

## Research Article

# Time Delay Estimation of AIS Signal Based on Three-Order Cumulant

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Received 24 October 2017; Revised 5 March 2018; Accepted 4 April 2018; Published 10 May 2018

Academic Editor: Paolo Barsocchi

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A time delay estimation algorithm based on three-order cumulant with the different symmetry of modulation signal between training sequence and information sequence of AIS (Automatic Identification System) is presented, which aims to overcome the large time delay and strong noise in the receiver side of AIS. Theoretical analysis and simulation results show that this algorithm successfully suppresses the influence of Gaussian noise. Moreover, under non-Gaussian and correlative noise, it can also obtain a fine estimated performance. Simulation results show that this algorithm is superior to the correlative and data-aided time delay estimation ones.

## 1. Introduction

In AIS (Automatic Identification System), the channel varies quickly with time, and the signal noise ratio (SNR) is low [1]. The performance of time delay estimation will greatly affect the detection and separation of AIS signal. So how to achieve the accurate estimation of time delay in low SNR has become an urgent problem. For MSK and GMSK base-band signal, [2] presents a high-order autocorrelation algorithm based on the front feedback structure. It is a widely used algorithm of time delay estimation at present, but the estimation performance is affected by the autocorrelation step; [3] designs a self-correlation time delay estimation algorithm based on one-bit difference with the use of the prior information of frame structure of AIS signal. This algorithm can overcome the influence of frequency offset and phase shift, but it is sensitive to noise; [4] proposes a two-order self-correlation algorithm and it extracts the cosine function which contains time delay from the autocorrelation function of received signal, but the computation burden is large, and estimation accuracy is relatively low. Besides the estimated range is only half of the symbol period; In [5], there is a time-phase-combined estimation algorithm based

on the principle of maximum likelihood. Time delay can be obtained by the joint likelihood function of the delay and the phase of the received signal. The algorithm has a certain improvement in the estimation accuracy, but it relies on high-precision frequency offset correction. When the error exceeds 5, time delay estimation accuracy will deteriorate rapidly; [6] proposes a time delay estimation algorithm based on digital phase-locked loop, but the estimation performance is not fit for GMSK modulation signal; [7] deduces the relationship between phase information of power spectrum and autocorrelation function from frequency domain, and then the phase spectrum of the regression straight line is weighted averagely, but the estimation performance of the algorithm is greatly affected by the power spectrum method; a delay difference estimation algorithm based on one-dimensional slice of three-order cumulant is proposed in [8]. This algorithm needs two sensors to deal with two receipt signals, which is not in conformity with the requirements of AIS receiver; [9, 10] present maximum likelihood algorithms of time delay estimation. This algorithm requires that the conditional probability density should be known, so as to construct a delay probability density function to estimate the time delay by the peak. But in actual communication, the

TABLE 1: Message structure of AIS.

8 bits	24 bits	8 bits	168 bits	16 bits	8 bits	24 bits
RU	TF	SF	Data	Fcs	EF	Buffer

probability density of signal is so difficult to obtain, that it is hard to achieve in application; [11] presents an adaptive time delay estimation algorithm based on high-order cumulant. But this algorithm needs the design of high-order filter and the repeated iteration. The complex computation is not conducive to the realization of AIS receiver hardware.

The algorithm in this paper makes use of the different symmetry of modulation signal between training sequence and information sequence of AIS. Then define the delay measurement function based on three-order cumulant and estimate the time delay by minimization. As a result, it successfully restrains the influence of noise on the time delay estimation of AIS signal and the accurate estimation is achieved.

