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A Novel Receiver Algorithm for Coherent Underwater Acoustic Communications

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***Abstract*—**In this paper, we proposed a novel receiver algorithm for coherent underwater acoustic communications. The proposed receiver is composed of three parts: (1) Doppler tracking and correction, (2) Time reversal channel estimation and combining, and (3) Joint iterative equalization and decoding (JIED). To reduce computational complexity and optimize the equalization algorithm, Time reversal (TR) channel estimation and combining is adopted to simplify multi-channel adaptive decision feedback equalizer (ADFE) into single channel ADFE without reducing the system performance. Simultaneously, the turbo theory is adopted to form joint iterative ADFE and convolutional decoder (JIED). In JIED scheme, the ADFE and decoder exchange soft information in an iterative manner, which can enhance the equalizer performance using decoding gain. The simulation results show that the proposed algorithm can reduce computational complexity and improve the performance of equalizer. Therefore, the performance of coherent underwater acoustic communications can be improved greatly.

***Keywords*—**Underwater acoustic communication, Time reversal (TR) combining, joint iterative equalization and decoding (JIED)

I. Introduction

THE underwater acoustic channel, especially shallow water, is characterized as doubly spread: (1) delay spread due to strong multi-path arrivals, and (2) Doppler spread due to channel variations as well as transmitter and/or receiver motion, leading to a channel with significant inter-symbol interference (ISI). At present, the effective approach to eliminate the ISI caused by multi-path propagation is that adaptive decision feedback equalizer (ADFE) integrates with spatial diversity and phase tracking. That is multi-channel adaptive decision feedback equalizer (MC-ADFE) which is applied in [1-2] represents a more general approach to spatial and temporal signal processing. In the MC-ADFE, the total number of adaptive feedforward taps increases with the number of channels. The implementation complexity of the MC-ADFE is high for a moderate to large number of line array channels, which often are needed to achieve reliable performance in dynamic ocean environments. In order to alleviate the complexity issue, channel estimation and combining can be incorporated into the DFE structure [3].

A relatively simple time reversal (TR) approach has been introduced in underwater acoustics. TR exploits spatial diversity to achieve spatial and temporal focusing in a complex environment such as underwater channels. Temporal focusing (pulse compression) mitigates the ISI while spatial focusing achieves a high signal to noise ratio (SNR) at the intended receiver. TR approach is combined with ADFE to conduct active and passive acoustic communications with lower implementation complexity [3-6].

In order to further improve the system performance, the convolutional code is adopted. In receiver, the turbo theory is applied to ADFE and convolutional decoder. And thus, the joint iterative equalization and decoding (JIED) is formed to enhance the performance of ADFE utilizing decoding gain provided by decoder such that the performance of communication system is improved.

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II. System describtion

Consider an underwater acoustic transmitter and receiver deployed in shallow water. At the source, a binary information sequence $x(n)$ is transformed into the baseband continuous wave $x(t)$. Then $x(t)$ is modulated onto the carrier frequency f_c and transmitted from a sound transducer. The receiver usually is equipped with multiple hydrophones. Let the total number of the hydrophones be K and $y^{(k)}(t)$ be the received baseband signal at the kth hydrophone. The analog waveform $y^{(k)}(t)$ is sampled at *T/2* fractional symbol interval [7]. The channel impulse response between the source and the kth hydrophone can be described by a combination of $h^{(k)}(t)$ and $\phi^{(k)}(t)$, a factor attributed to carrier frequency offsets between the transmitter and receivers and phase fluctuations from the environment. Therefore, simplified model for the purposes of this discussion becomes the following

$$y^{(k)}(n) = e^{j\phi^{(k)}(n)} h^{(k)}(n) * x(n) + w^{(k)}(n) \qquad (1)$$

Where, $w^{(k)}(n)$ represents the additive noise, * represents convolution operation.

To recover the transmitted symbols from $y^{(k)}(n)$, a novel receiver structure is proposed. As shown in Fig.1, the proposed receiver is composed of three parts: (1) Doppler tracking and correction is used to compensate carrier phase offset. At the beginning of a data packet, a preamble, is used to perform Doppler estimation, (2) Time reversal (TR) operation is used to perform channel estimation and multi-channel combining, and (3) using JIED to eliminate residual ISI and residual phase fluctuations.

