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# 천해환경에서 적응 알고리즘을 이용한 음향 등화기의 성능 비교

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# Performance Comparison of Acoustic Equalizers using Adaptive Algorithms in Shallow Water Condition

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## 요 약

천해 환경에서의 수중 음향 통신 채널은 전형적으로 시변 다중 경로 페이딩 채널 특성을 나타낸다. 이러한 채널 전송을 통해 수신된 신호는 시간 지연 및 진폭의 중첩에 의해 심볼 간 간섭을 유발한다. 이를 보완하기 위해 여러 기술이 사용되었으며, 그 중 하나가 음향 등화기이다. 본 연구에서는 심볼 간 간섭을 보상하기 위해 feed-forward equalizer (FFE), decision direct equalizer (DDE), decision feedback equalizer (DFE) 및 DFE와 결합된 DDE의 4 종류의 등화기와 등화기의 계수를 조정하기 위해 normalized least mean square (NLMS) 알고리즘과 recursive least square (RLS) 알고리즘의 2 종류의 알고리즘을 적용하였다. 그 결과 비선형 등화기에서는 신호 대 잡음비 6 dB 이상에서 상당한 성능 향상을 발견할 수 있었으며, DFE와 DDE의 조합은 어떤 경우에도 최고의 성능을 발휘하였다.

## ABSTRACT

The acoustic communication channel in shallow underwater is typically shown as time-varying multipath fading channel characteristics. The received signal through channel transmission cause inter-symbol interference (ISI) owing to multiple components of different time delay and amplitude. To compensate for this, several techniques have been used, and one of them is acoustic equalizer. In this study, we used four equalizers - feed forward equalizer (FFE), decision directed equalizer (DDE), decision feedback equalizer (DFE) and combination DDE with DFE to compensate ISI. And we applied two adaptive algorithms to adjust coefficient of equalizers - normalized least mean square algorithm and recursive least square algorithm. As result, we found that it has a significant performance improvement over 6 dB on SNR in nonlinear equalizer. By combination of DFE and DDE has almost best performance in any case.

**키워드** : 수중음향통신, 적응 알고리즘, 적응 등화기, BPSK 통신 시스템

Key word : Underwater acoustic communication, Adaptive algorithms, Adaptive equalizers, BPSK communication system

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## Ⅰ. Introduction

In the field of rapid development of the underwater acoustic communication system, such as shallow water, it is known to characterize a frequency selective fading channel [1-3]. The main problem of multipath channels is the inter-symbol interference (ISI). In order to improve the communication quality and reduce the ISI, the equalization technique is usually used at the receiving end [4, 5]. In this study, we tried to transmit the data signal using continuous transmission method with binary phase shift keying (BPSK) modulation and demodulation system in shallow water. And four acoustic equalizers are adopted that are the feed forward equalizer (FFE), the decision directed equalizer (DDE), the decision feedback equalizer (DFE) and combination of the DFE and the DDE with two adaptive algorithm with normalized least mean square (NLMS) algorithm and recursive least square (RLS) algorithm [6, 7]. Here the FFE and the DDE are linear equalizer with transversal structure. The DFE belongs to nonlinear equalizer and the coefficients of the equalizer are adjusted by feedback. The purpose is to reduce the loss of the data transmission for multipath channel and compare the performance of the four kinds of equalizers with two adaptive algorithms. As the same time, we also compared the convergence speed and stability of the two adaptive algorithms.

## Ⅱ. Configuration of simulations

Fig. 1 shows a configuration of a sea experimental condition and its sound velocity profile in a shallow channel located near to Busan, Korea. The experimental parameters are as following; the range between the transmitter and the receiver are respectively set to be 100 m and 400 m. The depths of the transmitter and the receiver are respectively set to be 7 m and 10 m. The water depth is set to be 14.7 m, 15.7 m, and 16.7 m respectively. The transmitted data is a standard Lena image with 35 × 35 pixels size consisting of 8 bits per pixel, which corresponds to 9800 bits in data. The sampling and carrier frequencies are respectively chosen as 128 kHz and 16 kHz. Fig. 2 shows channel impulse responses for the numerical simulation from this experimental condition. We assumed that the channel impulse response have 5 multipath signals – the direct, the bottom reflected, the surface reflected, the bottom surface reflected, and the surface bottom reflected signals.

