17th SYMPOSIUM ON HYDROACOUSTICS

Jurata May 23-26, 2000

NOVEL EQUALIZER FOR SHALLOW WATER ACOUSTIC COMMUNICATIONS

Y-h. Yoon, A. Zielinski

University of Victoria,
Victoria, B.C., CANADA
Adam.Zielinski@ece.uvic.ca

ABSTRACT

The signal transmitted through an underwater acoustic channel is corrupted by multipath propagation caused by surface and bottom reflections. This delay spread causes intersymbol interference (ISI). The surface waves produced by wind affect the signal delay spread at the receiver. The delay spread is large for a calm sea and decreases with rough sea agitated by winds. To compensate for distortion introduced by the multipath channel, adaptive equalizers have been utilized to improve the performance of a communication system. These equalizers have been designed to handle the worst case of a signal delay spread condition associated with a calm sea and have a suitable and fixed number of taps. However, it is known that the power consumed by the processor increases with the number of computations performed per time unit. In this paper, we propose a novel equalizer with a variable number of taps which adaptively changes depending on channel conditions in order to conserve power. The proposed equalizer was tested by computer simulations using a model of an underwater acoustic channel. Results indicate savings in computational load up to 54% for a selected case. Power savings were also obtained when a directional receiver is used together with the proposed equalizer. Lower power consumption is particularly desirable for battery operated systems.

1. INTRODUCTION

The signal transmitted through an underwater acoustic channel suffers distortion due to multipath propagation as well as phase shifts caused by Doppler effects due to relative motions of receiver and transmitter [1, 2]. The channel condition varies in time with sea state as affected by surface wind. Both the time varying multipath and Doppler effects can be mitigated by jointly optimized adaptive equalization and synchronization at the receiver. More specifically, a receiver structure which employs a decision feedback equalizer (DFE) combined with a digital phase-locked loop (DPLL) has been shown to be effective in ameliorating these effects [3, 4]. While adaptive equalization is effective in compensating for the distortions introduced by multipath propagation, a DPLL is used to acquire and to track carrier reference needed for coherent demodulation.

In this paper, a model of a time varying underwater acoustic channel is used [7]. When the sea state changes in response to wind speed, the signal delay spread at the receiver varies greatly. It decreases with wave height caused by higher wind velocity. This is due to higher losses encountered during signal reflections from a rough surface. Presently used equalizers [2, 4] are designed to handle the worst case conditions (large signal delay spread) and have a required and fixed number of taps. This leads to unnecessary operations in favorable channel conditions (rough sea). As demonstrated in [5], a greater number of operations per unit of time results in more power consumption of the system which in turn shortens the operation period of a battery operated system.

We propose a novel equalizer structure with a variable number of taps. The performance of the proposed structure in a time-varying underwater acoustic channel is investigated by computer simulations using a channel model described in [6, 7].

2. VARIATION OF WIND SPEED

The effect of variation of wind speed as well as the directionality of the receiver on the channel impulse response are investigated. The following parameters of the channel and system were assumed for this analysis: range; 10 km, ocean depth; 50m, transmitter and receiver depth; 25 m, carrier frequency; 10 kHz, system bandwidth: 2 kHz, transmission rate: 4 ksymbols/s

We have investigated the envelope of impulse response of the system for three different wind speeds (5, 8 and 10 knots) for a non-directional receiver. We noted that the variation of signal delay spread can be as large as 1:20 as the wind speed varies between 5 and 10 knots. To investigate the effect of using a directional receiver, a simplified beam pattern was postulated. The main beam has a constant level within its beamwidth. A constant side lobe is suppressed by X_{dB} = 20 dB with respect by the main lobe. As expected we observed reduction of delay spread as the result of the receiver directivity, particularly for narrow beam.

To summarize the above results, we note that there is a large variation in signal delay spread caused by wind speed for a non-directional receiver. Application of a directional receiver will reduce this variation to a certain extent. The proposed method of variable-tap equalizer is particularly effective for a non-directional receiver.

