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A Review on Adaptive Equalizer Algorithms

Engineering
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ABSTRACT

Today's wireless world demands transmitting data without noise. Additive white Gaussian noise(AWGN) gets added to the signal and however due to the presence of multipath channel and band limited channel time spread symbols reaches the receiver and act as noise for subsequent symbols giving rise to intersymbol interference(ISI). Thus the most favored solution to this problem is use Adaptive Equalizer. In this paper we will discuss in brief about the Adaptive equalizer and main algorithms used by equalizer.

I. Introduction (Heading 1)

Wireless communication has gone through major changes in the last few decades. While it mostly had been used for satellite, terrestrial links and broadcasting until the 1970s, cellular and wireless networking and other Personal Communication Systems (PCS) presently dominate the technology of modern wireless communications. Basic Electromagnetic (EM) wave propagation phenomena such as scattering that occur along these paths further increases the number of the paths between the communicator. In this paper we will give a brief idea about basic phenomenon like AWGN and ISI and how the signal can be made noise free by using adaptive equalizer in the channel and then about various algorithms used by the equalizer for overcoming the effects of ISI and AWGN, thus making the signal noise free.

II. Additive White Gaussian Noise

Additive white Gaussian noise (AWGN) is that kind of noise in which the only impairment to communication is a linear addition of wideband or white noise with a constant spectral density (expressed as watts per hertz of bandwidth) and a Gaussian distribution of amplitude. It does not account for fading, dispersion, interference, nonlinearity or frequency selectivity. It produces simple and tractable mathematical models which are useful for gaining insight into the underlying behaviour of a system before these other phenomena are considered.

Main sources of wideband Gaussian noise are thermal vibrations of atoms in conductors (referred to as thermal noise or Johnson-Nyquist noise), shot noise, black body radiation from the earth and other warm objects, and from celestial sources such as the Sun.

Many channels practically are bandpass and often respond with different frequency components to different inputs, i.e., they are dispersive.

Band limited channel, the simplest dispersive channel having the ideal low pass filter channel impulse response results in the effect of spreading of symbol with the adjacent symbols while transmission of sequence of symbols. This results in Intersymbol Interference, resulting in degrading error performance of communication system. However two ways to eliminate the effect of ISI are, first is to design Nyquist pulses, which are band limited transmission pulses which will minimize the effect of ISI and second method is filtering received signal for cancelling ISI added by channel impulse response, and this approach is known as equalization.

III. Intersymbol Interference

In communication intersymbol interference is a kind of distortion in which interference of one symbol with subsequent symbols takes place. It is an unwanted and unreliable phenomenon as the previous symbol act as noise. ISI is generally caused by inherent non linear frequency response or the multipath propagation causing blurring of successive symbols. Errors are introduced in the decision device at the receiver output due to the presence of ISI. Therefore while designing transmitting and receiving filter , the main objective is to deliver data to its destination with minimum error rate and thus minimizing isi. So equalization plays an important role for doing the same.

IV. Causes

- One of the major cause of ISI is multi path propagation where a wireless signal reaches the receiver through many different paths from transmitter. Main causes for the same are reflection, refraction, diffraction etc. as a result of which different versions of the same signal reaches at different times resulting in delay means that part of a symbol will be spread into subsequent symbols thus interfering with correct detection of symbols.
- Where the frequency response is zero above a certain frequency is another cause of ISI, known as band limited channel. Passing of signal from such a channel removes the frequency components above cutoff frequency and can sometime attenuates amplitude and phase of frequency components below cutoff frequency. This filtering will result in spreading of subsequent symbols at receiver and thus spread pulse of individual symbol will interfere with the subsequent symbols.

V. ADAPTIVE EQUALIZER

Applying an equalizer at the receiver will attempt to undo the effect of communication channel by applying an inverse filter .the main aim of equalizer is to remove ISI and AWGN to maximum extent. These equalizers are used for overcoming the negative effects of channel. Adaptive equalizer as also stated in [1] considers channel to be time varying and tries to design an filter whose coefficients will vary as per change of channel thus eliminating ISI and noise at each time assuming the channel to be varying slowly. Figure 1 shows how an adaptive equalizer is implemented in a communication channel generally.

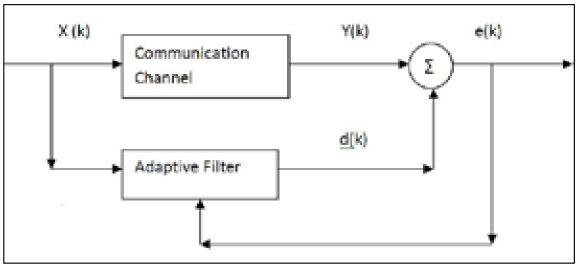

Fig.1. An Adaptive Equalizer

VI. Working

The adaptive equalizer works on following principles

1. The transmitted signal is applied to receive filter which is only a low pass filter that rejects all out of band noise

2. Sampling of receive filter output is done either at symbol rate of twice that of symbol rate
3. This sampled signal is applied to adaptive equalizer whose main aim is to adapt the coefficients for minimizing ISI and AWGN at the output.
4. This adaptation is driven by an error signal

Two basic modes in which adaptive equalizer works are as

1. Decision Directed Mode- in this the decision of the receiver are used by the equalizer to generate error signal and with the help of this error signal equalizer adapts. This method is very effective in tracking slow variations in the channel response but it is not much effective during channel acquisition.
2. Training Mode- for intial acquisition duration a training signal is sent to receiver consisting of known data symbol sequence and the receiver then substitutes the known training sequence in place of slicer output, which is substituted and actual data transmission begins once the agreed time has elapsed

VII. Algorithms

1. Zero Forcing Algorithm

It is a kind of linear equalization algorithm used where inverse frequency response of the channel is applied, to restore the signal after the channel. The name zero forcing corresponds to bringing down the ISI to zero in a case having no noise and it is very useful when ISI is significant compared to noise [2].

