Proceedings of the 2nd International Conference on Computing Innovation and Applied Physics (CONF-CIAP 2023)
DOI: 10.54254/2753-8818/5/20230569

Sound source signal location and tracking system based on STM32

Xingcheng He

Hunan Agricultural University, Changsha, Hunan, China, 410000

he18931915579@163.com

Abstract: With the progress of science and technology, sound source localization technology has a wide range of applications in urban traffic, digital hearing AIDS, mechanical systems and other fields, especially in noise monitoring and other aspects. In order to solve problems such as failure risk, this technology can accurately determine the location of the problem. At present, low-end manual technology is used in many fields, which is inefficient and a waste of time. Therefore, this paper designed a sound source signal positioning and tracking system based on STM32. The system takes STM32F407VET6 as the control core, and the speaker as the sound source that can emit self-defined regular sound. After the amplification circuits, it is sent to the microcontroller for processing. After sampling, the distance and offset Angle between the test point and the sound source are calculated by calculating the time deviation of the sound signal collected by the two microphones. Then control the steering gear rotation and laser alignment to realize the positioning and tracking of the sound source, and display the distance and offset information between the test point and the sound source in the terminal. With the development of modern technology, sound source localization can be widely used in daily life, and with the development of machine learning, cloud computing and electronic technology, this technology will have a broader application prospect.

Keywords: STM32F407VET6; Time Difference of Arrival (TDOA) Estimation; device design; Low pass filter; Positioning means

1. Introduction

Whether in the civil or military fields, it plays an important role in many fields .However, compared with the research progress abroad, the research on sound source localization technology in China started late, but still made significant progress. However, compared with foreign research, the accuracy of the current sound source localization algorithm in our country is not high, and there are still many imperfections in the algorithm, and the application in the real system is not much [1]. In this paper, TDOA algorithm is used to improve the sound source localization method of the same phase shift delay. Hardware design amplification chip uses more precise OPA2333 chip, making the data collected by STM32 more accurate, so as to locate and track the sound source. And through the experiment, the collection of test data, with good real-time and intelligent. This technology has received the attention of many countries, and the research on target localization based on sensor array has very important practical significance [2]. It will show great superiority and market potential after further practical application.

2. Methodology

2.1. Research proposal

2.1.1. System main control module scheme demonstration and selection. Scheme One: STM32F103ZET6 processor is used. Its core is Cortex-M3, the maximum operating frequency of 72MHz. High performance, low cost, low power consumption, relatively high function integration, low price, but the main frequency is low.

Scheme Two: STM32F407VET6 processor is used. The Cortex-M4 core has FPU single precision and supports all ARM single precision data processing instructions and data types. It integrates the new DSP and FPU instructions, and has a high performance of 168MHz, which can effectively improve the execution speed and code efficiency of the control algorithm.

After comprehensive consideration, this study chooses Scheme Two.

2.1.2. Demonstration and selection of sound source localization calculation scheme. Through the sampling array 400hz (signal frequency) sampling for a half period, a total of 300 points are collected and stored into an array, the subscript of the maximum value of the array is calculated, and the difference between the two subscripts is calculated to fit the function curve to determine the distance and offset Angle between sound sources.

*2.1.3. Hardware design and software selectio*n. Altium Designer is widely used in the education industry. PADS Layout and Candences software are more difficult to use and suitable for more professional personnel. Lichuang EDA is a domestic EDA introduced to the market in 2017, which collects online design, online simulation, project management, PCB customization, project open source community and other functions of web pages and clients. It has been widely used in the education industry and personal production [3].

Therefore, this study finally chooses Lichuang EDA.

2.2. System scheme design

The system is shown in Figure 1, firstly, the sound signal is collected through the microphone to transform into continuous electrical signal, the collected sound information is transformed into audio analog signal and amplified by the designed sound amplification circuit, and then the amplified signal is converted into digital to analog through the AD7606 data acquisition module and then entered the STM32F407VET6 for sampling processing. By calculating the signal delay and extracting the data from the display module after calculating the algorithm, the time deviation of the sound signal was calculated to calculate the distance and offset Angle between the test point and the sound source. Then the positioning and tracking module of the sound source was realized by controlling the rotation of the servo and the laser alignment.

