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Adaptive Synchronization of Rayleigh Fading Channel Using Kalman Filter

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Abstract: ***In this paper, we use the adaptive kalman filtering to approximate the impulse response of the system such that the impulse response of the system is synchronized with the ideal system response. Adaptive synchronized Rayleigh fading channel requires the knowledge of channel information. To evaluate the performance of the system knowledge of channel parameter is the basic requirement. Kalman filter predict the value of the weights then measure the weights and then update the real values of the weights. It is also helpful to determine the minimum mean square error that is MMSE. It is purely a time domain filter where as other filters are formulated in the frequency domain and then transformed back to the time domain for implementation. The Results reveals that the performance of the Rayleigh fading channel in terms of kalman gain and minimum mean square error (MMSE).***

Index Terms: **Kalman filtering, Adaptive Synchronization, Kalman gain and Minimum mean square error.**

I. INTRODUCTION

For the last few decays, the application of signal processing of adaptive algorithm for linear channel equalization [1]. The further increasing demand on high data rates in wireless communication systems has arisen in order to support broadband services. To combat the channel dynamics, the channel approximation algorithm is frequently used for rapid convergence and improved MSE performance [2]. But it requires optimum forgetting factor such that the estimator error is minimized. Many problems in science require estimation of the state of a system that changes over time using a sequence of noisy measurements made on the system. In this paper, we will concentrate on the state-space approach to modelling dynamic systems, and the focus will be on the discrete-time formulation of the problem. There are numerous fields that require the use of estimation theory. Some of these fields includes Interpretation of scientific experiments, Signal processing [3]-[5], Clinical trials, Opinion polls, Quality control, Telecommunications [6], Control theory, Actuator changes with time and Network intrusion detection system [7]. The problems of identification, state estimation with tracking, and adaptive control of systems with unknown parameters have been studied mainly in the fields of control and aerospace for over 20 year [8]. In the area of signal processing similar ideas are known as ‘‘adaptive algorithms” whereas in statistics the methods are usually called ‘‘sequential parameter estimation [9]. In this paper, we use the adaptive kalman filtering to approximate the impulse response of the system such that the impulse response of the system is synchronized with the ideal system response. Adaptive synchronized Rayleigh fading channel requires the knowledge of channel information. To evaluate the performance of the system knowledge of channel parameter is the basic requirement. kalman filter predict the value of the weights then measure the weights and then update the real values of the weights. It is also helpful to determine the minimum mean square error that is MMSE. The rest of the paper is organized as follows: In section II, explain the channel estimation used for the approximation of the impulse response of the channel. Section III explains the Kalman filter for filter gain and minimum mean square error (MMSE). In Section V, shows the true values of weight, estimated tap weight for the approximation of the impulse response of the channel using extended kalman filter. Finally, a conclusion is made.

II. CHANNAL ESTIMATION

Channel: In its most General sense can describe everything from the source to destination of the radio signal including the physical medium. Here the “Channel” refers to the physical medium.

Channel Model: It represent the mathematical model of the transfer characteristics of the physical medium. By observing the characteristics of the received signal the Channel is model.

Estimation: The term estimator is related to the filtering. It is used to refer to a system i.e. designed to extract the information about a prescribed quantity of interest from the noisy data.

Channel Estimation: The term channel estimation allow the receiver to approximate the effect the channel on the signal.

In its most general sense, a channel can describe everything from the source to the sink of a radio signal. This includes the physical medium between the transmitter and the receiver through which the signal propagates. An essential feature of any physical medium is, that the transmitted signal is received at the receiver, corrupted in a variety of ways by frequency and phase-distortion, inter symbol interference and thermal noise [8]-[12]. The channel estimation model is given in figure 1.

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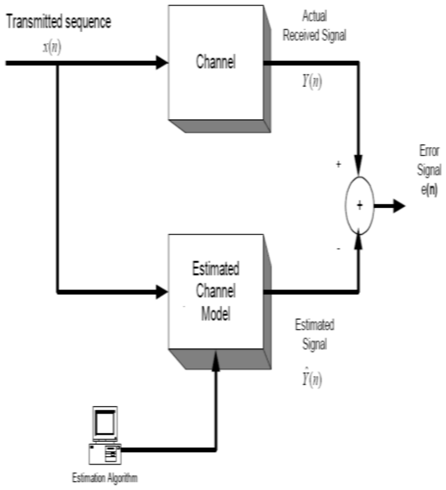

Fig. 1 Channel Estimation Process

A channel model is a mathematical representation of the transfer characteristics of this physical medium. On basis of physical phenomenon or it could be formed by fitting the best mathematical/statistical model on the observed channel behaviour the model based. Different mathematical models that explain the received signal are then fit over the accumulated data. Usually the one that best explains the behaviour of the received signal is used to model the given physical channel.

III. KALMAN FILTERING

The kalman filter is a recursive estimation. The Kalman filter also allow us to predict the Tap weights before the actually Tap weights are calculated. Kalman is an adaptive filter which contains various advantages over filters. The kalman estimator is shown in figure 2.

