A New Method for Frame Synchronization in Acoustic Communication

Zheng Guo
Key Laboratory of Information Technology for AUVs,
Institute of Acoustics, Chinese Academy of Sciences
University of Chinese Academy of Sciences
Beijing, China, 100190
guozheng@mail.ioa.ac.cn

Shefeng Yan, Lijun Xu
Key Laboratory of Information Technology for AUVs,
Institute of Acoustics, Chinese Academy of Sciences
Beijing, China, 100190
sfyan@mail.ioa.ac.cn, idle_xu@mail.ioa.ac.cn

***Abstract*—This work considers the problem of false frame synchronization for underwater acoustic communication (UWAC), and a new frame synchronization approach based on Linear Frequency Modulation (LFM) signal parameters estimation is proposed. Two parameters estimation techniques including Fractional Fourier Transform (FRFT) estimation and Maximum Likelihood (ML) estimation have been presented in this paper. Traditional frame synchronization methods are based on a correlator, the reference signal used in the correlator has the same parameters as the transmitted signal; while in the signal propagating process, the received signal suffers severe distortion, so it is obvious not the optimal way for frame synchronization. By avoiding a complex design of synchronization signal waveform and prolonging the synchronization signal time, this method we proposed can locate the frames of the received signal effectively; therefore, communication rate will not be reduced. And a simplification of the algorithm has been derived to save computational cost. Simulation and experimental results to show the performance of this method are also addressed in this paper.**

Keywords—frame synchronization; underwater acoustic communication; parameters estimation; ML; FRFT

I. Introduction

Frame Synchronization is critical in communication system, since data streams are followed by a unique frame format, only with an accurate frame synchronization can we recover information from the modulated signal.

In order to get a robust frame synchronization, practical methods are focused on the design of synchronization signal waveform (SSW), prolonging the synchronization signal time or increasing the sampling rate, etc. The basic theory of frame synchronization is presented in [1] and design of frame markers is discussed, and five attributes of a good frame synchronization system have also been discussed. Lots of researches have been done on synchronization to meet the growing needs of reliable underwater acoustic communication. A multi-correlator is used to avoid frames losses instead of a mono-correlator in [2]; In [3], LFM and HFM waveform used as synchronization signal have been analyzed in terms of their performance in underwater acoustic (UWA) communication under different signal-to-noise ratio and different relative velocity, but it still needs two SSWs to do Doppler compensation; A method [4] which is focused on the design of the preamble in a frame is proposed, it contains two different parts known as wake-up signal and calibration signal, but communication rate can be reduced in this case, the same problem exists in [5] which mainly discusses synchronization signal structure.

As for the SSW detection algorithm, Massey [6] derived the maximum likelihood (ML) rule for detecting periodically embedded synchronization patterns for the AWGN channel, and the optimal detection metric is derived under the hypothesis testing theory and the likelihood ratio test in [7]. These techniques inevitably either reduce the communication rate or increase the design complexity of the synchronization signal.

Traditional methods of frame synchronization are based on a correlator, and the synchronization signal is assumed to have been received when the correlation coefficient between the received signal and the local signal is higher than the threshold. Here we propose a new method for frame synchronization based on LFM signal parameters estimation. This frame synchronization technique takes LFM signal as the synchronization signal, a complex LFM signal is given by

$$s(t) = \begin{cases} Ae^{j(2\pi f_0 t + \pi k t^2)} & 0 \leqslant t \leqslant T \\ 0 & else \end{cases} \tag{1}$$

where k is the frequency rate (chirp factor) and f_0 is the initial frequency. And the reasons we choose LFM signal as SSW include:

- Good autocorrelation and spectral characteristics.
- LFM signals have a knife-edge or ridge ambiguity function.
- LFM signals also have good peak-to-average power ratio (PAPR) characteristics.