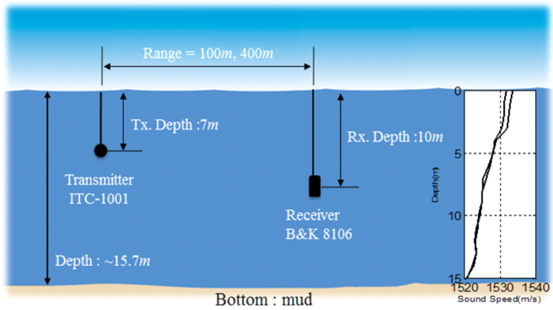



**Fig. 1** Configuration of a sea experimental.

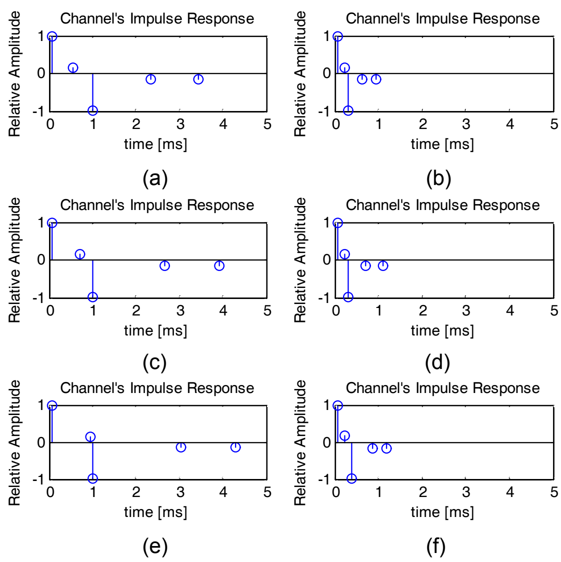



**Fig. 2** Channels' impulses responses according to distances and depths: (a) distance 100 m and depth 14.7m, (b) 400m and 14.7m, (c) 100 m and 15.7m, (d) 400m and 15.7m, (e) 100m and 16.7m, and (f) 400m and 16.7m.

## Ⅲ. Equalizers and adaptive algorithms

As the reflections from surface and bottom boundaries in shallow water acoustic communication, its channel's characteristics shows a frequency selective fading channel affected by a multipath delay spread. The performance of underwater acoustic communication degrades owing to ISI. To compensate this ISI effect, the equalizer based on the channel estimation is usually adopted.

Fig. 3 shows the structure of four acoustic equalizers - FFE, DDE, DFE and combination DFE with DDE - that are used in this paper. In all equalizers, $x[n]$, $u[n]$, $y[n]$, $z[n]$, and $e[n]$ mean the original signal to be transmitted, the input signal through the communication channel on the receiver, the filter's output, the decision results from $y[n]$, and the error signal between desired signal and the signal through the filter, respectively. $H_{tf}[z]$ is the transfer function of the underwater acoustic communication channel and $\eta[n]$ is the noise.

The underwater acoustic communication channels are based on Fig. 2's characteristics. The way of achieving errors on FFE is just one time training using a delayed version of the transmitted data. However, after training, the channel might change during data transmission, so adaptation should be continued. The DDE continues estimating the errors through the decision feedback after training [8, 9]. The reference signal should be equal to the recovered output data. If the signal is time-varying, adaptive filter adjusts the equalizer coefficients through the adaptive algorithm, making filter characteristics change with the change of signal and noise in order to achieve the optimum filtering effect [10]. The DFE is a decision feedback equalizer that uses the previous decisions when trying to estimate the present symbol with a symbol by symbol detector.

Next, we compare NLMS algorithm with the RLS algorithm from their iterative expression. The output $y[n]$ of input sequence $x[n]$ of tapped delay line equalizer is composed of a sum of discrete convolution defined as Eq. (1).

$$y[n]=\sum_{k=0}^{N} w_k x[n-k] \tag{1}$$

Here, $w_k$ is tap weight of the tap, the N+1 is the total number of taps. Then, the error signal is expressed by Eq. (2).

$$e[n]=d[n]-y[n] \tag{2}$$

Here, $d[n]$ is the desired signal, $e[n]$ is the difference between expected data and actual output. For equalizer with NLMS algorithm due to the existence of the error signal, we must constantly adjust the weights of the equalizer. So the update value of the tap weight can be expressed as Eq. (3).