3. Theoretical Analysis and Calculation

3.1. Near Field Data Analysis

The schematic diagram of the system distribution is shown in Figure 2, where C is the position of the sound source, A and B are the positions of the microphone head, $AE = 0.8$ m and O is the observation point in this system. When the position of point C changes, the Angle θ and the OC distance are calculated.

DOI: 10.54254/2753-8818/5/20230569

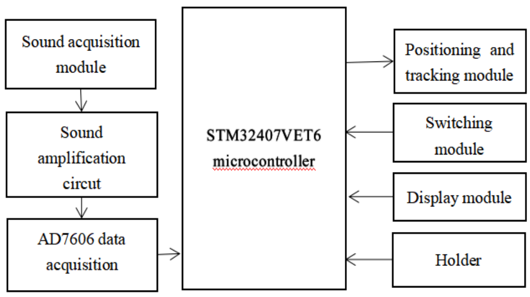

Figure 1. Schematic diagram of the system structure

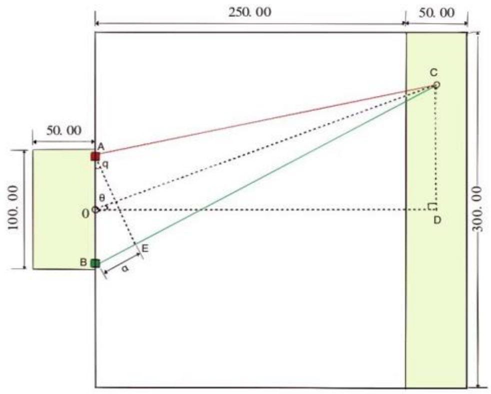

Figure 2. System distribution diagram

In Figure 2, $|CE| - |CA|$, Since the Angle θ is small, it can be approximately considered $AE \perp OC$, Therefore, the distance difference between the two microphones and the sound source is:

$$a = AB\sin q = 0.8\sin q \tag{1}$$

In the above equation, the distance difference can be calculated by sampling the phase deviation of the two sound signals, so we can deduce θ, thus, the distance of AC is calculated:

$$|OC| = \frac{2.75}{\cos q} \tag{2}$$

Error analysis: The above calculation process is approximated by calculating that when point C is at the edge point, the Angle θ takes the maximum value, and the Angle error at this time can be calculated. In this case, the theoretical value of θ can be calculated by Eq. 3:

$$\tan q = \frac{1.5}{2.75} \tag{3}$$

Calculated $q = 28.6°$

$$|AC| = \sqrt{2.75^2 + 1.1^2} = 2.9618 \tag{4}$$

$$|BC| = \sqrt{2.75^2 + 1.9^2} = 3.3425 \tag{5}$$

$$a = |BC| - |AC| = 0.3807 \tag{6}$$

It can be obtained by formula 1:

$$\sin q = \frac{a}{0.8} = \frac{0.3807}{0.8} \tag{7}$$

Calculated $q = 28.4°$, it follows that at the limit position, the Angle deviation is 0.2 degrees, which can be ignored, and at the rest of the positions, the deviation Angle is smaller, so Equation (1) can be used to calculate the Angle θ. And through the calculation of the phase difference $X = X1 - X2$, and finally get the Angle.

3.2. TDOA algorithm

The sound source localization algorithm based on TDOA has the characteristics of small amount of computation, simple algorithm, high positioning accuracy and low hardware cost, which makes the algorithm widely used in practice, and can realize real-time positioning. According to the system design requirements, the sound source is set as (x, y), and the coordinates of the receiving module for the ith sound are (x_i, y_i). Therefore, Equation (8) is satisfied between the transmitting sound and the collection system.

$$Riy = \sqrt{(x_i - x)^2 + (y_i - y)^2} \tag{8}$$

Then it can be known that Riy represents the distance difference of the ith receiving module of the transmitting sound, so the transmitting sound (x, y) and pickup (x_i, y_i) in hyperbolic positioning satisfy Equation (9).

$$\mathrm{Riy} = \sqrt{(x_i - x)^2 + (y_i - y)^2} + \sqrt{(x_v - x)^2 + (y_v - y)^2} \tag{9}$$

Using the method of phase detection, the phase difference between two sinusoidal signals is converted into a voltage signal, and the phase difference can be reversely calculated by using ADC to measure the voltage signal, so that it can be transformed into an equation solution problem to realize the localization of sound source [4, 5].