Since our objective is to locate the synchronization signal from observation, we need to detect the arrival of LFM signal first. So we come to parameters estimation techniques, useful tools include ML estimation, dechirp method and some time-frequency analyze tools like FRFT and Radon-Wigner

This work was supported in part by the National Natural Science Foundation of China under Grant 61431020, S. Yan is the corresponding author.

978-1-4673-9724-7/16/$31.00 ©2016 IEEE

transform, etc. In this paper, we mainly discuss ML and FRFT parameters estimation used for frame synchronization.

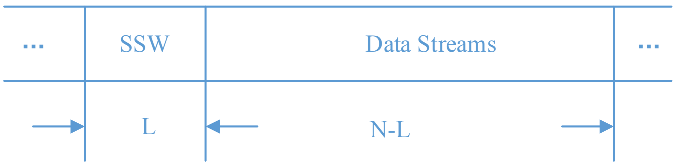

Fig. 1. Frame Structure of the communication system

The method we proposed here can reduce the false synchronization probability effectively, without sacrificing the communication rate. The factors that lead to a relatively high cross correlation between the local reference signal and the received interference signal include short synchronization signal duration, spectrum overlap with modulated signal, and the complex underwater acoustic channel. False frame synchronization caused by it turns out to reduce the reliability of communication. With techniques like FRFT and ML, we can distinguish the LFM signal and the cosine signal, thus false synchronization can be avoided. This method shows its superiority compared to the traditional schemes.

The rest of this paper is arranged as follows: Section II introduces the communication signal model used in this paper and gives the detailed approach of frame synchronization, Section III gives the simulation results and experimental results, and Section IV summarizes the paper.

II. Synchronization Based on Parameters Estimation

The model of frame structure is shown in Fig. 1, each frame consists of a synchronization sequence of L consecutive symbols followed by $(N-L)$ data symbols. The data streams we discussed here can be phase shift keying (PSK), frequency shift keying (FSK) and amplitude shift keying (ASK) modulation.

The received synchronization signal in the AWGN channel is expressed as

$$r(t) = s(t) + w(t) \tag{2}$$

Without considering the interference of multipaths, the synchronizer needs to choose from two possible hypotheses according to the hypothesis testing theory

$$\begin{aligned} H_0 &: r(t) = s(t) + w(t) \\ H_1 &: r(t) = y(t) + w(t) \end{aligned} \tag{3}$$

where $y(t)$ can be the modulated signal or other interference signal, $w(t)$ denotes the white Gaussian noise process, and for a certain time t_0: $w(t_0) \sim N(0, N_0/2)$, where N_0 is the one-sided noise spectral density.

So the problem of synchronization becomes to detect $s(t)$ from observation. But underwater acoustic communication channel exhibits both time and frequency dispersive fading which is caused by the Doppler and delay spread of the medium and by the transmitter and receiver motion, so $s(t)$ often suffers severe distortion from the original transmitted signal. Under hypothesis of H_0, $r(t)$ is described by

$$\begin{aligned} r(t) &= Ae^{j[2\pi f_0(1+\delta)(t-\tau)+\pi k(1+\delta)^2(t-\tau)^2]} + w'(t) \\ &\overset{\tau=0}{=} Ae^{j[2\pi f_0(1+\delta)t+\pi k(1+\delta)^2 t^2]} + w'(t) \\ &= Ae^{j(2\pi f'_0 t+\pi k' t^2)} + w'(t) \end{aligned} \tag{4}$$

where $\delta = v/c$, which represents Doppler factor, and v represents the relative moving velocity between the transmitter and the receiver, c is the underwater sound propagation speed, and $f'_0 = f_0(1+\delta)$, $k' = k(1+\delta)^2$. So the instantaneous frequency of $r(t)$ is expressed as

$$\begin{aligned} f(t) &= f_0(1+\delta) + k(1+\delta)^2 t \\ &= f'_0 + k't \end{aligned} \tag{5}$$

In the presence of Doppler spread, the received LFM signal and the transmitted LFM signal mismatch[4], resulting in the decrease of peak detection capacity and timing accuracy of synchronization, as shown in Fig. 2.