## 2. AIS Signal Model

AIS signal mainly uses the protocol of SO-TDMA. According to HDLC high-level data link control procedures, AIS signal uses grouped-structure. There is 256 bits per message. The message frame is shown in Table 1 [12]

The modulation of AIS is GMSK. The binary information of the ship is represented as  $\{a_i\}$ . After coding of NRZI (No Return Zero-Inverse),  $\{a_i\}$  take on the value  $\pm 1$ . The base-band signal of GMSK is shown as follows:

$$s(t) = e^{j\psi(a_i;t)} \quad (1)$$

$$\psi(a_i;t) = \pi \sum_i a_i q(t - iT_b) \quad (2)$$

$$q(t) = \int_{-\infty}^t g(\tau) d\tau \quad (3)$$

$$g(t) = \frac{1}{2T_b} \left\{ Q \left[ \frac{2\pi B}{\sqrt{\ln 2}} \left( t - \frac{L+1}{2} T_b \right) \right] \right\} \quad (4)$$

$$Q(t) = \frac{1}{\sqrt{2\pi}} \int_t^{\infty} e^{-\tau^2/2} d\tau, \quad (5)$$

where  $q(t)$  is the phase pulse response,  $g(t)$  is the rectangular impulse response of Gaussian filter,  $L$  is the correlation length,  $B$  is the band-width of 3 dB, and  $T_b$  is symbol period.

Rectangular impulse response of Gaussian filter is infinite in theory. In order to put Gaussian filter into practice,  $g(t)$  is truncated with correlation length  $L = 3$ .

$$q(t) = \begin{cases} 0, & t < -\frac{(L-1)T_b}{2} \\ \frac{1}{2T_b} \int_{-\infty}^t g\left(\tau - \frac{1}{2}T_b\right) 1d\tau, & -\frac{(L-1)T_b}{2} \leq t \leq \frac{(L+1)T_b}{2} \\ \frac{1}{2}, & t > \frac{(L+1)T_b}{2}. \end{cases} \quad (6)$$

## 3. Time Delay Estimation Algorithm Based on Third-Order Cumulants

**3.1. Symmetry Analysis of AIS Signal.** The symmetry of signal can be expressed by the skewness [12]. If the skewness is greater than zero, positive deviation value is larger and probability density function drags the long tail on the right. To the contrary, negative deviation value is larger, the long tail is on the left. The more close to 0 the value of skewness is, the more close to the symmetrical distribution the signal will be. If signal obeys normal distribution, the skewness is equal to the three-order center distance where value is 0.

From Table 1, training sequence of AIS signal is composed of 24 bits alternate codes between 0 and 1. After it, the start flag is made up of 01111110. Then the feature sequence is made up by training sequence and the first two codes of start flag, where the length of feature sequence is  $N$ ,  $N = 26$ . Through the modulating by GMSK, the modulation signal of feature sequence is denoted by  $s(k)$  and the skewness of  $s(k)$  is named as  $J$ . Then  $J$  is defined as

$$J = E \left[ \frac{s(k) - E[s(k)]}{\sqrt{\text{var}[s(k)]}} \right]^3, \quad (7)$$

where  $\text{var}[s(k)]$  is variance of the signal and  $E[s(k)]$  is average.

According to the definition of (7), the skewness value of feature sequence modulation signal is  $6.9244 \times 10^{-6}$ . Similarly, skewness of 100 groups of modulation signal of AIS information sequence is calculated. Then get the mean of them to be  $1.1669 \times 10^{-4}$ . Based on the comparison of two values, it is clear that the symmetry of modulation signal of feature sequence is much stronger than the information sequence. At the same time, three-order cumulant of symmetric distribution is 0, and the more close to the symmetric distribution the signal is, the smaller third-order cumulants will be. So, third-order cumulant can achieve accurate time delay estimation by different symmetry between the modulation signal of feature sequence and information sequence. Meanwhile it can also restrain the influence of Gaussian noise.

**3.2. Principle of Time Delay Estimation Algorithm.** Assume  $\{h_1, h_2, \dots, h_n\}$  is  $n$ -dimensional random variable; its  $r$  order joint cumulant is denoted by  $C_r$ :

$$C_r = \frac{1}{j^r} \frac{\partial^r \Psi(\omega_1, \omega_2, \dots, \omega_n)}{\partial \omega_1^{r_1} \partial \omega_2^{r_2} \dots \partial \omega_n^{r_n}} \Big|_{\omega_1=\omega_2=\dots=\omega_n=0}, \quad (8)$$

where the order  $r = r_1 + r_2 + \dots + r_n$ , and  $\Psi(\omega_1, \omega_2, \dots, \omega_n)$  is described by

$$\Psi(\omega_1, \omega_2, \dots, \omega_n) = \ln E \left[ e^{j(\omega_1 h_1 + \omega_2 h_2 + \dots + \omega_n h_n)} \right]. \quad (9)$$

Preset the receipt signal:

$$y(t) = x(t - \tau) + n(t), \quad (10)$$

where  $x(t)$  is the sending signal and  $n(t)$  is the noise.  $\tau$  is time delay. Assume that  $\tau_d$  is normalized time delay; then  $\tau_d = \tau/T_b$ .

