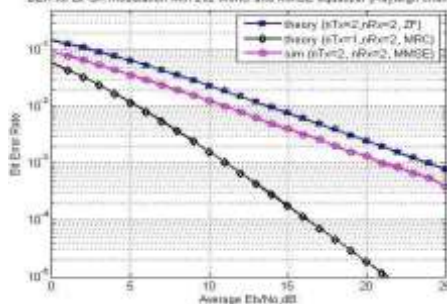   (f) Repeat the multiple value of and plot the simulation and theoretical result
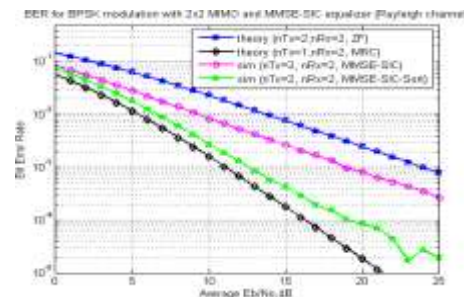

Fig4 :BER for BPSK with 2×2 MIMO&MMSE

Fig 5:BER for BPSK with 2×2 MIMO&MMSE-SIC

5. Zero Forcing Equalizer

Zero Forcing Equalizer is a linear equalization algorithm used in communication systems, which inverts the frequency response of the channel. This equalizer was first proposed by Robert Lucky. The Zero-Forcing Equalizer applies the inverse of the channel to the received signal, to restore the signal before the channel. The name Zero Forcing corresponds to bringing down the ISI to zero in a noise free case. This will be useful when ISI is significant compared to noise. For a channel with frequency response F(f) the zero forcing equalizer C(f) is constructed such that $C(F) = 1/F(f)$ Thus the combination of channel and equalizer gives a flat frequency response and linear phase $F(f)C(f) = 1$.If the channel response for a particular channel is H(s) then the input signal is multiplied by the reciprocal of this. This is intended to remove the effect of channel from the received signal, in particular the Inter symbol Interference (ISI). let us consider a 2x2 MIMO channel, the

Channel is modelled as, The received signal on the first receive antenna is,

$$y_1 = h_{1,1}x_1 + h_{1,2}x_2 + n_1 = \begin{bmatrix} h_{1,1} & h_{1,2} \end{bmatrix} \begin{pmatrix} x_1 \\ x_2 \end{pmatrix} + n_1 \quad (5)$$

$$y_1 - k_{2,1}r_1 + k_{2,2}r_2 + n_2 = [h_{1,1} \ h_{1,2}]\binom{x_1}{x_2} + n_2 \quad (6)$$

Where

y_1, y_2 are the received symbol on the first and second antenna respectively, is

$h_{1,1}$ the channel from 1st transmit antenna to 1st receive antenna,

$h_{1,2}$ is the channel from 2nd transmit antenna to 1st receive antenna, is

$h_{2,1}$ the channel from 1st transmit antenna to 2nd receive antenna, is

$h_{2,2}$ the channel from 2nd transmit antenna to 2nd receive antenna,

x_1, x_2 are the transmitted symbols and

n_1, n_2 are the noise on 1st and 2nd receive antennas.

The equation can be represented in matrix notation as follows

$$\binom{y_1}{y_2} = \binom{h_{1,1} \ h_{1,2}}{h_{2,1} \ h_{2,2}}\binom{x_1}{x_2} + \binom{n_1}{n_2} \quad (7)$$

Equivalently

$$y = Hx + n \quad (8)$$

To solve for x, we need to find a matrix W which satisfies $WH = I$. The Zero Forcing (ZF) detector for meeting this constraint is given by

$$W = (H^H H)^{-1} H^H \quad (9)$$

Where W - Equalization Matrix and

H - Channel Matrix

This matrix is known as the Pseudo inverse for a general m x n matrix where

$$H^H H = \begin{pmatrix} h_{1,1}^* & h_{2,1}^* \\ h_{1,2}^* & h_{2,2}^* \end{pmatrix}\begin{pmatrix} h11 & h12 \\ h21 & h22 \end{pmatrix} = \begin{bmatrix} |h1,1|^2 + |h2,1|^2 & h_{1,1}^* h12 + h_{2,1}^* h22 \\ h11 h_{1,2}^* + h21 h_{2,2}^* & |h1,2|^2 + |h2,2|^2 \end{bmatrix} \quad (10)$$