4. Circuit and Program Design

4.1. Device design process

Firstly, the sound signal of fixed frequency is played by the self-made sound source, and then the signal is collected by the pickup and the voltage difference is generated for easy reading. After the current bias, the signal is input into the capacitor filter circuit, as shown in Figure 3, the capacitor filter circuit is known to filter out the noise above 600Hz, and set R3=10KΩ according to the actual situation. According to the filtering formula $f = \frac{1}{2} * \pi * R * C$, the capacitance can be calculated as 33nF, and the noise at other frequencies will be filtered out. After filtering, the signal will enter the amplifier circuit. However, the signal itself is very weak, and the measurement accuracy of the measurement system is limited by the size change of the weak signal itself and the influence of the amplifier amplification noise.

Generally, the accurate method of measuring the weak signal is to amplify the weak signal and measure it. Therefore, different industries have studied the weak signal amplifier circuit [6-8].We choose OPA2333AIDR double high precision op amp chip, the chip adopts TI high performance and high precision mixed signal CMOS manufacturing technology, using a single power supply. An auto-zeroing calibration technique provides near-zero drift as the temperature changes. Filter circuit output electrical signal is very weak, can not be directly displayed for these energy weak signals, further processing is also difficult, they must be amplified to a certain magnitude, for different applications to design different amplifier circuit, according to the actual input signal and the required output signal voltage and current, amplifier circuit is divided into four types: Voltage amplifier circuit, current amplifier circuit, mutual resistance amplifier circuit and mutual conduction amplifier circuit [9]. The input and output resistance, gain, cut-off frequency, frequency characteristics and output distortion are important performance indicators of the amplifier [10]. This paper designed an integrated operational amplifier OPA2333 as shown in Figure 4 as the core component, the differential amplifier circuit with high input impedance is multi-layered, so that the amplifier circuit has ideal common-mode ability and the ability to suppress signal drift, and the voltage gain accuracy is high. The waveform obtained by the simulation in EDA is also ideal, and then the components and PCB design are carried out in the standard mode. The electrical signal is filtered and input from J1 and amplified by 23,230 and 2300 times using the formula $U = (1 + \frac{R}{R/})$, respectively. The electrical signal obtained in this subject is mainly amplified by 230 times, that is, input from J1 and output from J5.

