Optimisation of Adaptive Partial Response Equalisation for Data Storage Channels

M. Abuitbel [1], N. Abouzakhar [2], D. King [3] and G. Manson [4]

[1,3] University of Manchester, [2,4] University of Sheffield

Abstract: The process of reshaping the channel frequency response, correcting the channel-induced distortions and lowering the intersymbol interference is termed as channel equalisation. In practice, the equaliser in the system receiving electronics with continuously adjusted parameters used to handle the channel characteristic variations in an attempt with the objective of optimising the channel frequency response. This is to enhance the storage capacity, maximising the signal to noise ratio, decreasing the channel error rate and eventually improving the system performance. A mechanism of obtaining partial response equalisation through the optimisation of the FIR filter tap weights is discussed. The optimised scheme is based on least mean square (*LMS*) algorithms. Making the filter tap weights to be varying with time in a tightly controlled manner according to a specified criterion is refereed to an adaptive equalisation. Adaptive optimisation is mainly relies on an objective function of an iterative algorithm for adjusting its parameters.

1 Introduction.

Adaptive equalisation systems have become an integral part of many modern practical designs and successfully used techniques for many years. By virtue of their structure, these devices adjust themselves to an ever-changing environment, such that their performance improves through a continuing interaction with their surroundings. Their superior performance in non-stationary environment results from their ability to track slows variations in the statistics of the signals and to continually seek optimal design [10].

Fig. 1 illustrates the basic principles of adaptive equalisations. The processed signals could be a received signal from a communication system or a replayed signal from a magnetic storage media. The reference sequence *y(n)* with the combination of the input sequence *x(n)* are used to update the filter tap weights by the minimisation of the error difference. This paper introduces a mechanism of finding the minimum mean square error (*MSE*) at the output of the equaliser that shapes the replayed signal to a defined response through the optimisation of the equaliser tap weights [4].

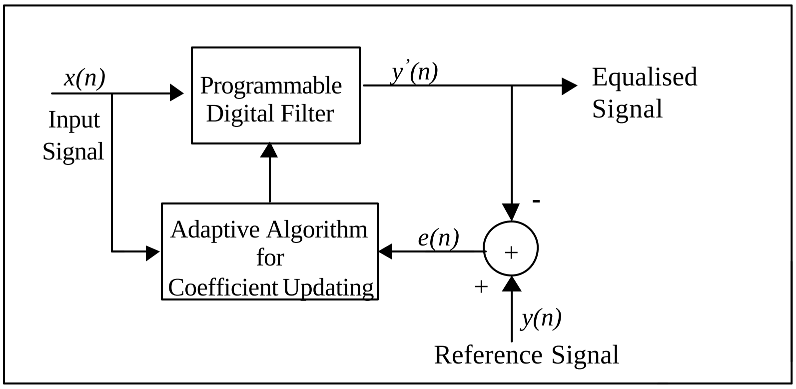

Figure 1: Principles of an adaptive equalisation

Generally, the versatility of adaptive systems have evolved so rapidly over the years as a result of significant advances in integrated circuits (ICs) design, digital computer technology and in adaptive signal processing disciplines. Adaptive structures that perform multiple tasks have become prominent parts of many electronic devices, ranging from a multitude of consumer products to advanced military electronics. Notable applications include the suppression of interference arising from noisy measurement sensors, the elimination of distortion introduced when signals travel through transmission or storage channels (channels equalisation), and the recovery of signals embedded in a multitude of, or many, echoes created by multi-path effects in mobile communications [10].

2. The Appeal of Adaptive Equalisation in Data Storage Channels.

The performance of digital magnetic recording systems suffers degradations through the decline of signal levels (pulse amplitudes) and increasing inter-symbol interference (ISI) as the data packing densities increase [1,2,3]. In practical recording systems as shown in Fig.2 the frequency response of the channel is not always known with sufficient precision to allow the playback electronics to compensate for possible channel impairments and increased inter-symbol interference. Advanced replay equalisations have been adopted in the magnetic recording technology to shape the channel pulse response to some specified target shape which has a shorter duration (higher bandwidth) and this is called partial-response signalling or equalisation. The idea of implementing partial response equalisation in recording systems is adopted from communications theory. The partial response equaliser is normally followed by a Viterbi detector that is matched to the target shape. Maintaining precisely the desired partial response shape through adaptive equalisations at the channel output permits the Veterbi-detector to be efficiently realised and hence improving the bit detection quality. The overall task of the detector is to recover the encoded data that was originally recorded on the magnetic medium.

