Research on Positioning Accuracy of Passive Acoustic Positioning System Based on Feature Matching in Air

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Abstract. Passive acoustic localization is an important interdisciplinary research branch which combines acoustics, communication and signal processing. Passive acoustic positioning system, as an important part of battlefield sensing and monitoring system, has been paid more and more attention. Due to the disturbance of battlefield and other environments and the existence of measurement errors, the positioning accuracy often cannot meet the requirements. Passive acoustic positioning system, as an important part of battlefield sensing and monitoring system, has been paid more and more attention. Due to the disturbance of battlefield and other environments and the existence of measurement errors, the positioning accuracy often cannot meet the requirements. In multi-sensor passive location, the measured data of each array element is not only related, but also compensatory and redundant, so some people use the optimal data fusion algorithm as post-processing algorithm to improve the positioning accuracy. In this paper, the positioning accuracy of ground passive acoustic positioning system is analyzed based on feature matching method, and some useful conclusions are drawn.

Keywords: Passive acoustic location, location accuracy, sensor.

1. Introduction
Passive acoustic location is an important technology in the acoustic composite fuze technology of anti helicopter weapon system, which includes passive acoustic direction finding and passive acoustic range finding. The accuracy of direction finding and ranging is affected by many factors such as measurement environment interference and measurement system error, which is difficult to meet the requirements [1]. If high-precision sophisticated weapons are compared to spears, then the study of passive acoustic location is equivalent to a hard shield, defending the country and the people [2]. As an important part of battlefield sensor monitoring system, passive acoustic positioning system has been paid more and more attention [3]. After receiving the echo or echo, these signals are analyzed and processed to obtain the azimuth angle, running speed, distance and other parameters of the detection target [4]. The linear acoustic array passive target location method can not accurately determine the direction, while the plane cross array passive target location can basically determine the azimuth and elevation angle of the target, but the positioning accuracy is low, and when the elevation angle of the target is large, the error is increasing [5]. Sonar technology and radar technology are the main methods of target location in military field. Active detection is the main method for them to complete target detection. Radar uses the
transmission and reception of electromagnetic wave to detect, while sonar uses the transmission of acoustic wave and the reception of echo to detect [6].

Passive acoustic positioning in air is developed from underwater acoustic positioning, but because the speed of sound wave in the air is generally about meters per second, compared with supersonic aircraft and intercontinental missiles and other flying objects that are often times the speed of sound, it is not very important to completely use underwater acoustic positioning theory for positioning [7]. For the development of anti helicopter and anti tank passive acoustic detection intelligent mine, people have designed various small array positioning schemes in the air to estimate the target azimuth [8]. In the modern battlefield with precision guided weapons as the main strike force, the importance of artillery position reconnaissance is more and more prominent [9]. As early as during the Second World War, the rapid development of science and technology and the completion of the industrial revolution brought about earth shaking changes in the form and tactics of war [10]. In the early 1970s, artillery location detection and calibration radar rose and played an important role in the Gulf War and other local wars. But with the application of electronic jamming, anti radiation missile and stealth technology, they are facing serious challenges. To analyze the factors that affect the positioning accuracy of passive acoustic positioning system and find out the way to improve the positioning accuracy is one of the key problems that must be solved in passive acoustic positioning technology [11]. Based on the feature matching method, this paper analyzes the positioning accuracy of passive acoustic positioning system, and draws some useful conclusions.

