

Implementation Adaptive Decision Feedback Equalizer for Time-Reversal Communication in Shallow Water Environment

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Abstract: Underwater wireless communications are growing very fast along with human needs for applications such as defense, state security, underwater control and monitoring systems. Until now, an acoustic signal is a practical way to achieve long distance communication in the ocean. However, the underwater acoustic channel faces many challenges including limited available bandwidth, long delays, time-variability, and Doppler-spread. These challenges can reduce the reliability of the communication system and the achievement of high data-rate becomes a challenge. Adaptive decision feedback equalization is a method to compensate for the distortion of information signals on the underwater acoustic channel. On the other hand, time reversal is an effective method of overcoming intersymbol interference (ISI) problems which is the effect of multipath phenomena in underwater channels. Spatial focusing on time reversal can reduce the co-existing system disturbances and its temporal focusing makes the received power concentrated within a few taps so that the equalizer design work becomes much simpler. The temporal focusing can also increase the transmission rate. This paper shows that the combination of time reversal and adaptive DFE (TR-DFE) has superior performance than TR and DFE itself. By modifying the step-size parameters in the adaptive DFE, the TR-DFE level of convergence and performance can be improved. The geometry-based modeling which is used proves that distance and multipath variation greatly affect the quality of time reversal communication on the underwater acoustic channel.

Keywords: underwater acoustic communication; passive time-reversal; adaptive DFE; multipath propagation; inter-symbol interference (ISI)

INTRODUCTION

The underwater acoustic communication (UAC) system has received much attention from most researchers today. Underwater acoustic sensor networks are widely studied for having potential applications in marine fields such as marine exploration, underwater robots, offshore oil industry exploration, pollution monitoring and many other examples of applications. Currently, designing a reliable underwater communication system is an active research area. It is important to know the purpose of using underwater acoustic communication before designing a communication system. Take knowledge of the direction of communication on an underwater acoustic channel as an example. Vertical underwater communication has a lighter challenge because there is usually only a small amount of multipath. While horizontal underwater communication is strongly influenced by long and unpredictable multipath reflections, especially in high-frequency underwater communications. In the deep ocean, the signals rarely interact with the sea surface. This causes communication channels to be time-invariant, possibly sparse and widespread delays. In shallow water environments, experiencing reflections with sea surface and sea bottom can't be avoided and faced with obstacles more often. In addition, signals interact with noise derived from ocean waves, shipping traffic, and marine life. Depth, geometry system, water column chemistry are unique parameters found in every marine environment. Therefore until now, there has been no universal model of the underwater acoustic channel. To get an underwater channel model is usually adjusted based on in-situ measurements.

The reason why most acoustic signals are used for underwater communication is due to the facts that electromagnetic waves and laser always experience with high path loss in the underwater environment. Electromagnetic waves are also commonly used in terrestrial communication, but when applied for short-range in underwater communication, they will experience with high attenuation. In the range of megahertz to gigahertz radio frequency is only able to propagate a few meters before weakening and in the optic range (especially blue-green light) can reach about hundreds of meters. In contrast, the acoustic signals can propagate at long distance in the sea and have relatively low attenuation. Acoustic signals can also reach up to thousands of kilometer and are used in almost all underwater environments (Kilfoyle and Baggeroer, 2000). However, the level of complexity in underwater acoustic communication is higher when compared to radio channel communications. One of the main obstacles of underwater acoustic communication with limited-bandwidth is the inter-symbol interference (ISI). If the spread of multipath on the radio channel moves about two or three symbol intervals, then the acoustic channel underwater can increase up to several tens of symbol intervals. Over the last ten years, much research has focused on equalizer design to overcome ISI and compensate for channel variations (Stojanovic, 1993); (Yang, 2000). This equalization technique requires complex computation, algorithm stability, and channel parameter selection (Yang, 2005). Moreover, to improve the performance of the communication system, the single-channel receiver is developed into a multichannel receiver system. However, the multichannel system requires the equalization technique with high computational complexity (Stojanovic, 1993); (Qureshi, 1985).