After discretization, the receipt signal is shown as follows:

$$y(k) = x(k - D) + n(k), \quad (11)$$

where  $D$  is the discrete time delay.

As is known to all, AIS signal can be approximated as stationary random signal. So the three-order cumulant of  $y(k)$  can be expressed by the joint one- of three-dimensional random variable  $\{y(k), y(k - \lambda_1), y(k - \lambda_2)\}$  and denoted as  $C_{3y}(\lambda_1, \lambda_2)$ . If joint three-order cumulant is represented as symbol  $\text{cum}(\cdot)$ , then

$$C_{3y}(\lambda_1, \lambda_2) = \text{cum}(y(k), y(k + \lambda_1), y(k + \lambda_2)). \quad (12)$$

In a similar way, the cumulant of sending signal can be noted as follows:

$$C_{3x}(\lambda_1, \lambda_2) = \text{cum}(x(k), x(k + \lambda_1), x(k + \lambda_2)). \quad (13)$$

When the time delay  $\tau$  is an integer multiple of symbol period,  $y(k)$  is the sampled signal of  $y(t)$ . Sample period is  $T$ , and the total of sampling point in each symbol period is  $N_b$ , where  $N_b = T_b/T$ . Then it is obvious that the time delay  $D$  is an integer multiple of  $N_b$ . Now according to the length of the modulation signal of the feature sequence, gradually slide and intercept the corresponding part of  $y(k)$ . Then define the delay measurement function in accordance with three-order cumulant. Finally, the time delay is estimated by minimization.

Sliding value  $d$  is given as follows:  $d \in \{0, 1, \dots, M\}$ , where  $M = \tau_m \cdot N_b$  and  $\tau_m$  is the largest normalized time delay of AIS with its value being an integer and larger than  $\tau_d$ .

The truncation signal can be described by

$$y_d(k) = y(k + d), \quad k = 0, 1, \dots, N_t, \quad (14)$$

where  $N_t = N \cdot N_b$ .

Set time delay measurement function  $P(d)$ :

$$P(d) = C_{3y_d}(\lambda_1, \lambda_2). \quad (15)$$

Let  $\lambda_1 = 0, \lambda_2 = 0$ . According to (14) and the character of three-order cumulant, (15) can be expanded as follows:

$$\begin{aligned} P(d) &= C_{3x_d}(0, 0) + C_{3n_d}(0, 0) \\ &= \text{cum}(x_d(k - D), x_d(k - D), x_d(k - D)) \\ &\quad + \text{cum}(n_d(k), n_d(k), n_d(k)), \end{aligned} \quad (16)$$

where  $x_d(k) = x(k + d), n_d(k) = n(k + d), k = 1, 2, \dots, N_t$ .

If  $n(k)$  is Gaussian noise, the value of three-order cumulant of  $n(k)$  is 0; if  $n(k)$  is non-Gaussian and correlative noise

with high SNR, just ignore the influence of noise. Then  $P(d)$  can be simplified as follows:

$$P(d) = C_{3x_d}(0, 0). \quad (17)$$

From (17), when  $x_d(k)$  represents the modulation signal of the feature sequence,  $P(d)$  obtains minimum value. So, assume

$$\widehat{D} = \arg \min_d \{P(d)\}. \quad (18)$$

Then the time delay is estimated as follows:

$$\widehat{\tau} = \widehat{D}T. \quad (19)$$

When the normalized time delay  $\tau_d$  is a decimal,  $N_b$  should satisfies the following:

$$N_b = \xi \cdot 10^v, \quad (20)$$

where  $v$  is the order of magnitude of  $\tau_d$  and  $\xi$  is an integer larger than zero.