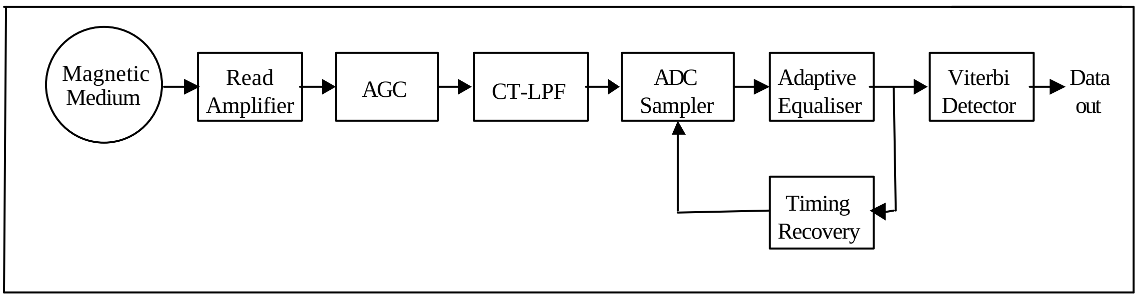

Figure 2: A block diagram of a partial-response read (data receiver to convert the read amplifier analogue output signal into the original user data) channel

The most common partial-response systems are defined by the polynomial $P_n(D)=(1-D)(1+D)^n$ with n being a positive integer and D is the delay operator (could be one-bit channel period). Partial response provides an adequate means of recovering playback waveforms by controlling the inter-symbol interference rather than trying to eliminate it completely at the detector [6]. Hence no intensive equalisation is required which boosts the system noise to a significant level. The reduced noise at the detector and the known amount of the inter-symbol interference provide a means to improving the high recording capacity [4,5].

Practically, partial response equalisation is established using analogue signal processing or in digitally oriented scheme in a form of a linear tapped delay line or transversal filter as shown in Fig. 3. Common *direct-form* FIR or transversal filter is the simplest equalisation structure to study and implement because of their stability, robust and precise design. The applicability of analogue filters in magnetic recording systems is gradually diminishing because analogue filters are inherently difficult to tune or adapt to wide variability in magnetic media, recording heads and other storage materials [7]. Presently digital FIR filters play the major role in achieving the signal equalisations [5,6]. Filter tap weights adaptation algorithms are continually becoming cheaper with advances in digital circuit technology as they are easily programmed. Presenting the replayed signal in a digital form allows the subsequent signal processing to be carried out digitally. As a result optimal system performance and accurate characterisation of the playback signal shape can be achieved with less cost and time consumption.

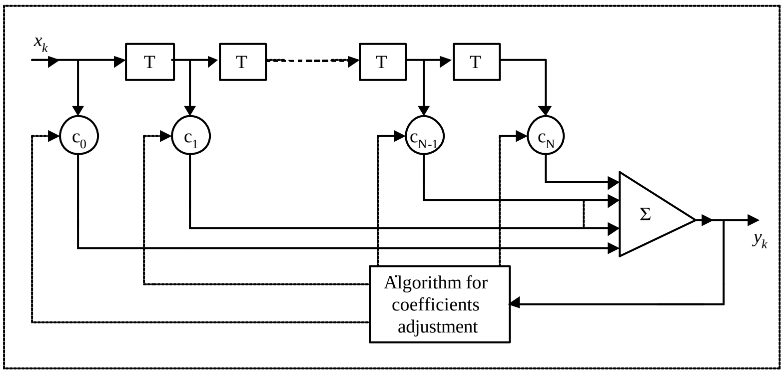

Figure 3: FIR as an adaptive equaliser

The FIR filter tap weights c_N are adaptively adjusted directly from the replayed data to minimise the error between the filter output (in response to sampled input data) and the specified partial response target. Therefore the criterion for selecting the filter coefficients c_N is based on the minimisation of mean squared error (*MSE*) to force the filter output to have a specified shape through the utilisation of an effective optimisation strategy [9].

