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Comparison of adaptive algorithm on the equalizer for Underwater Acoustic Communication in shallow water

Ming Chuai†, Kyu-Chil Park, and Jong Rak Yoon
(Pukyong Nat'l Univ., KOREA))

1. Introduction

Adaptive filtering algorithm is one of the most active research topics in the current adaptive signal processing. Adaptive algorithm is widely used in system identification, echo cancellation, adaptive channel equalization in many fields. In shallow water, the underwater acoustic communication channel is a typical time- varying multipath fading channel After passing the channel transmission, receiving signal can be deemed to have a different path to reach, with the superposition of multiple components of different time delay and amplitude, it will cause variation in the amplitude, phase change and inter-symbol interference. [1-2] To compensate for this, several techniques have been used, and one of them is the underwater acoustic equalizer, in this paper, we choose two typical adaptive algorithms: least mean square algorithm and recursive least square algorithm (RLS) on finite impulse response filter (FIR). To compare the performance of recursive least square and least mean square algorithm in underwater communication. The performance comparisons were carried out on different situations like in different step size factor, the initial weight of case to comparison of their convergence speed, stability and so on. Through the derivation of the two algorithms, we can clearly understand the theoretical system of the adaptive algorithm. [3-5]

2. Equalizer Structure

Figure 1 shows the structure of a complex coefficient equalizer. FFE is feed forward equalizer and a finite impulse response filter, finite impulse response filter is a commonly used filter structure, most commonly used in the linear filter of equalization is a transversal filter. In FFE, x(n), u(n), y(n), z(n) and e(n) mean the original signal to be transmitted, the input signal after passing communication channel on receiver, the filter's output, the decision results from y(n), and the error signal between desired signal and the signal after through filter, respectively. $H_{tc}(z)$ is transfer function of the underwater acoustic communication channel base on channel response. And the adaptive algorithm is used to adjust the coefficient of equalizer in order to achieving the best filter effect.

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kcpark@pknu.ac.kr

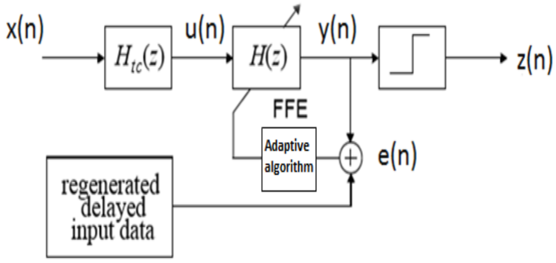

Fig. 1 The structure of feed forward equalizer

3. Experimental Conditions

Figure 2 shows the configuration of a sea experimental configuration, multipath intensity profile and its impulse response for the simulation. The specific experimental parameters are given in **Table I**. The range between the transmitter and the receiver is set to be 100 m, and the depths of the receiver and transmitter are set to be 7 and 10 m, respectively. We assumed that the channel response had only 5 multipath, namely, direct, bottom reflected, surface reflected, bottom-surface reflected, and surface-bottom reflected signals. The transmitted image is the standard Lenna image consisting of 35x35 pixels and 8 bits per pixel, which amounts to 9,800 bits of data.

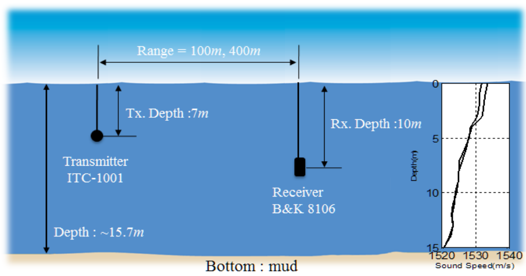

Fig. 2 Experimental configuration

Table I. Simulation and experimental parameters

Carrier frequency (kHz)	16 kHz
Sampling frequency (kHz)	128 kHz
Data Transmission Type	Packet
Tx and Rx range (m)	100
Tx and Rx depth (m)	7, 10
Depth (m)	~15.7
Bottom property	Mud
Data (bits)	Image 9,800 bits

4. Comparison of LMS and RLS algorithm with Numerical simulations and results

Least mean square algorithm was first proposed by B.Widrow and Hoff in 1959. [1] Its remarkable characteristic is simple, does not need to calculate the related function, also does not require the matrix operation and so on. The LMS algorithm is proposed based on the minimum mean square error criterion and the gradient descent method. **Figure 3** shows the results of LMS when weight is the same and at conditions of different step size. But the convergence rate of LMS algorithm is slow. In order to achieve fast convergence, the complex algorithm with additional parameters can be used.