For a channel with frequency response F(f) the zero forcing equalizer C(f) is constructed by

$$C(f)=1/F(f) \qquad (1)$$

Therefore a flat frequency response and linear phase is given by the combination of channel and equalizer

$$F(f)C(f)=1 \qquad (2)$$

But in practical zero-forcing equalization does not work because of the following reasons:

The equalizer's impulse response needs to be infinitely long, Even though the channel impulse response has finite length.

For some frequencies the received signal may be very weak. To compensate for it, the magnitude of the zero-forcing filter ("gain") grows very large. Due to which, any noise added after the channel gets boosted by a large factor and destroys the overall signal-to-noise ratio. In addition to this, the channel may have zeroes in its frequency response that cannot be inverted at all, as gain multiplied by zero will be equal to zero.

Having more limiting condition is the second problem. Such types of problems are addressed by making a small modification to the denominator of C(f)

$$C(f)=1/(F(f)+k) \qquad (3)$$

as such in MMSE where k is related to the channel response and the signal SNR.

When the channel transfer function or the channel response for a channel is H(s), then the input signal is multiplied by its reciprocal to remove the effect of channel from the received signal, especially ISI. The zero forcing is ideal when the channel is noiseless. When channel is noisy, it will amplify the noise greatly at frequencies where the channel response has small magnitude, inverting the channel completely.

2. Least mean square

LMS is a stochastic gradient optimization algorithm [3] based on Method of steepest descent which is a traditional optimization technique. LMS has following advantages

1. When the mean square error surface is plotted against the filter coefficients is with an unique minimum, quadratic and bowl shaped.
2. The negative of the gradient always points towards minimum, conversely the gradient of function always points uphill

Instead of using expected quantities, LMS algorithm simplifies the gradient calculation by use of instantaneous quantities. From the available data, it makes the use of the estimate of the gradient vector [4]. LMS involves an iterative procedure making successive corrections in the weight vector in the negative direction of the gradient vector eventually leading to the minimization of mean square error. In comparison to other algorithms, LMS is very simple as it does not require matrix inversions or correlation function calculation. Thus making it most commonly used algorithm. Because of being iterative it can be used in environment which is highly time varying. Against different signal conditions it gives stable and robust performance. However it does not have fast convergence speed as compared to other algorithms thus resulting a correlation matrix possessing a large eigen spread.

It uses multiple input signals for calculating the mean square error in estimating the reference signal. Steepest-descent based algorithm is then used for searching the Weiner solution by using following equation

$$w(k+1)=w(k)-\beta g\text{^'}(k) \qquad (4)$$

for k=0,1,2,3…, and g'(k) represents gradient vector estimation with respect to filter coefficients of the objective function. The resulting gradient estimate is given by

$$g\text{^'}(k)=-2e(k)x(k) \qquad (5)$$

So if the instantaneous square error replaces the objective function i.e the mean square error, then the above gradient represents the true gradient vector. Thus the gradient based algorithm so resulted is known as Least Mean Square LMS algorithm with following updating equation.

$$w(k+1)=w(k)+2\beta e(k)x(k) \qquad (6)$$

3. Recursive least square algorithm

In RLS algorithm, the inverse of the covariance matrix of the process in the update equation is incorporated that resembles the Weiner optimum solution in adaptive filter techniques [5]. It gives poor final performance measures. a general system setup where the received signals are generated by a process that could be the unknown wireless communication channel.

In general following operations are performed by RLS algorithm in order to update coefficients of an adaptive filter

a) Output signal y (k) is calculated for calculating error signal.
b) Error signal e(k) is calculated by using equation
$$e(k)=d(k)-y(k) \qquad (7)$$

c) Updating the filter coefficients by using equation
$$w'(k+1)=w'(k)+e(k).g'(k) \qquad (8)$$

where is the filter coefficients vector and is the gain vector and is inversely proportional to the λ, where λ is the forgetting factor[6]. Forgetting factor specifies the speed by which the filter forgets or updates the past sample information and has range typically between $0.95 \le \lambda \le 1$.

CONCLUSION

In this paper we presented a comprehensive study for Zero forcing algorithm, LMS and the recursive least-squared (RLS) algorithm. However practically zero forcing algorithm is not used because of her magnitude and gain related problem. For LMS algorithm its simplicity and easy implementation are there whereas RLS is complex in terms of implementation, but on other hand RLS algorithm provides faster convergence rate and minimum error rate when unknown system is employed. Also RLS algorithm has higher SNR as compare to LMS still LMS provides superior performance when it comes in terms of narrowband signals.

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