$$w[n+1]=w[n]+2\frac{\mu}{x^T[n]x[n]+c}e[n]x[n] \tag{3}$$

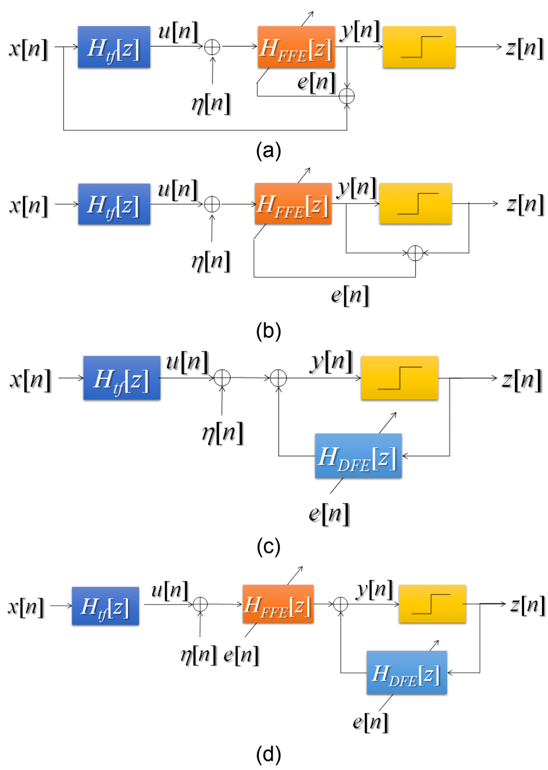



**Fig. 3** Schemes of four types of complex coefficient equalizers: (a) FFE, (b) DDE, (c) DFE, and (d) Combination of DFE with DDE.

Here, $\mu$ is a parameter for the step size, $c$ is a positive small constant denominator in order to avoid the denominator on the fraction is 0. The $x^T[n]$ matrix is called the transpose of $x[n]$. The 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[11]. In addition, a weighting factor (a forgetting factor) is used to introduce into the error function. It can greatly improve the convergence properties of the adaptive equalizer. The mean square error can be expressed by Eq. (4).

$$\epsilon[n] = \sum_{i=1}^{n} \beta[n, i] e^2[i] \tag{4}$$

Here, $\beta[n, i]$ is a weighting factor, also called the forgetting factor satisfies: $0 < \beta[n, i] \leq 1$, $n = 1, 2, 3, \ldots, k$. The most commonly used form of forgetting factor is the exponential weighting factor. It can be expressed by Eq. (5).

$$\beta[n, i] = \lambda^{n-i}, \; i = 1, 2, \ldots, k \tag{5}$$

Here, $\lambda$ is nearer to 1 but less than 1. The tap weight update can be given by Eqs. (6) – (7).

$$g[n] = \frac{p[n-1]x[n]}{\lambda + x^T[n]p[n-1]x[n]} \tag{6}$$

$$w[n] = w[n-1] + g[n]e[n] \tag{7}$$

Here, $g[n]$ is a gain vector, $e[n]$ express error signal, it is defined as the difference between the expected response $d[n]$ and the actual response $y[n]$ of the equalizer, $p[n]$ is the inverse matrix of the auto-correlation matrix.

We evaluated two types of adaptive algorithm that are the NLMS algorithm and the RLS algorithm, and two kinds of experiment method was adopted, one is training and transmission at water depths are 14.7 m, 15.7 m and 16.7 m, respectively and transmission distance is 100 m and 400 m. The other is transmission data is divided into data segments for transmission, it means that training at 15.7 m, actual transmission divided into three kinds of situations, the first one is to transmit half of the information at the 15.7 m water depth, the remaining half of the data at the 14.7 m water depth transmission. The second one is to transmit half of the information at the 15.7 m water depth, the remaining half of the data at the 16.7 m water depth transmission. The third one is to transmit a quarter of the data at the 15.7 m water depth, a quarter of the data at the 14.7 m water depth transmission, a quarter of the data at the 15.7 m water depth transmission, a quarter of the data at the 16.7 m water depth transmission.