Note that the off diagonal elements in the matrix $H^H H$ are not zero, because the off diagonal elements are non zero in values. Zero forcing equalizer tries to null out the interfering terms when performing the equalization, i.e. when solving for $x1$ the interference from $x2$ is tried to be nulled and vice versa. While doing so, there can be an amplification of noise. Hence the Zero forcing equalizer is not the best possible equalizer. However, it is simple and reasonably easy to implement. For BPSK Modulation in Rayleigh fading channel, the BER is defined as

$$P_b = \frac{1}{2}\left(1 - \sqrt{\frac{\frac{E_b}{N_0}}{\frac{E_b}{N_0} + 1}}\right) \quad (11)$$
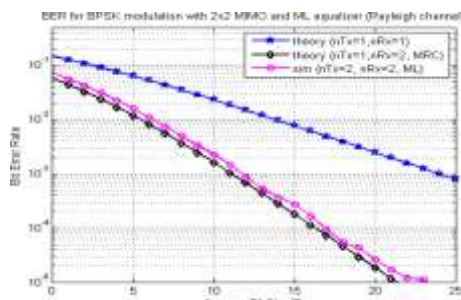
Where - Bit Error Rate

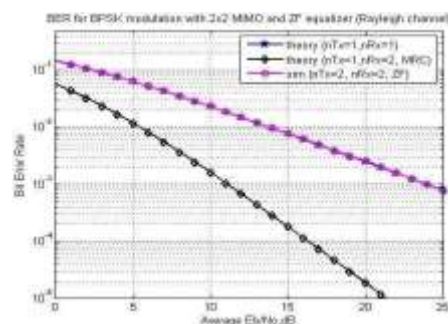**Figure6:** BER for BPSK with 2×2 MIMO&ML

Figure7: BER for BPSK with 2×2 MIMO&ZF

6. Maximum Likelihood (ML)Receiver

The Maximum Likelihood receiver tries to find $|y - Hx|$ which minimizes, **J** ^{2} Since the modulation is BPSK, the possible values of x_1 is +1 or -1 Similarly x_2 also take values +1 or -1. So, to find the Maximum Likelihood solution, we need to find the minimum from the all four combinations of x_1 and x_2. The estimate of the transmit symbol is chosen based on the minimum value from the above four values i.e

if the minimum is , $J_{+1 +1}$=> [1 1]

if the minimum is , $J_{+1 -1}$=> [1 0]

if the minimum is $J_{-1,1}$ => [0 1]

and

if the minimum is . J_{-1-1}=> [0 0]
Simulation Model

(a)Generate Random binary sequence 1's& -1's

(b)   Group them into pair of symbols and send two symbols into one time slot (c)Multiply the symbol with the channel and then add white Gussian noise (d)Find the minimum four possible transmit symbol combinations

$\frac{Eb}{N}$ (e)   Based on the minimum chose the estimate of transmit symbol

(f)Repeat the multiple values of   and plot the simulation and theoretical result

7. CHANNEL EQUALIZATION

ISI problem is resolved by channel equalization [12] in which the aim is to construct an equalizer such that the impulse response of the channel/equalizer combination is as close to $z^{-\Delta}$ as Possible , where is a delay. Frequently the channel parameters are not known in advance and more over they may vary with time, in some applications significantly. Hence, it is necessary to use the adaptive equalizers, which provide the means of tracking the channel characteristics. The following figure shows a diagram of a channel equalization system.
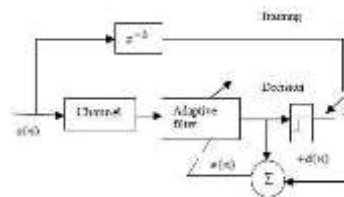

Fig7. Digital transmission system using channel equalization

In the previous figure, s(n) is the signal that you transmit through the communication channel, and x(n) is the distorted output signal. To compensate for the signal distortion, the adaptive channel equalization system completes the following two modes: Training mode - This mode helps you determine the appropriate coefficients of the adaptive filter. When you transmit the signal $s(n)$ to the communication channel, you also apply a delayed version of the same signal to the adaptive filter. In the previous figure, $z^{-\Delta}$ is a delay function and $d(n)$ is the delayed signal, $y(n)$ is the output signal from the adaptive filter and e(n)is the error signal between $d(n)$ and $y(n)$. The adaptive filter iteratively adjusts the coefficients to minimize $e(n)$ After the power of e(n) converges, $y(n)$ is almost identical to $d(n)$,which means that you can use the resulting adaptive filter coefficients to compensate for the signal distortion.

