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Power optimization algorithm for OFDM underwater acoustic communication using adaptive channel estimation

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Abstract: An adaptive channel estimation algorithm for the channel length is proposed to construct a channel estimation model suitable for orthogonal frequency division multiplexing (OFDM) underwater acoustic communication signals for the dependence of traditional channel estimation algorithms on channel length information. This algorithm can be adopted to evaluate channel estimation quality in real time and to adaptively adjust the channel length of the channel estimation algorithm according to the evaluation result, which satisfies the need of accurate estimation of unknown underwater acoustic channels and communication application; based on the study on the relationship between the OFDM communication bit error rate and the subcarrier signal to noise ratio, a self-adjusting optimization scheme for OFDM subcarrier transmitting power is proposed, which realizes underwater communication with the low bit error rate through higher energy efficiency. The validity of the research content is verified through simulation and field experiments.

Keywords: channel estimation, underwater acoustic communication, orthogonal frequency division multiplexing (OFDM), adaptive.

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1. Introduction

As an efficient multi-carrier underwater acoustic communication scheme, orthogonal frequency division multiplexing (OFDM) has the advantages of high bandwidth utilization and strong inter-code interference resistance. It is the focus of application and research in the field of underwater acoustic communication in recent years [1 – 4]. In order to achieve high efficient underwater data communication, OFDM adaptive underwater acoustic communication technology based on underwater acoustic channel information has drawn widespread attention [5 – 8]. According to the estimated underwater acoustic channel state, OFDM spectrum resources, transmission power, modulation mode, communication rate and other indicators are automatically optimized and adjusted to achieve the optimal matching between the underwater acoustic communication signal and the harsh underwater acoustic channel, thereby improving the effectiveness and reliability of the OFDM underwater acoustic communication system [9 – 11].

At present, most of underwater acoustic channel estimation algorithms are carried out under the assumption that the channel length is known [12 – 17]. However, the actual underwater acoustic environmental interface, random reflection and scattering of scatterers can cause time-varying multipath structures, resulting in the changes in the channel length, and multipath expansion in different underwater acoustic channels ranging from microseconds to seconds [18,19]. Therefore, the selected channel length L in underwater acoustic channel estimation algorithms will have a great impact on the channel estimation quality, real-time performance of the algorithm and timeliness of the adaptive underwater acoustic communication [20]. For this reason, this paper analyzes the influence of the channel length on channel estimation accuracy and underwater acoustic communication effect, and proposes an adaptive underwater acoustic channel estimation algorithm, which can evaluate the channel estimation equality in real time and adaptively adjust the actual underwater acoustic channel length. In addition, based on reliable channel estimation evaluation results, the adaptive optimization scheme for OFDM subcarrier power is studied from the perspective of improving the reliability and energy efficiency of underwater acoustic communication.

2. Study on length adaptive channel estimation algorithm based on OFDM signal

OFDM is a communication technology that performs parallel data transmission on multiple orthogonal subcarriers [21]. The modulation and demodulation of signals can be realized by V points inverse fast Fourier transform (IFFT)

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and fast Fourier transform (FFT) respectively [22]. The time domain signal of a single symbol period T is as shown in (1):

$$x(n) = \frac{\mu}{\sqrt{V}} \sum_{k=0}^{V-1} X(k) \mathrm{e}^{\mathrm{j}\frac{2\pi}{V}kn} \quad (1)$$

where $X(k)$ is the corresponding complex baseband subsymbol on the subcarrier k, which includes the data information to be transmitted on the subcarrier. According to the FFT rule, the frequency corresponding to the subcarrier k can be calculated by using (2). f_{sam} is the sampling frequency of the digital time domain signal $x(n)$. In practice, it is necessary to select N subcarriers located in the available underwater acoustic communication bandwidth from all $V/2$ subcarriers [23].

$$f_k = k \cdot f_{\mathrm{sam}}/V \quad (2)$$

In ideal cases, after $x(n)$ is sent to transmission channels, the receiver can recover the source information $X'(k)$ and complete signal demodulation by performing corresponding V points FFT operation on the symbol receiving sampling signal. However, due to the serious multipath effect in the underwater acoustic channel, in order to reduce the distortion of the multipath effect on communication signals, OFDM needs to add a cyclic prefix (CP) of length Δ before each symbol [24]. Generally the CP length is required to be greater than the maximum multipath delay spread value of the underwater acoustic channel to overcome inter-carrier interference and inter-symbol interference caused by multipath effects. At this time, the symbol period signal received by the receiving end without CP $\boldsymbol{y} = [y(0), \dots, y(V-1)]^{\mathrm{T}}$ can be written in the form of (3) [25], where $\boldsymbol{X} = [X(0), \dots, X(V-1)]^{\mathrm{T}}$ is the transmission sequence and $\boldsymbol{W}$ is an FFT matrix with its elements $\boldsymbol{W}_{n,m} = \mathrm{e}^{-2\pi(n-1)(m-1)i/V}/\sqrt{V}$. $\boldsymbol{C}$ is a $V \times V$-dimensional channel cyclic matrix. The first column of $\boldsymbol{C}$ is the underwater acoustic channel impulse response $[c_0, \dots, c_{L-1}, 0, \dots, 0]^{\mathrm{T}}$, and L is the real channel length. $\boldsymbol{n}$ is the noise sequence. It should be noted that there is inevitably a Doppler shift in underwater acoustic channels, but it can be compensated by the corresponding method [26].

$$\boldsymbol{y} = \boldsymbol{C}\boldsymbol{W}^{\mathrm{H}}\boldsymbol{X} + \boldsymbol{n} \quad (3)$$

In order to achieve tracking and estimation of underwater acoustic channels through the received signal $\boldsymbol{y}$, it is necessary to add a known pilot sequence in the OFDM symbol period. Assuming that the transmitted information column vector $\boldsymbol{X}$ can be written in a form $\boldsymbol{X} = \boldsymbol{X}_p + \boldsymbol{X}_v$, where $\boldsymbol{X}_p$ contains only the pilot sequence, and the valid transfer information is stored in $\boldsymbol{X}_v$. In order to extract the information determined by $\boldsymbol{X}_p$ in $\boldsymbol{y}$, $\boldsymbol{y}$ can be processed according to (4), where $\boldsymbol{N}$ is a $V \times V$-dimensional matrix with all its elements equal to 0, except the element on the position (s, s) equals 1 if the sth element $X(s)$ of the transmission sequence $\boldsymbol{X}$ is a pilot element.