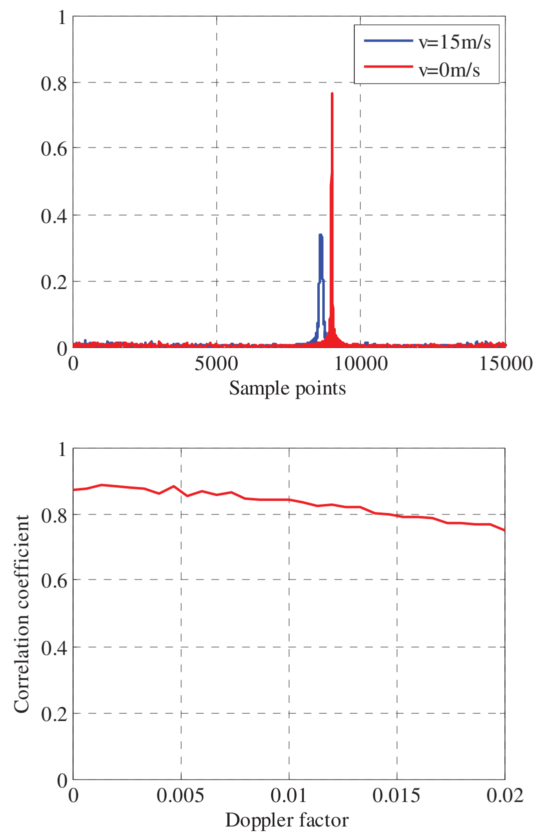

Fig. 2. Top: Correlation of LFM signal. Bottom: Correlation coefficient of LFM signal with Doppler

Compared to the output of LFM signal under relative velocity of 15m/s, the sharper red line in (a) shows the

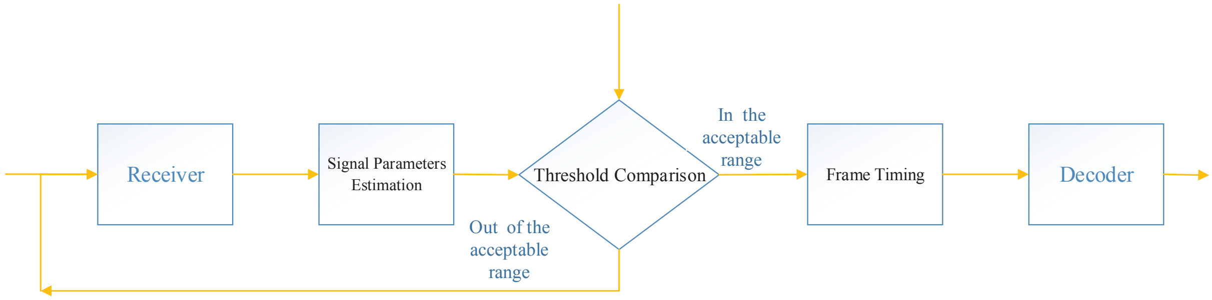

Fig. 3. Frame Synchronization Process

output from matched filter (MF) of LFM signal without Doppler; and (b) shows the correlation coefficient of LFM signal under different Doppler factor. From the result of the simulation, it is clear to see that the timing and detection capability of regular MF get worse under Doppler interference. And the faster the relative moving velocity is, the poorer the detection capacity will be under a certain threshold.

This frame synchronization scheme we proposed here takes two steps. The first step is to detect the synchronization signal. LFM signal parameters estimation technique is applied to detect the LFM signal, and techniques for parameters estimation discussed here include FRFT and ML estimation, so the initial frequency $\widehat{f_0}$ and frequency rate $\widehat{k}$ of the right LFM signal can be estimated, then a comparison with the threshold (f_0, k) is made. These two parameters we estimated should be in the right range if the right synchronization signal has arrived. Once the synchronization signal has been detected, a Matched Filter will be applied to get the accurate synchronization time. The detailed process of frame synchronization is shown in Fig. 3.