Decision-directed mode - After you determine the appropriate coefficients of the adaptive filter, you can switch the adaptive channel equalization system to decision-directed mode. In this mode, the adaptive channel equalization system decodes the signal and $y(n)$ produces a new signal, which is an estimation of the signal $s(n)$ except for a

 delay of taps .Here, Adaptive filter plays an important role. The structure of the adaptive filter is showed in Fig.3
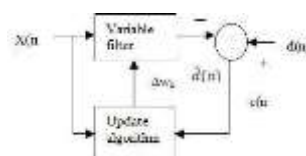

Fig 8. Adaptive filter

To start the discussion of the block diagram we take the following assumptions:

The input signal is the sum of a desired signal $d(n)$ and interfering noise v (n)

$x(n) = d(n) + v(n)$ {12}

 The variable filter has a Finite Impulse Response (FIR) structure. For such structures the impulse response is equal to the filter coefficients. The coefficients for a filter order p are defined as

$w_n = [w_n(0), w_n(1), ..., w_n(p)]^T$ {13}

the error signal or cost function is the difference between

the desired and the estimated signal

$e(n) = d(n) - \hat{d}(n)$ {14}

The variable filter estimates the desired signal by convolving

the input signal with the impulse response. In vector notation this is expressed Where

$\hat{d}(n) = w_n \cdot x(n)$ {15}

is an input signal vector. Moreover, the variable filter updates the filter coefficients at every time instant

$x(n) = [x(n), x(n-1), ..., x(n-p)]^T$ {16}

$$\Delta w_n \qquad (17)$$

Where $w_{n+1} = \Delta w_n$ is a correction factor for the filter coefficients .The adaptive algorithm generates this correction factor based on the input and error signals

8. ADAPTATION ALGORITHMS

This section briefly introduces two well-known algorithms that possess different qualities in terms of the performance i.e one is least mean square (LMS) and other is Recursive least square filter (RLS).

A. Least Mean Squares Algorithm (LMS)

The least mean-square (LMS) algorithm is probably the most widely used adaptive filtering algorithm, being employed in several communications systems. It has gained popularity due to its low computational complexity and proven robustness. The LMS algorithm

is a gradient-type algorithm that updates the coefficient vector by taking a step in the direction of the negative gradient [1] of the objective function, i.e.,

$$W_{k+1} = W_k - \frac{\mu}{2}\frac{\partial J_w}{\partial w(k)} \qquad (18)$$

Table 1.1: The Least Mean Square Algorithm

LMS ALGORITHM
for each k { $w(k+1) = w(k) + \mu e(k) x(k)$ }

where μ is a step controlling the stability, convergence speed, and misadjustment.

To find an estimate of the gradient, the LMS algorithm uses as objective function the instantaneous estimate of the MSE, i.e resulting

$$J_w = e^2(k)$$

in the gradient estimate

$$\frac{\partial J_w}{\partial w(k)} = -2e(k)x(k) \qquad (19)$$

The pseudo-code for the LMS algorithm is shown in Table 1.1. In order to guarantee stability in the mean-squared sense, the step size should μ

be chosen in the range $\quad 0 < \mu < 2/tr\{R\}\quad$ where tr{.} is the trace operator and

is the input-signal autocorrelation matrix. The upper bound should be considered optimistic and in practice a smaller value is recommended [1]. A normalized version of the LMS algorithm, the NLMS algorithm [6], is obtained by substituting the step size in Equation (1.4) with the time-varying step size $\mu/\|X(k)\|^2$ where $0 < \mu < 2$

The NLMS algorithm is in the control literature referred to as the *projection algorithm* (PA) The main drawback of the LMS and the NLMS algorithms is the slow convergence for colored

noise input signals. In cases where the convergence speed of the LMS algorithm is not satisfying, the adaptation algorithms presented in the following sections may serve as viable alternatives.

