Adaptive Kalman Filter based Channel Equalizer

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***Abstract*- Channel Equalization is a necessity of the communication system. In this paper a channel Equalizer based on Adaptive Kalman Filter is presented. The performance indexes used for measurement are mean square error (MSE), Rate of convergence and signal to noise ratio (SNR). This analysis is compared with some other Adaptive Equalizer like recursive least square (RLS) and experimental results shows that this approach gives a less mean square error which is better than other equalizer with fast rate of convergence. Also experimented for different communication system like QAM (64 QAM, 16 QAM, 4QAM), QPSK and BPSK, results shows that this Equalizer is quite compatible with different digital modulator.**

Keywords*- *Channel Equalization, LMS Filter, RLS Filter and Kalman Filter, AWGN channel.

I. INTRODUCTION

Channel equalization is the process in which the transmitting signal affected by the unwanted signals during transmission process is trying to become noise free [1]. The ISI (Inter Symbol Interference) imposes the main obstacles for achieving increased digital transmission rates with the required accuracy. Traditionally, inter symbol interference problem is resolved by channel equalization. A channel equalizer is an important component of a communication system. The equalizer depends upon the channel characteristics. The recent digital transmission systems impose the application of channel equalizers with short training time and high tracking rate. These requirements turn attention of researcher to channel equalization algorithms.

The channel will affect the transmitting signal because of the channel noise and dispersion which are leading to the Inter symbol Interference (ISI) phenomenon, so it is to pass the received signal at the receiver through an equalizing filter to minimize the channel effect [9]. The adaptive equalizer and the decision device at the receiver compensate the ISI created by the channel. Thus it may be necessary for the channel equalizer to track the time varying channel in order to provide reasonable performance. The main key goal of this adaptive filter based equalizer is to minimize the mean square error of equalized signals before reaching to the receiver.

This paper presents the Adaptive Kalman Filter based Channel Equalizer for AWGN Channel, in section II a Channel Equalizer [2] is explained, in section III Adaptive Filter Applications is discussed, in section IV Channel Equalizer using Adaptive Kalman Filter is analyzed whose experimental results are shown in section V, model performance is concluded in section VI.

II. CHANNEL EQUALIZER

Channel Equalizer is very much needed not only to mitigate the noise effect or ISI but to provide an optimum signal to the decision device at the receiver so that decision device can take a good decision in favor of original signal. The existing equalizers like decision feedback, zero forcing etc [2] are not preferable over Adaptive algorithm based Equalizers. Unlike normal digital filters most of the Adaptive Filters are able to work in recursive manner which helps to reduce the computational complexity on the basis of particular application. The key goal of working with Adaptive filters is to minimize the MSE as minimum as possible.

Figure 1 shows the basic process required for channel equalization [3] using adaptive filter. The equalizer plays a important role for providing the equalized input to the receiver.

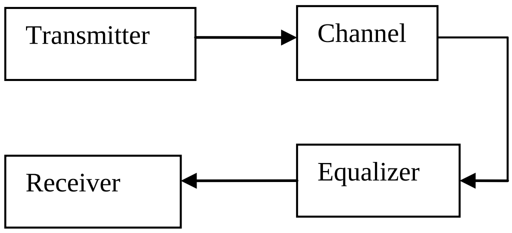

Fig. 1 General block of Channel Equalization

The Equalizer is placed before the receiver so the aim of the equalizer is to produce an output for which the receiver can take appropriate action because the equalizer is the most expensive component of a data demodulator and can consume over 80% of the total computations needed to demodulate a given signal. Figure 2 shows the general steps required for adaptive equalizations using adaptive algorithms.

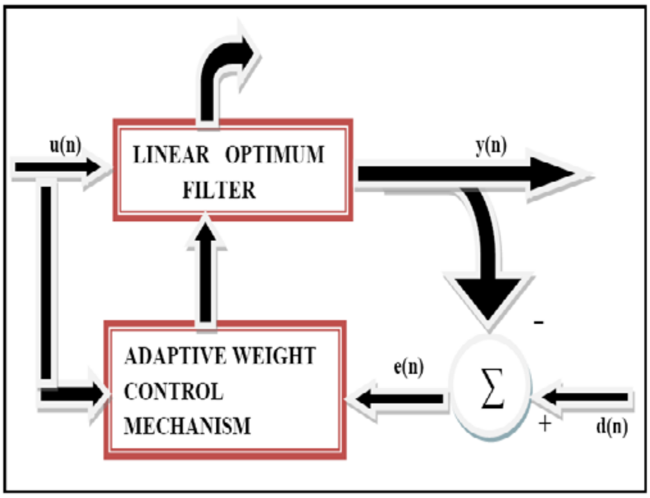

Fig.2 Adaptive Equalization [1]