$$\boldsymbol{b} = \boldsymbol{W}^{\mathrm{H}}\boldsymbol{N}\boldsymbol{W}\boldsymbol{y} \quad (4)$$

Since $\boldsymbol{C}$ is a cyclic matrix, $\boldsymbol{W}\boldsymbol{C}\boldsymbol{W}^{\mathrm{H}}$ is also a diagonal matrix. Thus, by substituting (3) into (4), (5) can be obtained, which means that $\boldsymbol{b}$ is the received symbol period signal if the information source only transmits the pilot sequence $\boldsymbol{X}_p$ in the transmitted information column vector $\boldsymbol{X}$, without $\boldsymbol{X}_v$.

$$\boldsymbol{b} = \boldsymbol{W}^{\mathrm{H}}\boldsymbol{N}(\boldsymbol{W}\boldsymbol{C}\boldsymbol{W}^{\mathrm{H}})(\boldsymbol{X}_p + \boldsymbol{X}_v) + \boldsymbol{n}'' = \boldsymbol{C}\boldsymbol{W}^{\mathrm{H}}\boldsymbol{X}_p + \boldsymbol{n} \quad (5)$$

Let $\boldsymbol{d} = \boldsymbol{W}^{\mathrm{H}}\boldsymbol{X}_p$. According to the cycle characteristics of $\boldsymbol{C}$, (5) can be transformed into (6), where $\boldsymbol{D}$ is a $V \times L$ dimensional truncated cyclic matrix whose first column element is $\boldsymbol{d}$. $\boldsymbol{c}$ is the underwater acoustic channel $[c_0, \dots, c_{L-1}]^{\mathrm{T}}$ to be estimated. A signal estimation model for OFDM underwater acoustic communication signals is established based on it.

$$\boldsymbol{b} = \boldsymbol{D}\boldsymbol{c} + \boldsymbol{n}' \quad (6)$$

According to the least squares (LS) criterion, the channel response estimate $c_{LS}^{\wedge}$ can be obtained as shown in (7) [27]:

$$\boldsymbol{c}_{LS}^{\wedge} = (\boldsymbol{D}^{\mathrm{H}}\boldsymbol{D})^{-1}\boldsymbol{D}^{\mathrm{H}}\boldsymbol{b}. \quad (7)$$

Although the LS algorithm is a general method for channel estimation, it involves complex high-dimensional matrix inversion operations, which is computationally intensive. More importantly, because $\boldsymbol{X}_p$ contains many 0 elements, $\boldsymbol{D}^{\mathrm{H}}\boldsymbol{D}$ approximates a singular matrix whose inverse operations require more iterations and the true value can even not be calculated. It can be seen that the traditional LS channel estimation algorithm and the improved ThLS, SpLS algorithm [28] are not directly applicable to the channel estimation of (6).

In fact, underwater acoustic channels have their unique sparse weight feature, that is, only a very small number of weights c_i are not zero. With the sparse property, the estimation of the underwater acoustic channels can be evolved into linear combination approximation of the received data $\boldsymbol{b}$ with fewer columns of $\boldsymbol{D}$. The positions of these columns are the important propagation paths of the underwater acoustic channel, that is, the locations where the non-zero weights are located and the corresponding linear combination coefficients are the corresponding channel path weights. For such problems, the matching pursuit (MP) algorithm [29] can be used to achieve fast and suboptimal solution estimation of the underwater acoustic sparse channel with a small amount of computation. The specific steps are as follows.

(i) Let $\boldsymbol{b}_0 = \boldsymbol{b}$. The estimated channel weights are initialized to 0.

(ii) Let $\boldsymbol{m}_i = \boldsymbol{D}^{\mathrm{H}}\boldsymbol{b}_i$. Find the most matching column d_{k_p} with $\boldsymbol{b}_i$ in $\boldsymbol{D}$. k_p is the location of the important path of the underwater acoustic channel.

$$k_p = \max_l(|(\boldsymbol{m}_i)_l|^2/\|\boldsymbol{d}_l\|^2) =$$

$$\max_l(|\boldsymbol{d}_l^{\mathrm{H}}\boldsymbol{b}_i|^2/\|\boldsymbol{d}_l\|^2), \quad l = 1, \dots, L \tag{8}$$

(iii) Calculate the underwater acoustic channel weight coefficient of the determined path k_p by

$$c_{k_p}^{\wedge} = |\boldsymbol{d}_{k_p}^{\mathrm{H}}\boldsymbol{b}_i|^2/\|\boldsymbol{d}_{k_p}\|^2. \tag{9}$$

(iv) Remove the contribution of the k_pth column of $\boldsymbol{D}$ in the linear approximation to $\boldsymbol{b}$ through (10).

$$\boldsymbol{b}_{i+1} = \boldsymbol{b}_i - (\boldsymbol{d}_{k_p}^{\mathrm{H}}\boldsymbol{b}_i)\boldsymbol{d}_{k_p}/\|\boldsymbol{d}_{k_p}\|^2 \tag{10}$$

(v) Repeat (2) – (4) until $c_{k_p}^{\wedge}$ is less than the threshold T_{MP} and the algorithm ends.