3. STRUCTURE OF THE DFE/SYNCHRONIZER

A. Equalizer Optimization Algorithm

The structure of a receiver suitable for joint equalization and synchronization was proposed by Falconer [3] and studied further for the underwater channel by Stojanovic *et al.* [4]. This structure has been generalized by the authors to allow a variable number of equalizer taps. The equalizer consists of the adaptive feed-forward and feed-back transversal filters $\boldsymbol{A}_{k-1}$ and $\boldsymbol{B}_{k-1}$ with a variable number of equalizer taps shown in Figure 1 as shaded boxes.

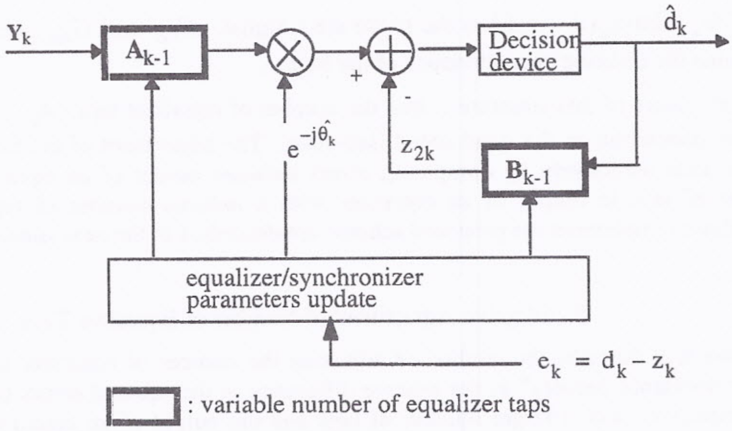

Figure 1. Structure of the DFE/Synchronizer

In this study, we selected a decision feedback equalizer where the feed-forward filter is fractionally spaced with spacing T/2 while the feed-back filter is symbol spaced at spacing T. The sampled complex signal $\boldsymbol{Y_k}$ is passed through the adaptive transversal filter $\boldsymbol{A_{k-1}}$ intended to minimize the intersymbol interference. The most commonly used criterion for the optimization of the equalizer coefficients is the minimization of the mean square error (MSE), that is,

$$J = \min([E\,|\,e_k\,|^2]), \tag{1}$$

where e_k is the error signal:

$$e_k = d_k - z_k \tag{2}$$

while d_k and z_k are the actual transmitted data and the equalizer output at $t = kT$, respectively. For an ideal channel, $d_k = z_k$.

The minimization of the MSE is accomplished recursively by the use of the least mean square (LMS) algorithm [2].

$$\begin{aligned} \boldsymbol{A}_k &= \boldsymbol{A}_{k-1} + \mu e_k \boldsymbol{Y}_k^* e^{-j\hat{\theta}_k} \\ \boldsymbol{B}_k &= \boldsymbol{B}_{k-1} - \mu e_k \hat{\boldsymbol{D}}_k^* \end{aligned} \tag{3}$$

where μ is a positive adaptation constant (step size), $\hat{\theta}_k$ denotes the estimated phase error and * denotes the complex conjugate of the vector.

The equations to track the carrier phase $\hat{\theta}_k$ can be derived using the steepest descent (gradient) approach [2, 3]

$$\hat{\theta}_{k+1} = \hat{\theta}_k + G_{1,\theta_k} + \sum_{i=0}^{k} G_{2,\theta_i} \Phi_i, \tag{4}$$

where $\Phi_k = \mathrm{Im}\{z_{1k}[d_k - z_k]^*\}$ is the phase error signal, $G_{1,\theta k}$ and $G_{2,\theta k}$ are loop gains which determine the tracking characteristics of the loop.

