Fading Channel Prediction Using Adaptive Linear Predictor In QPSK System

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Abstract – **In this paper we consider performance evaluation of QPSK transmission system operating in a frequency nonselective fading channel. A algorithm that combines decision feedback and adaptive linear prediction (DFALP) [1] by using tentative coherent detection and the least mean square (LMS) algorithm is used for tracking the phase and amplitude of the fading channel. The channel gain is predicted by FIR and IIR filters employing LMS algorithm. Simulation results are given for the system's performance using FIR and IIR filters. The special case of IIR filter having length of unity is well-known Kalman-LMS filter.**

Keywords – **fading channels, adaptive filtering, channel identification**

I. Introduction

To detect an information sequence transmitted coherently and reliably over a fading channel, it is necessary to estimate the channel phase and amplitude. This is motivated by the fact that coherent detection of signals over fading channels is superior to non-coherent detection if accurate channel state information (CSI) is available.

One approach was proposed by Moher and Lodge [2] to track frequency nonselective fading channels, where one training symbol is sent for every K_t - 1 data symbols, and linear interpolation is used to estimate channel gains. This idea was extended by Irvine and McLane [3] using decision-feedback and noise smoothing filters. However, such filtering results in large decision delay.

It is well-known that fading channels are correlated. Therefore, past channel gain estimates may be used to predict the channel gain using linear prediction theory. This paper investigates adaptive linear prediction for fading channel amplitude and phase prediction. Two different predictors are used, both employing LMS algorithm. One of them uses FIR filter, and the other uses IIR filter.

It should be noted that a non-adaptive linear predictor was used by Young and Lodge [4]. However, the algorithm reported in [4] may not outperform a conventional differential detector when the signal to noise ratio (SNR) is less than 20 dB.

In the next Section, we review the fading channel model, and describe the DFALP algorithm using both predictors. Simulation results are presented in Section III.

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II. Fading Channel Model

Let I_k denote a binary information sequence, and x_k the of a low-pass equivalent discrete-time encoder/modulator. The complex signal x_k is transmitted over a frequency-nonselective Rayleigh or Rician fading channel. The received low-pass equivalent discrete-time signal is then

$$y_k = x_k c_k + n_k \tag{1}$$

where c_k is channel gain, a complex Gaussian process with memory. The mean of c_k is $a = \mathrm{E}\{c_k\}$. When $a = 0$, the fading channel is Rayleigh. Otherwise it is Ricean. The covariance function of c_k, is

$$r_{k,k-n} = r_n = E\left\{(c_k - a)(c_{k-n} - a)^*\right\} \tag{2}$$

in general case. A special case of the above model is the Jakes-Reudink fading channel with r_n given by

$$r_n = r_0 J_0(2\pi f_m nT) \tag{3}$$

where $J_0()$ is the zeroth order Bessel function, T is the symbol period and f_m is the maximum Doppler frequency given by $f_m = v / \lambda$, with v and λ defined as mobile vehicle speed and transmission wavelength, respectively.

The channel gain c_k, can be divided into two parts: the line-of-sight (LOS) part with average power a^2 and the random scattering part with average power r_0. The K factor is defined as the ratio $K = a^2 / r_0$. If r_0 is normalized to 1, then $K = a^2$. The K factor is equal to zero for Rayleigh fading channels and is greater than zero for Rician fading channels. The average signal-to-noise (SNR) ratio per symbol is then

$$\gamma_s = \frac{a^2 + r_0}{\sigma_n^2} \tag{4}$$

where σ_n^2 is the variance of the additive white Gaussian noise (AWGN) n_k.

We now describe the algorithm using decision feedback and adaptive linear prediction (DFALP) [1] to track frequency nonselective (flat) fading channels.

If x_k is a known training symbol and if the signal-to-noise ratio (SNR) is high, a good estimate of c_k can be easily computed as

$$c_k \approx \frac{y_k}{x_k} = \widetilde{c}_k \tag{5}$$

according to Eq.(l), where y_k is the received signal. However, most of the received symbols are not training symbols. In these cases the available information for estimating c_k can be based upon prediction from the past detected data-bearing symbols $\bar{x}_i$ $(i < k)$. Since a fading channel is usually correlated, it is possible to use an adaptive linear filter to estimate the current complex channel gain c_k using the past detected symbols $\bar{x}_i$ $(i < k)$ and the current observed signal y_k.

The block diagram of the receiver is shown in Fig. 1.

