Adaptive Block Linear Equalizer for Block Data Transmission Systems

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Abstract— **This paper presents an adaptive algorithm that is considered as a modification to the Block Linear Equalizer which is used with Block Data Transmission Systems. The two main factors that influence the noise performance of the Block Data Transmission System are the distortion introduced by the channel and the block size. The proposed algorithm considerably improves the noise performance and hence the transmission efficiency of the system by adaptively reducing the block size of the transmitted signal-elements when the distortion introduced by the channel is high and increasing the same when the channel distortion is low. This advantage is achieved with only a slight increase in the equipment complexity.**

Index Terms— ***Wireless, Block transmission, Channel distortion, Equalizer***

I. Introduction

In the study of detection processes for distorted digital signals, techniques of both linear and nonlinear equalization of the channel have been widely studied, The non-linear equalization of the channel usually gives a better tolerance to additive white Gaussian noise than linear equalization , normally requiring a lower average signal to noise power ratio for a given error rate. An even better tolerance to noise can be achieved through the use of more sophisticated detection processes which do not equalize the channel. Many of these processes, however, involve considerable equipment complexity [1-4].

An interesting technique called the Block Data Transmission system (BDTS) has recently been proposed which for certain applications can achieve a similar standard of performance as the more sophisticated processes just mentioned, but with relatively simple equipment [5-8]. The system is a synchronous serial data transmission system and employs transmission of alternating blocks of data and training/zero valued symbols. In contrast to the recursive symbol-by-symbol detection approach usually employed, each data block is here detected as a unit. The Block Linear Equalizer (BLE) and Block Decision Feed Back Equalizer (BDFE) have been designed for the system and the system has applications in wireless local loop, wireless LAN's and indoor communications in general [6-8]. Techniques for estimating the channel characteristics have also been proposed for implementing BLE in an adaptive manner [9-12].

Block Data Transmission System mentioned above has some useful advantages over systems with continuous transmission**.** Firstly, exact equalization of the channel is, in every case, achieved**.** Secondly, complete loss of signal cannot result from an unfortunate combination of signal-element values and the channel impulse response. Thirdly, there are no error extension effects from one block of elements to the next, regardless of the detection process used. Finally, detection process achieving a high tolerance to additive Gaussian noise can be implemented quite simply [5-8, 11].

Assuming that there are m signal-elements in a group and g training/zero elements then in the BDTS adjacent blocks of m signal-elements at the transmitter, are separated by g training/zero elements which carry no information. (m+g)T seconds are required to transmit the information carried by the m signal-elements of a group, where 1/T is the element transmission rate. If groups of signal.-elements are replaced by a continuous stream of elements without the g training/zero, then for the same information rate, m signal-elements must now be transmitted, with no gaps, over (m+g)T seconds. This means that the element transmission rate in the equivalent continuous transmission, is reduced from 1/T to 1 /T' where T' = (m+g/m)T. Thus for a given information rate, the bandwidth required in the case of BDTS is wider than that required for the continuous system. This reduces the tolerance to noise of the BDTS and partly offsets the basic advantages gained by the arrangement.

In order to improve the tolerance to noise of the BDTS, the value of m should be kept large compared to g the number of training/zero elements. However, it is shown that when the channel introduces severe amplitude distortion the tolerance to noise of the BDTS decreases as the group size m is increased. Thus in order to utilize the advantages offered by BDTS the group size of the block should be changed in an adaptive manner. When the distortion in the channel increases the group size should be reduced and when the distortion in the channel decreases the group size may be increased.

This paper presents an adaptive algorithm where the group size m of the transmitted signal-elements is changed adaptively depending on the estimates of the channel impulse response. The resulting block linear equalizer is called the Adaptive Block Linear Equalizer (ABLE). Results of computer simulation show that ABLE achieves considerable advantage in tolerance to noise over the BLE together with

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higher transmission efficiency. These advantages are achieved with only a slight increase in equipment complexity.

