A Research on Underwater Acoustic Channel Modeling and Simulation of Shallow Sea

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Abstract. In order to research the shallow underwater communication system further and conveniently, the paper study the characteristics of underwater acoustic propagation and choose "Ray Model" as the simulation. Then we describe this model from the point of geometry and analyze some parameters of shallow underwater acoustic communication. According to these factors, the received signal delay and loss can be calculated and the received signal can also be expressed. This paper uses the technique of adaptive equalization to solve the serious Inter symbol interference (ISI) from the multi-path signals by giving principles and formulas of the least mean square (LMS) error algorithm. Finally, the shallow underwater acoustic channel model can be simulated by MATLAB in 2ASK modulation. In the results of simulation, the severe ISI can be observed in the received signal and can be eliminated by equalization. We also get the bit error rate curve successfully. This simulation can provide the foundation for other underwater acoustic communication researches and works.

Keywords: Shallow underwater acoustic channels · Channel modeling · Ray model · Adaptive equalization

1 Introduction

With the continuous development of science and technology, together with the increase of world's population, scientific researchers in the major research institutions and universities are increasingly concerned about the ocean, known as a valuable resource to be explored. To realize marine survey, resource exploitation and utilization is one of the most concerned problems in coastal countries. By constructing shallow underwater communication system, we can monitor and gather military intelligence, along with detect conditions of ports and coast to realize group management. In addition, the construction can also provide a large amount of data for the marine environment, such as the sea temperature, salinity, and so on. In conclusion, the high-speed reliable shallow underwater acoustic communication technology has become a heated research direction in the field of underwater acoustic communication. However, the underwater acoustic channel is an extremely complex channel.

Because the electromagnetic wave attenuation is serious in seawater, low frequency acoustic wave is the most widespread shallow sea communication technology.

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However, this has caused the limitation of bandwidth of the shallow water communication. Moreover, the low propagation velocity of the 1500 m/s and the multipath effects caused by the shallow water propagation are inevitable. Worse physical properties are imposed on shallow water acoustic channels, due to the Doppler frequency shift generated by transceiver movement together with the random Doppler frequency shift generated by the sea movement. In order to facilitate the further study of the shallow seawater communication system, the modeling of the underwater communication channel is required [1]. Since the study of underwater acoustic communication, several underwater sound propagation model were promoted through different approximation and ameliorate method, which are ray model, normal mode model, parabolic equation model, fast field model and so on.

Description of the sound ray model is by the sound line to transmit energy, from the sound source of the sound line to follow a certain path to receiver, the received acoustic energy is all reach the superposition of voice. The ray model can describe the transmission path between the transmitter and the receiver, and gives the propagation loss and the time of each path.

Therefore, when analyzing and simulating the propagation characteristics of the multipath channel, the ray model is a common tool for analysis. In this paper, the basic characteristics of acoustic wave propagation in shallow water environment are studied. Based on the wave propagation, the most commonly used simulation model of underwater acoustic channel ray model is applied. In the end, the reasonable equalization technique is adopted to eliminate the multipath interference, and realize the transmission of the shallow water acoustic signal.

2 The Analysis of Underwater Acoustic Propagation Characters

This article will analyze the characters of underwater acoustic propagation from the following aspects:

(1) The propagation velocity of the sound waves in the sea: The propagation velocity of the sound waves has a great effect on the time we need to get the information from the receivers, which may influence the signal we receive. Therefore, this coefficient is of great importance. The average velocity of the sound in the sea is 1500 m/s. In the acoustic researches, conclusions have been drawn that the propagation velocity can be affected by the temperature, salinity and static pressure. Usually we use empirical formulas to demonstrate the relationship of the coefficients because of the complicated influence of environment. If accuracy is not required to be very high, Ude empirical formula can be used:

$$c = 1450 + 4.21T - 0.037T^2 + 1.14(S - 35) + 0.175P \tag{1}$$

where T denotes temperature; S is salinity, P is static pressure.