Currently, PPC (passive phase conjugation) or time reversal has been widely applied to several studies in the field of underwater acoustic communication with a relatively simple approach and has a low level of computational complexity (Rouseff, et.al, 2001); (Widiarti et.al, 2020). Temporal focusing of TR property can minimize ISI, while its spatial focusing can reduce the channel fading effect and improve the signal-to-noise ratio (SNR). The concepts and experiments of active time reversal communication have been shown (Rouseff, et.al, 2001); (Song. et.al, 2006). To reduce the complexity of a receiving array, the TR passive technique requires only one-way communication. In passive TR spatial diversity is obtained by sampling the sound field with multiple receivers. Reduction of multipath which is obtained by PPC has its own challenges in the presence of a time-varying channel environment so that the perfect channel estimation can't be gained. The delayed multipath arrivals cause inter-symbol interference (ISI) so they can damage the transmitted information, therefore an equalizer is needed to eliminate the ISI. Although the focusing property of TR can overcome ISI, the resulting side lobes can't be simply removed. There are several commonly used equalization techniques such as zero-forcing, MMSE, and block equalization. Decision Feedback Equalization (DFE) is an effective equalization technique because it eliminates ISI in addition to compensating for channel changes. If the communication channel is not known, then the adaptive equalization technique can be used to extract the channel response from the symbol training sequence first and then compensate for channel distortion in the arrival data symbol (Proakis, 1989).

This paper presents a combination of passive time reversal and adaptive equalization techniques by using adaptive DFE to overcome the residual ISI. This paper also models the shallow water environment by using geometry-based model and environmental parameters. The geometry-based model used refers to the real conditions of the towing tank where the experiment will be conducted (Widiarti et.al, 2018). The channel impulse response obtained by weighting of multipath signals that are attenuated by reflection, absorption, and so on. The contributions of this paper are:

1. Performance analysis of passive reversal time communications system combined with adaptive equalization (TR-DFE) with data simulations on the multipath environment that varies in a shallow water environment
2. The analysis of modified DFE and unmodified DFE equalization to achieve the better system performance.

METHODOLOGY

The underwater acoustic channel has time-varying multipath characteristics due to the reflection of the sea bottom and sea surface, as well as transmitting through the internal wave (Blomgren. et.al, 2008). To compensate for the effects of the channel then an adaptive equalizer is required. In conventional signal processing, the tap coefficient on the equalizer is determined using the minimum MSE criteria, while the tap count is determined by the physics-multipath channel (19). Most of the poor performance gained comes from a neglected time range.

1. Passive Time-Reversal Technique

Basically, the concept of PPC and TR are the same, both use vertical arrays to suppress ISI. In the communication process, the information signal consists of a series of symbols that are notated I_n and each symbol has a duration of T , then the baseband data signal can be expressed as follows:

$$s(t) = \sum_n I(n)g(t - nT) \quad (1)$$

Where $g(t)$ is a pulse shape function for each symbol, so that:

$$g(\tau) = \begin{cases} 1, & \text{for } 0 \leq \tau < T \\ 0, & \text{otherwise} \end{cases} \quad (2)$$

Signals received on the j -th receiver in the UAC can be expressed by the following equation:

$$r_j = h_j * s(t) + w_j(t) \quad (3)$$

In the above equation, h_j is the impulse response channel, w_j is a band-limited noise, while the notation $*$ shows the convolution of the transmitter filter and the receiver filter with the impulse response function. In the equation, the match-filtering process on the received signal is applied, and the output produced after the match-filtering process can be expressed as follows:

$$\begin{aligned} z_j &= h_j(-t) * r_j(t) \\ &= h_j(-t) * (h_j(t) * s(t) + w_j(t)) \\ &= h_j(-t) * h_j(t) * s(t) + h_j(-t) * w_j(t) \\ &= Q_j(t) * s(t) + \zeta_j \end{aligned} \quad (4)$$

$Q_j(t)$ in the above equation is the autocorrelation of the impulse response function $h_j(t)$, while $\zeta_j(t)$ is filtered noise. The performance of the passive reversal time depends on the function of $Q_j(t)$. If $Q_j(t)$ does not approach the Dirac function, then the side lobes of $Q_j(t)$ may cause ISI. The ISI can be reduced by time reversal refocusing. It is assumed that there are K receivers and using equal weight combining, then one channel output can be written as follows:

$$z(t) = \sum_{j=1}^K z_j(t) = \sum_{j=1}^K Q_j(t)s(t) + \sum_{j=1}^K \zeta_j(t) = Q(t) * s(t) + \zeta(t) \quad (5)$$

Where $Q(t)$ represents the total autocorrelation response of $Q_j(t)$ summed for all K receivers/channels.

