Variable Tap-Length Blind Equalization for Underwater Acoustic Communication

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Abstract. In view of the characteristics of underwater acoustic channel, a blind equalization algorithm for underwater acoustic communication based on variable tap length is proposed. On the basis of the normalized modified constant modulus algorithm (MCMA), the algorithm adjusts the length of the tap through the update algorithm to realize the blind equalization of the underwater acoustic channel. The simulation shows that the algorithm adaptively adjusts the length of the taps to the optimal. Compared with the traditional blind equalization algorithm, the performance of the system is improved.

Keywords: Underwater acoustic communication · Equalizers
Modified constant modulus algorithm (MCMA) · Variable tap-length

1 Introduction

In the underwater acoustic communication system, the communication quality of the underwater acoustic channel is greatly affected due to the existence of poor inter symbol interference (ISI). In order to overcome this weakness, channel equalization is commonly used in underwater acoustic communication systems to reduce ISI [1–5]. The traditional linear equalizer works by sending known training sequences. However, in the complex and time-varying underwater acoustic channels, the training sequence extraction fails due to the serious distortion of the receiving waveform, and the training sequence is sent periodically to cause the channel bandwidth waste and reduce equalizer performance. Therefore, the channel is equalized using a blind equalization

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technique that only needs to the received signal and statistical characteristics of transmitted signal.

Godard proposed a constant modulus algorithm (CMA) [6], which is simple and easy to implement, but the convergence rate is slow. In this connection, scholars have proposed a number of corresponding improved algorithms. The author in [7], proposed orthogonal constant modulus algorithm. Because the orthogonalization matrix is needed in the iterative process, the computation is heavy, when the signal dimension is large. In [8], the squared norm of the input signal is used as the normalization factor to improve the convergence speed. Due to the phase error of the equalized output signal constellation, MCMA proposed in [9] can achieve blind equalization and carrier recovery simultaneously. All of the above researches are based on fixed tap blind equalization. Due to the time-varying of underwater acoustic channels, the traditional fixed tap-length blind equalization cannot determine the optimal tap length, and is not suitable for complex and variable underwater acoustic channels.

In this paper, to solve the above problems, combined with the normalized MCMA, a blind equalization algorithm based on variable tap length is proposed, which adaptively adjusts to the optimal tap length and improves the performance of the system.

2 System Model

In Fig. 1, $a(n)$ is transmission sequence, $h(n)$ is the underwater acoustic channel impulse response, which is obtained from the BELLHOP model, $n(n)$ is the additive white Gaussian noise, and received signal can be seen in the following equations:

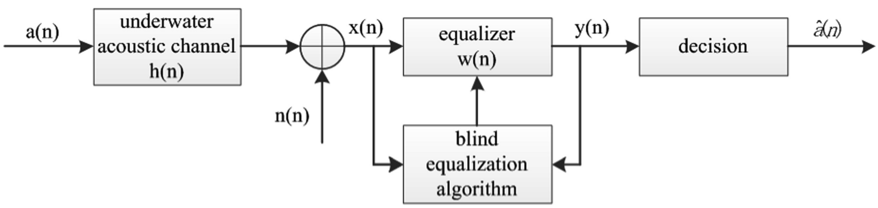

Fig. 1. The equivalent baseband model of a blind equalization system

$$x(n) = \sum_{i=0}^{N-1} h(i)a(n-i) + n(n) \tag{1}$$

And the output of the equalizer is expressed as the following formula

$$y(n) = X^T(n)W(n) \tag{2}$$

where $W(n) = [w_0(n), w_1(n), \ldots, w_{N-1}(n)]^T$ is the weight vector and $X(n) = [x(n), x(n-1), \ldots, x(n-N+1)]^T$ represents the input signal vector. In addition, N indicates the length of the equalizer.