Fig. 2 Kalman Estimator

IV. KALMAN FILTERING FOR CHANNEL ESTIMATION

In this section, we describe how the kalman filter is used to estimate the channel. The process of the kalman filter is explained in fig. 3. As now that kalman filter provide the prediction of the data before actually data obtain. Firstly Kalman filter predict the value of the tap weights and then proceeds to measure the actual weights. After obtaining the real value the kalman filter is used to calculate the kalman gain [5]. This kalman gain is helpful to estimating the error from the signal. The process of estimating the coefficient of the channel is shown in fig, 4. The diagram explained that firstly kalman filter predict the value of the weights. Then it estimates the actual value of the weights. This value is called estimated tap weights of the channel. Then kalman gain is calculated by applying the kalman gain formula. After calculating the kalman gain the minimum mean square error is calculated.

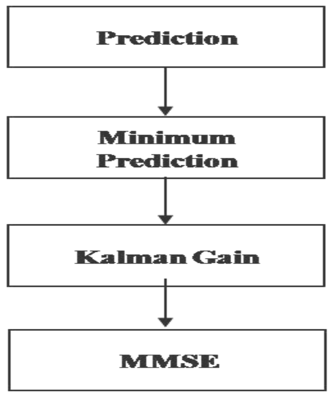

Fig. 3 Basic Process of Kalman Filter

Many transmission channels can be characterized as being linear but not time invariant. These are referred to by various names such as fading dispersive channels or fading multi path channels.

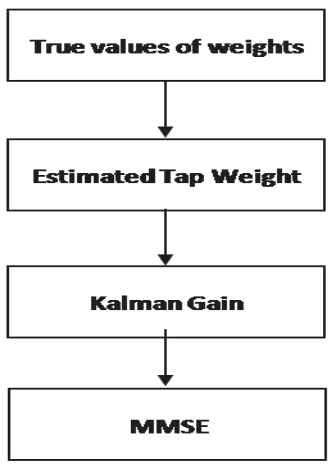

Fig. 4 Estimate Coefficient of Channel

They arise in communication problems in which the troposphere is used as a medium or in sonar in which the ocean is used. In either case medium act as a linear filter causing an impulse to appear as a continuous waveform at the output as shown in the figure 5. This effect is a result of a continuum of propagation paths, i.e. multi path, each of which delays and attenuates the input signal. Additionally, however, a sinusoid at the input will appear as a narrow band signal at the or one whose amplitude is modulated (the fading nature). This effect is due to the

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changing character of the medium, for example, the movement of the scatterers. A little thought will convince that the channel is acting as a linear time varying filter. If we sample the output of the channel, then it can be shown that a good model is the low-pass tap delay line model as shown in next figure. This input-output description of this system is

$$y[n] = \sum_{k=0}^{p-1} h_n[k]v[n-k] \quad (1)$$

and the curves depicting the input output relationship is shown in the figures below

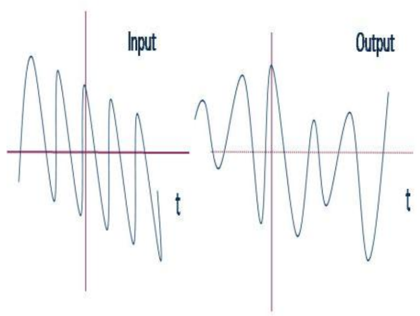

Fig. 5 input and output signal of the fading Channel

This is really nothing more than a FIR filter with time varying coefficients. To design effective communication or sonar systems it is necessary to have knowledge of these coefficients. Hence the problem becomes one of estimating $h_n[k]$ based on the noise corrupted output of the channel

$$x[n] = \sum_{k=0}^{p-1} h_n[k]v[n-k] + w[n] \quad (2)$$

Where $w[n]$ is the observation noise.

From above equation the observations for p=2 and assuming that $v[n]$=0 for n<0, are

$$x[0] = h_0[0]v[0] + h_0[1]v[-1] + w[0] = h_0[0]v[0] + w[0] \quad (3)$$

$$x[1] = h_0[0]v[1] + h_1[1]v[0] + w[1]$$

$$x[2] = h_2[0]v[2] + h_2[1]v[1] + w[2]$$

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It is seen that for $n \geq 1$ we have two new parameters for each new data sample. Even without corrupting noise we cannot determine the tapped delay line weights. A way out of this problem is to realize that the weights will not change rapidly from sample to sample.

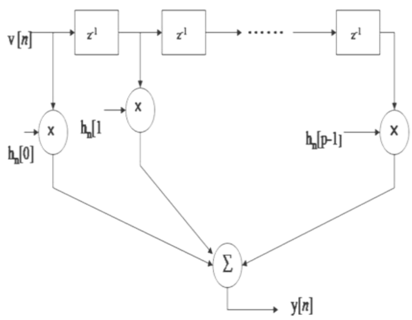

Fig. 6 Tapped Delay Line Channel Model

The use of Gauss-Markov model allows us to fix the correlation between the successive values of a given tap weight in time. Hence, we suppose that the state vector is

$$h[n] = Ah[n-1] + u[n] \quad (4)$$

where $h[n] = [h_n[0]\ h_n[1] \ldots\ h_n[p-1]]^T$, A is a known $p \times p$ matrix and $u[n]$ is a vector WGN with covariance matrix **Q**. A standard assumption that is made to simplify the modelling is that of uncorrelated scattering. It assumes that the tap weights are uncorrelated to each other and hence independent due to jointly Gaussian assumption. As a result, we can let A, Q, and Ch, the covariance matrix of h[-1], be diagonal matrices. The vector Gauss-Markov model then becomes p independent scalar models.

