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수중음향 통신을 위한 혼합형 송수신기에 관한 연구

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A Hybrid Transceiver for Underwater Acoustic Communication

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ABSTRACT: *In this paper, we propose a hybrid transceiver for underwater acoustic communication, which allows the system to reduce complexity and increase robustness in time variant underwater channel environments. It is designed in the digital domain except for amplifiers and implemented by using a multiple digital signal processors (DSPs) system. The digital modulation technique is quadrature phase shift keying (QPSK) and frame synchronization is an energy (non-coherent) detection scheme based on the quadrature receiver structure. DSP implementation is based on block data parallel architecture (BDPA).*
We show experimental results in the underwater anechoic basin at KRISO. The results indicate that the frame synchronization is performed without PLL. Also, we show that the adaptive equalizer can compensate frame synchronization error and the correction capability is dependent on the length of equalizer.

1. Introduction

The underwater acoustic channel can be characterized as a time-varying multi-path channel with fast fading and Doppler effect [5]. Especially, horizontal channels have multiple scattering by surface and bottom, which causes multi-path delay spread up to several tens of symbol intervals, while vertical channels have little time dispersion. For these reasons, frequency shift keying/differential phase shift keying (FSK/DPSK) modulation was commonly considered as robust techniques of digital underwater transmission, as an energy detection rather than phase coherent reception.

In recent years, many kinds of phase coherent receiver were introduced. These are based on the joint optimization of equalizer, phase-locked loop (PLL), and delay-locked loop (DLL) [1], [2]. [1] is the first work to demonstrate feasibility of phase coherent techniques. It optimized jointly carrier synchronization and fractionally spaced decision feedback equalizer (DFE) which parameters are adaptively adjusted using a combination of the recursive least squares (RLS) algorithm and second-order digital PLL, and applied to experimental data from several underwater acoustic channels.

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The acoustic transmitters and receivers can be implemented without the analog mixer in digital domain except for amplifiers because the carrier frequency of underwater acoustic telemetry is typically lower than 10MHz and there are many high-speed commercial analog to digital converters (ADCs) and digital signal processors (DSPs). In the underwater acoustic communications without network, system parameters (modulation technique, sampling rate, and carrier frequency, etc.) should be changed according to the channel conditions (range and depth). Almost all system parameters such as the modulation technique, sampling rate, and carrier frequency, etc., can be easily changed by simple program modifications in the all-digital implementation.

These facts induced all-digital implementation of acoustic telemetry [3], [4]. A digital acoustic image transmission system between autonomous underwater vehicle (AUV) and mother ship was proposed [3]. The modulation techniques were phase shift keying (PSK) and quadrature amplitude modulation (QAM). Its disadvantages are the complicated synchronization processes, the expensive optical repeater, and limitation to vertical channels. It is under test and should be verified in the real underwater channels. DPSK modulation method was used in [4].

In this paper, we present a hybrid transceiver for underwater acoustic communication in the digital domain except for amplifiers and it is implemented by using a multiple DSPs system. Also we describe a frame synchronization method using 13-symbol Barker sequence. It is based on the quadrature receiver structure and has no need of a PLL . To verify the performances of the designed transceiver, we show experimental results in the underwater anechoic basin at KRISO.

2. Hybrid Scheme for Underwater Acoustic Transceiver

2.1 Transmitter

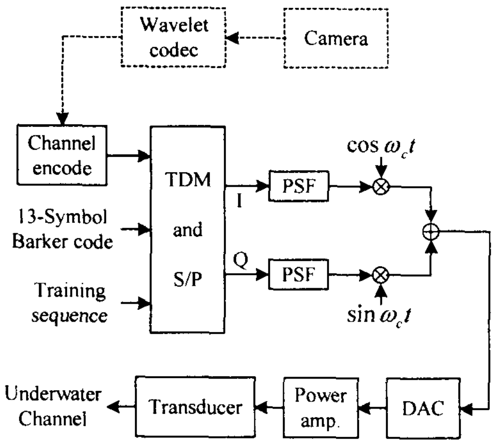


Fig. 1 Designed transmitter structure

The block diagram of the transmitter is shown in Fig. 1. The signal frame consists of 13-symbol Barker codes, training sequence and data, which are time multiplexed as shown in Fig. 2. Frame synchronization is accomplished by matched filtering to Barker code. Transmission gaps exist between Barker code and training sequence and between Data and Barker code for successful matched filtering in the frame synchronization processes. Training sequences are used for an adaptive equalizer and beamformer. The signal frame is organized in blocks so as to provide periodic frame synchronization and retraining for the adaptive signal processing techniques, which allows to meet effectively with time varying underwater channel.