A. *FRFT Estimation*

Namias[8] first introduced the mathematical definition of FRFT in 1980, now its application in time-frequency analysis has attracted more and more attention in signal processing society.

The FRFT of signal $x(t)$ is defined as

$$X_\alpha = F^p[x(t)] = \int_{-\infty}^{+\infty} x(t) K_\alpha(t,u)dt \tag{6}$$

Where p is the order of the FRFT, $\alpha = p\pi/2$ which stands for the angle of rotation[9], and $K_\alpha(t,u)$ is the kernel of the transform

$$K_\alpha(t,u) = \begin{cases} \sqrt{\frac{1-j\cot\alpha}{2\pi}} exp(j\frac{t^2+u^2}{2}\cot\alpha - tu\csc\alpha) & \alpha \neq n\pi \\ \delta(t-u) & \alpha = 2n\pi \\ \delta(t+u) & \alpha = (2n\pm1)\pi \end{cases} \tag{7}$$

The representation of a signal in the Fractional Fourier domain contains the information in both time and frequency domains of the signal. The concentration of the LFM signal energy can be reached if we select a proper rotation angle, while cosine signal energy is concentrated in the Fourier Transform domain. This property can be used to detect LFM signals under the interference of cosine signal. Then we only need to find the peak in the signal's FRFT domain, and then the signal's parameters $(\widehat{f_0}, \widehat{k})$ can be estimated. So we can recognize the right synchronization signal by a comparison with threshold (f_0, k). An acceptable range of the two parameters need to be set, and the right synchronization signal will be determined if the estimated parameters are in this range. Simulation results are shown in Fig. 4.

Result shows that FRFT can detect LFM signal and estimate its parameters effectively, but for the moment, the fast implementation of the discrete Fractional Fourier Transform is yet unknown [10], in order to find a method which can be used in real time communication, we use ML Estimation instead.

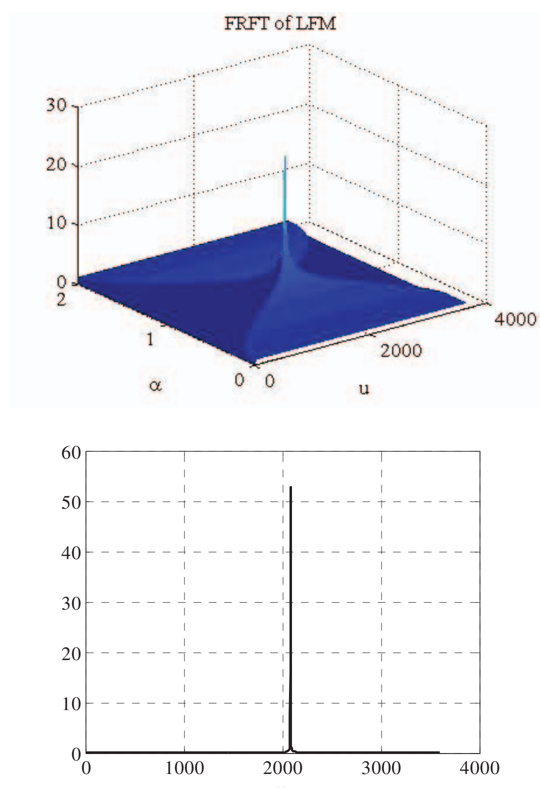

Fig. 4. Top: FRFT transform of LFM signal. Bottom: The profile of the peak of LFM estimation

B. ML Estimation

The ML technique has been used to estimate parameters of LFM signal for a long time, Shimon and Boaz proposed an estimation algorithm based on ML estimation in [11], and analyzed its asymptotic performance, they proved that the performance of the method is close to the Cramer-Rao lower bound when SNR is 0 dB and above. Massey[6] has shown that the performance of ML rule used for synchronization pattern detection is substantially better than that of the correlation rule. And Nielsen [12] found that in condition of high SNRs, ML rule approximation provides the same performance as the optimal rule while it is much simpler to implement.