From (20), this algorithm relies too much on the sampling frequency and therefore fits the sample signal with cubic spline interpolation algorithm. Assume there are  $N_s$  interpolating points in each sample period  $T$ ; the interval of interpolation is  $T_s$ , where  $T_s = T/N_s$ . Then  $N_b$  just needs to satisfy the following:

$$N_b \times N_s = \xi \cdot 10^v. \quad (21)$$

Same as the previous estimate algorithm in this paper, if the normalized time delay is a decimal, the time delay can be estimated by

$$\widehat{\tau} = \widehat{D}T_s. \quad (22)$$

## 4. Simulation Results

Simulation parameter setting is shown in Table 2. All the simulation parameters are set according to AIS protocol. There is an assumption that the transmitter and receiver are UTC synchronization.

Normalization in this paper is achieved with  $T_b$ . There are four cases of simulation in this table. In Case 1, because time delay is an integral multiple of the symbol period, the signal does not need to be interpolated. But in Case 2/3/4, the interpolation can make estimation more accurate. The simulation results are as follows.

### (1) Time Delay Estimation under Gaussian Noise

*Case 1.* When normalized time delay  $\tau_d$  is an integer, simulation results are shown in Figures 1 and 2. In the simulation, frequency offset is set to be zero.

By the graph under different SNR (Signal-to-Noise Ratio), this algorithm can get completely accurate estimation. The estimation performance is not affected by the value of SNR or time delay.

*Case 2.* When normalized time delay  $\tau_d$  is a decimal, set frequency offset to be zero, too. The simulation results are

TABLE 2: Simulation parameter.

System parameter settings			
Symbol rate	$R = 9600$ bit/s		
Symbol period	$T_b = 1/R$		
Modulation system	GMSK		
Normalized maximum time delay	$12T_b$		
Time delay estimation under Gaussian noise			
Case 1: integral normalized time delay		Case 2: decimal normalized time delay	
Time delay	$2T_b, 4T_b, 8T_b$	Time delay	$0.221T_b, 1.414T_b, 8.331T_b$
Sample period ( $T$ )	$T_b/10$	Sample period ( $T$ )	$T_b/10$
		Interpolation period ( $T_s$ )	$T/100$
Time delay estimation under non-Gaussian and correlative noise			
Case 3: integral normalized time delay		Case 4: decimal normalized time delay	
Time delay	$2T_b, 4T_b, 8T_b$	Time delay	$0.221T_b, 1.414T_b, 8.331T_b$
Sample period ( $T$ )	$T_b/10$	Sample period ( $T$ )	$T_b/10$
Interpolation period ( $T_s$ )	$T/100$	Interpolation period ( $T_s$ )	$T/100$

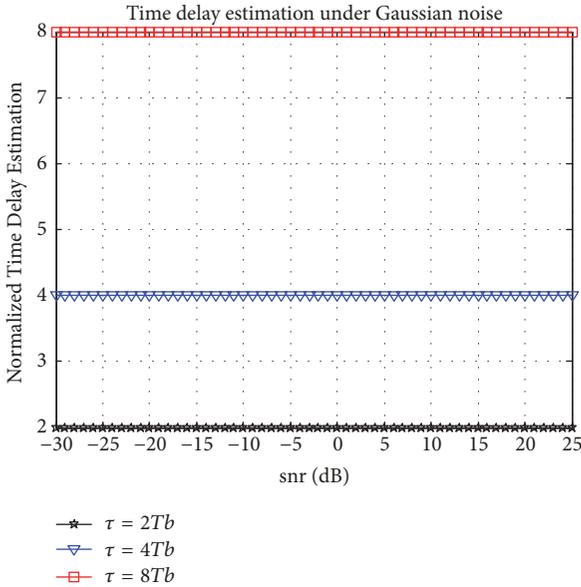


FIGURE 1: Normalized time delay and its estimation with different SNR.