It can be seen that the MP algorithm can quickly determine the important path position of the channel k_p and its weight coefficient $c_{k_p}^{\wedge}$ by finding the matching degree of each column in $\boldsymbol{D}$ with $\boldsymbol{b}$ based on the linear approximation contribution, which is very suitable for the estimation of the sparse hydrophobic acoustic channel. However, it can be seen from (8) and (10) that the MP algorithm needs each column of the matrix $\boldsymbol{D}$ to satisfy the orthogonality, but in the actual situation, such orthogonality is difficult to guarantee, so estimation error is introduced for the MP algorithm. In order to improve the channel estimation accuracy, the orthogonal MP (OMP) algorithm can be used [30]. However, the LS operation included in the OMP algorithm still requires matrix inversion. When there are many important paths in the underwater acoustic channel, the calculation amount is large. Therefore, from the practical perspective, it is necessary to improve the estimation accuracy of the MP algorithm with small calculation. The improvement is carried out in two aspects. Firstly, important paths of the underwater acoustic channels should be selected accurately. The exponential dynamic path decision threshold T_{MP} shown in (11) can be introduced to the MP algorithm which makes T_{MP} change from $\alpha + \beta$ to β automatically with the signal-to-noise ratio (SNR) raising from 0 to infinity. According to (11), β plays the important role when the SNR is high, and this favorable acoustic environment makes precise channel estimation possible. In general, the value β is less than 0.02 to ensure the estimation accuracy. On the other hand, when the environment is poor and the SNR is relatively low, we have to discard the less important paths less than $\alpha + \beta$ (when SNR equals 0) in order to avoid the false paths causing error estimations, in general $\alpha + \beta$ values about 0.15 which discards most of the false paths and also leaves the important key acoustic channel paths.

$$T_{MP} = \alpha \cdot \mathrm{e}^{-\gamma \cdot \mathrm{SNR}} + \beta \tag{11}$$

In addition, because the orthogonality of each column of $\boldsymbol{D}$ is difficult to guarantee, the newly calculated path may have an influence on the previously calculated paths. Inspired by the OMP algorithm, we can replace (9) with (12) and implement the second iteration correction to nonzero weight coefficients of important paths until the coefficient correction amount $|\Delta c|$ is less than a certain smaller value ε, and the algorithm ends, thereby further improving the estimation precision of weight coefficients. $c_{k_p,i}^{\wedge}$ represents the weight coefficient of the k_pth position estimated by the ith iteration.

$$c_{k_p,i}^{\wedge} = c_{k_p,i-u}^{\wedge} + \Delta c = c_{k_p,i-u}^{\wedge} + |\boldsymbol{d}_{k_p}^{\mathrm{H}}\boldsymbol{b}_i|^2/\|\boldsymbol{d}_{k_p}\|^2 \tag{12}$$

In the typical underwater acoustic channel given in [31] $\boldsymbol{H}_{t1} = [0.83, 0_{(5)}, 1, 0_{(3)}, -0.69, 0_{(7)}, 0.476, 0_{(5)}, 0.081]$ (the subscript of 0 represents the number of consecutive 0 coefficients) where the channel length $L = 25$, the estimation error of different channel estimation algorithms is shown in Fig. 1. The horizontal axis of Fig. 1 represents SNR and the vertical axis represents the dB value of the channel estimation error calculated with (13). In simulations, the decision threshold T_{MP} of the traditional MP algorithm is set to 0.06; the threshold decision parameters of the improved MP algorithm are $\alpha = 0.13$, $\beta = 0.02$ and $\gamma = 0.05$. The sampling frequency f_{sam} of OFDM signals is 100 KHz and the points of IFFT V for generating OFDM communication signals in (1) values 1 024 which means the duration of an OFDM symbol is 10.24 ms. The index of OFDM subcarriers is used for transmitting values from 103 to 234 which means the transmission sequence $\boldsymbol{X} = [0, \dots, 0, X(103), \dots, X(234), 0, \dots, 0]^{\mathrm{T}}$ and the communication bandwidth occupied is between 10 KHz and 23 KHz. In the 132 used subcarriers, each subcarrier uses QPSK as its communication mode and the pilot elements are inserted in $\boldsymbol{X}$ with the interval of four whose indices are 103, 107, …, and 234. Thus the number of pilot elements is 34 in an OFDM symbol.

$$E_{es} = 10\lg(\sum_{i=0}^{L-1}|c_i - c_i^{\wedge}|/L) \tag{13}$$

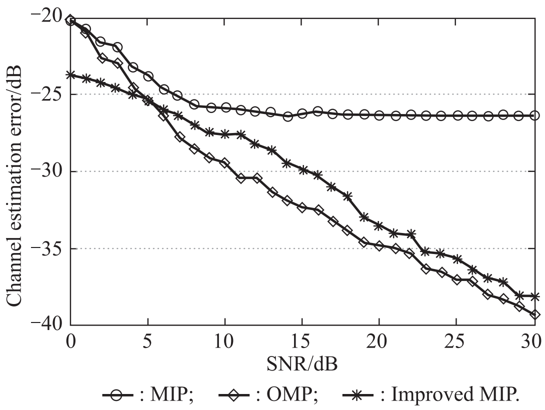


Fig. 1 Comparison of estimation accuracy among channel estimation algorithms

The simulation shows that the improved MP algorithm adopts the exponential adjustment mechanism for threshold T_{MP}, which improves the discriminating ability of channel paths, reduces the adverse effects of false paths on the estimation algorithm, and improves the channel estimation accuracy under the low SNR. In addition, the iterative correction of the path weight coefficients reflects the interlacing effect of non-orthogonality in each coefficient estimation, which makes the improved MP algorithm break through the convergence bottleneck of the traditional MP algorithm, and achieves the estimation accuracy equivalent to that of the OMP algorithm (about 2 dB lower), but with a much smaller amount of calculation. Thus it can be seen that the improved MP algorithm has good comprehensive performances.

Based on (6), all the above estimation algorithms work under the condition that the channel length L is known. However, in practice, the multipath extension length of the underwater acoustic channel is difficult to estimate, and the length of the underwater acoustic channel will slowly change due to the random dispersion and reflection of the interface and scatterers. It is assumed that the channel length of the channel estimation model at the receiving end is M, that is, the matrix $\boldsymbol{D}$ in (6) becomes a $V \times M$ dimensional truncated cyclic matrix at this time. When SNR is 15 dB, in channel $\boldsymbol{H}_t$, the relationship between the channel estimation accuracy of the improved MP algorithm and the channel estimation length M is as shown in Fig. 2, where the abscissa represents the variation factor α, and when L is constant, the channel estimation length $M = \alpha L$.

It can be seen from Fig. 2 that with the reduction of estimated channel length M, the estimation accuracy of the channel estimation algorithm shows a downward trend as a whole, especially when M is lower than the real length L of the underwater acoustic channel. The performance of the algorithm drops sharply and the channel estimation error is large, which cannot be used to adjust the parameters of OFDM underwater acoustic communication. When M is longer than the real channel length L, although the increase of M can improve the channel estimation accuracy, the improvement space is very limited. In addition, relatively large M will bring about a large increase in the channel estimation calculation amount, affecting the computational efficiency of the channel estimation algorithm and real-time performance of it.

