Application of ARIS and Adaptive Filtering Algorithm in Underwater Acoustic Communication

Tianrui Zhang, Shuangshuang Han, and Ziyuan Yang

***Abstract*—As the core of marine communication, underwater acoustic communication technology significantly improves the performance of underwater communication by using technologies such as modulation, multiple access, and anti-interference. Among them, adaptive filtering technology plays a key role in eliminating noise and channel equalization. This paper explores the novel application of acoustic reconfigurable intelligence surface (ARIS) and adaptive filtering in underwater acoustic communication. We introduce the concept of ARIS assisted underwater acoustic communication, integrating the least mean squares (LMS) algorithm to dynamically estimate and mitigate oceanic noise. This approach further enhances signal clarity and communication quality by leveraging orthogonal frequency division multiplexing (OFDM) technology. Simulation results show that the underwater acoustic channel with ARIS is superior to the traditional channel without ARIS in terms of bit error rate and anti-interference performance. This technology significantly enhances the reception quality of underwater communication signals, making it suitable for applications in marine monitoring, military communication, and other complex environments.**

***Index Terms*—Underwater acoustic communication, adaptive filtering technology, acoustic reconfigurable intelligence surface, orthogonal frequency division multiplexing**

I. INTRODUCTION

With the development and utilization of marine resources, the demand for efficient and reliable communication in the ocean is increasingly urgent in the fields of marine environmental monitoring, marine scientific research, seabed exploration, marine military, etc. Due to the extremely serious attenuation of radio waves in water, the traditional radio communication technology cannot be effectively applied in the marine environment [1]. Therefore, underwater acoustic communication technology based on acoustic transmission has gradually become the mainstream choice of marine communication. Especially in deep sea and ocean environment, underwater acoustic communication has become a reliable communication mode, supporting the cooperative work and data transmission of marine equipment such as unmanned underwater vehicles, underwater sensor networks, and underwater robots [2–4]. These devices rely on efficient communication links to achieve real-time data transmission and remote control to ensure the smooth progress of marine observation and operations.

At present, the core technologies of underwater acoustic communication include modulation technology, multiple access technology, anti-multipath interference technology, and error control coding technology. Modulation technologies such as frequency shift keying (FSK) [5], phase shift keying (PSK) [6], and orthogonal frequency division multiplexing (OFDM) [7] are the key to achieving efficient data transmission. By optimizing spectrum utilization and improving transmission efficiency, the performance of underwater acoustic communication is significantly improved. Multiple access technologies such as time division multiple access (TDMA) and code division multiple access (CDMA) solve the communication problem in the multi-user environment, enabling multiple devices to work together efficiently in the same spectrum [8, 9]. The anti-multipath interference technology effectively copes with the multipath effect in the complex underwater environment by means of equalizer and adaptive filter, ensuring the integrity and stability of the communication signal [10, 11]. Huang et al. [12] proposed an optimized autoencoder method, which can better identify underwater signals by comparing the error between damaged features and target features. McCarthy et al. [13] proposed a prediction model based on the optimal receiving position of underwater acoustic channel using machine learning method, which is faster and more accurate than the traditional model. Li et al. [14] proposed to deploy acoustic devices in underwater vehicles. With the movement of underwater vehicles, they can adjust their pointing angle and improve the stability of underwater acoustic communication. From the perspective of energy allocation, Zhang et al. [15] proposed a model based on competitive game to balance the power allocation between sensor nodes and unmanned underwater vehicles, which improves the transmission efficiency of underwater acoustic signals. Zhu et al. [1] proposed a new type of underwater communication

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system, which can better transmit acoustic data. The sensor layer is composed of many types of sensors, and the fusion layer uses double cache technology, which provides a useful reference for designing communication system in complex environment.