The major parts of the receiver will be discussed in next section.

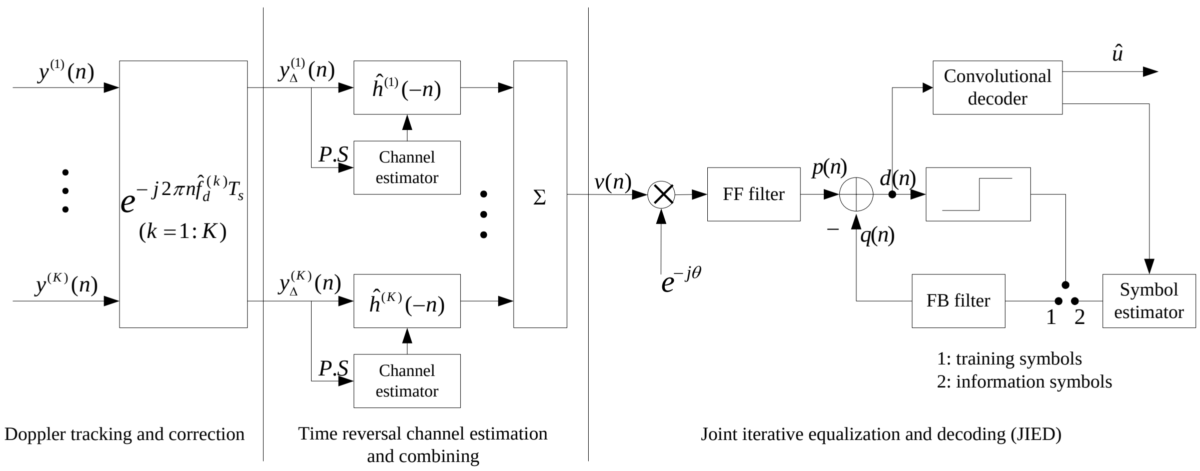

Fig.1 The proposed receiver is composed of three parts: (1) Doppler tracking and correction; (2) Time reversal channel estimation and combining; (3) Joint iterative equalization and decoding (JIED)

III. The Proposed Receiver Algorithm

As shown in Fig.1, the received signal performs Doppler correction firstly. And then, channel estimation and multi-channel combining is done using TR. A single channel ADFE with phase compensation is used to equalize the ISI in $v(n)$. The fast self-optimized LMS (FOLMS) algorithm [8] is used to update the equalizer tap weights. The residual carrier phase offset in $v(n)$ is compensated by a second order phase locked loop (PLL) embedded in the adaptive channel equalizer. The phase correction based on the PLL output is implemented at the input to the ADFE feedforward (FF) filter. In order to further improve the system performance, the turbo theory is applied to ADFE and convolutional decoder to form joint iterative equalization and decoding (JIED) scheme.

A. Doppler tracking and correction

In this paper, The Doppler estimate at the kth channel is obtained by the Chirp correlation. And then, the interpolator is used to correlation the Doppler [9].

B. Time reversal channel estimation and multichannel combining

According to TR theory, channel estimation can be obtained from Doppler corrected probe signal (P.S), therefore

$$\hat{h}^{(k)}(n) = p(n) * h^{(k)}(n) * p_\Delta^*(-n) \tag{2}$$

Where, a*b denotes convolution of a and b, a* denotes complex conjugation of a, $p_\Delta(t)$ is P.S after Doppler correction. In this paper, Gaussian signal is adopted as P.S.