(2) Sea surface: The sound waves can be reflected and scattered on the surface of sea. The wind and sea waves make the reflection various. Because of the regularity and the randomness of the waves on the surface, the researches of the waves' statistical characters will help us to analyze the surface's influence on the propagation. The reflection coefficients can be calculated in this way. The Bechmann-Spezzichino model, put forward by Coates, makes it possible to compute the reflection coefficients by the velocity of the wind and other variants. In addition, this model can be used perfectly in shallow sea underwater acoustic channel. Concrete details about calculation will be discussed in Sect. 3.2.
(3) Seabed: Different topographic texture and diverse sediments can influence the reflection coefficients and reflection loss. It affects the distance of the propagation in the sea. As the research show, the reflection coefficients are highly related to the seabed topographic texture. When it comes to shallow sea, the smooth silt and sand on the bottom provide little reflection loss. Combined with the other factors of the shallow sea, the grazing angle ψ is small on the reflecting surface. Based on the NUSC model, when ψ is smaller than 5°, the frequency of sound signal is less than 50 kHz and the osmotic coefficient is smaller than 0.5 (fine sands, silts satisfy the condition), the reflection coefficient is close to 1, which means no loss.
(4) Noise: The noise of the sea environment has a great effect on the propagation of the signal. Severe noise may disturb the receivers' demodulation and decision. Usually, we take these sea noises into consideration: the noise from winds and waves, live beings, raindrops and environmental thermal noise. However, these factors are time-variable and also space-variable. Hence, we can only attain rough spectrum of the shallow sea noises. So the noises can be regarded as White Gaussian Noise when having theoretical analysis.
(5) Doppler shift: There are two kinds of Doppler shift. One is based on the relative motion of the receiver. Assumed the velocity of the transmitter is v_s, the velocity of the receiver is v_r, the frequency of the signal is f_s, the acoustic speed is c, and then the frequency of received signal under the effect of Doppler shift is $f_r = f_s(c - v_r)/(c - v_s)$.

Due to the velocity of the sound is far from that of the electromagnetic wave, and the frequency of the sound is too low, the Doppler shift can be severe.

Another situation is that the receiver does not have relative motion, and the Doppler Shift is caused by the waves on the surface and the current. This kind of Doppler shift may cause frequency expansion, which may lead to the time selective fading of the shallow water acoustic channels.

Doppler shift will cause great influence on the correct demodulation of the received signal. Hence, Doppler compensation is needed. If only frequency shift is considered, the compensation of the shift can be compensated easily. However, if frequency expansion occurs and it is frequency modulation, the interspace of the frequencies is required. When it comes to theoretical analysis, the Doppler frequency shift can be regarded as a compensated factor, and has no effect on the received signal.

3 Model Selection for Underwater Acoustic Channel

Received data from underwater sensor shows that the received signal of the underwater acoustic channel is a multipath signal, which is the superposition of the reflection of transmitted signal on the sea surface and seabed. Because of the small grazing angle and small boundary reflection loss caused by the shallow water environment, the received signal is a multipath signal. This has led to the amplitude of a number of non-diameter signals at the receiver are no less than the direct path, thus these signals cannot be ignored. In order to be able to consider the influence of environmental parameters and reflect multipath, "Ray Model" is used to describe and analyze the underwater acoustic channel. The impulse response of the channel can be attained based on the reflection and attenuation of multipath signals, which is convenient for the subsequent study of the received signal.

3.1 Description of the Ray Model

This model mainly analyzes more in the aspects of mathematics and physics, which is described by the method of geometry. In this way, it is more easily to reflect the propagation of rays, at the same time, more helpful for the further calculation and understanding of the parameters of the model. The model diagram is shown in Fig. 1.

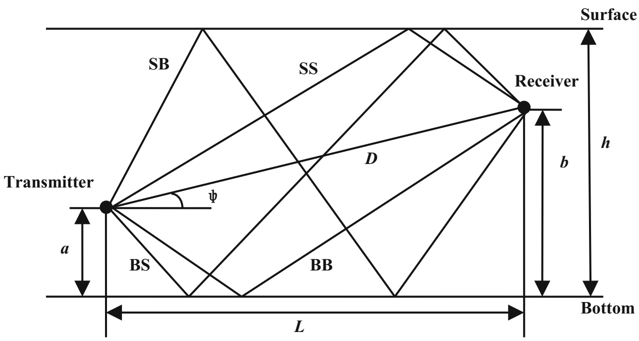

Fig. 1. Underwater ray model of shallow sea

In Fig. 1, a denotes the distance between transmitter and the bottom of the sea; b is the distance between receiver and the bottom of the sea; L stands for the horizontal distance between the transmitter and receiver; ψ is glancing angle (grazing angle).