2. Passive acoustic location based on multi element spatial acoustic sensor array

Using the transducer array arranged in a certain array, the spatial signals are collected, and then the collected data are processed, usually by the way of weighted summation, and the output result of one beam is obtained. The multi-dimensional acoustic sensor array positioning algorithm is not only related to the number of acoustic sensors, but also has an important relationship with the spatial arrangement and distribution of acoustic sensors. Appropriate arrangement and distribution can not only simplify the calculation process and reduce the computational complexity, but also help to improve the positioning accuracy. The acoustic path difference between acoustic sensors is converted into the time delay between acoustic sensors according to equation. Compared with the three-dimensional planar acoustic sensor array positioning algorithm, the five-dimensional spatial acoustic array passive acoustic positioning algorithm requires a high degree of time synchronization between acoustic sensors, but the complexity of the whole algorithm is not high, and most of them are addition and subtraction operations between numerical values, so that the position coordinates of the sound source in the acoustic sensor array can be obtained. As far as the location algorithm is concerned, it is difficult to get the theoretical variance of all parameter estimation because there are many steps in parameter estimation. The processor that completes the beamforming process is called a beamformer. The beamformer that processes narrowband signals is called narrowband beamformer, while the beamformer that processes wideband signals is called wideband beamformer. For the narrowband beamformer, when weighting the data of each array element, selecting different weighting vectors can enhance the signal in a certain direction, or suppress the signal in this direction to effectively improve the signal-to-noise ratio.

According to the cumulative effect of errors, the more the front-end estimation in the estimation process, the greater the impact on the final parameter estimation accuracy. Therefore, the estimation performance of time and time gradient is analyzed first. Different array structures, changes in the number of elements and different beamforming algorithms will change the spatial filtering characteristics of transducer arrays. However, the formation structure can't be too complicated, and because of the influence of machining accuracy and installation errors, the formation structure can't meet the ideal requirements, and the number of elements can't be increased indefinitely. Therefore, improving the algorithm under the fixed array has become a main way to improve the performance of array signal processing. When an interfering target appears near the measured target, due to the correlation between the jammer signal and the target signal, the shape of the correlation peak of the correlation function will be affected to a certain extent, even distorted, so that the accurate time delay estimation of the target
signal cannot be obtained. The spatial characteristics of the array depend on the composition and structure of the array elements. When the array elements with different orientations form different formations, the array will show different spatial characteristics. Under different application backgrounds, different beamformers are designed to meet different system requirements by using relatively fixed formations. According to the implementation, beamformers can be divided into time domain beamformers and frequency domain beamformers. If the position of a group of sound pressure sensors is known, we can accomplish passive location by using the time when the shock wave reaches each sensor.

Image rotation mainly refers to the process of forming a new image by rotating the selected image around a certain point, which is usually the center point of the image. After rotating \((x_0, y_0)\) clockwise by \(\alpha\) degrees, the new coordinate position \((x_1, y_1)\) is shown in Figure 1.

Figure 1. Rotation diagram

Among them, \(r\) is the length from \((x_0, y_0)\) to the origin, and \(b\) is the angle between \(r\) and the \(x\) axis. Before and after the picture is rotated, \(r\) will not change. Suppose the coordinates of point \((x_0, y_0)\) in the figure are:

\[
\begin{align*}
x_0 &= r \cos b \\
y_0 &= r \sin b
\end{align*}
\]  

(1)

The coordinates after rotation are:

\[
\begin{align*}
x_1 &= r \cos(b - a) = r \cos b \cos a + r \sin b \sin a = x_0 \cos a + y_0 \sin a \\
y_1 &= r \sin(b - a) = r \cos b \cos a - r \sin b \sin a = -x_0 \cos a + y_0 \sin a
\end{align*}
\]  

(2)

The coordinate system of the rotation schematic diagram is different from that of the image, and the conversion relationship between the two coordinate systems is shown in Figure 2.

Figure 2. Coordinate system interchange
In practice, it is difficult for each array element to have the same directivity and flat frequency response due to production and processing problems, so the sensitivity of the array element is related to the position and frequency in practice. The azimuth estimation of target aircraft will be affected to a certain extent, resulting in a certain degree of error, and even a large deviation may occur, thus losing its application value. In the case of narrow band, the frequency domain beamforming processing is to carry out complex weighted summation on the collected signals to obtain complex beam output. If the collected data is in non-complex form, it needs to be converted into real part and imaginary part by orthogonal demodulation, and then weighted and summed respectively for calculation. Because there is a time difference when each signal arrives at the array element, time domain beamforming processing is realized by weighting and summing the time delays. A supersonic moving target generates shock waves in the air. In a short distance, we can approximately think that the target moves in a straight line at a constant speed, and the shock wave generated by it is a standard conical surface [13]. The velocity and flight path of the target are determined by the time when the shock wave reaches each sound pressure sensor. Time-domain beamforming processes signals continuously, which ensures the continuity of processing results, but the amount of data processing is large. However, the data processing capacity of frequency domain beamforming is relatively reduced, but it is difficult to guarantee real-time performance by block processing.

