Five Sensors Acoustic Array Location Model and Target Signal Recognition Method

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Abstract

Based on poor measurement accuracy problems for location system of acoustic array with four sensors or five sensors, this paper proposes an improved location processing algorithms of acoustic array with five sensors and researches an improved target location calculation model of acoustic array with five sensors. According to an improved acoustic location calculation model, this paper makes error derivation and analysis for three-dimensional coordinates by differential method and gives an error calculation principle; this paper researches target signal recognition algorithm of wavelet transform and gives detailed analysis of recognition reasoning; this paper researches the relationship between location error and pitch angle, matrix length and makes derivation and analysis for best array size and pitch angle. Calculation and test results show that location error is smaller when the target distance is closer. Some acoustic sensor signals are relatively weak and the error increases significantly when the target keeps away from acoustic sensor. It is effective to reduce error when detection distance is far and target signal is not obvious and improve measurement accuracy of test system by target recognition algorithm.

Key words: ACOUSTIC ARRAY WITH FIVE SENSORS, TIME-DELAY ERROR, TARGET DETECTION

1. Introduction

Passive acoustic location technology has a very wide range of applications in many fields. The principle of passive acoustic location system is to detect the time from target acoustic signal to the microphone through microphone, calculate acoustic path, and accomplish location with the acoustic array geometry model and a series of calculations[1]. Acoustic array layout is an important location tool of the passive acoustic location system. From the current situation at home and abroad, it often use acoustic array with five sensors and seven sensors in space location track system. In the passive acoustic location technology, there is a very important technology--target and signal processing technology. The microphone transforms received acoustic signals into electrical signals by the sensitive elements, outputs the amplified signal in form of voltage after the enlargement process. Acoustic targeting technology is to make correlation analysis for acoustic signal by signal analyzers or computers, calculate the time-delay of two signals, and then estimate the target azimuth angle and distance[2]. That is the time-delay algorithm. Because of the impact of non-target, the noise signal will also affect the accuracy of TDE, thus affecting direction and distance accuracy of acoustic array with five sensors. Especially, for passive time-delay estimation system, it does not emit a signal and determine target parameters by receiving electromagnetic waves or acoustic waves emitted by target[3]. This method can not control the size of the received signal energy, but it has good concealment. It has important significance for military applications. In order to enhance direction and distance stability and accuracy of acoustic detection array and improve extraction way of acoustic signal, this paper uses wavelet analysis to research information recognition and detection of target and microphone.

2. Model of acoustic array with five sensors

2.1. Calculation model of acoustic array with five sensors

Calculation model of acoustic array with five sensors is shown in figure 1, and it is three-dimensional coordinate system with microphone S_0 as the coordinate origin. Coordinates of four microphones are $S_1(0,l,l)$, $S_2(l,0,-l)$, $S_3(0,-l,l)$, $S_4(-l,0,-l)$, acoustic target is T, its coordinate is (x,y,z), it is assumed that the distance from acoustic target T to microphone S_0 (coordinate origin) is r, azimuth angle is φ, pitch angle is θ.

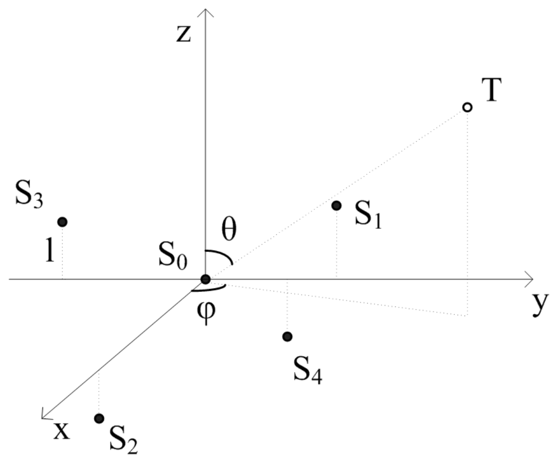

Figure 1. Calculation model of acoustic array with five sensors

τ_i indicates time difference that acoustic signal of target arrives at microphone S_i and microphone S_0, c indicates acoustic speed, Δd_i indicates distance difference that acoustic signal of target arrives at microphone S_i and microphone S_0, that is acoustic path difference. Then formula (1) can be got:

$$\Delta d_i = \tau_i * c \tag{1}$$

According to the geometric relationship of model, equations (2) can be got,

$$\begin{cases} x^2+y^2+z^2=d^2 \\ x^2+(y-l)^2+(z-l)^2=(d+\Delta d_1)^2 \\ (x-l)^2+y^2+(z+l)^2=(d+\Delta d_2)^2 \\ x^2+(y+l)^2+(z-l)^2=(d+\Delta d_3)^2 \\ (x+l)^2+y^2+(z+l)^2=(d+\Delta d_4)^2 \end{cases} \tag{2}$$

In (2), d is the distance between target T and S_0, formula (3) can be got:

$$d=ct \tag{3}$$

In (3), c is acoustic speed, t is the time that can be detected by microphone, that is called acoustic path.