Time reversal multichannel combining uses $h^{(k)}(-n)$ to match filter the Doppler corrected signals on each channel $y_\Delta^{(k)}(n)$ and then combines the results. The output of TR combining is

$$v(n) = \sum_{k=1}^{K} y_\Delta^{(k)}(n) * h^{(k)}(-n) = x(n) * q(n) + v(n) \tag{3}$$

Where, $v(n)$ is noise component,

$$v(n) = \sum_{k=1}^{K} (\hat{h}^{(k)}(-n)) * (w^{(k)}(n) e^{-j2\pi n \hat{f}_\Delta^{(k)} T_s}) \tag{4}$$

and $q(n)$ is the q function of the time reversed channel,

$$q(n) = \sum_{k=1}^{K} (\hat{h}^{(k)}(-n))^* * (h^{(k)}(n)) \tag{5}$$

C. Joint iterative equalization and decoding (JIED)

In the JIED scheme, the ADFE (including phase compensation) and decoder exchange soft information in an iterative manner. Specifically, at the output of the equalizer, the demapper computes soft information of coded bits based on the symbol estimation $\hat{d}(n)$. This soft information is delivered to the maximum a posteriori (MAP) decoder. In addition to providing the decoded output, the decoder also computes soft information on the coded bits, which is converted to soft estimates of the symbols. These soft symbol estimates are used to aid the operation of the ADFE and its adaptive weight update algorithm.

1. Single channel ADFE

According to minimum mean square error (MMSE) scheme, an error signal is used to update receiver parameters. After the carrier phase compensation, the input signal samples are represented in a matrix

$$\mathrm{V}(\mathrm{n}) = V(n) e^{-j\theta} \tag{6}$$

where ,

$$V(n) = \begin{bmatrix} v(nT + N_1 T/2) \\ \vdots \\ v(nT - N_2 T/2) \end{bmatrix} \tag{7}$$

The input of symbol decision device is given by

$$\hat{d}(n) = p(n) - q(n) = p(n) - \boldsymbol{b}^{\mathrm{H}}\tilde{\boldsymbol{d}}(n) \tag{8}$$

Where, $\mathrm{b}^{H} = [b_1 \cdots b_M]^*$ is the coefficients vector of the feedback part of the equalizer, and $\tilde{\mathrm{d}}(n) = \left[\tilde{d}(n-1)\cdots\tilde{d}(n-M)\right]^T$ denotes the vector of M previously detected symbols stored in the feedback filter. $p(n)$ represents the output of the linear part of the equalizer, it can be written as $p(n) = a^H V(n) e^{-\mathrm{j}\theta}$. $^{\mathrm{H}}$ denotes transpose conjugate, $^{\mathrm{T}}$ denotes transpose.

As shown in Fig.1, the residual phase offsets is compensated using PLL technology [7], therefore:

$$\hat{\theta}_{k+1}(n) = \theta_k(n) + K_{f_1}\phi_k(n) + K_{f_2}\sum_{m=1}^{n}\phi_k(m) \tag{9}$$

Where, $\Phi_k(n+1) = \mathrm{Im}\left\{p_k(n)\left[p_k(n) + e(n)\right]^*\right\}$, K_{f_1} and K_{f_2} are constants, $K_{f_2} \le K_{f_1}$.

Denote the overall equalizer vector as

$$\boldsymbol{w}(n) = \begin{bmatrix} \boldsymbol{a}(n) \\ -\boldsymbol{b}(n) \end{bmatrix} \tag{10}$$

The input data for this update is a composite vector

$$\boldsymbol{x}(n) = \begin{bmatrix} \mathrm{V}(n) \\ \tilde{\boldsymbol{d}}(n) \end{bmatrix} \tag{11}$$

A fast self-optimized LMS (FOLMS) algorithm [8] is used to update the equalizer vector $\mathrm{w}(n)$. But in [8], the formulations are conducted based on the line equalizer (LE). In this paper, we extent it to ADFE and consider the effect of carrier phase compensation. So, we can rewrite the FOLMS algorithm as follows

$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) + \mu(n)\boldsymbol{x}(n)e^*(n) \tag{12}$$

$$\mu(n+1) = \mu(n) + \alpha\,\mathrm{Re}\left[\boldsymbol{G}^{\mathrm{H}}(n)\boldsymbol{x}(n)e^*(n)\right] \tag{13}$$

$$g(n) = \boldsymbol{x}^{\mathrm{H}}(n)\boldsymbol{G}(n) \tag{14}$$

$$x'(n) = e^*(n)/\mu(n) \tag{15}$$

$$\xi(n) = x'(n) - g(n) \tag{16}$$

$$\boldsymbol{G}(n+1) = \boldsymbol{G}(n) + \mu(n)\boldsymbol{x}(n)\xi(n) \tag{17}$$