2. Decision Feedback Equalizer

As nonlinear equalizer DFE has a common form as follows:

$$\hat{I}_k^n = \sum_{j=-N_{ff}+1}^0 a_k^j V_k^{n-j} + \sum_{j=1}^{N_{fb}} b_k^j \hat{I}_k^{n-j} \quad (6)$$

Where $a = \{a_k^0, \dots, a_k^{N_{ff}-1}\}^*$ and $b_k = \{b_k^1, \dots, b_k^{N_{fb}}\}^*$ are tap coefficient vectors for feedforward and feedback filters with each length N_{ff} and N_{fb} . \hat{I}_k^{n-j} denotes the symbol of the best decision result that

approximates the estimated symbol \hat{I}_k^{n-j} . Figure 1 shows the block diagram of DFE. When compared to the linear equalizer, in the DFE there is a feedback loop where filter $B_k(z)$ uses \hat{I}_k^n as input so that DFE becomes non linear. W_n^k denotes the additive noise, $A_k(z)$ and $B_k(z)$ denotes the z transforms of tap, a_k and b_k coefficients for forward feed filters and feedback where $z = e^{j\omega t}$. The b_k value is uniquely determined by the CIR and the value of $N_{fb} = L-1$ is absolutely necessary. Although N_{fb} is independent of CIR, a_k and b_k are interrelated. Assuming that the previous symbol is correctly detected in the feedback filter, the tap coefficient is obtained by minimizing the MSE output. TR communication based on back-propagation utilizes focusing property to overcome ISI. The focusing on TR consists of two aspects: (1) temporal focusing determined by independent time delayed arrivals, and therefore the rate of channel variation determines the duration of time from temporal focusing, (2) spatial focusing obtained from spatial diversity [21] so that the physical condition of the medium affects the quality of spatial focusing.

$$J_{mse}(a_k, b_k) = E(|I_n - \hat{I}_k^n|^2) \quad (7)$$

The focusing of the TR can be interpreted as matched-field processing when the sound field in a water column is sampled with multiple independent hydrophones. It is assumed that the channel information is captured perfectly by many hydrophones, so the focusing function denoted by the Q function approaches the sinc function [19] where the width of the main lobe is determined by the signal bandwidth. This physical limitation that causes ISI can't be removed simply by focusing TR. In fact, the underwater channel is time variant, where focusing will decrease with time. Therefore, an adaptive equalizer is required to eliminate residual ISI as well as to detect the channel variations.

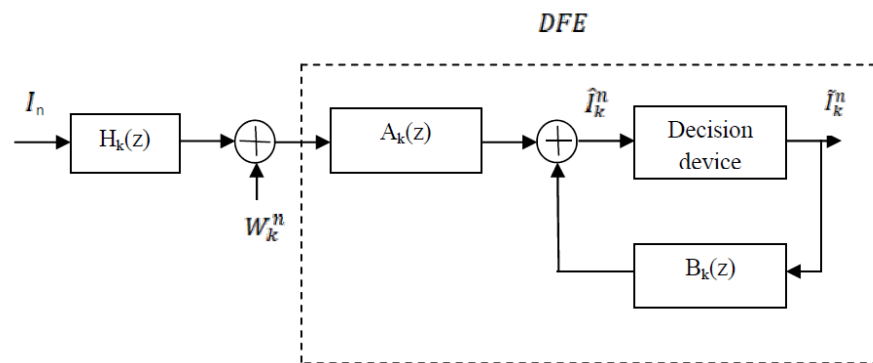


Figure 1. Block diagram of adaptive DFE

Figure 2 shows a block diagram of the TR receiver combined with adaptive DFE. The least mean squares (LMS) algorithm is used in adaptive equalization here. The advantage of LMS is in its simplicity and low level of computational complexity. The LMS algorithm also does not require the measurement of correlation or inverse matrix functions. The convergence rate is determined by spreading the Eigenvalue of the correlation matrix from the input symbols. The downside of the LMS algorithm is that when the Eigenvalue spread is large the LMS algorithm will be slow because it requires a certain number of iterations to converge. For that required step modification to get a better convergence level.