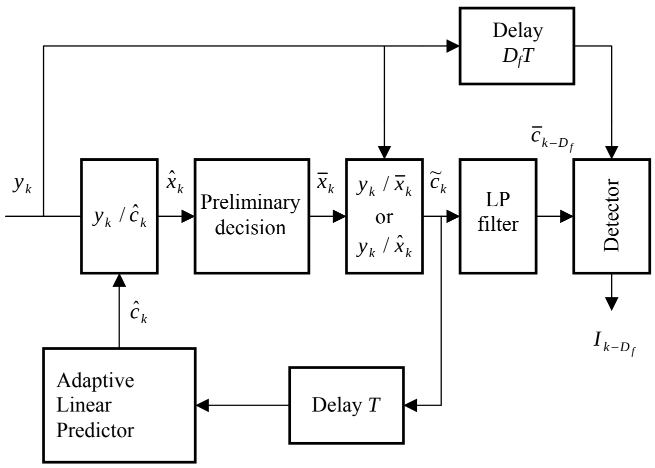

Fig. 1. Receiver block diagram

First, we estimate the data symbol using the predicted channel gain

$$\hat{x}_k = \frac{y_k}{\hat{c}_k} \tag{6}$$

where y_k, is the current received signal plus noise, and $\hat{c}_k$ is a channel estimate given by the linear predictor. Second, we use the minimum distance decision rule

$$\min_{x_k \in D} | \hat{x}_k - x_k | \tag{7}$$

where D is the signal constellation of the modulated complex low-pass equivalent signal x_k. For QPSK, $D = \{e^{jn\pi/4}, n = 1,$ $3, 5, 7\}$. Let $\bar{x}_i$ denote the detected data symbol, i.e.,

$$| \hat{x}_k - \bar{x}_k | = \min_{x_k \in D} | \hat{x}_k - x_k | \tag{8}$$

Using the detected data symbol $\bar{x}_k$, we formulate a new estimate of the channel gain

$$\frac{y_k}{\bar{x}_k} \tag{9}$$

There exist two possibilities for the decision rule (8). One possibility is that the decision is correct, i.e., $\bar{x}_k = x_k$. Then the estimate $y_k / \bar{x}_k$ would be reliable. On the other hand, if the decision is wrong, i.e., $\bar{x}_k \neq x_k$, the estimate $y_k / \bar{x}_k$ will certainly be very poor. To solve this problem, we use a thresholding idea. In most cases, if the decision is correct, the distance between the predicted channel gain $\hat{c}_k$ and the decision feedback estimate $y_k / \bar{x}_k$ would not be large, i.e., the probability that $| \hat{c}_k - y_k / \bar{x}_k | < \beta$ would be high, where β is a chosen threshold. On the other hand, if the decision is wrong, the distance between the predicted channel gain $\hat{c}_k$, and the decision-feedback estimate $y_k / \bar{x}_k$ would be large, i.e., the probability that $| \hat{c}_k - y_k / \bar{x}_k | > \beta$ would be high. Therefore the corrected channel estimate may be expressed as

$$\tilde{c}_k = \begin{cases} y_k / \bar{x}_k & | \hat{c}_k - y_k / \bar{x}_k | < \beta \\ \hat{c}_k & | \hat{c}_k - y_k / \bar{x}_k | \geq \beta \end{cases} \tag{10}$$

There exists no analytical approach to choosing the threshold β. In our experiments we determined optimal value for the treshold β to be 0.7 for both FIR and IIR filter.

The predicted fading channel gain, for FIR filter, at time k is

$$\hat{c}_k = \sum_{i=1}^{N} b_i^* \tilde{c}_{k-i} \tag{11}$$

where

$$(\tilde{c}_{k-1}, \tilde{c}_{k-2}, ..., \tilde{c}_{k-N})^T = \tilde{\mathbf{c}}(k) \tag{12}$$

is a vector of past corrected channel gain estimates and

$$(b_1, b_2, ..., b_N)^T = \mathbf{b}(k) \tag{13}$$

are the filter (linear predictor) coefficients at time k. The superscript T stands for transpose. The constant N is the order of the linear predictor. The LMS algorithm computes the filter coefficients $\mathbf{b}(k + 1)$ of the next time-step using the current filter coefficients $\mathbf{b}(k)$ and the estimation error $\tilde{c}_k - \hat{c}_k$. Formally, the algorithm is

$$\mathbf{b}(k+1) = \mathbf{b}(k) + \mu(\tilde{c}_k - \hat{c}_k)^* \tilde{\mathbf{c}}(k) \tag{14}$$

where μ is the adaptation parameter controlling the convergence rate and steady-state error of the algorithm.