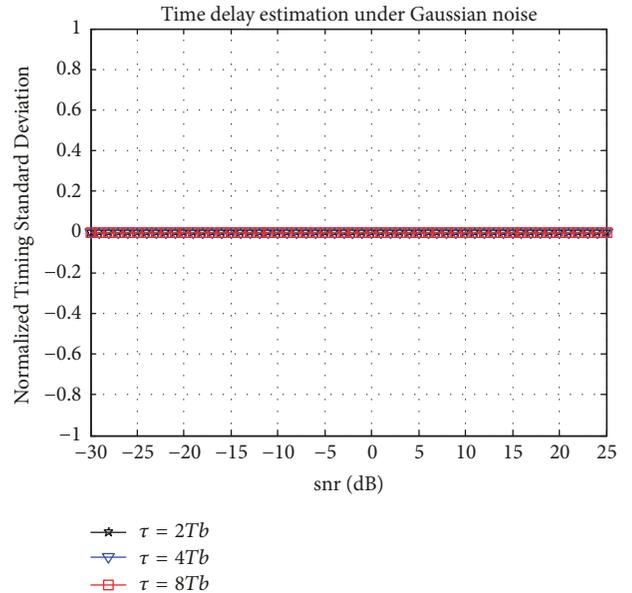


FIGURE 2: Standard deviation of normalized time delay estimation with different SNR.

shown in Figures 3 and 4. From the graph, when the number of interpolation points is large enough, the signal estimation performance is the same as the normalized time delay being an integer. Besides it is also not affected by the value of time delay or SNR.

In Figure 5, the time delay estimation proposed in this paper is compared with those presented in [3, 4]. For the sake of simplicity, let “1-bit-difference algorithm” denote the algorithm from [3] and “M&M algorithm” denote the one from [4]. According to the fact that the range of estimation of algorithm from literature [4] is  $[-0.5T_b, 0.5T_b]$ , let  $\tau = 0.375T_b$ . From the graph, the estimated performance in this paper is much better than others, especially under the lower SNR.

From Figure 6, different SNR do not affect the performance of time delay estimation with the same frequency offset. But the performances will drop rapidly with the increase of frequency deviation. As the frequency offset is 5 Hz, normalized MSE is larger than  $10^{-3}$ . Therefore, this algorithm needs higher frequency deviation correction accuracy.

(2) *Time Delay Estimation under Non-Gaussian and Correlative Noise.* In this subsection, the noise is set as follows:

$$n(k) = x(k) \times w(k), \quad (23)$$

where  $w(k)$  is Gaussian noise. The relationship between intensity  $\kappa$  of Gaussian noise and SNR is shown in Table 3.

TABLE 3: Noise Intensity and SNR.

$\kappa$	-30	-29	-28	-27	-26	-25	-24	-23	-22	-21
SNR	60.03	57.95	56.03	53.99	52.07	50.02	47.99	46.08	43.91	42.00
$\kappa$	-20	-19	-18	-17	-16	-15	-14	-13	-12	-11
SNR	40.03	38.00	35.95	34.01	31.92	30.00	27.93	26.09	23.86	22.01
$\kappa$	-10	-9	-8	-7	-6	-5	-4	-3	-2	-1
SNR	20.03	18.00	16.13	14.01	11.98	10.06	8.02	6.02	3.96	1.97
$\kappa$	0	1	2	3	4	5	6	7	8	9
SNR	-0.04	-1.91	-3.98	-5.90	-7.99	-10.08	-11.97	-13.99	-15.82	-18.00

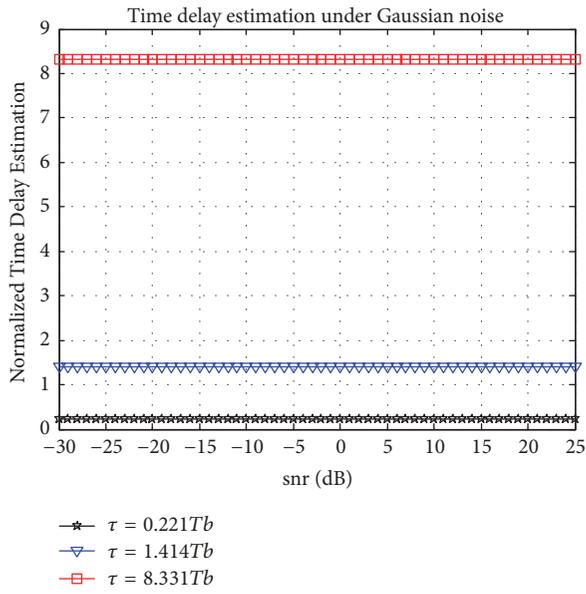


FIGURE 3: Normalized time delay and its estimation with different SNR.