Where, $\mathrm{Re}(\cdot)$ denotes the real part of data, $\mathrm{w}(n)$ and $\mathrm{x}(n)$ are shown in Eq.(10) and Eq.(11) respectively; $\mu(n)$ is step-size factor for controlling the convergence ratio of the equalizer, which can be adaptively updated, $G(n)$ is a temporary variant for updating $\mu(n)$, α is constant.

2. Convolutional decoder

For MPSK, the corresponding $m = \log_2 M$ coded bits are mapped to an M-ary signal. The LLR value of *m*th coded bit of *k*th received symbol can be calculated as follows:

$$\Lambda_{k,m} = \ln\frac{\sum_{d\in \mathrm{B}_{m=1}} \exp\left(-\frac{\left(\hat{d}_k - d\right)^2}{2\sigma^2}\right)}{\sum_{d\in \mathrm{B}_{m=0}} \exp\left(-\frac{\left(\hat{d}_k - d\right)^2}{2\sigma^2}\right)} \tag{18}$$

where, $\mathrm{B} = \{d_1, d_2, \cdots, d_M\}$ denotes the finite alphabet used for MPSK signals, $\mathrm{B}_{m=1}$ and $B_{m=0}$ denote the sets of all possible symbol values, in which the *m*th coded bit is 1 and 0 respectively.

In this paper, log-map algorithm [10] is used to simplify calculation through transforming multiplication into addition.

3. Soft symbol estimation

As shown in Fig.1, the LLR values of coded bits, output from the decoder, are used to implement symbol estimation. And then, these symbols are fed back to feedback (FB) filter of ADFE to perform JIED scheme. So, the symbol estimation is key module.

There are two methods to estimate data symbol: hard estimation and soft estimation. Compare with hard estimation, soft estimation can void error symbols spread during the course of iterations. What's more, soft estimation can more sufficiently utilize decoding gain to update system performance. In this paper, soft method is adopted to estimate data symbols.

The soft symbol estimation can be obtained as follows:

$$\hat{d}_k = \sum_{d\in\mathrm{B}} d \prod_{i=1}^{m} p(b_i) \tag{19}$$

where, $\mathrm{B} = \{d_1, d_2, \cdots, d_M\}$ denotes the finite alphabet used for MPSK, b_i denotes the *i*th coded bit, $i = 1, 2, \cdots m$.

The probability distributions $P(b_i)$ of coded bits can be obtained from the corresponding LLR values $L(b_i)$, therefore

$$P(b_i) = \frac{e^{b_i \cdot L(b_i)}}{1 + e^{L(b_i)}} \tag{20}$$

IV. Simulation Results

In this section, we use simulation experiments to verify the performance of the proposed receiver algorithm. The system parameters of computer simulation are shown in Table 1. The convolutional code encoder is 4-state 1/2 code rate for QPSK modulation.

TABLE I
THE PARAMETERS OF COMPUTER SIMULATION

Carrier frequency	3.5 KHz
Data rate	2 Kbps
Array elements	6
Training symbols	200 sys

A. *Performance of the proposed receiver algorithm*

The channel impulse responses are shown in Fig.2.

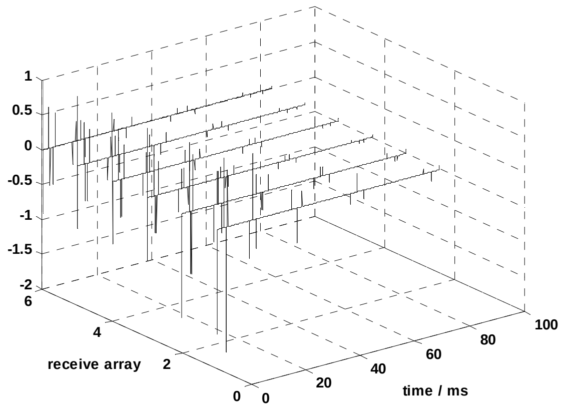

Fig.2 Channel impulse response (6 channels)

Fig.3 shows the BER curves of the proposed receiver algorithm with soft iteration for QPSK modulation. As show in Fig.3, the proposed receiver algorithm can effectively correct Doppler and enhance the equalizer performance using decoding gain such that the system performance is improved and the data transmission with lower BER can be obtained using small iterations.