Similarly, for IIR filter, Eqs. (11) and (14) are defined as

$$\hat{c}_k = b_1^* \tilde{c}_{k-1} + \sum_{i=2}^{N} b_i^* \hat{c}_{k-i} \tag{15}$$

$$\mathbf{b}(k+1) = \mathbf{b}(k) + \mu(\tilde{c}_k - \hat{c}_k)^* \hat{\mathbf{c}}(k) \tag{16}$$

The corrected channel estimate $\tilde{c}_k$ is then low-pass filtered using a linear phase low-pass filter (LPF) with $2D_f + 1$ taps to reduce the noise. That is, the final channel gain estimate is

$$\bar{c}_{k-D_f} = \sum_{i=0}^{2D_f} h_i \tilde{c}_{k-i} \tag{17}$$

where h_i is the impulse response of a LPF with $2D_f + 1$ taps. The filter cutoff frequency is equal to Doppler frequency.

III. Numerical Results

In following figures we simulated a typical digital cellular telephone channel, where the carrier frequency is 800 MHz, symbol rate is 24000 symbols/sec, and the fading channel is Rayleigh. The low-pass filter length is set to $D_f = 10^3[-7.526(f_mT)^3 + 3.6729(f_mT)^2 - 0.3981(f_mT) + 0.0153]$, which is determined in [1] to be optimal. In all the following figures, solid line stands for IIR filter performance, and dashed line stands for FIR filter.

Error probability as a function of LMS algorithm adaptation parameter is shown in Fig. 2. It can be seen than the lowest error probability can be achieved for $\mu \approx 10^{-4}$ for both FIR and IIR filter. One can also note that FIR filter is less sensitive to variation of parameter μ than IIR filter, i.e. the error probability for FIR filter is very close to minimal for μ within the range of 10^{-2} to 10^{-5}.

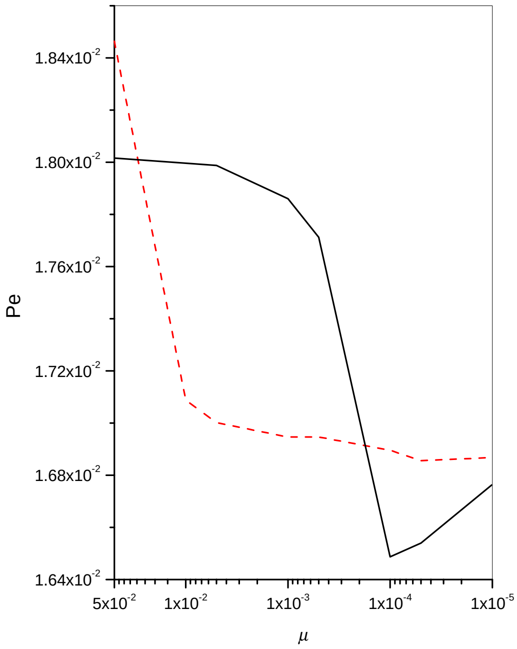

Fig. 2. Error probability as a function of LMS algorithm adaptation parameter (v = 50 km/h)

Figs. 3 and 4 show the error probability as a function of adaptive linear predictor length, for receiver velocity of v = 50 km/h and v = 160 km/h, respectively. Curves labeled with *a* stand for signal to noise ratio of SNR = 5 dB, and curves *b* stand for SNR = 15 dB. It can be noted that FIR filter performs better than IIR one for receiver velocity v = 160 km/h, regardless of signal to noise ratio. On the other hand, for v = 50 km/h and SNR = 15 dB, the predictor employing IIR filter has better performances for $N \leq 5$. Because of that, IIR filter should be used in case of lower receiver velocities and high signal to noise ratios, when high predictor length is not allowed. It should be also noted that the length of IIR filter should be set to a optimal value. The higher the signal to noise ratio the less optimal length is.