3. Analysis of integrated positioning error

The larger the signal-to-noise ratio is, the closer the time delay estimation is to the true value, and the smaller the error of target azimuth estimation is. Therefore, in actual combat, if the distance between the simultaneous targets and the array is quite different, the signal-to-noise ratio is also large, and the method based on time delay estimation can be used to estimate the azimuth of the closer targets. With the decrease of signal-to-noise ratio, the error of time delay estimation will increase. Through the research in section, it is found that the pitch angle estimation of the target sound source is seriously distorted by the five-element spatial acoustic array passive acoustic positioning algorithm, and because the estimation performance of the first-order discrete Gaussian wavelet transform time delay estimation algorithm is poor, the introduction of this algorithm into the acoustic array for passive acoustic positioning will cause more serious distortion. When different frequencies and sounds exert pressure on the capacitance diaphragm, the capacitance value will change, and then the charge between the plates will change, thus realizing acoustic-electrical conversion. Although this microphone is widely used, it is difficult to design the microphone with small volume, and its sensitivity can not meet the demand of high-precision acquisition and system application.

In the process of first-order discrete Gaussian wavelet transform, many convolution operations and Fourier transforms are involved. This algorithm increases the complexity of hardware operation in hardware implementation, and the passive acoustic localization algorithm of multi-dimensional acoustic sensor array requires parallel processing of multi-channel signals and strict time synchronization, which further aggravates the burden of hardware processing. The main difference between electret condenser microphone and traditional condenser microphone is that the diaphragm of electret condenser microphone adopts a material with permanent polarization charge to replace the polar plate of condenser microphone. Therefore, the size of the microphone is reduced, the sensitivity is greatly improved, and the microphone has better consistency. Since the azimuth estimation is calculated from the first three time delay estimates, the results presented with the change of angle should be related to all three. Algorithm error, system error and random error, among which the latter two errors have great influence on system accuracy. The algorithm errors are mainly caused by different time delay estimation algorithms and different deployment methods, which can be reduced by adopting reasonable deployment and high-precision time delay estimation.

As the target machine gets farther away from the array and the jammer gets closer, the SNR will decrease continuously, so the error will increase. The frequency response and directivity of microphone are two important performance indexes, which are directly related to the reliability of the results given by the sound source localization algorithm, so it is necessary to test whether these two indexes meet the
system requirements. Data acquisition, visual positioning and data receiving modules transmit data through interfaces. The data format is shown in Table 1.

| Name          | Length |
|---------------|--------|
| Start flag    | 7      |
| Data length   | 6      |
| Command word  | 9      |
| Data part     | 12     |
| Termination code | 6     |

Table 1. Data format

In practice, it is difficult for each array element to have the same directivity and flat frequency response due to production and processing problems, so the sensitivity of the array element is related to the position and frequency in practice. Figure 3 shows the planning results of critical chain method.

With the in-depth study of wavelet analysis theory, it is found that the traditional wavelet analysis is developed from Fourier analysis, so it is limited by Fourier analysis to some extent. If wavelet analysis and multi-resolution analysis are collectively called the first generation wavelet, a new wavelet analysis method which does not depend on Fourier transform is proposed, which is called lifting wavelet. However, the addition of two acoustic sensors brings great challenges to practical application or programming, and even time synchronization between sensors. Therefore, blindly increasing the number of acoustic sensors may not improve the positioning accuracy, which is often counterproductive. Lifting wavelet not only retains the analysis characteristics and advantages of the first generation wavelet, but also overcomes the invariance of translation and expansion. Lifting wavelet improves the algorithm of the first generation wavelet. By analyzing the shortcomings of the first generation wavelet, the wavelet transform is simplified and improved [14]. The signal output by the microphone is a single-ended signal, so it is necessary to provide a bias voltage to the microphone when using the microphone. The system hardware design described in some documents does not consider the influence of the microphone bias voltage on the signal, but directly adopts the conventional DC power supply. In order to get accurate target location by multivariate spatial acoustic sensor array location algorithm, we need to consider on the other hand. In multivariate spatial acoustic sensor array passive acoustic location algorithm, we also need to estimate the time delay between target sound source and different acoustic sensors.