Table 1.2: The Recursive Least Square Algorithm

RLS ALGORITHM

$$R^{-1}(K) = \frac{1}{\lambda}\left[R^{-1}(K-1) - \frac{R(K)X^T(K)}{\lambda + X^T(K)R(K)} \right]$$

$R^{-1}(0) = \delta^{-1}$ δ Small positive constant

$$\frac{R^{-1}(k-1)x(k)}{K(K)}$$
$$\lambda + x^T(k)x(k)$$

for
each k
{

K(k) —

R(K) —

$$e(k) = d(k) - W^T(k)x(k)$$
$$w(k+1) = w(k) + e(k)K(k)$$

}

B. **The Recursive Least-Squares (RLS) Algorithm**

Convergence of the LMS algorithm operating in colored environment, one can implement the recursive least-squares (RLS) algorithm [6, 10]. The RLS algorithm is a recursive implementation of To overcome the problem of slow the least- squares (LS) solution, i.e., it Minimizes the LS objective function. The recursions for the most common version of the RLS algorithm, which is presented in its standard form in Table 1.2, is a result of the weighted least-squares (WLS) objective function

$$J_w = \sum_{i=1}^{k} \lambda^{k-i} e^2(i) \qquad (21)$$

Differentiating the objective function J_w with respect to $\mathbf{w}(k)$ and solving for the minimum yields the following equations

$$(i)]w(k) = \sum_{i=1}^{K} \lambda^{K-I} X(i) d(i) \qquad (22)$$

Where $0 < \lambda \leq 1$ is an exponential scaling factor often referred to as the forgetting factor.

Defining the quantities

$$R(K) = \sum_{i=1}^{k} \lambda^{k-i} x(i) x^T(i) \qquad (23)$$

and

$$P(i) = \sum_{i=1}^{K} \lambda^{K-I} x(i) d(i) \qquad (24)$$

the solution is obtained as $W(k) = R(k) P^{-1}(k)$ (25)

The recursive implementations is a result of the formulations

$$R(k) = \lambda R(K-1) + X(k) X^T(k) \qquad (26)$$

And

$$(27)$$

The inverse $\mathbf{R}^{-1}(k)$ can be obtained recursively in terms of $\mathbf{R}^{-1}(k-1)$ using the *matrix inversion lemma*1 [10] thus avoiding direct inversion of $\mathbf{R}(k)$ at each time instant k. The main problems with the RLS algorithm are potential divergence behavior in finite-precision environment and high computational complexity, which is of order N^2

9. Adaptive Equalization Technique

Different kinds of Equalizer are available in the text like Fractionally Spaced Equalizer, Blind Equalization, Decision-Feedback Equalization, Linear Phase Equalizer, T-Shaped Equalizer, Dual Mode Equalizer and Symbol spaced Equalizer. But most widely used equalizers are discussed as follow:

A. Decision-Feedback Equalization

The basic limitation of a linear equalizer, such as transversal filter, is the poor perform on the channel having spectral nulls. A decision feedback equalizer (DFE) is a nonlinear equalizer that uses previous detector decision to eliminate the ISI on pulses that are currently being demodulated. In other words, the distortion on a current pulse that was caused by previous pulses is subtracted. Figure 7 shows a simplified block diagram of a DFE where the forward filter and the feedback filter can each be a linear filter, such as transversal filter. The nonlinearity of the DFE stems from the nonlinear characteristic of the detector that provides an input to the feedback filter. The basic idea of a DFE is that if the values of the symbols previously detected are known, then ISI contributed by these symbols can be cancelled out exactly at the output of the forward filter by subtracting past symbol values with appropriate weighting. The forward and feedback tap weights can be adjusted simultaneously to fulfil a criterion such as minimizing the MSE. The DFE structure is particularly useful for equalization of channels with severe amplitude distortion, and is also less sensitive to sampling phase offset.
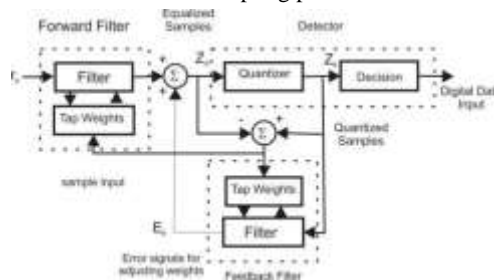

