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USING OF AN ADAPTIVE MATCHED FILTER IN SELF ADAPTIVE EQUALIZATION

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ABSTRACT

This paper deals with a new proposal technique for improving the performance of a communication link. This is carried out by using an adaptive filter cascaded with the matched filter at the receiving end. When this proposal is used to equalizing a communication channel , no need to a training data sequence at the start of the system operation. So, it is important to use it in self adaptive equalization [1]. The matched filter is used to maximize signal to noise ratio at the receiving end. Since the matched filter is designed to match the transmitted signal, which is different from the received signal due to the channel effect the output of the matched filter is different from the designed one. An adaptive filter may be used to avoid these drawbacks caused by the channel since it has the ability to track the channel variation. The paper presents a system based on using adaptive filter together with a matched filter at the receiver. To check the performance of the proposal, a BARKER code is used to modulate four level data. At the receiver, the adaptive filter is cascaded with the matched filter and adapts its weights to get ideal auto-correlation function of the BARKER code. The basic idea in the adaptation process is based on the shape of the auto-correlation function, which has alternative zeroes. So, the adaptive filter adapts its output to zero at these instants and stop adaptation at the next instants, which solves the problem of the distortion in the matched filter output. The main advantage of the adaptation processor of the adaptive filter will modify the tap weights at double the baud interval i.e, slower processor is sufficient to be used. Computer simulation is carried out to test the performance of the proposal. It clarifies the improvement caused when the proposal is used.

KEY WORDS

Adaptive matched digital filter

* Modern Academy in Maady

NOMENCLATURE

SAE Self adaptive equalizer
ACF auto-correlation function
ISI inter symbol interference
SNR signal to noise ratio

INTRODUCTION

Coding techniques are important in many areas such as pulse compression in radar systems, data synchronization, system identification, code division multiplexing and digital communication. The matched filter at the receiving end considers the media between transmitter and receiver as ideal and fixed, which is practically impossible. In practical communication systems the performance is greatly affected by ISI, non-stationarity and noise caused by the channel.

Adaptive filters may be used to avoid the drawbacks caused by the channel [2]. If the channel is stationary then the performance surface remains fixed in its coordinate system [3]. So, the adaptation process consists of starting at some point on the surface and seek the minimum point. If the channel is non-stationary the performance surface, will be fuzzy and moving in its coordinates system and the adaptation process consists not only of seeking the minimum but also tracking it as it moves [3].

In the proposed system, Barker code will be used [4, 5]. It is an example of binary codes. The main advantage of binary codes is that they are easy in generate, so the transmitter is simple and also it is easy to implement its matched filter.

This paper is organized as follows. Section 2 gives the basic idea, and theoretical analysis to the proposed scheme. Computer simulation of the communication system is presented in section 3. Also, the effect of various parameters on the output of the receiver is introduced . Finally, conclusion is given in section 4.

BASIC IDEA OF THE PROPOSED SCHEME

Complete block diagram of the proposed scheme is shown in Fig. 1.

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Fig.1 : The block diagram of the proposed scheme

In this Fig. , the following notations will be considered: b(t) is the binary signal, m(k) is the 4 level pulse amplitude modulated signal, c(k) is the encoded signal, x(k) is the received signal, h(k) is the impulse response of the channel, y(k) is the output of the matched filter, z(k) is the output of the adaptive filter and t = kτ where τ is the sampling period and k = 1,2,3, … and b(k) is the data of the training mode.

Ideal output of the matched filter occurs in case of noise free system, stationary channel and no ISI. This ideal output will be the ACF of the Barker code multiplied by m(k) as shown in Fig.2. where m(k) $\in$ {-3,-1,1,3} and all the four values are equally likely to occur i.e. P(-3)=P(-1)=P(1)=P(3)=0.25

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Fig.2: Ideal output of the matched filter

The output of the matched filter differs from the ideal auto-correlation function due to channel impairments mentioned in [1]. To overcome this problem an adaptive filter is used in cascade with the matched filter to adapt it's output to the ideal auto-correlation function shown in Fig.2. From this figure it is noted that the ideal output of the adaptive filter is zero at t= τ,3τ,5τ … . The idea is to adapt the output of the adaptive filter to zero at these instants and stop adaptation at t =2τ,4τ,6τ,…This proposal of adaptation has many advantages; first, in this method there is a complete ignorance about the value of the transmitted bit. Second, the processor used in the adaptive filter modifies the tap weight of the adaptive filter each 2τ seconds. This means that a slower processor is sufficient to be used.

