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An Overview of Adaptive Channel Equalization Techniques and Algorithms

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Abstract: ***Wireless communication system has been proved as the best for any communication. However, there are some undesirable threats of a wireless communication channel on the information transmitted through it, such as attenuation, distortions, delays and phase shifts of the signals arriving at the receiver end which are caused by its band limited and dispersive nature. One of the threats is ISI (Inter Symbol Interference), which has been found as a great obstacle in high speed communication. Thus, there is a need to provide perfect and accurate technique to remove this effect to have an error free communication. Thus, different equalization techniques have been proposed in literature. This paper presents the equalization techniques followed by the concept of adaptive filter equalizer, its algorithms (LMS and RLS) and applications of adaptive equalization techniques.***

Keywords: Channel Equalization, Adaptive Equalizer, Least Mean Square, Recursive Least Square.

1. Introduction

Nowadays, digital based systems are used for the transmission of voice, data and video. Digital communication systems are simple, more flexible, and highly reliable and are more tolerant to noise than analog systems [1]. But a phenomenon called Inter Symbol Interference (ISI) [2], could be a great threat in the process of digital communication. It arises when the data transmitted through the channel is band limited and hence dispersive, which causes the spreading of transmitted data/pulse and results in overlapping of adjacent pulse, due to which interference occurs in the transmitted signals [3]. It then becomes difficult to recover the original data from one channel sample, e.g. in fig.1 [4] there is a four level PAM signal, x(t) transmitted through the channel with impulse response h(t), the noise n(t) is added then at receiver, a distorted signal r(t), is received.

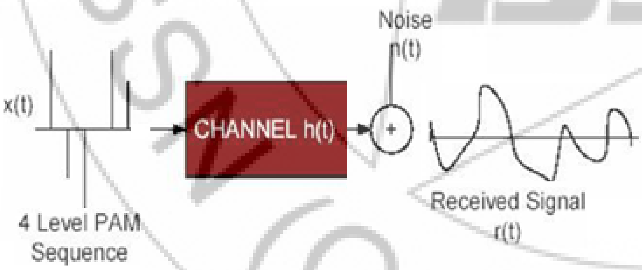

Figure 1: effect of noise in channel

This ISI has been recognized as the major obstacle to high speed data transmission over mobile radio channel. Thus it is very crucial to minimize ISI, so as to have an error free communication [5]. For this, a technique called Equalization is used to combat ISI. It is an inverse filter placed at the front end of the receiver. The transfer function of the equalizer is just inverse of the transfer function of the channel. It is an iterative process of reducing mean square error, the difference between the desired response and output of filter used in equalizer. It compensates for the average range of expected channel amplitude and delay characteristics [6].

The rest of the paper is organized as follows. In section II channel equalization followed by the description of adaptive equalization. Adaptive Equalizer (Filter) and Adaptive Algorithms are discussed in section III. Existing Adaptive Equalization Techniques are discussed in section IV. The application of adaptive equalization is presented briefly in section V. The conclusions along with future scope are given in section VI.

2. Channel Equalization

As discussed in introduction the equalization is used to eliminate ISI, comes broadly in two types [7]. The first is Maximum likelihood sequence estimation (MLSE), in which channel impulse response is measured and then receiver is made to adjust to the transmission environment (e.g. Viterbi equalization). The second one is Equalization with filters, which uses filters to compensate the distorted pulses. As the channel is generally unknown and time varying , Adaptive Equalizer are used, in which an equalizer filter is designed, whose filter coefficients are varying in nature according to the change of channel and try to eliminate ISI and additive noise at each time. In adaptive equalization the received signal is applied to receiver filter, the output of which is sampled at the symbol rate or twice of it, and then sampled signal is applied to adaptive filter equalizer. The coefficients are adapted to minimize the noise and ISI at the output. The adaption of equalizer is driven by an error signal.

3. Adaptive Filter Equalizer and Algorithms

Adaptive filter is defined by four aspects discussed below [8], [9]:

1. The signal being processed by the filter. Two modes are there:

   1.1 Decision directed mode: This means that receiver decisions are used to generate error signal. It is efficient in tracking slow variations in channel but is not efficient during initial acquisition.

   1.2 Training mode: To make equalization suitable in initial acquisition duration, a training sequence is needed (desired signal). In this mode, transmitter generates a data symbol sequence, known to receiver. The receiver therefore, substitute this known training sequence in place of slicer output. Once an agreed time elapses, the output is substituted and actual data transmission begins.

2. The structure that defines how the output signal of the filter is computed from its input
   2.1 The structure used could be IIR and FIR or transversal filter. The focus of paper is on FIR.
3. The parameters (filter response and weights) within this structure that can be iteratively changed to alter the filter's input-output relationship
4. The adaptive algorithm that describes how the parameters are adjusted from one time instant to the next.
   4.1 LMS and RLS are discussed in the paper.

