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SIMULATION OF ADAPTIVE CHANNEL EQUALIZATION FOR BPSK,QPSK AND 8-PSK SCHEMES

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Abstract - The distortion and inter symbol interference caused by multipath effects of channel degrades the quality of signal transmission in transmission system of digital baseband. Adaptive channel equalization is used commonly to compensate these effects so as to increase the reliability of propagation. Recursive Least Squares (RLS) algorithm is most commonly used adaptive algorithm because of its simplicity and fast convergence. In this work, simulation model of finite impulse response adaptive equalizer based on RLS is developed to reduce distortion caused by channel. The constellation diagram before and after equalization is obtained. It is observed that bit error rate is decreased by fifty percent after equalization. Hence this shows that the algorithm appears to reduce channel effects effectively and achieves channel equalization.

***Keywords**: Adaptive channel equalization, RLS algorithm, inter symbol interference, bit error rate..*

I. INTRODUCTION

The digital communication quality depends on channel characteristics. Phase shift keying (PSK) is an M-ary digital modulation scheme similar to conventional phase modulation. The input binary information is encoded into groups of bits before modulating the carrier[1]. PSK is widely used nowadays in military and communication systems because it offers lowest probability error[2].

In the transmission system of digital baseband, multipath transmission of channel will generate inter symbol interference in order to compensate these channel effects inverse filtering also called as equalization is used[3].Traditional filters can be used for equalization if known channel characteristics are considered. In practical channel is time varying and channel characteristics are unknown. Adaptive filters, whose weights are adjusted automatically based on some criteria, are used for equalization at receiving end to compensate distortions caused by channel. These filters play important role in many diverse applications such as communications, acoustics, speech, radar, sonar, seismology and biomedical engineering[4][5][6]. Recursive Least Squares(RLS) is the most well known adaptive filter that are used . It is designed to get minimum quadratic sum of difference between desired signal and output signal of filter. RLS is preferred because of its fast convergence and simplicity. In this work RLS algorithm is used for equalizing multipath transmission channel.

This paper is organized as follows. Section II describes the adaptive filter and equalization principle. Section III presents the RLS algorithm. Section IV presents simulation results and discussion. Section V presents conclusion.

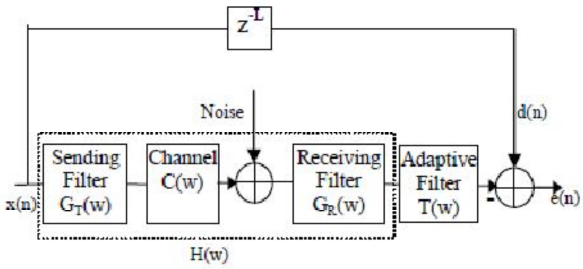
II. ADAPTIVE FILTER AND EQUALISATION PRINCIPLE

Equalization can be done in frequency domain (or) time domain. In frequency domain equalization, frequencies characteristics of equalizers and channels are synthesized to meet no inter symbol interference condition by modifying frequency characteristics of system. In time domain equalization impulse response of equalizers and channel are used to meet no inter symbol interference condition [7]. Time domain equalization is the most commonly used method. Figure 1 shows schematic of channel equalization by utilizing adaptive filter compensating for channel distortions. Its transfer function is an estimate and information transmitted by channel at that instant of time[8].In figure 1, x(n) is input signal ,e(n) is output signal, d(n) is the desired signal, L is the delay period, channel is Rayleigh fading and additive white Gaussian noise is used here.

Inter symbol interference is multiplication interference and it depends on presence (or) absence of signal .If the overall transfer characteristics $H(W)= G_T(W)C(W)\ G_R(W)$ of band limited channel is close to ideal rectangular features, then the output signal, equivalent to signal passed through transceiver device and channel will have reduced inter symbol interference. Let $H'(W)=G_T(W)T(W)$ and $H'(W)$ is suited to conditions of nyquist’s theorem

$$\left(\sum_{i=-\infty}^{\infty} H'\left(W+\frac{2i\pi}{TS}\right)\right)=T_S,\ |W|\leq\frac{2i\pi}{TS}$$

to meet equalization[4]



