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Two-dimensional planar sound source localization based on microphone array

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Two-dimensional planar sound source localization based on microphone array

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Abstract. In recent years, people's awareness of wildlife protection has been increasing, and the demand for wildlife observation and recording has increased. However, the existing sound source localization system needs to continuously collect sound signals and convert the time-domain signals of the sound into frequency-domain signals. These shortcomings will increase the power consumption of the device, adversely affect the battery life, and the sound source localization result is easily interfered by the echo of environmental noise and target sound waves. In this study, the algorithm and hardware are improved on the basis of the existing sound source positioning system, which can improve the positioning accuracy and reduce the power consumption and cost of the whole machine. The adaptive circuit is used to filter and adjust the sound signal in real time to eliminate the influence of the difference between the individual microphones in the microphone array, which makes the sound source localization system more robust to the environmental noise and the echo of the target sound wave, and more suitable for the forest scene of wildlife protection.Through the research of this sound source localization system, we hope to provide some new ideas for producing economical and practical sound source localization equipment and lay a foundation for the intelligent development of wildlife protection in the future.

1. Introduction

Due to the high-speed development of computer technology and artificial intelligence, sound source localization technology has been greatly improved in recent years. For the indoor sound source localization system using small size microphone, the feature vector can be constructed by the sound intensity of the target sound wave, and the characteristic information of the target sound wave can be extracted. At present, sound source localization technology is widely used in wildlife protection. For example, passive acoustic monitoring technology composed of directional pick-up array was used to monitor the activity of western black crested gibbons [1], and the sound of birds singing was collected by tape recorder to locate birds [10].Although the accuracy of sound source localization technology applied in the field of wildlife protection can meet the demand at present, these schemes have the following disadvantages: the device needs to collect sound signals for a long time and carry out a lot of data processing on the collected signals so as to obtain accurate sound source localization results. Continuous sound acquisition will increase the power consumption of the equipment, while a large number of data analysis tasks put forward higher requirements on the data processing ability of the equipment, and the above two defects will increase the overall manufacturing and maintenance cost of the equipment. In addition, the existing sound source localization scheme needs to arrange many sets of complex sound collection devices in the habitat of wild animals [1], which has the risk of affecting the normal life of animals.

In order to improve the localization of sound sources, the researchers usually adopt methods such as manual noise removal, noise removal with low signal-to-noise and even deleting poor data [8]. Although these methods can effectively improve the accuracy of sound source localization, the manual correction of original data has a large workload and low efficiency, which can not be popularized in a wide range. More importantly, it takes a long time to correct the original data manually, which makes the results of acoustic source localization lack timeliness.

In order to solve the shortcomings of the above-mentioned positioning device, the sound source positioning scheme based on microphone array in this study can obtain the positioning data directly in the time domain by using the hardware circuit to process and analyze the signals in real-time. This scheme is low-cost and easy to operate, and has the following advantages:

(1) The device does not need to collect the sound continuously, it only needs to collect the target sound in a short time.

(2) Real-time adjustment of sound acquisition threshold by hardware has better robustness to environmental noise.

(3) The algorithm is simple and reliable, and the operation ability is low.

(4) The real-time analysis of sound signal by hardware can get the sound source localization results in a short time, which makes the data have strong timeliness.

(5) The sound source location device is simple and reliable in structure and low in power requirement. It is suitable for use in forest scene without fixed power supply.

In this study, the sound source localization system of microphone array is studied systematically, which is expected to provide more practical and lower cost selection for sound source localization technology, and lay a foundation for the intelligent development of wildlife protection in the future.

2. Main research content

2.1. Hardware description

This sound source location system uses LM358P audio operational amplifier chip, electret microphone and a small number of peripheral components to form sound source acquisition and conversion module (hereinafter referred to as acquisition module); LM393 voltage comparator chip, STC8A series MCU and a small number of peripheral components to form sound source analysis and location module (hereinafter referred to as analysis module). As shown in Figure 1: the hardware structure includes microphone, borderless filter, LM358P sound source amplifier, LM393 voltage amplifier and STC8A MCU.

After the electret microphone collects the target sound wave, the sound signal is converted to an electrical signal. Then, the passive filter in the acquisition module filters the electrical signals, and part of the ambient noise is filtered out. Subsequently, the LM358P audio operational amplifier chip amplifies the weak electrical signal, amplifies the amplitude of the voltage to an easily detected level, and filters it again through a passive filter. After that, the processed electrical signal is input to the LM393 voltage comparator chip, in which the electrical signal and the threshold voltage are continuously compared. When the amplitude of the electrical signal is less than the voltage threshold, the output of the voltage comparator maintains a high level. In this case, the STC8A MCU is in sleep state, and the whole sound source localization system runs continuously at low power. As soon as the magnitude of the electrical signal exceeds the voltage threshold, the voltage comparator outputs a low level. At this time, the STC8A microcontroller is triggered to change from low power consumption state to normal state and perform operation processing on the signal. At the same time, the microcontroller temporarily blocks the trigger signal output by the voltage comparator, and within a period of time, the microcontroller will not be triggered again. Since the target sound wave always reaches the microphone before the reflected wave, the microcontroller can block the trigger signal in a short time after being triggered by the target sound wave, which can effectively prevent the sound source localization result from being interfered by the reflected sound wave.