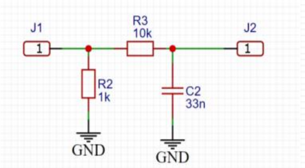

Figure 3. Capacitor filter circuit

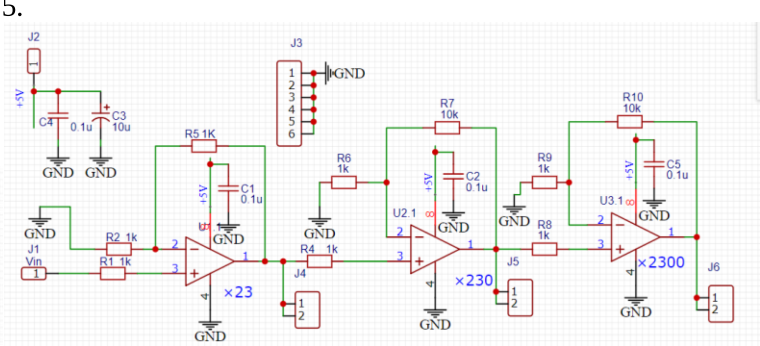

Figure 4. Circuit Diagram of Sound Signal Amplifying Circuit

DOI: 10.54254/2753-8818/5/20230569

After stabilization, the waveform is then sampled by ADC. In order to make up for the shortcomings of most sound positioning technology on the market, such as the use of radar search target facing electronic interference, imitation of human binaural sound source positioning in the scene of multiple interference will face the problem of accurate positioning, this system mainly uses TDDO algorithm, is that the system can achieve a degree of accurate positioning [4, 11], the simplified process is shown in Figure 5(a). The STM32F407ZET6 used also controls the different functions separately through the IO port, as shown in Table 1. The processor obtains data information through the TDOA algorithm, and needs to collect multiple groups of data for analysis during the whole modulation process of the system. The collected data are shown in the Table 2. Through the analysis of the data Table 2, the formula coefficient is obtained. After obtaining the position information, it is transmitted to the two-dimensional pan-tilt-head through the serial port. The laser positioning device waits for the platform to stabilize before turning on the laser pointing to the sound source. Thus, the specific position coordinates are also displayed through the OLED screen, so that users can consult the coordinate information more quickly and easily, the simplified process is shown in Figure 5(b)&(c) .

4.2. Device process example

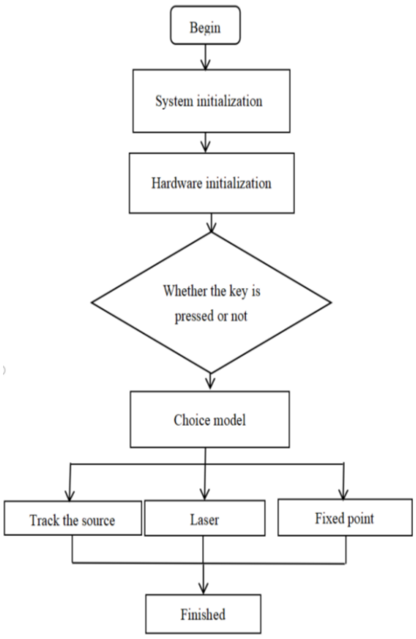

Figure 5(a). Main program design flow chart

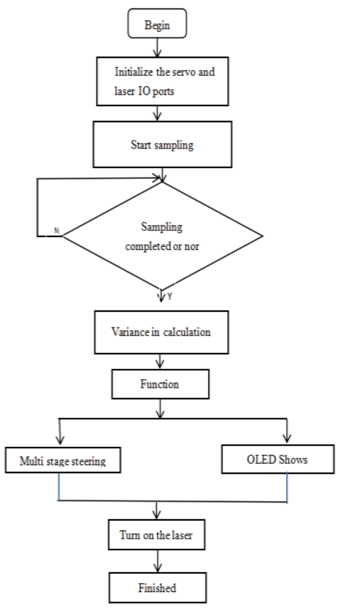

Figure 5(b). Track sound source, plus laser mode

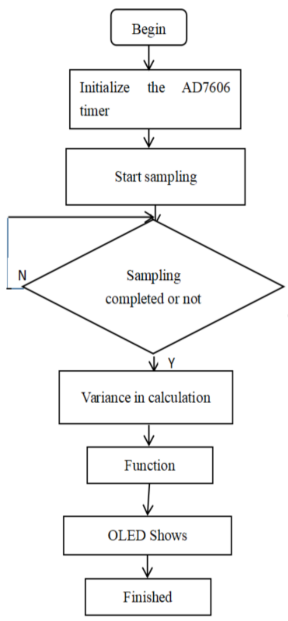

Figure 5(c). Fixed-point mode

4.3. Test plan and the test results

4.3.1. Test program. The true distance and Angle were measured manually after the sound source was placed inside the test area. After recording the actual data, the measurement device is started, the distance and Angle data collected by the device are recorded, and the above measurement process is repeated many times by changing the position of the sound source and recording the relevant data.

4.3.2. Test Data. As shown in Table 3, the functional test results, were obtained by measuring.

DOI: 10.54254/2753-8818/5/20230569

Table 3. Functional Test Results

Serial number	Test length (cm)	Error(cm)	Angle (°)	Error angle (°)
1	296	10	20	5
2	270	16	23	3
3	285	5	27	9
4	300	10	35	6
5	290	8	18	12
6	289	6	30	6

5. Conclusion

The system can locate the sound source position more accurately, so that it can be better applied in practice. However, mixed noise and three-way ADC acquisition still have some limitations, and the relevant information still has deviations compared with the accurate value.This would be a fruitful area for further work.