## Ⅳ. Estimation of SNR from correlation functions

As we assumed that the channel's impulse response had 5 multipath signals - direct, bottom reflected, surface reflected, bottom surface reflected, and surface bottom reflected signal as shown in Fig. 2. From this condition, the origin signal to be transmitted $x[n]$ and the input signal through the communication channel on the receiver $u[n]$ can be represented by

$$u[n] = \sum_{k=0}^{L} A_k x[n - \tau_k] + \eta[n] \tag{8}$$

where $A_k$, $\tau_k$ , and $\eta$ are an amplitude, a delay, and the ambient noises of the kth received signals, respectively. The L is chosen by 4.

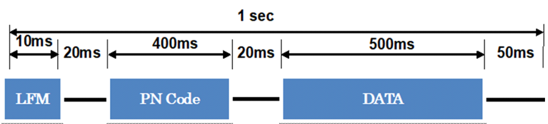



**Fig. 4** Frame structure for packet data transmission.

Fig. 4 shows the frame structure for packet data transmission. The transmission time of each frame was set to be 1 s. For measuring of the channel's characteristics and symbol timing alignment on the data transmission, a linear frequency modulation (LFM) and a pseudorandom noise (PN) code were used [12].

Fig. 5 shows (a) the LFM's time signal and (b) its frequency response as shown in Fig. 4. It has all frequency components between 13 kHz and 23 kHz.

The signal to noise ratio (SNR) estimation is mainly used for performance improvement in speech signal processing or wireless communication system [13-18]. As a method of estimation, there have been proposed a method using a correlation function [13, 14], communication channel data [15, 16], and learning-based model [17, 18].

To estimate SNR, we used correlation functions using the LFM signal. From Eq. (8), the cross-correlation between $x[n]$ and $u[n]$ can be represented by

$$\begin{aligned}\gamma_{xu}[k] &= \frac{1}{N+M}\sum_{k=0}^{N+M-1} x[n]u[n+k] \\ &= E\{x[n]u[n+k]\}\end{aligned} \tag{9}$$

where the N and the M is the length of each signal. E{·} is the expected value operator. Using the correlation function's property, the $\gamma_{xu}[k]$ can be represented by

$$\begin{aligned}&E\{x[n]u[n+k]\} \\ &\quad= E\{x[n](Ax[n+k]+\eta[n+k])\} \\ &\quad= E\{Ax[n]x[n+k]\}+E\{x[n]\eta[n+k])\} \\ &\quad= A\gamma_{xx[k]}+\gamma_{x\eta}[k] \\ &\quad\approx A\gamma_{xx[k]}\end{aligned} \tag{10}$$

where $\gamma_{xx}[k]$ and $\gamma_{xh}[k]$ are the auto-correlation of x $x[n]$, and the cross-correlation between $x[n]$ and $h[n]$, respectively. There is no relation between $x[n]$ and $h[n]$, the $\gamma_{xh}[k]$ would be considered as zero. From this equation, the amplitude of the signal could be estimated.

Next, to estimate the noise level, the auto-correlation of the $u[n]$ was calculated as following,

$$\begin{aligned}&E\{u[n]u[n+k]\} \\ &\quad= E\{(Ax[n]+\eta[n])(Ax[n+k]+\eta[n+k])\} \\ &\quad= A^2\gamma_{xx[k]}+2A\gamma_{x\eta}[k]+\gamma_{\eta\eta}[k] \\ &\quad\approx A^2\gamma_{xx[k]}+\gamma_{\eta\eta}[k]\end{aligned} \tag{11}$$

where $\gamma_{hh}[k]$ is the auto-correlation of noise $h[n]$, and it will be given by a delta function because the noise has not any relations with adjacent points. From Eqs. (10) and (11), the SNR can be estimated at each frame.