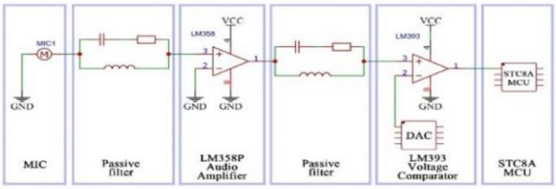

Figure 1. Hardware structure description.

2.2. Principle of sound positioning

The acquisition module completes the selection, filtering and characteristic signal acquisition of the sound waveform. The analysis module adjusts the target acoustic wave acquisition threshold of the four front-end analog sampling modules, adjusts the amplification amplitude of the input signal in real time, and only collects the wave crest signal required by the sound source positioning system.

When the microphone of the first acquisition module receives the target sound wave, the timer in the analysis module starts timing and measures the time T1, T2 and T3(where T1< T2< T3) of the remaining microphone level jump. Then the time difference △t1=T2-T1, △t2=T3-T2, △t3=T3-T1. Figure 2 shows the spatial position and distance relationship between the four microphones. Figure 3 is the schematic diagram of sound source positioning system: the trigger signals sent by four microphones are or processed and sent to MCU Timer Trigger. Because the propagation speed of sound in the air is relatively fixed at the same temperature, according to the geometric knowledge, the angular calculation formula can be obtained as follows:

$$\theta = \frac{1}{3}\sum_{n=1}^{3}\arcsin\frac{c\Delta t_n}{L} \tag{1}$$

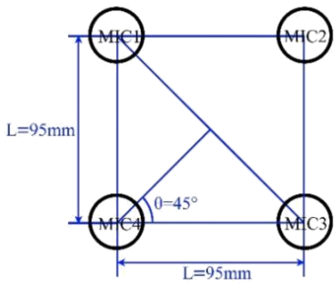

Figure 2. Microphone Location.

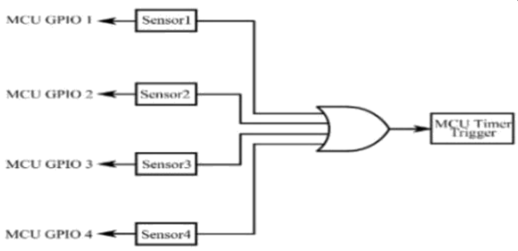

Figure 3. Schematic diagram of sound source positioning system.

2.3. Advantages of this method in system structure

In addition to simple structure and low cost, the sound source positioning system adopted in this paper has the following advantages:

(1) Use the hardware circuit to complete the selection, filtering and characteristic signal acquisition of sound waveform, so as to reduce the processor burden.

(2) When there is environmental noise but the target acoustic wave does not appear, although the front-end analog sampling module is in the working state, the power consumption is low [5]. Through calculation, the average power consumption of the front-end analog sampling module in the working state is about 7.95 mW [6]. When the single-chip microcomputer is in the standby low power consumption state, the average power consumption is about 1.32μW. By triggering the signal acquisition circuit through the hardware circuit, the processor does not need to continuously acquire the analog signal of sound signal, thus further reducing the cost and power consumption. This sound source location system has low power consumption and small operation amount of single-chip microcomputer, which is suitable for field observation equipment powered by battery.

2.4. Advantages of this method in algorithm

At present, the sound source localization algorithm applied to the microphone array can be roughly divided into three types[3]:

(1) Positioning algorithm based on high-resolution spectral estimation;
(2) Based on controllable beam sound source localization algorithm;
(3) Location algorithm based on arrival time difference (TDOA);

The method of sound source localization in this paper is mainly based on TDOA algorithm, combined with inverse trigonometric function and angle range judgment, the accurate localization results can be obtained. The algorithm of this sound source localization system is simple, the calculation quantity is small and the accuracy is high (the error of the sound source position angle is less than 4.59°). Compared with the method based on controllable beam positioning, the algorithm has the advantage of real-time adjustment of the amplification amplitude of the input signal. By using the hardware target acoustic trigger technology, only the wave crest signal required by the acoustic source positioning system is collected, which reduces the interference caused by noise and acoustic reflection to the accuracy and has better anti-noise performance. When the number of microphones is equal, the accuracy is higher.

3. Work flow of sound source positioning system

After the start-up of the sound source positioning system, the acquisition module sends the amplified and filtered acoustic analog voltage signal to the analysis module. The sound processing module is composed of sound source acquisition and conversion module, sound source analysis and positioning module, and the connection relationship is shown in Figure 5. Under the control of microcomputer, the voltage comparator in the analysis module compares the acoustic voltage signal with the set threshold voltage in real time. By analyzing the output signal of the comparator, the single-chip microcomputer determines the noise intensity of the current environment, calculates the target acoustic threshold voltage according to the noise intensity, and controls the digital-to-analog converter to lock the target acoustic threshold voltage and transmit it to the voltage comparator. Then the single-chip microcomputer enters the standby low-power state, and the front-end analog sampling module continues to sample the acoustic signal. The system work flow chart is shown in Figure 4.