The adaptive filter adapts its output to zero each 2τ interval at instants (k=1,,3,5…). By using the steepest descent algorithm

$$W_{K+2} = W_K + \mu\, e_k Y_K \qquad (1)$$

$$W_{K+1} = W_K \qquad (2)$$

where

$$W_k = [\, w_k^1 \; w_K^2 \; w_K^3 \; \ldots \; w_K^M \,]^T \qquad (3)$$

$$Y_k = [\, y_k \; y_{k-1} \; \ldots \; y_{k-M+1} \,]^T \qquad (4)$$

It is clear that W_{K+2} is the modified vector of W_k and T denotes transpose, and M is the order of the filter. e_k is the error used to control the tap weights, given by

$$e_k = d_k - z_k \qquad (5)$$

In our situation $d_k = 0$, then, $e_k = - z_k$. From (1) and (5) we get

$$W_{K+2} = W_K - \mu\, z_k Y_k \qquad (6)$$

From (3), (4) and (6) we get

$$[\, w^1_{K+2}\; w^2_{K+2}\; w^3_{K+2}\; \ldots\; w^M_{K+2}\,]^T = [\, w_k^1\; w_K^2\; w_K^3\; \ldots\; w_K^M\,]^T - \mu z_k\, [y_k\;\; y_{k-1}\; \ldots\; y_{k-M+1}\,]^T \qquad (7)$$

$$w^j_{k+2} = w^j_k - \mu z_k\, y_{k-j+1} \qquad (8)$$

Where j = 1,2, … ,M. The output of the adaptive filter is given by

$$z_k = \sum_{j=1}^{M} w_k^j\, y_{k-j+1} \qquad (9)$$

The output of the adaptive filter at no adaptation is

$$z_{k+1} = \sum_{j=1}^{M} w_k^j\, y_{k-j+2} \qquad (10)$$

To check the performance of the system, computer simulation will be carried out in the next section.

SIMULATION RESULTS

The simulations are carried out according to the block diagram given in Fig. 3, in which the “channel” includes the effects of the transmitter filter, the modulator, the transmission medium, and the demodulator.

Fig. 3 Block diagram of a communication link used in simulation

The input output relation of the channel is given by:

$$x_k = \sum_{i=k_1}^{k_2} h_i\, d_{k-i} \qquad (11)$$

where $[h_i, -k_1 < i < k_2]$ is the channel impulse response, K_1 and K_2 are the number of channel coefficients. Three channel models are considered. First one has mild distortion and the second has severe distortion. Impulse responses of these models are given in Fig. 4. The second model is a typical telephone channel [7].

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.k .

First model Second model

Fig. 4 Channel Models used in simulation

The third channel model is non stationary. Its impulse response is considered to be time varying according to

$$h_i(k) = h_i(0)[1 + c\,\sin(2\pi k/f)] \qquad , K_1 < i < K_2 \qquad (12)$$

where $2\pi/f$ is the channel variation parameter, $h_i(0)$ is the initial value of the impulse response , and c is a constant. The data to b transmitted are equally distributed over L possible values. We have considered the case of four level data (L=4), with the data alphabet { ±3 , ±1 } modulated by Barker code. The additive noise, $\{n_k\}$, at the channel output is modeled by a zero mean white Gaussian process with power σ^2. The signal-to –noise ratio (SNR) is defined as SNR= d^2/σ^2 , where $d^2= E(d^2_k)$ is the data power.