3. LMS Algorithm Optimisation.

In addition to analytical approaches the use of powerful computers and software with efficient algorithms makes it possible to model and optimise the behavioural mechanisms of adaptive filtering. The least mean square (*LMS*) is one of the most widely used stochastic gradient-based algorithms for adaptively minimising the mean square error (*MSE*). It is less complicated compared to other optimisation techniques where no matrix inversion or other complicated calculations are involved. The combination of channel coding and the *MSE*-based equalisation offer a practical way to achieving high storage data capacity. Standard *LMS* algorithms aim at obtaining a *MMSE* between the received samples and the desired response by adjusting the equaliser tap weights through several iterations cycles [8,9]. The algorithm repeatedly updates the tap weights of the adaptive filter according to the following update equation:

$$C_{k+1} = C_k + m e_k sign(X_k) \qquad (1)$$

where: C_k is a vector of tap weights to be optimised, m is the algorithm step size, e_k is a vector of measured error, *sgn* is a signum function and X_k is a sampled input data.

The *LMS* iteration algorithm can be summarised as follows:

1. Starts the iteration process with known initial conditions where each tap weight in the vector $\boldsymbol{C_k}$ is set to some arbitrary value such as zero.
2. Compute the filter output signal response.
3. Compute the ideal target reference.
4. Calculate the error difference between step 2 and 3.
5. Use the update equation to determine the next filter tap weights.
6. Repeat the preceding steps for few iterations.
7. Stop the iteration process once the *LMS* converges with *MMSE* ($E\{e_k^2\}$).

4. Modelling Results.

Fig.3 shows the performance of the equaliser for shaping the channel-isolated pulse to desired partial-response pulse. The obtained *MSE* value is found to be $8x10^{-13}$ and the set of the equaliser tap weights are also determined. The obtained results demonstrate that, the presented technique can be applied to successfully to a series of data pattern in which the more iterations are performed the better reshaping is resulted.

Fig.5 shows a comparison curve called learning curve obtained by plotting *MSE* versus the number of performed iterations with step-size μ is being 0.01, 0.05 and 0.1. Learning curves are suitable for choosing the precise value of μ. Reducing the value of µ will result in algorithm converge at a slower rate but with more accurate set of tap weights [8,9].

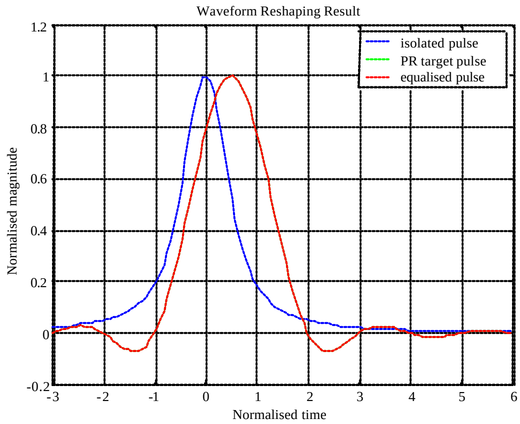

Figure 4: The result of equalising the channel basic isolated pulse to a partial response target

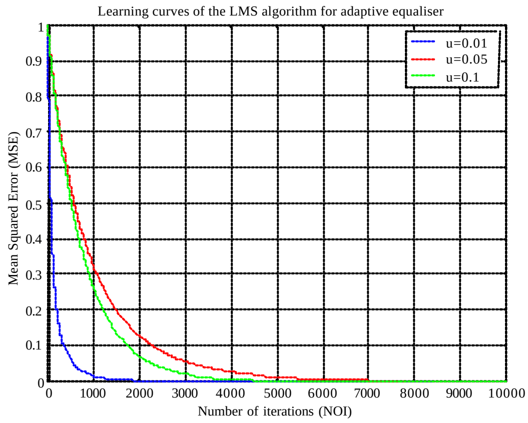

Figure 5: Convergence of the adaptive equalization algorithm

5. Conclusions.

Introducing partial-response equalisations in data storage channels allow better control of the channel impairments. The introduced equalisations enforce spectral properties at the channel playback electronics hence more efficient utilisation of the given channel bandwidth is permitted. A successfully optimisation scheme based on *LMS* algorithms for calculating the tap weights of an adaptive partial response equaliser is developed in *MatLab-based* programming. For efficient data equalisation, the introduced algorithm has been made adaptive. Adaptive equalisation precisely maintains the desired partial response shape at the storage channel output. The adaptation algorithm is based on calculating the error difference signal between the desired equalised signal and an ideal target reference. The adaptation training is terminated when no further improvement in the value of the error signal at the equaliser output was observed. Adaptive equalisations offer the possibility of a robust data recovery process that changes to suit the time varying characteristics of a recording channel

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