Adaptive filter adjusts current filter parameters automatically based on previous filter parameters. Adaptive filter involves two basic process which include filtering process, to produce an output in response to a input data and adaptive process, to provide a mechanism for adaptive control of adjustable set of parameters used. Adaptive filter structures are of two types Finite Impulse Response(FIR) and Infinite Impulse Response(IIR).The FIR filter is most preferred and widely used because it has only adjustable zeros and is free of stability problem associated with adaptive IIR filters. In this work FIR transversal filter structure is used as adaptive filter structure and is shown in Fig 2. X(n),d(n) are two input signals and y(n),e(n) are two output signals to adaptive filter Output signal y(n) is generated by processing input signal x(n) through adjustable parameters of filter .The adaptive algorithm is used to adjust the filter parameters to minimize the error signal e(n) expressed in (1) ,

$$e(n) = d(n) - y(n) \tag{1}$$

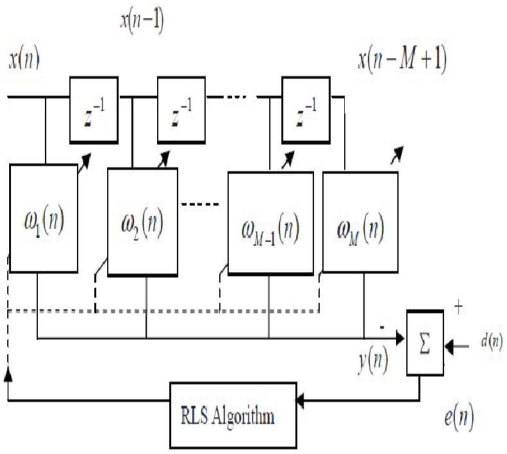


Fig 2 Adaptive trasversal filter structure

Tap input signals x(n),x(n-1),…………… x(n-M+1) are elements of tap input vector X(n) of order M×1 . M-1 is the number of delay cells where M is number of taps(filter length) and n is variable data length .In this filter model, order M is less than (or) equal to n [9].In this work M is taken as 12. Elements of tap weight vector W(n) are tap weights $w_0(n),w_1(n),w_2(n),\ldots.w_{M-1}(n)$.If desired signal d(n) is given, adaptive filter can calculate priori estimation error e(n).

III. RLS ALGORITHM

The weights of adaptive filter adjust automatically based on some criteria to yield better estimate of desired signal as output .The criteria according to which filter weights are adjusting corresponds to different adaptive algorithms .In this work RLS algorithm is used in which the minimization criteria is error square average value .The cost function J(n) constituted by sum of error square become minimum by selecting tap weights and next update the weights to achieve optimal adaptive iterative algorithm based on estimate gradient vector of input signals in iterative process

$$e(n)=d(n)-y(n)=d(n)-x(n)*w(n) \tag{2}$$

The cost function of adaptive RLS algorithm is constituted by sum of exponentially weighted error signal[9] which is shown in (3)

$$J(n) = \sum_{i=1}^{n} \lambda^{m-1} |e(i)|^2 \tag{3}$$

λ is called forgetting factor (or) weighting factor and it is $0<\lambda\leq 1$ .when i is closer to n the value of λ is larger and when i is far away from n , λ value is very small . here n is variable data length.

To minimize sum of error squares, $\dfrac{\partial J(n)}{\partial w} = 0$.

Tap weight vector update equation is given by

$$w(n) = w(n-1) + k(n)e^T(n) \tag{4}$$

Here k(n) is gain vector and given by

$$k(n) = \frac{p(n-1)x(n)}{\lambda + x^T(n)p(n-1)x(n)} \tag{5}$$

P(n) is inverse autocorrelation matrix of input signal. $R_x(n)$ is deterministic input autocorrelation matrix and given by

$$p(n) = R_x^{-1} = \lambda^{-1}p(n-1) - \lambda^{-1}k(n)x^T(n)p(n-1)$$
(6)

Tap weight vector initial value is taken as zero. (w(0)=0). In cases of non-stationary, the initial value p(0) is

$$p(0) = \delta^{-1}I \tag{7}$$

I is unit matrix [6] and δ is regularization parameter which is usually small positive number like 0.01 (or) smaller.