It is convenient to define the instantaneous mean square error $E\{(d_k-y_k)^2\}$ at the output of the equalizer. Its estimate ϵ_n is calculated over the interval [(n-1)M, nM-1] according to:

$$\epsilon_n=[\sum (d_k - y_k)^2]/M; \qquad n=1,2, \ldots \qquad (13)$$

Where M is few tens of samples . The performance of an equalizer is evaluated in terms of the convergence time T_c and the residual mean square error ϵ_s; T_c is considered as the time after which ϵ_n fluctuates around a constant value and ϵ_s is the average value of ϵ_n after convergence has been settled.

In practice, the main factor used in measuring the system performance is the steady state probability of error, P_e. An approximate relation between the steady state error, ϵ_S , and P_e is derived in the following . This relation is beneficial due to the fact that the evaluation of ϵ_S , from simulations, is much simpler than that of P_e especially in cases where P_e is small. Indeed, in such cases, the time interval needed to evaluate P_e, and consequently the simulation program, will be very long. The evaluation of ϵ_S , on the other hand, can always be carried out over few hundreds of baud intervals.

In the case L=4 considered above, the output y_k of the equalizer can be expressed as:

$$y_k = d_k + e_k \qquad (14)$$

Where e_k is the output error of the equalizer. The function of the decision block is as follow

d_k= -3 if $y_k < 2$, d_k=-1 if $y_k > -2$ and $y_k < 0$,d_k =1 if $y_k > 0$ and $y_k < 2$, and d_k=3 if $y_k > 2$.

Hence, the probability of error is given by:

$$P_e = P(d_k =1)\, P(|e_k|>1) + P(d_k =-1)\, P(|e_k|> 1)$$
$$+P(d_k=-3)\, P(e_k > 1) + P(d_k =3)\, P(e_k < -1) \qquad (15)$$

To simplify the evaluation of P_e d_k is assumed equipropably and e_k is a zero mean Gaussian random variable with the variance ϵ_s,. and P_e is given by

$$P_n = \int_{|x|>d/2}^{\infty} 1/\sqrt{2\pi\varepsilon_n}\exp(-x^2/2\varepsilon_n)dx = 2Q\left(\dot{d}/2\sqrt{\varepsilon_n}\right) \quad (16)$$

Where

$$Q(u) = \int_{u}^{\infty} (1/\sqrt{2\pi})\exp(-z^2/2)\,dz \quad (17)$$

Under these assumptions

$$P_e = 1.5\, Q\,(1/\sqrt{\hat{\varepsilon}_s}) \quad (18)$$

The channel model with mild distortions (the 1st model) is considered with SNR equal to 20 dB. The proposed filter is a transversal filter with length N=11. At the start of the adaptation, the center coefficient is set to unity while remaining ones are set to zero. The results are given in Fig. 5 showing $\epsilon_{n}=\epsilon_n/d^2$ versus n. In this figure, the curve 1 is the evolution of ϵ_n in the case when matched filter used. It is observed that no convergence occurs and the data communication cannot be achieved. When using the proposed SAE, the convergence occurs and ε_s equals .02 which corresponds to $P_e=3*10^{-6}$ (from equation (18)) . Simulation results of the 2nd model with greater distortion than the above model are shown in Fig. 6. In such a case , by using the matched filter alone no convergence occurs. On the other hand, the proposed scheme converges. The steady state error is 0.05 which corresponds, approximately, to an error rate of the value 3×10^{-4} from equation (18) .

To check the performance of the proposed SAE with non stationary channel, the 3rd model is considered. The same results, as discussed in the previous cases, are found (Fig. 7). To show the effect of different values of ISI and SNR , simulations are carried out with many values of ISI and SNR, and the results are shown in Figs. 8,9 and 10 demonstrate the improvement in the performance by using the proposed technique.

CONCLUSION

The results of this paper show that the matched filter output is distorted due to channel defects and using adaptive filter with the matched filter minimizes the distortion. The adaptive filter has the ability of tracking channel variations. The method used in adaptation in the proposed system is suitable for SAE and it modifies the tap weights at some instants and stop modifying at the next instants .Also, it improves the performance of communication links.

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