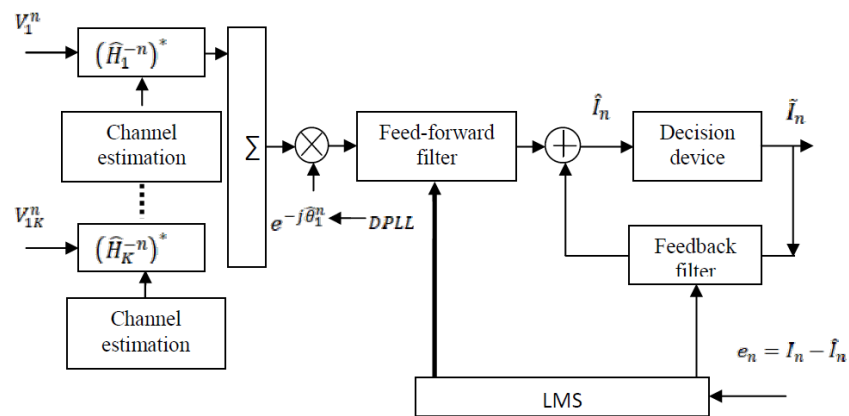


Figure 2. Block diagram of joint passive time reversal and adaptive DFE

RESULTS AND DISCUSSION

The geometry-based channel model is done in accordance with the towing tank conditions used for the experiment. The towing tank has dimensions of 200 m x 12 m and a depth of 6 m. The medium condition is uniform, there are no waves, and no transient noise in the environment around the towing tank. Table 1 is an underwater acoustic channel parameter used to calculate the length of the direct and multipath. In this paper, it is assumed that the main error source is multipath interference and the background noise is assumed to be Gaussian distributed.

Table 1. Underwater Acoustic Channel Parameters

Parameter	Values
Depth	6 m
Sound speed	1498 m/sec
Transmitter height (Tx)	3 m
Receiver height (Rx)	4 m
Tx-Rx distance	40 m; 80 m; 100 m

Assuming that the wind speed in the room is 0.22 knot and referring to the paper (Santoso et.al., 2016) the speed of sound propagation in water is 1498 m / sec. The distance between the transmitter and receiver is 100 m, and the transmitter height from the bottom of the pool is 3 m, while the receiver's height from the bottom of the pool is 4 m. Figure 3 shows the performance of a TR communication system in the underwater acoustic channel. The bit error rate (BER) parameter of the Signal to Noise Ratio (SNR) value is used to see how the combined passive time reversal and adaptive DFE works. In this paper, the modulation scheme used is BPSK with carrier frequency used is 10 kHz. The data sequence containing 10^6 bits and 5×10^3 bits training symbols are transmitted from the transmitter to the receiver through the underwater acoustic channel. The simulation compares the performance of a single carrier system between MSE linear equalization, combined time reversal and MSE linear equalization, time reversal without equalization, DFE adaptive equalization, and the combination of time reversal and adaptive DFE.