IV. SIMULATION RESULTS

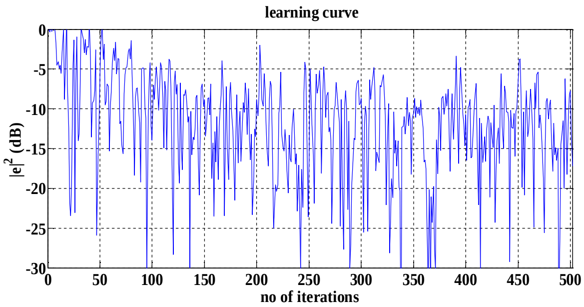


Fig 3 Learning curve for BPSK

The variation in mean square error in db with respect to number of iterations which is called as learning curve is shown in fig.3. It shows that mean square error decreases by a considerable amount and is about -10db after 100 iterations .

The constellation diagram for BPSK scheme before and after equalization is shown in fig.4 .The BER of received constellation before equalization is 0.51 and after equalization BER has come down to 0.001 i.e before equalization 51%error was present after equalization 0.1% error is present .It is observed that error has decreased by 50.9% due to equalization.

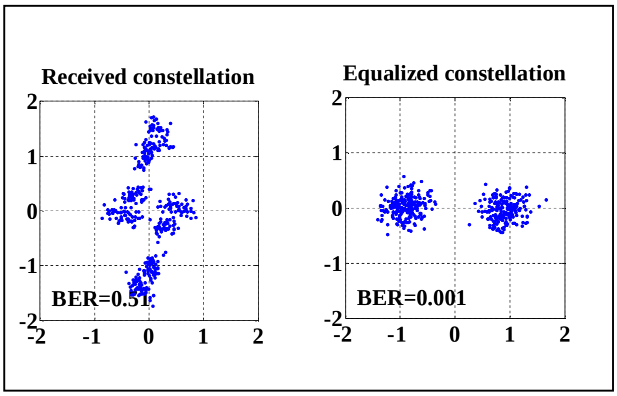


Fig 4 constellation diagram of BPSK(a)before equalization and (b)after equalization.

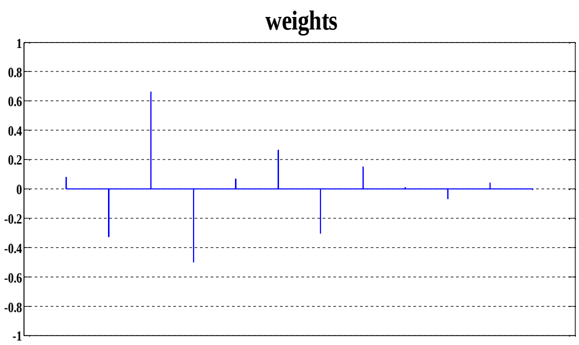


Fig 5 Filter tap weights after equalization for BPSK scheme

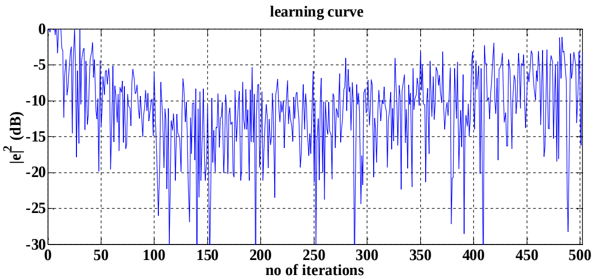


Fig 6 Learning curve for QPSK

The converged filter tap weights after equalization is shown in fig.5. The variation in mean square error in db with respect to number of iterations for QPSK scheme is shown in fig.6. It shows that mean square error decreases by considerable amount and is about -10db after 150 iterations.

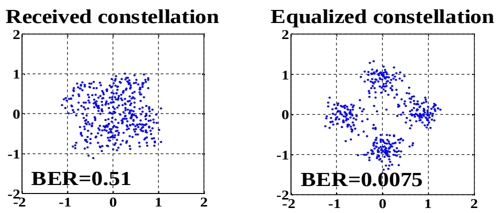


Fig 7 constellation diagram of QPSK(a)before equalization and (b)after equalization.