The propagation paths can be divided into direct path D and multipath. Multipath signals can be divided into four categories according to the type of reflection: symbol SS denotes the signals which before reaching the receiver the first and last reflection is happened on the sea surface. Index is used to indicate the sequence of the signal. SB, BS, and BB are in the same way. These four types of signals are represented in Fig. 1 when $n = 1$. In order to facilitate the calculation, we assume that A < B.

In the shallow seawater acoustic environment ($h \ll L$), grazing angle ψ is very small. When the grazing angle ψ is less than the "boundary grazing angle (total internal reflection angle)", the boundary reflection can be regarded as specular reflection. This is a very important condition for the calculation of the model.

Each path propagation length can be calculated as $D \simeq L + (b-a)^2 \big/ 2L$, $SS_n \simeq L + (2nh - a - b)^2 \big/ 2L$, $SB_n \simeq L + (2nh - a + b)^2 \big/ 2L$, $BS_n \simeq L + (2nh + a - b)^2 \big/ 2L$, and $BB_n \simeq L + [2(n-1)h + a + b]^2 \big/ 2L$.

3.2 The Signal Loss

(1) Extension loss (spread loss): The extension loss is due to the amplitude (energy) attenuation caused by the propagation of sound waves in the sea. Generally speaking, the propagation loss caused by the extension can be indicated as $TL = n * 10 \lg r (\mathrm{dB})$.

According to different propagation conditions, n may take different values: n = 0: plane wave propagation, no expansion loss; n = 1: cylindrical wave propagation, the wave front is expanded according to the law of the cylinder, which is equivalent to the propagation condition of ideal waveguide, which is consisted of the total reflection sea surface and total reflection sea bottom; n = 2: spherical wave propagation, the wave front is extended by spherical surface.

In this study, we assume the acoustic wave is transmitted from the transmitter and expand in the sea in the form of spherical wave expansion, that is n = 2. Propagation loss (energy loss) is proportional to the square of the propagation length. Along the propagation direction of harmonic radius r of spherical wave, the pressure can be expressed as $p = \frac{p_0}{r} \exp[-i(wt - kr)]$, in which p_0/r is the sound pressure amplitude of the spherical wave, which is inversely proportional to the distance r.

In conclusion, as long as the propagation length of each path is known, the extension loss of this path can be expressed by the reciprocal of the length of the transmission.

(2) Absorption attenuation: The absorption attenuation is the sound intensity decrease caused by the absorption surface, which is related to the distance. Generally, $\alpha$is used as a representation of the absorption coefficient, which unit is dB/m. In seawater, absorption attenuation and the loss caused by scattering presents at the same time, and it is difficult to separate them when measuring. However, when considering the absorption of homogeneous medium, it is found that the value of absorption coefficient of a majority of the liquid will be far greater than its theoretical value. We call this difference super absorption. As a result, we cannot use replace absorption attenuation with the theoretical value.

The structure relaxation theory proposed by Hall has explained the super absorption: in seawater, the propagation of acoustic waves is dissolved in the $MgSO_4$ of salt water. Because of its low solubility, the equilibrium of the original dissolution is destroyed, and a new equilibrium is reached. This process absorbs the energy of sound waves.

After a large amount of data testing, the empirical formula of the absorption coefficient [2] is $\alpha(f) = \frac{0.102f^2}{1+f^2} + \frac{40.7f^2}{4100+f^2}(\mathrm{dB/km})$.

(3) Reflection loss: Sea Surface: The reflection coefficient is dependent on wind speed. The empirical formula [3] has been given in the Bechmann-Spezzichino model. The application condition of the model is also a low grazing angle of the shallow sea underwater acoustic channel, thus this formula can be applied to this study. Empirical formula is $|r_s| = \sqrt{\left[1+(f/f_1)^2\right]\Big/\left[1+(f/f_2)^2\right]}$, where $f_2 = 378\omega^{-2}; f_1 = \sqrt{10}f_2$.