The measurement model is

$$x[n] = [\,[v[n]\ \ v[n-1] \ldots \ldots \ldots .v[n-p+1]]\ h[n] + w[n] \quad (5)$$

$$v^T[n] \quad = \quad = [\,[v[n]\ \ v[n-1] \ldots \ldots \ldots .v[n-p+1]]$$

Where $w[n]$ is assumed to be WGN with variance σ^2 and the $v[n]$ sequence is assumed known (since we provide the input to the channel) [2]. It can now from the MMSE estimator for the tapped delay line weights recursively in time using the kalman filter equations for a vector state and scalar observations. With obvious changes in notation we have

Prediction:

$$\hat{\mathbf{h}}[n|n-1] = \mathbf{A}\,\hat{\mathbf{h}}[n-1|n-1] \quad (6)$$

Minimum Prediction MSE:

$$\mathbf{M}[n|n-1] = \mathbf{A}\mathbf{M}[n-1|n-1]\mathbf{A}^T + \mathbf{Q} \quad (7)$$

Kalman Gain:

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$$K[n] = \frac{M[n|n-1]v[n]}{\sigma^2 + v^T M[n|n-1]v[n]} \qquad (8)$$

Correction:

$$\hat{\mathbf{h}}[n|n] = \hat{\mathbf{h}}[n|n\text{-}1] + \mathbf{K}[n]\,(x[n]\text{-}\,\mathbf{v}^T[n]\,\hat{\mathbf{h}}[n|n\text{-}1]) \qquad (9)$$

Minimum MSE:

$$\mathbf{M}[n|n\text{-}1] = (\mathbf{I} - \mathbf{K}[n]\,\mathbf{v}^T[n])\,\mathbf{M}[n|n\text{-}1] \qquad (10)$$

And is initialized by $\hat{\mathbf{h}}[-1|-1] = \boldsymbol{\mu}_h$, $\mathbf{M}[-1|-1] = \mathbf{C}_h$.

V. SIMULATION RESULTS

Matlab simulation results of time varying channel approximation for impulse response using kalman filter. The channel is act as a tapped delay line modelled with n no of weights.

$$A = \begin{bmatrix} 0.554 & 0 & 0 & 0 \\ 0 & 0.869 & 0 & 0 \\ 0 & 0 & 0.59 & 0 \\ 0 & 0 & 0 & 0.45 \end{bmatrix}$$

$$Q = \begin{bmatrix} 0.01 & 0 & 0 & 0 \\ 0 & 0.01 & 0 & 0 \\ 0 & 0 & 0.01 & 0 \\ 0 & 0 & 0 & 0.01 \end{bmatrix}$$

$$\hat{h}[-1|-1] = 0$$

$$M[-1|-1] = 40I$$

$$\sigma^2 = 0.1$$

The no. of tap weights is assumed as per requirement. Fig. 7 shows the Adaptive gain different filter coefficient. Fig. 8 demonstrates the minimum mean square error for different filter coefficient.

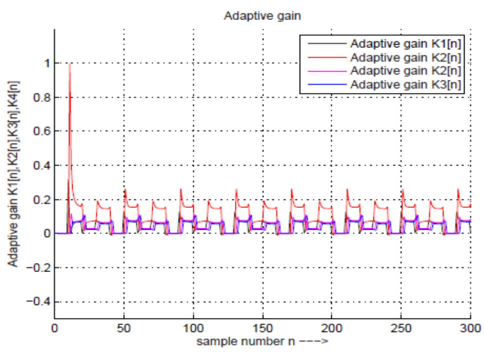

Fig. 7 Adaptive gain different filter coefficient

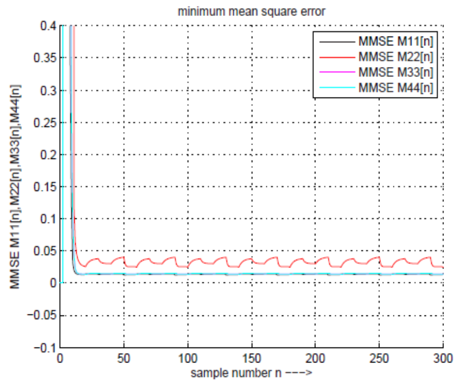

Fig. 8 minimum mean square error for different filter coefficient

VI. CONCLUSIONS

In this paper, we use the adaptive kalman filtering to approximate the impulse response of the system such that the impulse response of the system is synchronized with the ideal system response. To evaluate the performance of the system knowledge of channel parameter is the basic requirement. Kalman filter predict the value of the weights then measure the weights and then update the real values of the weights.

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