Appendix

Table 1. STM32 IO port allocation table

Peripherals	Reference port	Peripherals	Reference port
AD7606	PC0-3, PC10-11, PE4-6	debug interface	PA9-10
OLED	PE9-13	PTZ	PB9-8
key matrix	PD7-PD0	system warning	PD8

Table 2. Experimental data

<table>
<tr><th>Array element</th><th colspan="2">Phase shift different (peak different)</th><th colspan="2">Phase shift mean</th><th>Array element</th><th colspan="2">Phase shift different (peak different)</th><th colspan="2">Phase shift mean</th></tr>
<tr><td rowspan="3">00</td><td>95</td><td>199</td><td rowspan="3">98</td><td rowspan="3">194</td><td rowspan="3">20</td><td>97</td><td>232</td><td rowspan="3">96</td><td rowspan="3">235</td></tr>
<tr><td>96</td><td>192</td><td>102</td><td>239</td></tr>
<tr><td>104</td><td>191</td><td>90</td><td>236</td></tr>
<tr><td rowspan="3">01</td><td>76</td><td>208</td><td rowspan="3">75</td><td rowspan="3">210</td><td rowspan="3">21</td><td>103</td><td>210</td><td rowspan="3">103</td><td rowspan="3">207</td></tr>
<tr><td>75</td><td>209</td><td>103</td><td>207</td></tr>
<tr><td>74</td><td>214</td><td>103</td><td>204</td></tr>
<tr><td rowspan="3">10</td><td>195</td><td>83</td><td rowspan="3">204</td><td rowspan="3">84</td><td rowspan="3">30</td><td>75</td><td>220</td><td rowspan="3">76</td><td rowspan="3">224</td></tr>
<tr><td>212</td><td>89</td><td>78</td><td>225</td></tr>
<tr><td>206</td><td>82</td><td>75</td><td>227</td></tr>
<tr><td rowspan="3">11</td><td>158</td><td>77</td><td rowspan="3">169</td><td rowspan="3">78</td><td rowspan="3">31</td><td>74</td><td>232</td><td rowspan="3">73</td><td rowspan="3">230</td></tr>
<tr><td>169</td><td>85</td><td>74</td><td>229</td></tr>
<tr><td>181</td><td>73</td><td>74</td><td>232</td></tr>
</table>

References

[1] Buyun Tong. Research on Sound Source Localization Method Based on Time Delay Estimation Technology [D]. North China Electric Power University, 2019.
[2] Yishi Jia, Juncheng Wang, Yue Zhao, Huachen Ma. Sound localization system Based on STM32 [J]. Electronic World, 2021, (02):172-173
[3] Chengbin Ye, Xianyu Chen, Lingfeng Chen. Application of Lichuang EDA in University

DOI: 10.54254/2753-8818/5/20230569

Electronic Innovation Laboratory [J]. Microcomputer Application, 2012, 38(01):164-166+177.
[4] Xiaoyang Li, Shihao Li, et al. Sound source Localization and Tracking System Based on Raspberry Pi [J]. Science and Technology Innovation, 2022(32):33-36.
[5] Honghao Zhang, Yuhao Duan, Binbin Han. Research and implementation of sound source localization algorithm based on TDOA [J]. Journal of Tianjin University of Technology, 2020, 33(2):1-2.
[6] Changqing Li, Xinli Mei, Ming Qi. Implementation of weak light signal detection circuit [J]. Applied Optics, 2010, 31(05):724-727.
[7] Jiangfeng Wang. Research on Low Frequency Weak Signal Detection System Based on embedded System [J]. West China Science and Technology, 2015, 14(07):42-44.
[8] Yansheng Zhu, Yong Gao, et al. Simulation of phase-locked Amplifier Circuit for Weak Signal Detection [J]. Journal of Hunan University of Arts and Science (Natural Science Edition), 2023, 35(01):20-25.
[9] Zhuling Yin. Research on Low Frequency Small Signal Amplifier Circuit [D]. Beijing University of Posts and Telecommunications, 2012.
[10] Mingquan Wang, Jiaxin Yang, Tang Yu. Amplifying circuit characteristic test device design [J]. Journal of electronics, 2022, 30 (15): 59-63.
[11] Guochang Zhang. Research on robust acoustic source localization method [D]. Beijing. Institute of Acoustics, Chinese Academy of Sciences, 2019.