In Fig.2 the process required for Adaptive equalization is explained Where u(n) is the filter input, y(n) is the filtered output, e(n) is the error between the filtered output and the desired response d(n). The Adaptive Weight Control Mechanism means the Adaptive algorithm which is to be able to update the filter coefficients and adjust the filter weights on the basis of error occurred and again this process is repeated to get the optimum output. An important consideration in the use of an adaptive filter is the criterion for optimizing the adjustable filter parameters. The criterion must not only provide a meaningful measure of filter performance, but it must also result in a practically realizable algorithm.

The most common criterion used in Adaptive filtering is MMSE (Minimum Mean square Error) which is known as cost function. The formula for MMSE is given by-

$$MMSE = E[|e_0|^2] \qquad (1)$$

Where e_0 is the optimum error between the measured output y (n) and desired signals d (n).

$$e_0(n) = y(n) - d(n) \qquad (2)$$

Adaptive filters have very effective features so in next section application of adaptive filter is discussed and gives exposure about them.

III. ADAPTIVE FILTER APPLICATIONS

The ability of an adaptive filter to operate satisfactorily in an unknown environment and track time variations of input statistics makes the adaptive filter a powerful device for signal processing and control applications [8]. Indeed, adaptive filters have been successfully applied in such diverse fields as communications, radar, sonar, seisinology, and biomedical engineering. Although these applications are indeed quite different in nature, nevertheless, they have one basic common feature: an input vector and a desired response are used to compute an estimation error, which is in turn used to control the values of a set of adjustable filter coefficients. Adaptive filters are self-designing using a recursive algorithm. And it is useful if complete knowledge of environment is not available a priori. There are lots of applications [4] where Adaptive filters playing a key role some of them are discussed below-

A. Equalization and Inverse Modeling

The basic idea of inverse modeling, also known as deconvolution or equalization [10], the input signal is filtered through a physical system, which might be a communication channel for instance.

B. Adaptive Linear Prediction

Adaptive Linear prediction is the process where the next state prediction is found out for the upcoming process or simply the delayed input is calculated for next state.

There are some most common Adaptive filters which can be used for equalizations with noticeable performance based on particular application such as LMS (Least Mean Square), RLS (Recursive Least Square) [5] and Kalman filter. The least-mean-square (LMS) algorithm [6] is a member of the family of stochastic gradient algorithm and it is a linear adaptive algorithm [1] that consists of two basic processes filtering process and an adaptive process.

In next section a new approach based on Kalman filer is presented. Our interest in the Kalman filter is motivated by the fact that it provides a unifying framework for the derivation of an important family of adaptive filters known as recursive least-squares filters, by which not only the rate of convergence but computational complexities will also be reduced as compared to other Adaptive algorithms [11].

IV. KALMAN FILTER BASED EQUALIZER

A distinctive feature of the Kalman Filter is that its mathematical formulation is described in terms of state-space concepts and also its solution is computed recursively applying without modification to stationary as well as non-stationary environments. It is known from the theory [1], that the Kalman filter is optimal in case that the model perfectly matches the real system, the entering noise is white and the covariance of the noise is exactly known. Because Kalman filter is a recursive estimator this means that only the estimated state from the previous time step and the current measurement are needed to compute the estimate for the current state. In contrast no history of observations and/or estimates is required.

A model which is employed in this paper consists of Random integer generator, a modulator, AWGN channel, and Adaptive Kalman Filter as an Equalizer, and is implemented using MATLAB block set as shown in Figure3.