A novel feature of this structure is that the number of equalizer taps ($\boldsymbol{A}_{k-1}, \boldsymbol{B}_{k-1}$) can be made variable depending on the wind speed (sea-state). The adjustment of the number of equalizer taps is done recursively by comparing errors between output of an equalizer with a larger number of taps to output of an equalizer with a reduced number of taps. Details of the algorithm to implement the proposed scheme are described in the next subsection.

B. Adaptive Adjustment of Number of Equalizer Taps

Here, we will describe the method of adjusting the number of equalizer taps. Let us define "the performance penalty" as the relative difference in the squared errors between the output of an equalizer with a larger number of taps and the output of an equalizer with a reduced number of taps. The number of equalizer taps are decreased whenever two conditions are simultaneously met:

- the squared error at the output of an equalizer, $|e_k|^2$ as given by Eq. (2), is below the predefined value, PSE; and
- the performance penalty is below a certain specified value, PP.

The first condition guarantees the operation of the equalizer with a performance better than a certain minimum allowable level, PSE. The second condition controls the level of the performance degradation due to a reduced number of taps. In other words, assuming a larger PP will produce a larger value of squared error at the output of equalizer, but with a smaller number of equalizer taps. We expect that this will translate to a larger bit error rate (BER) for a system operating with a certain ambient noise condition [8]. This expectation is supported by results of computer simulations. This method is based on the observation that when the system impulse response becomes shorter in rough seas, those equalizer taps outside of system impulse responses will have a quite small effects on the computation of the equalizer output. This allows us to reduce the number of taps. When the number of taps becomes smaller than is needed to cover the system impulse response, the PP rapidly increases. This is due to the fact that taps with significant contributions start to be truncated. For a faster convergence, the number of taps to be increased/decreased in each iteration, Δ_n, can be larger than one. To stabilize the proposed equalizer operation, we restrict the number of taps within a certain range ($N_{eq,min}$, $N_{eq,max}$).

4. SIMULATION RESULTS

In this section, computer simulation results for the proposed structure using the postulated parameters of the channel and system are presented. We assume a Doppler frequency shift of f_d = 3.33 Hz corresponding to a relative velocity between transmitter and receiver of 1 knot. Figure 2 shows the variation of number of equalizer taps N_{eq} when wind speed is abruptly changed. The operation of an equalizer starts with an initial number of taps N_o = 225 associated with a calm sea. The initial increases in the number of equalizer taps after wind speed has abruptly changed is

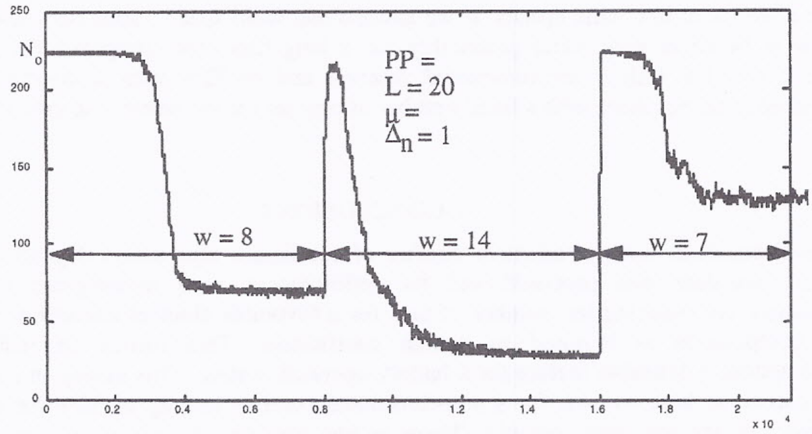

Figure 2. The variation of number of equalizer taps when wind speeds are changed

caused by a fact that a sudden change to impulse response with a new sea state results in mis-adjustments of tap coefficients which increases the squared error at the output of the equalizer. An equalizer needs time to adjust to a new system impulse response and it will converge to a reduced number of taps at steady state. The results illustrate that the proposed method works as expected; the number of taps is adjusted depending on the wind speed.

