EXPERIMENTAL ASSESSMENT OF SPARSE CHANNEL ESTIMATIONS FOR PASSIVE-PHASE CONJUGATION COMMUNICATIONS

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ABSTRACT

This paper investigates sparse channel estimations to improve the performance of passive-phase conjugation (PPC) communications. PPC processing achieves pulse compression for time delayed arrivals at the receiver. This property is used for underwater communications with a reduced computational load. The channel estimation is required by PPC. In multipath channels, the sparse channel estimations can be used to improve the communication performance. The dominant arrivals are estimated by matching pursuit (MP) processing for PPC processing and the remained arrivals are treated as zeros. Using experimental data collected in a range dependent acoustic channel, MP processing is assessed in improving the performance of PPC communications.

Index Terms— Passive-phase conjugation, match pursuit, underwater acoustic communication, decision feedback equalizer, sparse channel.

1. INTRODUCTION

Passive-phase conjugation (PPC) communications take advantage of pulse compression [1] to achieve underwater coherent communications with a reduced computational load for a receiver [2, 3]. Extended multipath in the channel results in an increased complexity for a channel equalizer to remove intersymbol interference (ISI) [4]. PPC processing requires the channel information which could be obtained by the channel probe or training symbols. There are some methods to obtain the channel estimations. For example, the least square (LS) method [5] and the adaptive method [6] have been used for time reversal communications. However, underwater acoustic channels are frequently characterized as sparse due to reflections and refractions, especially in deep water propagation scenarios.

In a sparse channel, dominant arrivals are estimated for adaptive channel equalization [7], and this property can be applied to PPC communications. With the estimated dominant arrivals, the remaining arrivals are tuned into zeros in PPC processing. The dominant arrivals contribute less time-varying effects and the zeros also reduce correlation noise level. Matching pursuit (MP) processing has been used in underwater communications in sparse channels [8-10]. In this paper, we have investigated sparse channel estimations for PPC communications. Since there is lack of a model which precisely predicts the acoustic channel for high frequency scenarios (e.g., 10-14 kHz), a recent sea experiment was conducted to assess MP processing for two receiver structures. One is the time reversal receiver structure realized by PPC plus one channel decision feedback equalizer (PPC-DFE) [3], and the other is joint PPC and multichannel DFE (PPC-McDFE) [11].

This paper is organized as follows. The MP is introduced in Section 2. In Section 3, two receiver structures are shown. Section 4 illustrates the experimental setup. The information about the experiment, sound speed profile and examples of channel response are shown. Section 5 presents communications results and analysis. Conclusions and discussions are given in Section 6.

2. MATCHING PURSUIT

MP processing [7, 10] estimates the dominant channel taps and corresponding tap coefficients in an iterative process. The received baseband signal V_k^n of the nth symbol interval at kth hydrophone can be written as

$$V_k^n = \sum_{l=0}^{L-1} H_k^l I_{n-l} + W_k^n \tag{1}$$

where H_k^l is the lth tap of channel impulse response of L taps and W_k^n is the bandwidth limited noise at the kth hydrophone. By combining M observed symbols, we can rewrite Equation (1) in the matrix format as

$$V_k = IH_k + W_k , \tag{2}$$

where

$$\begin{bmatrix} V_k^{n-M+1} \\ V_k^{n-M} \\ \vdots \\ V_k^n \end{bmatrix} = \begin{bmatrix} I_{n-M+1} & \cdots & I_{n-M-L+2} \\ \vdots & & \vdots \\ I_n & \cdots & I_{n-L+1} \end{bmatrix} \begin{bmatrix} H_k^0 \\ H_k^1 \\ \vdots \\ H_k^{L-1} \end{bmatrix} + \begin{bmatrix} W_k^{n-M+1} \\ W_k^{n-M} \\ \vdots \\ W_k^n \end{bmatrix}$$

and I is a Toeplitz matrix that consists of training symbols.

In MP processing, at the pth step, the l_pth column $(I)_{l_p}$ in the matrix I which is best aligned with the residual signal r_{p-1} is selected, where $r_0 = V_k$ at initial step. In practice, the l_p is given by

$$l_p = \arg\max_l \left\{ \left\| (I)_l^H r_{p-1} \right\| / \left\| (I)_l \right\| \right\}. \tag{3}$$

Correspondingly, the tap value $\hat{H}_k^{l_p}$ is estimated by

$$\hat{H}_k^{l_p} = \frac{(I)_{l_p}^H r_{p-1}}{\left\| (I)_{l_p} \right\|^2}, \tag{4}$$

and r_p is updated by

$$r_p = r_{p-1} - \frac{(I)_{l_p}^H r_{p-1}}{\left\| (I)_{l_p} \right\|^2} (I)_{l_p}. \tag{5}$$

This iteration is terminated until the preset P taps have been estimated. In practice, one column in I is probably selected more than once. To avoid this situation, we can exclude previous selected columns in the searching process as shown in Equation (3), or the tap value calculated in Equation (4) can be added to the value found at previous steps [8]. In this paper, we exclude previous selected column in the searching process.