This paper is organized as follows. In Section II, a brief description of BLE is given. The design and analysis of the adaptive algorithm is presented in Section III. In Section IV results of computer simulation are presented and the performance of BDTS with and without the proposed algorithm is compared. Vectors and matrices are represented by block capital letters.

II. BLOCK LINEAR EQUALIZER

Fig. 1 shows the model of the data transmission system considered. Each impulse $s_i\,\delta(t\text{-}iT)$ at the input to the channel is the corresponding input signal-element and it may be either binary or multilevel. The signal elements are assumed to be antipodal and statistically independent.

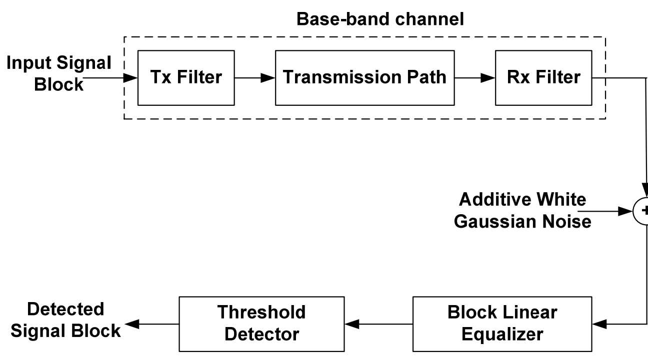

Fig.1 Model of the block transmission system

The linear baseband channel has an impulse response y(t) and includes all transmitter and receiver filters used for pulse shaping and linear modulation and demodulation. The impulse response h(t) of the transmitter and receiver filters in cascade is assumed to be such that h(0) = 1 and h(iT) = 0 for all nonzero integer values of i. This impulse response is achieved by using the same transfer function $B(f) = H(f)^{½}$ for the transmitter and receiver filters, where

$$H(f) = \begin{cases} \frac{1}{2}T(1+\cos(2\pi ft)) & -\frac{1}{T} < f < \frac{1}{T} \\ 0 & \text{else where} \end{cases} \tag{1}$$

White Gaussian noise is introduced at the output of the transmission path. The noise has zero mean and a two sided power spectral density of σ^2, giving the zero mean Gaussian waveform w(t) at the output of the receiver filter. Thus the resultant signal at the output of the receiver filter is

$$r(t) = \sum_i s_i y(t\text{-}iT) + w(t) \tag{2}$$

The received signal is sampled at time instant t=iT , where T is the symbol interval. Here, consecutive blocks of *m* information symbols at the input to the transmitter filter are separated by blocks of *g* zero level symbols as shown in Fig. 2, where *g* is the largest memory length of the channel y(t), and $y = [y_0, y_1, y_2 \ldots. y_g]$ is its sampled impulse response.

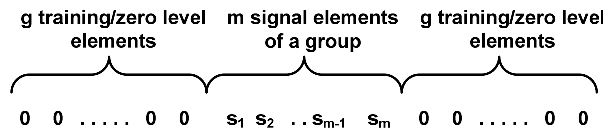

Fig. 2 Structure of transmitted elements in a block transmission system.

For each received group of m signal-elements, there are n=m+g sample values at the receiver input that are dependent only on the m elements, and independent of all other elements. The detector uses these *n* values in the detection of the symbol block. With correct detection , the detected values are used for the estimation of the channel sampled impulse response using the same equipment [11].