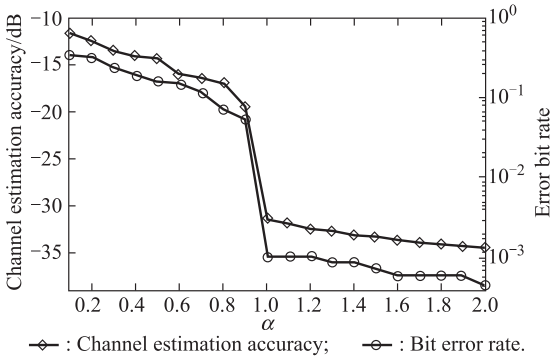


Fig. 2 Channel estimation accuracy and bit error rate of the improved MP algorithm under different M

In order to reasonably select M of the channel estimation algorithm and adapt to the underwater acoustic channel requirements with time-varying multipath delay (microsecond to second), a means for real-time evaluation of channel estimation quality is needed, and M should be adjusted according to it. When the channel estimation quality is found to be poor, M can be increased to improve the estimation accuracy, on the contrary, when the estimated quality meets the application requirements, M can be kept or reduced to improve the calculation efficiency. Under the premise that the channel information is completely unknown, how to evaluate the channel estimation quality is the key to the adaptive channel estimation algorithm.

For underwater acoustic communication, after the signal sequence $x(n)$ sent by the source is transmitted through the underwater acoustic channel, the received symbol signal $\boldsymbol{x}' = [x'(0) \cdots x'(V-1)]$ will be greatly distorted, which seriously affects the correct recovery of the frequency domain transmission information $X(k)$. OFDM generally needs to rely on the frequency domain equalizer to overcome the adverse effects of multipath effects [32]. To this end, the channel response $\boldsymbol{c}^{\wedge}$ estimated by the improved MP algorithm can be utilized to calculate the frequency domain equalization filter coefficient $H_e(k)$, and the received information sequence $X'(k)$ can be equalized and recovered by (14). When the channel estimation quality is good, the information $X_r(k)$ after equalization should be able to reflect the relevant features of the source sequence $X(k)$ very well. Therefore, the quality of the channel estimation can be evaluated by comparing the feature of recovered sequence $X_r(k)$ with the real feature of the source sequence $X(k)$.

$$X_r(k) = X'(k)H_e(k) = (\boldsymbol{W}\boldsymbol{x}')_k/(\sqrt{V}\cdot\boldsymbol{W}\boldsymbol{c}^{\wedge})_k \quad (14)$$

Under the simulation conditions same with Fig. 2, after equalization compensation of (14), the bit error rate (BER) obtained by the OFDM underwater acoustic communication system is shown by the "◯" line in Fig. 2, and it can be seen that when M is smaller than L, worse channel estimation quality results in equalization failure and BER is high. On the other hand, when M is greater than L, the BER level is basically maintained at a satisfactory low level. Fig. 2 clearly shows that the actual BER of underwater acoustic communication can be used to evaluate the channel estimation quality, which is a relatively intuitive method, but requires a large number of known training sequences, resulting in large bandwidth waste. Another more effective method is to use the relevant statistical information of the known transmission sequence $X(k)$, such as the constellation of phase modulation. In ideal equalization compensation, the recovery information $X_r(k)$ should be located near the specified U constellation points C_s $(s = 1, \dots, U)$ [33]. In this way, by calculating the degree of dispersion $ds(k)$ between $X_r(k)$ and specified constellation points C_s, the channel estimation quality and equalization quality can be quickly evaluated. The calculation of $ds(k)$ is shown as (15). When the average value $\overline{ds(k)}$ is greater than the minimum threshold $T_{\rm com}$ required for reliable underwater acoustic communication, it means the channel estimation quality is poor and the equalization effect is not good, the estimated channel length M should be increased; on the contrary, it means the equalizer can work effectively, the current estimated channel length M can meet the channel estimation requirements and can even be appropriately reduced to improve the computational efficiency, thereby realizing the adaptive adjustment of the channel estimation algorithm to the real underwater acoustic channel length.

$$ds(k) = \min_{s=1,\dots,U} |X_r(k) - C_s| \quad (15)$$

Fig. 3 shows the simulation comparison between channel estimation accuracy and QPSK constellation $(U = 4)$ dispersion, which shows that the constellation dispersion degree really can reflect the channel estimation quality and provide basis for adaptive adjustment of the channel estimation algorithm.

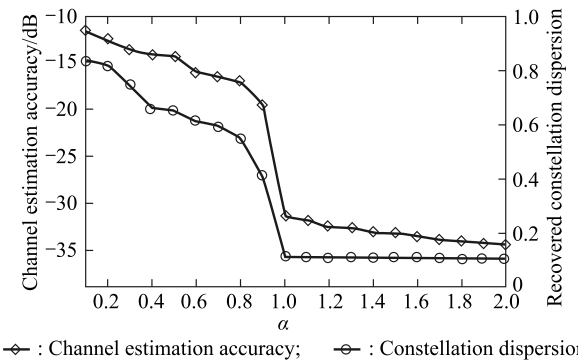


Fig. 3 Channel estimation accuracy and recovery constellation dispersion of the improved MP algorithm under different M

Fig. 4 shows a comparison of the channel length adaptive channel estimation algorithm and the fixed channel length ($M = 10, 16, 22$ and 50) estimation algorithm under the channel $\boldsymbol{H}_{t1}$. The initial estimated channel length of the improved MP channel estimation algorithm is $M = 10$. For discussion, Fig. 4 uses a simple equal-length search strategy to adjust M with constant amount of increment (decrement) of six each time.