3. PPC COMMUNICATIONS

The time reversal receiver structure is shown in Fig. 1. The channel responses are estimated by training symbols, where the LS and MP methods are used. The time reversed channel responses act as matching-filters for the received data signal. Carrier phase tracking is realized using a second order digital phase-locked loop (DPLL) [12]. One channel DFE removes residual ISI posterior to the refocusing, but it cannot eliminate ISI [13]. The recursive least square (RLS) algorithm is used to update the tap coefficients of DFE, as it has a fast convergence rate [14].

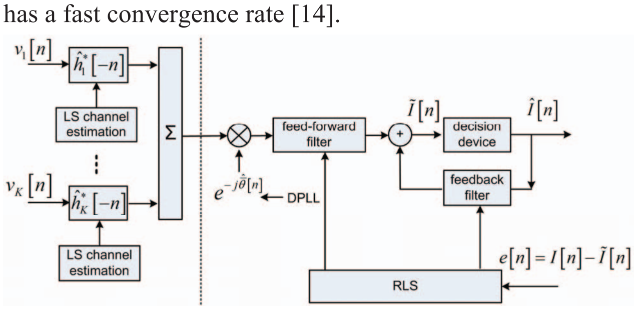


Fig. 1. The time reversal receiver structure PPC-DFE.

Fig. 2 shows the receiver structure presented by Zhang *et al* [11]. Posterior to pulse compression by PPC processing, adaptive diversity combining is performed by PPC-McDFE, where spatial diversity is exploited. The RLS algorithm jointly updates the tap coefficients of the K-channel DFE. Both structures are assessed by processing real data collected in a range of 3 km.

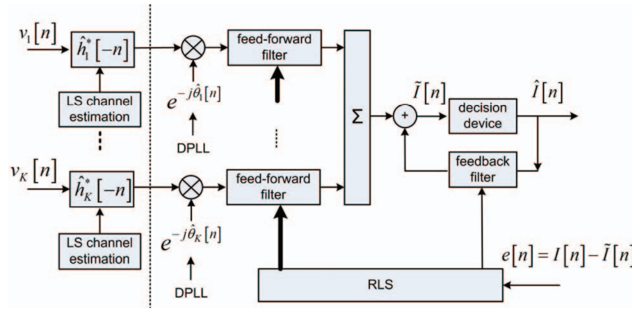


Fig. 2. The receiver structure of PPC-McDFE.

4. THE EXPERIMENT

The communication experiment was conducted on June 16, 2011, in Trondheim harbor in Norway shown in Fig. 3. The shallow region which is less than 20 m extends about 100 m offshore, and the sea depth varies from tens of meters near the island Munkholmen (in the center of the figure) to hundreds of meters. The red spot denotes the position of the transmitter in a distance of 3 km to a cross receiving array, which is denoted by the black spot. The receiving array consisted of 8 hydrophones, and it was near-shore deployed in a water depth less than 10 m. The array consisted of a vertical receiving array of 4 hydrophones (hydrophone No. 1-4) with 1 m element spacing and a horizontal receiving array of 4 hydrophones (hydrophone No. 5-8) with 1.5 m element spacing. Hydrophone No.1 was 0.5 m below the sea surface, and the depth of hydrophones No. 5-8 was 4.5 m. The transmitter used a hemispherical acoustic transducer suspended at a depth of 40 m from the NTNU research vessel R/V Gunnerus, whose dynamic positioning system was activated to reduce drifting during the trail.

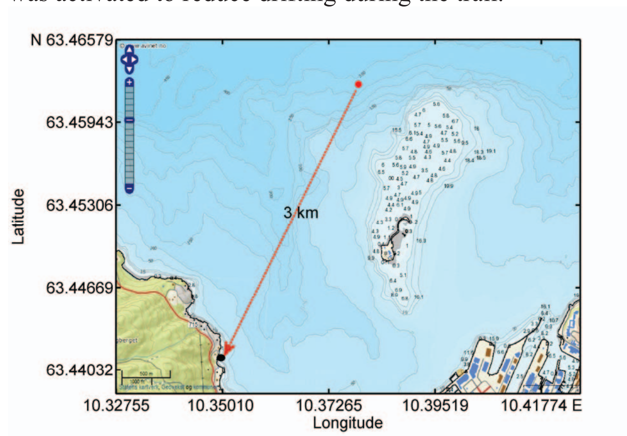


Fig. 3. Experimental area in Trondheim harbor.