Figure9. Decision feedback equalizer

10. Simulation of DFE&MLSE channel equalizer

This example constructs and implements a linear equalizer object and a decision feedback equalizer (DFE) object. It also initializes and invokes maximum likelihood sequence estimation (MLSE) equalizer. The MLSE equalizer is first invoked with perfect channel knowledge, then with a straight forward but imperfect channel estimation technique.As the simulation

progresses, it updates a BER plot for comparative analysis between the equalization methods. It also shows the signal spectra of the linearly equalized and DFE equalized signals. It also shows the relative burstiness of the errors,

indicating that at low BERs, both the MLSE algorithm and the DFE algorithm suffer from error bursts. In particular, the DFE error performance is bustier with detected bits fed back than with correct bits fed back. Finally, during the "imperfect" MLSE portion of the simulation, it shows and dynamically updates the estimated channel response.

**A.SIMULATION MODEL FOR DFE**

(a)Set parameters related to the signal and channel

(b). Use BPSK without any pulse shaping, and a 5-tap real-valued symmetric channel impulse response

(c) Set parameter values for the linear and DFE equalizers.

(d) Use a 31-tap linear equalizer, and a DFE with 15 feed forward and feedback taps.

(e) Use the recursive least squares (RLS) algorithm for the first block of data to ensure rapid tap convergence.

(f) Use the least mean square (LMS) algorithm thereafter to ensure rapid execution speed.

B.SIMULATION MODEL FOR MLSE

(a) Set the parameters of the MLSE equalizer.

(b) Use a trace back length of six times the length of the channel impulse response.

(c) . Initialize the equalizer states.

(d) Set the equalization mode to "continuous", to enable seamless equalization over multiple blocks of data.

(e) Use a cyclic prefix in the channel estimation technique, and set the length of the prefix.