If only the i^{th} signal-element in a group is transmitted, in the absence of noise and with s_i set to unity, the corresponding received n sample values used for the detection of m elements of a group are given by

$$Y_i = \overbrace{0 \;\ldots\ldots\; 0}^{i-1}\;\overbrace{y_o \;\; y_1 \;\; \ldots \;\; y_g}^{g+1}\;\overbrace{0 \;\ldots\ldots\; 0}^{m-i} \tag{3}$$

where y_h must be non-zero for at least one h in the range 0 to *g*. The sum of the *m* received signal elements in a block in the absence of noise is, therefore, given by the n components of the vector R, where

$$R = \sum_{i=1}^{m} s_i Y_i = SY \tag{4}$$

S is the m-component row vector whose i^{th} component is s_i and represents the transmitted signal block of size m. Y is an m×n matrix whose i^{th} row Y_i is given by (3). Since at least one of the y_h is non–zero, the rank of the matrix **Y** is always m, so that the m rows of Y are linearly independent. Note that the impulse response of the channel completely determines the matrix Y.

When AWGN is present, the n sample values corresponding to a received signal block at the detector input are given by the n-component row vector R **,**where

$$R = SY + W \tag{5}$$

W is the *n*-component noise vector whose components are sample values of statistically independent Gaussian random variables with zero mean and variance σ^2.

Assume now that the detector has prior knowledge of Y_i , but has no prior knowledge of the s_i or σ^2. Knowledge of the Y_i of course implies knowledge of the channel impulse response. Since the detector knows Y, it knows the *m* dimensional subspace spanned by Y_i and hence the subspace containing the vector SY, for all s_i . Since the detector has no prior knowledge of s_i , it must assume that any value of S is as likely to be received as any other, and in particular, as far as the detector is concerned, s_i need not be ±1 . For a given vector R the most likely value of SY is now at the minimum distance from R. Clearly, if R lies in the subspace spanned by the Y_i , then the most likely value of SY is R. In general, R will not lie in this sub-space, and in this case, the best estimate the detector can make of S is the *m*-component vector X, whose components may have any real values and are such that XY is at the minimum distance from R. By the projection theorem [13], XY is the orthogonal projection of R onto the m-dimensional subspace spanned by the Y_i . It follows that R-XY is orthogonal to each of the Y_i, so that

$$(R-XY)Y^T = 0 \tag{6}$$

Or

$$X = RY^T(YY^T)^{-1} = RF \tag{7}$$

$F = RY^T(YY^T)^{-1}$ is a real n×m matrix of rank m. Since the *mxn* matrix Y has rank m, the *mxm* autocorrelation matrix (YY^T) is symmetric positive definite and its inverse will always exist. Thus if the received signal vector R is fed to the n input terminals of the linear network represented by the matrix F, the signals at the m output terminals are the components x_i of vector X, where

$$\begin{aligned} X &= RY^T(YY^T)^{-1} = [SY + W]\, Y^T(YY^T)^{-1} \\ &= S + WY^T(YY^T)^{-1} = S + WF = S + U \end{aligned} \tag{8}$$

The n-input m-output linear network F is called the Block Linear Equalizer (BLE). BLE can also be implemented in an adaptive manner for channels that are varying with time [9-12]. The *m*-component row-vector U is the noise vector at the output of the network F. From (8) each component u_i of the vector U is a sample value of a Gaussian random variable with zero mean and variance

$$\eta_i^2 = \sigma^2 \sum_{i=1}^{n} f_{ji}^{\,2} \tag{9}$$

f_{ij} is the element of matrix **F** in i^{th} row and j^{th} column. The average value of the noise variance in the detection of the received block of m signal-elements is therefore, given by

$$\eta^2 = \sigma^2 \left(\frac{1}{m} \sum_{i=1}^{m} \sum_{j=1}^{n} f_{ji}^{\,2} \right) \tag{10}$$

s_i is now detected by testing the corresponding x_i against the appropriate threshold level.

III. ADAPTIVE ALGORITHM

From (10) the average noise variance η^2 in the detection of the received signal-elements of a group and hence the bit error rate (BER) of the BDTS is directly proportional to the factor Γ, where Γ is given by

$$\Gamma = \sqrt{\left(\frac{1}{m} \sum_{i=1}^{m} \sum_{j=1}^{n} f_{ji}^{\,2} \right)} \tag{11}$$

and the probability of error in terms of the factor Γ is given by

$$P_e = \frac{1}{2} \operatorname{erfc}\left(\frac{1}{\Gamma} \sqrt{\frac{E_b}{N_0}} \right) \tag{12}$$

E_b is the energy per bit of the transmitted signal.