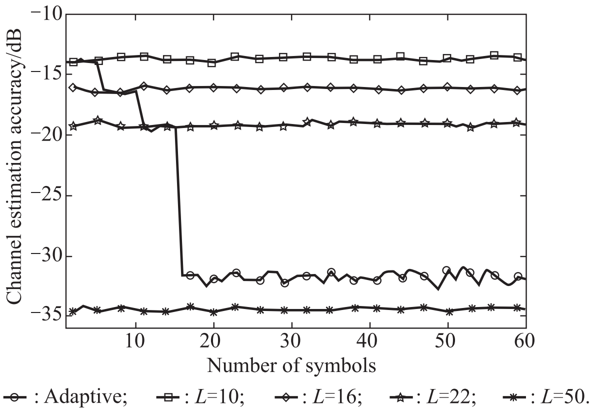
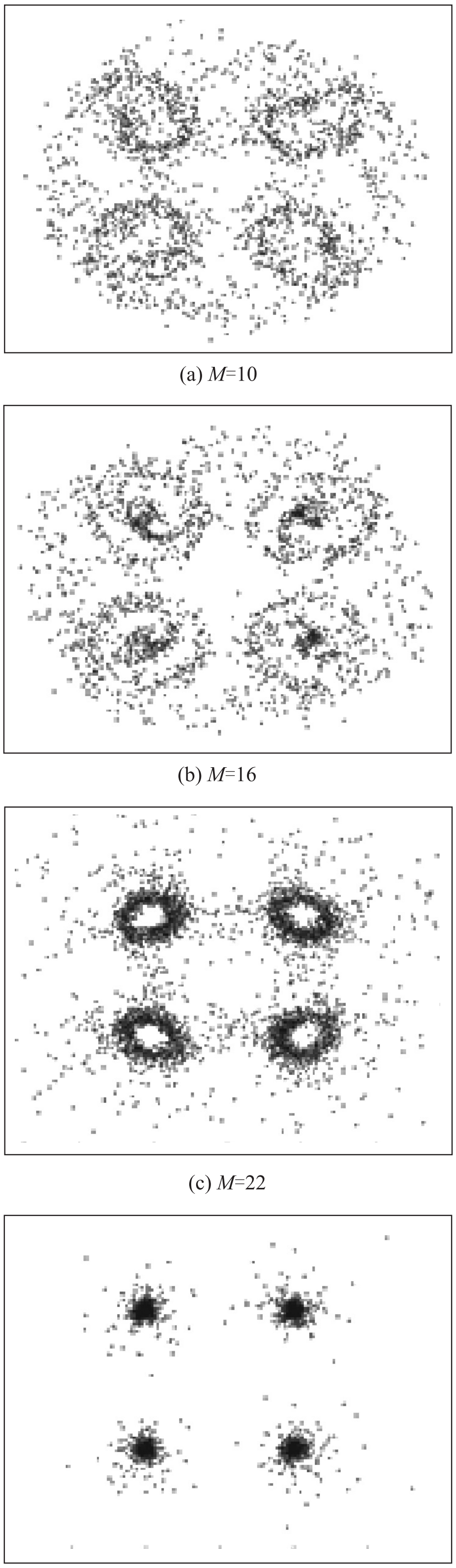


Fig. 4 Comparison of channel length adaptive estimation algorithms

As can be seen from Fig. 4, the adaptive channel estimation algorithm has experienced four adjustment stages of $M = 10$, 16, 22, and 28. When M is less than the real channel length $L = 25$, the equalizer output constellation is chaotic, as shown in Fig. 5(a) – (c) and the BER is high. The larger constellation dispersion is used as the channel estimation quality feedback and M is increased automatically to adapt to the underwater acoustic channel. When M is equal to 28 (greater than L), the clear constellation, as shown in Fig. 5(d), indicates that the current channel estimation algorithm can provide good estimation accuracy and there is no need to continue to increase M. Actually, even if M is larger and becomes 50, the effect of improving the channel estimation accuracy is very limited as shown in Fig. 4, but the amount of calculation is greatly increased.



(a) M=10

(b) M=16

(c) M=22

(d) M=28

Fig. 5 Equalized constellation under different M

Therefore, it can be seen that the adaptive channel estimation algorithm can adapt the real underwater acoustic channel length automatically and achieve accurate estimation of the underwater acoustic channel with efficient calculation efficiency.

3. OFDM underwater acoustic communication subcarrier power optimization adjustment algorithm

The adaptive channel estimation algorithm proposed in Section 2 can accurately estimate the unknown and time-varying underwater acoustic channels, and provide a reliable basis for parameter optimization of OFDM underwater acoustic communication systems. Many underwater systems (underwater vehicles, underwater ad hoc networks) are mostly energy-constrained systems. Therefore, under certain BER constraints, minimizing the energy consumption of underwater acoustic communication is the key to maintaining long-term reliable operation of these underwater systems. In [34], a symbol power adjustment method is proposed, which calcualtes the environmental SNR and adjust the symbol power parameter μ in (1) under the constrain of a certain BER based on channel estiamtion results. However, this method does not consider the individual differences on OFDM subcarriers and treat each subcarrier indistinguishably, resulting in waste of energy on some subcarriers. To this end, this paper takes each subcarrier as the basic unit of power correction, and studies the power optimization algorithm that independently adjusts the transmitting power of each subcarrier to meet the final BER requirement. Based on (1), the OFDM underwater acoustic communication transmission signal is rewritten as shown in (16), where $\mu(k)$ is the power adjustment factor for the kth subcarrier.

$$x(n) = \frac{1}{\sqrt{V}} \sum_{k=0}^{V-1} \mu(k)X(k)\mathrm{e}^{\mathrm{j}\frac{2\pi}{V}kn} \tag{16}$$

Let the frequency domain transmission information of the kth subcarrier of OFDM be $X(k)$, the underwater acoustic channel attenuation on the kth subcarrier be $Cd(k)$, and the noise power be $\sigma_n^2(k)$. Then the frequency domain SNR on the subcarrier is shown in (17). The BER of each OFDM subcarrier communication varies with the subcarrier frequency domain SNR can be calculated for BPSK, QPSK and 8PSK modulated signals respectively, as shown in Fig. 6.

$$C\_snr(k) = 10\lg[\frac{|\mu(k)X(k)Cd(k)|^2}{\sigma_n^2(k)}] \tag{17}$$

Since the underwater acoustic channel has the frequency selection characteristics [35], it can be seen from (17) that strong fluctuations in channel attenuation at different frequencies will cause great differences in SNR on different OFDM subcarriers, directly leading to great differences in respective BER. In order to meet the specified communication BER requirement BER_r, the corresponding subcarrier frequency domain SNR matched to BER_r can be found according to Fig. 6. Therefore we use the channel attenuation $Cd^{\wedge}(k)$ obtained by the channel estimation algorithm proposed in Section 2; the sampled environmental subcarrier noise power $\sigma_n^{2\wedge}(k)$ and the known average amplitude $\overline{X}(k)$ of transmitted modulation signal, and the power adjustment factor $\mu(k)$ for the kth subcarrier satisfied with BER_r can be calculated by (17), thereby adjusting the overall power of the OFDM underwater acoustic communication signal by adjusting respective power factor $\mu(k)$ for each subcarrier, as shown in (16).