This paper, we use normalized MCMA, the cost function of it is as follows:

$$J(n) = J_R + jJ_I = E\left\{(y_R^2(n) - R_{2R})^2\right\} + jE\left\{(y_I^2(n) - R_{2I})^2\right\} \tag{3}$$

where $y_R(n)$ and $y_I(n)$ represent the real and imaginary parts of the equalizer output, respectively. R_{2R} and R_{2I} are the real and imaginary parts of the input signal statistics. The transmitted signal constant modulus:

$$R_{2R} = \frac{E\{a_R^4(n)\}}{E\{a_R^2(n)\}} \tag{4}$$

$$R_{2I} = \frac{E\{a_I^4(n)\}}{E\{a_I^2(n)\}} \tag{5}$$

We use the random gradient descent method for the derivation of the cost function and seek the minimum, thus we get the error signal of MCMA and divide it into the real part and the imaginary part.

$$e_R(n) = y_R(n)(y_R^2(n) - R_{2R}) \tag{6}$$

$$e_I(n) = y_I(n)(y_I^2(n) - R_{2I}) \tag{7}$$

The normalized iterative recursive formula of the tap weight vector is

$$W(n+1) = W(n) + \frac{\mu}{\delta + X^T(n)X(n)} e(n)X^*(n) \tag{8}$$

where μ is step-size, δ is some small constant. $e(n)$ is an error signal, the formula is as follows:

$$e(n) = e_R(n) + je_I(n) \tag{9}$$

The MCMA algorithm equalizes the real and imaginary parts of the output signal separately. Since the MCMA algorithm utilizes the amplitude and phase information of the received signal, it can well solve the problem of phase deflection or offset after equalization (Fig. 2).

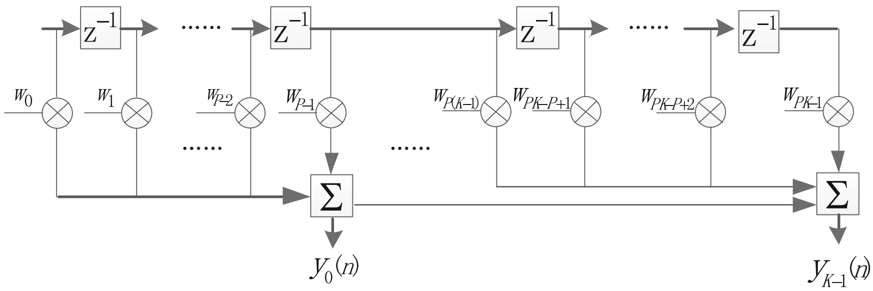

Fig. 2. Segmented equalizer with K segments of P taps

3 Variable Tap Length-Based MCMA (VT-MCMA)

The received signal is equalized by the VT-MCMA, finally recover the signal. The idea of VT-MCMA is to use a piecewise filter, that is, to divide a FIR filter with M taps into K segments, each of which contains P taps $(M = PK)$. Each segment produces an estimate of the transmitted signal $y_m(n)$ $(1 \leq m \leq K)$:

$$y_m(n) = X^T(n)W_m(n) \tag{10}$$

where $W_m(n)$ is the weight vector in segment m. The Eq. (11) is a normalized weight vector equation:

$$W_m(n+1) = W_m(n) + \frac{\mu}{\delta + X^T(n)X(n)} e_m(n)X^*(n) \tag{11}$$

where μ is step-size, δ is some small constant to avoid diverging the algorithm when the $X(n)$ is zero. * represents conjugate transpose operation. The real and imaginary part of the error signal for each segment can be expressed as:

$$e_{mR}(n) = y_{mR}(n) \times \left| y_{mR}(n)^2 - R_{2R} \right| \tag{12}$$

$$e_{mI}(n) = y_{mI}(n) \times \left| y_{mI}(n)^2 - R_{2I} \right| \tag{13}$$

We let $y_{mI}(n)$ and $y_{mR}(n)$ denote the real and imaginary parts of the output $y_m(n)$ signal respectively. In addition, we denote the real and imaginary parts of the input signal statistics using R_{2R} and R_{2I} respectively. Then the error signal formula of each segment is as follows:

$$e_m(n) = e_{mR}(n) + je_{mI}(n) \tag{14}$$

The mean square error (MSE) can be calculated by error signal of each segment.

$$MSE_m(n) = E\{|e_m(n)|^2\} = \frac{\sum_{i=1}^{n} e_m(i)^2}{n} \tag{15}$$

We define the accumulated squared error (ASE) for each segment as:

$$A_m(n) = \sum_{i=1}^{n} e_m(i)^2 \tag{16}$$

The tap-length of equalizer can be adjusted with the following length update algorithm.

$$A_L(n) = \sum_{i=1}^{n} \beta^{n-i} \left| e_L(n)^2 \right| \tag{17}$$

$$A_{L-1}(n) = \sum_{i=1}^{n} \beta^{n-i} \left| e_{L-1}(n)^2 \right| \tag{18}$$

If:

$$A_L(n) \leq \alpha_{up} A_{L-1}(n) \tag{19}$$

then increase one segment.