However, the traditional filtering technology is difficult to deal with the complexity of underwater environment, and the application of adaptive filtering technology can maintain stable communication quality in dynamic underwater environment [16, 17]. Modern underwater acoustic communication system uses advanced adaptive algorithm to further improve the anti-interference ability and data transmission reliability. Xue et al. [18] proposed an improved method of Kalman filter, which can ignore the statistical information of noise and is more flexible when dealing with unknown system parameters. In underwater signal processing, Wang et al. [19] proposed a recursive Kalman filter algorithm, which can reduce the impact of noise on signal by estimating the signal envelope. Guo and Zhao [20] combined the adaptive filter with data association model, which can effectively improve the accuracy of data tracking and prevent data loss. Li et al. [21] proposed a distributed active noise suppression system based on the diffusion filtered least mean squares (DFLMS) algorithm, which overcomes the dependence of the traditional distributed algorithm on acoustic path symmetry by increasing the adaptation and combination strategies between nodes. Zhu et al. [22] proposed an enhanced sparse sensing recursive least squares (ESSRLS) algorithm, which improves the performance and robustness of the algorithm by introducing multiple optimization parameters, such as non-uniform norm, forgetting factor, etc. Bereyhi et al. [23] decomposed the error covariance matrix in RLS algorithm into two parts, processed them separately, and introduced the concept of exponential distribution. The improved algorithm has high convergence speed and stability. The above algorithms show good channel estimation performance in limited training time.

On the other hand, in addition to reducing the error of channel estimation through adaptive algorithm, the application of reconfigurable intelligence surface (RIS) technology in underwater acoustic communication has become a research hotspot in recent years. Its core idea is to use programmable surfaces to enhance or suppress signal transmission [24]. Zhou et al. [25] proposed to introduce RIS into the drone relay network and forward it to the destination through RIS. The entire communication system has high transmission rate and security. Girdher et al. [6] studied the application of RIS in the Internet of Vehicles, and optimized signal transmission by adjusting the reflection coefficient and phase offset. They also proved that this method can effectively reduce noise interference by Lyapunov central limit theorem. In millimeter wave communication systems as above, RIS will produce absorption or scattering effects from the surrounding environment in practice. Ma et al. [26] proposed a new channel estimation algorithm based on Bayesian algorithm. Compared with the traditional channel estimation method, it is more efficient. Zheng et al. [27] solved the problem of how to maximize the total system rate by jointly optimizing the beamforming matrix and RIS coefficient. Hossain et al. [28] proposed an underwater communication scheme based on RIS, which utilizes RIS to adjust the phase to maximize the received signal-to-noise ratio (SNR) and meet the underwater communication requirements under different conditions.

Acoustic reconfigurable intelligence surface (ARIS) can dynamically adjust the communication path and signal strength in complex underwater environment by intelligently regulating the phase and amplitude of sound waves, so as to enhance the communication quality and reliability [23]. The intelligent reflector can be deployed on the seabed, hull, or other underwater structures to reduce signal attenuation and interference by accurately controlling the reflection and refraction path of sound waves [29, 30]. The introduction of this technology is expected to significantly improve the performance of underwater acoustic communication system, solve the challenges that traditional communication methods are difficult to overcome, and further expand the application scenarios and coverage of underwater acoustic communication. Its typical application scenario is shown in Fig. 1. The underwater sensor transmits the collected data information to the shore based base station through ARIS.

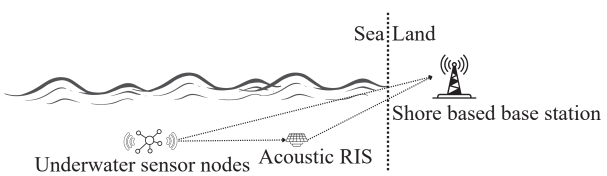

Figure 1 Underwater acoustic communication scene.

However, unlike traditional RIS, ARIS reflects sound waves, which is quite different from the principle of traditional RIS reflecting electromagnetic waves. Therefore, in order to successfully use RIS in the ocean, RIS needs to be redesigned. Wang et al. [31] designed three key components of ARIS for the above problems, and realized signal transmission by using the principle of acoustic reflection, avoiding some limitations encountered in traditional wireless communication. It is expected to combine with other technologies to form a more perfect communication system.