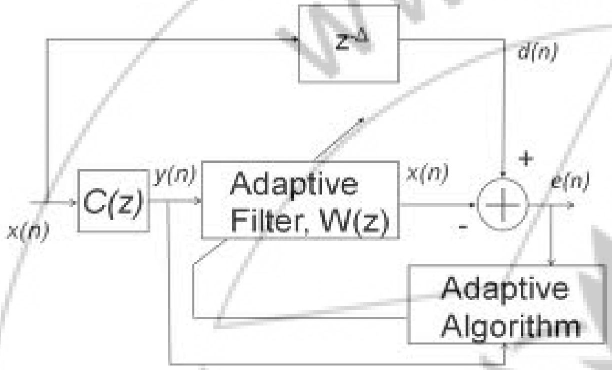

Figure 2: Adaptive Filter

The error signal is fed into a procedure which alters or adapt the parameter of the filter from time k to time (n+1), oblique arrow on the adaptive filter in fig.2 shows adaption. The e (n) is reduced to get the better desired response d(n)and this is achieved by adaptive algorithm, which adjust the parameters of the adaptive filter.

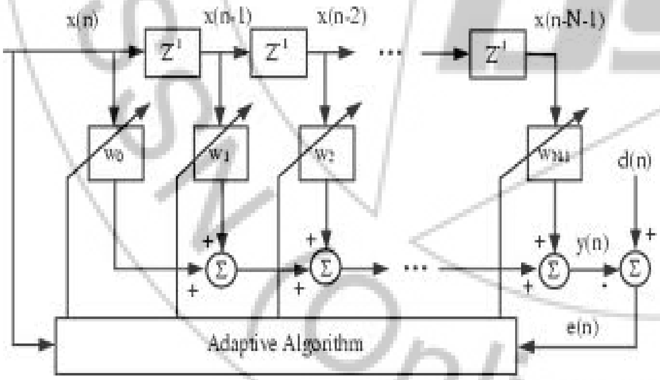

Figure 3: Structure of an FIR filter

The parameter w (t) correspond to impulse response values of filter at time n. The output signal is

$$y(t) = \sum_{i=0}^{N-1} w_i(n).x(n-i)$$

$$= W^T(n).X(n)$$

$$X(n) = [x(n), x(n-1) \ldots . x(n-N+1), \ldots input\ vector$$

$$W(n) = [w_0(n), w_1(n) \ldots . w_{i-1}(n)]^T, coefficient\ \text{Vector}$$

The general form of an adaption FIR filtering algorithm is

$$W(n+1) = W(n) + \mu(n)G(e(n), x(n), \varphi(n)$$

Where G (.) is a particular vector valued non-linear (depends on the cost function chosen), μ (n) is the step size parameter, e (n) and x (n) are the error and input signal respectively. Φ (n) is vector of states that stores information about input signal and error signal.

Now, the problem is x (n) and d (n) relation varies with time, so adaptive filter must continuously change its parameter values to adapt the change. For these following two algorithms are discussed.

A. Least Mean Square Algorithm (LMS)

Least Mean Square (LMS) algorithm comes under adaptive filter that finds least mean square of error signal (d (n) – y (n)). It is the approximation of steepest-decent method and is based on Minimum Mean Square Error (MMSE) criterion [10]. It includes H∞ theory which provides the mathematical basis for the deterministic robustness of the LMS filters. A weight control mechanism is responsible for performing adaption process using transversal filter as illustrated in fig.4

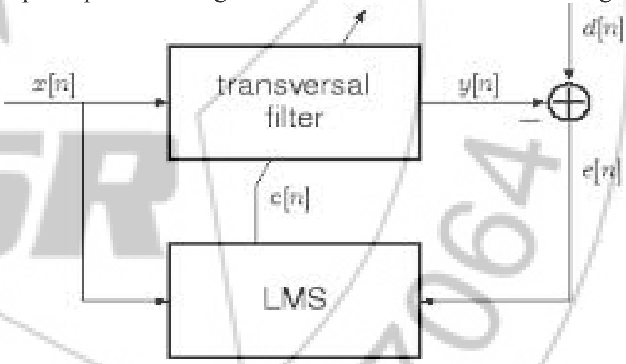

Figure 4: Equalization through LMS Algorithm

Steps,

$$filter\ output,\ \ y(n) = \sum_{i=0}^{N-1} x(n) - i\, w_i^*(n)$$

$$estimation\ error,\ \ e(n) = d(n) - y(n)$$

$$tap\ weight\ adaption\ \ w_i(n+1) = w_i(n) + \mu\, x(n) - i\, e^*(n)$$

$$update\ value\ of\ weight\ parameter$$
$$= old\ value\ of\ weight\ paramet - learning\ rate\ parameter(input\ vector)(error\ vector)$$

The algorithm convergence or stability depends on the value of step size or learning rate μ. It is convergent if and only if it lies between zero and inverse of input signal power. LMS is simple to implement, gives stable and robust performance under different signal conditions and do not neglect the noise like zero-forcing equalization. The disadvantages are its slow convergence and its demand for using training sequence as reference, then decreasing the communication bandwidth.