When the acquisition module collects the target acoustic wave, the acoustic wave signal voltage is greater than the target acoustic wave threshold voltage. The voltage comparator will trigger the single-chip microcomputer in the analysis module to change it from the standby low-power state to the working state, and start to calculate the phase difference of the acoustic wave signal collected by the four acquisition modules, and calculate the time difference of the target acoustic wave arrival to obtain the sound source positioning result. After the sound source localization results are obtained, the single-chip microcomputer sets the target acoustic wave threshold voltage of the comparator again according to the environmental noise, and then the single-chip microcomputer enters the standby low power state and waits for the next trigger.

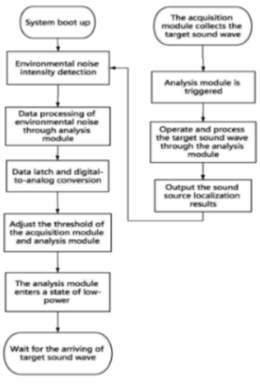

Figure 4. System work flow chart.

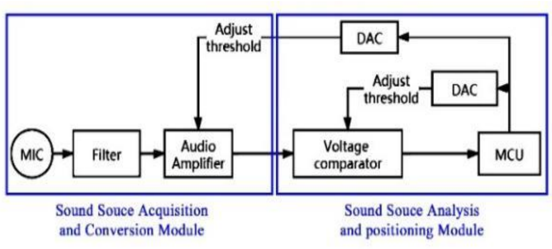

Figure 5. Module composition and connection diagram.

4. Analysis of location error and accuracy of sound source

4.1. Accuracy analysis of this sound source positioning system.

Calculation formula according to angle θ:

$$\theta = \arcsin\frac{c}{Lf} \tag{2}$$

Take the test temperature $t = 20℃, a = 0.0975\text{m}, v = 340 \text{ m/s}, f = 700\text{kHz}$, get $\theta = 0.285°$.

4.2. Accuracy analysis of other sound source localization methods.

In the microphone array-based sound source localization method, the distance error of sound source position is within $2m$ and the deviation percentage is between 0 and 10% through TDOA algorithm[4]. In the beam scanning sound source orientation method based on Bark subband, the angle estimation root mean square error is controlled within 7°. When the signal-to-noise ratio is greater than 10 dB, the angle estimation root mean square error is less than 2°[11].

5. Test of sound source positioning system

5.1. Test procedure of sound source positioning system

(1) Place a loudspeaker with a diameter of 100mm at 1m in front of the sound source positioning system, so that the angle between the loudspeaker and the geometric center of the sound source positioning system is $\alpha = 0°$. Figure 6 shows the waveform changes of Analog signal, Trigger source and MIC 1 and MIC 2 displayed by oscilloscope.

(2) Play the sound pulse signal with the frequency of 700kHz and the sound intensity of 80dB to the sound source positioning system by using the loudspeaker.

(3) Change the angle α between the loudspeaker and the geometric center of the sound source positioning system, and repeat steps (4) and (5) until 10 groups of data of 45°,90°,135°,180°,225°,270° and 315° are collected respectively.

Figure 6. Output waveform of hardware trigger circuit.

5.2. Data analysis

By using this sound source localization system to carry out the experiment, test the sound source and microphone matrix at different distances in 0°,45°,90°,135°,180°,225°,270° and 315° directions respectively, measure the time difference of the level jump of each microphone, and substitute it into Formula (4), and then judge the actual angle range simply according to the order of the level jump of each microphone, get the specific angle of the sound source calculated by the system, and compare it with the theoretical angle.

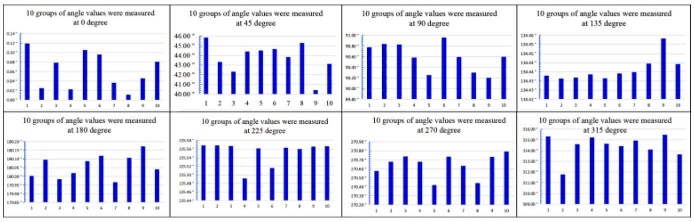

Figure 7. Comparison between actual angle and measured angle deviation histogram.

6. Conclusions and prospects

6.1. Conclusion

According to the analysis of verification experimental data, the error of the position angle of the sound source obtained by this sound source localization method is within 4.59°. By using digital potentiometer and voltage comparator to adjust the amplification amplitude of input signal in real time, it can avoid the influence of amplification circuit distortion on sound source location caused by excessive sound intensity. It can be applied to real environment with many environmental noise. The sound source signal can be processed by hardware circuit of the system to reduce the influence of external factors. Therefore, the sound source positioning system has higher precision and practicability.

6.2. Outlook

Acoustic localization based on microphone array is a research with wide application background. It is widely used in monitoring system, intelligent robot, video phone, hearing aid, video conference, robust speech recognition and other fields [2].In recent years, people's awareness of wildlife protection has increased, and the demand for wildlife observation and recording has increased. This sound source location system is practical, convenient and low-cost, and can meet the needs of most wildlife protection workers.

Acknowledgments

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