The basic principle of ML estimation is given by

$$(\widehat{f}_0, \widehat{k}) = arg \quad \max_{f_0,k} \left| \sum_{n=0}^{L-1} r(n) e^{-j(2\pi f_0 n + \pi k n^2)} \right| \tag{8}$$

The algorithm is accompanied by two one-dimensional searches for the maxima. The estimated parameters $\widehat{f}_0$ and $\widehat{k}$ can be used to do Doppler compensation of the reference signal, so a much accurate time synchronization can be reached by the MF. Although parameters of synchronization signal (f_0, k) is priori known to both the transmitter and the receiver in condition of communication, it still needs to conduct two one-dimensional searches to find the peak, computational cost is intolerable in real time communication. Here we propose a more simple way to accomplish this estimation, which doesn't take much calculation in searching for the maxima.

Under hypothesis of H_0, the output of ML estimator $L(f_0, k)$ is expressed as

$$\begin{aligned} L(f_0,k) &= \left| \sum_{n=0}^{L-1} r(n) e^{-j(2\pi f_0 n + \pi k n^2)} \right| \\ &= \left| \sum_{n=0}^{L-1} (A e^{j(2\pi f_0' n + \pi k' n^2)} + w'(n)) * e^{-j(2\pi f_0 n + \pi k n^2)} \right| \\ &= \left| \sum_{n=0}^{L-1} (A e^{j(2\pi f_0' n + \pi k' n^2)} * e^{-j(2\pi f_0 n + \pi k n^2)} + w'(n) * e^{-j(2\pi f_0 n + \pi k n^2)}) \right| \\ &= \left| A * D(f_0,k) + \sum_{n=0}^{L-1} w'(n) * e^{-j(2\pi f_0 n + \pi k n^2)} \right| \end{aligned} \tag{9}$$

where $D(f_0, k)$ is defined as

$$\begin{aligned} D(f_0,k) &= \sum_{n=0}^{L-1} e^{j(2\pi f_0' n + \pi k' n^2)} * e^{-j(2\pi f_0 n + \pi k n^2)} \\ &= \begin{cases} N & only\ if\ f_0 = f_0', k = k' \\ \sum_{n=0}^{L-1} e^{j(2\pi (f_0' - f_0) n + \pi (k' - k) n^2)} < N & else \end{cases} \end{aligned} \tag{10}$$

Since we have already known the relationship between f_0' and k', we treat the case when several rays with the same Doppler shifts of the received signal, so we get

$$\begin{cases} f_0' = f_0(1+\delta) \\ k' = k(1+\delta)^2 \end{cases} \tag{11}$$

Then we just need to search for the maxima in the δ domain, this greatly reduced the time in the maxima searching process. This method can be implemented at less computational cost.

The proposed method requires no strict carrier synchronization, the algorithm can detect the SSW even if carrier synchronization deviation occurs, and using the Doppler estimation result δ, the center frequency of each received signal can be adjusted. Superiority of this method is obvious when the Doppler shift is subject to rapid variations as the case like AUV communications.

C. Frame Timing

Once the synchronization signal has been recognized, the frame timing process begins. Time-scale of the received signal can be adjusted as Doppler factor δ of the received signal has already been estimated. We choose the MF to time the head of the frame, MF is functioned as a correlator, and the correlation peak of the MF determines the synchronization time. The reference signal of the MF is the LFM signal we used for frame synchronization at the transmitter which has been adjusted in time-scale.