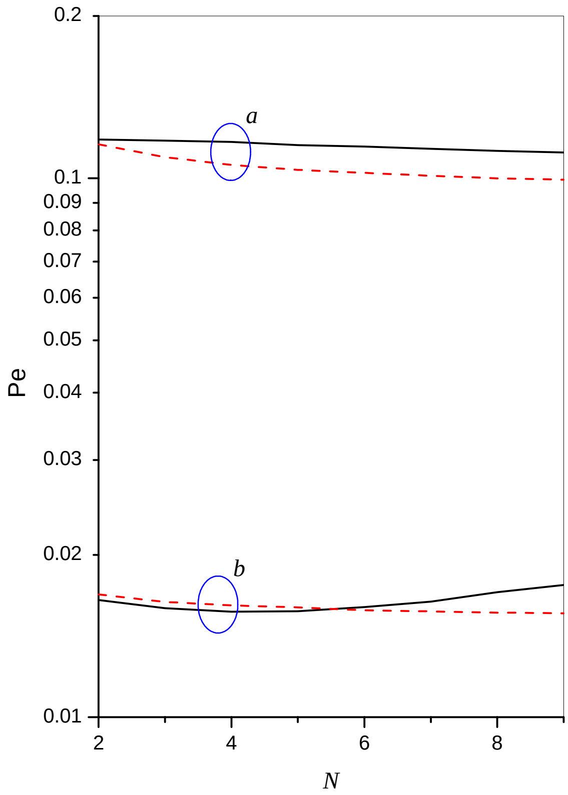

Fig. 3. Error probability as a function of adaptive linear predictor length (v = 50 km/h)

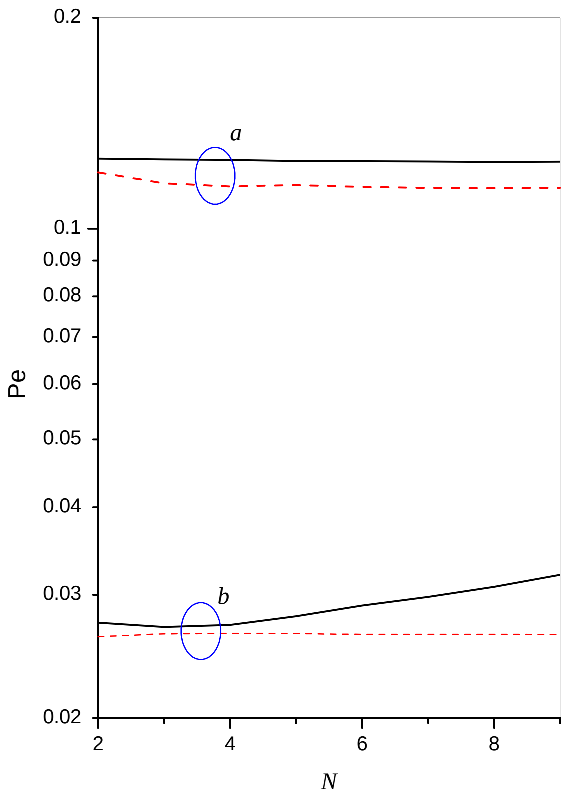

Fig. 4. Error probability as a function of adaptive linear predictor length (v = 160 km/h)

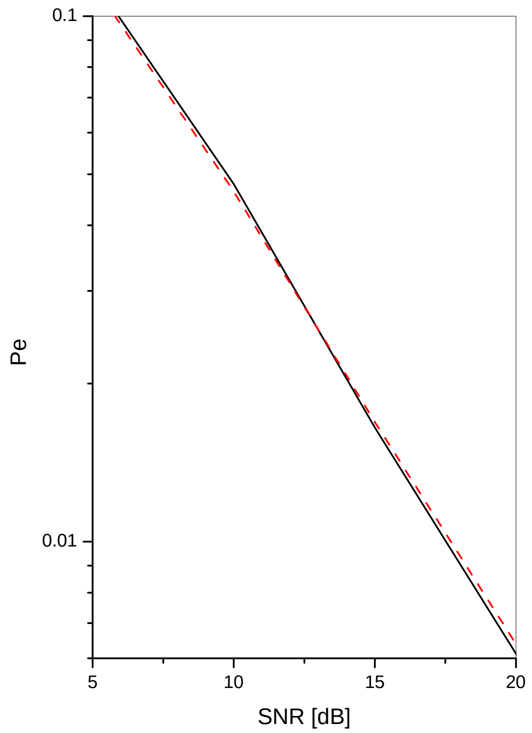

Fig. 5. Error probability as a function of signal to noise ratio (v = 50 km/h)

Error probability as a function of signal to noise ratio is shown in Fig. 5. The adaptive linear predictor length is set to $N = 2$. As in Fig. 3. it can be seen that IIR filter is better for high signal to noise ratio (SNR > 12.5 dB), while FIR filter is better for low signal to noise ratio.

IV. Conclusion

In this paper we considered performances of QPSK transmission system operating in a frequency nonselective fading channel. The channel gain is predicted by FIR and IIR filters employing LMS algorithm. It was shown that FIR filter is less sensitive to variation of LMS adaptation factor μ than IIR filter. Also, the optimal value of μ is approximately 10^{-4} for both filters. IIR filter should be used in case of lower receiver velocities and high signal to noise ratios, when high predictor length is not allowed. For high receiver velocities and low SNR, the FIR filter is better solution.

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