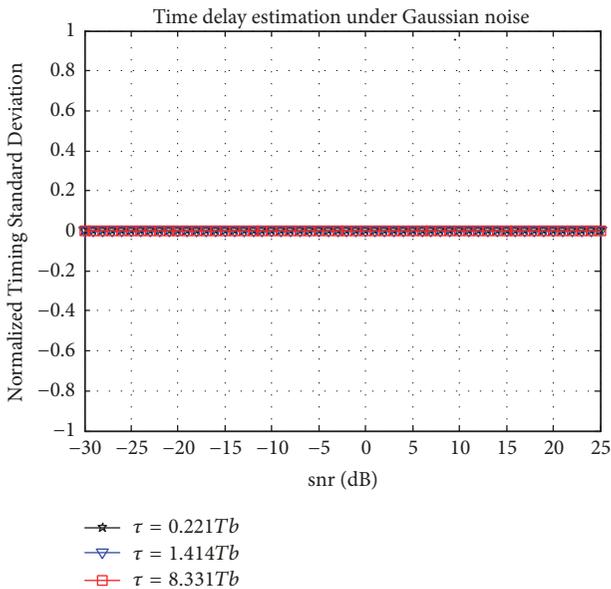


FIGURE 4: Standard deviation of normalized time delay estimation with different SNR.

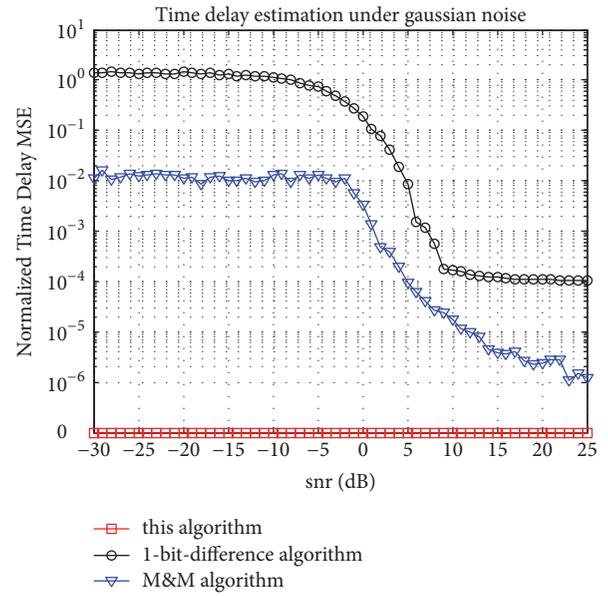


FIGURE 5: Comparison among different algorithms.

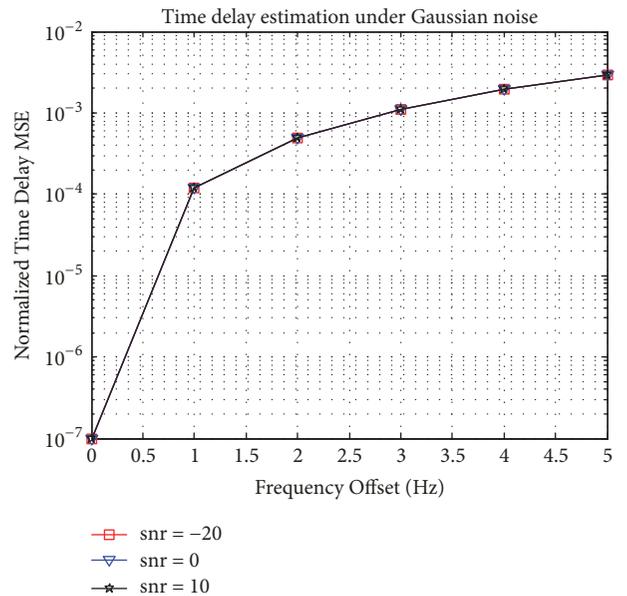


FIGURE 6: Influence of frequency offset.

