AN EFFICIENT SELF-RECOVERING ADAPTIVE ALGORITHM FOR BPSK SIGNALS TRANSMITTED THROUGH UNDERWATER ACOUSTIC CHANNELS

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ABSTRACT

The aim of this paper is to propose a receiver which performs fast self-recovering adaptive identification and equalization based on decision feedback equalization and reduced maximum likelihood sequence estimation. It runs on binary-phase-shift keying signals through underwater acoustic channels. Considerations on implementation are discussed. Its performance is demonstrated both on experimental data from long-range deep water channel and on simulated data from two discrete-time channels with severe amplitude distorsion and additive white gaussian noise.

1. INTRODUCTION

This work is focused on achieving reliable communication over Underwater Acoustic Channels (UWA). The UWA channel is characterized as a time dispersive channel with relatively rapid time variation and fading. In bandwidth-efficient digital communication system, multipath propagation causes InterSymbol Interference (ISI) which results in severe amplitude distorsion and phase fluctuation. For linear modulation system such as PSK format, its effects can be most easily described by a complex baseband equivalent model. For simplicity the noise in the baseband model is assumed to be white with jointly Gaussian real and imaginary components. In order to mitigate the effects of ISI, variety of receiver structures have been proposed. One widely used in practice due to its simple implementation is the linear equalizer. For a channel introducing only mild interference, the performance achievable by a conventional linear equalizer is often satisfactory. As the channel distorsion becomes severe such that there appears spectral nulls or deep valleys in the Nyquist band, efficiency of linear equalizer is limited by the noise enhancement. In this case Decision Feedback Equalizer (DFE) results in better performance. Unfortunately it tends to propagate errors due to incorrect decision feedback. To overcome strong ISI and lower received SNR, other solutions, based on other criteria, exist : Magee and Proakis [1] have proposed in a former paper the use of the Viterbi Algorithm (VA) directly in conjonction with a channel estimator. Nevertheless for channels with large time spread, this structure becomes too complex. These factors have led researchers to consider measures that might limit receiver complexity while still retaining much of the performance advantages of Maximum Likelihood Sequence Estimation (MLSE). Thus, several authors [2] [3] have proposed a receiver incorporating a linear adaptive filter used to limit the time spread of the channel and mitigate time variation before MLSE. Receivers differ in the form of constraint on the Desired Impulse Response (DIR) which has to be approximated by a feedback filter.

The second part of this paper is based on the approach of Qureshi and Newhall [3] who proposed a DFE-MLSE where both the forward filter and the feedback filter are jointly optimized to minimize the m.s.e. In this approach the first coefficient of the DIR has to be unity and the DIR has to be causal. The latter constraint is discussed further in part (2).

The third part is devoted to the subject of decision-directed convergence and self-recovering adaptive algorithms. The basic idea is that for a Binary-PSK modulation format a reduced VA with a short inherent delay, in parallel with a tap-limited channel estimator, may provide the identification of the principal arrivals without the knowledge of the input symbol sequence $\{U_n\}$. After convergence, the decisions at the output of the VA may serve as a training sequence for a DFE-MLSE. This simple scheme is made possible by the fact that for a binary transmission ($U_n = \pm 1$) the symbol error probability at the output of the VA is necessarily not greater than 0.5. As soon as the symbol error rate becomes inferior to 0.5 the channel identification starts, resulting in more reliable Viterbi decisions, hence, in a recursive manner, in better estimation. Further discussion about implementation is given in part (3) and a receiver which jointly performs blind identification,

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fractionally spaced decision feedback equalization, and reduced MLSE is presented.
In part (4) and (5) the algorithm is tested and proved efficient on both experimental data from a long-range deep water channel and on simulated data from non-minimum phase discret-time channels.