In the normal state, set the load learning cluster \( \{c_n, u_n\} \), introduce the cluster data in the high-dimensional space according to the nonlinear function \( \gamma(c) \), and obtain the linear expression of the regression function:

\[
g(c) = E^T \gamma(c) + m
\]
In the formula, \( E \) is the priority coefficient quantity, and \( m \) is the introduced variation setting quantity. Using DBZ calculation to optimize the regression target coefficient group is:

\[
\begin{align*}
\min & \|E\|^2 + \frac{1}{2}V \sum_{n=1}^{i} h_n^2 \\
& \left\{ u_n - E^T \gamma_{(c)} + m = r_n \\
& n = 1,2,\ldots,i \\
& \right. 
\end{align*}
\]  
(4)

In the formula, \( V \) is the reverse misjudgment coefficient; \( r_n \) is the deficit entity quantity. Do two-factor directional optimization conversion, you can get:

\[
A(e, n, h, f) = \min \|e\|^2 + \frac{1}{2}V \sum_{n=1}^{i} h_n^2 + \sum_{n=1}^{i} \left( e^T \gamma_{(c)} - n + r_n - u_n \right) 
\]  
(5)

For the amplifier circuit, the most important parameter is the gain. Generally, the voltage range of the analog-to-digital conversion chip at the back end is about 5 V, so the output signal of millivolt level can be guaranteed not to be clipped when the maximum gain is 60dB. However, due to the different angle or distance between the same sound source signal and microphone, the amplitude of the electrical signal will be different. Therefore, it is necessary to adjust the gain to meet the changing signal amplitude, so that the signal can keep high amplitude without clipping. The output signals of the receiving array subsystem are analog signals, which are directly sampled by oscilloscope. The output signals are visual, but the signal analysis is very inconvenient, so it is necessary to convert the output analog signals into digital signals for convenient analysis and processing. According to the principle analysis of lifting wavelet, compared with the first generation wavelet analysis and multi-resolution analysis, lifting wavelet has simple algorithm and fast processing speed, so it is suitable for parallel processing, and has small memory requirement, which is convenient for DSP or other hardware chips to realize [15]. There is no amplitude and phase difference between all signals, but in actual situation, the consistency of array elements and electronic components will make the amplitude and phase of signals shift. Therefore, it is necessary to test the amplitude and phase consistency of the receiving circuit and array element to verify whether the hardware design meets the consistency requirements. No matter from the error analysis of the five-element spatial acoustic sensor array positioning algorithm or the seven-element spatial acoustic sensor array positioning algorithm, the estimation of time delay is in a very important position in the passive acoustic positioning of the multi-element acoustic sensor array. The accuracy, accuracy and standard deviation of its estimation directly affect the accuracy and accuracy of the location of the target sound source.

4. Conclusions

When the signal-to-noise ratio is low, the direction estimation error of the target is large, and the meaning of estimation is lost. When the signal-to-interference ratio is large, the estimation error of the direction angle of the target is small, which is basically feasible. With the increase of the azimuth angle of the interfering target, the positioning error of the target also increases. The spatial characteristics of the array depend on the composition and structure of the array elements. When the array elements with different orientations form different formations, the array will show different spatial characteristics. In the actual environment, noise is often unpredictable, and it is another factor that affects the performance of wavelet time delay estimation algorithm. Under different application backgrounds, different beamformers are designed to meet different system requirements by using relatively fixed formations.

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