In the experiment, the signals shown in Fig. 4 were repeatedly transmitted for about 50 minutes. The carrier frequency of the transmitted signals was 12 kHz. A 0.1 s linear frequency modulation (LFM) chirp with a Hanning window was used for coarse time synchronization. The data symbols were generated by binary phase shift keying (BPSK), and the data rates were 1 and 2 kilobits/s. A root-raised cosine filter was used for pulse shaping, where the roll-off coefficient was 1, and therefore the bandwidths for the communication signals were 2 and 4 kHz, respectively. The received waveforms were recorded with a sampling frequency of 96 kHz for off-line processing in the laboratory.

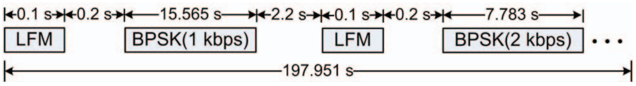


Fig. 4. The diagram of transmitted signals.

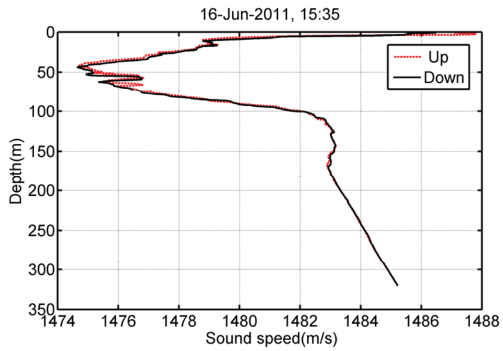


Fig. 5. Sound speed profile measured by the R/V Gunnerus.

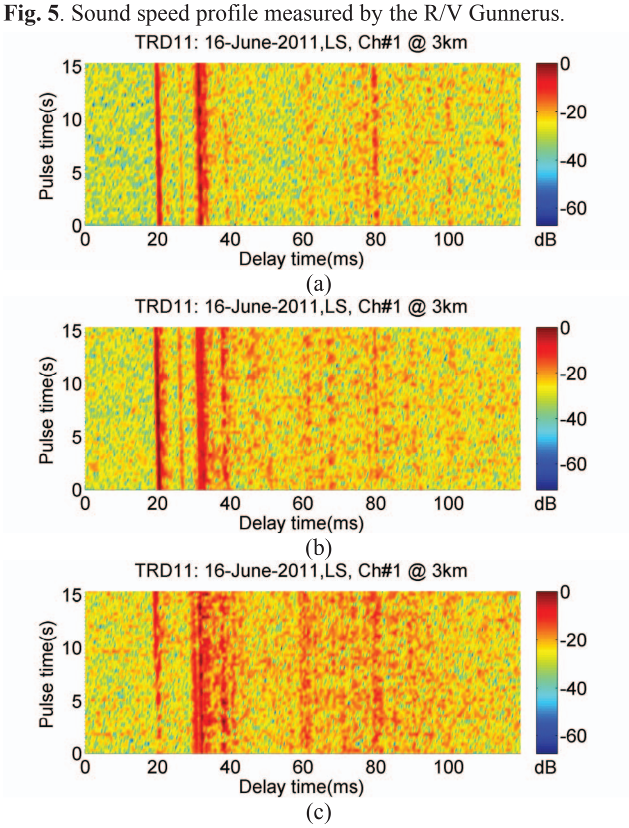


Fig. 6. Channel impulse response estimated by the LS method. (a) The 1st period. (b) The 9th period. (c) The 15th period.

Sound speed profile measured by R/V Gunnerus is shown in Fig. 5. The sea depth at the transmitter is about 320 m. It is indicated that there could be a sound channel at the depth of 45 m in the communication track. Fig. 6 shows examples of channel impulse response versus time, which are estimated by the LS method for the hydrophone No. 1. Within each period, the dominant arrivals are stable, and some arrivals change with time. In the large time scale, the multipath pattern changes with period. The examples indicate that sparse channel responses exist and can be estimated for PPC communications.

5. RESULTS AND ANALYSIS

The parameters listed in TABLE I for the two structures are fixed in processing the data. The feed-forward filter taps span 4 symbol intervals with an over sampling rate of N=4.