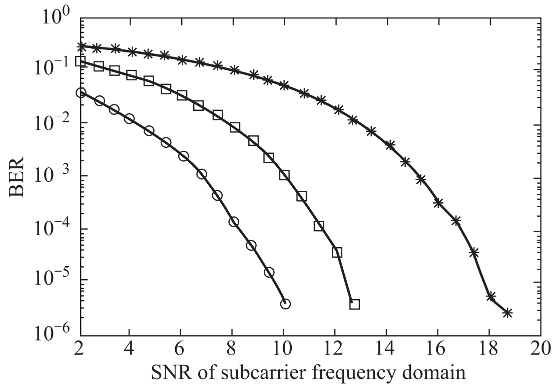


Fig. 6 Relationship curve between BER and SNR of subcarrier frequency domain

Fig. 7 and Fig. 8 respectively show the curve of time domain transmitting power needed by the proposed OFDM subcarrier power adjustment algorithm and that needed by the traditional symbol power adjustment algorithm proposed in [14], under the same simulation conditions of Fig. 1, with the BER requirements for audio transmission 10^{-3} and video transmission 10^{-4}. Also, the power consumptions saved by the proposed OFDM subcarrier power adjustment algorithm compared to the traditional algorithm are shown in Fig. 9 under the BERs of 10^{-3} and 10^{-4}.

The power adjustment is based on the estimation result of the adaptive channel estimation algorithm in Section 2 and it can be seen from Fig. 7 and Fig. 8 that as the environmental noise increases, both power adjustment algorithms need to increase the transmitting power to ensure the BER quality of underwater acoustic communication, and the higher the communication rate (8PSK $>$ QPSK $>$ BPSK), the greater the required transmitting power. Through simulation, it can be verified that both algorithms can meet the specified BER requirements after power adjustment, but the energy consumption costs of the two algorithms are different.

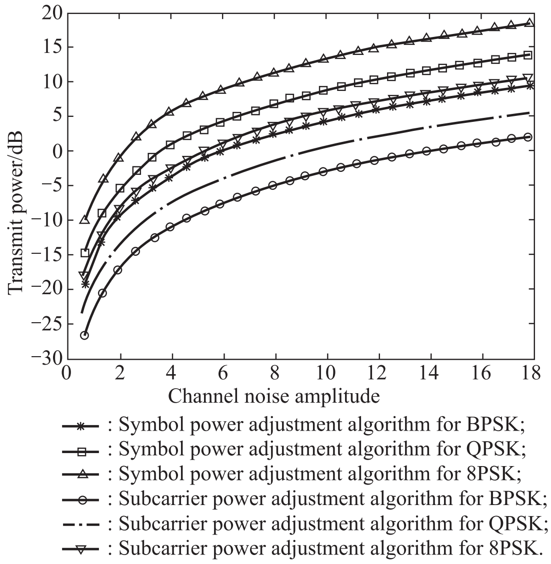


Fig. 7 Transmitting power of different algorithms under 10^{-3} BER requirement

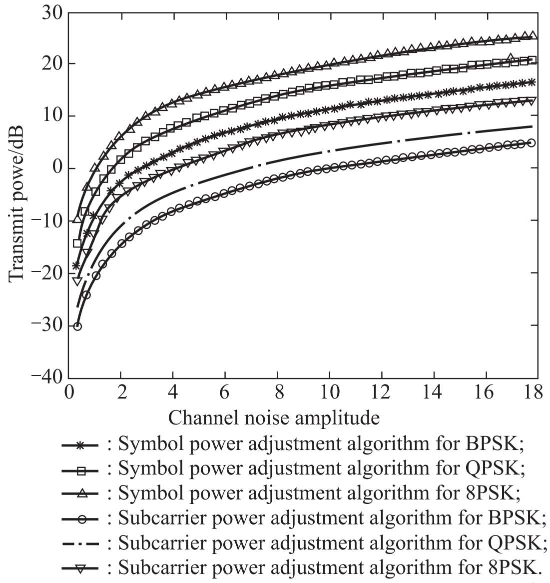


Fig. 8 Transmitting power of different algorithms under 10^{-4} BER requirement

From Fig. 9, it can be seen clearly that under different environmental noises, the proposed subcarrier power adjustment algorithm always gets a steady and similar power consumption save, which is nearly −8 dB under the BER of 10^{-3} and −12 dB under the BER of 10^{-4}.

The main factor that caused bad influence on BER is the deep fading OFDM subcarrier of the underwater acoustic channel. In order to overcome the serious errors caused by these deep fading, according to Fig. 6, the frequency domain SNR and corresponding power of these subcarriers need to be greatly improved. Because each subcarrier of the traditional symbol power adjustment algorithm has the same power adjustment gain, when the transmitting power at the depth fading subcarrier position is increased, the power of other subcarrier is also improved, but the power consumption of these non-depth fading subcarriers has little effect on the overall BER. Therefore, a large amount of transmitting energy is wasted. However, the algorithm proposed in this paper adjusts each subcarrier power independently according to their respective frequency domain SNR estimated by the channel estimation algorithm in Section 2. When the frequency domain SNR of one subcarrier is low relatively, the transmitting power of this subcarrier is increased correspondingly to meet the overall BER requirement and when the frequency domain SNR of one subcarrier is high relatively, the transmitting power is reduced reasonably to save energy, which achieves on-demand energy distribution and higher energy efficiency for underwater acoustic communications.

