ISSN: 2414-1895 DOI: 10.6919/ICJE.202207_8(7).0079

Array Microphone Acoustic Imaging Instrument Design based on K210 Chip

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Abstract

Noise is more common in our daily life and industrial production, for the problem of how to find the source of sound in space more quickly, design an acoustic imaging camera instrument based on K210 MCU and digital microphone array. According to the intensity signal of the sound detected by different microphone units on the array microphone assembly, the distribution map which can distinguish the direction of the sound is formed by certain algorithm after correction and then overlapped with the video screen captured by OV2640 camera, so as to achieve the effect that the location of the sound source can be clearly seen on the display through the video screen. Experimental results show that the instrument can display the location of the sound source in real time and stable.

Keywords

Digital Microphone Array; K210 MCU; Sound Source Localization.

1. Overview

With the development of various maintenance industries, maintenance often some failure points will be accompanied by some noise, sometimes difficult to distinguish the source of sound through the human ear, for this type of maintenance failure to the maintenance staff brought some trouble and trouble. The acoustic imaging instrument studied in this paper can effectively solve this problem[1].

2. Hardware Circuit Composition and Design

The hardware of the system includes: core processor Kendryte K210, OV2640 camera module, digital array microphone module, 3.2-inch LCD, the hardware block diagram is shown in Figure 1.

The core processor of the hardware uses the Kendryte K210 chip, which is manufactured using TSMC's ultra-low power 28nm process and has a dual-core 64-bit processor. As shown in Figure 2, the core PCB board where the processor is located has integrated camera and LCD display line interface, which can facilitate the direct installation of the camera and display[2].The core function of K210 is machine vision and hearing, so it has good stability and reliability for visual and auditory processing, which can reduce the cumbersome design of the program.In addition, the K210 has a rich set of peripheral units to meet a large number of applications, its internal integrated audio bus I²S, can be connected to the audio processing module, the input can support up to 32-bit precision audio signal transmission, up to support the input of eight audio data streams, enough to meet the digital microphone multiple inputs at the same time.

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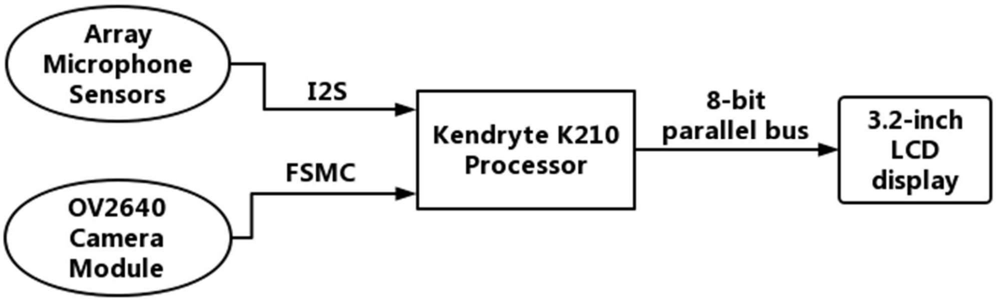

Figure 1. Hardware block diagram

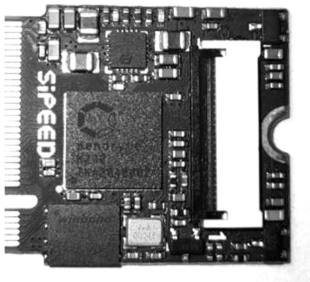

Figure 2. K210 Core PCB

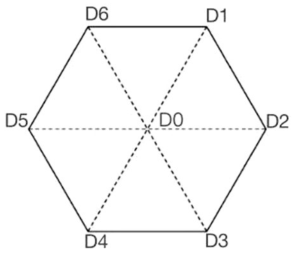

Figure 3. Distribution of seven digital microphones

As shown in Figure 3, the array microphone assembly consists of seven MEMS digital microphones, model INMP441, distributed in the center and at the vertices of a square hexagon, denoted D0 to D6. The digital microphone features a digital interface with high-precision 24-bit data, high signal-to-noise ratio of 61 dBA, sensitivity of -25 dBFS, stable frequency response from 60 Hz to 15 kHz, low current consumption of 1.4 mA, left and right channel selection, and a standard I^2S interface[3]. The microphone transmits data as a digital signal, which eliminates the need for analog-to-digital conversion and the need for a system audio codec, resulting in significant savings in development costs.