According to $\Delta d_i = d$, equations (4) can be got:

$$\begin{cases} x = \dfrac{\Delta d_4{}^2 - \Delta d_2{}^2 + 2d\bullet\Delta d_4 - 2d\bullet\Delta d_2}{4l} \approx \dfrac{d}{2l}(\Delta d_4 - \Delta d_2) \\ y = \dfrac{\Delta d_3{}^2 - \Delta d_1{}^2 + 2d\bullet\Delta d_3 - 2d\bullet\Delta d_1}{4l} \approx \dfrac{d}{2l}(\Delta d_3 - \Delta d_1) \\ z = \dfrac{\Delta d_4{}^2 + \Delta d_2{}^2 + 2d\bullet\Delta d_4 + 2d\bullet\Delta d_2 - \Delta d_3{}^2 - \Delta d_1{}^2 - 2d\bullet\Delta d_3 - 2d\bullet\Delta d_1}{8l} \\ = \dfrac{d}{4l}(\Delta d_4 + \Delta d_2 - \Delta d_3 - \Delta d_1) \\ d = \dfrac{8l^2 - (\Delta d_4{}^2 + \Delta d_2{}^2 + \Delta d_3{}^2 + \Delta d_1{}^2)}{2(\Delta d_4 + \Delta d_2 + \Delta d_3 + \Delta d_1)} \end{cases} \tag{4}$$

According to equations(2) and equations(4), combined with formula(3), equations(5) can be got:

$$\begin{cases} \tan\varphi = \dfrac{y}{x} = \dfrac{\tau_3-\tau_1}{\tau_4-\tau_2} \\ \sin\theta = \dfrac{\sqrt{x^2+y^2}}{d} \\ = \dfrac{c}{2l}\sqrt{(\tau_4-\tau_2)^2+(\tau_3-\tau_1)^2} \\ d = \dfrac{8l^2 - c^2(\tau_1+\tau_2+\tau_3+\tau_4)^2}{2c(\tau_1+\tau_2+\tau_3+\tau_4)} \end{cases} \tag{5}$$

The above equations are the location equations of acoustic targets. According to equations, we can know that azimuth angle and pitch angle are concerned with five microphones. So the five-dimensional space detection array also has good orientation accuracy and good detection accuracy of low pitch angle. Distance d is concerned with five sensors, therefore the detection array has a better fixed distance accuracy.

2.2. Location error analysis and simulation of acoustic array with five sensors

Location accuracy of acoustic array with five sensors is concerned with acoustic speed, time-delay estimation accuracy and arrangement of acoustic array by calculating the above model. Acoustic speed can be corrected according to site factors, layout of acoustic arrays also can be reduced and avoided errors by accurate measurement[4]. So analysis of time-delay accuracy is an important factor in accurate location of passive acoustic location system.

(1) Orientation error analysis

According to above equations, equations(6) can be got by calculating the azimuth angle partial derivative:

$$\begin{cases} \dfrac{\partial\varphi}{\partial\tau_1} = -\dfrac{\partial\varphi}{\partial\tau_3} = \dfrac{1}{1+\tan^2\varphi(\tau_1-\tau_3)^2} \\ \dfrac{\partial\varphi}{\partial\tau_2} = -\dfrac{\partial\varphi}{\partial\tau_4} = \dfrac{\tau_4-\tau_2}{1+\tan^2\varphi(\tau_1-\tau_3)} \end{cases} \tag{6}$$

Relationship between azimuth angle error and time-delay variance can be got from equations(6):

$$\sigma_\varphi = \sqrt{(\frac{\partial\varphi}{\partial\tau_1})^2\bullet\sigma_\tau + (\frac{\partial\varphi}{\partial\tau_2})^2\bullet\sigma_\tau + (\frac{\partial\varphi}{\partial\tau_3})^2\bullet\sigma_\tau + (\frac{\partial\varphi}{\partial\tau_4})^2\bullet\sigma_\tau} = \frac{\sqrt{2}c}{2l\sin\theta}\bullet\sigma_\tau \tag{7}$$