Barker code	Tx. gap	Training sequence	Data	Tx. gap

Fig. 2 Signal frame

The digital modulation technique is quadrature phase shift keying (QPSK). The channel encoded data, Barker code, and training sequences are time multiplexed through the time division multiplexing (TDM) module and mapped to I and Q branches by the serial-to-parallel (S/P) block. The I and Q branches are then filtered by the pulse shaping filter (PSF) and QPSK modulated. The PSF is a raised cosine filter and truncated to satisfy following requirement on modulation accuracy

$$\sqrt{\frac{\int_{-\infty}^{\infty}\{h(t)-\bar{h}(t)\}^2 dt}{\int_{-\infty}^{\infty} h^2(t)dt}} < \alpha$$

where the threshold value α can be changed according to the given conditions.

The maximum carrier frequency is bounded by the conversion time for a "full scale" signal jump of the digital to analog converter (DAC). It is also restricted by the conversion time of the ADC in the receiver. The wavelet codec and camera indicated by dashed box are need for image application.

2.2 Receiver

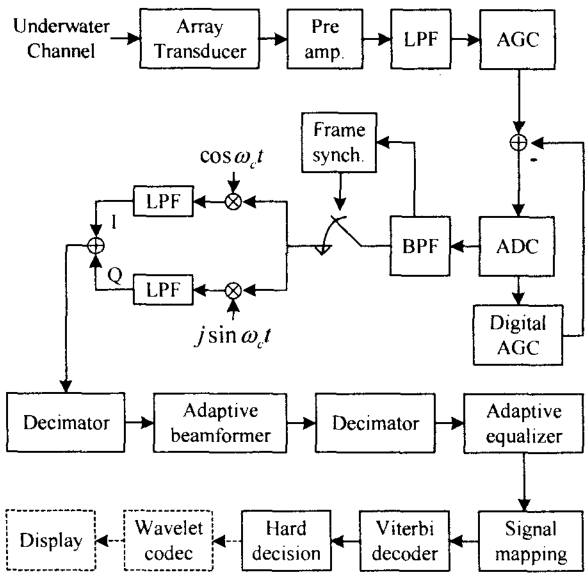


Fig. 3 Designed receiver structure

The block diagram of the all-digital receiver is shown in Fig. 3. The received signal, after being passed through the underwater acoustic channel, is filterd and sampled in the A/D converter. The low-pass filter (LPF) cuts off the high frequency components of the received signals. The sampling rate of the ADC can be determined as follows. 1) It should be larger than the Nyquist rate from the Nyquist theorem. 2) It should be a multiple of $R \times N$ for easy implementation of the decimator, where R is the data rate and N (oversampling rate) is the number of sample for one symbol period.

The quantization noise and saturation noise caused by ADC can be compensated by the digital automatic gain control (AGC) module. Following the ADC there is the band pass filter (BPF) which extracts the only interested frequency band within bandwidth.

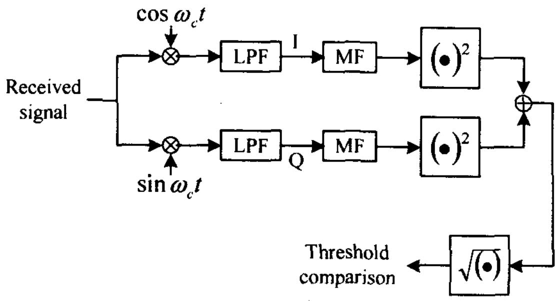


Fig. 4 Frame synchronization scheme

The frame synchronization process is performed as shown in Fig. 4. Note that the "Received signal" in Fig. 4 is the digital signal sampled and filterd by BPF. The channel probe for the frame synchronization is a 13-symbol Barker code (See the signal frame of Fig. 2.). The term "13-symbol Barker code" means that the same 13-bit Barker code is inserted into the both I and Q channels data at transmitter. Low-pass filtered I-channel and Q-channel outputs are correlated with 13-bit Barker sequence, and then they are squared and added to ignore the carrier phase error. Therefore, the frame synchronization process is independent on the carrier synchronization, so we have no need of PLL. It will be demonstrated by experiments.