2. ADAPTIVE FILTER FOR MLSE

As Qureshi s' approach, the gist of DFE-MLSE is that the task of the reception of digital signals is shared by an equalizer and a Viterbi detector, as depicted in Block 1b of Fig. 1. The equalizer is a forward filter D(z) with N coefficients fractionally spaced at T/2-second intervals (T is the symbol duration). Its function is to model the Whitened Matched Filter (WMF) proposed by Forney [4]. In our application its aim is to limit Viterbi complexity.
The decision-feedback part consists of a Q-tap filter K(z) in the form $[k_1 Z^{-1} + k_2 Z^{-2} + \ldots + k_Q Z^{-Q}]$. Its length doesn't need to be longer than the causal ISI.
Adaptation of the receiver coefficients for both filters is carried out by the stochastic gradient algorithm as proposed by Benveniste [5] and Bragard [6]. The latter differs from the conventional LMS by the fact that the step-size parameter (with the initial value Δ_1) which controls the rate of adaptation and stability of the algorithm is time-updated according to a second equation with a constant step-size parameter Δ_2. This significant improvement realizes the best trade-off between rapid convergence and small fluctuations in the equalizer coefficients during steady-state operation. Besides, since the transversal part needs fewer taps and is not required to remove ISI completely, Benveniste LMS is a particularly suitable mean to simply adjust DFE.
As regards implementation let us focus on two points : first, in order to force the DIR to be causal, the delay τ_2 must be equal to the delay through the VA ; secondly, to prevent instability it is advisable to duplicate the forward filter rather than to update its coefficients by means of a feedback loop with delay.
An important consideration on optimum receiver is that at low SNR the m.s.e. criterion does not allow the fractionally spaced equalizer to perfectly model the WMF. Actually, only a zero-forcing formulation might constraint the forward filter to produce zero anticausal ISI and white noise. However, since the causal DIR is strong enough to compensate for channels with spectral nulls, the forward filter D(z) in the structure of Block 1b approaches the WMF at moderately high SNR, resulting in uncorrelated noise at the VA input.
The crucial problem of the selection of the DIR, i.e. the filter " 1+K(z) ", is let to part (3). Concerning this point, let us emphazise that the selection of the DIR is directly related to the arrivals which are taking into account during the blind identification stage (Block 1a) and thus to the computation of the training sequence used to adjust DFE-MLSE.

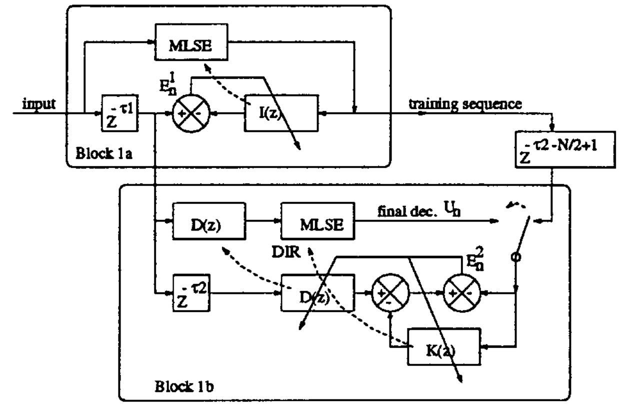

Figure 1: Block-diagram of the receiver

3. SELF-RECOVERING ADAPTIVE RECEIVER

Most blind equalization methods proposed in recent years have been based on higher-order statistics. The main reason is that for nonminimum phase channels, second order statistics of a stationary process do not contain the phase information. However the major shortcomming of these methods is the low convergence rate. To overcome this drawback several methods based on other criteria [7] or based on array antenna approach [8] have been proposed. Yet these methods are not well adapted to our particular application : communication through a noisy channel with severe time-varying distorsion.
As mentioned in the introduction the principle of our proposal is to use two reduced VA as described in the general block-diagram of Fig. 1. The first one, in Block 1a, works in conjonction with the FIR transversal filter I(z). The latter is the channel estimator. Block 1a is the classical Viterbi detector scheme, but here, the purpose is to select only the main (Q+1) arrivals. Its (Q+1) coefficients are adjusted with the previous modified LMS and then fed to the MLSE-based VA for use in the metric computations. At this stage it is noteworthy that all decisions provided by the first VA dont need to be correct. In fact it is the VA in Block 1b which makes reliable decisions on the assumption that the DIR is the actual overall channel response. This second Viterbi also works with (Q+1) taps. After a training period, these decisions are fed to the feedback

part of the equalizer.
This scheme is of interest from two points of view : first, it allows blind identification/equalization with a fast convergence rate, secondly, it provides a mean to achieve a good frame-synchronization resulting in a well chosen DIR.
To sum up, the first stage (block 1a) makes tentative decisions that yield convergence of the DFE and the second stage (block 1b) produces final reliable decisions.