B. Recursive Least Square (RLS)

Recursive Least Square is another algorithm for determining the coefficients of an adaptive filter [11]. In contrast to the LMS algorithm, the RLS uses information from all past input samples to estimate the autocorrelation matrix (inverse) of input vector. To decrease the influence of input samples from the far past, a weighting factor for the influence of each sample is used.

The RLS algorithm for a p[th] order RLS filter is given as

Parameters, p=filter order, λ=forgetting factor,

$\delta = value\ to\ initialize\ p(0)$

$initialization,\ w(n = 0), x(n) = 0, (n = -p \ldots . -1)$

Computation, for n=1, 2, 3…

$$x(n) = \begin{bmatrix} x(n) \\ x(n-1) \\ \ldots x(n-p) \end{bmatrix}$$

$$error,\quad e(n) = d(n) - x^T(n)w(n-1)$$

$gain\ vector,$

$$g(n) = P(n-1)x^*(n)\{\lambda + x^T(n)\,P(n-1)x^*(n)\}^T$$

$$P(n) = \lambda^{-1}P(n-1) - g(n)x^T(n)\lambda^{-1}P(n-1)$$

$$w(n) = w(n-1) + e(n)g(n)$$

4. Adaptive Equalization Techniques

Different kinds of equalization techniques are there [12], but most widely used are discussed below:

A. Zero-Forcing Equalizer

Zero-Forcing Equalizer refers to a form of linear equalizer algorithm which applies the inverse of the channel frequency response to the received signal, to restore the signal after channel [13]. The name zero-forcing corresponds to bring down the ISI to zero in noise free case. This will be useful when ISI is significant compared to noise. For a channel with frequency response C_h (z), the zero-forcing equalizer, C(z) is constructed by

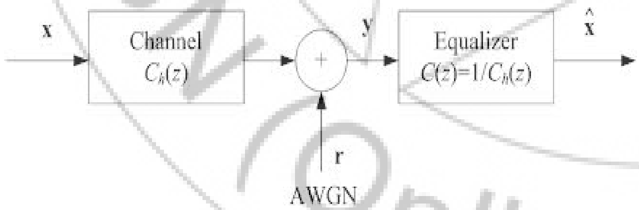

Figure 5: Zero Forcing Equalizer

$$C(z) = {}^{1}/_{C_h(z)}$$

Thus the combination of channel and equalizer gives a flat frequency response and time phase

$$C_h(z)C(z) = 1$$

Zero-Forcing equalizer removes all ISI and is ideal when the channel is noiseless. But sometimes zero-forcing equalizer does not work in most applications, e.g. at some frequencies the received signal may be weak. To compensate, the magnitude of the zero forcing filter ('gain') grows very large, as a result any noise added after the channel gets boosted by a larger factor and destroys the overall signal to noise ratio (SNR). Also, the channel may have zeroes in its frequency response that cannot be inverted at all (gain* 0 still equals zero).

B. Decision feedback Equalizer (DFE)

A decision feedback equalizer is a non-linear equalizer that uses previous decisions to remove ISI caused by previously detected symbols on the current symbols to be detected [14]. The DFE consist of two filters, a feed forward and a fractionally spaced FIR filter with adjustable coefficients (feedback filter). The output of feedback filter is subtracted from the output of feed forward filter to form input to the detector.

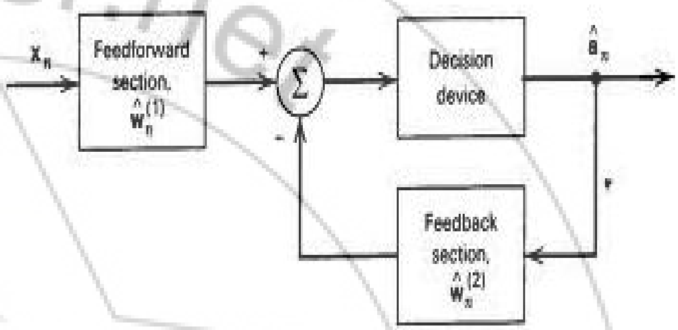

Figure 6: Decision feedback Equalizer

$$z_m = \sum_{n=-N_1}^{0} c_n\, y(mT - n\tau) - \sum_{n=1}^{N_2} b_n\, I'_{m-n}$$

$[c_n]\&[b_n] = adjustable\ coefficients\ of\ feedforward\ and\ feedback\ filter\ response$

$I_{m-n}, n = 1, \ldots N_2 = previously\ detected\ symbols$

$N_1 + 1\ \&\ N_2 = length\ of\ feedforward\ \&\ feedback\ filter$

Based on the input z_m, the detector determines which of the possible transmitted symbol is closest in distance to the input signal I_m. Thus it makes its decisions and output $\tilde{I}m$. The advantage of a DFE implementation is the feedback filter, which is additionally working to remove ISI, operates on noiseless quantized levels, and thus its output is free of channel noise.