In the above formula, the working frequency of the acoustic signal is f, and its unit is kHz; wind speed is expressed as ω, and its unit is knots. 1knots = 1.852 km/h = 0.514 m/s.

Sea Bottom: In order to simplify the model parameters, in the following calculation and simulation, it is assumed that the seabed reflection coefficient $|r_b| = 0.9$.

Considering each of the reflection of the sea surface and sea bottom is a specular reflection, as a result a 180° phase shift will be produced, so $r_s = -|r_s|$; $r_b = -|r_b|$. On the basis of the multipath classification, the reflection coefficient of each multipath signal in the sea surface and the sea bottom is different, and the reflection coefficient of each path can be obtained: $R_{SS_n} = r_s^n r_b^{n-1} = -r_s^n * 0.9^{n-1}$, $R_{SB_n} = r_s^n r_b^n = r_s^n * 0.9^n$, $R_{BS_n} = r_s^n r_b^n = r_s^n * 0.9^n$, where $n = 1, \ldots, \infty$.

3.3 The Signal Propagation Time Delay

Signal time delay is another important parameter in the received signal impulse response. According to the previous analysis, the condition of specular reflection can help to calculate the propagation length of each path. The propagation sound velocity in seawater can be regarded as constant c = 1500 m/s, so the propagation delay of each path can be expressed as: $t_D = D/c$, $t_{SS_n} = SS_n/c$, $t_{BS_n} = BS_n/c$, $t_{SB_n} = SB_n/c$, $t_{BB_n} = BB_n/c$.

3.4 The Joint Response of Received Signal

The multipath signals received by each path can be expressed by the sum of the sum of each path:

$$r(t) = \alpha \frac{e^{jw(t-t_D)}}{D} + \alpha \sum_{n=1}^{\infty} \left[\frac{R_{SS_n}}{SS_n} e^{jw(t-t_{SS_n})} + \frac{R_{SB_n}}{SB_n} e^{jw(t-t_{SB_n})} + \frac{R_{BS_n}}{BS_n} e^{jw(t-t_{BS_n})} + \frac{R_{BB_n}}{BB_n} e^{jw(t-t_{BB_n})} \right] \tag{2}$$

Because there are numerous multipath signals may be represented in this way, in this study, in order to limit the number of multipath, the multipath signal which amplitude is less than 1% of the signal of the direct path will be omitted. In this way, finite multipath signals are used to represent the received signal of a shallow sea underwater acoustic channel.

4 Underwater Acoustic Channels Equalization of Shallow Sea

4.1 The Work Mode and Structure of Adaptive Equalizer

Adaptive equalizer has the following two work modes [4–6]:

(1) Training Mode: The transmitter sends a fixed length set of training sequences (the typical training sequence is a binary pseudo-random sequence or a set of advance specified data). The equalizer makes compensation for the channel through recursive algorithm, which evaluates the channel characteristics, and constantly revises the filter coefficients. The filter coefficients tend to the optimal value after continuous recursive iterative process [7, 8]. (Recursive algorithm will be described in detail later.)
(2) Tracking Mode: After the most optimal filter coefficients are known, in order to eliminate ISI, we equalize the follow-up received signal.

According to the previous analysis, the environmental parameters of the underwater acoustic channel change slowly, that is, the channel is slowly time varying. So the channel characteristics can be regarded as the same for a time. Therefore, assume in this period, after continuous recursive iteration, the optimal filter coefficients of adaptive equalizer in the training mode can effectively eliminate inter symbol interference of multipath signal. For the next period, the adaptive equalizer will enter training mode again, which ensures the time varying characteristics of adaptive equalizer [9, 10].

The structure of the adaptive filter is shown in Fig. 2.

Figure 2 is a simple form of an adaptive filter: a linear transversal equalizer, in which there are a total of N tap coefficients. $y(n) = \sum_{i=0}^{N-1} w_i(n)x(n-i)$ is the equalized signal, which is obtained by the linear superposition of the current signal and the delay of the signal.