Acoustic communication technology is more and more widely used in the ocean, the introduction of adaptive filtering technology and ARIS has injected new vitality into the future development of underwater acoustic communication. On the existing basis, the contributions of this paper are as follows:

(1) At present, the traditional RIS is mainly used in the field of electromagnetic wave, and its application in underwater acoustic communication is still in exploratory stage. This paper introduces a new material, named as ARIS, which can intelligently control the phase and amplitude of sound wave. By dynamically adjusting the communication path and signal strength, the quality and reliability of underwater acoustic communication can be improved. This marks a good start for the application of ARIS technology in underwater environment.

(2) Although the above literatures discuss how to use multiple access technology and modulation technology to improve the performance of underwater acoustic communication, these methods often do not consider the noise suppression and channel equalization in dynamic environment. In this paper, a new concept of ARIS assisted underwater acoustic communication is proposed. At the same time, the LMS algorithm is used to dynamically estimate and suppress the unstable ocean noise, which improves the communication quality.

(3) This paper establishes a complete system model including OFDM signal generation, cyclic prefix (CP) addition, underwater acoustic channel transmission, ARIS auxiliary signal transmission, and LMS filtering. Through MATLAB simulation, it is proved that the underwater acoustic communication system assisted by ARIS is superior to the traditional system in bit error rate (BER), and successfully transmits a real audio signal, which verifies the effectiveness and feasibility of the underwater acoustic communication assisted by ARIS. By combining ARIS and LMS algorithm, it provides a new solution for underwater acoustic communication.

II. Analysis of Underwater Acoustic Communication

A. Signal Modulation Method

Underwater acoustic communication technology uses the characteristics of sound wave propagation in water to realize the transmission of information under water. Compared with other means of communication, sound wave has significant advantages in a wide range of marine environment because of its long transmission distance and small attenuation in water. Data collected by underwater sensor (including water temperature, salinity, dissolved oxygen content, etc.) are transmitted through acoustic communication. Initially, the analog data are transformed into digital signals via an analog to digital converter (ADC). Subsequently, digital signal processing (DSP) technology processes these signals, encompassing operations like filtering, sampling, and quantization, in preparation for modulation. DSP technology can optimize signal quality, reduce noise interference, and improve signal clarity and accuracy. In order to transmit in underwater environment, digital signals need to be modulated to sound waves. Common modulation technologies include PSK, FSK, quadrature amplitude modulation (QAM), and OFDM.

In this paper, we use OFDM technology to convert digital signal into acoustic signal, so that it can spread in underwater environment. OFDM is an efficient modulation technology, which divides the high-speed data stream into several low-speed substreams, and then these substreams are modulated to orthogonal subcarriers. In this way, the original data are converted into a series of OFDM symbols, and each symbol contains the modulation results of multiple subcarriers. This technology can improve spectral efficiency, combat multipath propagation and frequency selective fading, and provide better performance in underwater acoustic communication. At the same time, the underwater acoustic channel is usually multipath, that is, the signal will reach the receiver along multiple paths in the process of propagation, which will lead to the delay diffusion and frequency selective fading of the signal. OFDM can effectively deal with this multipath effect by allocating data on its subcarriers, and improve the reliability and rate of data transmission.

B. Application of Adaptive Filter

The application of adaptive filter in underwater acoustic communication is mainly used to overcome the influence of complex and changeable underwater environment on the communication signal. Common problems in underwater acoustic communication include multipath propagation, signal attenuation, noise interference, and so on. The adaptive filtering algorithm can dynamically adjust the filter parameters to minimize the error and optimize the receiving quality of signal. LMS algorithm is a commonly used adaptive filtering algorithm. Its basic idea is to adjust the filter coefficient by minimizing the error between received signal and desired signal. The structure of underwater acoustic channel combined with LMS algorithm is shown in Fig. 2. The original signal is modulated by OFDM to generate a time domain signal containing multiple subcarriers. The modulated signal is transmitted through the underwater acoustic channel, and the signal is affected by multipath effect and noise. The received signal is demodulated by OFDM, and the time domain signal is converted back to the frequency domain to extract the data symbols carried by each subcarrier. The demodulated signal is then adaptively filtered to remove noise and interference brought by channel and recover a cleaner signal. In LMS algorithm, the error $e(n)$ between received demodulated signal $y(n)$ and ideal original signal $d(n)$ is calculated with

$$e(n) = d(n) - y(n) \tag{1}$$

$$y(n) = w^{\mathrm{T}}(n)\,x(n) \tag{2}$$

where n stands for time index, $w(n)$ is the filter coefficient, and $x(n)$ is the input signal.