C. Minimum Mean Square Error Equalizer (MMSE)

The MMSE equalizer suppresses both the interference and noise components, whereas the ZF equalizer removes only the interference components. It is based on the mean square error (MSE) criterion [15]. This implies that the mean square error between the transmitted symbols and the estimate of the equalizer is minimised. Let x be an unknown random variable and let y be a known random variable. An equalizer x^(y) is any function of the measurement y, and its mean square error is given by

$$MSE = E\{(\hat{x} - x)^2\}$$

The coefficient values of an adaptive filter equalizer obtained at this minimum are the ones that minimize the power in the error signal e (n), indicating that y(n) has approached d(n). It is a simple equalizer for adjusting the parameters of an FIR filter.

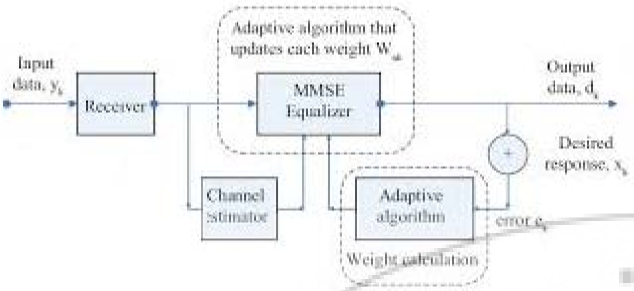

Figure 6: MMSE Equalizer

D. Blind Equalization

In many high data rate band limited digital communication systems; the transmission of a training sequence is either impractical or very costly in terms of data throughput. Conventional LMS adaptive filters depending on the use of training sequences cannot be used. For this reason, blind adaptive equalizers that do not rely on training sequence have been developed [16].

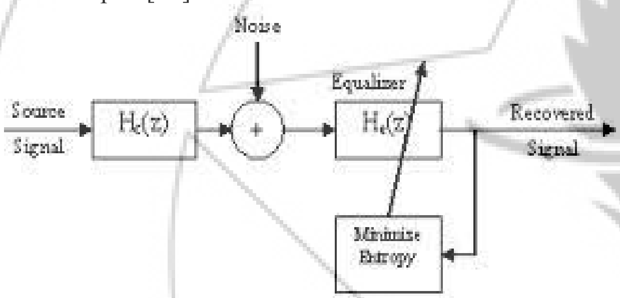

Figure 7: Blind Equalizer setup

In blind equalization, the desired signal or input to the channels is unknown to the receiver, except for its probabilistic or statistical properties. As both the channel and its input are unknown, the objective of blind equalization is to recover the unknown input sequence based solely on its probabilistic properties. Blind equalization algorithms [17] that have been proposed are the constant modulus algorithm (CMA) [18] and multimodal algorithm (MMA) [19]. This reduces the mean square error (MSE) to desired levels. In this equalization, the output of the equalizer is quantized and the quantized output is used to update the coefficients of the equalizer. Due to the absence of a training signal, it is important to exploit various available information about the input symbol and the channel output [20] such as the power spectral density (PSD) of the channel output signal, which contains information on the magnitude of the channel transfer function and the higher order statistics (HOS) of the sampled channel output, which contains information on the phase of the channel transfer function.

5. Application of Adaptive Filtering

Adaptive filters have become invaluable system component in modern industry. Without adaptive filters, many of the systems we currently rely on in our daily life would not exist. 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. There are four most common applications of adaptive filters currently used in commercial system [21], [22], [23], [24].
1. System Identification
2. Inverse System Modelling
3. Signal Prediction
4. Interference Cancelation

Class of adaptive filtering	Description	Application
System identification	Model the channel to design distortion compensation	Echo Cancellation, VOIP
Inverse System Modelling	Filter attempts to estimate unknown system's inverse, reduce ISI	Channel Equalization
Signal Prediction	Reduce speech transmission bandwidth	DPCM speech quantize
Interference Cancelation	One or more sensor signals used to remove interference and noise	Array processing for radar and communication

6. Conclusion

In this section, we have presented an overview of channel equalization, description of adaptive filter equalizer, its algorithms and techniques with emphasizing the applications that have already proven they are useful in practice. Despite the many contribution in the field, research efforts in adaptive filter equalization continue at a strong pace and it is likely that new application, techniques and algorithms will be developed in the future.

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