The factor Γ itself depends on the following three parameters

- the elements fij of the n x m network F
- the number of training/ zero elements ‘g’
- the length of the signal block ‘m’

Moreover parameters f_{ij} and g in turn depend on the impulse response of the channel. For severe signal distortion and large channel memory g, the correlation between the rows of the matrix Y is high and hence the elements of the m x m correlation matrix YY^T and its inverse $(YY^T)^{-1}$ and consequently the elements fij will have large values [12]. This will result in large values of Γ and hence the noise performance of BDTS will deteriorate. On the other hand, for phase distortion and small values of g, the correlation between the rows of the matrix is small and the elements of $(YY^T)^{-1}$ and consequently the elements f_{ij} will have small values. This will improve. For a given input noise variance and channel characteristics, Fig. 3 shows the relationship between Γ and the BER. It is clear that the noise performance of the system deteriorates as Γ is increased.

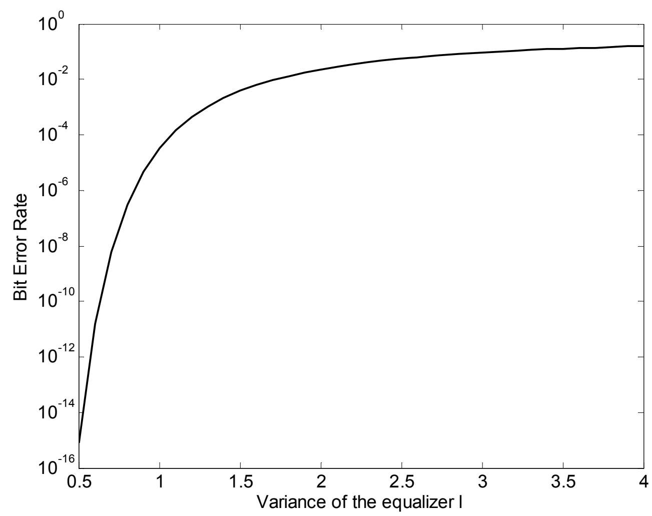

Fig. 3. Effect of equalizer variance on the performance

Another important factor that affects the system noise performance is the block length *m*. It can be seen from Eqns (9) & (11) that for a given channel distortion, as m increases the value of Γ will also increase. Moreover, the increase in the value of Γ will be higher in cases where the channel introduces higher inter-symbol interference (greater amplitude distortion).Fig. 4 shows the degradation in the noise performance of the system as the block length is increased.

Thus in order to keep the BLE operating at an optimum block size, it is desirable to adaptively change the size of the signal block as and when the cannel distortion changes. Such an adaptive implementation will take the form of an adaptive