III. Simulation and Experimental Results

A. Simulation Results

The performance of the proposed method has been evaluated by computer simulations and experimental results. In our simulations, frequency of LFM signal ranges from 13 kHz to 18 kHz, we take 2048 samples of LFM signal, the relative moving velocity between the transmitter and the receiver is set to be 15m/s. The channel model used in the simulation refers to [13], as is shown in Fig. 5, it is a typical channel power delay profile obtained from experimentation which is used in UWA communication of computer simulation between a single source and a destination terminal. The performance of this synchronization procedure was checked via Monte Carlo simulation.

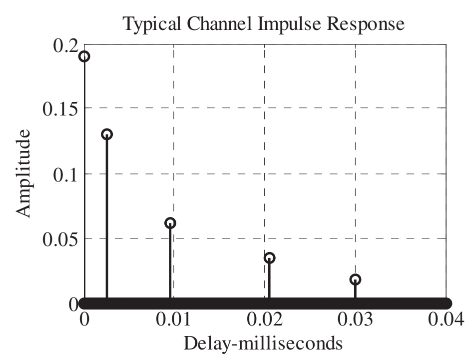

Fig. 5. Channel model used in the simulation

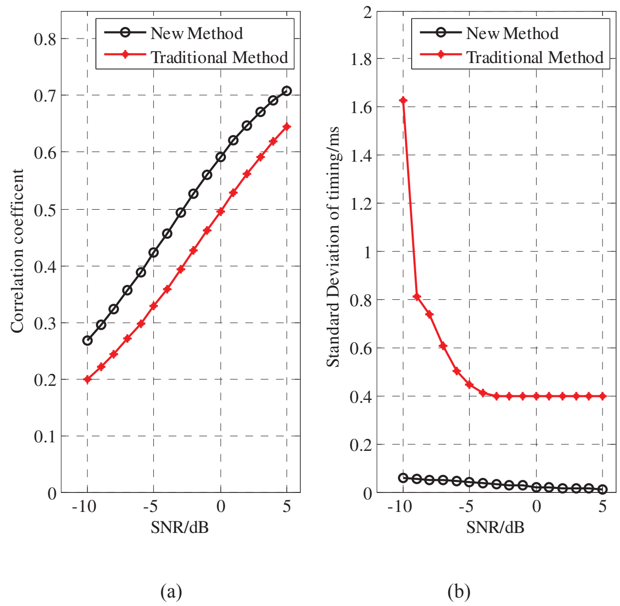

Fig. 6. Comparison of two synchronization method

Traditional frame synchronization method is based on a correlator, which takes the synchronization signal with the same signal parameters at the transmitter as the reference signal, while underwater acoustic signal suffers severe distortion due to severe signal degradations caused by multipath propagation and high temporal and spatial variability of the underwater acoustic channel conditions, so it is obvious not the optimal way of frame synchronization, we have compared the performance of this traditional method with the proposed one in this paper, results are shown in Fig. 6, where (a) shows the correlation coefficient between the received signal and the reference signal, and (b) shows timing deviation of these two methods.

The results show that the method we proposed have a larger correlation coefficient than the traditional one, and the new method offers more accurate timing precision in synchronization. As the signal-to-noise ratio drops to -4dB, the timing precision of traditional method worsens rapidly, while the new method keeps a low timing deviation even in low signal-to-noise ratio; since the correlation coefficient of the new method is higher, the false synchronization probability can be effectively avoided.

B. Experimental Results

We also did an experiment in Qiandao Lake to verify this synchronization method. Results are shown in Fig. 7.

This experiment was conducted on 28 June 2015, the modulation scheme used is multiple frequency shift keying (MFSK) and the signal frequency ranges from 10 kHz to 15 kHz, the sound speed profile of the lake is a typical negative gradient profile where the sound speed decreases steadily to the bottom. Because all ray paths interact with the bottom, the communication signal suffers from severe multipaths interference and distortion, the receiver couldn't obtain direct sound wave except at short range.