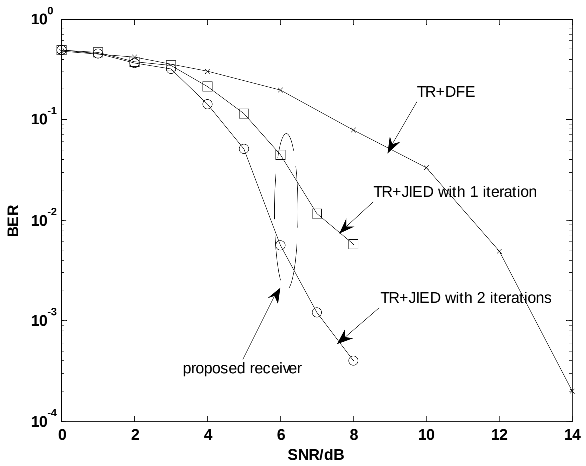

Fig. 3 Performance of the proposed receiver algorithm with soft iteration

B. The proposed receiver with dual mode JIED

The dual mode JIED is shown in Fig.4. It can be divided into two parts: (1) JIED, and (2) iterative decoding. As mentioned in Section III, the ADFE and decoder exchange soft information in an iterative manner in the JIED scheme. So, we can do decoding iteration before JIED. And thus, the accuracy of symbol estimation is further improved such that the equalizer performance can be improved greatly.

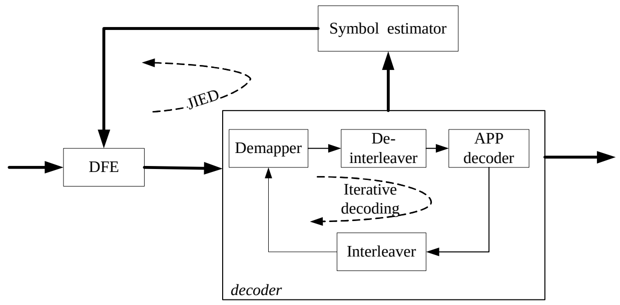

Fig. 4 dual mode JIED

As shown in Table 2，in dual mode JIED, the soft symbols estimation, based on the coded bits output from decoder, are more accurated. Therefore, the propagation errors can be futher reduced.

TABLE II
RECEIVER WITH DUAL MODE JIED

JIED number	1	
Decoder iteration number	0	1
3 dB	0.3491	0.3358
4 dB	0.2141	0.1712
5 dB	0.1151	0.0733
6 dB	0.0442	0.0158
7 dB	0.0117	0.0018
8 dB	0.0058	0.0006

V. CONCLUSION

In this paper, we proposed a novel receiver algorithm for coherent underwater acoustic communications. The proposed receiver is composed of three parts: (1) Doppler tracking and correction, (2) Time reversal channel estimation and combining, and (3) Joint iterative equalization and decoding (JIED). To reduce computational complexity and optimize the equalization algorithm, Time reversal (TR) channel estimation and combining is adopted to simplify multi-channel adaptive decision feedback equalizer (ADFE) into single channel ADFE without reducing the system performance. Simultaneously, the turbo theory is adopted to form joint iterative ADFE and convolutional decoder (JIED). In JIED scheme, the ADFE and decoder exchange soft information in an iterative manner, which can enhance the equalizer performance using decoding gain. What's more, the dual mode JIED scheme is proposed to obtain lower BER in order to meet the demands of higher system performance. The simulation results verify that the proposed receiver can obtain satisfied data transmission with the small iterations, especially the dual mode JIED.

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