The adaptive beamfomer is the class of wide-band beamformer which has tapped delay lines behind each sensor [9], [10]. The adaptive wide-band beamformer parameters, e.g., inter-tap delay and the number of taps per element, for the given fractional bandwidth can be determined from the previous results [6], [7], [8]. Note that the inter-tap delay can be easily modified by the programmable sampling rate. The weight updating method is the RLS algorithm.

The adaptive equalizer is a T/4-spaced transversal filter and the weight update algorithm is also RLS, where T is the symbol duration.

The Viterbi decoder is used as the maximum likelihood convolutional decoding algorithm. It operates on the 8-level soft decisions, which results in a performance improvement of approximately 2 dB in required signal to noise ratio (SNR) compared to hard decisions and a loss of approximately 0.2 dB compared to infinitely fine quantization. The decoding delay is 5K, where K is the constraint length of the convolutional encoder [12].

3. Experimental Results

3.1 Experimental Set-up

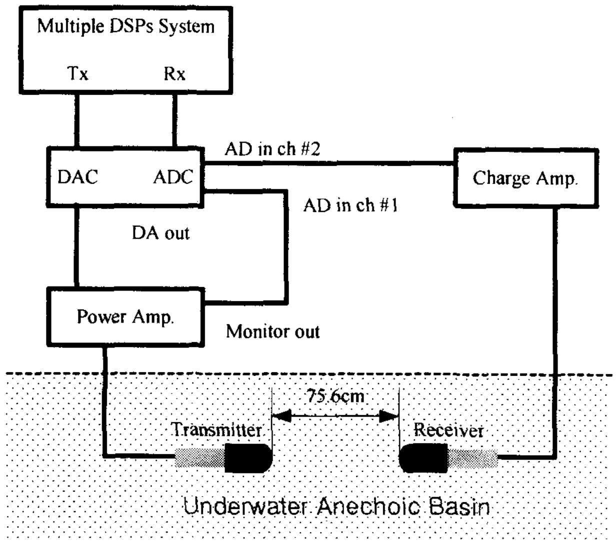


Fig. 5 Equipment set-up for transmitting and receiving acoustic signals in the anechoic tank using two omni-directional transducers

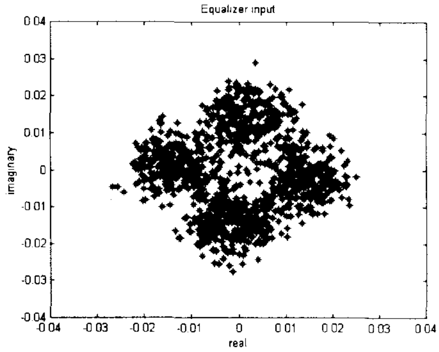
A simple transmitting and receiving experiment was performed in the anechoic tank at KRISO. The anechoic tank has a rectangular shape and its size is 914cm long, 282cm wide, and 500cm deep. It has cone-type anechoic lining at the four walls and bottom. It was designed to absorb acoustic waves frequency of which is higher than about 3kHz [11].

During the experiment two omni-directional acoustic transducers (one transmitter and one receiver) were used so that there was two communication paths: direct and surface-reflected ones. The transducers were located at 83.6cm deep and the distance between them was 75.6cm (See Fig. 6). One frame data (37200 points) was generated by the QPSK modulation method with the center frequency of 37kHz. The generation frequency at the AD converter was 149.25kHz (period: 6.7ms) and a power amplifier was used for amplifying transmitting signals. The received data was also amplified and filtered (100kHz LPF) via a charge amplifier and sampled at the same frequency with generation one, or 149.25kHz.

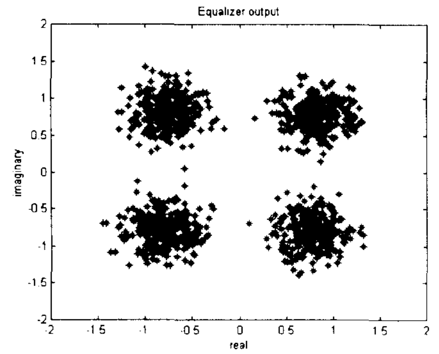
The important system parameters are summarized as follows. The data rate is 9,600- bps (4.8 KHz bandwidth) and the duration of one frame is 250ms (1200 symbols or 2400 bits). The training sequence is 50-bit pseudo-noise (PN) sequences.