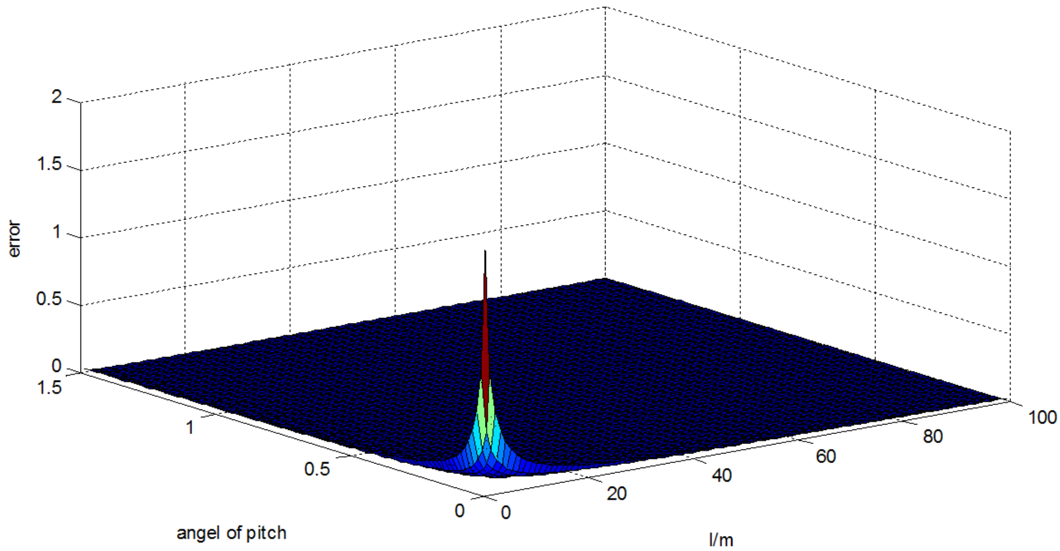

Figure 2. Three-dimensional simulation diagram of azimuth angle error

we know that the location error is related to array size l, sine of pitch angle $\sin\theta$ and time-delay error by combining with equations and simulation diagram. Increasing the array size and pitch angle can reduce the orientation error, and the greater time-delay error, the greater orientation error.

Similarly, pitch angle location error can be represented as:

$$\sigma_\theta = \frac{\sqrt{2}c}{2l\cos\theta}\bullet\sigma_\tau \tag{8}$$

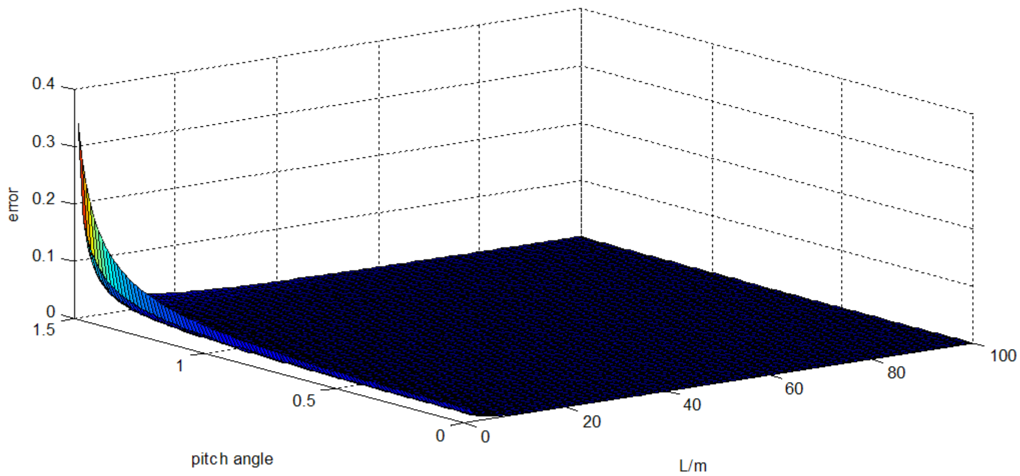

Figure 3. Relationship between pitch angle error and pitch angle, acoustic array size

We can know that the accuracy of pitch angle is related to time-delay estimation error, array size and cosine of pitch angle from the above analysis. Increasing array size and decreasing the pitch angle will improve the accuracy of pitch array.

(2) The analysis of fixed distance error

Formula (9) can be got from above equations:

$$\frac{\partial d}{\partial \tau_i}=\frac{-2c^2\sum_{i=1}^{4}\tau_i^2-\left(8l^2-c^2\sum_{i=1}^{4}\tau_i\right)^2}{2c^2\left(\tau_1+\tau_2+\tau_3+\tau_4\right)^2} \qquad (9)$$

Formula (10) can be got after simplifying formula (9):

$$\frac{\partial d}{\partial \tau_i}=\frac{-c\tau_i-d}{\tau_1+\tau_2+\tau_3+\tau_4} \qquad (10)$$

So, standard deviation of the distance d is :

$$\sigma_d=\sqrt{\sum_{i=1}^{4}\left(\frac{\partial d}{\partial \tau_i}\right)^2}\sigma_z \approx \frac{d\sqrt{d_2+2l^2}}{l^2\left(2-\sin^2\theta\right)}.\sigma_z \qquad (11)$$