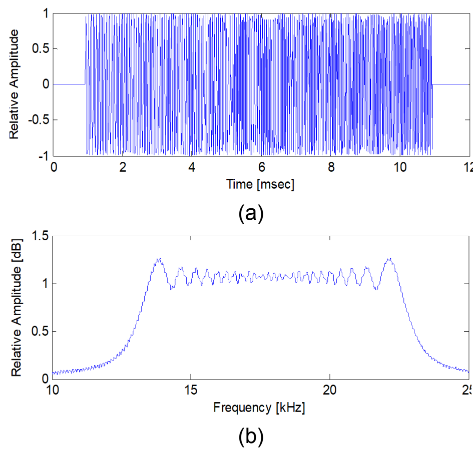
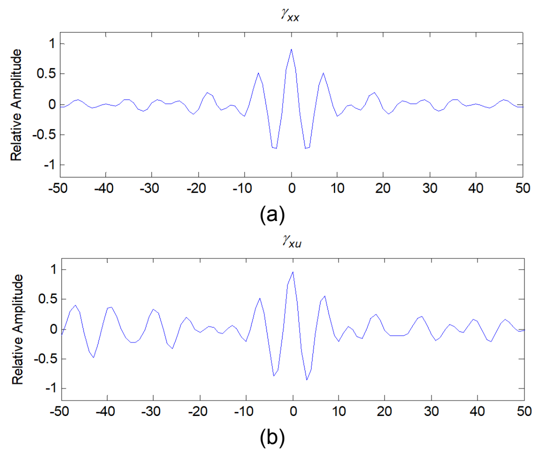
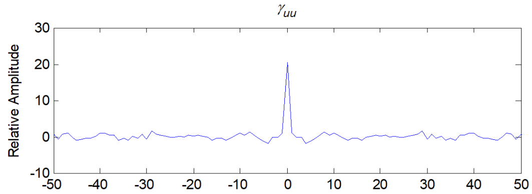



**Fig. 5** A LFM signal and its frequency response.

## Ⅴ. Simulation results

At first, we evaluated the SNR using Eqs. (9) - (11). Fig. 6 shows the results of the correlation functions $\gamma_{xx}[k]$, $\gamma_{xu}[k]$, and $\gamma_{uu}[k]$. The $\gamma_{xx}[k]$ is symmetry. The $\gamma_{xu}[k]$ and $\gamma_{uu}[k]$ are not symmetry owing the noise component. The $\gamma_{uu}[k]$ has a peak in center owing to a power of the noise. The $\gamma_{hh}[k]$ of the noise will be estimated by comparison $\gamma_{xx}[k]$ and $\gamma_{uu}[k]$ area with comparison using Eq. (10). Then $A^2$ will be estimated by subtract $\gamma_{hh}[k]$ from $\gamma_{uu}[k]$, and multiply by $\gamma_{xx}[k]$.



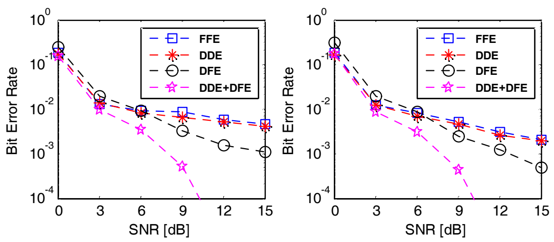



(c)

**Fig. 6** The results of the correlation functions, (a) gxx[k], (b) gxu[k], and (c) guu[k].

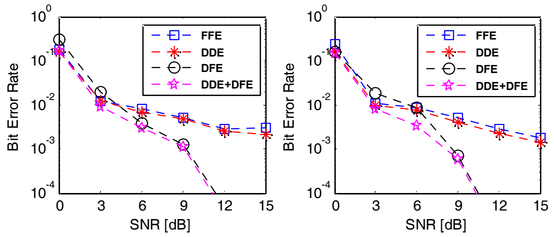
To check the accuracy, the amplitude of signal and noise were chosen by 12 and 2 that meant SNR was 12 dB. The estimated results were that the amplitude of signal and noise was 12.13(error rate: 1.07%) and 1.97(error rate: 2.90%), respectively. The SNR was estimated as about 12 dB with the 2.36% error rate.

Next, we compared four different kinds of equalizers in different underwater environments with two kinds of algorithms. The equalizers are a feed forward equalizer, a decision direct equalizer, a decision feedback equalizer, and a combining decision feedback equalizer with decision-directed equalizer, respectively. The depth of the underwater environment was chosen by 15.7 m and 16.7 m, respectively. And we also used the NLMS algorithm and the RLS algorithm for the adaptive algorithms, respectively. All simulations were performed with noise environment as given from 0 dB to 15 dB by 3 dB intervals.