Else if:

$$A_L(n) \geq \alpha_{down} A_{L-1}(n) \tag{20}$$

remove one segment.

Form the tap-length algorithm, we can see that if $A_L(n)$ is much smaller than $A_{L-1}(n)$, then adding taps can obviously improve the system performance. If $A_L(n)$ is close to or even greater than $A_{L-1}(n)$, then the increase of taps has no effect on system performance or even deteriorates the performance of the system. Where $\beta(\beta \leq 1)$ is forgetting factor, its roles is to weight the importance of the previous segment and the current segment. α_{up} and α_{down} are parameters that change the frequency of the tap update. L indicates the current active segment.

4 Simulation Results

Simulation build underwater acoustic channel from BELLHOP model. The carrier frequency is 12 kHz. The distance between transmitter and receiver is 100 m, they all located at a depth of 10 m. Wave height is 0.2 m. The transmitted sequences are 1000 bits. The variable tap length method has an initial tap length of 1 and a tap length increment of 4, the transmitted signal is modulated by 4QAM.

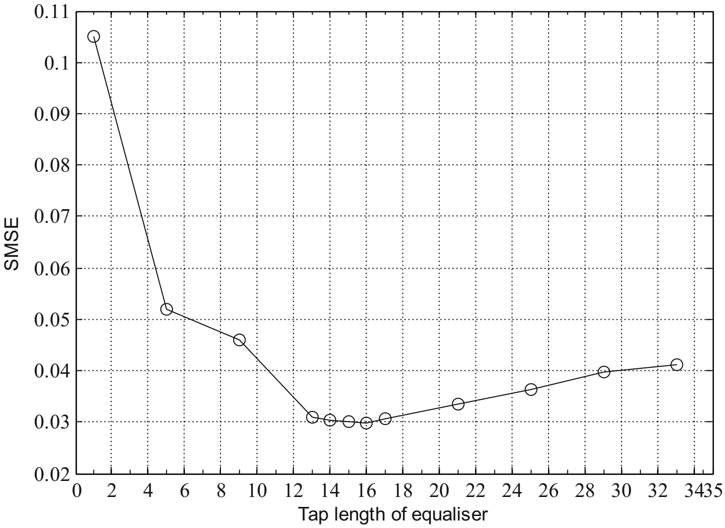

Fig. 3. The influence of the tap-length on SMSE

First, we simulate the tap length of the system performance, and determine the optimal tap length. Set the number of packets transmitted in the simulation to 300. Signal to noise ratio (SNR) is set to 15 dB. Herein, we define optimal tap length is the minimum tap length which close to the minimum steady mean square error (SMSE). The tap length is an important parameter affecting the SMSE. It can be seen from Fig. 3 that the system can achieve the minimum SMSE when the tap length is 16, but when

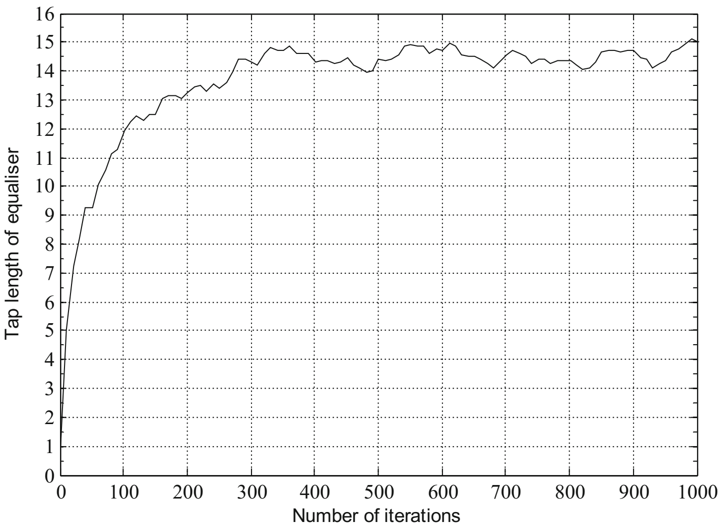

Fig. 4. Tap-length variation curve

the tap length is 13, it can not only achieve the near minimum SMSE, but also reduce the tap length, which reduces the arithmetic operation, so the optimal tap length is 14.