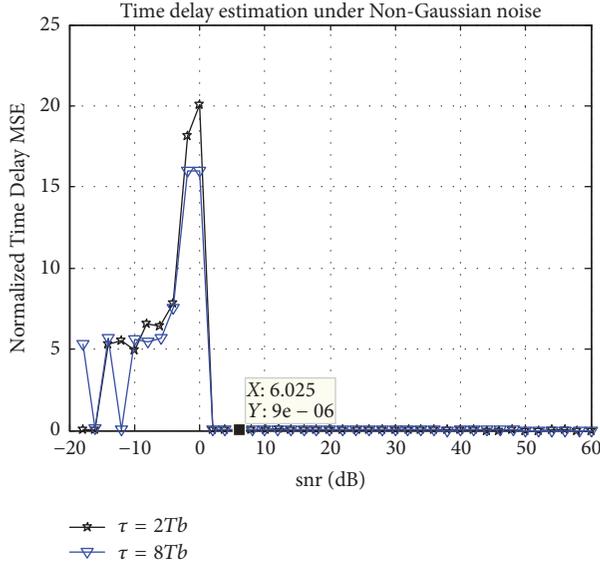


FIGURE 7: Normalized time delay estimation MSE with integral time delay.

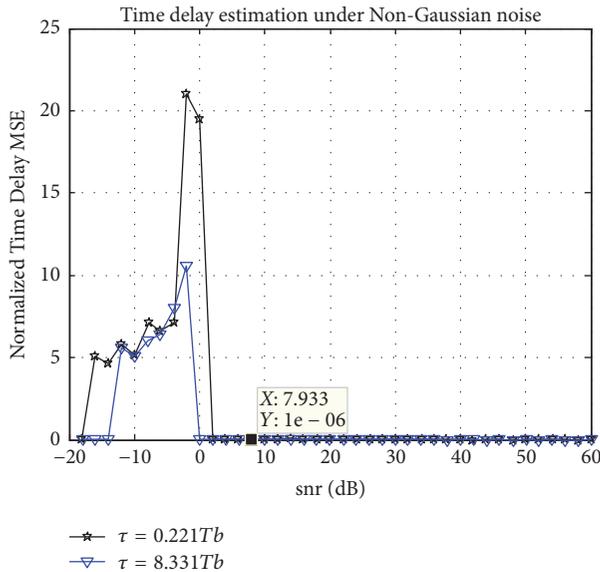


FIGURE 8: Normalized time delay estimation MSE with decimal time delay.

By Table 3, the SNR is decreasing with the increase of the intensity of Gaussian noise.

Because the noise is not additive Gaussian one and it is related to the signal, the three-order cumulant cannot completely suppress the influence of noise to the signal symmetry. The frequency offset is set to be zero.

*Case 3.* When normalized time delay  $\tau_d$  is an integer, simulation results are shown in Figure 7.

*Case 4.* When normalized time delay  $\tau_d$  is a decimal, the simulation results are shown in Figure 8.

From Figures 7 and 8, the algorithm has obvious noise threshold at 0 dB, but when SNR is larger than 0 dB, the normalized time delay estimation can still achieve  $10^{-5}$ .

## 5. Conclusion

In this paper, under Gaussian noise, the algorithm can accurately estimate the time delay. Its performance is more accurate than the 1-bit-difference algorithm and M&M algorithm. It is nearly not affected by the SNR, even if SNR is  $-30$  dB. Furthermore when the noise is non-Gaussian and correlative, the normalized time delay estimation MSE can achieve  $10^{-5}$  with SNR being larger than 0 dB. These two are the main contributions of this paper. Besides the algorithm has the same performance for all signals with similar characteristics and is not limited to AIS signals. However, this algorithm needs to know the order of magnitude of the delay in the received signal, so that we can set the Sample Frequency and the Interpolation Frequency. Otherwise the algorithm cannot get the perfect results.

## Conflicts of Interest

The authors declare that there are no conflicts of interest regarding the publication of this paper.

## Acknowledgments

This research work is supported in part by the National Natural Science Foundation of China under Grant 61601326.

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