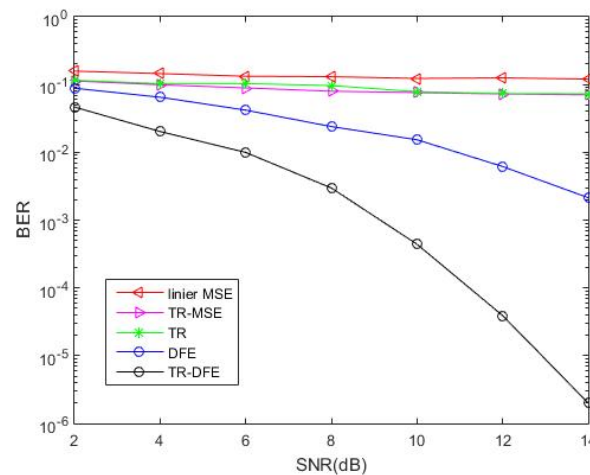


Figure 3. BER value of various methods

From the resulting curve, we can see that BER value on TR-DFE is the best among other methods used. There is a large gap between DFE and TR-DFE on the resulting BER value in the range of SNR value. While the curves generated by TR and TR-MSE tend to coincide. However, there is a little difference in the value of BER at SNR between 5 dB to 10 dB where TR-MSE is superior to TR without equalization. In general, the combined TR and equalization have superior system performance than TR without equalization and the adaptive equalization is superior to the linear equalization of underwater acoustic communication. Furthermore, the performance of TR-DFE is compared to the variation of the distance between the transmitter and receiver. By using the geometry-based model, the performance of TR-DFE was observed at 40 m, 80 m, and 100 m variations. The resulting BER value of TR-DFE at a distance of 40 m has the best performance compared to the other two distances.

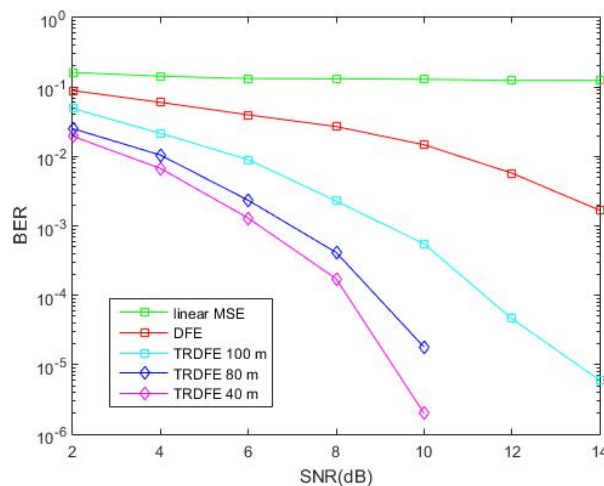


Figure 4. BER value comparison of TR-DFE at various distance

While at a distance of 80 m and 100 m the performance of TR-DFE is superior when compared to the communication with the single carrier using only adaptive equalization or linear equalization MSE at 100 m distance. The conditions where the system performance at a distance of 40 m is better than the system performance at distances of 80 m and 100 m are due to the greater distance resulting in attenuation experienced by the signal is also getting bigger. However, 40 m is too short for underwater acoustic communication, whereas the experiment uses a minimum distance of 100 m between the

transmitter and receiver. The resulting BER value can still be increased by adding the number of array elements to the receiver by considering the efficiency of the system. The result of the BER value is represented in Figure 4.

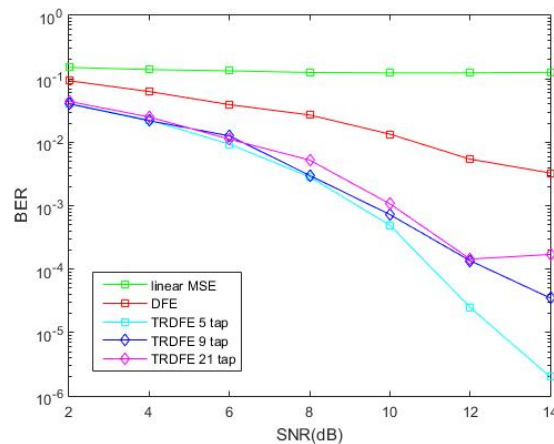


Figure 5. BER value comparison of multipath variations

Multipath in shallow water environment varies according to an environmental condition such as the number of multipath and the resulting attenuation coefficient value. This will affect the performance of the time reversal communication system on the underwater acoustic channel. Thus, by using the geometry-based channel model, the effect of the multipath variations and signal attenuation to the system performance can be analyzed. It can be shown in Figure 5 that the environment with the smallest number of multipath has a superior BER value. In the simulation, the system performance with 5 taps is the best compared to 9 taps and 21 taps multipath. In more widespread multipath environment conditions, the higher equalization capability is needed to eliminate ISI. The resulting BER curve shows that at the higher SNR there is a larger gap between TR-DFE with 5 taps and TR-DFE with 9 taps or 21 taps.