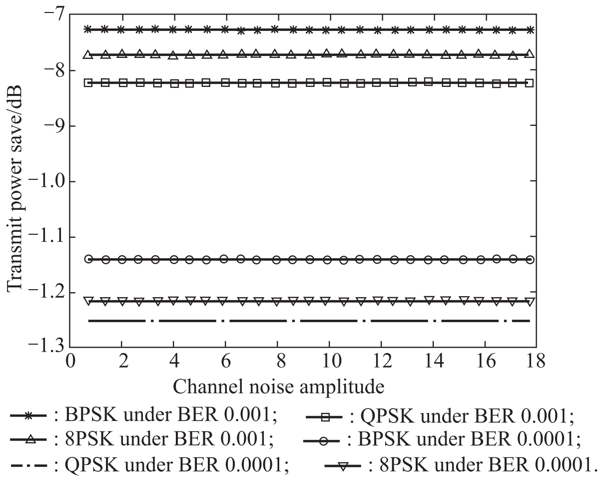


Fig. 9 Power save comparison with different modulations under 10^{-3} and 10^{-4} BER requirements

4. Test verification

In order to test the practical application effects of the new algorithm, a communication test with a distance of 700 m was carried out in Mulan Lake, and the 8PSK communication mode was adopted. Using the length adaptive channel estimation algorithm proposed in this paper, the underwater acoustic channel was estimated correctively and the channel length M was adjusted to 30 automatically. The real and imaginary parts of the underwater acoustic channel response estimated at a certain time are shown in Fig. 10(a), and the normalized power for each subcarrier corrected by the proposed subcarrier power optimization adjustment algorithm is shown in Fig. 10(b) which realizes on-demand distribution according to different subcarrier channel equalities. The recovered constellation obtained by equalization is shown in Fig. 10(c), and the BER is 0.074 3%, which achieves the required 10^{-4} BER requirement with higher energy efficiency.

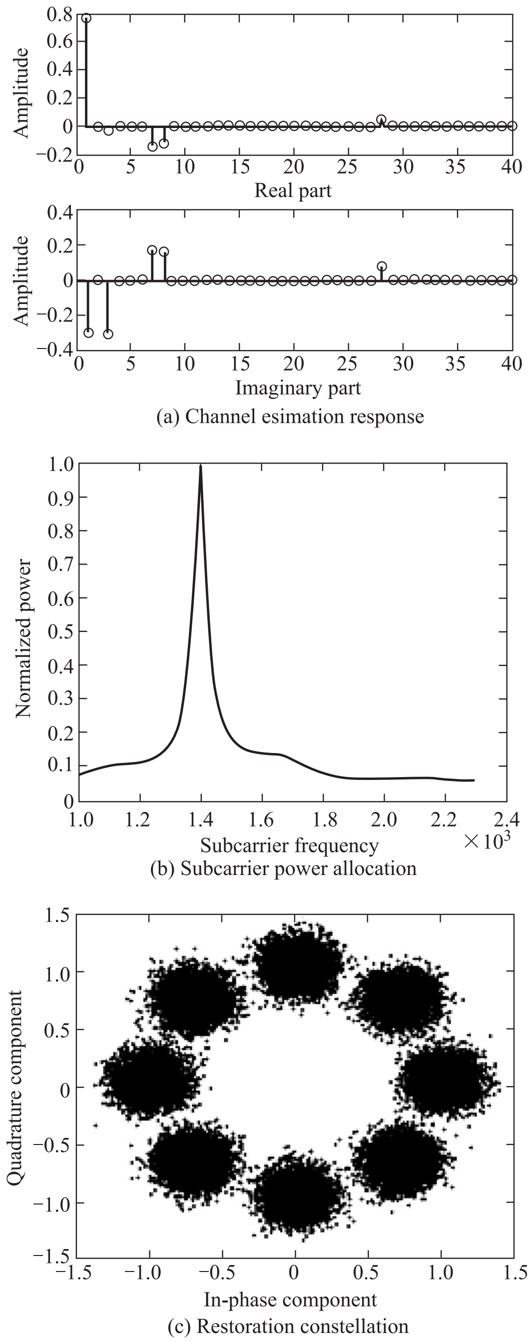


Fig. 10 Test results in Mulan Lake

5. Conclusions

Channel estimation is the premise and key for self-adjusting of underwater acoustic communication parameters. In this paper, for the dependence of the traditional OFDM channel estimation algorithm on the real channel length, a new channel length adaptive channel estimation algorithm is proposed which can be self-adjusted to reasonably select the channel estimation length in the unknown underwater acoustic environment by using the output signal of frequency domain equalizer to judge the channel estimation quality timely, and achieve accurate estimation of the underwater acoustic channel with higher computational efficiency. Based on the channel estimation result, the OFDM subcarrier transmitting power optimization adjustment algorithm is proposed. Compared with the traditional symbol adjustment algorithm, this new algorithm can achieve the specified BER with higher energy efficiency. Finally, the feasibility and effectiveness of the proposed algorithm are verified by simulations and experiments.

References

[1] ALAAELDEEN M E A, DUAN W Y. Overview on the development of autonomous underwater vehicles. Journal of Ship Mechanics, 2016, 20(6): 768 – 787.

[2] WANG J, YANG J, XIONG J, et al. SHAFA: sparse hybrid adaptive filtering algorithm to estimate channels in various SNR environments. IET Communications, 2018, 12(16): 1963 – 1967.

[3] MA D, SHI Y B, ZHANG W, et al. Performance analysis of V-OFDM for acoustic communication along drill strings. IET Communications, 2017, 17(4): 576 – 583.

[4] LI C G, SONG K, YANG L X. Low computational complexity design over sparse channel estimator in underwater acoustic OFDM communication system. IET Communications, 2017, 11(7): 1143 – 1151.

[5] GUO T L, ZHANG Z Y, ZHAO D F, et al. LS-OMP channel estimation algorithm for underwater acoustic OFDM systems. Technical Acoustics, 2017, 36(1): 10 – 16. (in Chinese)

[6] ZHU A F, LI X. Feedback strategies for iterative channel estimation in OFDM underwater acoustic communications. High Technology Letters, 2018, 24(4): 396 – 401.