Next, we evaluated the tap length adjustment capability of the improved method. Set the channel conditions as in Fig. 3. Figure 4 shows the equalizer tap length adjustment curve. It can be seen from the Fig. 4 that the tap length can converge to 14. Therefore, the proposed method can converge to the optimal tap length.

To evaluate the performance of the proposed blind equalization algorithm, we compared its bit error rate (BER) performance and convergence performance with the fixed tap length-based MCMA (FT-MCMA). Simulation using Monte-Carlo method, the number of packets transmitted 400.

The BER performance of the proposed VT-MCMA and existing FT-MCMA is shown in Fig. 5. It can be seen from the figure that the proposed method can achieve better BER performance. This is because the FT-MCMA can not predict the optimal tap length, resulting in the new system performance is limited. However, the proposed method can converge to the optimal tap length through adaptive algorithm, so as to improve BER performance.

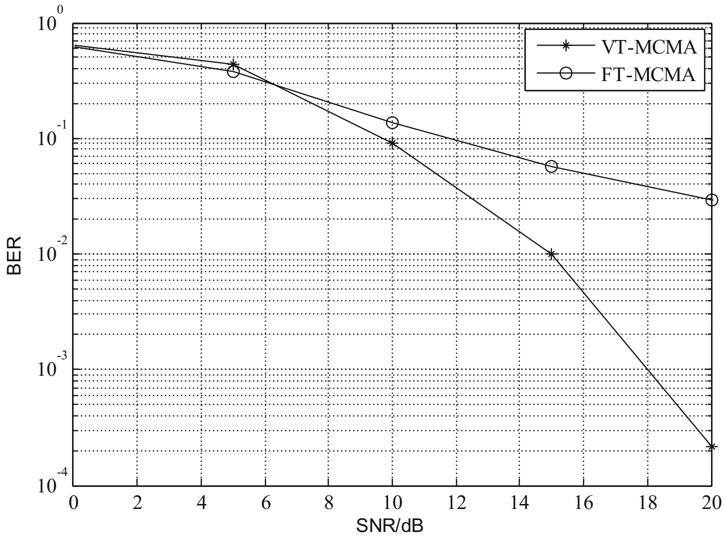

Fig. 5. Bit error rate performance comparison (SNR = 15 dB)

Finally, we compare the convergence performance, the simulation results shown in Fig. 6. It can be seen from the figure that VT-MCMA approaches the convergence speed of the traditional FT-MCMA. When reaching steady state, VT-MCMA achieves a smaller SMSE.

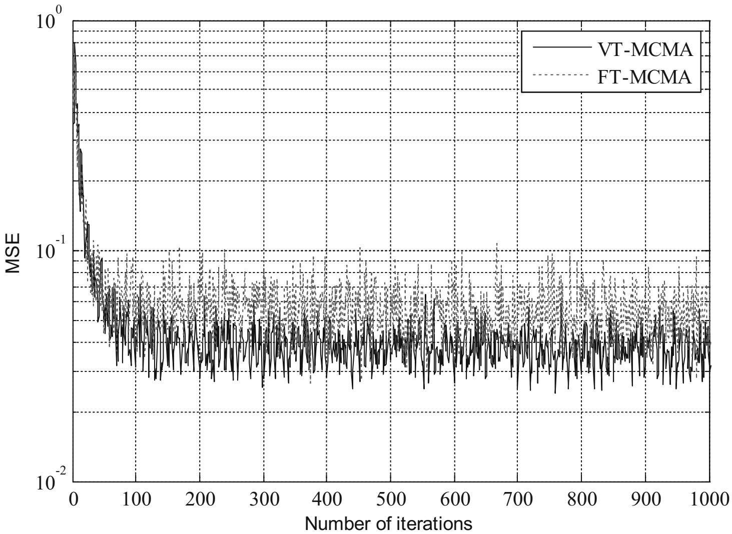

Fig. 6. Convergence curve comparison

5 Conclusion

The traditional FT-MCMA affected by the tap length, there will be a waste of computing and system performance constraints. Based on this improvement, this paper proposes VT-MCMA. It adopts the length updating algorithm, which can adaptively adjust to the optimal tap length according to the underwater acoustic channel environment. The feasibility of this method is proved through simulation, and its performance is compared with the existing FT-MCMA to verify the system performance is improved.

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