The filter coefficient is updated according to the error, which can be expressed as

$$w(n+1) = w(n) + 2\mu e(n)\,x(n) \tag{3}$$

where μ is the step size factor. The larger the μ is, the faster the convergence speed of the algorithm will be, but it may lead to instability of the algorithm. The smaller the μ is, the slower the convergence speed of the algorithm will be, but it can ensure that the convergence process is more stable.

C. Principle of RIS

RIS is well known in recent years by deploying reflective surfaces with adjustable characteristics in communication environment. These reflective surfaces are composed of a large number of low-cost passive reflective units, each of which can adjust the phase and amplitude independently. When the wireless signal reaches reflective surfaces, these surfaces can intelligently adjust the reflection path and reflection characteristics. Therefore, the signal can be

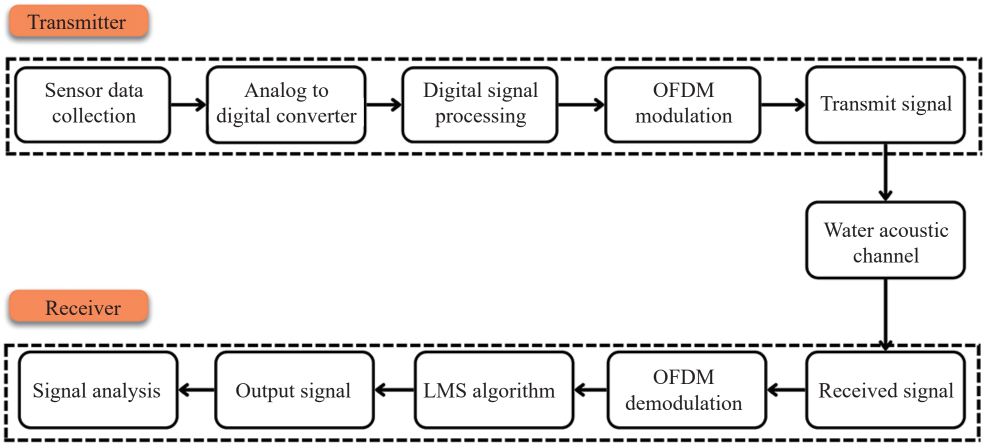

Figure 2 Structure diagram of underwater acoustic communication system.

enhanced or suppressed in the target direction, so as to optimize the path of signal propagation and improve the signal strength and coverage. Similar to the principle of RIS reflected signal, each piezoelectric reflector in ARIS has the ability of rapid response and immediate reflection. When the sound wave is incident on the piezoelectric reflectors, they will immediately generate corresponding vibration and reflect the sound wave. This reflection is automatic without external control or intervention. Due to its characteristics [23], ARIS can directly reflect the incident sound wave to the next destination node without intermediate digital data processing steps. At the same time, an important feature of ARIS is that its piezoelectric reflector and measuring circuit are independent. This means that the measuring circuit can independently measure the phase of the incident sound wave while the piezoelectric reflector reflects the sound wave. This independence enables ARIS to perform both reflection and measurement tasks at the same time, thus improving the functionality and flexibility of the system.

III. System Model and Signal Processing

Suppose there are K subcarriers in the OFDM system, the signal period is T, the interval length of each subcarrier is set to $1/T$, and its center frequency is f_c. The data collected by the sensor are converted into a binary sequence, and specific pilot tones are inserted at the same time. These pilots are equally spaced between data subcarriers. Each subcarrier uses 16QAM to map the above data sequence. Define $S(k)$ as the coding information on the k-th subcarrier, where the frequency of each subcarrier is

$$f_k = f_c + \frac{k}{T} \tag{4}$$

where $k = -\frac{K}{2}, -\frac{K}{2}+1, \ldots, \frac{K}{2}-1$. With the help of inverse fast Fourier transform (IFFT) method, the results of spectral analysis are converted back to time series data, that is

$$S(n) = \text{IFFT}\{S(k)\} = \frac{1}{K}\sum_{k=0}^{K-1} S(k)e^{j2\pi n f_k} \tag{5}$$

After IFFT, we add a CP after the OFDM symbol, which helps to resist inter symbol interference (ISI) caused by multipath effect. It can be expressed as

$$S_{cp}(n) = \begin{cases} S(K+n), \text{ for } n = -N_{cp}, -N_{cp}+1, \ldots, -2, -1; \\ S(n), \text{ for } n = 0, 1, 2, \ldots, K-2, K-1 \end{cases} \tag{6}$$

where N_{cp} represents the number of cp.