Performance Analysis of Modified Adaptive DFE

To improve the work performance of TR, then on the receiver side TR technique combined with adaptive DFE. In this study LMS algorithm based DFE is used. By using 5×10^3 bits training symbol and a step size of 0.045 then the convergence of TR-DFE can be achieved. However, characteristic of the LMS algorithm is a slower rate of convergence, hence required a modification to its step size. In this study modification of step size in LMS is done by multiplication with the absolute value from the difference of the last two errors. Thus, the modified DFE results are faster in convergence than before modification. At the 100th iteration, the MSE value in the modified DFE is close to zero. If compared to DFE without modification, the same MSE value is achieved at the 300th iteration. In Figure 6 can be shown that the TR-DFE system with modified LMS step size is better in terms of its convergence rate compared to TR-DFE before modification. The modified TR-DFE is shown by red curve and TR-DFE before modification is shown with the blue curve. The number of training symbol also affects the adaptive DFE performance. By transmitting 10^6 information bits, the superior system performance can be obtained when DFE uses 5×10^3 bits of training symbol. In this simulation, the number of bits in the training symbol is made varies 10^3 bits, 3×10^3 bits and 5×10^3 bits.

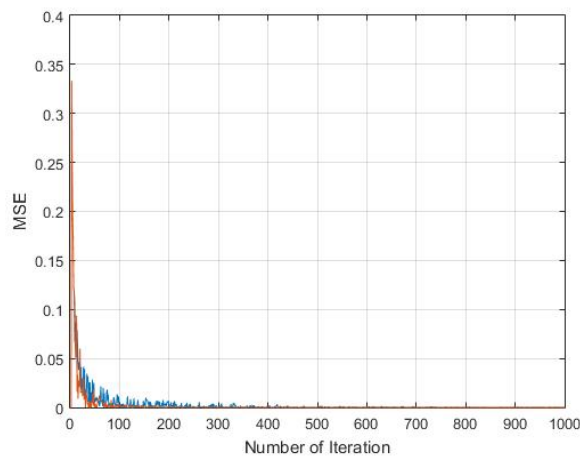


Figure 6. Convergence of TR-DFE with modified DFE

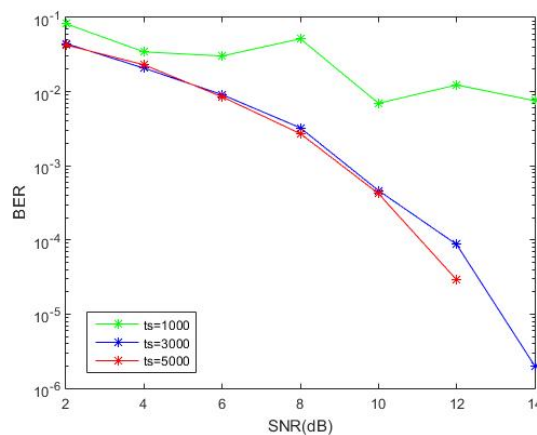


Figure 7. Comparison of BER value with a variation of the number of training symbols

The simulation result shows that the most superior BER is obtained in 5×10^3 bits training symbol. The optimal number of training symbols is required for efficiency and equalizer performance improvement. What should be avoided is the inefficiency of the number of training symbols more because the level of convergence will also decrease. The simulation result can be seen in Figure 7.

CONCLUSIONS

In this paper, the shallow water channel modeling based on geometry and environmental parameters have been done. TR-DFE is a powerful technique in overcoming multipath effects in the shallow water environment. The simulation results show that the combination of both adaptive DFE and TR techniques is capable of improving the performance of underwater acoustic communication system as evidenced by the resulting bit error rate (BER). The results also prove that the propagation losses are one of them influenced by the distance between transmitter and receiver. Multipath that varies in shallow water environment affects the system performance and DFE work. The sparse multipath distribution is more conducive than the dense multipath distribution in obtaining the superior system performance. LMS-based adaptive DFE performance can be increased in convergence rate with step size modification. By using 5×10^3 bits training symbol and a step size of 0.045 then the convergence of TR-DFE can be achieved. The MSE value and the convergence rate are used as adaptive DFE performance parameters. The optimal number of training symbols is also needed to get the better performance of TR-DFE system in underwater acoustic communication. By transmitting 10^6 information

bits, the superior system performance can be obtained when DFE uses 5×10^3 bits of training symbol.

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