TABLE I. RECEIVER PARAMETERS

Parameters	Description	Value
F_s	The sampling frequency at the receiver	96 kHz
f_c	Carrier frequency	12 kHz
R	The symbol rate	1, 2 kbps
M	The number of observation symbols	100
N	Over sampling factor	4
N_{ff}	The number of feed-forward filter taps	16
N_{fb}	The number of feedback filter taps	2
N_t	The number of training symbols	72
λ	RLS forgetting factor	0.999
K	The number of channels	8
K_1	Proportional tracking constant in PLL	0.01
K_2	Integral tracking constant in PLL	0.001

An example of MP processing is shown in Fig. 7. Three taps of the impulse response are estimated, and the weak arrivals are tuned into zeros. Comparing with the LS method, it is flexible for MP processing to obtain the dominant arrival estimations instead of the total impulse response within a time interval. As shown in Fig. 6, the sparse impulse responses are stable within 15 s, and the sparse impulse response estimations are applied to PPC.

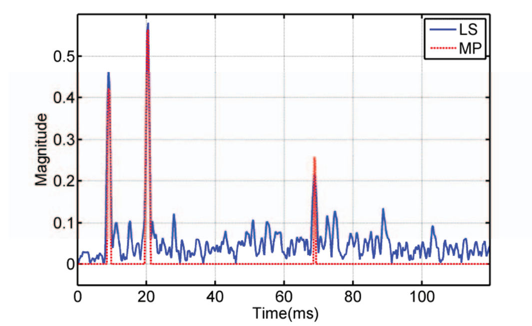
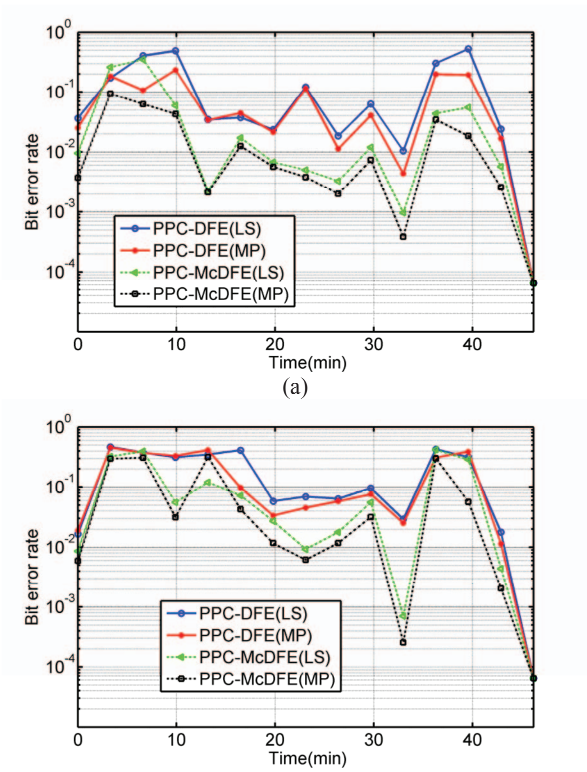


Fig. 7. MP processing compared with the LS method.

The recorded data of 8 receiving channels is processed. There are 15 periods in total. The results in terms of bit error rate (BER) are shown in Fig. 8, where 3 taps are estimated by MP processing within a time interval of 20 ms, and the LS method estimates the channel response within 20 ms. Time variant input signal-to-noise ratio (SNR) results in the variant output performance. In some periods, BERs are high due to low input SNRs. For both receiver structures, MP processing improves the performance in most periods. PPC-McDFE achieves superior performance at data rates of 1 and 2 kbps, respectively. For example, at a data rate of 2 kbps, PPC-McDFE reduces the BER in the 11th period from 2.9e-2 to 7.1e-4, where the LS method is used. Because of the highest input SNR of the 15th period, few errors occur in this period. The performance in terms of output SNR is shown in TABLE II. MP processing improves the performance for both structures, and PPC-McDFE improves the performance of PPC-DFE for both MP and LS channel estimation methods.



(a) (b)

Fig. 8. The receiver performance in terms of BER. (a) 1 kbps. (b) 2 kbps.

TABLE II. RECEIVER PERFORMANCE IN TERMS OF OUTPUT SNR

The structure	**1 kbps (dB)**		**2 kbps (dB)**	
	LS	**MP**	**LS**	**MP**
PPC-DFE	8.3	9.5	7.8	8.4
PPC-McDFE	11.0	11.8	10.4	10.8

6. CONCLUSIONS AND DISCUSSIONS

We have investigated sparse channel estimations for PPC communications in a time-varying sparse channel, where two receiver structures are assessed by processing real data collected in a range dependent underwater channel. The experimental assessment has demonstrated that MP processing improves the performance of the two receiver structures, as only dominant arrivals are estimated and the remaining arrivals are tuned into zeros for PPC processing. Since the parameters for MP processing are fixed in the data processing, the number of taps estimated by MP maybe not optimal. However, the improvement has been obtained most of the time within 45 minutes. In addition, PPC-McDFE has achieved superior performance, and it improves the performance of time reversal communications by adaptive combining to exploit spatial diversity.

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