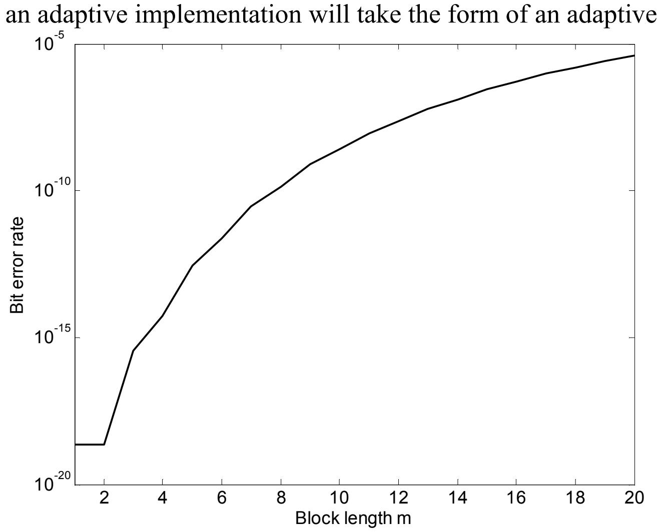

Fig. 4 Effect of block length on BER

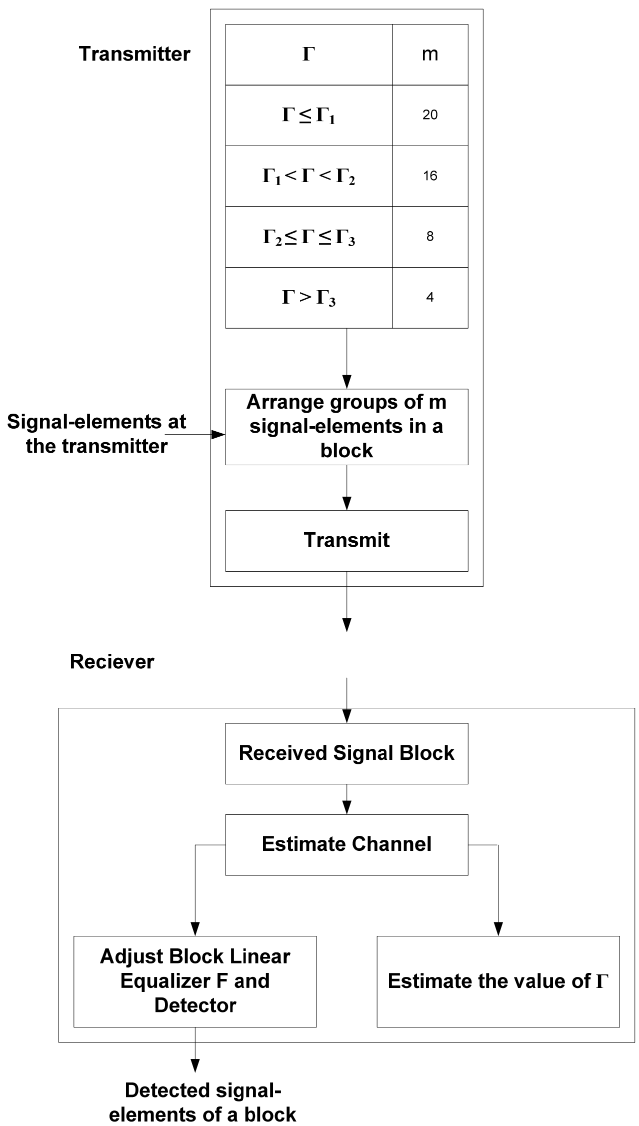

Fig. 5 Adaptive algorithm for block linear equalizer

algorithm that will use short signal blocks when the channel distortion is large (large values of Γ) and long signal blocks for low channel distortion (lower values of Γ). It is, therefore, required to calculate the value of Γ from the knowledge of the estimated channel impulse response and send this information back at the transmitter through the reverse channel. The signal block size will then be selected at the transmitter on the basis of a look up table. It may be pointed out that this process is to be repeated only when the channel is estimated and not before the transmission of each block of signal-elements. This algorithm is shown in Fig. 5. Only four distinct values of block size are considered for four distinct ranges of Γ. The resulting BLE is called the Adaptive Block Linear Equalizer (ABLE).

IV. RESULTS

Fig. 6 shows the noise performance of the Adaptive Block Linear Equalizer (ABLE). For the sake of comparison the performance of Block Linear Equalizer (BLE) is also shown. The results have been obtained through computer simulation. In this simulation, a total of 10000 block of data were transmitted with four distinct ranges of Γ. It was assumed that the channel is estimated correctly using any of the techniques referred to in the paper. For the BLE, the block size m was taken to be 8 so that a total number of 80,0000 signal bits were transmitted in the simulation.