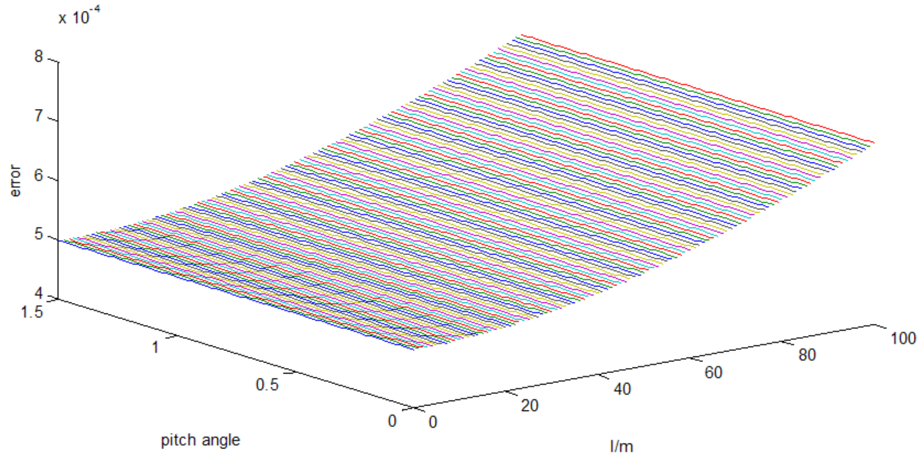

Figure 4. Relationship between fixed distance error and acoustic array size, pitch angle

We can know that target distance relative estimation error is related to time-delay accuracy, acoustic array size, the actual target distance and pitch angle of target from the above equations and simulation analysis. the greater acoustic array size and pitch angle, the smaller fixed distance accuracy, the greater actual distance, the greater time-delay.

3. Target signal recognition algorithm (wavelet algorithm) and simulation of acoustic location

It is necessary to know time difference τ_i that acoustic signal of target arrives at microphone S_i and microphone S_0 and initial coordinate of microphone for calculating azimuth angle of target by analyzing model of acoustic array with five sensors. The main method of improving location accuracy is to improve the accuracy of time difference τ_i from calculation model error analysis of acoustic array with five sensors[5]. In the passive acoustic location system, τ_i is often calculated by time-delay estimation algorithm. Target recognition is an important factor of influencing accuracy of time-delay estimation algorithm. Combined with Fisher criterion, wavelet decomposition is proposed to identify target and reduce time-delay error in this paper.

3.1. Signal model of time-delay estimation method

Time-delay estimation method is a key technology in the passive acoustic location. Time-delay estimation (TDE) refers to estimate and measure time-delay by using method of parameter estimation and signal processing[6].

In time-delay estimation, noise will make an influence on received signals through microphone, so the signal model is:

$$x_i(n)=\alpha_i(n-\tau_i)+s_i(n) \qquad (12)$$

In (12), α_i is acoustic wave propagation attenuation coefficient ($\alpha_i<1$), τ_i is propagation time-delay of microphone, $s_i(n)$ is additive noise.

3.2. Principle of wavelet transform

It is assumed that $\psi(t)$ is a square integral function, that is $\psi(t)\in L^2(R)$, $\Psi(\omega)$ is the Fourier transform of $\psi(t)$, when $\Psi(\omega)$ meets such conditions:

$$\int_{-\infty}^{+\infty}\frac{\left|\psi(\omega)\right|^2}{\omega}d\omega<\infty \qquad (13)$$

$\psi(t)$ can be regarded as a basic or mother wavelet, a wave sequence function can be got after making parallel movement and extension on $\psi(t)$ [7].

$$\psi_{a,b}(t) = a^{-\frac{1}{2}}\psi(\frac{t-b}{a}), a>0, b\in R \quad (14)$$

In (13), a is the telescopic factor, b is the shift factor.

Continuous wavelet transform (CWT) is defined as: It is assumed that $f(t)$ is a square inferable function and $\psi(t)$ is the complex conjugate of $\psi(t)$, continuous wavelet transform of $f(t)$ is:

$$WT_f(a,b) = \langle f(t), \psi_{a,b}(t)\rangle = \frac{1}{\sqrt{a}}\int_{-\infty}^{+\infty} f(t)\overline{\psi\left(\frac{t-b}{a}\right)}dt \quad (15)$$

For the discrete case, the wavelet sequence is shown as follows:

$$\psi_{(2^{-j},b)} = 2^{\frac{j}{2}}\psi(2^j(x-b)) \quad (16)$$

In equation (16), j is an as scale index, b is location coordinates.