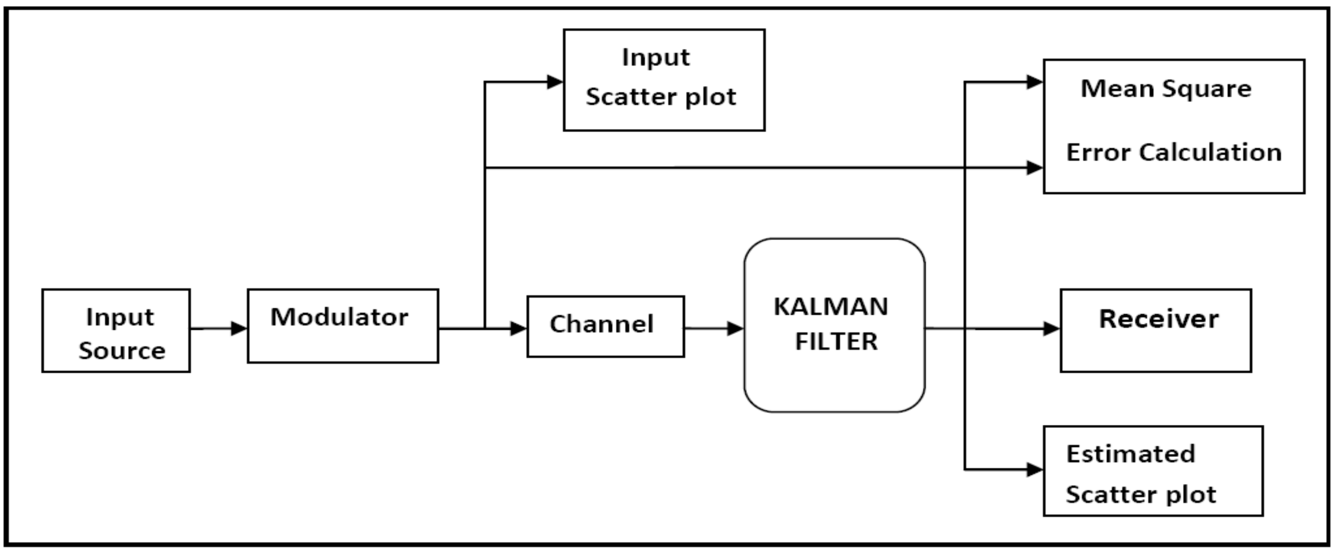

Fig. 3 Adaptive Kalman filter based Channel Equalization

The input source generates uniformly distributed random integers in the range [0, M-1], where M is the M-ary number. The M-ary number can be either a scalar or a vector. If it is a scalar, then all output random variables are independent and identically distributed. If the M-ary number is a vector, then its length must equal the length of the initial seed. If the initial seed parameter is a constant, then the resulting noise is repeatable.

Modulation is an important stage of communication. Modulator is needed before transmitting the signal. Here Different kind of modulator such as, BPSK (Binary phase sift keying), QAM (Quadrature amplitude modulation), QPSK (Quadrature phase sift keying) have been used for evaluating the performance of the Kalman Equalizer. As shown in Fig.3 a modulator is placed before the channel for necessary up gradation for transmitting the signal The Additive White Gaussian Noise (AWGN) Channel is applied and Kalman filter [7] is modeled to equalize this AWGN channel.

Kalman Filter mathematically Kalman Filter works in two steps namely Time update and Measurement update. The Time update is simply meaning the prediction and Measurement update means the correction on that value.

Process Equation:

$$x(n+1) = A\,x(n) + v(n) \qquad (3)$$

Where A is the M by M transition matrix and $v(n)$ is the M by 1 process noise vector and x(n) is the M by 1state at time n:

Measurement Equation:

$$y(n) = Bx(n) + w(n) \qquad (4)$$

In this model Kalman filter is placed before Receiver and it performs a very important role between channel and receiver. And equalized signal is provided to the receiver for necessary processing to get the transmitted signal. Mean square error (MSE) calculation as shown in Figure 3. MSE is taken as the cost function for calculating the performance of the Equalizer, shown in next section. The formula is given in Equation (1) mentioned in earlier section. Scatter plots are used for seeing the channel response. In next section the experimental analysis is taken place to evaluate the performance of the Adaptive Kalman filter based Adaptive Equalizer.

V. EXPERIMENTAL ANALYSIS

In this paper random sequence with 4 QAM modulated signal is applied as an input to the channel. No. of filters used in Adaptive Kalman filter are 10. The process noise covariance and measurement noise covariance applied to the Kalman filter are 0.1 and 0.01respectively.

A. Performance Analysis

The experimental results which have been taken for Kalman Filter based Channel Equalizer are shown in following figures.

1. Scatter plot of channel input:

Scatter plot of input applied at the AWGN channel is shown in figure 4.

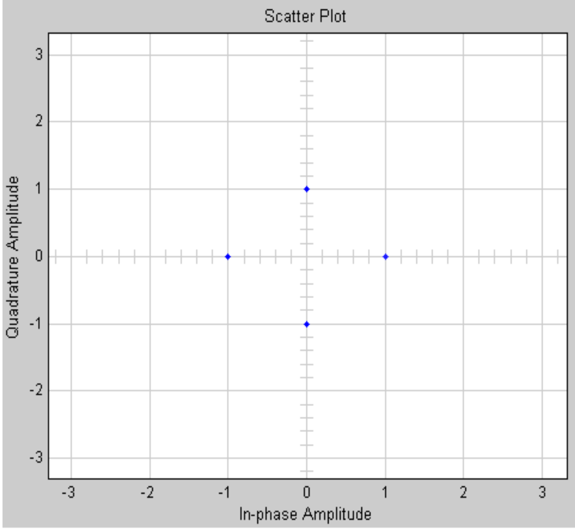

Fig. 4 Scatter plot of Channel Input

Fig. 4 shows the scatter plot of channel input taken out using 4 QAM modulation technique.