TABLE I.
Normalized Channel Impulse Response

Channel Sampled Impulse Response
$(1.000)^{-1/2}$ [1.000 0.000 0.000]
$(1.500)^{-1/2}$ [0.500 1.000 -0.500]
$(2.000)^{-1/2}$ [1.000 0.000 1.000]
$(6.000)^{-1/2}$ [0.707 2.234 0.707]
$(2.000)^{-1/2}$ [1.000 1.000 0.000]
$(6.000)^{-1/2}$ [1.000 2.000 1.000]
$(1.500)^{-1/2}$ [0.500 1.000 0.500]
$(19.00)^{-1/2}$ [1.000 2.000 3.000 2.000 1.000]
$(2.000)^{-1/2}$ [0.235 0.667 1.000 0.667 0.235]
$(1.945)^{-1/2}$ [-0.167 0.667 1 0.667 -0.167]

The impulse responses of the channels considered in this simulation test are shown in Table 1. In each case the impulse response is normalized to have unit energy. These channel characteristics are selected as these are close to the realistic channels [3,7,8] and cover wide ranges of channel lengths and distortions. The algorithm takes the block length of size 20 if the value of Γ is below 1.2, and the block length *m* = 16 if Γ lies between 1.2 and 2. If Γ lies between 2 and 3, the block length used is 8, and the last choice of block length is 4, which is taken for higher values of Γ.

From Fig 6 it can be seen that ABLE gains consider advantage in tolerance to noise over the BLE. Moreover, by using the adaptive algorithm proposed in this paper , the systems managed to send 125000 bits in the same 10000 block with an average of 12.5 signal bits/block. This indicates that ABLE uses the channel bandwidth more efficiently as compared to BLE. The performance of ABLE can be improved further by incorporating more ranges for Γ. However, this will increase the complexity of the system. How many ranges should be used for Γ will depend on the environment and application.

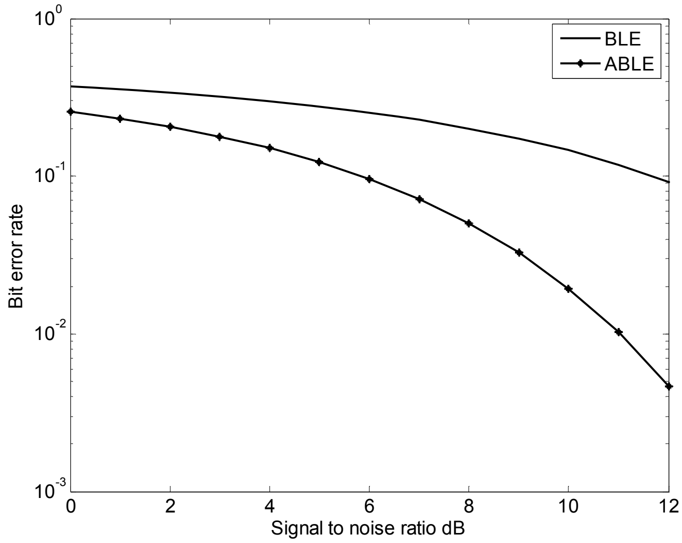

Fig. 6 Comparison between the BER for BLE and ABLE.

V. CONCLUSIONS

In Block Data Transmission System the noise performance of the Block Linear Equalizer depends on the channel distortion and the size of the transmitted signal block. The algorithm presented in this paper adaptively controls the block size by estimating the channel distortion from the estimated impulse response of the channel. For higher values of channel distortion the block size is reduced and for smaller values of channel distortion the block size is increased. This is done at the transmitter through a look up table. The resulting block linear equalizer is called the Adaptive Block Linear Equalizer (ABLE). It achieves considerable advantage in tolerance to noise over the BLE and has a higher information transmission rate for the same element transmission rate. This advantage is achieved with only a slight increase in equipment complexity.

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