Microphone output signals contain low frequency and high frequency signals, usually low frequency can be eliminated through filtering circuit in circuit processing, and microphones for target sound signals are high-frequency signals, so the same output signal is needed to detect and extract target signal[8].

For the above considerations, based on basic principle of wavelet transform, extraction of microphone signal characteristics only needs to focus on band-pass frequency characteristics, using 3-layer wave decomposition as an example, removing more than half closed frequency Wavelet coefficients d_1, that is to say we only need to extract wavelet coefficients c_3, d_3, d_2 with its bands energy as characteristics, feature vector can be set up:

$$X = (E_1, E_2, E_3) \quad (17)$$

3.3. Recognition target signal method of small Fisher

When using statistical method to pattern recognition, many problems are related to dimension, methods work out in the low dimensional space may not be feasible in high dimensional space, so Fisher discriminates method is to reduce dimension number and solve dimension reduction problems[9].

How to determine detection circuit of the microphone output signal contains information of target signal, the problem can be summed up as two types of discrimination model, the description is: setting up the entire G_1, G_2, G_1 is the background signal overall, G_2 is the sum of background signal and target signal. Assume r_1, r_2 are two average value of G_1 and G_2, their corresponding function value are $y_1 = C^T r_1$ and $y_2 = C^T r_2$. Discriminates function coefficients can be obtained by mathematical calculation, then discriminates function are obtained by coefficient, so we can determine discriminates threshold y_0. Data discriminates $X = (x_1, \cdots x_p)$ is applied in functions, then we can get result y. making $y_1 > y_2$, If $y > y_0$, then determine $X \in G_1$, if $y < y_0$, $X \in G_2$ is identified. By Fisher decision analysis, determine whether there is signal is to determine if it belongs to the general G_1 or G_2, according to the model, if the signal belongs to G_1, target signal does not existed, which indicate microphone circuit makes false triggering by larger noise, if belongs to G_2, then receiving sound signals come from target signal.

Based on wavelet transform theory, 3 layers of Wavelet is resolved and wavelet coefficients a_3, d_1, d_2 are extracted, using signal frequency band energy as it represents characteristics, feature vector is set up: $X = (E_1, E_2, E_3)$. Because time is short when target signals on microphone, the effectiveness of algorithms and real-time are particularly important, which effectively reduces feature dimension under low Wavelet Decomposition levels, improves target recognition in real-time. Set the passive sound localization system background signal feature dimension of number n_1 is $X_1^{(1)}, X_2^{(1)}, \cdots, X_{n_1}^{(1)}$, feature dimension contains number observation data is $X_1^{(2)}, X_2^{(2)}, \cdots, X_{n_2}^{(2)}$, through Fisher discriminates method and mathematical methods can be obtained by linear discriminates function In the detection process, the target signal is determined by the signal filtering and the Fisher discriminates criterion[10], and then the singular point of Fisher's output signal is detected by using the wavelet modulus maxima theory. Then using the singular point of the signal to find the target signal, moreover using the time delay algorithm and combining the algorithm of the acoustic detection array model and the geometric model to calculate the target's related parameters, such as azimuth and velocity.

4. Simulation calculation and experimental analysis

4.1. signal recognition simulation Simulation the Fisher discriminates analysis of wavelet with MATLAB, and assuming a gate signal as the target signal. The following figures are the simulation results, we discuss the Fisher decision rule of the different signal to noise ratio, from figure 5 to figure 10 respectively refer to the background signal, the containing target signal, the approximate component and 3 level detail component.