2. Scatter plot of channel input:

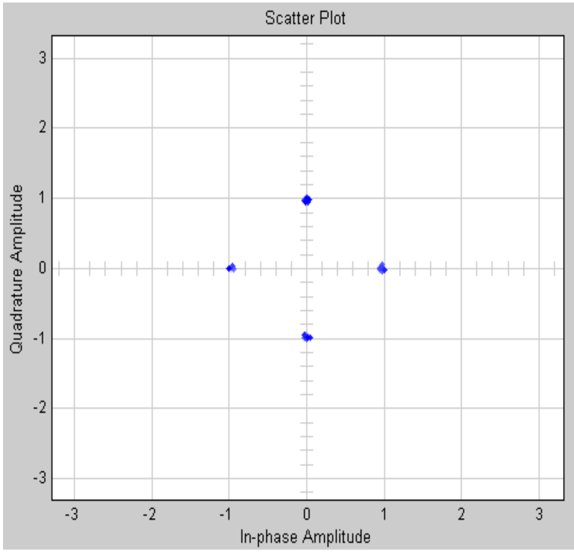

Fig. 5 Scatter plot of Equalized channel Output

In Fig. 5 the equalized output using Adaptive Kalman filter is shown. The Equalized Plot shown in Fig.5 is very close to the Channel input. The equalized scatter plot is taken using 4 QAM Modulator because it gives little better performance than other modulator as it is shown in Table 1.

3. MSE versus No. of Iterations:

Fig. 6 shows Mean square error with Iterations. From the Fig. 6 minimum mean square error achieved in Adaptive Kalman Filter based Equalizer is 0.004 which is very less as compared to RLS equalizer [11]. Also the rate of convergence is fast than RLS equalizer [11]. The expanded graph of MSE with iterations shows that using this new method the MSE can be reduced up to a remarkable level as compared to RLS Equalizer.

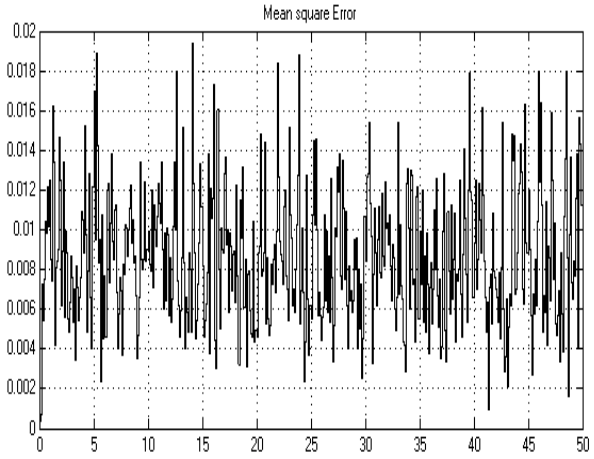

Fig. 6 MSE - vs. - Iterations

It is clear from the Fig.6 that the proposed equalizer not only converges fast but with a significant value of Mean Square Error (MSE).
As it is known that Adaptive Kalman filter has very good capability of estimation of channel coefficients which is one of the demanded applications [8] in many fields.

4. Estimation of channel coefficient:

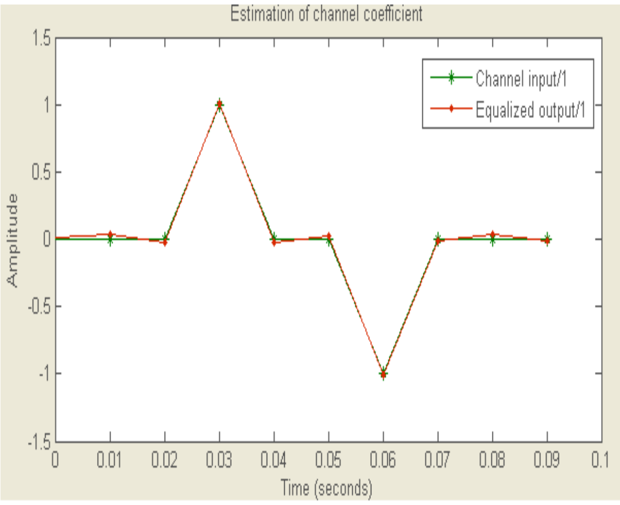

Fig. 7 Estimation of channel coefficient using Kalman Filter

From Fig.7 it is observed that Adaptive Kalman filter based Equalizer have good capability of estimation of channel coefficients.