[7] HUANG J G, WANG H, HE C B, et al. Underwater acoustic communication and the general performance evaluation criteria. Frontiers of Information Technology & Electronic Engineering, 2018, 19(8): 951 – 971.

[8] CHEN E, LIN N, SHUN H X, et al. Pre-coding based channel prediction for underwater acoustic OFDM. China Ocean Engineering, 2017, 31(2): 256 – 260.

[9] QIAO G, WANG W, LIU S Z. An improved decision feedback channel estimation algorithm for multiple-input multiple-output orthogonal frequency division multiplexing underwater acoustic communication. Acta Acoustic, 2016, 41(1): 94 – 104.

[10] JIANG Y X, MINN H, YOU X H, et al. Simplified frequency offset estimation for MIMO OFDM systems. IEEE Trans. on Vehicular Technology, 2017, 57(5): 3246 – 3251.

[11] CHEN Z R, WANG J F, ZHENG Y R. Frequency-domain turbo equalization with iterative channel estimation for MIMO underwater acoustic communications. IEEE Journal of Oceanic Engineering, 2017, 42(3): 711 – 721.

[12] QU F, WANG Z, YANG L. A journey toward modeling and resolving Doppler in underwater acoustic communications. IEEE Communications Magazine, 2016, 54(2): 49 – 55.

[13] DAOUD S, GHRAYEB A. Using resampling to combat Doppler scaling in UWA channels with single-carrier modulation and frequency-domain equalization. IEEE Trans. on Vehicular Technology, 2016, 65(3): 1261 – 1270.

[14] ZHAO Y, YU H, WEI G. Parameter estimation of wideband underwater acoustic multipath channels based on fractional Fourier transform. IEEE Trans. on Signal Processing, 2016, 64(20): 5396 – 5408.

[15] RENUKA B, SANGEETA J. Novel spectral efficient technique for MIMO-OFDM channel estimation with reference to PAPR and BER analysis. Wireless Personal Communications, 2019, 104(4): 1227 – 1242.

[16] CHANG Y K, UENG F B, SHEN Y S, et al. Joint channel estimation and turbo equalisation for MIMO-OFDM-IM systems. International Journal of Electronics, 2019, 106(5): 721 – 740.

[17] ZHAO J, PAUL T. Fast channel estimation and equalization scheme for offset-QAM OFDM systems. Optics Express, 2019, 27(2): 714 – 728.

[18] TAO J, WU J X. Enhanced MIMO LMMSE turbo equalization: algorithm, simulations, and undersea experimental results. IEEE Trans. on Signal Processing, 2011, 59(8): 3813 – 3823.

[19] JI W, REN C H, QIU L. Common sparsity and cluster structure based channel estimation for downlink massive MIMO-OFDM systems. IEEE Signal Processing Letters, 2019, 26(1): 59 – 63.

[20] ZHANG Y J, FENG W, GE N. Pilot power adaptation for tomographic channel estimation in distributed MIMO systems. IET Communications, 2017, 11(1): 112 – 118.

[21] ABDOLI H, NIKMEHR H, NASER M, et al. Improving energy efficiency of OFDM using adaptive precision reconfigurable FFT. Circuits Systems and Signal Processing, 2017, 36(7): 2742 – 2766.

[22] HAN J, ZHANG L L, ZHANG Q F, et al. Eigendecomposition-based partial FFT demodulation for differential OFDM in underwater acoustic communications. IEEE Trans. on Vehicular Technology, 2018, 67(7): 6706 – 6710.

[23] HAO J X, WANG J T, WU Y S. A new equalizer in doubly-selective channels for TDS-OFDM. IEEE Trans. on Broadcasting, 2015, 61(1): 91 – 97.

[24] AHMED A M, JASMIN E A. Adaptive modulation and interference cancellation techniques for MIMO-OFDM wireless networks. International Journal of Networking & Virtual Organizations, 2018, 18(3): 246 – 264.

[25] SUN X L, WANG P. Filter bank multicarrier technique simulation performance analysis. Electronic Measurement Technology, 2016, 39(5): 41 – 44. (in Chinese)

[26] KIM S W. A study on the correction of the channel equalizer decision error using channel estimator. Journal of the Korea Academia-Industrial Cooperation Society, 2017, 18(8): 18 – 24.

[27] JING H, ZHANG L L, GEERT L. Partial FFT demodulation for MIMO-OFDM over time-varying underwater acoustic channels. IEEE Signal Processing Letters, 2016, 23(2): 282 – 286.

[28] SAHOO H K, MOHANTY B, PATNAIK B. Block and fast block sparse adaptive filtering for outdoor wireless channel estimation and equalization. Wireless Personal Communications, 2018, 98(3): 3003 – 3019.

[29] LUO Y S, XU J H, HU H N. Research on self-adjusting OFDM underwater acoustic communication algorithm for transmission rate maximization. Journal of Electronics & Information Technology, 2015, 37(12): 2872 – 2876. (in Chinese)

[30] JOHANSSON H, HARRIS F. Poly phase decomposition of digital fractional-delay filters. IEEE Signal Processing Letters, 2015, 22(8): 1021 – 1025.

[31] LUO Y S, HU S L. Self-adjusting modulation algorithm for orthogonal frequency division multiplexing underwater acoustic communication. Journal of National University of Defense Technology, 2017, 39(1): 153 – 158. (in Chinese)

[32] RAMADAN K, DESSOUKY M I, ELAGOOZ S, et al. Equalization and carrier frequency offset compensation for underwater acoustic OFDM systems. Annals of Data Science, 2018, 5(2): 259 – 272.

[33] EKANT S, SAHIL R, KASTURI V. BER efficient interleaved OFDM system. Wireless Personal Communications, 2018, 98(1): 1531 – 1546.

[34] LIANG X W, ZHU Q. Energy efficient joint power and subcarrier assignment scheme in OFDM relay system. The Journal of China Universities of Posts and Telecommunications, 2015, 22(5): 56 – 62. (in Chinese)

[35] DIVNEET S K, AMIT K K. Channel estimation and long-range prediction of fast fading channels for adaptive OFDM system. International Journal of Electronics, 2018, 105(9): 1451 – 1466.

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