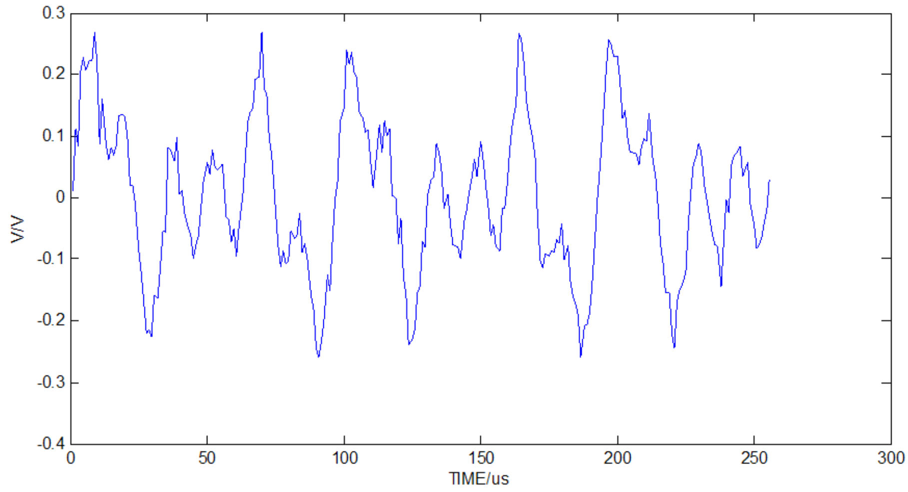

Figure 5. The background signal

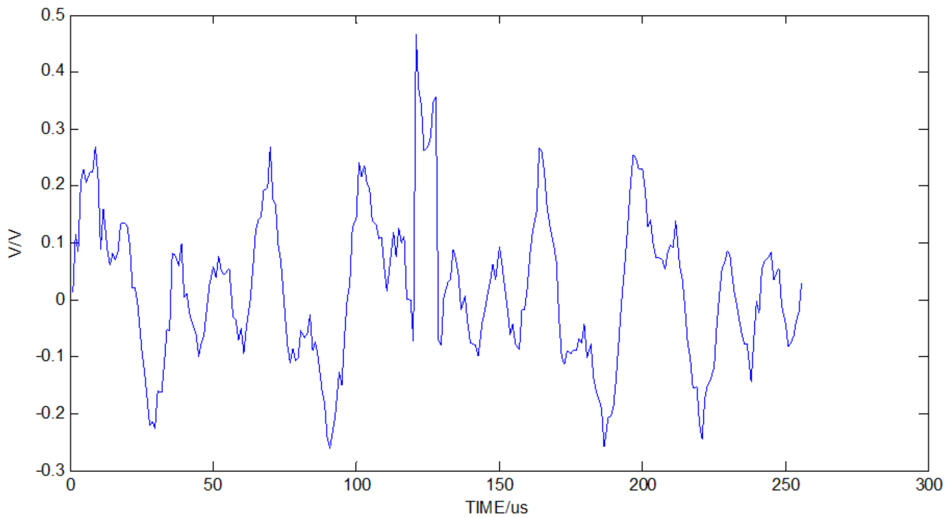

Figure 6. The containing target signal

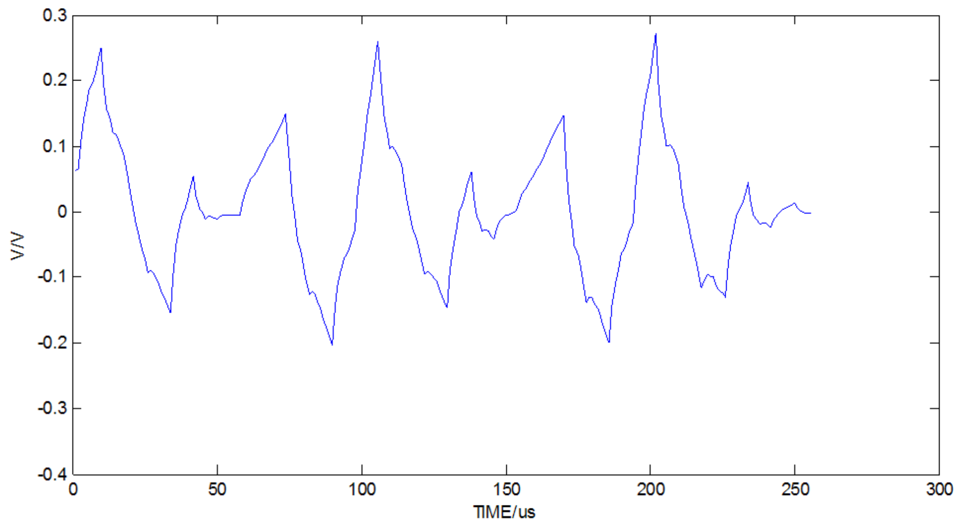

Figure 7. The approximate component

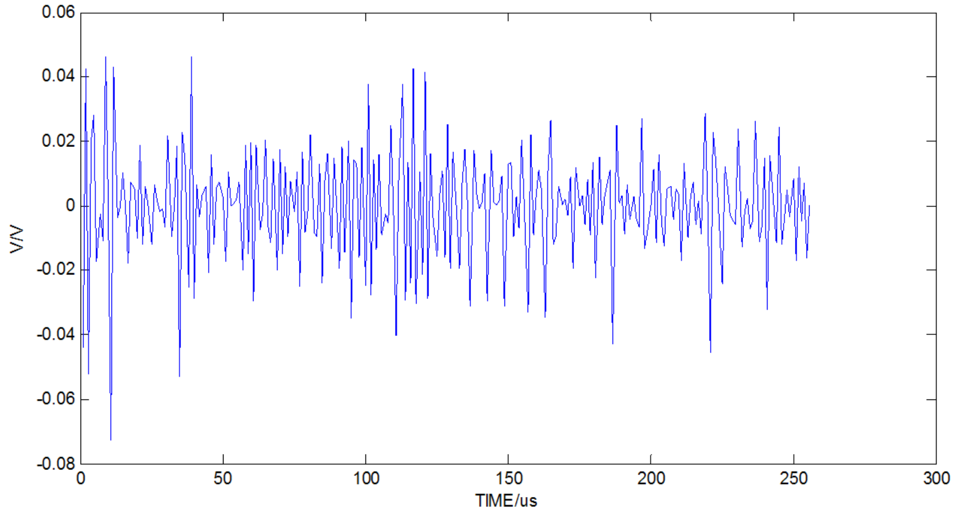

Figure 8. The first level detail component

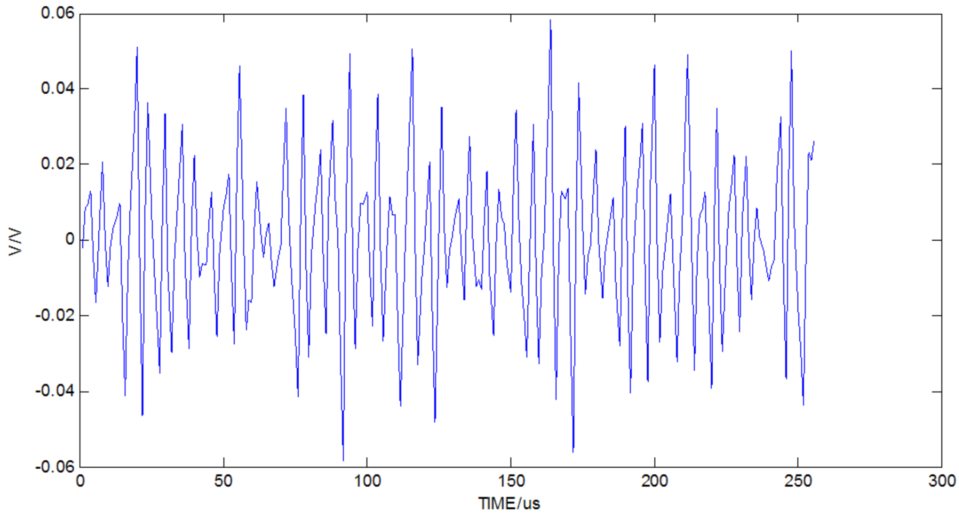

Figure 9. The second level detail component

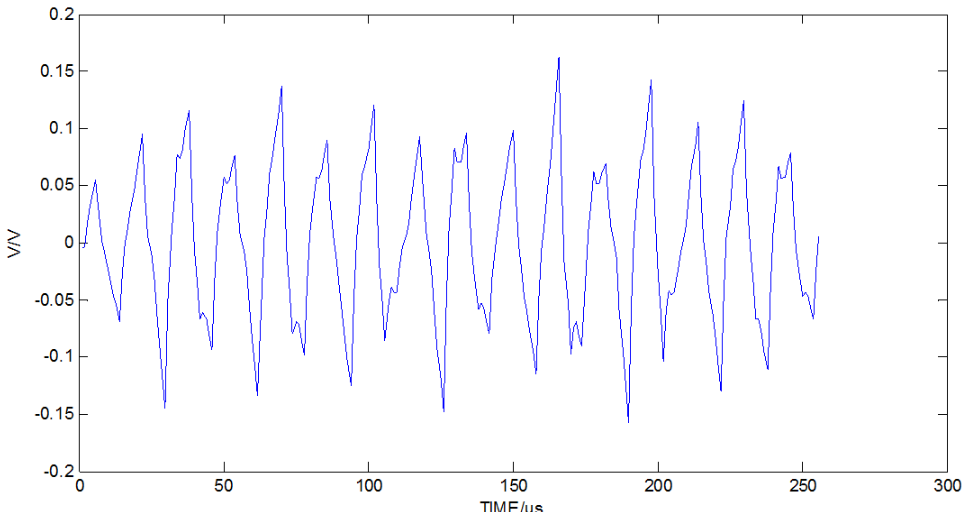

Figure 10. The third level detail component

According to the signal to noise ratio definition

$$SNR=10\lg(\sum_{i=1}^{N} s(i)^2 / \sum_{i=1}^{N} n(i)^2) \quad (18)$$

In (21), $s(i)$ is the output signal of a sound sensor detection part, $n(i)$ is a noise, N is the sampling point. Combining the figures and the 3 level wavelet decomposition, the energy of the corresponding frequency band is extracted according to the target signal extraction method, and the feature vector is $X=(E_1, E_2, E_3)$, respectively take value from the target signal and the noise signal values, as shown in Table 1. For the two groups of characteristic values, discriminates function can be obtained by discriminates analysis.