Assume that the estimated length of the channel impulse response is one sample longer than the actual length.
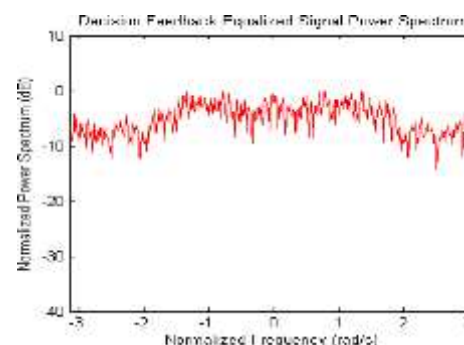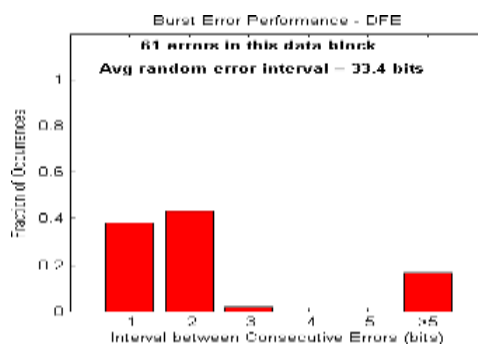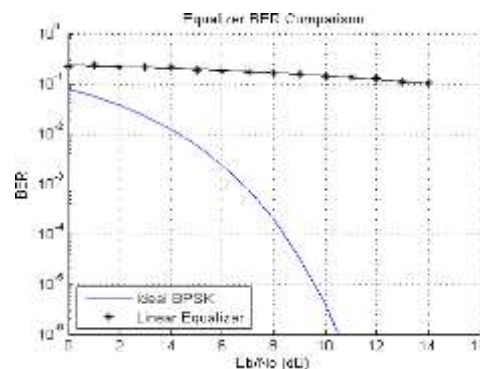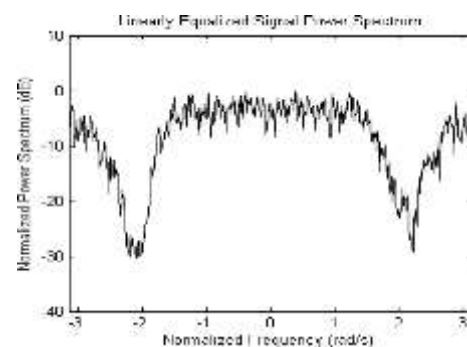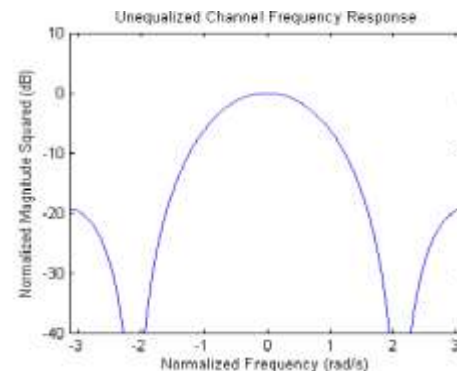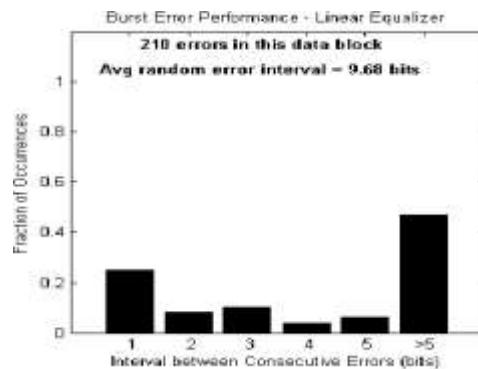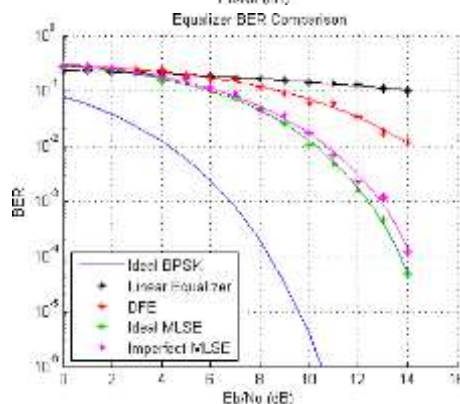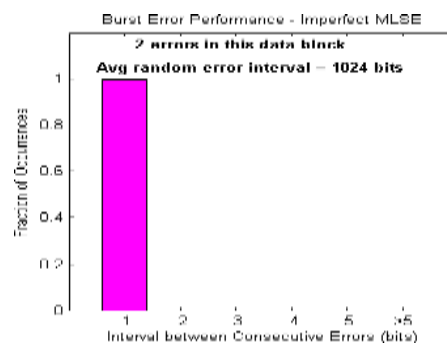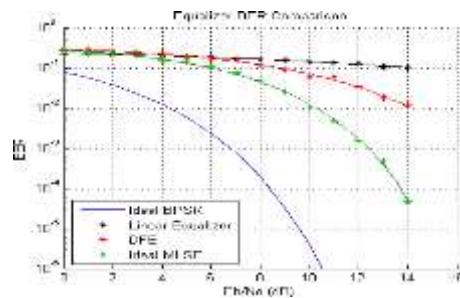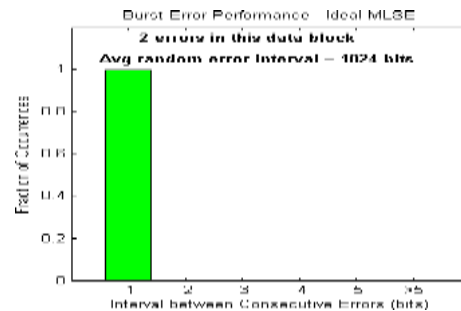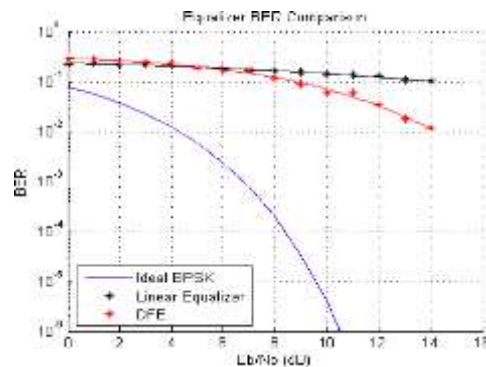
11. Conclusion:

In this paper we investigate the bit error rate performance characteristics of equalizers namely, ZF,SD, ML, MMSE ,DFE, and MLSE equalizers. It also shows the relative burstiness of the errors, indicating that at low BERs, both the MLSE algorithm and the DFE algorithm suffer from error bursts. In particular, the DFE error performance is bustier with detected bits fed back than with correct bits fed back. Finally, during the "imperfect" MLSE portion of the simulation, it dynamically updates the estimated channel response.

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