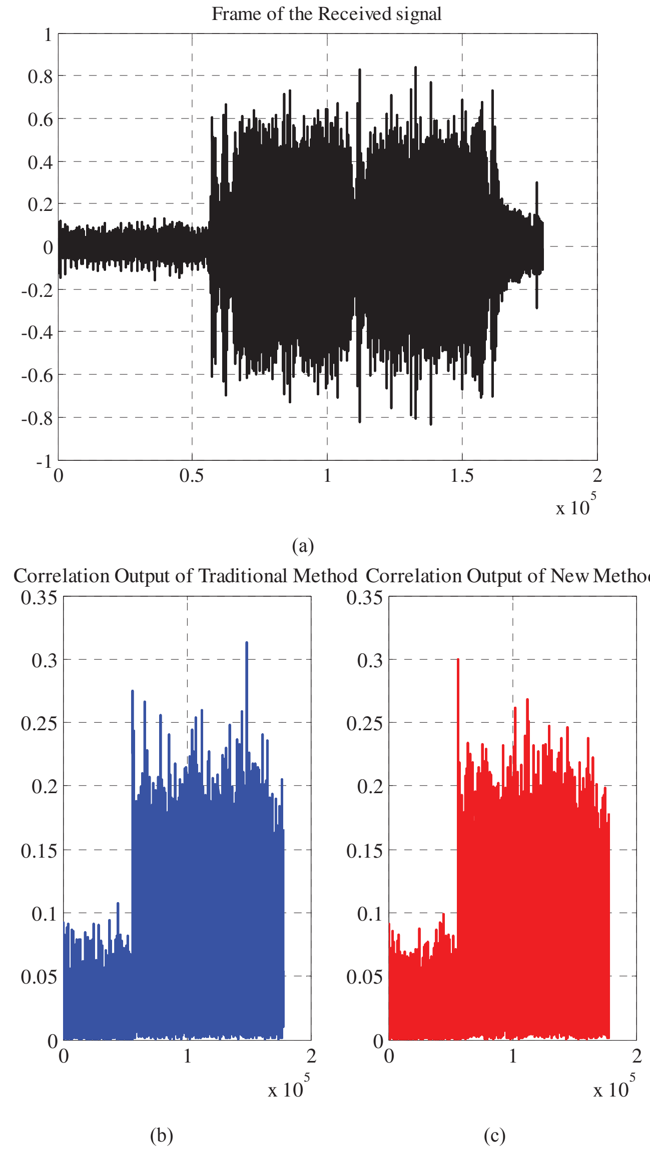

Fig. 7. (a) The received signal in lake experiment (b) Correlation Output of traditional method (c) Correlation Output of new method

In Fig. 7, (a) shows the received signal when communication distance reached 2.2 km, (b) and (c) shows the correlation output between reference signal and the received signal using traditional method and new synchronization method respectively; The reference signal used in the new synchronization method has been adjusted in time-scale, and the peak of the output indicates the head of the frame. It can be clearly seen that false synchronization occurs when using the traditional method, while the head of the frame can be successfully located by using this new method.

Serious distortion of the signal leads to low threshold of synchronous trigger, and false synchronization occurs because of the complex channel condition, traditional method on frame synchronization can't locate the head of frames correctly in such a communication condition. By using the method we proposed here, the synchronization signal can be successfully located. The experimental result shows that the new method is effective.

IV. CONCLUSION

In this paper, a new frame synchronization method in acoustic communication has been proposed and analyzed. This method we proposed, which is based on parameters estimation, has been simplified to save computational cost as to meet the needs of real time acoustic communication scenario. This paper discusses two parameters estimation techniques, including FRFT estimation and ML estimation, and we choose the latter one on account of computational cost. And the superiority of this method is clear especially when the Doppler shift is subject to rapid variations. A comparison has been made between the new method and the traditional one. Both simulation and experimental results have proved the effectiveness of this new method.

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