Table 1. The feature vector

signal	E_1	E_2	E_3
the background signal	2. 4877	0. 1555	1. 1826
	2. 5033	0. 1307	1. 1349
	2. 4377	0. 0864	1. 1665
	2. 3889	0. 1012	1. 1274
the containing target signal	2. 6513	0. 2912	1. 2058
	2. 8167	0. 2604	1. 1985
	2. 8212	0. 2342	1. 1827
	2. 7471	0. 2440	1. 1840

The method is better than the common method of judging the target by combining the simulation. And this calculation is simple and fast. Combing with simulation inference, this method prefer to common method to determine goals. And this one has a simple calculation and a faster computing speed.

4.2. Experiment and analysis

In passive acoustic positioning system, the analog signals which get measured and output by computer acquisition acoustic detection device can get the output signal acting on microphone which can smooth target after wavelet processing. Assuming that system sampling rate is f, the signal after filtering is y(i), whether it is target signal, it can be determined, and it can calculate the time and the sound path accurately, thus target azimuth speed can be calculated according to mathematical geometric method and sound array geometry.

To determine the position of the target utilizing the noise when the target moving is the key techniques to determinate the target position by passive acoustic detection system. The system measure the parameters of the target sound by passive mode and ensure the direction and distance of the target using the geometry of sound path difference and microphone array.

Time delay estimation is juxtapose the phase of the target sound source signals received by two microphones to get receiving unified signal time difference. Assuming that the signal received by two microphones with the sound source are $x_1(t)$ $x_2(t)$, the correlation is:

$$R_{x_1, x_2}(\tau)=x_1(t) * x_2(t)=\int_{-\infty}^{+\infty} x_1(t) x_2(t+\tau) dt \quad (19)$$

For stationary signals, corresponding to the correlation values $R_{x_1,x_2}(\tau)$ is the maximum. In the actual detection, microphone output signal is interfered by noise usually that can not distinguish the signal sent by the same sound source. Using the Fisher identification and determine rule, it can spot the signal received by two microphones with the sound source exactly, exclude interference noise, calculate using time delay estimation, to ensure locating or tracking detection exactly. Table 2 is the result by positioning whistle in five unit array Outdoor, no determination through Fisher criterion; Table 3 is in same environment, the data after Fisher criterion.

Table 2. Experimental data 1

	Azimuth φ (º)	Elevation γ (º)	Distance r(m)
Actual value	235. 7	30. 2	2. 1
Targeting values	221. 4	31. 0	1. 9
	247. 9	30. 6	2. 8
	216. 5	27. 3	2. 3

Table 3. Experimental data 2

	Azimuth φ (º)	Elevation γ (º)	Distance r(m)
Actual value	235. 7	30. 2	2. 1
Targeting values	231. 5	30. 6	2. 2
	237. 4	31. 3	2. 4
	236. 1	28. 2	1. 9

It is can be analyzed by the table data that the location results improved to a certain extent after Fisher criterion. Fisher criterion can make microphone identify the target better, thus using delay Algorithm can make a exactly calculation which quickly and easily, and there are advantage compared to other methods.

5. Conclusions

In this paper, using wavelet analysis of Fisher identification and modulus-maxima method combines local energy band feature extraction we extract and classify the target signal on the role of the microphone signals. By finding the target signal strictly and accurately, it make delay estimation algorithm eliminate the extraction signal inaccuracies caused by the problem of inaccurate positioning, while improving the precision of the system, with some engineering practice. This method is not the measures that determine whether a single a point of comparison voltage reaches a predetermined value, reducing the possibility of mistaken identification of the target, and increases the likelihood of the election to the same sound source. Fisher identification provide a reliable guarantee for the algorithm of delay estimation algorithm with more accurate and reliable measured data.

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HEVC Motion Estimation Algorithm on Motion Homogeneity

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Abstract
In view of highly complicated calculation problems of inter-frame motion estimation of high efficiency video coding (HEVC), a fast termination algorithm of motion estimation on motion homogeneity was proposed. The motion homogeneity of the same object in video sequence was adopted to make a reasonable selection for the division method of current coding unit (CU) and end motion estimation of partly less likely complex division mode in advance. Under the current cursive depth, the CU motion difference was adopted to determine whether the current CU was the similar motion regions. For CU in the similar motion regions, after decomposition of next recursive depth small prediction unit (PU) splitting motion estimation was cut to reduce amount of calculation of motion estimation and lower calculation complexity. The results show that compared with the original HEVC encoding algorithm, the proposed algorithm can reduce en-coding time by 41.79% and 41.98% on average with peak signal-to-noise ratio(PSNR) loss of 0.052dB and 0.041dB in the low-delay and random-access cases.
Key words: HEVC, MOTION ESTIMATION, MOTION DIFFERENCE