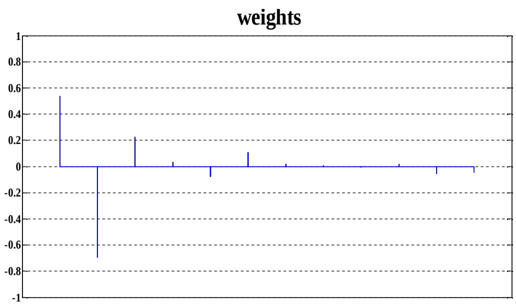


Fig 8 Filter tap weights after equalization for QPSK scheme

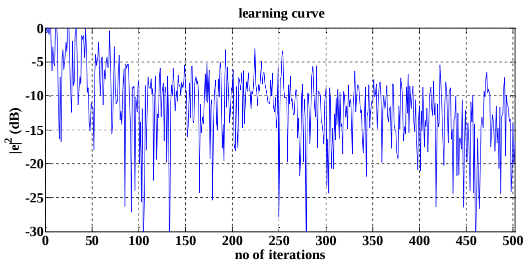


Fig 9 Learning curve for 8PSK

The constellation diagram for QPSK scheme before and after equalization is shown in fig.7 .The BER of received constellation before equalization is 0.51 and after equalization BER has come down to 0.0075 i.e before equalization 51%error was present after equalization 0.75% error is present .It is observed that error has decreased by 50.25% due to equalization. The converged filter tap weights after equalization is shown in fig.8.

The variation of mean square error in db with respect to number of iterations for 8PSK scheme is shown in fig.9 .It shows that mean square error decreases by considerable amount and is about -10db after 200 iterations.

The constellation diagram for 8PSK scheme before and after equalization is shown in fig.10 .The BER of received constellation before equalization is 0.54 and after equalization BER has come down to 0.085 i.e before equalization 54% error was present after equalization 8.5% error is present. It is observed that the error decreases by 45.5% due to equalization.

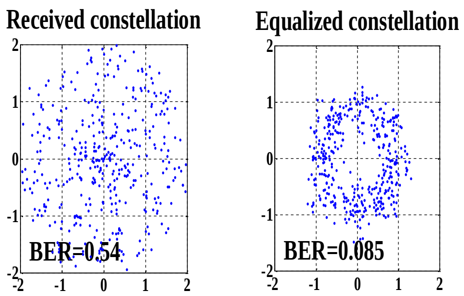


Fig 10 constellation diagram of 8PSK (a) before equalization and (b)after equalization.

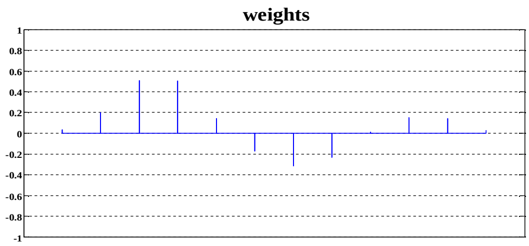


Fig 11 Filter tap weights after equalization for 8PSK scheme.

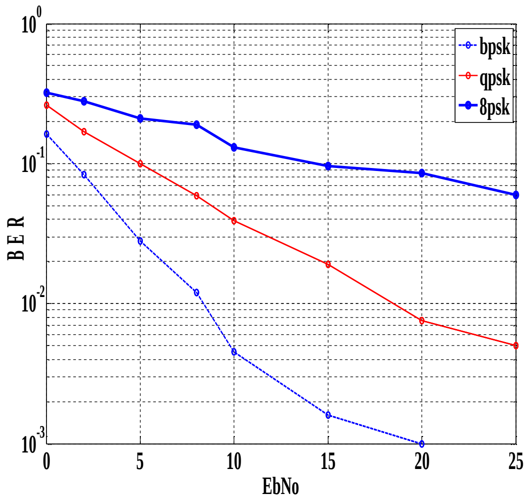


Fig 12 comparison of BER VS Eb/No(db) for BPSK,QPSK and 8PSK schemes

The variations of BER for different values of Eb/No in decibels are shown in fig.12. It shows that BER decreases with increase in Eb/No for BPSK,QPSK,8PSK schemes and BPSK has less BER when compared with QPSK and 8PSK. Decreasing order of Performance is BPSK, QPSK and 8PSK.

V. CONCLUSION

It is observed from the simulation curves that adaptive equalization improved the performance of signal transmission by reducing distortion and inters symbol interference caused by multipath channel effects and BER is decreased by 50.9%, 50.25%, and 45.5% for BPSK, QPSK and 8PSK respectively. It is also observed that as number of signals or number of M increases in M-ary based digital modulation schemes BER decreases monotonically with increasing values of Eb/No. RLS algorithm used in this work meets channel equalization by reducing channel effects. However RLS algorithm needs more computations. The performance can be further improved by using neural network equalization and so on.

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