These seven microphones share the same serial clock SCLK signal line and the same frame clock LRCK signal line, where D0, D1, D3, D5 set for the left channel output, D2, D4, D6 set for the right channel output, the same serial data line can transmit the signal of the left and right channels at the same time, so that you can use four serial data lines to transmit the audio data of seven microphones, the specific circuit is shown in Figure 4.

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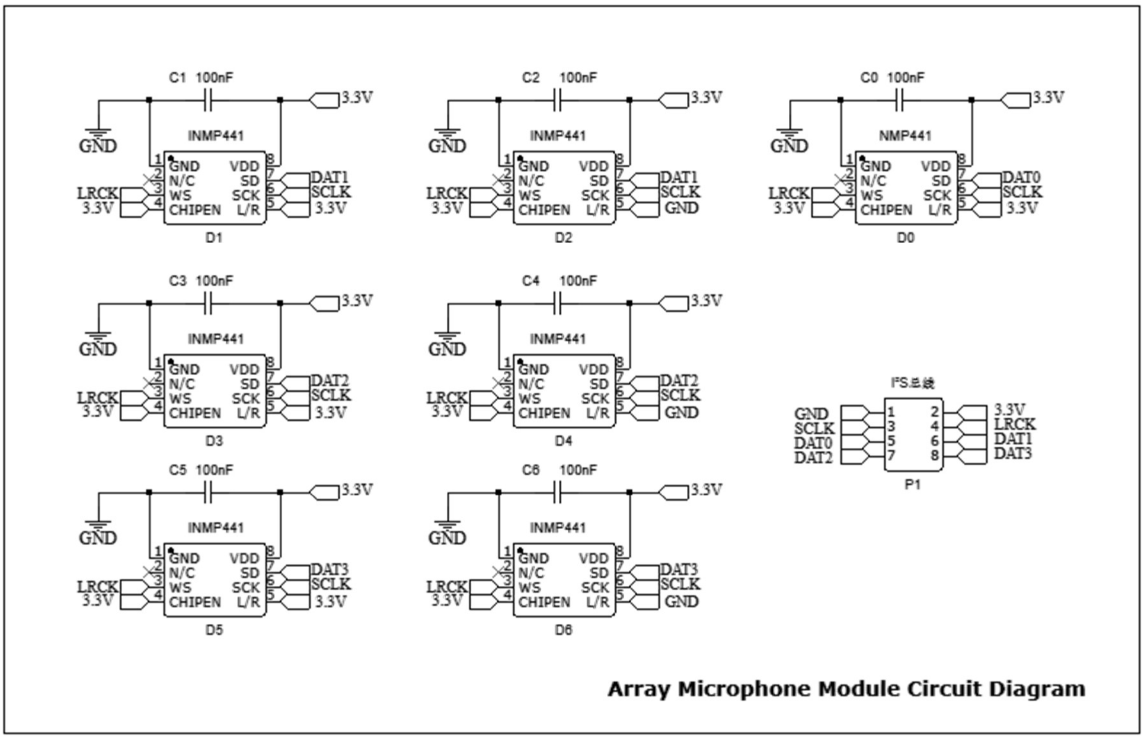

Figure 4. Array microphone circuit diagram

3. Theoretical Calculation and Analysis

The sound from different vertex positions of the spatial hexagon is sampled by the array microphone assembly, and the sound intensity data at the locations of these seven microphones are obtained after processing, and finally the sound direction is determined by vector synthesis.

L(1) to L(6) are the sound intensity captured by digital microphones D1 to D6, respectively, and the numbers 1 to 12 are used as the sound intensity level captured by each digital microphone.

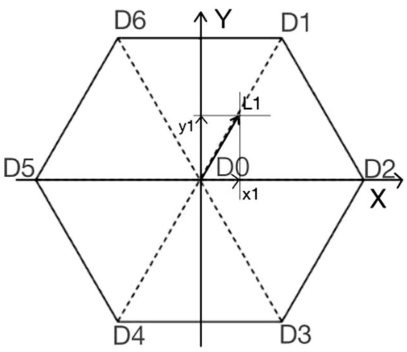

Figure 5. Decomposition diagram of vectors

As in Figure 5, establish a right-angle coordinate system with D0 as the center, and note that the side length of the square hexagon is a. The sound intensity from the six directions will be vectorially decomposed to the x-axis and y-axis, respectively. That is:

$$X(i)= L(i)\cos(i\pi/3) \qquad (i=1,2,3,4,5,6). \tag{1}$$

$$Y(i)= L(i)\sin(i\pi/3) \qquad (i=1,2,3,4,5,6). \tag{2}$$

Synthesized as:

$$X=X(1)+X(2)+X(3)+X(4)+X(5)+X(6). \tag{3}$$

$$Y=Y(1)+Y(2)+Y(3)+Y(4)+Y(5)+Y(6). \tag{4}$$

The synthesized angle equation is:

$$\theta=\arctan(X/Y). \tag{5}$$

The synthetic intensity is:

$$R=(X2+Y2)1/2. \tag{6}$$

Now we have obtained the direction and intensity of the sound in the xoy plane, and then combined with the central microphone D0 to obtain the sound intensity data L(0), so that we can get the angle between the sound direction and the xoy plane:

$$r=\arctan(L(0)/R). \tag{7}$$

Subsequently, the derived angle and intensity data will be used to depict the corresponding area on the screen, and finally corrected step by step according to the actual effect and combined with the camera's viewpoint. This method to obtain the sound source direction algorithm is relatively simple, easy to implement on a microcontroller, and thus save computing resources to facilitate development, but does not accurately calculate the direction of the sound source in three-dimensional space, the need for subsequent correction to achieve practical results, in the conditions of the accuracy requirements are not very high can be applied[4].

4. Program Design and Flow

The development environment of the system is python language, and the program flow is designed by modular program, the program has camera image acquisition, array microphone audio acquisition, algorithm for audio processing, drawing program, LCD display program and so on. At startup, each component is first initialized and set up, as shown in Figure 6 for the program flow.

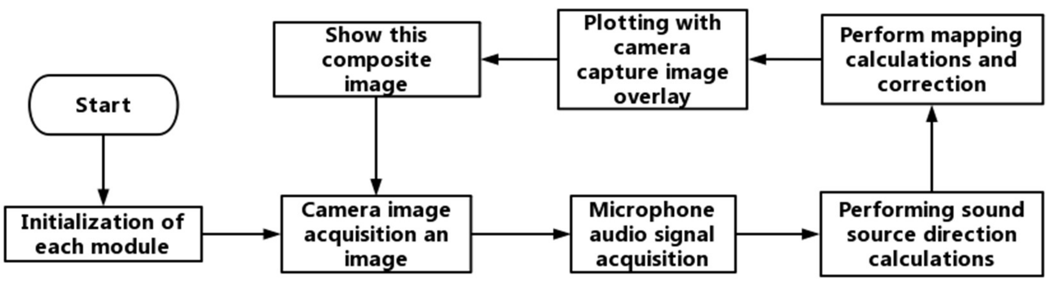

Figure 6. Program Flowchart

Volume 8 Issue 7, 2022
DOI: 10.6919/ICJE.202207_8(7).0079

5. Results and Analysis

As the camera of this acoustic imaging instrument has a certain distance from the center of the array microphone, when it is particularly close to the sound source, about 10 to 20 cm away from the sound source, the position of the sound source depicted on the display has a deviation visible to the naked eye from the actual situation, and the error is relatively small outside 0.5 m, which does not affect the position of the distinguished sound source. In the test with the cell phone as the sound source body, through the sound of different frequencies, the display screen are updated in time to mark the location, to achieve the desired effect, such as Figure 7 for a physical display effect.

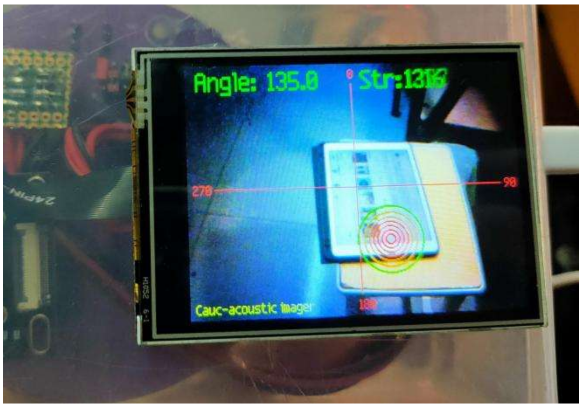

Figure 7. Physical display effect

Acknowledgments

Fund Support: Civil Aviation University of China (CAUC), Project No. 202110059142, University Student Innovation and Entrepreneurship Training Program Project Funding.

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