3.1 Experimental Results & Discussion

Fig. 6 shows signal constellations of the equalizer input and output, where the input has unknown carrier phase mismatch and the equalizer has 15 taps. Although SNR of unsynchronized input signal is severely low (commonly 30 dB lower than the synchronized one) we can obtain the same result with perfect carrier phase synchronization, i.e., SNR of the equalizer output is always same regardless of carrier phase. It comes from the fact that the frame synchronization process is not dependent on carrier phase error.



(a) Input constellation



(b) Output constellation

Fig. 6 Signal constellations

Fig. 7 shows the equalizer output SNR according to the number of tap. If the taps exceed 14 (3.5 symbols), the output SNR is slightly increased. It can be explained as follows. Because of the sound absorption at four walls and bottom, there are only two multi-paths in the communication channel. One is the path that goes from the transmitter to the receiver directly and the other is the one that is reflected at the free surface once. In this experiment, the delay of the surface reflection is 0.73ms and it corresponds to about 3.5 symbols duration [11].

To investigate the effect of frame synchronization error compensation of the equalizer, we observe the equalizer output SNR as shown in Fig. 9. In the figure, (a) and (b) are the cases of 15 taps and 40 taps, respectively. The value of 31 is equal to one symbol in the horizontal axis. When the frame synchronization error is lower than the time equivalent to the equalizer length the decreasing rate of SNR is slow. But, if it becomes larger than the threshold value SNR decreases very fast. In the case of (a), SNR varies within 1 dB when it is smaller than two symbols and it can be compensated up to 7 symbols without 3 dB loss of SNR in the case of (b). Hence, there is trade-off between frame synchronization error compensation and equalizer length. The equalizer has more taps, it corrects frame synchronization error more stably and widely.

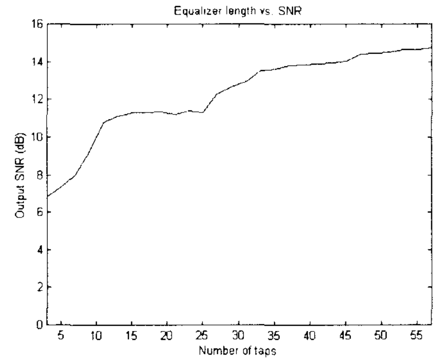
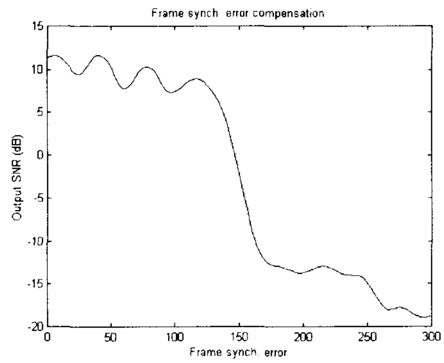


Fig. 7 Equalizer output SNR vs. equalizer length



(a) 15 taps

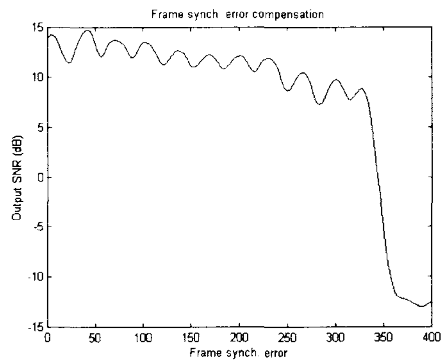


Fig. 8 Frame synchronization error compensation of the equalizer

4. Conclusions

We have designed a hybrid all-digital transceiver for underwater acoustic communication in the digital domain except for amplifiers and it was implemented by using the multiple DSPs system. The frame synchronization process had no need of PLL.

We have performed experiments using the designed systems in the underwater anechoic basin at KRISO. We have proved that the frame synchronization is well done without PLL. Also, it was found that the adaptive equalizer could compensate frame synchronization error and the correction capability was dependent on the length of equalizer.

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