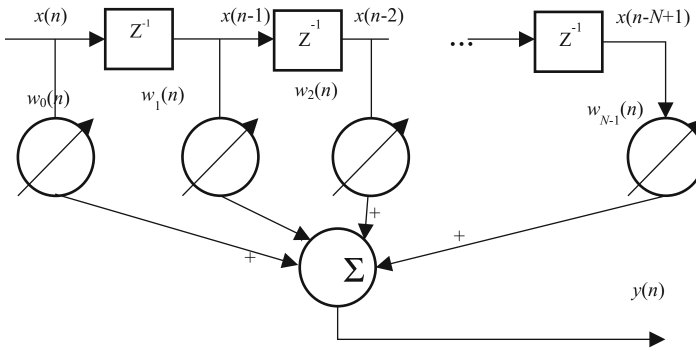

Fig. 2. The structure of adaptive filter

4.2 The Objective Function and Method of Iterative Recursive Adaptive Equalizer

(1) Objective Function of the Iteration: Determining the objective function iteration is to determine when the tap coefficient is best optimal. In this way, the tap coefficient can be used to equalize signal and ensure the output signal without inter-symbol interference.

Assuming that the expected signal (known training sequence) is $d(n)$, the error output sequence is $e(n) = d(n) - y(n)$. In the training mode, the tap coefficient $w(n)$ can be adjusted according to some algorithm, so that the cost of adaptive equalizer can be minimized. In this study, the least mean square error (LMSE) is adopted, which is the least mean square error between $d(n)$ and $y(n)$. This method is also referred to as the least mean square (LMS) algorithm.

(2) Iterative Method: The most widely used convergence principle, "steepest descent method" or said "gradient descent method" is adopted in this study to realize the iterative update of equalizer tap coefficients. Tap coefficients of next moment $w(n + 1)$ equal tap coefficients of this moment $w(n)$ add a negative square error gradient-$\nabla(n)$. Equation is expressed as $w(n + 1) = w(n) - \mu\nabla(n)$, in which μ is convergence factor for controlling the convergence rate.

Because the accurate calculation of the gradient is very complicated and difficult, the estimation of mean square error is used to approximate calculation, $\nabla(n) = -2e(n)x(n)$. Then, the iterative formula $w(n + 1) = w(n) + 2\mu e(n)x(n)$.

5 Simulation and Analysis

When simulating the system, the presumed channel and environmental parameters are: the depth of $h = 100$ m; horizontal distance between the transmitter and receiver $L = 1000$ m; distance between the transmitter and sea bottom $a = 20$ m; distance between the receiver and sea bed $b = 80$ m; working frequency of sound waves

f = 8 kHz; the speed of sound waves in the sea water c = 1500 m/s; the speed of wind on the surface of seawater speed_wind = 11.67 knots (6 m/s); sea bed reflection coefficient $|r_b|$ = 0.9; number of sent bit is 48 (modulation signal duration is 6 ms); modulation mode is 2ASK modulation; the channel signal-to-noise ratio SNR = 10; the training sequences are 8 pairs continuous 0,1 signals (the duration is 2 ms).

2ASK modulation is adopted. After bit 1 modulations the amplitude is 1, while after bit 0 modulations the amplitude is 0. In Fig. 3, the picture above is the entire received multipath signal, and the former blank is the time delay of the direct path from transmitter to receiver. The middle picture is the front part of the received signal. ISI can be found only in the latter part. This is because of the time delays difference between the second arrived path and the direct path. Even though the multipath signal is attenuated in the amplitude compared with the direct path, there is more than one path. As a result sometimes the maximum value of the received signal is greater than the direct path. The following picture is the whole the spectrum of the received signal. You can see the frequency is 8 kHz. The simulation of these instructions is successful.