B. Parameter based analysis

SNR Versus MSE for different modulator-The parameter indexes used for analysis are Mean Square Error, Signal to Noise Ratio and No. of Iterations. In Table 1, comparison of Mean Square Error with different Signal to noise Ratios values for different modulator is done. It is observed that 4QAM gives the best performance at 40 dB SNR.

The MMSE observed for Kalman Equalizer is 0.004, which is superior than other Adaptive Filter based Equalizers [11]. One of the interesting things is that with different modulators Kalman equalizer posses' consistence performance. This is concluded from the Table 1 that all Modulator gives very close performance so this proposed method is not a modulator dependent.

TABLE 1. COMPARISON TABLE WITH DIFFERENT MODULATORS

SNR(dB)	MSE (Mean Square Error)				
	BPSK	64 QAM	16 QAM	4 QAM	QPSK
10	0.5057	0.4943	0.4929	0.4932	0.5427
15	0.1577	0.1478	0.1467	0.1466	0.1753
20	0.05326	0.04421	0.04326	0.04284	0.05983
25	0.02336	0.01478	0.01394	0.01335	0.02383
30	0.01567	0.007353	0.006566	0.005882	0.01267
35	0.01423	0.006061	0.005306	0.00401	0.009284
40	0.01433	0.006246	0.005509	0.00474	0.008291
45	0.01467	0.006639	0.005912	0.00457	0.008021
50	0.01496	0.006955	0.00623	0.00543	0.007961

Table 1 concludes that 4 QAM gives best MSE than other modulators with significant value of MSE. Fig. 8 shows the mean square error for different modulators.

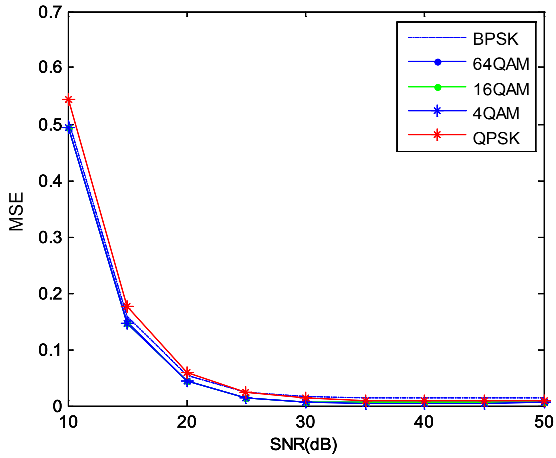

Fig. 8 MSE for Different Modulator at different SRNs

It is very clear that Kalman based Equalizer can be work with any kind of modulator because there is very minute variation with each modulator.

C. Comparison with Other Equalizer

Comparison with other Equalizer mean RLS (Recursive Least Equalizer) [11] has been done and the Comparison Table 2 shows that the Adaptive Kalman Filter based channel equalizer gives better performance than RLS equalizers [11].

TABLE 2 COMPARISONS WITH OTHER EQUALIZER

Type of Adaptive Equalizer	No. of Iteration	Channel	SNR (Signal to noise ratio)	MMSE (Min mean Square Error)
RLS Equalizer[11]	450	Multipath Channel	--	0.025
Kalman Equalizer	50	AWGN Channel	40dB	0.004

From Table 2. That Kalman Equalizer takes less no. of iterations than RLS [11] and minimum mean square error is very less than RLS. The rate of convergence in Adaptive Kalman Filter based Equalizer is 50 which is less than RLS equalizer, also the MMSE (Minimum Mean Square Error) which is obtained using 4QAM modulator is 0.004 also less than RLS Equalizer [11]. Table 1 and 2 shows that Kalman Equalizer gives consistence performance at High SNR values, which is required in some applications.

VI. CONCLUSION

A new approach of channel equalization based on Adaptive Kalman filter has been presented in this paper. This paper concludes that the proposed equalizer for AWGN channel is much more superior to RLS equalizer. This method of channel equalization has less computational complexity as a measure of rate of convergence also this approach gives less Minimum Mean Square Error (MMSE) than RLS [11] in less iteration. All the experiments with modulator shown in Table 1 and figure 7 also show that this Equalizer is quite compatible with different digital modulator.

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