4. EXPERIMENTAL RESULTS

The proposal receiver is demonstrated on experimental data from the Atlantic Ocean [1] . The transmission range is 63 km corresponding to a convergence zone. The transducer is 150 m below the surface. The receiver is a hydrophone at depth of 245 m.
The symbol rate is 511 symbols per second. The carrier frequency is 1022 Hz. The data are pseudonoise sequences.
Fig. 2 shows an example of the instantaneous overall channel impulse response. It indicates a multipath delay spread of about 24 ms corresponding to 12 symbols. The estimated input SNR is less than 10 dB.
Experimental results are obtained with a 24-tap prefilter D(z) whose taps are adjusted with $\Delta_1 = 0.001$, $\Delta_2 = 0.00005$, and a 6-tap DIR length (Q+1 = 6) with the adjustement parameters equal to 0.05 and 0.0005 respectively. Concerning the identification part, the adaptive channel estimation is accomplished by a 6-tap filter I(z) adjusted with $\Delta_1 = 0.01$ and $\Delta_2 = 0.0005$. The inherent Viterbi delays are $\tau_1 = 3T$ and $\tau_2 = 6T$.
The results are illustrated in Fig. 3 a,b and 4 a,b. Fig. 3a and 3b show the instantaneous quadratic error E_n^1 and E_n^2 over a data block of 3500 symbols. The initial training sequence consists of 200 symbols from the output of block 1a. After this period the coefficients of the DFE are adjusted in a decision-directed manner. Fig. 4a and 4b show the amplitude of the coefficients of the filters D(z) and "1+K(z)", at the end of the computation.
The main results are first, that the settling time of the DFE is reached in 200 points only, without the knowledge of the input symbol sequence, secondly, that the number of symbol errors over the last 2500 points is 0 at the output of DFE-MLSE instead of 61 at the output of block 1a .

[1] Data files were provided by the *Centre Technique des Systèmes Navals.*

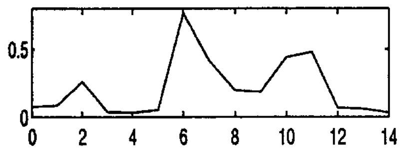

Fig. 2 : Channel impulse response

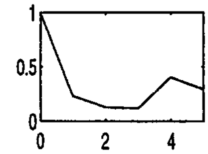

Fig. 4b : DIR

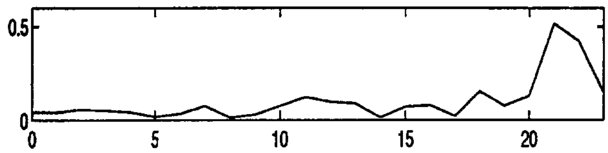

Fig. 4a : Tap-coefficients of D(z)

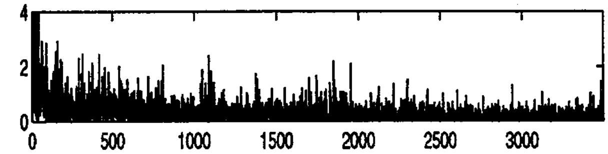

Fig. 3a : Instantaneous quadratic error of block 1a

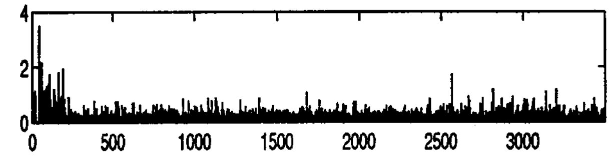

Fig. 3b : Instantaneous quadratic error of block 1b

5. SIMULATION RESULTS

Baseband communication system is modeled using discret-time channels sampled at half the symbol rate. Fig. 5 shows for each channel the impulse response represented by a linear 15-tap filter fractionally spaced at T/2-second intervals. In both cases the SNR is 10dB. Simulation is run with a 3-tap filter $I(z)$, for the identification (Fig. 5a), a 3-tap filter for modeling the DIR (Fig. 7b), and a 20-tap filter $D(z)$ fractionally spaced at T/2-second intervals (Fig. 7a). The inherent Viterbi delays are $\tau_1 = \tau_2 = 3T$.
The results are illustrated in figures 5a, 6 a,b and 7 a,b. We add Fig. 5a in order to point out that only the principal arrivals are selected during blind identification period.The number of symbol errors over the last 1500 points is 0 at the output of DFE-MLSE instead of 134 and 42 at the output of block 1a, respectively.