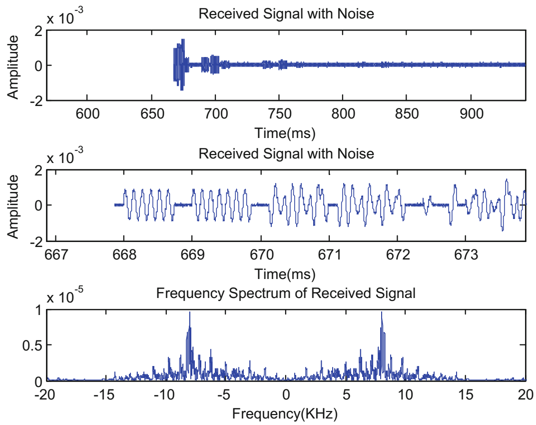

Fig. 3. The received multipath signal and its spectrum

In Fig. 4, the picture above is a modulated signal, and the following is an equalized signal of the received multipath signal. It can be discovered that the signal has a good recovery after the equalization. This is because the SNR of hypothetical channel is relatively high. As a result, the impact of noise on signal transmission is small.

According to Fig. 5, it can be seen that the error rate of the information transmission decreases with the increase of the channel SNR. There is no bit error in the channel of relatively high SNR. In this curve, I find that the bit error rate of the channel is relatively low when SNR is −1, for the equalizer can largely reduce the negative effect of the noise on the transmission.

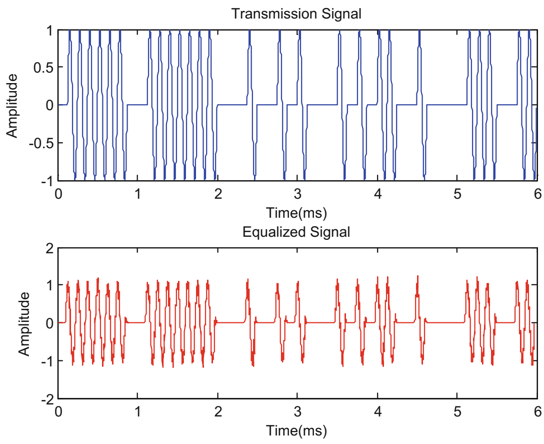

Fig. 4. The comparison between equalized signal and original signal

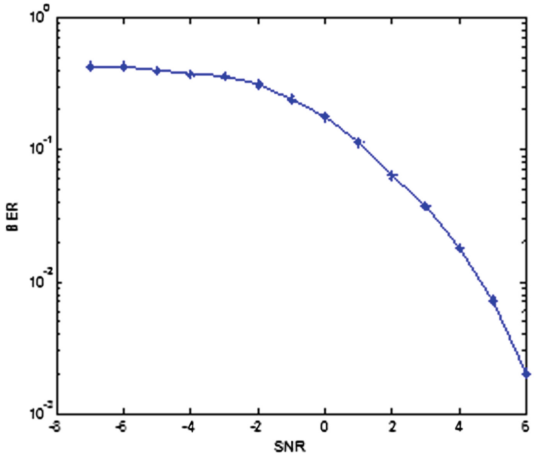

Fig. 5. The error rate curve of shallow seawater channel

6 Conclusions

(1) The characteristics of underwater acoustic communication are introduced, and the parameters of the underwater acoustic channel are analyzed from the perspective of physics. Then the "Ray Model" is chosen to simulate the channel, and the model is analyzed. After that, the model is described from the geometric point of view, and some parameters of the model are obtained. Then the delay and loss of the received signal are calculated. There are three kinds of loss, which is the extension attenuation, the absorption attenuation, and the attenuation. Finally, the expression of the received signal is given, which provides the basis for the follow-up work.

(2) In order to solve the inter symbol interference (ISI) caused by multipath effect of received signal, the adaptive equalization technique is adopted in this paper. After analyzing the reason of using the adaptive equalization technique, the working mode and structure of the adaptive equalizer is presented in this paper, and the principle and formula of iterative recursion is analyzed.

(3) MATLAB simulation of the underwater acoustic channel model is carried out using 2ASK modulation. In the simulation results, the severe Inter symbol interference can be observed, which indicates that the simulation of the channel can reflect the characteristics of the underwater acoustic channel. In the balanced simulation, the signal of the equalizer is almost the same as that before entering the channel, and the error rate curve is obtained, which indicates the equalizer has the ability to solve the severe Inter symbol interference.

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