RLS algorithm is a recursive least square algorithm, uses the known initial conditions to calculate, and using the information contained in the current input the new data to update the old filter parameters, thus, the data length is variable. RLS algorithm is based on the time carry out iteration. In other words the square of all the errors of the initial moment to the current time carry on average and make minimize. In addition, a weighting factor (the forgetting factor) is used to introduce into the error function. It can greatly improve the convergence properties of the adaptive equalizer. **Figure 4** shows the results of RLS when weight is the same and at conditions of different forgetting factors. In this section, we mainly discuss the application of LMS algorithm in adaptive equalizer.

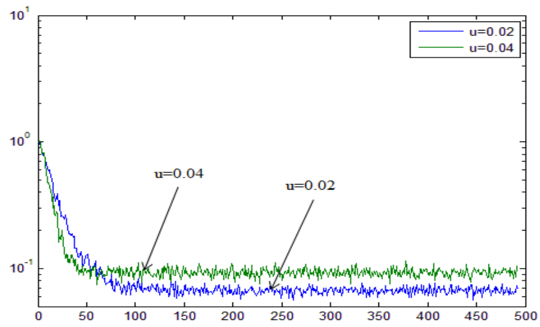

Fig. 3 results of different step size of LMS

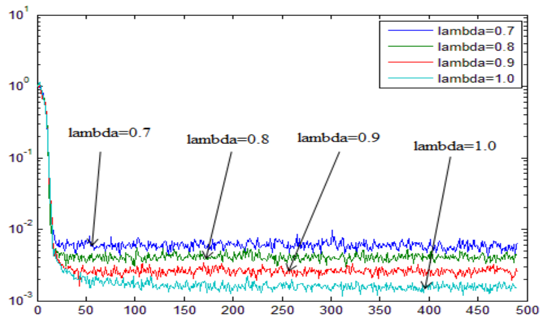

Fig. 4 different forgetting factors of RLS

Figure 5 shows comparison of convergence speed and stability of the two algorithms. From the results we can see RLS algorithm has better convergence speed and error rate. **Figure 6** shows a simulation on equalizer with RLS and LMS algorithm. The finite transversal filter will be used and adjust weight of tap using the two algorithm.

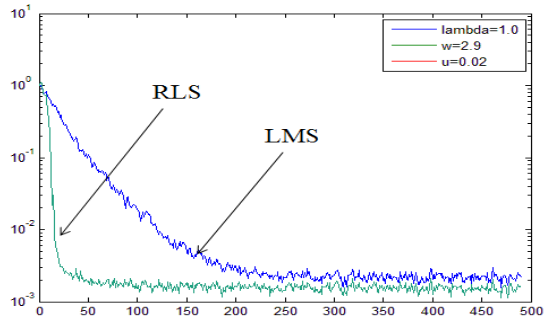

Fig. 5 Comparison between RLS and LMS

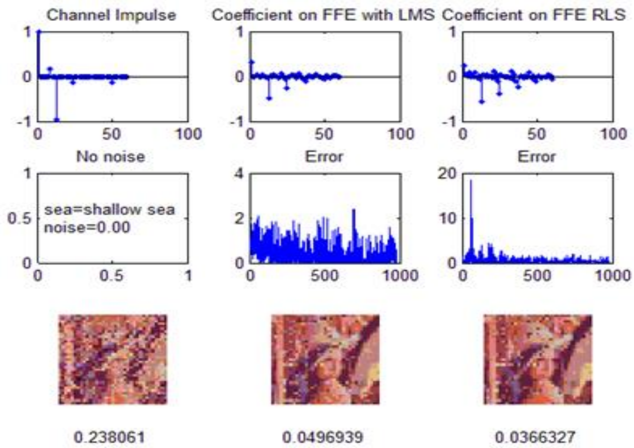

Fig.6 Performance comparison on FFE equalizer use the two algorithm

5. Conclusions

From the results we can see that the performance of the FFE equalizer with RLS is better than FFE with LMS. Because LMS is just a recursive method, input data and calculate output value through the difference between desired data and output data after equalizer. The RLS algorithm is based on the recursive algorithm. And use more complex coefficient called forgetting factor and RLS algorithm use current time to adjust the coefficient of the last moment

References

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