Concluding remarks :

- the second VA makes decisions on the assumption that the DIR is the actual overall channel response. In this manner VA works with only 3 taps instead of 7 (corresponding to a delay spread of 6T seconds as depicted in Fig. 5). This allows the MLSE to be made of practical size with only a small degradation in performance ;

- contrary to conventional linear self-recovering equalizers, the proposed algorithm successfully copes with severe nonminimum phase channels in taking advantage of nonlinear IIR structure. In this respect it is worthwhile to mention a new other self-recovering equalizer using nonlinear IIR filter [9]. Hence comparing both algorithms would be of great interest.

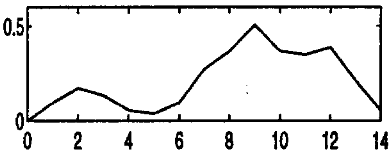

Fig. 5 : Channel impulse response

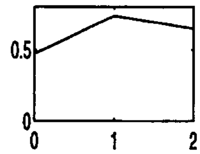

Fig. 5a : I(z)

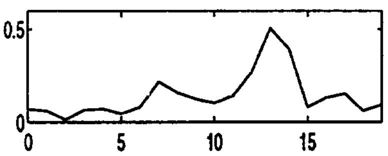

Fig. 7a : Tap-coefficients of D(z)

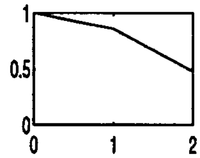

Fig. 7b : DIR

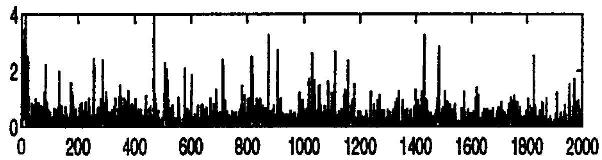

Fig. 6a : Instantaneous quadratic error of block 1a

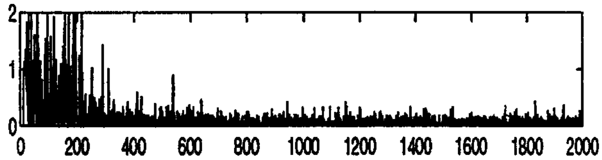

Fig. 6b : Instantaneous quadratic error of block 1b

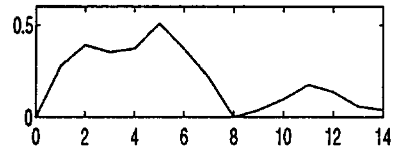

Fig. 5 : Channel impulse response

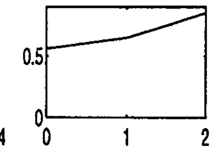

Fig. 5a : I(z)

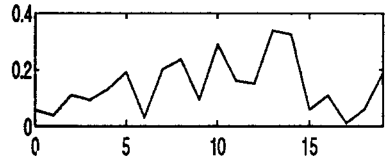

Fig. 7a : Tap-coefficients of D(z)

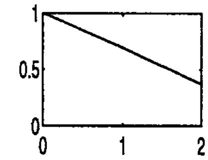

Fig. 7b : DIR

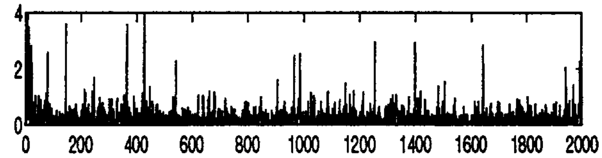

Fig. 6a : Instantaneous quadratic error of block 1a

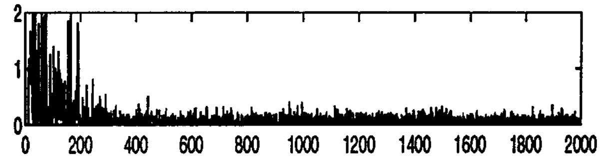

Fig. 6b : Instantaneous quadratic error of block 1b

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