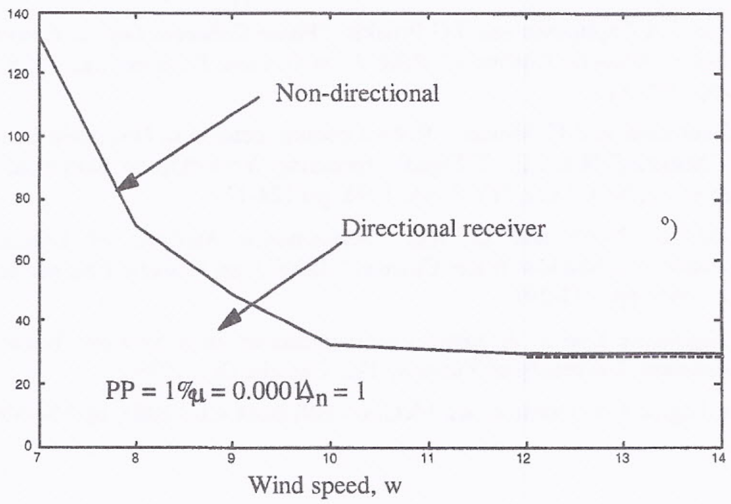

Figure 3. Number of equalizer taps vs. PP at steady state

Figure 3 shows the number of equalizer taps as a function of wind speed when a directional receiver is used with the proposed structure. As we can see, a directional receiver is much more effective at low wind speeds. If we assume that wind speed varies between w =7 knots and w = 14 knots with equal probability for a long time, the computational load can be reduced to 54% with a non-directional receiver and to 32% with a directional receiver compared to an equalizer with a fixed number of taps set for the worst case condition of w = 7 knots.

5. CONCLUSIONS

A novel equalizer with an adaptive number of coefficient taps which depend on the time-varying sea-state was proposed and its performances were investigated by computer simulation. By reducing the number of taps for a favorable channel condition (rough seas), less computation is required to update coefficients. This results in reduced power consumption, a desirable feature for a battery operated system. This saving in computational load can be as high as 54% for a non-directional receiver and up to 32% for a directional receiver for the assumed scenario. These results obtained by computer simulation for a specific case encourages further more general analysis of such a structure.

REFERENCES

1. J.Captipovic, "Performance Limitations in Underwater Acoustic Telemetry," *IEEE J. on Oceanic Engineering*, Vol. 15, 205-216 (1999)
2. J.Labat, "Real Time Underwater Communications," *Proc. of Oceans 94,* Brest, France, Sept. 1994, pp. III.501-III.506.
3. D.D Falconer, "Jointly Adaptive Equalization and Carrier Recovery in Two-dimensional Digital Communication," B.S.T.J., No.3, March 1976, pp.317-334.
4. M. Stojanovic, J.A Captipovic and J.G Proakis, "Phase Coherent Digital Communications for Underwater Acoustic Channels," IEEE J. on Oceanic Engineering, Vol. 19, No. 1, Jan 1994, pp. 100-111.
5. A.Eliaz, J.Tabrikian, and H. Messer, "Robust communication in Uncertain Shallow Water Acoustic Channels," Ninth IEEE Signal Processing Workshop on Statistical Signal and Array Processing, New York, NY, USA, 1998, pp.124-127.
6. A. Zielinski, Y. Yoon and L. Wu, "Performance Analysis of Digital Acoustic Communication in a Shallow Water Channel," IEEE J. on Oceanic Engineering, Vol. 20, No. 4, Oct. 1995, pp. 293-299.
7. Y.Yoon, "High-Rate Digital Acoustic Communications in a Shallow Water Channel," Ph.D. dissertation, University of Victoria, BC, Canada, Oct. 1999.
8. J.G Proakis, Digital Communications, McGraw-Hill Book Co., 1983, pp.554-598.