The generation method of CP is to copy the last N_{cp} sampling points of OFDM symbol, and then add these copy points to the beginning of the OFDM symbol as the CP. Therefore, the length of the modulation signal $S_{cp}(n)$ is $K+N_{cp}$, and the structure of each OFDM block is shown in Fig. 3, in which T_{cp} represents time length of CP, T_b represents length of 64 subcarriers, and T_s is the total length of OFDM blocks, respectively. Each symbol is composed of 64 subcarriers, and a CP with a length of 16 is added in front of each symbol. There are 48 subcarriers in each OFDM symbol for data transmission. The data on these subcarriers are represented by 16QAM. Each subcarrier can carry 4 bits of information.

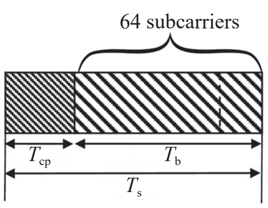

Figure 3 OFDM block structure diagram.

Then $S_{cp}(n)$ will be transmitted to the shore based base station through underwater acoustic channel with ARIS deployed. Ideally, for ARIS, by using the pilot signal of transmitter and the feedback signal of receiver, the angle of arrival (AOA) from the transmitted signal to the ARIS and the angle of departure (AOD) reflected by the ARIS can be estimated respectively. Once the angle information is determined, the ARIS can perform beamforming by adjusting the phase distribution of its reflector. Specifically, the ARIS can form a strong beam in the desired direction by allocating an appropriate phase offset to each reflecting unit. However, the research on ARIS is still in the theoretical stage, and there is no large-scale practical application case. Therefore, this paper temporarily regards ARIS as the relay equipment to strengthen the forwarding signal, and the modeling of time-varying underwater acoustic channel is not the focus of this paper, using consistent Doppler instead of non-uniform Doppler in underwater acoustic communication. After the introduction of ARIS, the channel model still needs to consider the additional path generated by the reflector. The amplitude, phase, and delay of these paths are controlled by the intelligent reflector. Therefore, the pulse response of underwater acoustic channel can be expressed as

$$h(n) = \sum_{i=1}^{s+d} g_i \delta(n-\tau_i) \tag{7}$$

where s and d represent numbers of additional and original multipaths introduced by ARIS, g_i and τ_i represent the gain and delay of the i-th path introduced by ARIS, and δ represents the impulse response, respectively. Therefore, the signal transmitted through the underwater acoustic channel can be expressed as

$$y_{\mathrm{cp}}(n) = S_{\mathrm{cp}}(n) \otimes h(n) + \xi_n,\ -N_{\mathrm{cp}} \leqslant n \leqslant K-1 \tag{8}$$

where $\otimes$ indicates convolution, ξ_n is additive white Gaussian noise (AWGN) and follows the Gaussian distribution, i.e., $\xi_n \sim \mathcal{CN}(0, \sigma_{\mathrm{n}}^2)$, where σ_{n}^2 is noise power.

At the receiving end, that is, the base station, the CP is removed to obtain the signal $y(n)$, and the discrete Fourier transform (DFT) is performed on it to convert the continuous domain signal into the discrete domain, and the following is obtained

$$Y(k) = H(k)S(k) + \xi(k) \tag{9}$$

where $Y(k)$, $H(k)$, and $\xi(k)$ represent the received signal (in the following LMS algorithm, it also represents the input signal of the filter), transmitted signal, and additive noise on the k-th carrier, respectively. At this time, the signal-to-noise ratio can be expressed as

$$\gamma(k) = \frac{|H(k)|^2 \sigma_{\mathrm{s}}^2}{\sigma_{\mathrm{n}}^2} \tag{10}$$

where σ_{s}^2 and σ_{n}^2 represent signal energy and noise energy, respectively.