(a) (b)

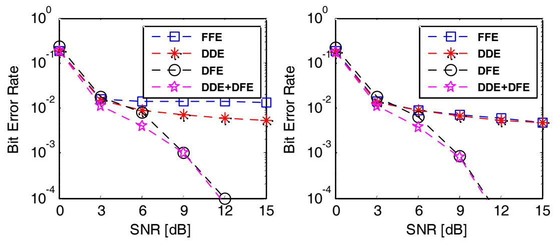
**Fig. 7** Simulation results by the NLMS algorithm at the transmission distance (a) 100 m and (b) 400 m with 15.7 m water depth.




(a) (b)

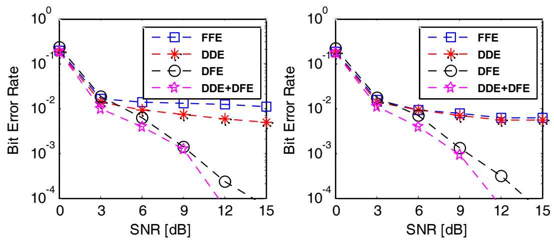
**Fig. 8** Simulation results by the RLS algorithm at the transmission distance (a) 100 m and (b) 400 m with 15.7 m water depth.

Fig. 7 shows the simulation results at transmission distance 100m and 400m, the water depth 15.7 m by NLMS algorithm, respectively. Fig. 8 shows the results at same situation with Fig. 4 by the RLS algorithm. From Figs. 7 and 8, we could found that RLS algorithm has a better performance than NLMS algorithm on same equalizers. And combination DFE with DDE has the best performance among all the equalizers.



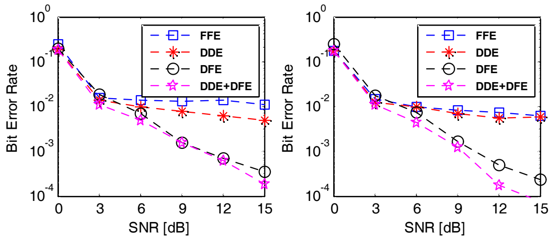

(a) (b)

**Fig. 9** Simulation results by (a) NLMS and (b) RLS algorithms, transmission distance at 100 m with depth changes from 15.7 m to 14.7 m.




(a) (b)

**Fig. 10** Simulation results by (a) NLMS and (b) RLS algorithms, transmission distance at 100 m with depth changes from 15.7 m to 16.7 m.




(a) (b)

**Fig. 11** Simulation results by (a) NLMS and (b) RLS algorithms, transmission distance at 100 m with depth changes from 15.7 m to 16.7 m, 15.7 m, and 14.7 m.

Next, we carried out the simulation with depth variations in the middle of transmitting signal to investigate the performance of environmental changes. The first environment is as follows. The first half is 15.7 m depth and the other half is 14.7 m depth. Fig. 9 shows the simulation results with transmission distance 100 m by NLMS and RLS algorithm. The second environment is as follows. The first half is 15.7 m depth and the other half is 16.7 m depth. Fig. 10 shows the results. The last environment is as follows. The first quarter is 15.7 m depth, then the others are 16.7 m depth, 15.7 m depth, and 14.7 m depth. Figs. 11 shows the simulation results.

## Ⅵ. Discussion and conclusion

Comparison the actual results with theoretical expectations, all the equalizers could effectively reduce the loss of the data in the transmission process. By combination of the linear equalizer with the nonlinear equalizer, it has the best performance from simulation results. Because linear equalizer can deal with precursor ISI and post-cursor, but DFE can only deal with post-cursor ISI. This is why when the SNR is relatively low and performance of DFE is poor.

From the results, we could see there are two distinct demarcation points. The performance of the DFE near 3dB is relatively poor below the 6 dB is also not significantly improved compared with linear equalizer. But over 6 dB has a significant performance improvement for nonlinear equalizer. For FFE and DDE comparison, with the increase of SNR, bit error rate is reduced in turn but performance of DDE is better than FFE. However, under any conditions, the combination of non-linear equalizer and linear equalizer has a significant improvement over the performance of the linear equalizer. Data transmission at a transmission distance of 400 m in the shallow water has a better transmission performance than at a transmission distance of 100 m under the water for signal transmission. The RLS algorithm is better than the NLMS algorithm in both stability and convergence. For data segments for transmission, the combination DFE with DDE also has the best performance and has the better transmission performance at nonlinear equalizers than all the data transmitted together.

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