The demodulated signal is applied to the LMS algorithm through an adaptive filter to calculate the error between the output signal of the filter $X(k)$ and the ideal original signal $S(k)$

$$e(k) = S(k) - X(k) \tag{11}$$

where $X(k) = W^{\mathrm{T}}(k)Y(k)$, $W(k)$ is filter coefficient and $Y(k)$ is the input signal of the filter.

Update the filter coefficient according to the error, and the equation can be rewritten as

$$W(k+1) = W(k) + 2\mu e(k)X(k) \tag{12}$$

To ensure the steady-state convergence of the algorithm, μ should be taken in the following range

$$0 < \mu < \frac{2}{\sum_{k=1}^{K} x(k)^2} \tag{13}$$

IV. Simulation and Analysis

In this experiment, a complete simulation process of OFDM communication system is implemented in MATLAB R2018a version, including signal coding, transmission through multipath channel, signal decoding, and performance evaluation.

First, a group of QAM data are randomly generated as the information to be transmitted. These data simulate the environmental information transmitted in the underwater acoustic communication channel. After inserting pilot symbols, performing IFFT transform, and adding cyclic prefix, they enter the underwater acoustic channel for transmission. This process simulates a simple multipath propagation environment with static fading characteristics, defines the different delays and powers of multiple paths, and simulates the delay spread and frequency selective fading experienced by the signal when passing through the complex channel.

At the same time, noises with different SNRs are added to the signal after multipath channel to simulate the noise interference in actual wireless communication. At the receiving end, the received signal affected by noise is decoded to try to recover the original constellation symbol. This process includes removing the cyclic prefix, performing FFT transform, and estimating and compensating the channel influence. Finally, the difference ratio between the received signal and the original transmitted signal is calculated to evaluate the transmission quality of the system. Several simulations are carried out for different SNR values, and two BER comparison curves are obtained, corresponding to the communication methods of underwater acoustic channel with and without ARIS deployment, as shown in Fig. 4.

It is obvious that the underwater acoustic channel with ARIS has significantly better performance in BER than that without ARIS. The introduction of ARIS into the system results in a notable reduction in BER across different SNR levels. This demonstrates the effectiveness of ARIS in enhancing the robustness of the communication link. This improvement is primarily due to the ability of ARIS to mitigate the effects of multipath propagation, which is a significant challenge in the complex underwater acoustic

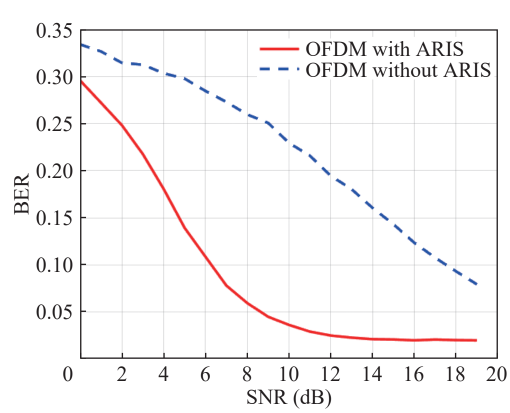

Figure 4 Performance comparison of communication systems with and without ARIS under different SNRs.

channel. Multipath propagation leads to signal fading and interference, which can severely degrade the communication quality. ARIS can effectively reduce the multipath effect, thereby reducing bit errors and improving the overall communication quality.

Moreover, the combination of ARIS with adaptive filtering techniques further improves the system's ability to handle the challenges posed by the underwater environment. The use of adaptive filters, such as the LMS algorithm, enables the system to dynamically adjust its filter coefficients based on the received signal. This results in a more precise estimation and cancellation of the noise components, contributing to the overall improvement in BER. At the same time, in another experiment, we send a real audio signal into the above channel by OFDM modulation. After demodulating the signal, we implement the LMS algorithm and observe the difference between the output audio signal waveform and the initial audio signal waveform. The filter order is set to 128 and the filter step size is set to 0.004. The results are shown in Fig. 5, in which the four subgraphs represent the original signal, noisy signal, denoised signal, and residual noise, respectively. After the initial audio signal is modulated by OFDM and transmitted through underwater acoustic channel, multipath propagation and noise will be introduced. After receiving the noisy signal, the receiver demodulates it by OFDM, trying to recover the original signal, and finally uses LMS algorithm to reduce the noise in the demodulated signal. Through the visual analysis of signal, the experimental results show that the underwater environment has a significant impact on signal transmission quality, and the LMS filter can effectively reduce the noise level and improve the signal definition. At the same time, this also proves the feasibility of the adaptive filtering algorithm in the deployment of ARIS underwater acoustic channel communication.

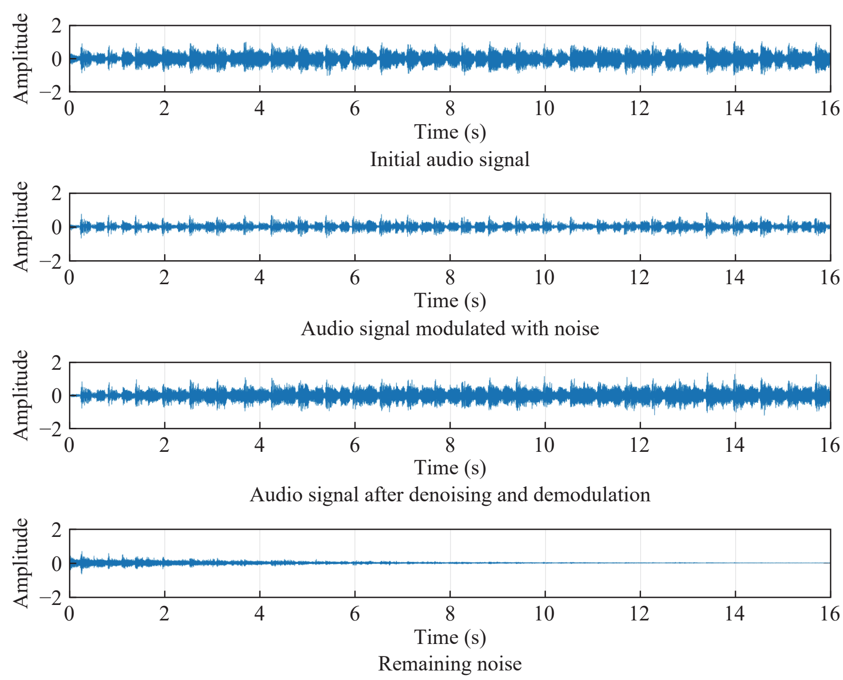

Figure 5 Waveform of original signal, noisy signal, denoised signal, and residual noise.

V. CONCLUSION

In this paper, the ARIS is introduced into the underwater acoustic channel. Through its gain effect on the channel, the stability of signal transmission is enhanced. At the same time, the LMS algorithm is combined to dynamically estimate and reduce the noise and interference in the underwater environment. Compared with the traditional underwater acoustic channel, the simulation results show that the underwater acoustic channel with ARIS achieves lower BER and better anti-interference performance. The ARIS is equivalent to providing additional control means for the communication system, which can be regarded as a channel optimization method, while adaptive filtering further improves the accuracy of signal processing. The proposed method improves the robustness of the system and reduces the risk of rising BER due to multipath effect and noise interference. However, the implementation of ARIS technology depends on complex hardware design and high-precision control algorithm, which has high deployment cost and may be restricted by environmental factors in practical application. At the same time, the convergence speed and robustness of LMS algorithm may be challenged in the face of complex and changeable underwater environment, especially in the case of large changes in noise environment. Despite the above challenges, ARIS and adaptive filtering technology still show great potential, and are expected to be more widely used in the fields of marine monitoring, military communications, and remote marine communications in the future. Future work will be devoted to solving the above problems and further improving the performance and practicability of ARIS assisted underwater acoustic communication system.

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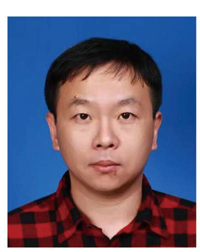

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